Lecture Notes in Electrical Engineering 312

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New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering

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ISSN 1876-1100 ISSN 1876-1119 (electronic)
ISBN 978-3-319-06763-6 ISBN 978-3-319-06764-3 (eBook)
DOI 10.1007/978-3-319-06764-3
Springer Cham Heidelberg New York Dordrecht London

Library of Congress Control Number: 2014956106

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Preface

This book includes the proceedings of the International Joint Conferences on Computer, Information, and Systems Sciences, and Engineering (CISSE 2013). The proceedings are a set of rigorously reviewed world-class manuscripts presenting the state of international practice in Innovative Algorithms and Techniques in Automation, Industrial Electronics, and Telecommunications.

CISSE 2013 is a high-caliber research four research conferences that were conducted online. CISSE 2013 received 150 paper submissions and the final program included 105 accepted papers from more than 80 countries, representing the six continents. Each paper received at least three reviews, and authors were required to address review comments prior to presentation and publication.

Conducting CISSE online presented a number of unique advantages, as follows:

- All communications between the authors, reviewers, and conference organizing committee were done online, which permitted a short six week period from the paper submission deadline to the beginning of the conference.
- PowerPoint presentations and final paper manuscripts were available to registrants for three weeks prior to the start of the conference.
- The conference platform allowed live presentations by several presenters from different locations, with the audio and PowerPoint transmitted to attendees throughout the internet, even on dial up connections. Attendees were able to ask both audio and written questions in a chat room format, and presenters could mark up their slides as they deem fit.
- The live audio presentations were also recorded and distributed to participants along with the PowerPoints presentations and paper manuscripts within the conference DVD.

The conference organizers and we are confident that you will find the papers included in this volume interesting and useful. We believe that technology will continue to infuse education thus enriching the educational experience of both students and teachers.

Bridgeport, CT, USA Khaled Elleithy
Bridgeport, CT, USA Tarek Sobh
April 2014

Acknowledgments

The 2013 International Joint Conferences on Computer, Information, and Systems Sciences, and Engineering (CISSE 2013) and the resulting proceedings could not have been organized without the assistance of a large number of individuals. CISSE was founded by Professors Tarek Sobh and Khaled Elleithy in 2005, and they set up mechanisms that put it into action. Andrew Rosca wrote the software that allowed conference management, and interaction between the authors and reviewers online. Mr. Tudor Rosca managed the online conference presentation system and was instrumental in ensuring that the event met the highest professional standards. We also want to acknowledge the roles played by Sarosh Patel and Ms. Susan Kristie, our technical and administrative support team.

We would like to express our thanks to Prof. Toshio Fukuda, Chair of the International Advisory Committee, and the members of Technical Program Committees.

The excellent contributions of the authors made this world-class document possible. Each paper received two to four reviews. The reviewers worked tirelessly under a tight schedule and their important work is gratefully appreciated.

April 2014

Contents

BORM-II and UML as Accessibility Process in Knowledge and Business Modelling 1
Vojtěch Merunka, Robert Pergl, and Jakub Tůma

Planning-Context Aware Mobile Recommendations 7
Chad A. Williams and Sean T. Doherty

Genetic Algorithms-Fuzzy Based Trade-Off Adjustment Between Software Complexity and Deliverability 15
Siddharth Lavania, Manuj Darbari, Neelu J. Ahuja, and Imran Ali Siddqui

Visual Servoing Based Positioning and Object Tracking on Humanoid Robot 19
Michael Bombile

Obtaining Agents and Entities from Natural Language 29
Luis Alfonso Lezcano, Jaime Alberto Guzmán-Luna, and Sebastián Alonso Gómez

A New Methodology Based on Cloud Computing for Efficient Virus Detection 37
Vasileios A. Memos and Kostas E. Psannis

New Generation Android Operating System-Based Mobile Application: RSS/News Reader 49
Taner Arsan, Mehmet Arif Erşahin, and Ebru Alp

Intelligent Emergency Response System for Police Vehicles in India 57
Ishan Ganeshan and Nasrullah Memon

Adaptive Instantaneous Traffic Signal Management Through Cascade Object Detection 63
Md. Sharifur Rahman and Md. Rafiqul Islam

Analysis of the Lifetime of Wireless Sensor Networks for Underground Communications in Wet Sand 71
Abdelrahman Elleithy and Gonhsin Liu

About One Way to Discover Formative Assessment Cheating 83
Jan Genci

Application of a Feed-Forward Control Structure 91
Wei-yu Louis Feng

Reactive Planning to Compose Learning Routes in Uncertain Environments 101
Ingrid-Durley Torres and Jaime Guzmán-Luna

Biodiesel Production in Stirred Tank Chemical Reactors: A Numerical Simulation 109
Alejandro Regalado-Méndez, Rubí Romero Romero, Reyna Natividad Rangel, and Sigurd Skogestad

Evaluation of Optimal Control-Based Deformable Registration Model 117
Naleli Jubert Matjelo, Fred Nicolls, and Neil Muller

The Effect of Mutual Coupling on a Microstrip Printed Antenna Array Operates at 5 GHz Using Three Different Substrates 125
Ali Elrashidi, Islam Ashry, Khaled Elleithy, and Hassan Bajwa

Variable Delay Optical Buffer Using Tunable Fiber Bragg Gratings 133
Islam Ashry, Ali Elrashidi, Dalia Sallam, Moustafa H. Ali, and Khaled Elleithy

An Exploratory Case Study of Offshore Outsourcing: Problems in Multicultural Settings 137
Jussi Koskinen, Veikko Halttunen, and Juha Mansikkaniemi

Comparison of M5' Model Tree with MLR in the Development of Fault Prediction Models Involving Interaction Between Metrics 149
Rinkaj Goyal, Pravin Chandra, and Yogesh Singh

Content Based Image Retrieval System with a Combination of Rough Set and Support Vector Machine 157
Maryam Shahabi Lotfabadi, Mohd Fairuz Shiratuddin, and Kok Wai Wong

Cloud Storage Using Matches Algorithms 165
Zyad Nossire, Julius Dichter, Eman Abdelfattah, and Subrina Thompson

Information Retrieval with the Use of Music Clustering by Directions Algorithm 171
Adam L. Kaczmarek

Using SQL Queries to Evaluate the Design of SQL Databases 179
Erki Eessaar and Janina Voronova

Improving Higher-Order Learning and Critical Thinking Skills Using Virtual and Simulated Science Laboratory Experiments 187
Nicole Simon

Handling a Device Changing from 3G to Wi-Fi Without Breaking Established Connections 193
Richard Sims and Carolin Bauer

Implementation and Analysis of a Morphological Algorithm for the Attributes Opening and Closing 199
D. Bachurin and A. Lakhtin

Document Classification Using Enhanced Grid Based Clustering Algorithm 207
Mohamed Ahmed Rashad, Hesham El-Deeb, and Mohamed Waleed Fakhr

Improving Identity Privacy in 3GPP-WLAN 217
Hiten Choudhury, Basav Roychoudhury, and Dilip Kr. Saikia

Performance of BFSA Collision Resolution: RFID Including Non-unique Tag IDs 225
Kirti Chemburkar, Zornitza Genova Prodanoff, Kenneth Martin, and Susan Vasana

Assessing Software Project Management Complexity: PMCAT Tool 235
Vyron Damasiotis and Panos Fitsilis

Performances of LEON3 IP Core in WiGig Environment on Receiving Side 243
Natasha Tagasovska, Paulina Grnarova, Aristotel Tentov, and Danijela Efnusheva

Clustering-Based Topic Identification of Transcribed Arabic Broadcast News 253
Ahmed Abdelaziz Jafar, Mohamed Waleed Fakhr, and Mohamed Hesham Farouk

Epilepsy Seizure Detection in EEG Signals Using Wavelet Transforms and Neural Networks 261
E. Juárez-Guerra, V. Alarcon-Aquino, and P. Gómez-Gil

A Bargaining Approach for Disseminating Context Information to Context-Aware Services 271
Elarbi Badidi

Automatic Generator of Decoupling Blocks Using Genetic Programming 281
M. Montes Rivera, M. Paz Ramos, and J.L. Orozco Mora

MEMS Dual Axis Accelerometer with H-T Shape Structure 291
Xingguo Xiong, Lidong Qiang, Linfeng Zhang, and Junling Hu

Prior Data Quality Management in Data Mining Process 299
Mamadou S. Camara, Djasrabe Naguingar, and Alassane Bah

Towards a Better Infrastructure Supporting the e-Education as a State Public Service 309
D. Kumlander

Novel Time-Stamped Digital Card (TSDC) to Determine User's Identity 317
Munif Alotaibi, Aziz Alotaibi, Adel Qahmash, and Zhengping Wu

PSPICE Implementation of Block-Wise Shut-Down Technique for 8 $\times$ 8 Bit Low Power Pipelined Booth Multiplier 325
Umatri Pradhananga, Xingguo Xiong, and Linfeng Zhang

Trust Models in Cloud: A Survey on Pros and Cons 335
Usha Divakarla and K. Chandra Sekaran

Face Recognition Security System 343
Michel Owayjan, Amer Dergham, Gerges Haber, Nidal Fakih, Ahmad Hamoush, and Elie Abdo

Evaluation of Sound Perception to Identify Candidate Frequency for Wireless Networking 349
Kuruvilla Mathew, Chong Eng Tan, and Biju Issac

The Efficiency of the Code Parallelization in Multi Core Environment on the Basis of Image Processing in 3D Space 361
Krzysztof Oleszko

Evaluation of the Contract-Aware Software Development Process in a Controlled Experiment 365
A. Derezińska and P. Ołtarzewski

Benchmarking GPenSIM 373
Reggie Davidrajuh

A Field Experiment of System to Provide Tourism Information Using Image Recognition Type AR Technology 381
Hidemi Fukada, Koichi Kasai, and Shou Ohtsu

Metrics for Data Warehouse Quality 389
Bharti Suri and Prerna Singh

A Framework Based on Model Driven Engineering to Support Schema Merging in Database Systems 397
Marcus Vinícius Carvalho, Denivaldo Lopes, and Zair Abdelouahab

Work in Progress: Model Design to Measure the Efficacy of Students Learning Preferences—Does Media Matter? 407
Ashfaque Ahmed Chowdhurya, Aruna Jayasuriya, and Elizabeth A. Eschenbach

Computerizing Exams: The Michigan Tech Testing Center 413
Amber J. Kemppainen, Gretchen L. Hein, and Michael R. Meyer

Gravitation Search Training Algorithm for Asynchronous Distributed Multilayer Perceptron Model 417
Natalya P. Plotnikova, Sergey A. Fedosin, and Valery V. Teslya

Application of Image Processing Techniques to the Identification of Phases in Steel Metallographic Specimens 425
Adarsh Kesireddy and Sara McCaslin

Requirement and Interaction Analysis Using Aspect-Oriented Modeling 431
Sagar Mohite, Rashmi Phalnikar, and S.D. Joshi

Fingerprint Orientation Image Estimation in the Frequency Domain 439
Victor-Valeriu Patriciu, Stelian Spînu, and Cezar Pleşca

Relationship Between Affordance and Cultural Conventions in the Design of IVR Systems for Oral Users 445
Tembalethu J. Ndwe and Nomusa Dlodlo

Analysis and Implementation of Frequency Domain Equalizer for Single Carrier System in the 60 GHz Band 451
Tatjana Chavdarova, Goran Jakimovski, Blagoj Jovanov, Aristotel Tentov, and Maja Malenko

Recent Trends and Developments for Standardization of Video Quality Assessment Metrics 457
Ranjit Singh and Naveen Aggarwal

An Efficient 64-Point IFFT Hardware Module Design 463
Danijela Efnusheva, Aristotel Tentov, and Natasha Tagasovska

The Relationship Between Psychological Distress and Human Computer Interaction Parameters: Linear or Non-linear? 471
Indika Karunaratne, Ajantha S. Atukorale, and Hemamali Perera

Leading Innovation in Universities: From Practice Ahead of Practice 479
Kristina Zgodavova and Matus Horvath

Implementation of Variants in Component Based Software Production 485
Yusuf Altunel and Abdül Halim Zaim

A Methodology for Designing Agent-Based Models: Agent from « UP » for Complex Systems 493
Ahmed Tidjane Cisse, Aboubacar Cisse, Alassane Bah, Jacques A. Ndjone, and Cheikh M.F. Kebe

Towards a Passive Gait: Modeling the Fully Actuated Humanoid NAO 505
P. Bohra and M. Bombile

An Architectural Model Framework to Improve Digital Ecosystems Interoperability 513
S. Shervin Ostadzadeh, Fereidoon Shams, and Kambiz Badie

Towards Multi-gigabit OFDM Throughput on FPGA: Limitations and Challenges 521
Goce Dokoski and Aristotel Tentov

Integral KNIT Cluster as a Source of Development Factors and Technological Frame for the Realization of Country Development Projects 525
Aleksandras Vytautas Rutkauskas and Viktorija Stasytytė

The Impact of Film-based Learning in Science Education 531
M. Hosťovecký and J. Štubňa

An Approach Based on BPMN to Detail Use Cases 537
Adriana Herden, Pedro Porfirio M. Farias, and Adriano Bessa Albuquerque

Design of a SCADA System to Prevent Freezing in Oil Pipes 545
M. Ghassoul, S.R.A. AlMishal, B.A. Ali, and A.A.N. Thabit

Optimizing Next-Generation Mobile Networks Using Frequent Sequential Pattern Mining 551
Zachary W. Lamb and Sherif S. Rashad

A Critical Review on World University Ranking in Terms of Top Four Ranking Systems 559
Farzana Anowar, Mustakim A. Helal, Saida Afroj, Sumaiya Sultana, Farhana Sarker, and Khondaker A. Mamun

Multi-agent Architecture for User Adaptive Information Retrieval Systems 567
Juan J. Pacheco-Reyes, Oleg Starostenko, Vicente Alarcon-Aquino, and Jorge Rodriguez-Asomoza

Case Study: Challenges and Issues in Teaching Fully Online Mechanical Engineering Courses 575
Sara McCaslin and Fredericka Brown

Finding Specifications of While Statements Using Patterns 581
Aditi Barua and Yoonsik Cheon

High-Linearity MEMS Accelerometer with Lateral Comb Finger Groups 589
Xingguo Xiong and Huanyu Li

A Quality-Driven Approach for Ranking Web Services 599
Eyhab Al-Masri

RFID-Based Approach for Monitoring Patient's Health Inside Hospitals 607
Eyhab Al-Masri and Mohamed Hamdi

An Evaluation of N-Gram Correspondence Models for Transliteration Detection 615
Peter Nabende

Design of an Optimized Crane Location System for Construction Sites 623
Soly Mathew Biju, AbdulLatif Tchan, Marina Helmy, and Kirolos Ayad

Route Tracking of Moving Vehicles for Collision Avoidance Using Android Smartphones 629
Ehab Ahmed Ibrahim, Said El Noubi, and Moustafa H. Aly

BORM-II and UML as Accessibility Process in Knowledge and Business Modelling

Vojtěch Merunka, Robert Pergl, and Jakub Tůma

Abstract

This paper presents two systems and knowledge modelling techniques that may be used as a tool to coordinate the communication between researchers and users from the agriculture problem domain. The paper is focused on th eusage of a general approach UML (Unified Modelling Language) and an innovative approach BORM-II (Bussiness Object Relation Modelling, second generation) as communication standards within research projects. The first part of this paper describes the framework, laying out the main aspects of both notations, metamodel and theoretical background as well as their advantages and disadvantages. The paper analyses practical examples from agriculture, rural and organization modelling domains. These innovation processes in both approaches are aplied on the same business process description and evaluates the impact on researchers and users of research. The main part is focused on the transformation model to model based on BORM-II. The transformation is in line with UML and SBVR (Semantics of Business Vocabulary and Rules) standards from OMG (Object Management Group). My predecessor worked on model transformation BORM to UML. This work follows Petr Šplíchals work and goes further. This transformation will be composed into a modelling tool and will be based on approach HOT (High Order Transformation). The objective of this research is to achieve documentation output like SBVR, and bridge the gap between business people (users) and designers (researchers) of information systems (IS). The paper concludes that the gap between IS designers (software engineers) and domain experts can be bridged by automated transformation of previously mentioned models. The main goal is to achieve a documentation output similar to SBVR, and ICT Accessibility for business people.

Keywords

Business process and knowledge modelling • ICT accessibility • Unified modelling language • Business object relation modelling • Model transformation • Tool • Semantics of business vocabulary and rules

Introduction

This paper presents two system and knowledge modelling techniques that may be used as a tool to coordinate the communication between researchers and users from the agriculture problem domain. The paper is focused on the use of a general approach UML (Unified Modelling Language)

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_1, © Springer International Publishing Switzerland 2015

and an innovative approach BORM-II (Business Object Relation Modelling, second generation) as communication standards within research projects.

BORM

Business Object Relationship Modelling [1, 2] is a object-oriented software engineering methodology, which has proven to be very effective in the development of business information and knowledge systems. Its effectiveness is achieved by unified and simple method for presenting all aspects of relevant model. The BORM methodology makes extensive use of business process modelling. BORM was designed as a method covering all phases of software development. BORM focuses mainly on the first phases of the project also known as business analysis. BORM uses only limited, easily comprehensible group of concepts for every life-cycle phase. This makes it easier to understand even for the first-time users with almost no knowledge of business analysis.

Another fact that makes the BORM methodology more expressive is that it does not need the division to static and dynamic views of the model and therefore does not bring a need of creation of different diagrams with a different viewpoints. BORM introduces the following types of diagrams:

- Business architecture diagram.
- Object relationship diagram.
- Class diagram.

BORM represents every concept with the same symbols in the data structure, the communication or other diagrams. For visual presentation of the information BORM uses simple diagrams that contain only a necessary number of concepts and symbols. These concepts and symbols cover most of the needs for the initial description of the model and its processes. The following symbols belong to the symbols used in the initial description:

- Participant an object representing the stakeholder involved in one of the modeled processes, which is recognised during the analysis.
- State sequential changes of the participants in time are described by these states.
- Association data-orientated relation between the participants.
- Activity represents an atomic step of the behaviour of the object recognised during the analysis.
- Communication represents the data flow and dependencies between the activities. Bidirectional data flow may be present during the communication.
- Transition connects the state-activity-state and represents changes of the states through activities.
- Condition expresses constraint that holds for the communication or activity [1].

UML

The Unified Modelling Language (UML) is a standardised notation for specifying object-oriented software systems [3, 4]. A UML model is a set of diagrams describing and documenting the structure, behaviour and usage of a software system. UML is used to model all kinds of software systems, including concurrent and embedded systems. There are commercial modelling tools available on the market to help the designer in creating the UML models and these tools can also generate program code from some diagrams of the model. UML is an expressive and rich language, their models must still be verified, since a model may contain unexpected behaviours from the designer.

Materials and Methods

Goal

The goal of this contribution is to present an approach for flexible modelling of business processes both at the management and operations level. The approach consists of combining a suitable modelling method and developing an original software tool to support it, as well as to perform automated model transformations.

The goal presented another view-point of this research is achieving documentation output like SBVR [5], and bridge the gap between business people (users) and designers (researchers) of information systems (IS) [6].

Methodology

1. First we set requirements for a suitable management-level business process model and operation-level business process model (BPM).
2. We describe how the selected modelling method and the OpenCASE support these requirements.
3. We present a case study to illustrate the results. Requirements for Management-Level and Operation-Level.

BPM For our purpose, let's define the management level is focused on the process orchestration, specifically:

1. terminology,
2. the logic of processes,
3. the relations of processes (transition, decomposition,
4. communications between participants,
5. optimisation of the overall process.

For the management level, the language of the model needs to support the mentioned aspects. This is why usually a combination of graphical and textual language is used.

The management level BPM specifies terms and their relations that are consequently manipulated in different ways:

- They need to be verified for correctness.
- They need to be communicated.
- They are used for reasoning.
- Various reports and statistics need to be calculated.
- They are often changed (they evolve).

This is why the management-level BPM needs to be sort of knowledge base, not just a set of diagrams (graphical objects). By the operational level here, we mean concrete process participants (staff, systems) performing the specified processes.

For this level, we specify the following requirements on the operation model:

- The language of the model is close to the language of the participant.
- The model is accurate.
- The model contains just necessary details to perform the operations.
- The model is up to date and consistent with the management-level model.

As systems participants provide quite a different category (being software and thus computer science and software engineering methods apply here), we will consider just human participants (staff) here. Staff at the operation level are not supposed to be interested in the big picture they just need accurate instructions for performing their tasks, i.e. the management needs to answer their questions:

- What are the steps I should follow to successfully complete a task?
- How should I make decisions and select correct approach?
- What are the inputs that I will get? From whom, how and when?
- What are the outputs that I should produce? To whom shall I handle them, how and when?

For operation levels, usually textual operation manuals are used, as operation-level staff is not supposed to prefer abstract notations.

Results and Discussion

The case study demonstrates the transformation from the management-level business process model into the operation-level business process model. As we specified in requirements, the operation-level model should be textual and tailored for each participant. This is where we utilize the OpenCASE the knowledge base and API and generate HTML page for each participant. HTML documentation output is like SBVR [5]. This is achieved by selecting the Project—Generate Report menu item in OpenCASE.

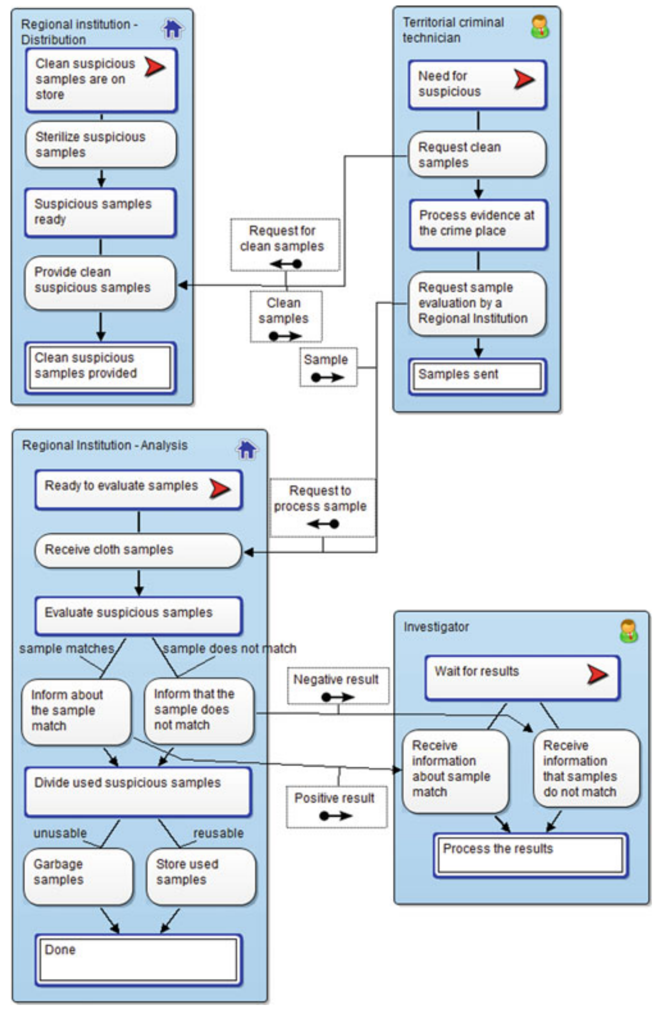

Fig. 1 Case study management process model in BORM: partial phase of MPI

Generation is based on publication [7] and this transformation is composed of a modelling tool and based on approach HOT (High Order Transformation) published.

The case study deals with the process of suspicious identification method (MPI) used in criminalistic. Just a part of the process is presented here due to space limitations. The case study is focused on collecting suspicious samples and their analysis. The process covers a collection of suspicious samples. The case study was written with cooperation of the Faculty of Agrobiology, Food and Natural Resources, namely we would like to thank Ing. Petr Vlasak from Canine Behavior Research Center. Here we will use just a simplified version to demonstrate the concepts presented in the paper. The MPI process is carried out by cooperation between four participants: Regional institution Distribution, Territorial criminal technician, Regional institution Analysis and Inspector. The whole case study is shown in Fig. 1 and in detail (Figs. 2, 3, 4, and 5).

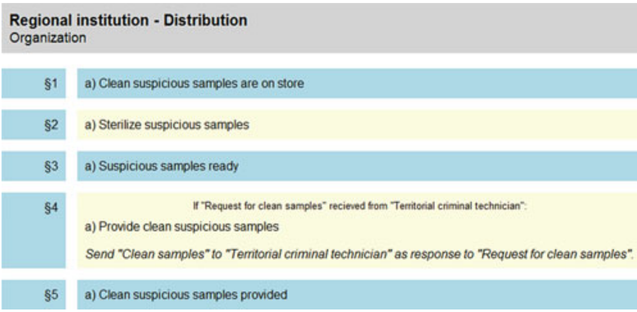

Fig. 2 Case study management process model in BORM: partial phase of MPI, detail of distribution

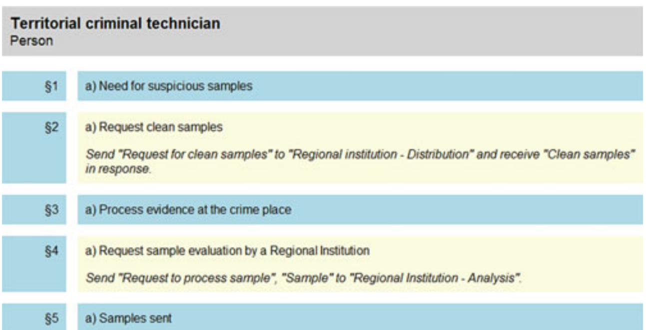

Fig. 3 Case study management process model in BORM: Partial Phase of MPI, detail of analysis

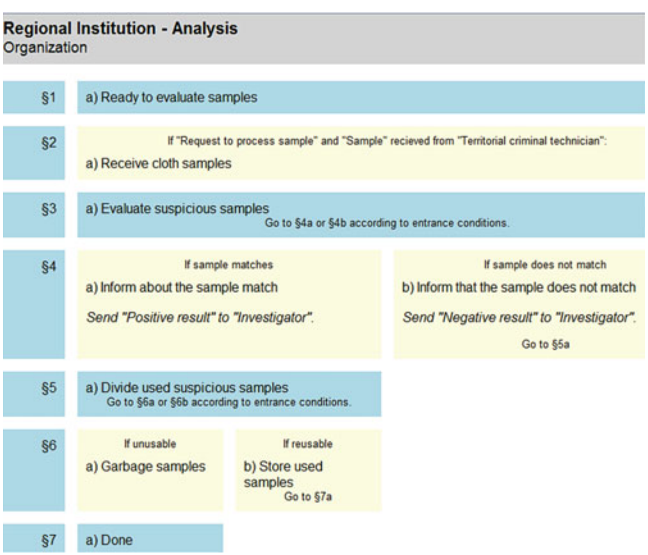

Fig. 4 Case study management process model in BORM: Partial Phase of MPI, detail of technician

The exporter then traverses through the inner diagram structure. The HTML operation manuals are generated for each participant (Figs. 6, 7, 8, and 9).

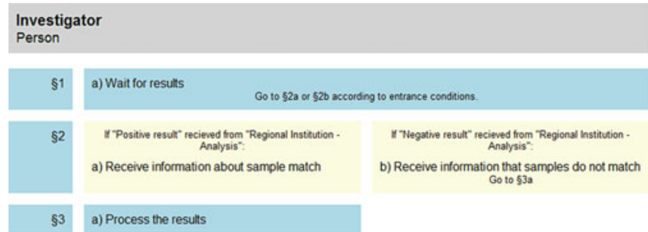

Fig. 5 Case study management process model in BORM: partial phase of MPI, detail of investigator

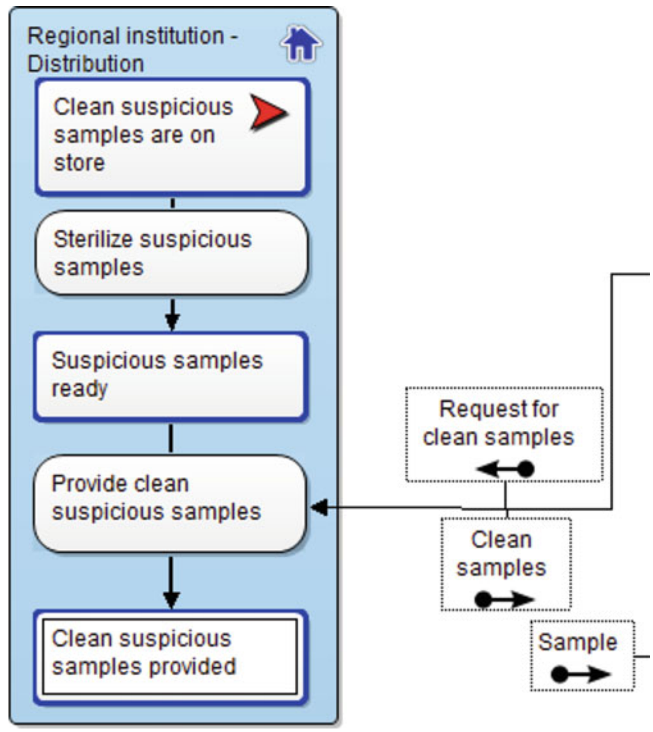

Fig. 6 Operation model (manual) for participant regional institution distribution

Conclusion

In this paper we presented our solution that supports business process engineering. It is a combination of a suitable method and notation (BORM) supported by a software tool (OpenCASE). The key aspect of the solution is that the modelled process is not just a diagram, but a whole knowledge base that may be used in operations, reporting, decision making and other areas. We presented one of its possibilities: automatic generation of operations manuals. Other possibilities include:

- Listings, like all input/output flows from/to a participant.
- Calculation of metrics (like numbers of states and activities in participants) that may be used for complexity estimations [8, 9].

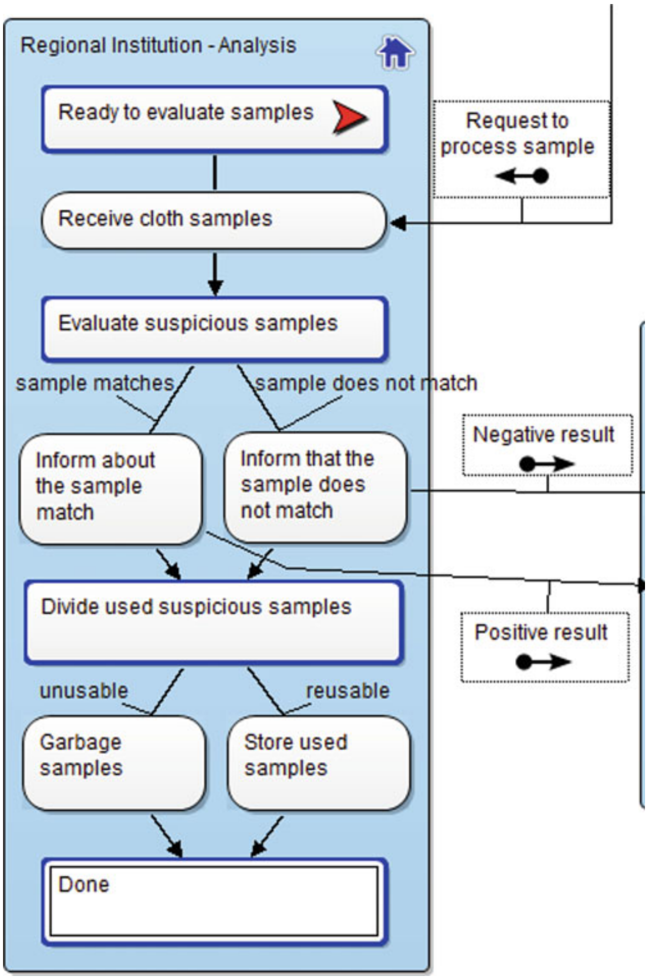

Fig. 7 Operation model (manual) for participant territorial criminal technician

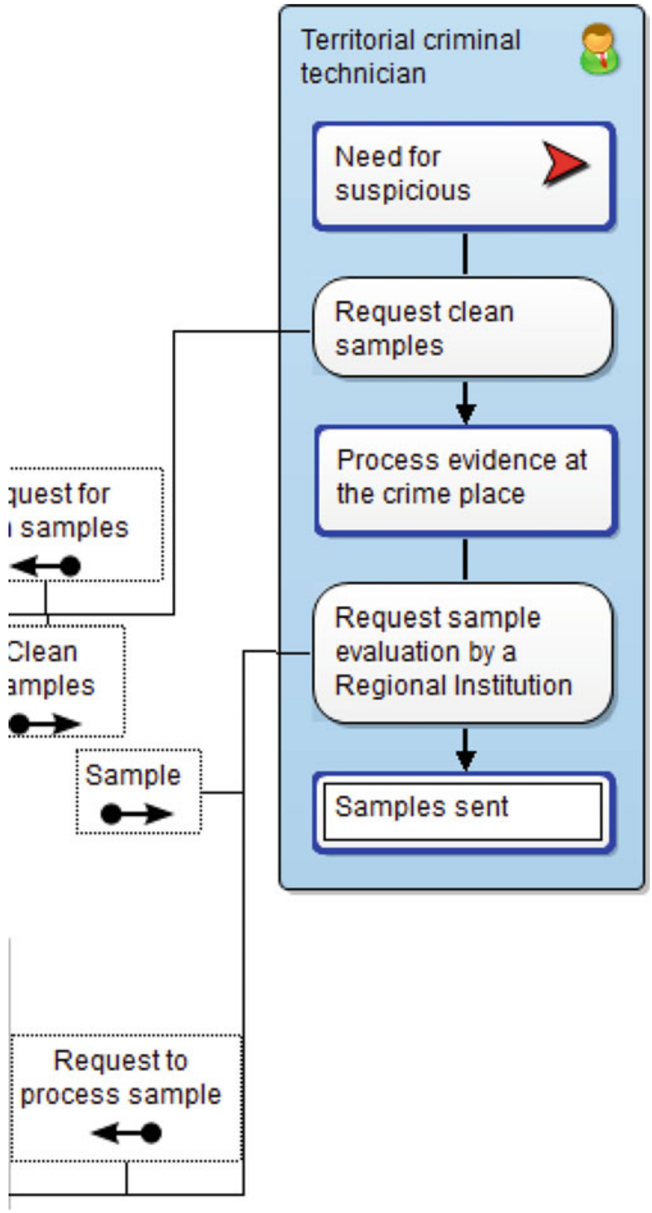

Fig. 8 Operation model (manual) for participant regional institution analysis

- Calculation of statistics, e.g. about data flows and communications (which participants communicate the most/least, above/below average, etc.).
- Semantics checks: there is a starting state in every participant, at least one final state.
- Conceptual normalization [10].
- Any further custom reporting/calculations/processing.

These areas provide many topics both for practice and research. The OpenCASE is implemented using open architecture based on Eclipse plugins, which makes it easily extensible and thus provides a platform for further studies.

Apart from this, it is already a stable tool for effective drawing of BORM and their management. Future work statistical research of developed methods in practice is planned.

Acknowledgment Many thanks to my supervisor Vojtěch Merunka and to the team group leader Robert Pergl for a lot hints and mental support. This paper documents summarise previous and future developing process of dissertation thesis. This article was realized with support of internal grant agency ČZU v Praze, IGA project number 20121059, Metody automatizovaných Transformací modelů v informačních systémech. I would like to thank proof readers Sarah Marlena, Angela Bilbilovska and Juraj Kardoš.

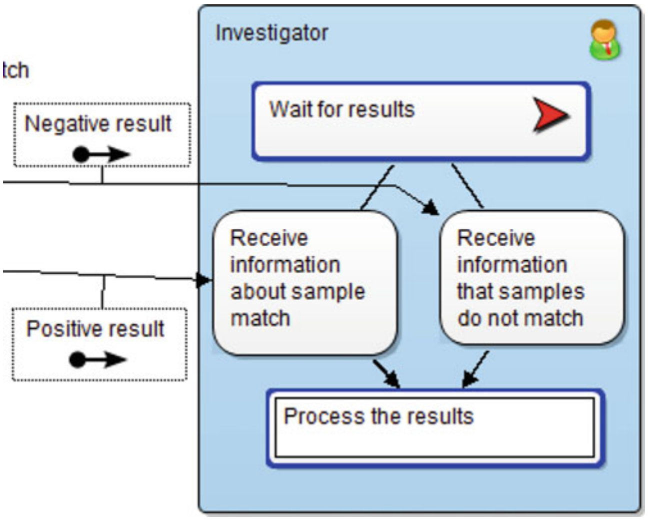

Fig. 9 Operation model (manual) for participant investigator

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Planning-Context Aware Mobile Recommendations

Chad A. Williams and Sean T. Doherty

Abstract
In the realm of mobile applications a significant effort has been made to develop recommender systems that customize results based off of one's current location and more recently even their inferred current activity. While this aspect of context has been shown to be quite successful, we suggest anticipating what they are currently planning for the future may help further improve the relevancy of the results as well. This work examines this problem as one of trying to predict the user's planning context, defined as what activities are currently being planned and how far in the future the event they are planning is going to be. An empirical analysis is made of the predictability of planning context and a discussion of the potential implications of this for mobile context aware recommenders.

Keywords
Mobile applications • Planning context • Locational context • Empirical analysis • Predictability

Introduction

Context aware recommender systems (CARS) have received significant focus in recent years as a way of increasing the relevance of results compared to traditional recommendation techniques. In this pursuit many different aspects of context have been examined regarding their ability to better understand what is of interest to the user at that particular point in time. As has been shown in previous work in the context of web recommendation, this type of insight can lead to superior predictions over basing recommendations on a general profile of the user alone. Within this work we study the benefits a similar approach might provide in the realm of mobile recommendations.

One of the key differences between a traditional web site experience and a mobile application experience is the availability of additional information about the user beyond just their profile and/or click stream. As a result users typically have an increased expectation that mobile applications are more tailored based on their context. As such, with mobile applications passive observation such as location history take a much more central role in understanding the context of what is relevant to a person.

Identifying mobile user context has been a focus of the ubiquitous computing community for several years. While numerous studies have focused on current context awareness, using that context for recommendation and prediction of future context has received far less attention. Another aspect of this is that while predicting the next location has been well studied, other aspects of context may be just as relevant to a user. Some of these aspects have received little attention such as understanding when an activity and associated trip are planned.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_2, © Springer International Publishing Switzerland 2015

We propose recognizing the context of a user in terms of their planning perspective to be a critical aspect in determining what is most relevant in mobile recommender system. For example, consider a user who is at their local coffee shop in the morning. Based on current models of context that focus on the immediate surroundings and/or time alone, the system might filter content to focus on what is nearby or related to their current activity. Consider, however, if the system knew the user was making plans in the near future of what they were going to be doing later that evening. In such a scenario, what the user considers to be most relevant extends beyond the immediate situation and also includes content that aids in making those plans.

This research focuses on including the planning behavior of mobile users as part of their recommender context. One of the key challenges addressed in this study is much of the data relevant to these aspects of context must be passively collected since user's typically do not explicitly identify this type of information. This work addresses how commonalities in planning behavior can be used to enhance what context is relevant to a mobile user outside of their immediate context despite limited information besides that that can be derived through passive data sources. The experiments conducted below attempt to identify both when plans are being made and the activity being planned. This work appears to be the first to address the integration of these two goals. This is followed by a discussion of the implications of the findings and directions for future work in this area.

Mobile User Contextual Factors

Transportation planners have studied travel behavior extensively over the years. More recently, focus has shifted from looking at travel alone to understanding why a trip was made and when the decision to make the trip occurred. One of the approaches used for this has been examining the activity needs/desires of the person as the reason the travel is made [1–4]. These studies have shown that at an abstract generic level a person's context can help determine their future activity and planning behavior. This study, however, examines modeling a personalized context aware approach rather than using a generalized traveler prediction model.

Determining what is relevant in these terms requires an examination of several different types of contextual factors.

Below we describe these utilizing the terminology framework established in Adomavicius et al. [5].

With mobile devices, two types of dynamic fully observable data from the crux of determining the relevant context. Specifically the two categories are passive and active collection. As the architecture in Fig. 1, adapted from Williams

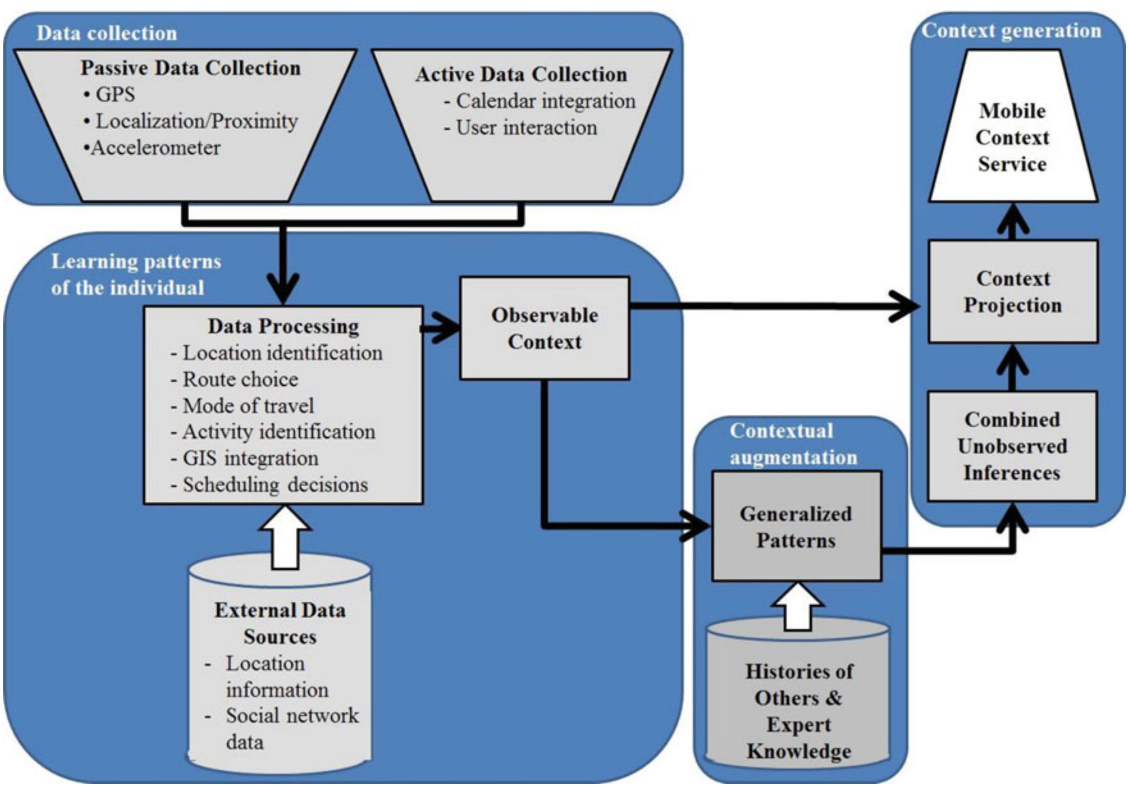

Fig. 1 Architecture of mobile context generation

and Mathew, shows, these two sources of data serve as the input to generating meaningful mobile user context.

Passive collection methods such as Global Positioning System (GPS) and accelerometer tracking are perhaps the most critical methods in mobile applications, because they provide location and movement context without requiring any active data entry by the user. Beyond the obvious location and route observations that can be collected, this data can often be processed to reveal partial context knowledge as well. For example by analyzing movement rate, stop frequency, and the accelerometer data, often the mode of travel such as walking, biking, train, bus, or car can be reliably inferred [6]. Combining location stop information with GIS map matching can sometimes be used to infer trip purpose [7]. Other sensors such as the device's microphones could be used to record ambient noise (e.g. voices, wind, and music) or determine whether the user is typing or talking providing additional environmental or use context. In theory, having tracked GPS data from all people would also allow passengers and social context to be inferred through examination of proximity and duration of contact [8]. Further processing of accelerometer data, especially from multiple body parts, can also be used to infer many specific human behaviors/contexts, such as sitting, standing, and specific exercise types [9–11].

In contrast, active collection methods involve manual data entry by the user. An example of active collection would be integration with calendar functionality, where calendar events that were entered manually could be used to ascertain event plans. There are obvious advantages to data entered explicitly by the user. One of these would be the potential for more accurate information, such as the specific activity that is actually taking place compared to just having the likelihood of what activity is occurring. Other advantages include the ability to collect information that simply cannot be captured completely through passive means such as the reasoning behind the planning decisions that are made. However, there are also significant limitations to active collection. Most notably any manual data entry puts a burden on the user, which in practice means users are less likely to provide the entry without incentive, particularly on an ongoing basis. However, one-time entry such as registration is often acceptable.

As noted in previous studies of transportation behavior, a demographic profile of a traveler can provide some significant insight in the types of activity patterns observed [12]. While some of this information might be explicitly given as part of a profile, such as age, overall it is partially observable as many demographic factors must be inferred to categorize the individual. Other types of information in this category such as work, school and home location, while not explicitly given, can often be reliably inferred through repeated past patterns.

Finally, within this work, what we are particularly interested in, and the goal of this study, is the largely unobservable dynamic context related to activity and travel planning. Towards this goal, this work examines the "Contextual augmentation" component of the architecture shown in Fig. 1, with the aim to output unobserved inferences of what the user is currently planning. We propose being able to predict when different aspects of plans are made is critical information to what is relevant to a user. As a result, recognizing that a plan is being made for a specific activity at a specific time window in the future may provide context that results in a significant improvement of determining relevant recommendations and thus improving the overall mobile user's experience.

Planning Context

"Planning horizon" is a term used to describe the length of time before a trip or activity took place for which its plans were made [13]. For example if it was 10 AM and John decided to go out with friends at 6 PM that evening, the planning horizon for that activity would be 8 h. Planning context can range from far in advance, such as a visit to the doctor planned several weeks ahead, to spur of the moment, such as an urgent stop to get gas. For many activities, the planning may have become rather routine over time, such as a child regularly going to school in the morning, or regularly going to the same coffee shop every morning, wherein little active planning or thought was made. A typical person's day has a mix of routine and actively planned activities with a more finite planning horizon.

The focus of this work is on scheduling behavior and time horizons. Studies have confirmed that most people do not make their plans for all activities at a single point in time [14]. Instead, plans are continuously made and finalized throughout the day for varying planning horizons [15]. For instance, a person does not go throughout the day continually making plans two hours in advance of each activity, nor would they plan everything completely impulsively. In reality, scheduling is more fluid where a person might be making reservations for dinner the next morning, followed by an impulsive decision to grab a snack, followed by making plans for meeting a friend for lunch in an hour. The main point being it is not as simple as the person is just considering the next activity or looking any fixed period in the future.

We propose that determining what is relevant to a user at a given moment in time is in part dictated by their planning context. This planning context is a combination of the type of activity they are looking to plan and the planning horizon dictating when that activity will be carried out. The combination of these factors plays a critical role in what is relevant at a given point in time. For example, if the application knows when John is interacting with the system at 10 AM

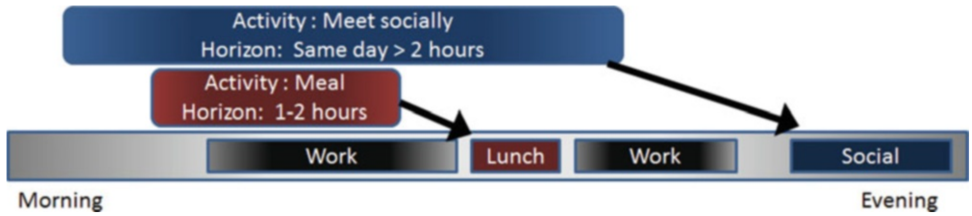

Fig. 2 Planning context throughout day

that he is looking to make plans for that evening not just considering what to do next; that may greatly help in recommending the most relevant information.

If we were to envision what is relevant to a user from a planning perspective it is likely to fluctuate throughout the day. Take for example the planning context John wants to make plans to get together socially with friends in the evening at least 2 h before the event. John also wants to make plans to meet a family member for lunch an hour or two before the event. From a modeling perspective, we can potentially have more than one relevant planning context at a single time as shown in Fig. 2. In a portion of the morning both contexts would be relevant, but as lunch approaches the planning context for the meal would no longer be relevant. As this example illustrates when we are considering what context(s) is relevant it may not be a single answer.

Methodology

The purpose of this work is to assess how well a person's planning context can be predicted at a point in time during a person's day, given information that could be obtained either through an initial profile or by passive sources. Data for this paper is derived from a Computerized Household Activity Scheduling Elicitor (CHASE) survey conducted in Toronto in 2002–2003 [16]. The CHASE survey captured a detailed accounting of the activity scheduling process of 271 households over a 1-week period. The data was recorded such that for each activity that took place, both observed attributes (start time, end time, location, involved persons, category of the activity) and a time frame for when it was planned were all captured, via a scheduling program completed regularly throughout the week. The activities were broken down into 11 categories: active recreation; drop-off/pick-up; entertainment; household obligations; meals; night sleep, other needs; other; services; shopping; social; and work/school. The planning time frame included choices such as routine, *X* number of days ago, more than 2 h before, 1–2 h before, less than 1 h before, and just prior.

To model the planning context for a participant the activities recorded and their planning data were used to construct a time line for each day. The day was discretized into time segments early morning [12 AM, 9 AM), late morning [9 AM, 11 AM), midday [11 AM, 1 PM), early afternoon [1 PM, 3 PM), late afternoon [3 PM, 5 PM), early evening [5 PM, 7 PM), and late evening [7 PM, 12 AM). For each of these time segments a series of possible planning context entries were created that consisted of the combination of a specific activity type and one of three planning horizons (1–2 h prior, more than 2 h prior, or 1–2 days before) for a total of 33 possible planning contexts for each time segment.

To identify which planning context(s) were active, each of the planning context entries was defaulted as being not active. Next, each activity entry was then iterated through and based on the planning time frame the appropriate planning context time segments would be marked as active. For example if a 'Shopping' activity took place that was noted as having been planned '1–2 h before' the planning context would be noted as active for the period(s) that corresponded between 60–120 min before the *start* of the activity. If the planning time frame was noted as being 'more than 2 h before' the planning context was marked as active from 121 min prior through to the beginning of the day. For the '1–2 days before', no time on the activity's actual day was marked but the two full previous days would have the planning context ['Entertainment', '1–2 days before'] active for example. The result of this effort was a planning context schedule that indicates at a given time segment, what planning context(s) were active for the participant.

The goal was then to build a classifier that given a time of day, day of week, information that could likely be obtained through passive means, and a user profile; it could predict which planning context(s) were active. For the information that could be obtained through passive means, we are approximating that to be the current high-level activity type. While in current practice this cannot always be determined exactly, recent research advancements discussed above continue to make what can be inferred more reliable. In addition to these, a selection of personal data was selected based on its information gain: age, employment status, and gender.

The training data was then created by creating a record for each activity entry with the participant's data and for each activity type and planning horizon whether that planning context was active during that activity. Based on this training data a C4.5 classifier was built.

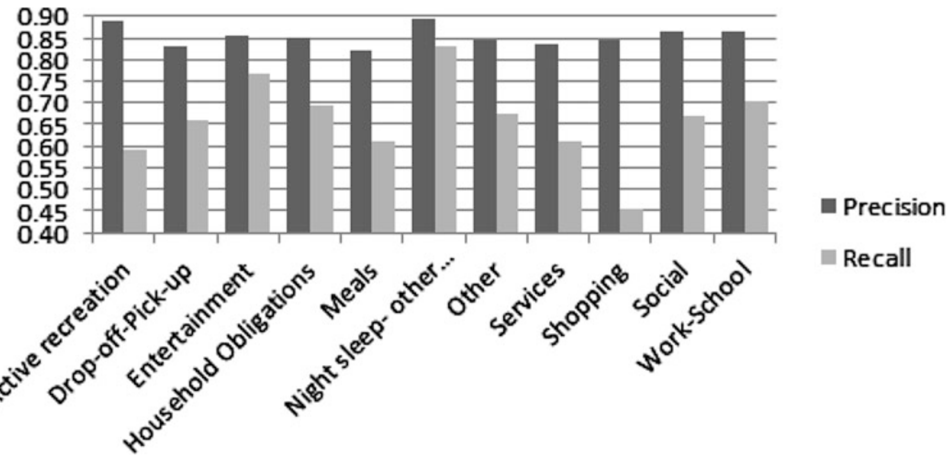

Fig. 3 Precision and recall for 1–2 h planning horizon

Evaluation Metrics

For measuring prediction performance, we use the information retrieval metrics of precision and recall [17]. The basic definition of recall and precision can be written as:

$$\text{precision} = \frac{\#\ \text{true positives}}{(\#\ \text{true positives} + \#\ \text{false positives})}$$

$$\text{recall} = \frac{\#\ \text{true positives}}{\#\ \text{true positives} + \#\ \text{false negatives}}$$

For the purpose of this study, # *true positives* is the number of times a specific activity and planning horizon (planning context) is predicted correctly; # *false positives* is the number of planning context values incorrectly predicted in the time step, and # *false negatives* is the number of planning context values predicted as not in context that were in fact in context.

		Actual class	
		Planning	Not planning
Predicted class	Planning	3290	555
	Not planning	999	14561
	1-2 hours before - Entertainment		

Fig. 4 Confusion matrix for 1–2 h planning horizon for activities entertainment related

		Actual class	
		Planning	Not planning
Predicted class	Planning	1473	235
	Not planning	725	16972
	1-2 hours before - Social		

Fig. 5 Confusion matrix for 1–2 h planning horizon for activities social related

Experimental Results

For our experiments, we examine a comparison of precision and recall for the variety of different activity types captured in CHASE across the various planning horizons. All results reported are based on a tenfold cross validation methodology. Figure 3 captures how well the prediction model performs at identifying, for a specific time window, which activity is currently being planned that will take place in 1–2 h. In a practical sense, this is essentially taking the profile of the individual, their current activity, and given a time of day inferring that in the current hour they are making plans to go shopping an hour or 2 in the future. This information could potentially be very useful in adapting recommendations for what is currently of interest to a user. As the precision results in Fig. 3 demonstrate, the C4.5 classifier performs reasonably well with results ranging from on the low end .82 for "meals" to .895 for "night sleep—other needs." From a context recommendation perspective, this indicates that if a specific planning context is predicted as being relevant there is a high degree of likelihood that it is correct. From the perspective of tailoring recommendations this particularly important because it means there is a low percentage chance that the system would be making use of planning context that is irrelevant. The recall, on the other hand, varies significantly more from .452 for "shopping" to .831 for "night sleep—other needs." From the perspective of recommendation, this would mean that the system did not recognize that some activity was being planned, and thus the missed planning context would not be utilized in identifying the most relevant information.

A more detailed analysis of the activity types is illustrated in the confusion matrices. For "entertainment" depicted in Fig. 4, the results show that during 22 % of the time windows examined "entertainment" activities were being planned for 1–2 h in the future. This activity type was the second most observed for this planning horizon. A similar look at "social" activity planning context, which was present during 11 % of the time windows, is displayed in Fig. 5. Both of these planning contexts show similar precision, but recall is 10 % lower for "social" activities.

Figures 6 and 7 show the confusion matrices for the planning contexts of "services" and "shopping" 1–2 h in the future. As these results show, while the precision remains high, recall drops considerably for these contexts. In terms of the reasons why the planning context for these two activity types was more difficult to predict, may be due to the nature of these activities as compared to "entertainment" and "social" activity types. "Entertainment" and "social" activities tend to be more discretionary, whereas "shopping" and "services" are more task oriented. Thus, while additional study is needed, this may indicate identifying when optional

		Actual class	
		Planning	Not planning
Predicted class	Planning	923	180
	Not planning	583	17719
	1-2 hours before - Services		

Fig. 6 Confusion matrix for 1–2 h planning horizon for activities services related

		Actual class	
		Planning	Not planning
Predicted class	Planning	934	169
	Not planning	1132	17170
	1-2 hours before - Shopping		

Fig. 7 Confusion matrix for 1–2 h planning horizon for activities shopping related

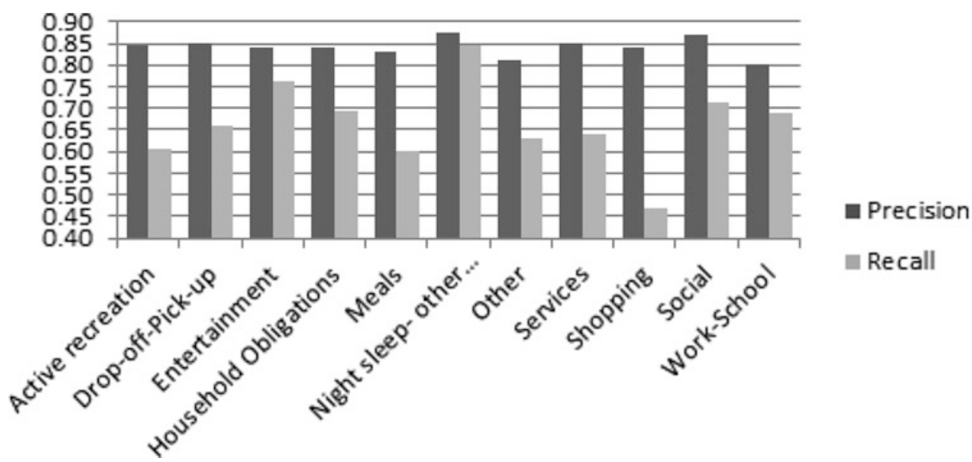

Fig. 8 Precision and recall for same day greater than 2 h planning horizon

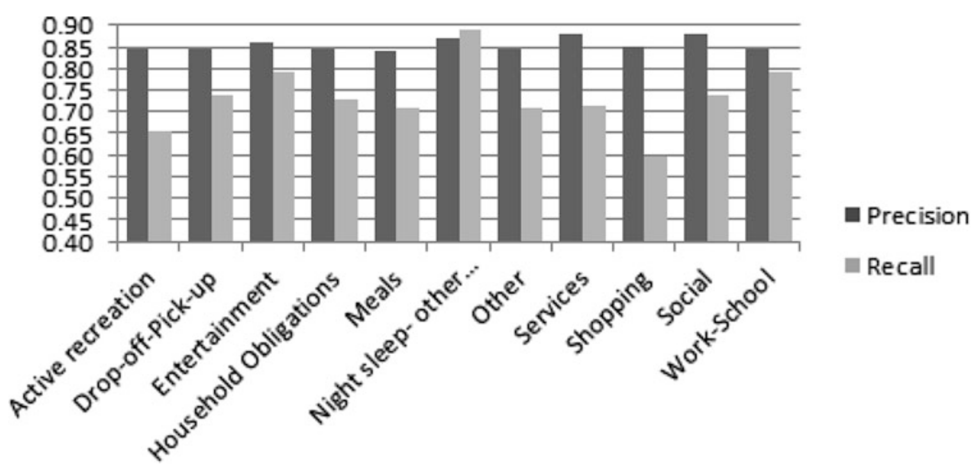

Fig. 9 Precision and recall for 1–2 day planning horizon

activities are being planned may prove to be an easier task than identifying when plans are made for activities related to maintaining the household.

Figures 8 and 9 depict the precision and recall for the activity types for the same day greater than 2 h and 1–2 days planning horizons respectively. The average precision results across the activity types for these two additional planning horizons is nearly identical to that of the 1–2 h planning horizon discussed above, approximately 85 %. For the recall results, the average across activity types for the same day greater than 2 h planning horizon (66.5 %) remains similar to that of the 1–2 h planning horizon (66 %). However, the average recall for the 1–2 day planning horizon rose to 73.2 %. Together these results indicate that, while there is room for improvement, there is a significant opportunity to take advantage of knowing what a person is currently planning. Furthermore since if a prediction is made it is likely correct, this increases the likelihood that if recommendations are made based on this predicted information there will be a have a lesser chance of decreasing the quality of recommendations.

Related Work

Several works have suggested other factors besides location of a mobile user that may help in determining what is relevant to users, but the vast majority has focused on context related to the immediate next activity. Some works have suggested that in addition to current location, current context refers to data such as time availability, real-time weather or traffic updates and real-time events such as accidents [18–20]. In addition, factors like financial situation and the group of people traveling along with the mobile user have been shown to affect the individual's real-time travel decision-making [21]. Other works have demonstrated that the makeup of the group involved influences travel decisions resulting from different groups of people having different preferences depending on their backgrounds [22, 23]. Other studies have examined what aspects of context and planning horizon affect the selection of destination/activity for tourists [24, 25].

We see these works as complimentary. Once the type of activity and timing are identified as part of context, as addressed in this work, that enhanced context can be combined with the findings of these related works to better determine how that information should be used to filter the relevance of the results.

Conclusions

This paper presents a novel approach to enhancing user context. While the majority of prior work in context-based recommendation for mobile applications has focused on the user's immediate situation, this study introduces the idea of extending the user's context to include what plans are being made beyond just the next location as an aspect of the user's context. An empirical analysis is conducted of how well predictions can be made of what activity type and how far in the future a user is currently planning with positive results. As this work demonstrates, this can be done with high precision potentially offering a significant opportunity to improve recommendations to mobile users through incorporating the aspect of what is being planning to context-based recommenders.

The findings of this study bring up many additional areas for future study. While using locational context on mobile applications can sometimes be intuitive such as tailoring results to what is nearby; how best to take advantage of a user's future activity plans that are not necessarily nearby is far less understood. Using mobile planning context to determine what is most relevant poses some very interesting challenges as the most relevant information is likely not just related to the activity being planned, but also must take into account location constraints of the user's upcoming schedule. Another area for future study is the adjustments in schedule a person makes throughout the day. For instance adding activities to a person's schedule can often impact other activities such as changing their timing or even the location. Recognizing when these points occur and identifying what information may be useful in adjusting existing activity plans may be beneficial as well.

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Genetic Algorithms-Fuzzy Based Trade-Off Adjustment Between Software Complexity and Deliverability

Siddharth Lavania, Manuj Darbari, Neelu J. Ahuja, and Imran Ali Siddqui

Abstract

The paper highlights the issue of the Trade-off between Software Deliverability and Complexity. It uses the technique of Messy Algorithm to generate chromosomes containing Fuzzy rules derived under various combinations. The proposed system makes the combination of Complexity and Deliverability more transparent and interoperable.

Keywords

Genetic Algorithms • Software Complexity • Software Deliverability • Fuzzy rules

Introduction

Ever since the software development came into existence, there has been a rift between the software developers and users. Software developers in order to ease their development cycle, try to embed multiple features in a single module making the usability of the module tougher.

It is the usability feature which plays a major role in having the product more sellable, but at the same time it should cater to all high level needs of the consumer. Although, there are number of companies developing certain guidelines about software development process, the major focus is on user-centered application development, software is evaluated with various tools like Cognitive tools and Complexity Matrices to find out the degree of acceptability amongst the users.

The Deliverability of the software in most of the cases is contextual i.e. Analytical or Empirical Methods. The Analytical method depends upon potential interaction with the system and finding out the flaw in the system. Secondly, Empirical evaluation method which is based on the actual usage data.

The main problem arises to maintain a balance between Complexity and Deliverability, as both the quantities are very much inter-related, it is very difficult to raise the Deliverability without increasing the Complexity of the system. The basic common criteria for deliverability are:

1. Ease of use
2. Task Support
3. Navigation
4. Help and
5. Scalability with disturbing the ease of use.

In general, complexity disturbs the ecological aspect of the messages in module. A complex information module can be represented in three dimensional formats as suggested in Albers [4, 6]: Knowledge Level, Detail Level and Cognitive Abilities. In order to increase all the three levels we have to compromise with the Usability Aspect form HCI preview.

The deliverable aspects nowadays focuses on Human Centered application where customer's involvement plays a major role in design phase, but the customer always tells the requirement in the form of stories which looks much simpler during requirement gathering stage but when implemented on real scale. The complexity of the software increases considerably.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_3, © Springer International Publishing Switzerland 2015

Paper [5] describes effective balancing strategy to manage the Complexity and Deliverability of the software with the following key areas:

- Representing the operation dialog in simple language.
- Clearly marked exists.
- Accelerators for Expert Users.
- Good error messages: The messages should be in the simple plain language.
- System should automatically prevent errors.

In order to balance the Trade-Off between complexity and Deliverability, we use Evolutionary Fuzzy Rule Generation using Messy Genetic Algorithm. Multi-Objective Evolutionary Algorithm is well known technique in finding out optimum solution in case of multiple goals. A single objective optimization problem may have a unique optimal solution while Multi-objective generates multiple solutions produces the vectors representing the value of Trade-Off.

Messy Genetic Algorithm

The focus of Messy Genetics is to arrange the chromosomes in tuples of values and these tuples will represent entire set of fuzzy rule (Fig. 1).

Messy Algorithm has some unique features like encoding of gene multiple times, it is encoded as a tuple and gene may be permutated in any way.

Messy Algorithm is a part of Pittsburg Approach. Pittsburg considers each chromosome as a complete set of rule and fuzzy inference represents a single individual.

Fuzzy Rule Based System

Fuzzy Rules Contains linguistic values [1, 2] which are supported by their intensity using IF-THEN-ELSE condition with other linguistic variable. Fuzzy Rule implication can be two kinds of logic inferences: modus ponens and modus tollens. A simple statement like: "If Complexity of Software is HIGH, then Deliverability is LOW". Since "Unix operating system is complex" according to modus ponens we can infer that "Unix Deliverability is LOW" while according to modus tollens "If Unix operating system is NOT Complex" we can infer that "Unix is Highly Deliverable". FRBS helps in generating a fuzzy model which consists of mapping functionality between set of input variables and set of output variables.

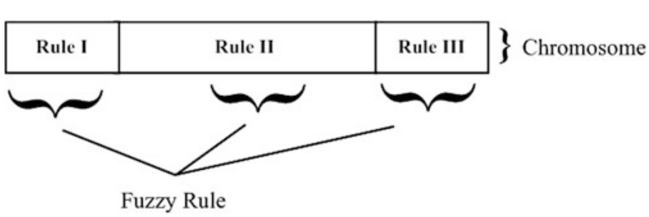

Fig. 1 Chromosomes embedded with fuzzy rules

In simpler terms we can write the Fuzzy Rule as (Fig. 2):

Complex Input problems are simplified in terms of linguistic variables to generate Fuzzy Rule Based. These Rules are matched with GA. These rules iterated and finally refined to eliminate GA generator encodes one complete set of duplicacy using Post-Processing stage. Each chromosome generated from Fuzzy Rule.

Implementation of the Above Framework to Find the Trade-Off Between Complexity and Deliverability

Consider FRBS for a situation showing various relationship conditions [3, 7–9]

Rule I: If Complexity is Moderate, Usability is also Moderate, Software is Popular.

Rule II: If Complexity is Very High, Usability is Poor, Software is Not Popular.

Rule III: If Complexity is Low, Usability is High, Software is Very Popular (Fig. 3).

Now converting the above rule in corresponding chromosomes for one particular situation we get:

```
{
At t = 0; / Counter initialized
Initialize  Population  ( );
Evaluate  Population ( );
while ( NOT termination condition satisfied ) do
{
Apply Crossover between Complexity Level, Usability Level;
Apply Mutation between Combinations generated;
Evaluate the new  population ( );
Alternate  Generation ( );
t + +; / * increase   the generate counter*/
}
```

Applying the algorithm for Genetic Mutation:

$$f = \begin{cases} 1\,;\ \text{if Deliverability} < 0 \\ 0.5\,;\ \text{if Deliverability} > 0 \\ \dfrac{\text{High} - x}{\text{High-Low}}\,;\ \text{if Low} \le x \le \text{High}\,;\ \text{Complexity} > 0, \text{Popularity} > 0 \\ 2 - \dfrac{\text{Complexity}}{\text{Complexity x Popularity}}\,; \quad \text{Popularity} > 0 \\ 0\,;\ \text{if Complexity} > \text{Deliverability} \end{cases}$$

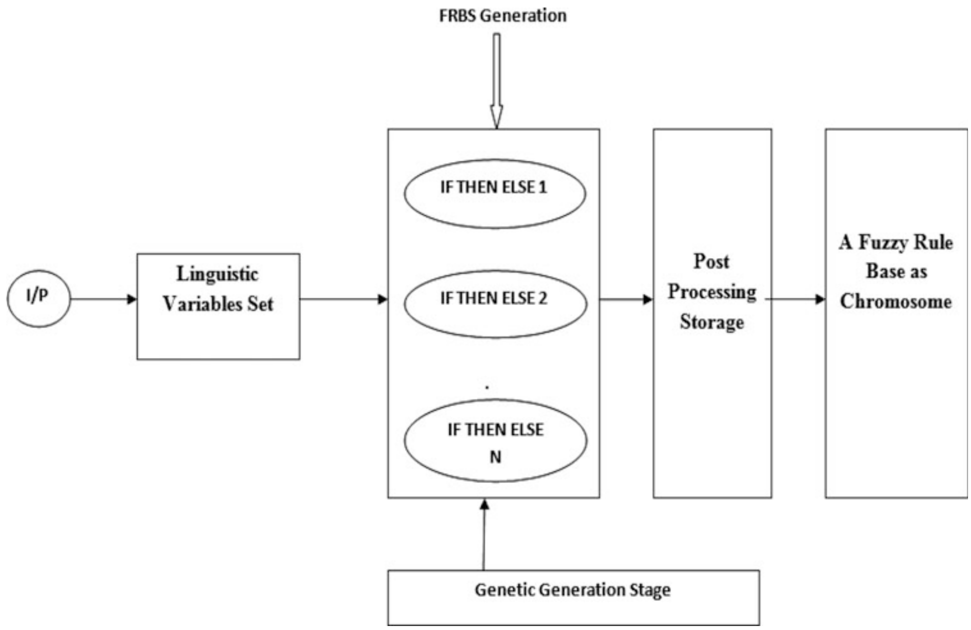

Fig. 2 Fuzzy-GA Conversion framework

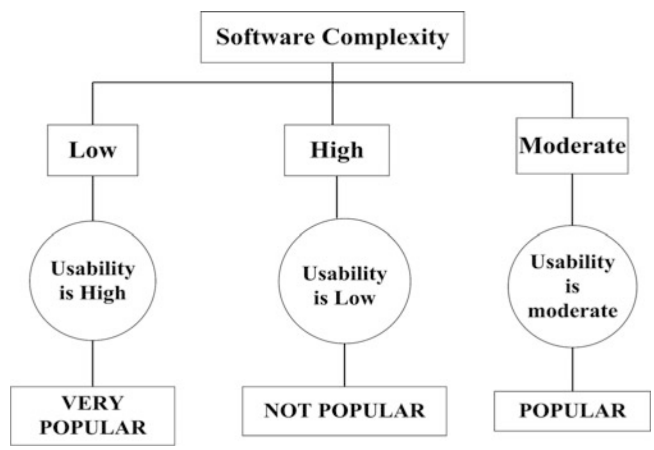

Fig. 3 Triangular membership of rule base

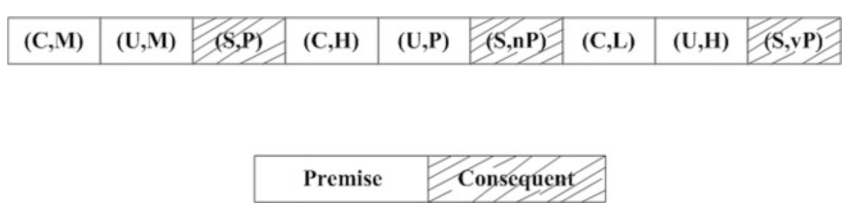

Fig. 4 Chromosomes and corresponding rules

Each generation was mutated by selecting 20 % of the Parent Population, and then these selected individuals were again mutated with a probability of 0.15.

For the above situation, we get the fitness function as a component of Complexity, Deliverability and Popularity:-

We finally get the Plot as shown in Fig. 4 which as the popularity of the software (i.e. its Deliverability features) increases steadily and plateau at complexity.

Simulating the above situation with simulating set to 20 fuzzy rules and real pair of chromosomes generated to be 60, we get the fitness curve as shown in Fig. 5.

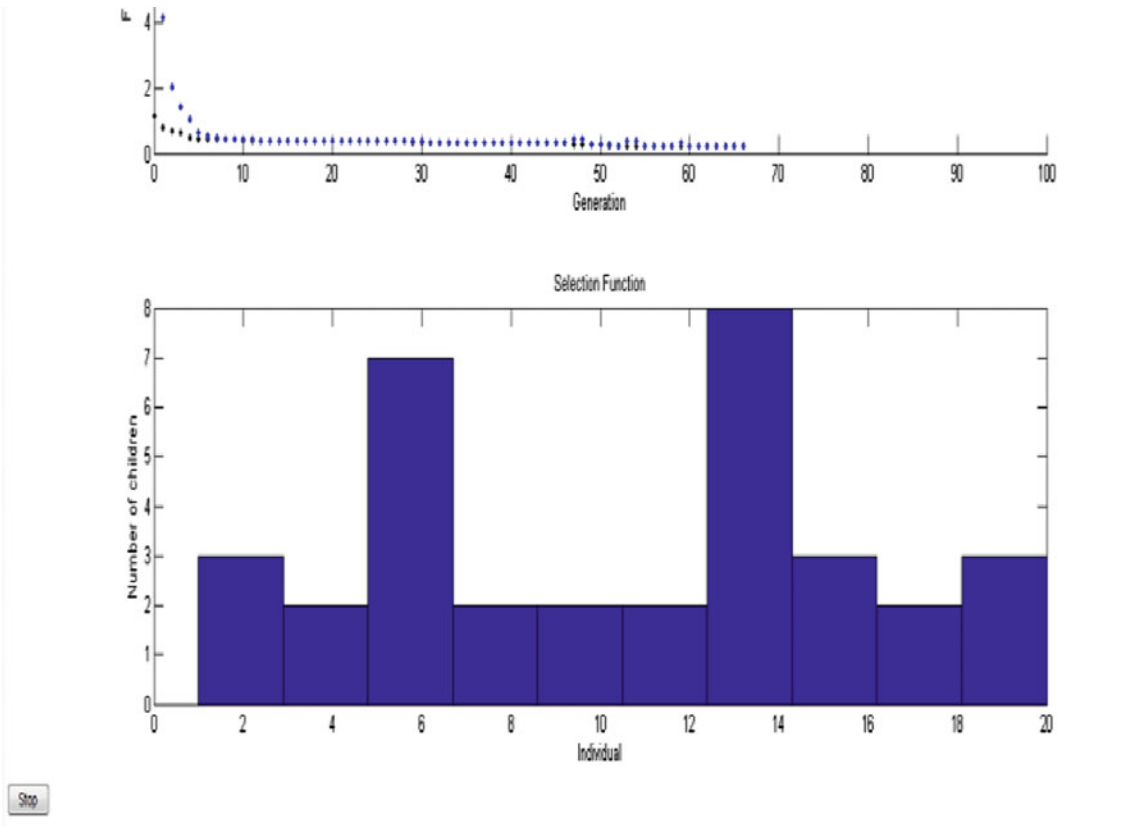

Fig. 5 Fitness curve based on 60 trials

Conclusion

The paper presents the problem of finding the Trade-off between Complexity and Deliverability aspect of software. Based the management of Trade-off [8–10], the Popularity index of the software is determined. Our Proposed algorithm of GA-Fuzzy approach makes the system more transparent and highly interoperable. The use Messy Algorithm along with Triangular fuzzy numbers provided a unique solution set which is a non-dominated solution set with multi-objective optimization. Fuzzy Rule Based is further tuned using GA.

Acknowledgment The author is thankful to **Mr. Mahesh Goel,** the **Managing Director** and **Dr. Devendra Agarwal, Director** of **Goel Group of Institutions, Lucknow** for their kind support and encouragement in promoting research activities in the Campus.

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Visual Servoing Based Positioning and Object Tracking on Humanoid Robot

Michael Bombile

Abstract

In this paper, we present a reactive vision based autonomous behavior on a humanoid robot. This perception action scheme enables the robot to walk toward a desired position or to track a given target and grasp it.

The proposed solution uses visual servoing as a reactive sensor based technique to address the problem of perception, thus to ensure reachability of the target object by generating appropriate motion commands. An omnidirectional velocity tracking gait generator is considered to solve the locomotion and balance problem. It is based on the framework developed by Herdt et al. (Advanced Robotics 24(5–6): 719–737, 2010; *2010 IEEE/RSJ International Conference on Intelligent Robots and Systems*, pp. 190–195, 2010), which has been extended for instantaneous rotational velocity in this work. The redundancy problem due to the high number of degrees of freedom associated with the mobility in operational space is handled by a task sequencing technique. This allows the improvement of reachability and manipulability while accounting for joints limits and obstacles avoidance. Finally, simulations on the humanoid robot NAO are shown to evaluate the effectiveness of the proposed scheme.

Keywords

Visual servoing • Walking • Humanoid

Introduction

Recently, there has been an increased interest in the field of service robotics, where one of the goals is the safe introduction of robots in environments designed for humans. The integration of robots in such complex, dynamic, and unstructured environments is a really challenging problem [3, 4]. The perception of the surrounding environment is required not only to perform tasks but also to adapt the robot's behavior accordingly.

This paper focuses on some applications of visual servoing on a humanoid robot. Visual servoing is the use of visual information in feedback, which offers good robustness to sensor calibration and modeling errors [5]. Using an on-board camera, our goal is to achieve fast and robust whole body positioning with respect to object or landscape, object tracking and grasping while walking. Thanks to visual servoing, the perception improves directly humanoid's actions, bypassing the time-consuming decision or planning phase.

Many papers have considered applications of visual servoing on humanoid robots. In [6], model based visual servoing is proposed and implemented on humanoid robot NAO for relative localization, walking and grasping. Within a task sequencing framework, in [7], visual servoing is applied as a secondary task on the humanoid robot HPR-2 to grasp a ball while walking. In [8] visual based manipulation task is implemented on the humanoid robot REEM. Feed-forward control is used to compensate for tasks interactions and redundancy is handled by a large projector operator [9].

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_4, © Springer International Publishing Switzerland 2015

While the above mentioned works emphasize grasping and manipulation tasks, Courty et al. [10], in a virtual reality framework, suggested visual servoing based whole body motion generation for highly reactive contexts. Michel et al. [11] combined visual sensing to a footstep planner on the humanoid robot ASIMO, thereby achieving autonomous walking while avoiding obstacles. However, these methods still rely on classical footstep planners, which need fast re-planning strategies if confronted with sudden obstacles. To address the reactivity issue, C. Dune et al. [12] proposed a visual based dynamic walk by coupling visual servoing directly with the online pattern generator developed by Herdt et al. [1, 2]. In his approach, no planning phase is required; the footsteps are controlled reactively by visual perception.

Our approach uses Herdt's online pattern generator as well, but with a slight modification to account explicitly for rotation instead of a predetermination. Unlike [12, 13], where the center of mass (CoM) is assumed to be rigidly linked to the camera, as such ignoring neck's joints, this paper proposes a vision based whole body motion. We extend the above reactive approaches to control not only the footsteps for positioning, but also for tracking moving objects and to control neck and hands for grasping while walking. Finally, taking advantage of the increased degrees of freedom, we show how to deal with redundancy to perform additional tasks such as joint limit and obstacle avoidance.

This paper is organized as follows. Section "Visual Servoing and Pattern Generator" recalls visual servoing concepts and proposes a small extension to the pattern generator. Section "Visual Servoing on Humanoid Robot" introduces visual servoing in a general framework for biped humanoid and explains the adopted positioning and tracking approaches. Section "Simulation Results" presents the simulation results using the NAO humanoid model. Finally Section "Conclusion and Future Works" concludes the paper.

Visual Servoing and Pattern Generator

This section outlines the basics of visual servoing and proposes a small extension to the online walking pattern generator developed by Herdt et al. [1, 2].

Visual Servoing Concepts

The visual servoing problem is usually formulated and solved in terms of regulation to zero a task function by appropriate of camera/robot motion [5]. This task function represents the robotic task to be performed, which is defined in terms of desired image features, and constitutes the goal to reach from a given initial configuration [14].

Let us consider the following task function:

$$e(t) = s(r(t)) - s^{\star} \tag{1}$$

where $s\,(r(t))$ represents a vector of selected visual features, $s^{\star}$ represents the desired value of $s(r(t))$ and $r(t)$ is an element of SE_3 (special Euclidean space) describing the situation between the scene and the camera at instant t.

The time derivative of the task function in camera space and in joint space is given by [15]:

$$\begin{cases} \dot{e}(t) = \boldsymbol{L}_e{}^{c}V + \dfrac{\partial e}{\partial t} \\ \dot{e}(t) = \boldsymbol{J}_e\dot{\boldsymbol{q}} + \dfrac{\partial e}{\partial t} \end{cases} \tag{2}$$

Here $\boldsymbol{L}_e = \frac{\partial e}{\partial s}L_s$ is the interaction matrix of the task function, where L_s is the interaction matrix associated with feature s [16]. ${}^{c}V = (\upsilon, \omega)$ is the kinematic screw of the camera relative to the object and expressed in the camera frame. $\boldsymbol{J}_e = \frac{\partial e}{\partial s}J_s$ is the Jacobian of the task function, where $J_s = L_s{}^{c}W_e{}^{e}J_q$ is the Jacobian of the visual features. ${}^{c}W_e$ is the twist transformation matrix between the end effector frame and the camera frame. ${}^{e}J_q$ is the robot Jacobian. $\frac{\partial e}{\partial t}$ is the variation of *e(t)* due to object unknown motion.

For an exponential decrease of the task function $(\dot{e}(t) = \lambda e(t))$, the control law is given by [14, 5, 15]:

$$\begin{cases} {}^{c}V = -\lambda\widehat{L_e^{\dagger}} + e(t) \\ \dot{\boldsymbol{q}} = -\lambda\widehat{J_e^{\dagger}} + e(t) \end{cases} \tag{3}$$

Here λ is a gain matrix. $L_e^{\dagger}$ is the Moore-Penrose pseudo inverse of L_e. In practice, only its approximation or estimation denoted by the "hat" symbol can be known.

The performances of the resulting closed-loop system depend on the choice of s, the associated form of L_s. When $s = (x_i, y_i)$, the coordinates of a point defined in image space, the corresponding interaction matrix is given by [16, 5]:

$$L_s = \begin{bmatrix} -\dfrac{1}{Z_i} & 0 & \dfrac{x_i}{Z_i} & x_i y_i & -\left(1 + x_i^2\right) & y_i \\ 0 & -\dfrac{1}{Z_i} & \dfrac{y_i}{Z_i} & 1 + y_i^2 & -x_i y_i & -x_i \end{bmatrix} \tag{4}$$

where Z_i is the depth of the corresponding 3D point $P = (X, Y, Z)$ expressed in the camera frame.

Explicit Orientation for Pattern Generator

Although it is able to follow a rotational velocity, the pattern generator as proposed in [1, 2], requires predetermination of the rotation. The extension aims at improving the pattern generator for high reactive motions.

Referring to *Herdt's* MPC formulation, particularly the feet trajectories and given a world frame $\sum_w$, the feet poses ${}^wF^{\alpha}_{k+1}$ on the ground over the predicted horizon NT can be written in compact form as:

$$ {}^wF^h_{k+1} = V^c_{k+1}\,{}^w\mathbf{f}^h_k + V^f_{k+1}\,{}^w\mathbf{f}^h_{k+1} \tag{5}$$

in which ${}^wf^h_k$ is the current pose of the foot on the ground, ${}^wf^h_{k+1}$ the following steps, and the superscript h represents the pose parameters (x, y, θ), assuming a horizontal ground. V^c_{k+1} and V^f_{k+1} are selection matrices associating steps to corresponding sampling time.

When accounting explicitly for orientation, Eq. (5) has to be reformulated, since the translations along x and y will be geometrically coupled.

Let us consider two complete walking cycles, the positions of the four steps with respect to Σ_w are given by:

$$\mathbf{F}^w_{s_{i+1}} = \mathbf{I}.F^w_{s_i} + \mathbf{R}_{k\_xy}\,\Delta\mathbf{F}_{s_i} \tag{6}$$

with

$$\mathbf{F}^w_{s_i+1} = \begin{bmatrix} F^w_{s_i} \\ F^w_{s_{i+1}} \\ F^w_{s_{i+2}} \\ F^w_{s_{i+3}} \end{bmatrix}, \quad \mathbf{R}_{xy} = \begin{bmatrix} 0 & 0 & 0 \\ R^w_{s_i} & 0 & 0 \\ R^w_{s_i} & R^w_{s_{i+1}} & 0 \\ R^w_{s_i} & R^w_{s_{i+1}} & R^w_{s_{i+2}} \end{bmatrix}, \tag{7}$$

$$\Delta\mathbf{F}_{s_i} = \begin{bmatrix} F^{s_i}_{s_{i+1}} \\ F^{s_{i+1}}_{s_{i+2}} \\ F^{s_{i+2}}_{s_{i+3}} \end{bmatrix}, \quad \mathbf{R}^w_{s_i} = \begin{bmatrix} \cos\theta_{s_i} & -\sin\theta_{s_i} \\ \sin\theta_{s_i} & \cos\theta_{s_i} \end{bmatrix} \tag{8}$$

where $F^w_{s_i} = \begin{bmatrix} {}^wf^x_k & {}^wf^y_k \end{bmatrix}^T$ is the position of step s_i with respect to Σ_w, $R^w_{s_i}$ and θ are respectively the orientation matrix and orientation angle of the foot at step s_i with respect to Σ_w.

Using expression of $F^w_{s_i}$ and Eqs. (6)–(8), we can write separately the x and y components of the following steps ${}^w\mathbf{f}^h_{k+1}$. We have:

$$\begin{cases} {}^w\mathbf{f}^x_{k+1} = \mathbf{1}_3.{}^wf^x_k + \mathbf{R}_x\Delta\mathbf{F}_{s_i} \\ {}^w\mathbf{f}^y_{k+1} = \mathbf{1}_3.{}^wf^y_k + \mathbf{R}_y\Delta\mathbf{F}_{s_i} \end{cases} \tag{9}$$

with

$${}^w\mathbf{f}^x_{k+1} = \begin{bmatrix} {}^wf^x_{s_{i+1}} \\ {}^wf^x_{s_{i+2}} \\ {}^wf^x_{s_{i+3}} \end{bmatrix}, {}^w\mathbf{f}^y_{k+1} = \begin{bmatrix} {}^wf^y_{s_{i+1}} \\ {}^wf^y_{s_{i+2}} \\ {}^wf^y_{s_{i+3}} \end{bmatrix}, \tag{10}$$

$$\mathbf{R}_x = \begin{bmatrix} c\theta_{s_i} & -s\theta_{s_i} & 0 & 0 & 0 & 0 \\ c\theta_{s_i} & -s\theta_{s_i} & c\theta_{s_{i+1}} & -s\theta_{s_{i+1}} & 0 & 0 \\ c\theta_{s_i} & -s\theta_{s_i} & c\theta_{s_{i+1}} & -s\theta_{s_{i+1}} & c\theta_{s_{i+2}} & -s\theta_{s_{i+2}} \end{bmatrix} \tag{11}$$

$$\mathbf{R}_y = \begin{bmatrix} s\theta_{s_i} & c\theta_{s_i} & 0 & 0 & 0 & 0 \\ s\theta_{s_i} & c\theta_{s_i} & s\theta_{s_{i+1}} & c\theta_{s_{i+1}} & 0 & 0 \\ s\theta_{s_i} & c\theta_{s_i} & s\theta_{s_{i+1}} & c\theta_{s_{i+1}} & s\theta_{s_{i+2}} & c\theta_{s_{i+2}} \end{bmatrix} \tag{12}$$

and the vector of relative foot steps

$$\Delta\mathbf{F}_{s_i} = \begin{bmatrix} f^{x\_s_i}_{s_{i+1}} & f^{y\_s_i}_{s_{i+1}} & f^{x\_s_{i+1}}_{s_{i+2}} & f^{y\_s_{i+1}}_{s_{i+2}} & f^{x\_s_{i+2}}_{s_{i+3}} & f^{y\_s_{i+2}}_{s_{i+3}} \end{bmatrix}^T \tag{13}$$

Substituting Eqs. (9) to (13) in (5), we obtain in compact form the expressions of the feet steps and their orientations over the predicted horizon NT

$$\begin{cases} {}^w\mathbf{F}^{x\_ref}_{k+1} = \left(V^c_{k+1} + V^f_{k+1}\mathbf{1}_3\right){}^wf^x_k + V^f_{k+1}\cdot\mathbf{R}_x\cdot\Delta\mathbf{F}_{s_i} \\ {}^w\mathbf{F}^{y\_ref}_{k+1} = \left(V^c_{k+1} + V^f_{k+1}\mathbf{1}_3\right){}^wf^y_k + V^f_{k+1}\cdot\mathbf{R}_y\cdot\Delta\mathbf{F}_{s_i} \\ {}^w\mathbf{\Theta}^{ref}_{k+1} = V^c_{k+1}\,{}^w\theta^c_k + V^f_{k+1}\,{}^w\boldsymbol{\theta}^f_{k+1} \end{cases} \tag{14}$$

Note that the analysis of Eqs. (9)–(14) shows that only three future values of θ are needed over two cycles. By formulating and computing rotation as such, we have avoided to propagate non-linearities all over the predicted values of x and y, thus making the system piecewise linear. In this way, we have addressed the nonlinearity challenge mentioned in [1, 2, 12].

Objective Function

The new objective function is similar to the one presented in [2], simply augmented with the orientation. In this way, MPC will generate the feet placements and their orientations automatically while following reference velocities. We have:

$$\min_{u_k}\{g(x,y) + g(\theta)\} \tag{15}$$

with $g(x,y)$ and $g(\theta)$ given by

$$\begin{cases} g(x,y) = \sum\limits_{h=x,y}\left\{\frac{\mu}{2}\left\|\widetilde{C}^h_{k+1}\right\|^2 + \frac{\sigma}{2}\left\|\dot{C}^h_{k+1} - \dot{C}^h_{ref}\right\|^2 \right. \\ \qquad\left. + \frac{k}{2}\left\|P^h_{k+1} - {}^wF^{h\_ref}_{k+1}\right\|^2 + \frac{\varepsilon}{2}\left\|EC^h_{k+1} - \dot{C}^h_{ref}\right\|^2\right\} \\ g(\theta) = \left\{\frac{\mu}{2}\left\|\widetilde{\theta}_{k+1}\right\|^2 + \frac{\sigma}{2}\left\|\dot{\theta}_{k+1} - \dot{\Theta}_{ref}\right\|^2 \right. \\ \qquad\left. + \frac{k}{2}\left\|\theta_{k+1} - {}^w\Theta^{ref}_{k+1}\right\|^2 + \frac{\varepsilon}{2}\left\|E\theta_{k+1} - \dot{\Theta}_{ref}\right\|^2\right\} \end{cases} \tag{16}$$

where μ, σ, k and ε are weighting factors, C^h and P^h represent the coordinates, respectively of the Center of Mass (CoM) and of the Zero Moment Point (ZMP), and E is a double diagonal matrix used to compute the average velocity of the CoM over two steps.

In canonical form, the objective can be written as:

$$\min_{u_k} \frac{1}{2} \cdot u_k^T \mathbf{Q}_k u_k + \mathbf{P}_\mathbf{k}^T u_k \tag{17}$$

with

$$u_k = \left[\dddot{C}^x_{k+1} \, \dddot{C}^y_{k+1} \Delta\mathbf{F}_{s_i} \, \ddot{\theta}_{k+1} \, \boldsymbol{\theta}^f_{k+1} \right]^T \tag{18}$$

$$\mathbf{Q_k} = \begin{bmatrix} Q_{x11} & 0 & Q_{x12} & 0 \\ 0 & Q_{y11} & Q_{y12} & 0 \\ Q_{x21} & Q_{y21} & Q_{xy22} & 0 \\ 0 & 0 & 0 & Q_\theta \end{bmatrix} \tag{19}$$

$$\mathbf{P_k} = \left[p_{k\_\dddot{C}^x} \quad p_{k\_\dddot{C}^y} \quad p_{k\_f^{x+y}} \quad p_{k\_\ddot{\theta}} \quad p_{k\_f^\theta} \right]^T \tag{20}$$

Using the system of Eq. (14) instead of (5), the elements of $\mathbf{Q_k}$ and $\mathbf{P_k}$ can be easily deduced from those computed in [2], from where the notation used above has been adapted.

For constraints formulation which account for balance, prevent legs overstretching, self-collision and other kinematic limitations, we refer the reader to [1], where more details can found. Nevertheless, the formulation has to be adapted to the new vector u_k given by Eq. (18).

Visual Servoing on Humanoid Robot

In this section, we start first by determining the geometric Jacobian of the robot which is necessary in the computation of the visual task Jacobian and then we present our approach to visual servoing based tasks on a humanoid robot.

Kinematic Modeling

A humanoid robot when standing on one leg can be seen as a serial-chain manipulator (leg joints and upper limbs joints) whose base is the stance foot and the end-effector could be either the head or one of both hands.

Using inverse kinematics to compute joints velocities, based on this assumption, may results in legs joints not satisfying the balance requirement. Thus, the modeling approach has to be reconsidered in order to account for walking.

Let us consider Fig. 1a, the head or/and hand chain can be viewed as a manipulator of joints vector $\mathbf{q}_m$ fixed on a moving base, of joints vector $\mathbf{q}_b = [x_b \; y_b \; z_b \;\; \alpha \; \beta \; \gamma]^T$, which undergoes three translations and three rotations (e.g. Euler angles) [17]. Within this model, all possible configurations of the base are determined by leg joint configurations as shown on Fig. 1b.

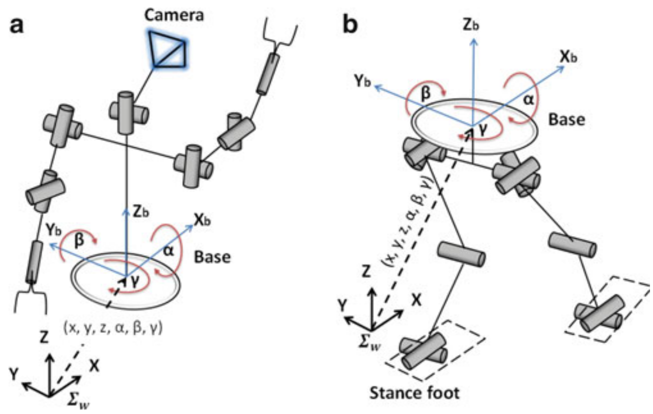

Fig. 1 Simplified kinematic model of Humanoid robot. (**a**) upper body part and (**b**) lower body part

Given this reduced model, the velocity screw of the end effector with configuration vector $r_e = f(q)$ is given by:

$${}^w\dot{\boldsymbol{r}}_e = \left[\frac{\partial f}{\partial q_m} \quad \frac{\partial f}{\partial q_b} \right] \begin{bmatrix} \dot{\mathbf{q}}_m \\ \dot{\mathbf{q}}_b \end{bmatrix} = {}^w\boldsymbol{J}_{mb}\dot{\mathbf{q}}_{mb} \tag{21}$$

where the robot Jacobian is given by:

$${}^w\boldsymbol{J}_{mb} = \left[\begin{bmatrix} \boldsymbol{R}_b^w & 0_3 \\ 0_3 & \boldsymbol{R}_b^w \end{bmatrix} \boldsymbol{J}_m^b \begin{bmatrix} \boldsymbol{I}_3 & -\left[\boldsymbol{R}_b^w t_{r_e}^b\right]_\times \boldsymbol{T}(\phi) \\ 0_3 & \boldsymbol{T}(\phi) \end{bmatrix} \right] \tag{22}$$

With $\dot{\mathbf{q}}_{mb} = \left[\dot{\mathbf{q}}_m^T \quad \dot{\mathbf{q}}_b^T \right]^T$.

In these equations, $\boldsymbol{J}_m^b$ is the Jacobian of the considered endeffector expressed in the base frame. ${}^w\boldsymbol{R}_b$ is the rotation matrix from the base frame to the world frame $\sum_w$. $t_{r_e}^b$ is the position vector of the end-effector $\boldsymbol{r}_e$ with respect to the base b. $\boldsymbol{T}(\phi)$ is the matrix that maps angular velocities to time derivative of the chosen Euler angles. Notation $[\cdot]_\times$ represents a skew symmetric matrix.

It can also be shown that:

$$\dot{\mathbf{q}}_b = \begin{bmatrix} \boldsymbol{R}_b^w & -\left[\boldsymbol{R}_b^{wb} t_{lg}\right]_\times \boldsymbol{R}_b^w \\ 0_3 & \boldsymbol{T}(\phi)^{-1} \boldsymbol{R}_b^w \end{bmatrix}^b \boldsymbol{J}_{lg} \dot{\mathbf{q}}_{lg} \tag{23}$$

where ${}^bJ_{lg}$ is the Jacobian of the leg's end-effector expressed in the base frame. $\dot{\mathbf{q}}_{lg}$ is the leg's joint velocity vector and ${}^bt_{lg}$ is the position vector of the leg's end-effector with respect to the base b.

Visual Servoing Based Humanoid's Positioning

In this task, from a given initial position, the humanoid robot walks toward a desired position with respect to a target object and stop at that desired pose. This task involves all legs joints and neck's joints. Using Eqs. (2) and (22), the associated visual task Jacobian is given by:

$$J_{e_{mb}} = L_{s_{tsk}}{}^{c}W_{w}{}^{w}J_{mb} \tag{24}$$

where $L_{s_{tsk}}$ is the interaction matrix associated to the chosen features vector s. ${}^{c}W_{w}$ is the twist transformation matrix from $\sum_{w}$ to the camera frame and is given by:

$${}^{c}W_{w} = \begin{bmatrix} {}^{c}R_{w} & \left[{}^{c}t_{w}\right]_{\times}{}^{c}R_{w} \\ 0_{3\times 3} & {}^{c}R_{w} \end{bmatrix} \tag{25}$$

with ${}^{c}R_{w}$ and ${}^{c}t_{w}$ respectively, the rotation and translation from $\sum_{w}$ to the camera frame, expressed in the camera frame. In this case $\sum_{w}$ is taken to be the stance foot frame. ${}^{w}\boldsymbol{J}_{mb}$ is given by (22) within which ${}^{b}\boldsymbol{J}_{m}$ is the head chain's Jacobian ${}^{b}\boldsymbol{J}_{head}$, with $\mathbf{q}_{m} = \mathbf{q}_{neck} = [\theta_{1n} \quad \theta_{2n}]^{T}$ and where θ_{in} are the neck joints.

1. *Redundancy resolution of Visual task* : Given that θ_{1n} and γ are rotation angles about the same axis Z, for any value of θ_{1n} there is a corresponding value of γ satisfying the task. Thus, there exist an infinite number of solutions to the inverse kinematics. To address this redundancy problem, we exploit the null space of the main task to perform a secondary one [18]. We chose to align the body with the direction of sight $(min|\theta_{1n} - \gamma|)$.
   Using the projected gradient of function $H(q(t)) = (\theta_{1n} - \gamma)^{2}$ which is minimal when $\theta_{1n} = \gamma$. From Eq. (3), the control law can now be written as follows:

$$\dot{\mathbf{q}}_{mb} = -\widehat{J^{\dagger}_{e_{mb}}}\lambda e - k\mathbf{P}_{mb}g_{1}(t) \tag{26}$$

   where $\mathbf{P}_{mb} = \left(\boldsymbol{I} - J^{\dagger}_{emb}J_{e_{mb}}\right)$ is the projection operator onto the null space of the task, $\dot{\mathbf{q}}_{mb} = \left[\dot{\theta}_{1n}\,\dot{\theta}_{2n}\,\dot{x}_{b}\,\dot{y}_{b}\,\dot{z}_{b}\,\dot{\alpha}\,\dot{\beta}\,\dot{\gamma}\right]^{T}$, $g_{1}(t)$ is the gradient of $H\ (q(t))$ and $k = k(\theta_{1n},\gamma)$ is an adaptive gain given by:

$$k = \eta\left(1 - \exp\left(-\frac{(\theta_{1n} - \gamma)^{2}}{\Delta q_{1}^{2}\gamma}\right)\right)$$

   with η a constant and $\Delta\ q_{1r}$ the admissible difference between θ_{1n} and γ.
   The action of $g_{1}(t)$ will not affect the task $\dot{e}(t)$ and will simply result in an internal motion on the robot.
2. *Integrating walking motion:* The pattern generator uses as references only three computed joints velocities $(\dot{x}_{b}, \dot{y}_{b}, \dot{\gamma})$ of the base $\dot{\mathbf{q}}_{b}$. Thus, the motion due to the roll angle α, the pitch angle β and the translation z will be considered as disturbances to the task. It, therefore, needs to be compensated by using the manipulator joints $\mathbf{q}_{m}$.
   The variation of the task function is now obtained as follows:

$$\dot{e}(t) = L_{s_{tsk}}{}^{c}W_{w}\left({}^{w}\boldsymbol{J}_{head}\dot{\mathbf{q}}_{neck} + {}^{w}\boldsymbol{J}_{b}\dot{\mathbf{q}}_{b\_walk} + {}^{w}\boldsymbol{J}_{b}\dot{\mathbf{q}}_{b\_dist}\right) \tag{27}$$

   After compensation, Eq. (27) can be written as:

$$\dot{e}(t) = L_{s_{tsk}}{}^{c}W_{w}\left({}^{w}\boldsymbol{J}_{head}\dot{\mathbf{q}}_{m\_walk} + {}^{w}\boldsymbol{J}_{b}\dot{\mathbf{q}}_{b\_walk}\right) \tag{28}$$

   With ${}^{w}\boldsymbol{J}_{m}\dot{\mathbf{q}}_{m\_walk} = {}^{w}\boldsymbol{J}_{head}\dot{\mathbf{q}}_{neck} + {}^{w}\boldsymbol{J}_{b}\dot{\mathbf{q}}_{b\_dist}$
   The compensating joint velocities are then given by:

$$\dot{\mathbf{q}}_{neck\_walk} = \widehat{\boldsymbol{J}^{\dagger}_{e_{head}}}\left(\dot{e}(t) - J_{e_{b}}\dot{\mathbf{q}}_{b\_walk}\right) + \mathbf{P}_{hd}\cdot g_{2}(t) \tag{29}$$

   $\mathbf{P}_{hd}$ is the projection operator associated to $\boldsymbol{J}_{e_{head}}$ and $g_{2}(t)$ is a joints velocities vector resulting from a secondary objective such as joint limits avoidance.
   Using Eq. (23), we rewrite Eq. (29) in terms of leg's joints, $\dot{\mathbf{q}}_{lg\_walk}$ walk resulting from the walking pattern generator which accounts already for balance. Then we have:

$$\dot{\mathbf{q}}_{neck\_walk} = \widehat{\boldsymbol{J}^{\dagger}_{e_{head}}}\left(\dot{e}(t) - J_{e_{leg}}\dot{\mathbf{q}}_{lg\_walk}\right) + \mathbf{P}_{hd}\cdot g_{2}(t) \tag{30}$$

   In Eqs. (29) and (30), $\boldsymbol{J}_{e_{i}} = L_{s_{tsk}}{}^{c}W_{w}{}^{w}\boldsymbol{J}_{i}$ with $i = head$, b, leg and ${}^{w}J_{leg}$ is the leg Jacobian related to the camera and expressed in $\sum_{w}$.

Visual Servoing Based Humanoid's Tracking

In tracking, unlike positioning, the robot does not stop at the relative desired pose, but has to maintain it while walking.

In the formulation of the positioning task, an additional term, $\frac{\partial e}{\partial t}$, accounting for the object's motion has to be included in the control law in order to ensure an exponential decrease of the task function [15].

Assuming knowledge of the object velocity, ${}^{w}\dot{r}_{object}$, with respect to the world frame, the variation of the task function due to object motion, while using image based visual servoing (IBVS) with feature points, is given by:

$$\frac{\partial e_{i}}{\partial t} = \begin{bmatrix} \frac{1}{Z_{i}} & 0 & -\frac{x_{i}}{Z_{i}} \\ 0 & \frac{1}{Z_{i}} & -\frac{y_{i}}{Z_{i}} \end{bmatrix}{}^{c}W_{w}{}^{w}\dot{r}_{object} \tag{31}$$

From Eq. (2) and (31), the control law becomes:

$$\dot{\mathbf{q}}_{mb} = -\widehat{J^{\dagger}_{e}}\lambda e - \widehat{J^{\dagger}_{e}}\frac{\partial e}{\partial t} - k\cdot\mathbf{P}\cdot\left(\nabla_{q}H\right) \tag{32}$$

with the compensating neck joints velocities as follows:

$$\dot{\mathbf{q}}_{neck\_walk} = \widehat{J^{\dagger}_{e_{head}}}\left(-\lambda e - \frac{\partial e}{\partial t} - \mathbf{J}_{e_{leg}}\dot{\mathbf{q}}_{lg\_walk}\right) \quad (33)$$

Simulation Results

In this section, all results presented have been tested on a simulator designed by the author for the purpose of the research. The simulator uses kinematic parameters of the NAO robot V4.0 that can be found in [19]. IBVS is used with four coplanar points (square of side 0.10 m) to control the humanoid walk. Its corresponding interaction matrix is a stack of four matrices given by (4). The required image points depths are assumed to be roughly known. A weighted sum of the current and the desired image Jacobian ($L_{s_{tsk}} = (1-\rho).L_s + \rho.L_{s*}$) has been used to prevent ill conditioning or singularities during task execution [20].

All velocity inputs to the pattern generator (PG) have been filtered to avoid high jerk at the beginning and during the walk. Four simulations are presented; for all of them the robot starts at the origin of the axes and the desired relative pose of the camera with respect to the object is (−0.15, 0.10, 0.10) m and (pi/2, 0, pi/2) rad. The camera is located at the head (0.485 m from the ground).

The first simulation is a translation in X direction. The pose of the target is (0.80, −0.10, 0.365) m and (0, 0, 0) rad. As can be seen in Fig. 2, the image feature points converge to their desired values with some oscillations due to the sway motion induced by the walk. In 3D space, the camera trajectory goes straight toward the target. The x velocity input to the PG decreases exponentially and the effect of the filter at the beginning can also be observed.

The second simulation is a combined motion involving X and Y translation and rotation about the Z axis. The target pose is (0.80, −0.40, 0.365) m and (0, 0, −pi/3) rad. Figure 3 shows that the robot still converge to its desired pose. It can also be noticed that far from the target, the sway motion in image is mechanically compensated by neck joints, but this solution becomes limited when close to the target. The observed orientation velocity $\dot{\gamma}$ to the PG after x and y settling, is due to internal motion which, as secondary objective, tries to align the body and the sight direction (θ_{1n} and γ).

The third experiment is a tracking of the target moving with 0.04 m/s and −0.04 m/s in X and Y directions respectively. The initial target pose is (0.50, −0.30, 0.365) m and (0, 0, −pi/4) rad. As shown in Fig. 4, the robot walked toward the target, oriented itself to face the target and maintained the relative desired pose. This can be clearly seen on the velocity plots where $\dot{y}_b$ and $\dot{\gamma}$ settle about zero, while $\dot{x}_b$ has now the resultant velocity.

The fourth experiment is a complex tracking motion. The target describes a curve in space by rotating its frame at 0.04 rad/s about the origin of axes. The target initial pose is (0.60, −0.40, 0.365) m and (0, 0, −2pi/15) rad. In this case, the robot is also able to track the target as can be seen in Fig. 5, but it appears that an almost constant features error causes some image feature points to leave the field of view. This experiment shows that the feed-forward term (31) which was derived for the velocity, cannot compensate for such motion, since the acceleration is no longer zero.

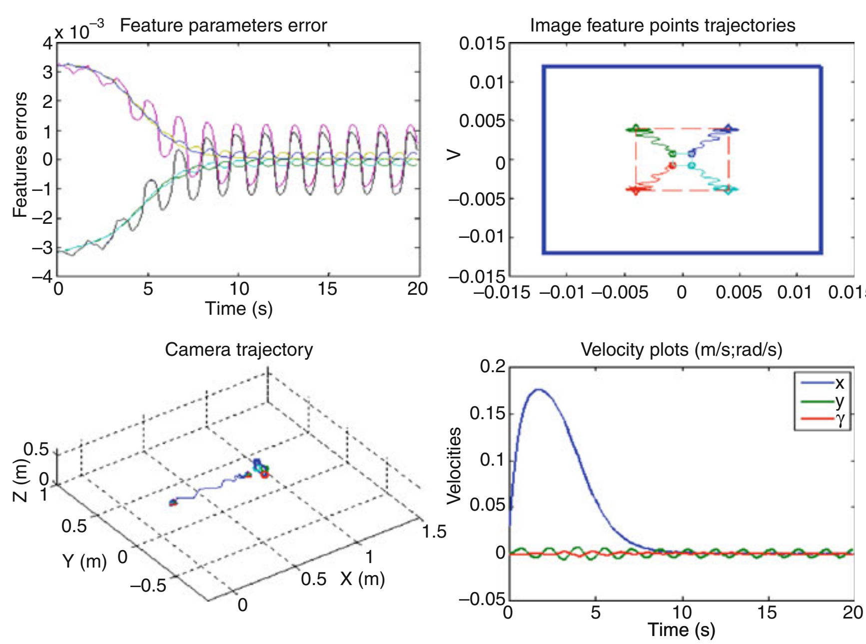

Fig. 2 Positioning task with robot facing the target

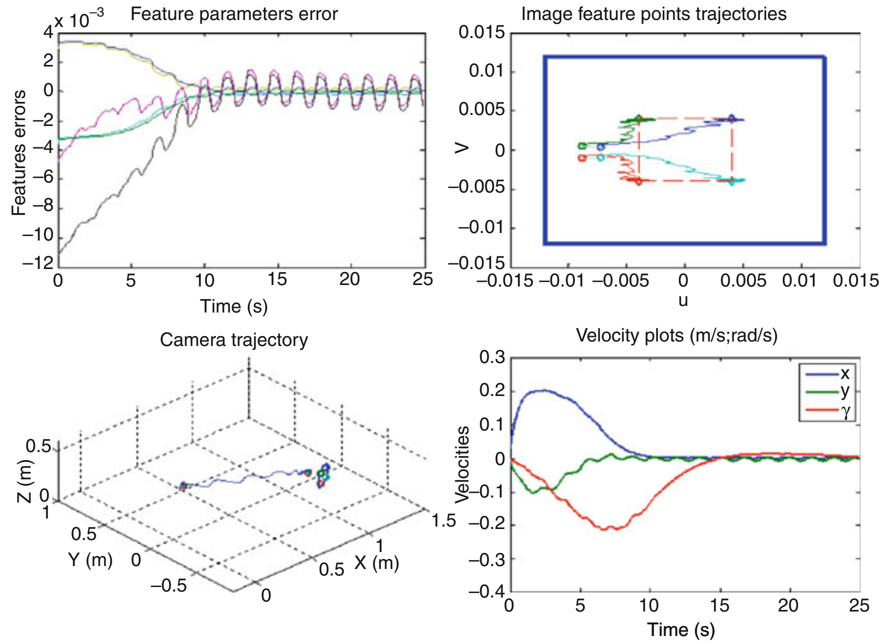

Fig. 3 Positioning task with target in general configuration

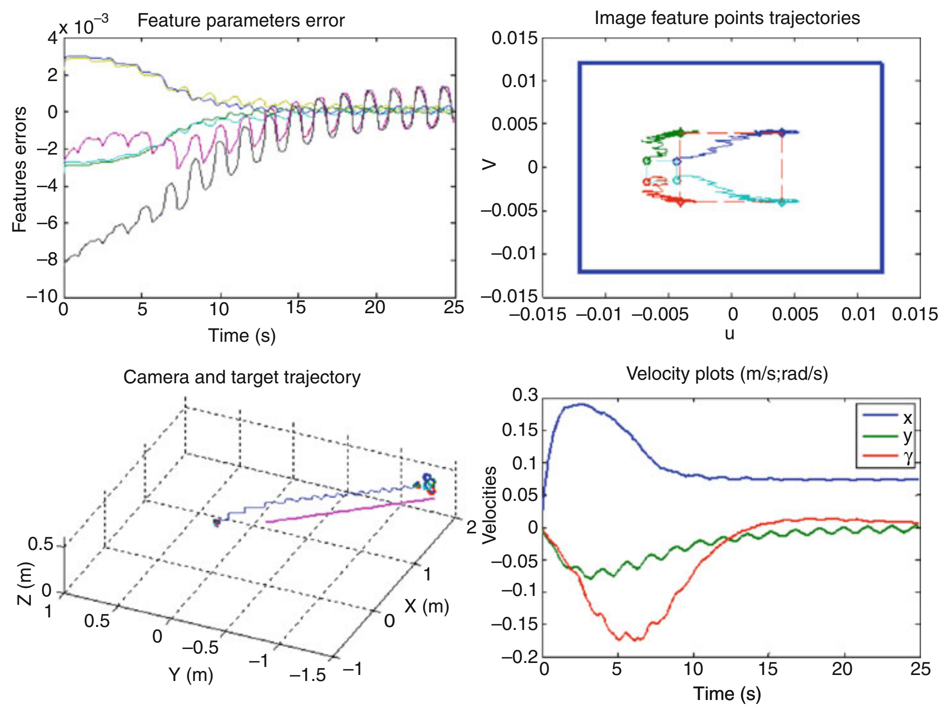

Fig. 4 Tracking task with translating target

Finally, Fig. 6 shows a 3D simulation of the grasping task after reaching the desired pose.

Conclusions and Future Works

In this paper, we have shown through simulations how visual servoing based whole body motion can be performed on a humanoid robot. The positioning and tracking task for grasping have been formulated and discussed. An easy way to combine visual servoing with the online pattern generator, while including all joints and handling redundancy has been presented. IBVS has been applied for both tasks. The tracking task needed a feed-forward term to compensate for target motion; it has exhibited, in simulation, good performance for constant velocity tracking. However, the feed-forward term used has revealed some limitations in tracking varying

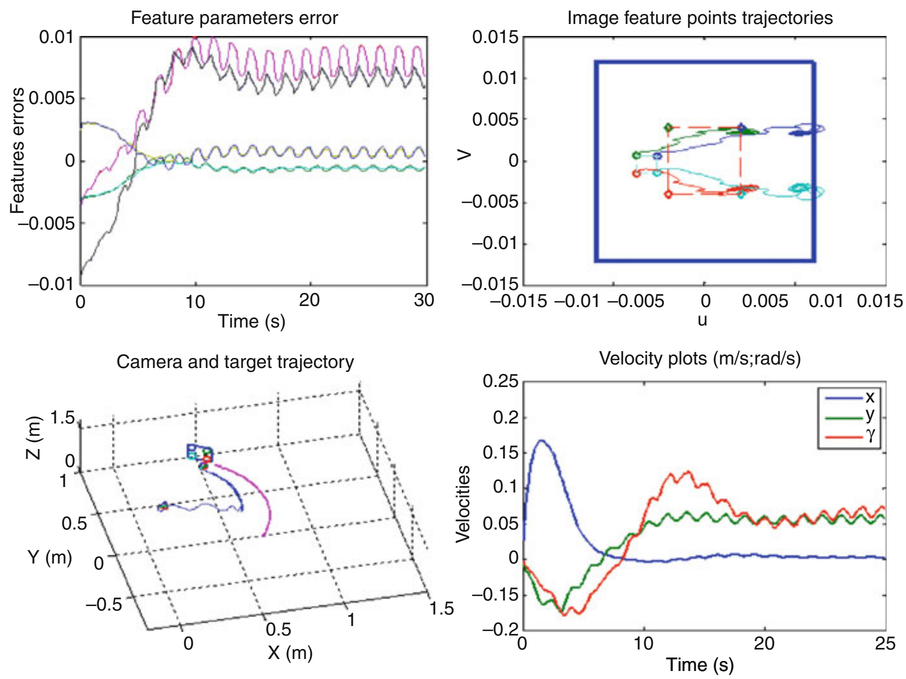

Fig. 5 Tracking task of target with curvilinear motion

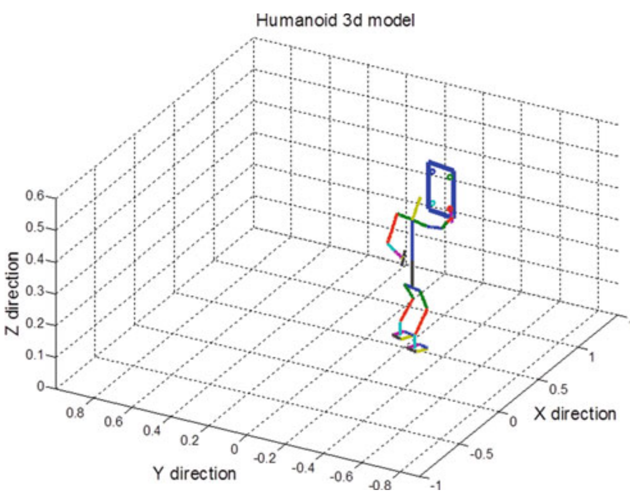

Fig. 6 Humanoid grasping the target at the desired pose

velocities such as in curvilinear target motion, therefore it needs to be extended.

Future works will be devoted to the implementation on the real robot, where better controllers will have to be designed in order to improve tracking performances for all motions and to account for the dynamics of the robot. In addition, the target motion which was assumed to be known will have to be estimated directly from the image.

Acknowledgment The author would like to thank Prof. M. Braae for his valuable comments.

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Obtaining Agents and Entities from Natural Language

Luis Alfonso Lezcano, Jaime Alberto Guzmán-Luna, and Sebastián Alonso Gómez

Abstract

This paper presents a method for identifying and classifying agents. Likewise, it also shows a method for identifying entities and attributes based on the definition of the KAOS goal model. The starting point is the specification of software requirements expressed in Spanish. The works presented in the literature fail to identify the traceability that should exist between the natural language and the components of the KAOS goal diagram. The method proposed in this paper makes it possible to achieve the consistency that should exist between natural language and the components (entities with attributes and agents) of the KAOS goal diagram. In other words, said method guarantees coherence between the elements that are being modeled and the natural language. This process serves as a starting point for: (i) identifying the other components of the KAOS goal diagram, and (ii) automatically creating said diagram.

Keywords

Agent • Entity • Requirement • Software development

Introduction

The Software Engineering process involves applying scientific principles which allow for the ordered representation of a problem or need in a software-developed solution. This also includes lifetime maintenance. The tasks involved in Software Engineering begin before the code is written and/or processed, and continue after the initial version of the piece of software has been finished [1].

In [2], the authors indicate that the first two phases of the software's life cycle (Definition and Analysis) allow for the development of processes for identifying, analyzing and validating the requirements that the software must fulfill. However, the processes involved in these phases still present problems due to the gap in communication between the software analyst, who uses technical language, and the stakeholder, who uses a form of natural language that is particular to the problem's domain.

With the aim of minimizing the existing gap in communication between the analyst and the stakeholder [3, 4], propose using the KAOS goal diagram, which includes entities, operations, actors, requirements and agents among others. Furthermore we believe that by using this diagram one is able to identify and justify the importance of the future software to the stakeholders. Nevertheless, in KAOS it falls on the analyst to interpret and identify, based on the stakeholder's discourse (which is expressed in natural language), and the components of the goals diagram. In most cases, however, the technical process performed by the analyst results in the creation of components in the diagram (entities, operations and agents, etc.) which are not consistent with the stakeholder's discourse.

The structure of the rest of the paper is as follows: the related work which is the basis of this proposal is presented in Section "Related Work". Similarly, the agent

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_5, © Springer International Publishing Switzerland 2015

identification and classification process is detailed in Section "Agents Identification and Their Classification", while entity and attribute identification is described in Section "Entities and Attributes Identification". In addition, Section "Experiments" shows the experiments, results and discussion, and Section "Conclusion and Future Work", the conclusions and further study topics derived from this paper.

Related Work

Letier [5] shows that there are different techniques used to elicit requirements, which allow the analyst and stakeholder to carry out the first phases of the software's life cycle. Some of these techniques are described below: (i) originating with the user: this is one of the most intuitive techniques, and lets the stakeholders openly express their needs. This technique has the following shortcomings: (a) stakeholders do not have a clear idea of what they want; (b) they sometimes have difficulty expressing and/or transferring their knowledge; and (c) they use vocabulary unfamiliar to the analyst; (ii) Form Analysis: a compilation of information structured using variables which supports data entry and its respective recovery. It is considered a formal model as it does not present the inconsistencies of natural language. Relevant aspects of the problem domain may be omitted using this technique, as the stakeholder does not actively participate in constructing the forms; (iii) Interviews: this is one of the most widely used techniques in eliciting requirements. In it the analyst and stakeholder study the different aspects of the domain area together. Some of these aspects are: (a) the goals of the domain area; (b) the processes used to achieve these goals; and (c) the actors involved in these processes. However in the majority of cases this technique only allows for an overall view of the problem domain to be obtained, i.e. it does not yield detailed domain information; (iv) Scenarios; this technique uses a series of steps to describe the properties that the future software must possess. The scenarios used in the analysis phase are considered the primary input for identifying the goals. However there is a contradiction in this technique, since the goal diagram is used in the definition phase, which is prior to the analysis phase but required in order to validate the requirements of the future software between the analyst and stakeholder; (v) Tasks Analysis: this refers to a set of processes which analyze and describe the way in which stakeholders develop their tasks in terms of: (a) the activities they carry out; (b) the knowledge necessary to carry out their activities. This technique centers on the tasks of the existing system, i.e. it does not include the requirements of the future software; and (vi) Goals Analysis: the fundamentals of a piece of software are established based on the goals of the organization in which the software application will be in production. Generally the goals are defined as the goals which are to be fulfilled by the piece of software and its environment. However for this technique it is still not clear how the analyst and stakeholder should go about validating the requirements that the piece of software must fulfill.

The authors of [3, 4] use the goal-based technique and, as a result of this, the KAOS methodology has been obtained and presented. This methodology is based on temporal logic and refining techniques belonging to Artificial Intelligence. According to Fig. 1 (taken from [6]) a variety of elements should be used in the KAOS methodology in order to guarantee that the KAOS goal diagram is obtained, one of such elements is the entity and agent. However in [3, 4] the authors were unable to identify the traceability that should exist between the natural language, the entities with their respective attributes and the agents.

In [7] a meta-model of the KAOS goal diagram is presented, based on the UEML approach. The goal of this meta-model is comparison, consistency verification, an updating process, synchronization and translation between models. A segment of this meta-model is shown in Fig. 2. In it one can see the entity, which is displayed as an inheritance relationship of the object type. In this proposal, the authors were unable to identify the traceability that should exist between the natural language the entities with their respective attributes and the agents.

Agents Identification and Their Classification

Process for Obtaining Agents Using Natural Language

Based on studies from [8–11] a method is proposed for identifying components which could be associated with agents. The proposed method includes the following steps: (i) obtain the agent description document created by the user or analyst. This document constitutes the text that is to be processed, and is used throughout the entire method; (ii) the use of a lexo-syntactic parser to morpho-syntactically tag the text; (iii) lemmatize each word in the text, and (iv) identify empty words. These last three steps are performed automatically by the NL2KAOS software (developed in this proposal) with the articulation of other applications which allow this method to be automated. Subsequently, and to decide if the agent is human, a device or software, the list of hyperonymic is consulted, and from this the list of software an environment agents is obtained. Finally the name of the agent is acquired and a definition is drawn from a lexical database.

A document in Spanish is taken as input data, free from orthographic or grammatical errors or ambiguities, which

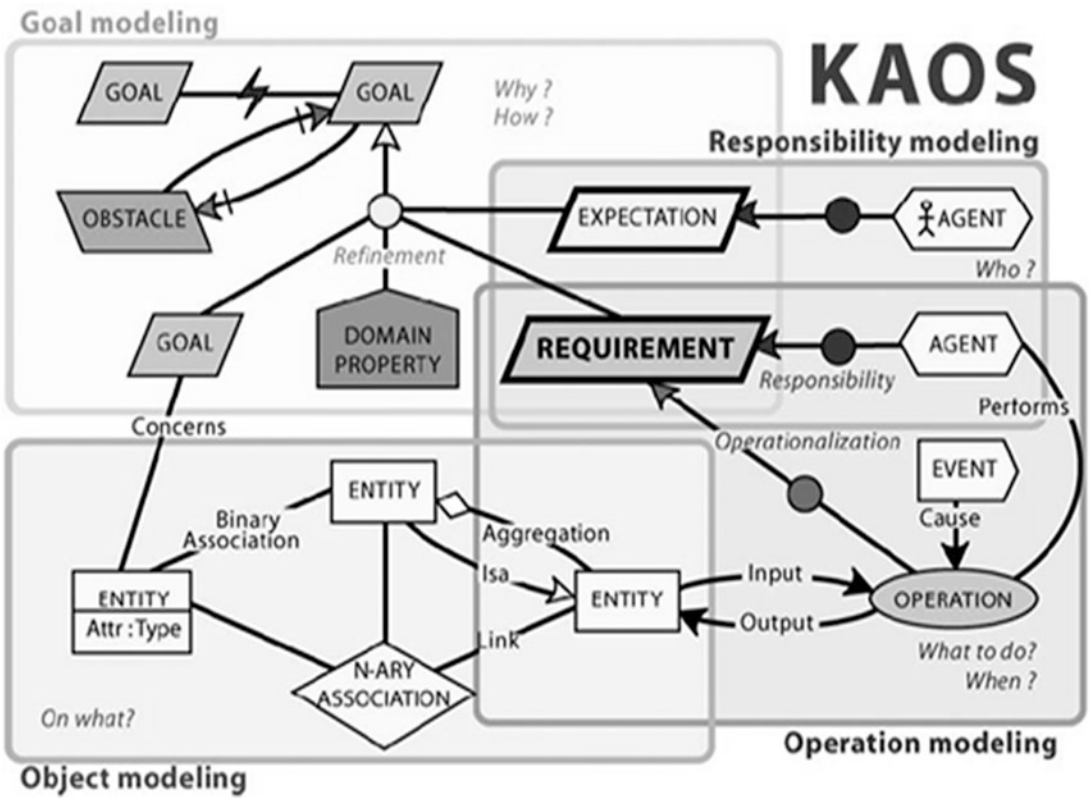

Fig. 1 Basic symbols of the KAOS goal diagram. Source: [6]

contains the specifications of the system's functional requirements, and which describes the agents (environment or software) that form part of the system in question. They may be associated with goals or directly with each one's operations. The morphosyntactic structure of the text which one is hoping to process is restricted to sentences in the active voice, as it is these which clearly indicate the subjects of the action. The order would therefore be: Noun–Verb–Complement.

All phrases should also explicitly contain the subject of the phrase. This is a simplification which is performed with the aim of avoiding working with the anaphora resolution. This is an additional problem which falls outside the scope of the proposed method, but which could be resolved in future studies. The process that was used is described in detail below.

Mophosyntactic Tagging

A morphosyntactic analysis of the input document, which complies with the aforementioned characteristics, is carried out with the aim of tagging each word in the text based on its grammatical category, i.e. its function within the context. This process is performed using the Freeling 3.0 (An Open-source Suite of Language Analyzers) tool for processing natural language [12–14]. The result of this process is a set of tags associated with all words and punctuation found in the text. The system that automates the method uses the punctuation tags to identify noun phrases; however it is not necessary to store these tags for this process, as only word tags are taken.

Lemmatization

The lexemes (or lemmas) are the smallest unit of lexical meaning of a word, and are important for comparing words without being affected by conjugations, plurals or any other possible modifications.

Identifying Empty Words (Stop Words)

Empty words are a set of words which have no meaning in and of themselves, such as articles, pronouns and prepositions. In order to identify them in the document the words in the text are used, as neither the morpho syntactic tags, nor the lexemes are of interest for this action. This is because the list takes into account all possible cases of empty

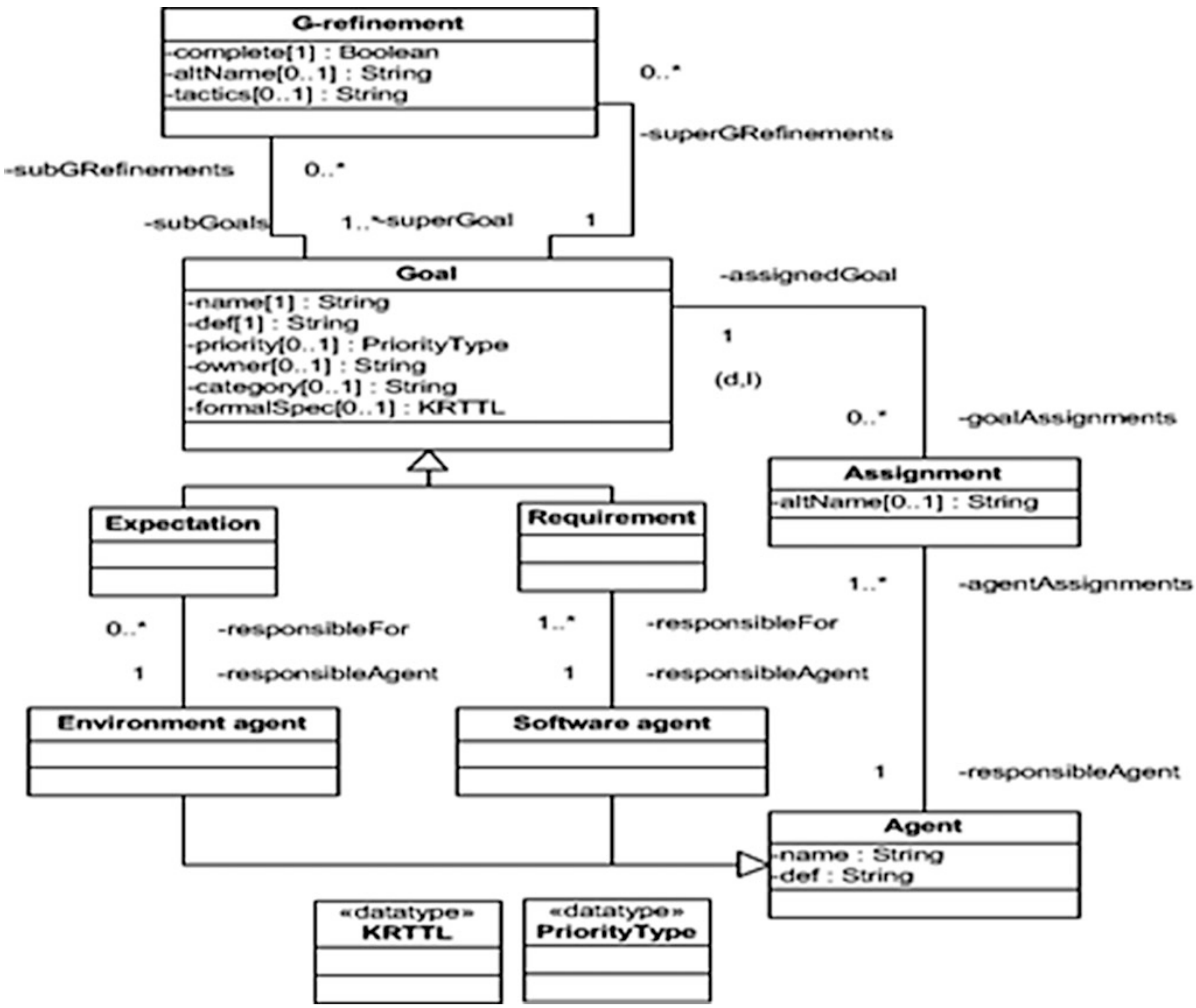

Fig. 2 A fragment of the KAOS metamodel. Source [7]

words associated with each lexeme. Each one is then compared with the Spanish empty word list which appears in [15]. Words which appear in both groups are stored in the {stopwords_encontradas} list.

Definition of the Domain Model

The analyst should define an Ontology which represents domain knowledge. This domain Ontology offers the possibility of identifying the classes relevant for the system. One should avoid describing the whole problem, taking as a starting point an Ontology which addresses the most generic concepts there may be in the domain. For example if a pizzeria is mentioned, then the Ontology could simply describe a restaurant and not necessarily a pizzeria. It must however take into account components such as employees, products or similar information.

Identifying Concepts Relevant for the Domain Model

With the aim of obtaining components which are truly relevant, one begins by supposing that if an agent is important, then it will be mentioned with some frequency. The "RA1" rule, presented later in this paper, carries out this selection. However to apply it we need to know the occurrence and frequency of each word. Therefore the set of words from the document is used and empty words eliminated; the lexemes are then chosen and the following calculations made: (i) total number of words Σ in the documents; (ii) Number of occurrences Oi of each word; (iii) frequency F of each word using the Eq. (1).

$$F = \frac{O_i}{\sum} \tag{1}$$

The calculations carried out with the lexemes should be taken into account in this process. For example if "controlador" then "controladores" appears anywhere in the text, this will constitute two occurrences, as the lexeme for both is "controlador". The candidate agents should however be nouns. In order to identify them the noun phrases for each sentence obtained by the tagging tool are taken and stored in the {conceptos} list. Components which do not relate to any action, i.e. which are not noun phrases, are not stored, as an agent should fulfill an action and is an active component [11]. In order to be able to compare the concepts and know, for each component on the {conceptos} list, if they are being linked with the system domain, a search for terms related semantically with the same concept should be made. To do this, lexical databases such as MultiWordNet [16] are used. This tool groups words into sets of synonyms, for hyperonymic word A is from the same type as B and hyponyms for multiple languages, including Spanish. For two words A and B, synonyms indicate that both A and B have a similar or identical meaning; for hyperonymic word A is from the same type as B; and for hyponyms word B is from the same type as A. This allows us to ascertain the semantic relationship of words, located within a hierarchy which helps the concepts to be categorized.

In accordance with the above, for each component on the {conceptos} list one looks for the hypernym and synonym in MultiWordNet and stores it in the {hiperónimos} and {sinónimos} lists respectively.

If nothing is found in this database, then another search is performed for each word separately, whenever it is not found on the {stopwords_encontradas} list. For example, as "personal de ambulancia" is not found, one then proceeds to examine "personal" and "ambulancia" (but not "de" as it is an empty word).

The above procedure should also be applied to find synonyms and hyperonymic for all of the Ontology's concepts, and to store them in the {sinónimos_de_la_ontología} and {hiperónimos_de_la_ontología} lists.

Then the differences between the {hiperónimos} groups for each concept on the {conceptos} list, and the {hiperónimos_de_la_ontología} group are calculated. For each concept, if there are more than three differences, and there is no match between the {sinónimos} and the {sinónimos_de_la_ontología}, one proceeds to eliminate them from the list, as it would indicate a large semantic gap between the concepts. In other words there is not much of a relation between that concept and those which are expressed in the domain Ontology, therefore this concept would not be relevant to the domain. Rules for identifying agents are listed below:

RA1: Eliminate infrequent and inactive words: If a concept occurs only once in the document, or has a frequency of less than 2 %, then it can be ignored. Moreover, if it is not followed by a verb then it will be the complement for a phrase, and not the subject, and therefore would not be specified as an agent. As a result it will be eliminated.

RA2: Identifying software Agents: If a concept is related to concepts such as "software", "sistema" or "dispositivo", then it is assigned to the {agentes-de-software} list. In order to do this the last levels of the hypernym list are checked, i.e. the most generic conceptual category which covers each concept, according to MultiWordNet. If the {hiperónimos} list for the concept in question contains any of the following terms {sistema, mtodo, código, equipo, equipamiento, instrumento, instrumental, utillaje, artefacto, cosa, objeto, objetofísico, objetoinanimado, entidad} then we can say that it is a candidate to be a software agent. The above list is the result of evaluating the "software", "sistema" and "dispositivo" hyperonymic (software agents may be automatic components, for example electronic devices which have some function within the system).

RA3: Identifying environment Agents: If a concept is related to people then it is assigned to the {agentes-de-ambiente} list. This arises from the affirmation made in [17] where it states that an agent is software or a person/people, and as we already have the agents for the software that is be developed, only the people that interact with it would remain. Naturally there may be software components from outside the system. These, despite being software, should be categorized as environment agents. However, supposing that the software under development is to interact exclusively with humans, it would true to say that any environment agent is referred to as a person. Therefore one searches within MultiWordNet for if each concept, in the last hyponyms categories, relates to "ser_vivo (human being)", "humano" (human), "agente" (agent), "agente_causal" (causal agent), "individuo" (individual), "organización" (organization), "grupo_social" (social group).

RA4: Defining Agents: Software agents, such as environment agents, are subtypes of Agent. It would therefore be necessary to specify their name and definition. The name is taken from the concept list, i.e. the names of the agents identified in the system are each one of the components which make up the {agentes-de-software} and {agentes-de-ambiente} lists. Fig. 3 shows, in general terms, the results of identifying the agents from natural language (Spanish).

Fig. 3 Example of agent identification. Source: the authors

Expression in Natural Language	Type of Identified Agent		KAOS Graphical Representation
	Environment	Software	
El personal de ambulancia envia información.	personal de ambulancia		personal de ambulancia
El sistema CAD puede proponer otra ambulancia.		sistema CAD	sistema CAD

Table 1 Grammatical elements. Source: [18]

Grammatical elements used in this proposal	
Name	Definition
Noun Phrase (NP)	Phrase whose nucleus is a noun, thus allowing it to perform any of the functions of a noun in a sentence
Noun	It is the word that acts as the subject of a sentence. It has gender and number
Subject	In a sentence, the subject is the person, animal or thing that performs the action expressed by the verb

Entities and Attributes Identification

Process for obtaining entities and attributes from natural language (Spanish).

It is possible to use natural language as the starting point for the characterization of the entities and attributes of the KAOS goal diagram. Moreover, since the input is natural language, namely Spanish, it is necessary to define the following grammatical elements (see Table 1) in order to obtain said KAOS components. Additionally, it is worth noting that these grammatical elements are used in literature and referenced in [18].

Based on the previous definitions the grammatical rules are presented below. These rules allow the entities and their attributes to be obtained from natural language.

R1: if in a phrase there are two nouns found joined by the prepositions “de” or “del”, then the second noun will be an entity, and the first will be a candidate to be the attribute for the previously identified entity.

R2: if in a phrase there are two nouns found joined by the verb “tiene”, then the first noun will be an entity, and the second noun will be a candidate to be an attribute for the previously identified entity.

R3: if a noun or noun phrase is identified as an entity and as a possible attribute, then the entity takes priority.

Experiments

With the aim of automatically extracting the KAOS methodology components (entities and agents) from the universe of discourse, characterized through the rules presented in the previous chapter, the NL2KAOS software tool has been developed in this proposal (in JAVA and PHP). The following IT tools were also used for this IT solution:

(i) Freeling 3.0 [12]: This allows natural language texts to be processed. The SpanishPlugin module was used for this process, the role of which is to process documents in Spanish.

(ii) MultiWordNet [15]: This is a lexical database of semantic relations between terms, such as hyperonymic, hyponyms and synonyms.

In order to carry out the experiment the case study entitled, “London Ambulance service system”, adapted from [5], was used.

The validation tests demonstrate that, using the automatic proposal presented in this paper, it was possible to identify the same amount of entities and attributes, in total 9 (nine) and 10 (ten) respectively, in relation to those identified manually. This leads us to conclude that, through this proposal, it was possible to comply with the analysis of the completeness properties (number of elements) and consistency (the diagram is consistent with the specification) in less time, given that the manual process took 7 min while the automatic process took only 16 s. Fig. 4 shows a portion of entities and attributes identified.

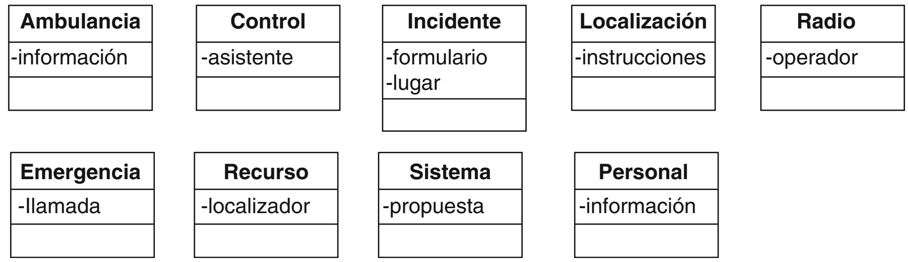

Fig. 4 Entities and attributes identified using the automatic proposal presented in this paper. Source: the authors

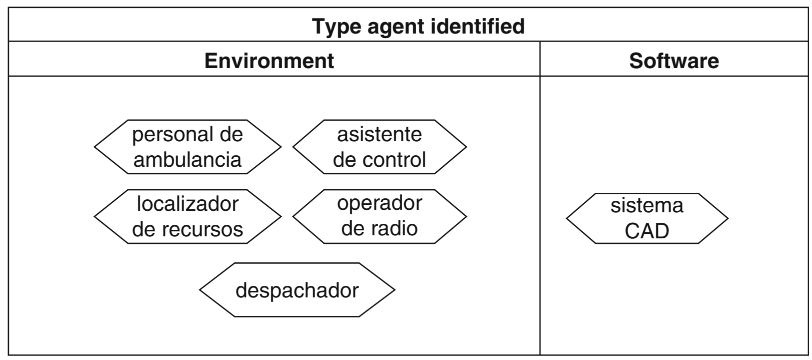

Fig. 5 Results—automatically obtained agents. Source: the authors

Six agents were found in the manual process (carried out by the agent) using the proposed rules. Using the automated proposal presented in this paper one was able to obtain the same amount of agents with relation to those identified manually. This leads us to conclude that, in this proposal, one was able to comply with: (i) the completeness characteristics with relation to the number of components that were identified; (ii) the consistency (the diagram is consistent with the specification), i.e. the traceability that should exist between the natural language and the KAOS components. These were identified in less time given that manual process took 11 min, whereas the automated process took 16 s. The results can be seen in Fig. 5.

Conclusion and Future Work

In order to provide a solution to the need to identify the traceability that should exist between the natural language and the entities, the natural language and agents (from the KAOS goal diagram) and its respective attributes, 3 (three) grammatical rules were defined which allow for the characterization of entities with their respective attributes based on natural language, and 4 (four) grammatical rules were defined which allow for the characterization of agents with their respective attributes based on natural language.

Likewise the Freeling 3.0 and Multiwordnet tools were used in order to automatically extract the elements that have been characterized using the rules presented in this paper from the universe of discourse. Finally a case study is presented which allows the proposal to be validated. The validation tests demonstrate that, using the automatic proposal presented in this paper, it was possible to identify all entities and all agents, in total 9 (nine) entities and 6 (six) agents respectively, in relation to those identified manually. This leads us to conclude that, through this proposal, it was possible to comply with the analysis of the completeness properties (number of elements) and consistency (the diagram is consistent with the specification) in less time, given that the manual process took 7 min average while the automatic process took only 16 s.

This paper has given rise to new areas of study which may provide continuity to this research. Some of these are listed below: (i) defining a new set of grammatical rules which allow the other components (expectations, domain properties and connectors, etc.) which make up the KAOS goal diagram to be identified; (ii) characterizing a new set of rules which can generate variations in the components (entities and agents) identified in this paper; and (iii) resolving the anaphora resolution.

Acknowledgment This paper is part of the research Project entitled "A Terminological Processing Model for Obtaining Software Requirements Based on the KAOS Goal Diagram". The project's code is 202010011022, and it is sponsored by the Research Directorate of Universidad Nacional de Colombia—Sede Medellín (DIME) through the "DIME 2012 Research Project Financing" program.

The authors thank specially to Systems Engineering student Carlos Andrs Vlez Carvajal for the unconditional support to the project "A Terminological Processing Model for Obtaining Software Requirements Based on the KAOS Goal Diagram".

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A New Methodology Based on Cloud Computing for Efficient Virus Detection

Vasileios A. Memos and Kostas E. Psannis

Abstract

Antivirus software programs use specific techniques to detect computer viruses, malware and other network threats. The basic, most common and oldest antivirus detection technique is "virus signature scanning", whereby antivirus programs use unique byte sequences for each virus so as to identify potential presence of malicious code in each file investigation procedure. Despite its advantages, this technique has many weaknesses that are highlighted in this paper. In lieu, this paper proposes a new hybrid security model for optimized protection and better virus detection, which merges the "Sandboxing Method", "System-Changes-based Signatures" and "Cloud Computing".

Keywords

Antivirus techniques evaluation • Cloud technology • Sandboxing method • System-changes-based signatures

Introduction

Today viruses and other malicious software—malware—have increased dramatically and spread rapidly every day. In addition, new unknown malware, known as zero-day threats, appear day by day and in combination with advanced virus concealment methods that are used by sophisticated virus programmers make the work of antivirus software very difficult. The current protection methods which antivirus vendors use are not adequate to solve these problems and produce many false positives. So, they should develop new methods and techniques for their software for more efficient malware detection. The main reason for these antivirus weaknesses is that they are based on "virus signatures" which consist of specific byte sequences to identify malicious code [1].

This problem urged us to make this research to prove the main problems of this technique and suggest a new security model that will not be based on specific byte sequences, but on signatures of system changes which malicious processes cause, in combination with some innovative techniques, such as sandboxing [2] and cloud technology [3] which have been used over the last years in many antivirus programs. In this research, we use a malicious file—Trojan horse type—which is transmitted via social networks, such as Facebook, to make a series of tests to prove the above antivirus weaknesses. We also use a file splitter to divide the Trojan to smaller files, an antivirus program for scans, the windows Command Line and a clear system file. The paper is organized as follows: In section "Related Work", we cite all the related work about the problem of signature scanning detection method. In section "Problem Definition", we extensively analyze the problems of this method. In section "Experiments", we prove these problems with a series of tests. Section "Proposed Approach: Methodology" includes our proposed methodology for better virus detection and more efficient protection against network attacks. Section "Conclusion" concludes the paper.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_6, © Springer International Publishing Switzerland 2015

Related Work

Nowadays antivirus programs are quite advanced and identify viruses by using various static and dynamic techniques, but these techniques are not sufficient. Virus programmers are usually a step ahead of antivirus programs, because they use many techniques, such as space filling, compressing and encryption, to make their viruses avoid the detection by antivirus scanners. In addition, new viruses appear every day, which current antivirus programs cannot identify immediately, because of the fact that they are mainly based on virus detection by using string signatures [4, 5]. In 2006, Seon Yoo and Ulrich Ultes-Nitsche presented a non-signature-based virus detection approach, using Self-Organizing Maps (SOMs). In 2008, Essam Al Daoud, Iqbal H. Jebril and Belal Zaqaibeh indicate the need of a new model that could be able to detect all metamorphic virus variants by using new trusty monitoring techniques by attaching a digital signature and certificate to each new program [6]. Unlike the traditional detection methods, this SOM-based approach would detect infected files without virus signatures. In their tests on 790 different infected files (including polymorphic and encrypted viruses), their model detected 84 % of them, with a false positive of 30 % [7]. In addition, in 2009, Min Feng and Rajiv Gupta, developed a new algorithm for matching a new type of virus signatures—dynamic signatures—which was based on the run-time behavior of a piece of malware. Tests proved that this algorithm was very effective in recognizing different variants in a malware family with a single dynamic signature and without false positives [8]. In 2011, Madhu K. Shankarapani, Subbu Ramamoorthy, Ram S. Movva and Srinivas Mukkamala developed algorithms which could detect known malware without the use of signatures [9].

Problem Definition

Antivirus vendors have developed many methods to improve the protection that their software provides to users. The most common and oldest technique is "virus signature scanning" that antivirus software uses unique byte sequences as a signature for each virus to recognize it. The following Section includes detailed description of this technique.

This signature-based detection model has many advantages, such as the low memory and system resources that it needs and the speed of scanning lots of files per second [10], but it has inevitably many weaknesses, too. It is dangerously easy for sophisticated hackers to change a virus signature and make viruses undetectable by the antivirus programs. In addition, virus signatures dramatically increase the false positive rate of antivirus scanners [11]. False positives can also cause major problems to the system operation. This happens because new viruses and malware are discovered daily, so there are lots of signatures and, in consideration of the amount of files that exist, a signature is often difficult to be absolutely unique.

In addition, antivirus virus database should be constantly updated to remain reliable [12]. However, new viruses are discovered per second [13], "0-days threats" as they are known, so virus identification should be based on other detection techniques too, which would not examine the binary code of the files to identify specific sequences of bytes, but would try to guess the behavior of each file. One such method is "heuristics" which is used additionally by many antivirus programs and are based on looking up for suspicious instruction sequences that may be related to malware existence. Although this method provides better detection capabilities, it gives a lot of false positives when it is adjusted for maximum detection rate [14, 15]. And of course, such behavior detection methods take up many system resources, so they slow down computers' operation and overload networks.

Virus programmers are usually one step ahead of antivirus programs, because of the fact that they first launch malware to attack their targets and the only thing that antivirus vendors can do is to find and defend against these attacks by finding their antidote to launch it as virus definition file update to protect users' computers. Note that often a considerably long time is needed to manage it.

In addition, hackers use many mutation and encryption methods to conceal their viruses and bypass antivirus scanners, such as Stealth technique [16], Self Modification technique [17], Virus Encryption technique (with a variable key) [9], Polymorphic technique [18], Metamorphic technique [18] and Avoiding Bait-Files Technique [19].

All the above problems are considerable vulnerabilities of antivirus software, so new techniques should be discovered to deal with them for better detection rate and lower false positives frequency.

Experiments

This section includes three tests that present the main weaknesses of virus signature method based on string scanning by using a known methodology which describes the way antivirus programs operate. This section is organized as follows: In section "Finding Virus Signature" we use some tools to find the virus signature of a malicious file. In section "The Problem of False Positives", we present a test to show the problem of false positives that are caused by using this method. Section "Virus Concealment" includes another test which indicates the problem of virus concealment.

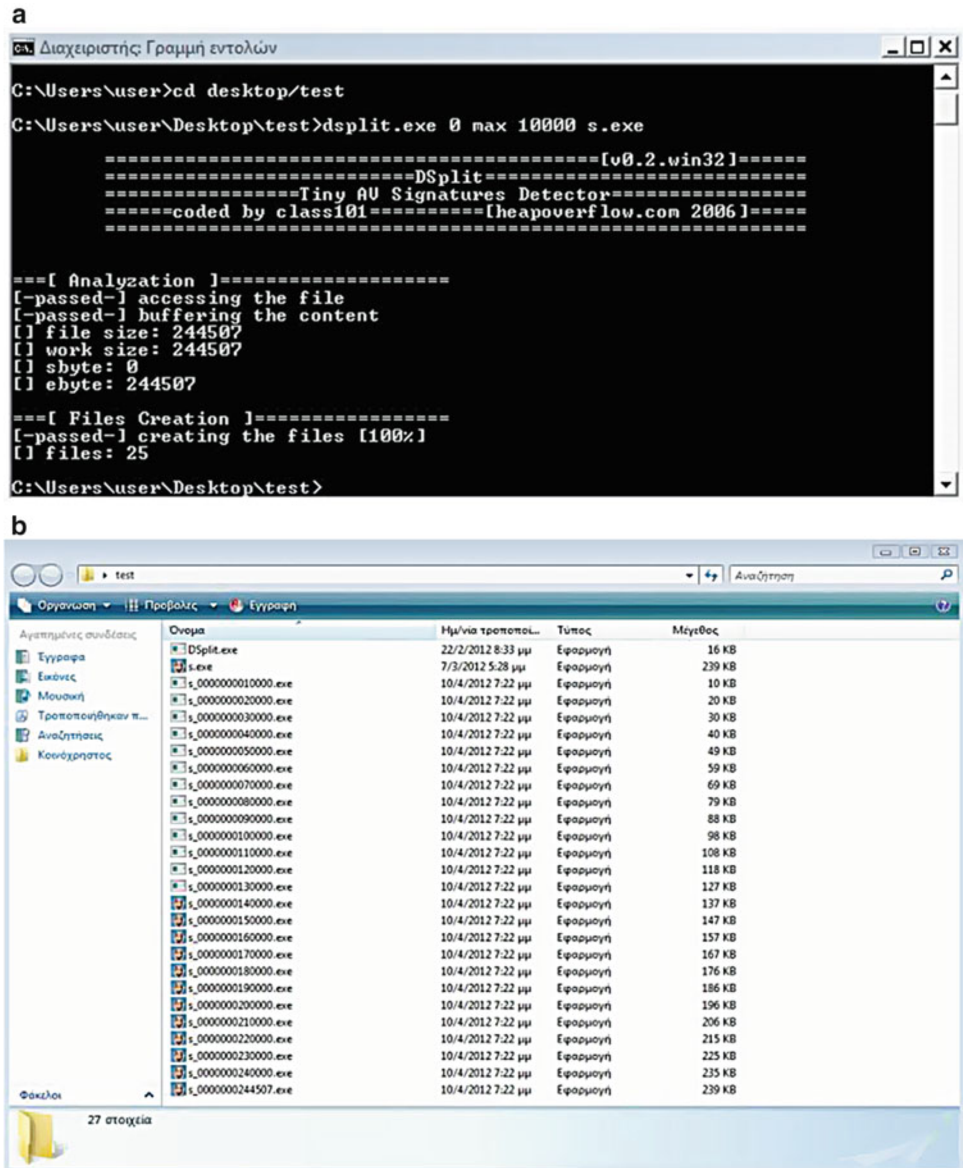


Fig. 1 The initial dsplit command (**a**) which divides the s.exe file in 25 new files (**b**)

Finding Virus Signature

In this test we study and test the malicious file "*s*", 238 KB size, which looks like a picture, but it is an executable file (*s.exe*) which is proved by clearing the "Hide extensions for known file types" check box in the windows folder options control panel. It is actually a Trojan horse and it is transmitted via social networks, such as Facebook. In this test we try to find its virus signature that has been used by an antivirus scanner to identify this threat. Note that each antivirus program uses a different signature to identify a certain threat.

For this test we used four tools: the Windows Command Line, DSplit file splitter, HxD hex-editor program and Avira antivirus as an on-demand scanner for our scans.

Firstly, we create a "test" folder on the computer desktop and put into "s.exe" and "dsplit.exe" files. We type *cd desktop/test* to gain access to the created folder and then give: *dsplit.exe 0 max 10000 s.exe*, as it is depicted in Fig. 1a. This command divides *s.exe* file to smaller ones so that each file is 10,000 bytes larger than its previous one and 10,000 bytes smaller than its next one. Figure 1b depicts the result of the above command with the 25 new created files.

We now scan the 25 new created files with Avira antivirus and as it is depicted in Fig. 2, Avira detects as "TR/Offend.kdv.49932" the last 16 files, from the 100 KB file (*s_0000000100000.exe*) to the final one. This actually means that some of the added bytes into the 90 KB file make the 100 KB one to be detected as malware. To find which of the 100 added KB make the difference, we delete all the created files except these two files.

The new command now is: *dsplit.exe 90000 100000 1000 s.exe*, which creates new files between 90 and 100 KB ones,

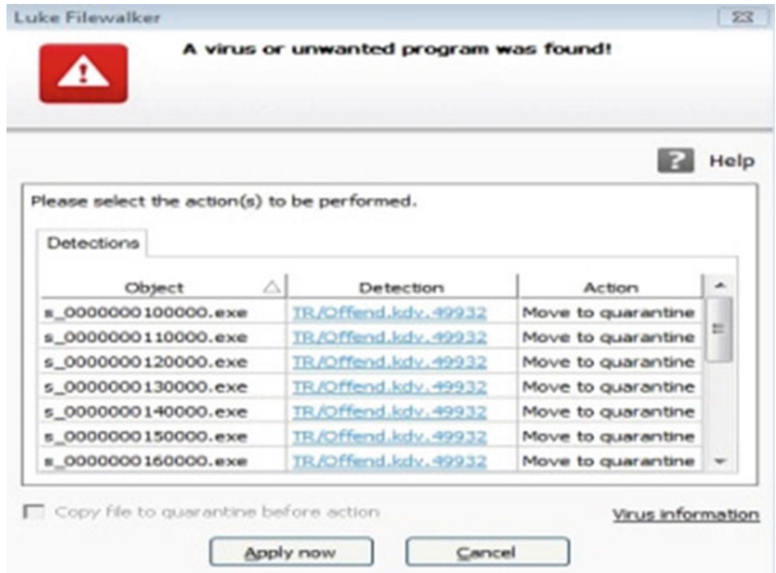


Fig. 2 Avira antivirus scanner detects 16/25 files, from the 100 KB file and all files after it

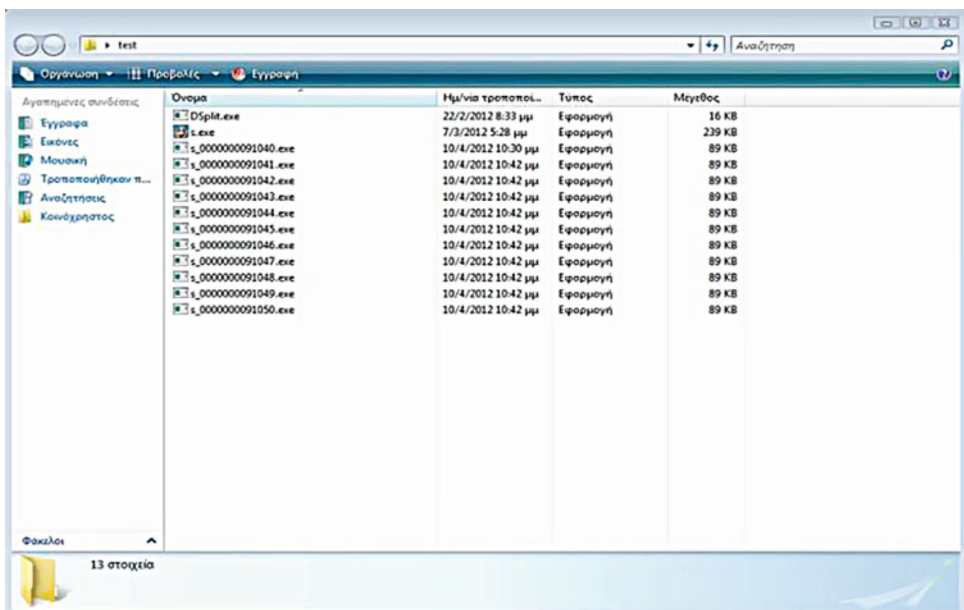


Fig. 3 All files have only 1B difference between each other

so that each of them is 1,000 bytes larger than its previous one and 1,000 bytes smaller than its next one. We scan these new files with Avira and observe that 9/11 files detected as Trojan horse, from 92 KB file onwards. It means that there is something in 92 KB file that is not in the previous 91 KB one.

The process is a routine and as previously we delete all other files and hold these two files and type the proper command, which is now: *dsplit.exe 91000 92000 100 s.exe*. By repeating the above steps, it will be needed to give the commands *dsplit.exe 91000 91100 10 s.exe* and *dsplit.exe 91040 91050 1 s.exe* too. Thus, we will manage to have files with only 1 byte difference between each other. Figure 3 depicts this final step of this loop, where there are 11 created files. By scanning these files again with Avira antivirus, we see only one file detected as Trojan, the 91050B one, as it is depicted in Fig. 4. It means that it has one more byte than the previous 91049B file, which causes the detection by the antivirus scanner.

We now use HxD hex-editor to open these two files and compare their hexadecimal codes, as it is depicted in Fig. 5.

We note at the end of the codes that this byte which is added and causes the Trojan detection is the hexadecimal *41* or the string *A*. So, we just find the end of the string signature which Avira antivirus use to identify this Trojan horse. To find the start of the signature we need a clear—non-malicious—file. We use *taskmgr.exe* system file, which is the executable part of the Windows Task Manager.

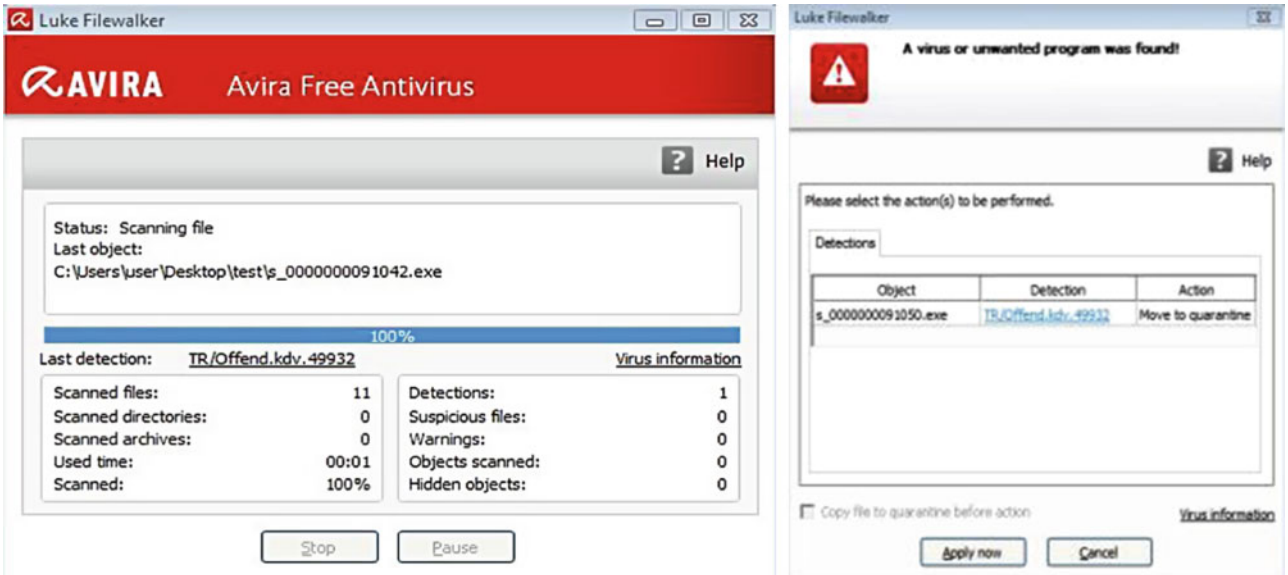


Fig. 4 Avira detects only one of the files

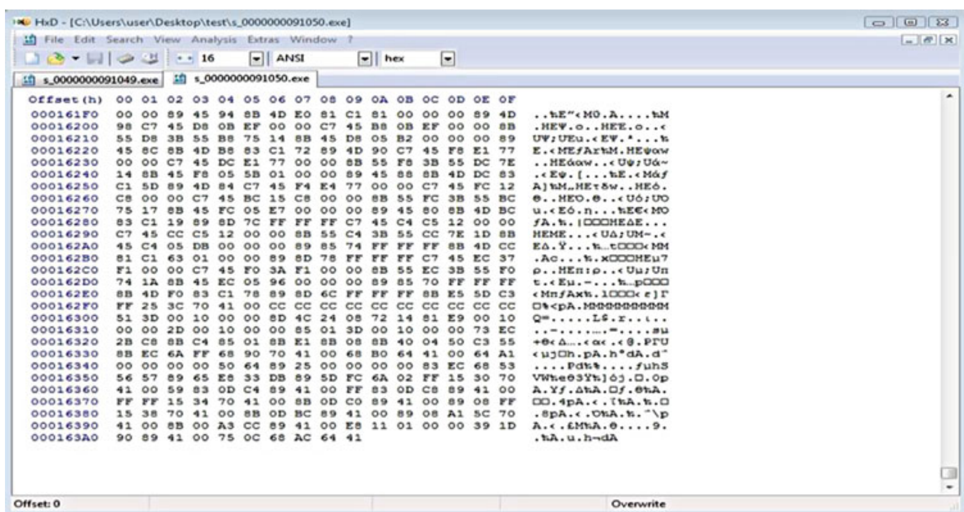


Fig. 5 HxD editor with hexadecimal code of '91049' file and '91050' one

We open *taskmgr.exe* file with HxD, too, and try copying some of the last bytes from the 91050B file into it. Then, we scan it with Avira antivirus. It is a routine again by adding byte-to-byte from the detected file to this clear one, until we can manage to make this clear file detectable as a Trojan, too.

A virus signature must be long enough to generate as fewest false positives as possible. So, choosing a short signature, such as: *55 8B EC 6A FF*, is not a good solution. This is because of the fact that such short byte sequence is possible to be present in many other files that are not malicious. That is why the length should be considerably long, more than 50 bytes [20]. By continuous tests we find and depict in Fig. 6 that Avira uses in this case a 123-byte length signature to detect the *s.exe* file as "TR/Offend.kdv.49932", which is consisted of the following:

55 8B EC 6A FF 68 90 70 41 00 68 B0 64 41 00 64 A1 00 00 00 00 50 64 89 25 00 00 00 00 83 EC 68 53 56 57 89 65 E8 33 DB 89 5D FC 6A 02 FF 15 30 70 41 00 59 83 0D C4 89 41 00 FF 83 0D C8 89 41 00 FF FF 15 34 70 41 00 8B 0D C0 89 41 00 89 08 FF 15 38 70 41 00 8B 0D BC 89 41 00 89 08 A1 5C 70 41 00 8B 00 A3 CC 89 41 00 E8 11 01 00 00 39 1D 90 89 41 00 75 0C 68 AC 64 41

The Problem of False Positives

In the previous section, in our try to find the virus signature of the malicious file, we make the clear *taskmgr.exe* file to be detected by the antivirus as a Trojan. But is this file actually "TR/Offend.kdv.49932" as Avira says? The

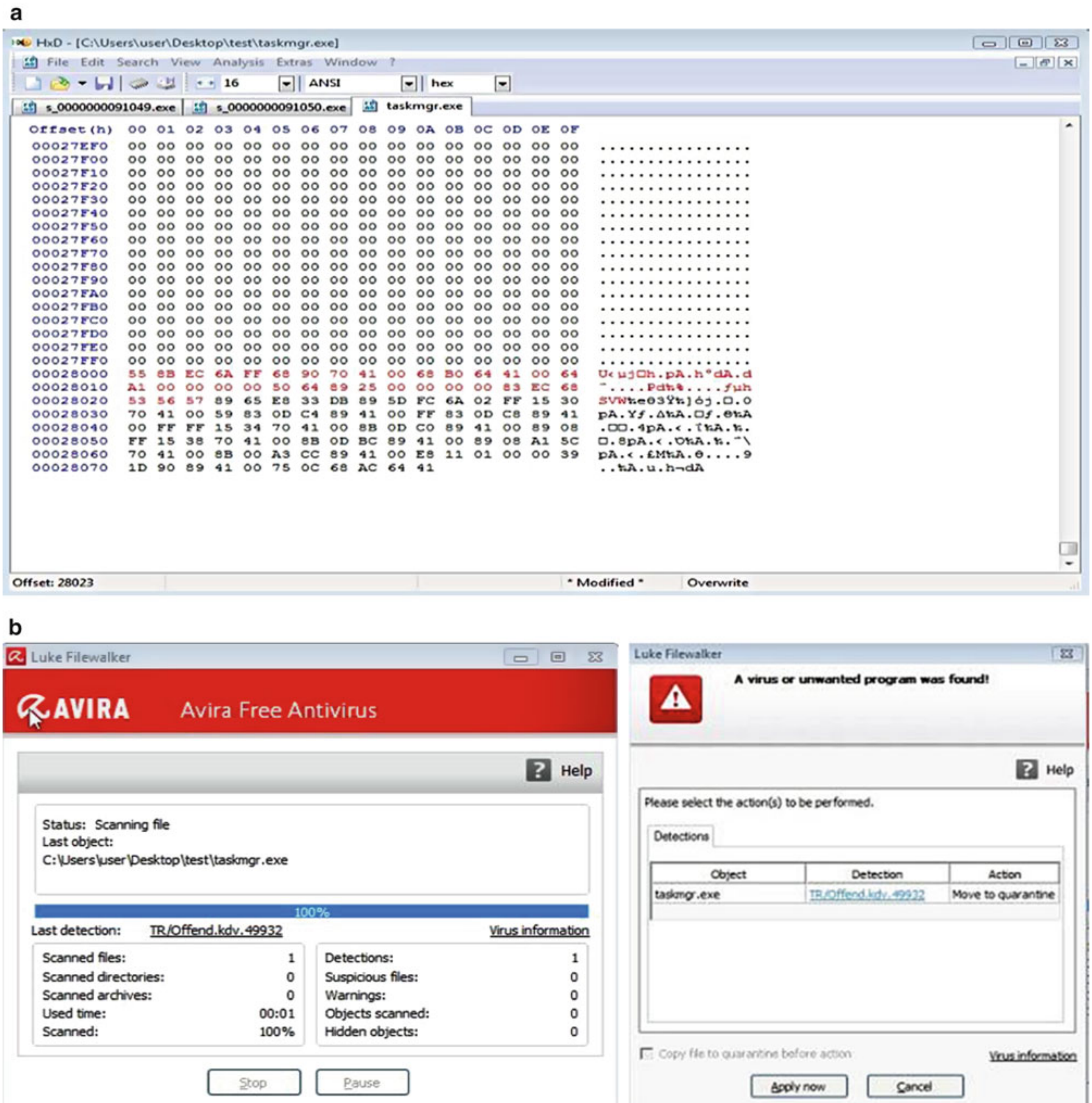


Fig. 6 The signature that is added in clear file taskmgr.exe (**a**) and makes it detected falsely as Trojan by Avira (**b**)

answer is no. The 123 additional bytes in the original code of the *taskmgr.exe* are no more than a useless code and it neither affect the operation of the task manager file, nor makes it malicious. This conclusion can be verified by uploading the modified *taskmgr.exe* file to VirusTotal site, which includes 42 online antivirus scanners. As it is depicted in Fig. 7, only Avira finds this file as malicious. It is a false positive by Avira, because of the 123B sequence existence inside the *taskmgr.exe* code. Same false positives we would also have if we made our tests with other antivirus scanners.

The above problem is more important than it seems, because of the fact that Avira and each other antivirus often recommends user to set this file to "quarantine" by isolating and making it inactive or to "delete" it, when it is unable to "repair" it. But system files are important and necessary for the proper system operation, and therefore this action will cause inconstancy to the system, making it unbootable or crash.

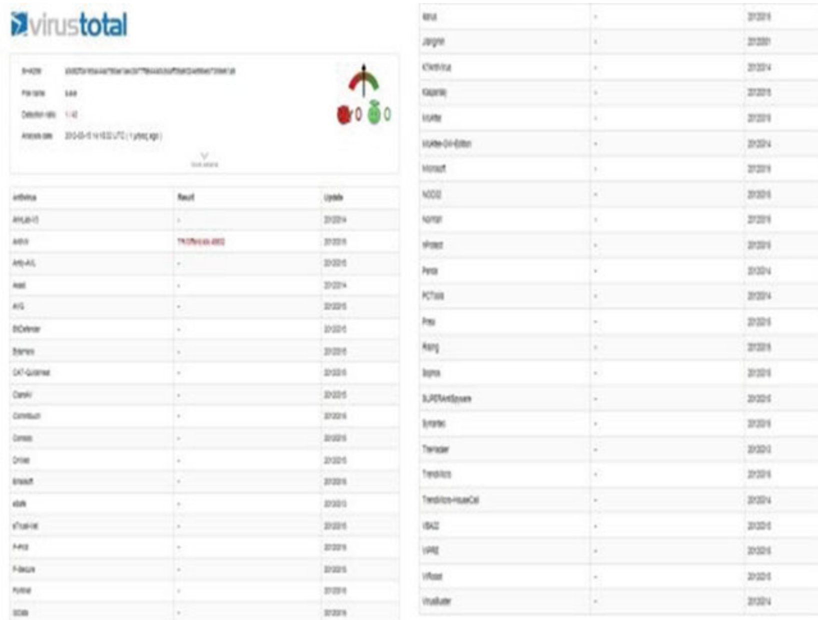


Fig. 7 Detection results of 42 antivirus by uploading the new taskmgr.exe file in VirusTotal site

Virus Concealment

In this section, we present a simple way to conceal a virus from detection by antivirus programs. To manage it, we need to change properly the bytes sequence which antivirus use as virus signature to identify—in our example—*s.exe* file as Trojan horse variant, without change its malicious operation.

Figure 8a depicts the highlighting of the signature we found in the previous section in the original *s.exe* malicious file.

By changing properly an alphabet string, such as “U” to “u” (55–75 in hexadecimal code), signature will change, but not the file’s destructive function. If we scan the new changed s.exe file now, we will observe that Avira does not detect it as Trojan (Fig. 8b).

Note that this process is only a simple, but important example to indicate how easily a virus can be concealed from an antivirus scanner. The problem is much more serious if we consider that there are polymorphic viruses too, whose code is self-replicated and self-concealed. In addition, there are many specialized programs—crypters—that use various complicated encryption methods to make a virus file non-detectable, easily and quickly. These tools are programmed and released by hackers over the Internet.

Proposed Approach: Methodology

Model Description

As we describe and analyze with experiments in the previous Sections, the already existing methods which antivirus software use, have many weaknesses. For optimized virus protection, we propose the development of a new hybrid security model that will merge “Cloud Computing” and “Sandboxing Technology”—methods which are already implemented gradually in some antivirus programs—with a new creation methodology of Virus Signatures that will be based on “System-Changes” which a malware causes to the infected computer.

This new cloud-based antivirus software will be used as a client and will monitor in real-time the activity of computer files in the cloud, such as any changes they cause to hard disk, system directory, registry etc, and if it recognizes malicious changes, they will not become permanent, thanks to the sandbox. If the changes are not clearly malicious, the antivirus will alert the user to decide whether the suspicious file will continue to run sandboxed or outside the sandbox feature. For a better performance, cloud technology would come in handy to contribute again, as the user will have the access to see in real-time collecting statistics, marks and comments by other users connected to the cloud for the suspicious file activity.

General Description of the Used Methods

Sandboxing Technology A sandbox provides an additional protection level against infection by harmful viruses. It is an entirely isolated environment which the programs that are running inside it have not access to the local computer drivers, so there is no risk for the system files. So, if the user runs a file that is malicious, the changes it will cause to the system will not be permanent. In this case, the installed

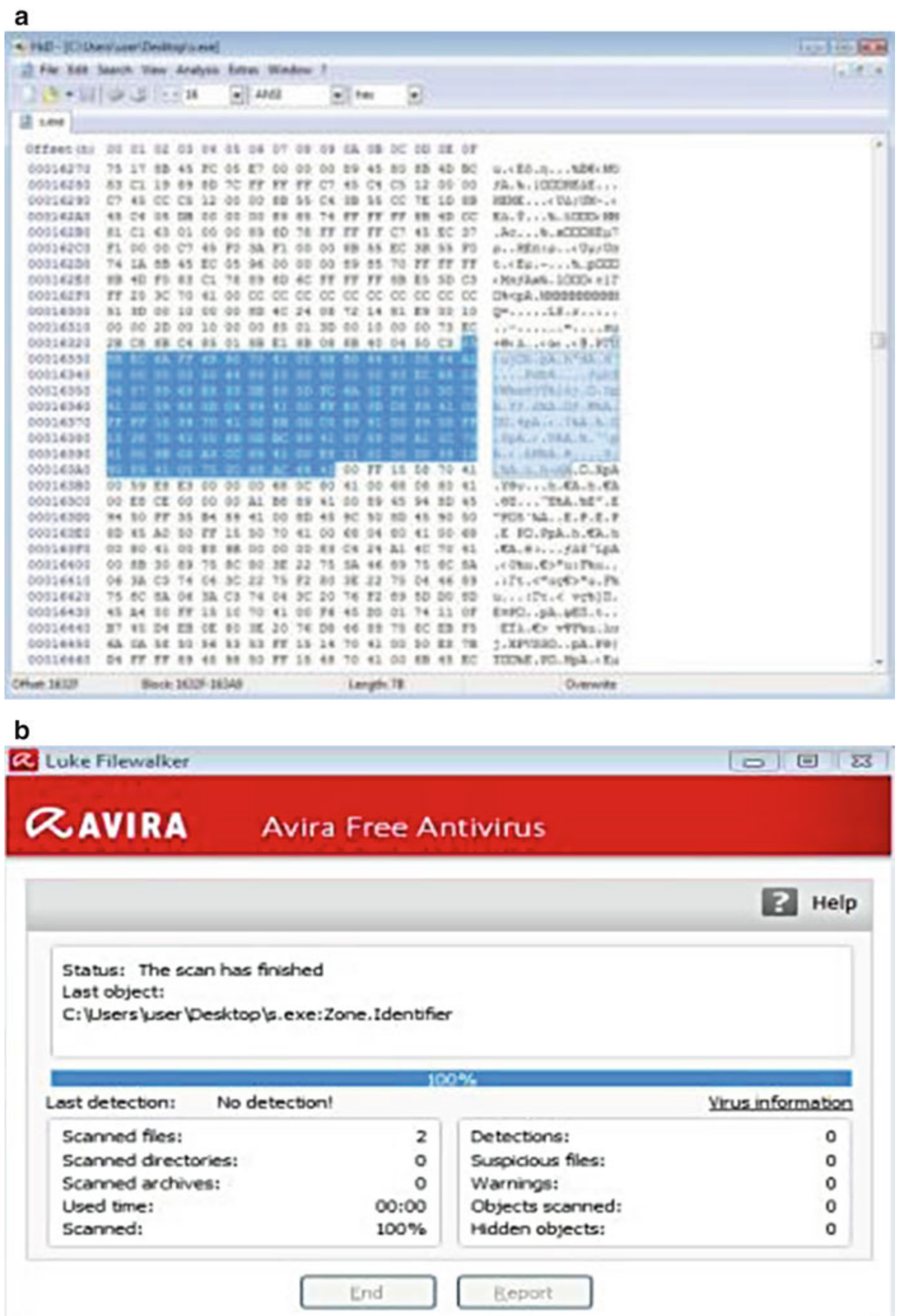


Fig. 8 The finding and change of the Trojan signature (**a**) and the scan process and scan results by Avira (**b**)

antivirus program blocks the malicious process of the file and the system resets to its previous mode [2].

System-Changes-Based Virus Signatures As we proved in the previous section, the "string-based" signature detection method has many weaknesses. Signatures change dangerously easily, making viruses undetectable and also this method has many false positives. In addition, new unknown malware are discovered, so it is necessary to change the way that signatures are created. This new method will not be based on specific byte strings, but on changes which malicious files make on the system computer. This technique is similar to heuristics; the main difference, however, is that each malicious behavior will have its different signature. So, in this new method the antivirus scanner will not look up for specific "bytes sequences", but for specific "system-changes sequences" to recognize viruses.

Cloud Computing Technology Cloud computing is a new approach for quicker response to new threats, which spread constantly on the internet. This architecture is based on communication between servers that are somewhere on the Internet—"cloud"—and computers that are connected to this "cloud". The connected computers have installed a small program—in our case the antivirus program—that is used as a client. Most processes of the program take part in the connected server by the web service, which is running in the cloud. Thus, the computer does not need to process and store a large amount of data—in our case, virus signatures. At regular intervals, the client automatically scans the computer for virus existence, using the information it takes from the web service's database of the cloud server that it's connected to. In addition, because of the fact that web service runs in the cloud and not locally in the computer, cloud antivirus programs consume low memory and system resources, without supercharge the computer even if it is not meet the minimum requirements that have the most current computers [21].

By using cloud technology, antivirus analysts can collect easily and quickly all the suspicious processes from users' computers to analyze them. So, if they find malicious process, they launch the suitable antidote signature in the cloud, so automatically within minutes, all the computers that are connected to it, are protected by this threat, without the need of downloading by the antivirus program often in a day large maybe patches to stay protected from the latest malware that spreads rapidly via the Internet all over the world [21].

Generally, the cloud method requires a 24/7 Internet connection, but it is not indispensably needed. Cloud antivirus always keeps a cache of malware information on the local computer for offline access, too. This cache doesn't include all the virus database of the cloud server, but only the basic and most common threat signatures, and it is updated every time it finds an Internet connection [21].

Proposed Model Analysis

As it is depicted in Fig. 9, our proposed model is a hybrid security system that will consist of:

m cloud virtual sub-servers that constitute the Home Cloud Server

n terminals connected to *m* cloud virtual sub-servers

i sandboxes contained in *n* terminals

k files inserted into *n* terminals

l virus signatures that are collected and contained to the Home Cloud Server's Virus Database

l virus signatures that are transmitted from the Home Cloud Server's Virus Database to *m* cloud virtual sub-servers' Virus Databases, where: $i = 1, 2 \ldots, n = 1, 2 \ldots, m = 1, 2 \ldots, k = 1, 2 \ldots, l = 1, 2 \ldots, i = n$.

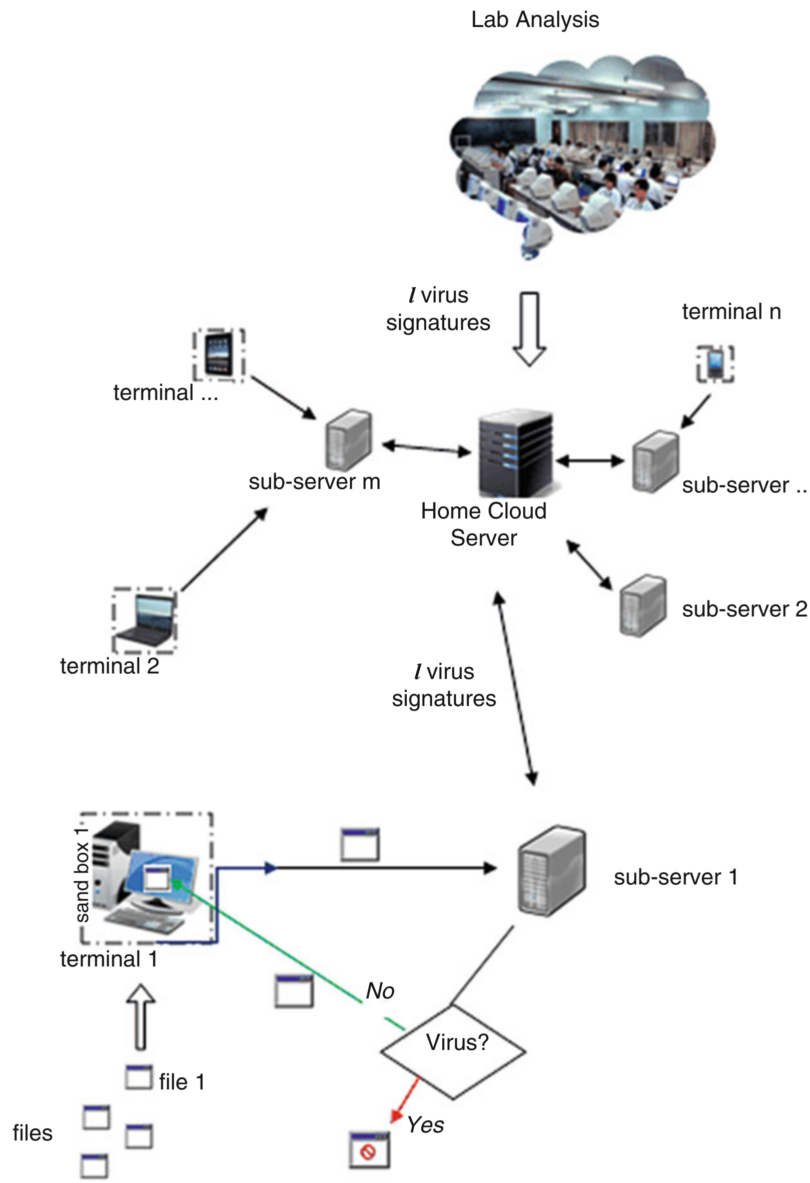
Example In the Home Cloud Server Virus Database, there have been collected all the present *l virus signatures* from the specialized analysts who work in the cloud. *L virus signatures* transmitted from Home Cloud Server Virus Database to *sub-server 1* (virus database) and each other sub-server, connected to Home Cloud Server. So, *sub-server 1* database has *l virus signatures*. *File 1* is inserted to *terminal 1* (e.g. by downloading). Firstly, the file is running in the terminal's 1 sandbox—*sandbox 1*—in the background, while antivirus (client) of *terminal 1* communicate with a cloud virtual sub-server of the Home Cloud Server, *sub-server 1*. A deep analysis takes part in the cloud *sub-server 1* to ascertain if the changes that cause the *file 1* inside the *sandbox 1* of the *terminal 1* match to any signature of the *l virus signatures* contained in the *sub-server 1* virus database. If they match to a virus signature, the process of the *file 1* is automatically stopped and the file is blocked. If they do not match to any signature, the *file 1* continues to run, but outside the *sandbox 1* therefore now.

Conclusion

In this paper, we have analyzed the problems of traditional antivirus software which are mainly based on the string signature detection technique. This method presents major problems, such as many false positives, and thus constitutes an easy method for specialized hackers to fool and bypass. The proposed method merges System-Changes-based Virus Signatures, Cloud Computing and Sandboxing techniques. This new hybrid security model will not be based on string signatures—specific byte sequences—for virus identification, but on specific system-changes sequences that will be made by malicious processes to the computer, in combination with the two innovative technologies, cloud technology and sandboxing method. The former is applicable for faster detection of new, unknown types of malware and lighter antivirus software, whereas the latter is aimed at offering enhanced computer protection by running malicious processes in an isolated virtual environment until the their functions are verified. Future work will include deep a study and analysis of the capability to adjust to the new security model in four areas: (a) capability of creation signatures based on specific system-changes, (b) capability of improvement the response time of sandbox mode for action in collaboration with the cloud server, (c) the user's data that will be collected for analysis by the cloud server, (d) how cloud technology architecture could be redesigned to eliminate the high false positive rate problem as far as possible. This is also a prospective research direction.

Acknowledgement This work was supported in part by the Research Committee of the University of Macedonia, Greece, under grant 80749 for the advance of Basic Research.

Fig. 9 The proposed model



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New Generation Android Operating System-Based Mobile Application: RSS/News Reader

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Abstract

RSS (Rich Site Summary)/News Reader is a web-based Android OS application developed by using PhoneGap framework. HTML5, CSS and JavaScript are basically used for implementation, instead of native Android programming language. This application has a production process like a web application because it is actually a fully working web program which is wrapped by PhoneGap framework. This means the application could be used on almost every mobile platform with making some basic arrangements.

RSS/News Reader mobile application takes advantage of both flexibility of web design and built-in features of the device it is installed. This combination provides a complete mobile application which eliminates the need to use different native languages with its hybrid form. This hybrid structure makes mobile programming faster and easier to implement.

In this new generation operating system-based mobile application, a combination of PhoneGap framework, HTML5, CSS3, JavaScript, jQuery Mobile, Python and Django is used for implementation.

Keywords

Mobile application development • Software architecture

Introduction

Technology continues to evolve rapidly, especially in mobile telephone development and production technology. There are many companies in the mobile telephone market such as; Nokia, Blackberry, Samsung, HTC, Sony, Apple, etc. In the last half-decade, smartphones and tablet PCs have become the main actors of technology. It could be said mobile software sector has surpassed non-mobile a major role in these developments during mobilization period.

With these important developments, computing is not only considered as a work on PCs and mainframes anymore. Operating systems are not only limited to Windows, Mac and UNIX. Single target platform application development can be admitted as a lack of concept. Since digital mobile platforms not only consist of PCs and but also a wide range of tablets, handhelds, and smartphones; the focus has shifted to mobile application development.

Developers aim to reach all these various types of platforms and on the other hand they try to reduce development and maintenance costs while producing consistent software on each counterpart platform. Developing applications on different platforms is a challenging issue because all these separate platforms need different type of requirements and development environments [1].

The basic concept of PhoneGap framework comes from the idea of coding software once and run it on different platforms. This only idea holds multiple solutions for problems caused by cross-platform development needs.

As well as mobile platforms have diversified, they met with the daily needs of platform users. Many people require

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_7, © Springer International Publishing Switzerland 2015

reaching different types of information with their various types of needs. Online data constitute a significant part of this information flow. Diversity of information creates a necessity to browse high numbers of online sources to get useful data. This may cause a certain amount of time and energy loss.

Starting point of RSS technology comes from this issue. Visiting each source everyday may be tiring and time consuming. Therefore collecting these sources on a single platform can be very useful. With RSS technology, readers can collect information sources by using a reader application.

With mobile revolution, people want to keep all these mentioned platforms in their pockets and utilize these small information sources with maximum usability. By this demand, interfaces of applications have become very important factor to be preferred by users. We analyzed this reality therefore usability and integrity of websites were our main design and development goals. In this way, we decided to build an RSS reader application on Android Platform [2, 3].

Android platform is preferred as a development environment because of its universality, open-source support, and large scale application distribution facility [4]. Android platform allows us to deploy application and reach users rapidly with advantages mentioned before.

In this paper, RSS (Rich Site Summary)/News Reader is a web-based application which developed with PhoneGap framework [5]. In this mobile application, a combination of PhoneGap framework, HTML5, CSS3, JavaScript, jQuery Mobile, Python and Django is used for implementation of Android-based Mobile Application.

The rest of this paper is organized as follows. In section "Methodology", we explain the details of methodology. The software architecture design steps, features and use case diagrams are given in section "Features and Use Case Diagrams". In section "Implementation", we present all details of implementation of mobile application development. Finally, conclusions are then given in last section.

Methodology

This application is a web-based application works on Android mobile platform. By using PhoneGap framework, a web-based software, can converted easily to a fully working native platform application. It takes advantages from both web languages and applied platform devices. This makes a hybrid structure which has faster and easier to implementation process.

In this application, we used a combination of PhoneGap framework [5], HTML5 [6, 7], CSS3 [8], JavaScript [9], jQuery Mobile, Python and Django.

Our application structure could be summarized as a combination of interface design, interactive components and data processing. These parts can create a complete web application. PhoneGap framework helps to pack and convert the software into a hybrid mobile application as shown in Fig. 1.

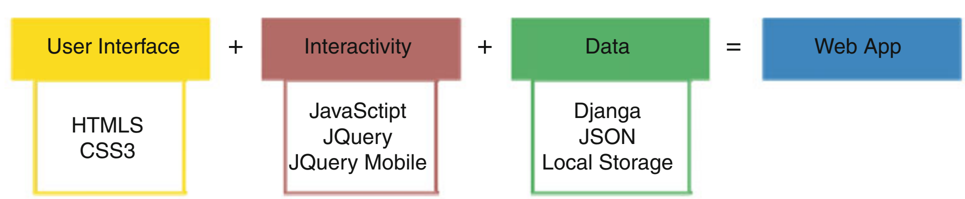
PhoneGap

PhoneGap [5] is an open source mobile platform to create cross-platform, means platform independent, mobile applications with web languages and tools like HTML, CSS, and JavaScript.

PhoneGap allows developers to create applications without using device-specific languages. This also provides the advantage that the exact same application can be used on different platforms without changing any interface. PhoneGap workflow is given in Fig. 2.

The applications developed with PhoneGap are called as 'hybrid'. It means they use both functions from native side and the web side.

PhoneGap is based on open source Apache Cordova Project. Apache Cordova is a collection of mobile application APIs that makes possible web based mobile applications with a suitable framework combination. In our application we used jQuery as a mobile framework.



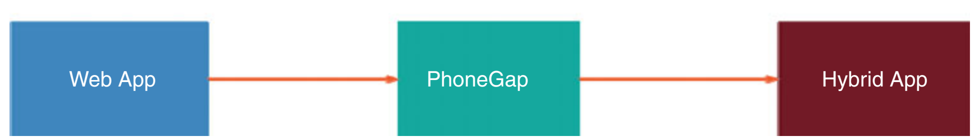


Fig. 1 Software platform on RSS/News Read

PhoneGap framework was created by Nitobi Software in 2008. The starting point of PhoneGap was that Nitobi developers did not want to spend their time with learning native mobile languages and instead, they want to use their web development skills. So they developed this framework to embed and run their web codes in mobile platforms. After their success with PhoneGap system, Adobe purchased Nitobi in 2011. The developers of PhoneGap believe the web is the optimal solution cross platform problem. They express the web is not accepted as a trusted application development platform and may lack at some areas but they try to fill these gaps with PhoneGap and make the web an acceptable development platform.

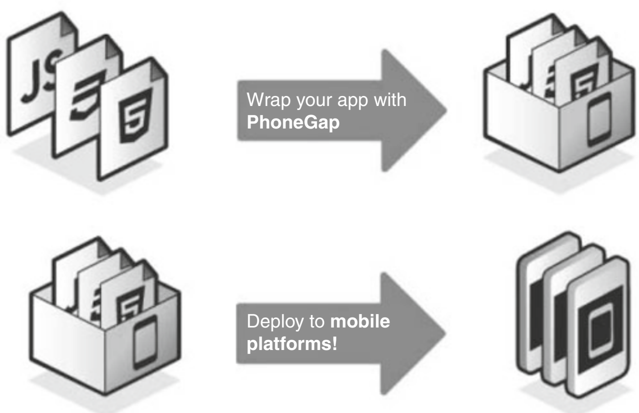


Fig. 2 PhoneGap workflow

Supported platforms by PhoneGap are given in Fig. 3 with the technical details. PhoneGap framework was created by Nitobi Software in 2008. The starting point of PhoneGap was that Nitobi developers did not want to spend their time with learning native mobile languages and instead, they want to use their web development skills. So they developed this framework to embed and run their web codes in mobile platforms. After their success with PhoneGap system, Adobe purchased Nitobi in 2011. The developers of PhoneGap believe the web is the optimal solution cross platform problem. They express the web is not accepted as a trusted application development platform and may lack at some areas but they try to fill these gaps with PhoneGap and make the web an acceptable development platform.

One of the main advantages of using web technology for mobile is that it is easy to check the code with any web browser and make changes on it. This provides faster coding process and an easier implementation. Also PhoneGap applications are packaged like native applications and can be distributed through mobile app stores.

HTML5

HTML5 is the last form of the core language of web: HTML. It has increased functionality of web design with its revolutionary features. Also, it is possible to develop mobile applications with these new features.

	iPhone / iPhone 3G	iPhone 3GS and newer	Android	Blackberry OS 5.x	Blackberry OS 6.0+	WebOS	Windows Phone 7	Symbian	Bada
Accelerometer	✓	✓	✓	✓	✓	✓	✓	✓	✓
Camera	✓	✓	✓	✓	✓	✓	✓	✓	✓
Compass	X	✓	✓	X	X	✓	✓	X	✓
Contacts	✓	✓	✓	✓	✓	X	✓	✓	✓
File	✓	✓	✓	✓	✓	X	✓	X	X
Geolocation	✓	✓	✓	✓	✓	✓	✓	✓	✓
Media	✓	✓	✓	X	X	X	✓	X	X
Network	✓	✓	✓	✓	✓	✓	✓	✓	✓
Notification (Alert)	✓	✓	✓	✓	✓	✓	✓	✓	✓
Notification (Sound)	✓	✓	✓	✓	✓	✓	✓	✓	✓
Notification (Vibration)	✓	✓	✓	✓	✓	✓	✓	✓	✓
Storage	✓	✓	✓	✓	✓	✓	✓	✓	X

✓ - supported feature
X - unsupported feature due to hardware or software restrictions

Fig. 3 PhoneGap compatibility table

Native applications may have better performance for mobile platforms but after the development of HTML5, web based applications are started to catch up native ones [6, 7]. HTML5 offers web based application with a different idea from classic web development fundamentals. It outs forward web applications instead of served side programming.

There could be some limitations and lacks of HTML5 to make a native-like application but they can be overcame by wrapping web applications with a platform framework which makes it connected with native APIs. We used PhoneGap to accomplish this as mentioned.

HTML5 Local Storage

With the new local storage feature of HTML5, developers do not have to use always server side storage for application data. HTML5 provides a new client side storage feature that minimizes necessity to connect remote server.

Local storage has some significant advantages. One of them is offline usage. After the application synchronized with the server side storage, it is possible to use obtained data offline. Furthermore, waiting times for data load are significantly low with local storage because there is no need to visit server every step of data usage. Lastly, since no server structure is needed, it is much easier to implement and program this type of storage technology.

CSS3

CSS3 is the latest and extended form of current CSS technology [8]. It is the standard way used to give the styles web pages and make them more dynamic.

Some of the most important features of CSS3 are:

- Animations
- Backgrounds and Borders
- Selectors
- Box Model
- 2D/3D Transformations
- Text Effects
- Multiple Column Layout

With these features, CSS3 has many capabilities to compose a mobile application indistinguishable from natively coded software. It is possible to make web pages and applications more dynamic and decorative easily. Also it works as an embedded component of web page in the application so there is no delay or extra workspace.

For mobile application development, HTML5 in combination with CSS3 is very useful, because it allows the creation of user interfaces similar to native apps, without the restrictions of each platform and without the need to program separately for them. This makes easier and more powerful to develop cross-platform applications.

JavaScript

JavaScript is a scripting language designed for adding interactivity to web pages and creating web applications.

JavaScript is a language used in the traditional web development to improve the user interface offered by the pure HTML pages with the aim to achieve a more dynamic web experience [9].

There could be some features of JavaScript which cause some compatibility problems with mobile devices. However, these problems can be solved by help of jQuery Mobile framework.

jQuery Mobile

jQuery Mobile is a mobile programming framework that makes available to use mobile functions in web based programming [10]. jQuery Mobile is a developer friendly technology because it offers progressive and stylish features to make easy to implement a web based mobile application. Also it has HTML5 standard code and this makes it to be supported by more platforms. It is supported by almost every mobile development platform today.

Some important features of jQuery mobile are:

- Open-source and free
- Cross-platform compatible
- Optimized for touch devices
- Customizable design
- Uses HTML5 semantics
- Based on well-supported jQuery core.

Python, Django and JSON

To download and use and RSS data; Django, a web framework of Python programming language, is used. With its many useful functions, Django is very helpful to format and employ the data of chosen RSS sources without using any server side programming language codes. It converts non-standardized and different RSS data into a uniform JSON (Java Script Object Notation) format.

JSON is data-interchange format which is based on usage of name/value pairs and an ordered list of values. It has a platform independent structure because it semantics is compatible with many significant programming languages.

Also, JSON is a lightweight text based format which has a very easy syntax for both computer and developer side. With its perfectly optimized structure, it needs significantly low storage space. Therefore, it fastens data transfer processes.

With this usable, compatible and fast structure; JSON is an optimal choice for a data-based RSS reader application.

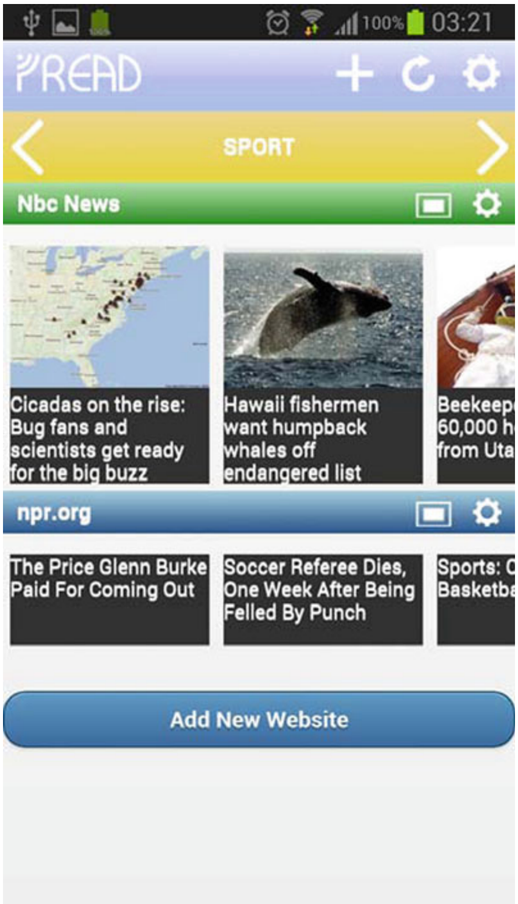


Fig. 4 Category screen

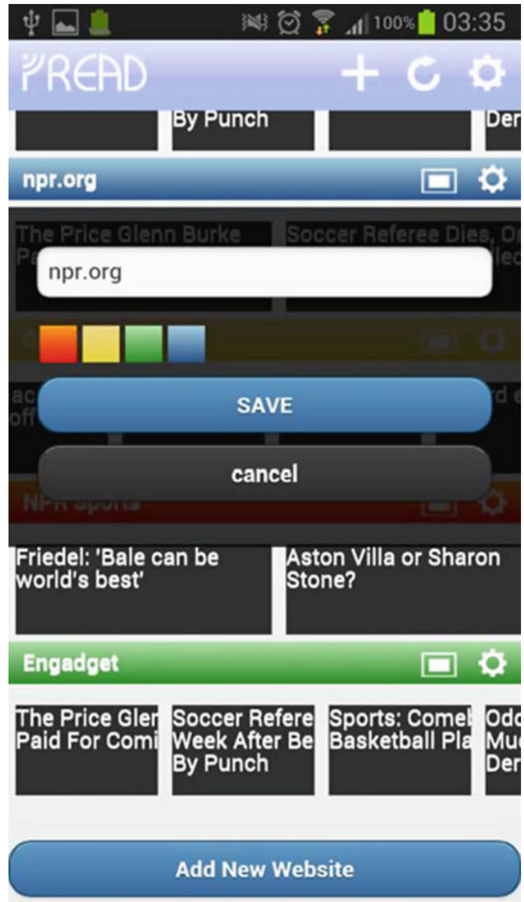


Fig. 5 Link settings screen

Features and Use Case Diagrams

Our main aim with this project is to create a RSS/News reader with useful and elegant design. This reader is going to have an option to add or remove RSS links and modifying them with categories.

The application is based on slider menu structure. Each source is implemented as a horizontal scroll menu and each category will contain a vertical list of these sliders. Each slider contains a modifiable amount of news cells.

With this aims our program features can be listed as;

- A cross-platform application which is easy to implement and deploy.
- Easy access and comfortable reading for RSS data.
- A better reading experience with a flexible interface.
- Ability of modification for users.

Use Cases

In this section, eight different use cases are described and users-system responses are examined. These are start-up application, view news, add category, add RSS link to a category, remove category, remove links from a category, settings and refresh.

1. Use Case: Start up Application

Actor action (user)	System response
1. The user starts up the application	2. The system loads information from local storage
3. The user ends the application	4. Until task is killed by the user, the service keeps on to process in background

2. Use Case: View News

Actor action (user)	System response
1. The user clicks on one of news picture	2. News reading screen opens
3. The user clicks ‘open in browser’ button	4. New link opened in embedded local web browser

3. Use Case: Add Category

Actor action (user)	System response
1. The user clicks add category button	2. Add category screen opens
3. The user choose category name and banner color	4. A new category with selected features is added

4. Use Case: Add RSS Link to a Category

Actor action (user)	System response
1. The user clicks add link button in any category	2. Add link screen opens
3. The user chooses link header, banner color and size of news cell	4. A new link with selected features is added in current category

5. Use Case: Remove Category

Actor action (user)	System response
1. The user clicks remove button for any category	2. Confirmation screen opens
3. The user confirms removal	4. The system removes selected category

6. Use Case: Remove Links from a Category

Actor action (user)	System response
1. The user selects any of links from a category	2. Confirmation screen opens
3. The user confirms removal	4. Selected links are removed by the system

7. Use Case: Settings

Actor action (user)	System response
1. The user clicks settings button	2. Settings screen opens
3. The user changes any of given settings	4. The changes are applied by the system

8. Use Case: Refresh

Actor action (user)	System response
1. The user clicks refresh button	2. The system refreshes all the links in all categories

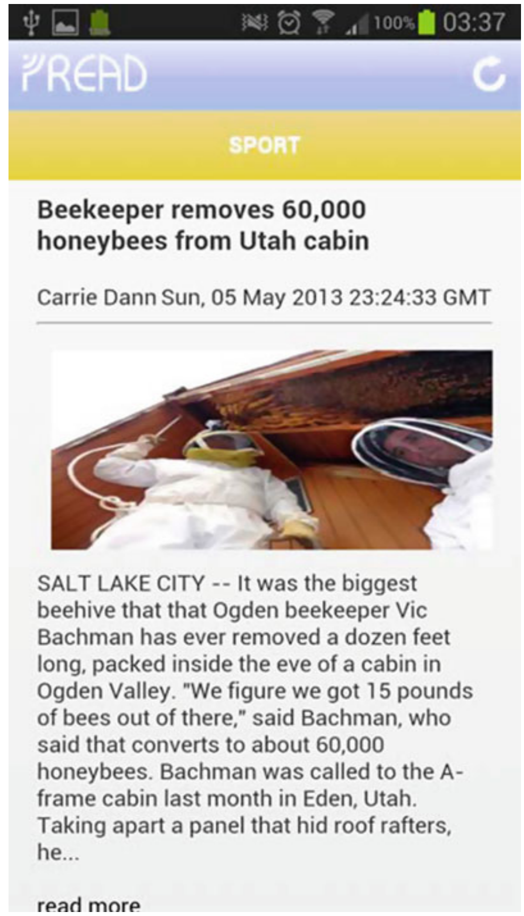


Fig. 6 News reading screen

Implementation

We tried to design our interfaces to provide a more flexible and clean application usage. With the web functions implemented with PhoneGap technology, it becomes easier to arrange and maintain the interfaces [11].

In this section, some screenshots of mobile application design are given. These screenshots demonstrate how the web-based application design appears on a real Android device.

1. *Category Screen*
   Category Screen is the most important part of the application design and it is given in Fig. 2, because of being the home page of the application. Link Setting Screen, News Reading Screen and Add Category Screen are also given in the following subsections in Fig. 4, in Fig. 5, in Fig. 6 and in Fig. 7.
2. *Link Settings Screen*
3. *News Reading Screens*
4. *Add Category Screen*

Conclusions

RSS/News reader is an android operating system mobile application. The difference between this application and the other news applications is that RSS/News is a web-based application. Therefore, it can also be used on other mobile platforms by adapting the core web application with simple configuration changes. This feature comes with PhoneGap framework. This framework acts as an adapter between web application and native platforms. By using this framework, there will not be any necessity to know specific programming languages for different operating platforms.

Another important point for RSS/News reader is that it has simple, clean and understandable user interfaces. Because of its simple graphical user interface, people can take advantage of a really comfortable reading experience. This application improves usability for people even not familiar to mobile devices.

It is decided to distribute the application on Android platform because it is the most suitable one at first stage. The application will be deployed to other platforms on demand.

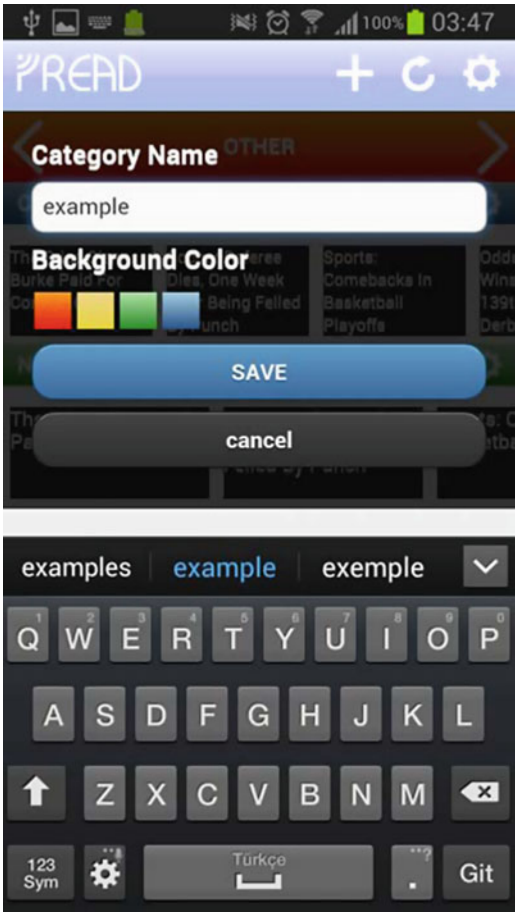


Fig. 7 Add category screen

In conclusion, RSS/News Reader is a reliable, simple, extensible mobile application which offers a more user-friendly experience among its counterparts. Our future work will be developing multi-platform version of RSS/News Reader mobile application.

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Intelligent Emergency Response System for Police Vehicles in India

Ishan Ganeshan and Nasrullah Memon

Abstract

When faced with emergency situations there might be several critical factors that could preclude the possibility for the victims to call for the help. In situations like kidnapping, rape, robbery making use of the traditional voice based methods to call for the help might alert the offenders and put the victim's life at a greater risk. This paper proposes an Emergency Response System (ERS), which focuses on developing an alternate mechanism through human computer interaction whereby the help can be called through a single press of a button and the locations of the callers are tracked in real time by the police vehicles. In the proposed system, the administrator can view the performance of all the police vehicles at any time through a web portal. The system used traditional data mining algorithms in order to analyze crimes in different areas of a city and at different times of the day. Based on this crime mapping, the administrator assigns patrol schedules for different police vehicles throughout the day. The proposed system would make it very easy for people to call for the help, and the police authorities to know the locations of the callers and identify crime hot spots and the administrator to keep track of the performance of each police vehicle.

Keywords

Critical factors • Data mining • Intelligent emergency response system • Police vehicles

Introduction

In India, there has been an alarming increase in crimes with every passing year. Every 20 minutes there is a rape happening somewhere in India [1]. The study shows that during 2001–2011 there has been a 31.4 % increase in crimes committed under the Indian Penal Code (IPC) [2]. Recent events [3] have highlighted the poor state of the emergency response in India. There are many problems plaguing the current system. Telecom Regulatory Authority of India [4] has identified many loopholes in the contemporary emergency response system. One notable loophole in the existing system is the lack of crime based police patrolling [5]. The crime taking place in a city needs to be analyzed in order know the locations and the victims against whom specific types of crimes occur. Knutsson and Clarke [6] have mentioned that there is a very large difference between the different times of the day at which there is crime and the police patrolling at those times. For example, crime peak hours in a specific area in a city might be 8–9 pm and police is not patrolling in that area at that time. The main objectives of this paper are: (1) to enable people to call for the help only with just a press of a button; (2) the automatic selection of the nearest police vehicle, (3) location tracking of the victim (4) crime mapping using apriori data mining algorithm and

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_8, © Springer International Publishing Switzerland 2015

on the basis of which setting the patrolling the schedules for the police vehicles, (5) to develop a web portal so that an administrator can keep track of the performance of the different police vehicles and (6) to log the activity of each police vehicle for each and every second of the day into a central database which would keep the police accountable throughout the day.

This article is organized as follows. We start in section "System Architecture of ERS" describing the system architecture of the proposed Emergency Response System (ERS). In section "Math" we present a methodology of the conceptual design of main components of the proposed system. We show in section "Results" the results of our novel application; in addition, data mining analysis and a clear indication of the crime patterns in a particular area in order that the patrolling schedules of the police vehicles are set more efficiently. Finally, conclusions and future work are presented in "Conclusion".

System Architecture of ERS

The system architecture of the proposed system is shown in Fig. 1. The system consists of three clients namely: Subject Client, Response Unit Client and the Administrator Client. A central server coordinates the actions of all the clients and retrieves/stores data in a database. The server is optimized to service thousands of subjects (people) and hundreds of response units at any point in time.

Each subject would be provided with a subject client device which would have a help button. The response unit client is fitted inside each response unit. The response units would receive the patrolling schedules set by the administrator in their respective response unit clients at the start of the day. The server controls and coordinates the data transfer between all the clients. It is the responsibility of the server to authenticate the different clients, analyze crime data patterns, and provide data from the database to the different clients. The administrator client is a web application through which an administrator gets access to the real time state of the different response units and subjects. The administrator sets the patrolling schedules for the different response units on the basis of the crime analysis performed by the data mining agent and also view the locations of the response units and the subjects on a digitized map.

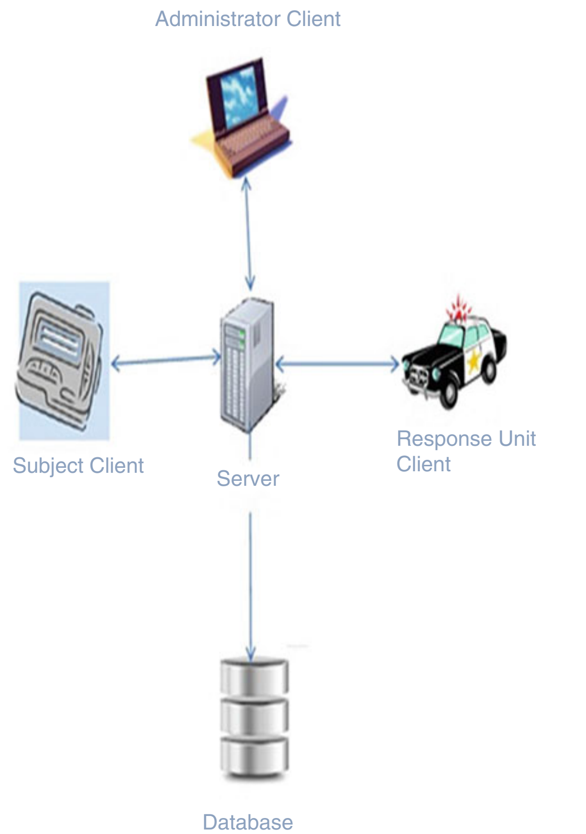

Fig. 1 The architecture of Emergency Response System

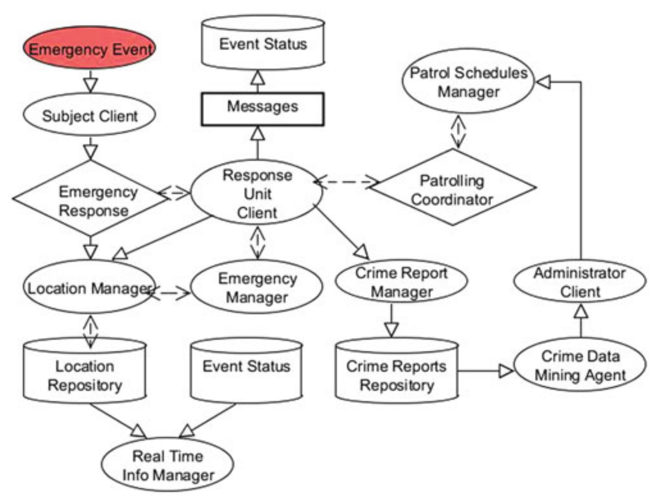

Fig. 2 The description of Process and Information Flow in the Emergency Response System

Description of the Process and Information Flow in ERS

The emergency response process starting from the occurrence of an emergency event to the collection of responses and actions by the different actors of the system is illustrated in Fig. 2.

Once a request is accepted, the details of the subject are passed on to the response unit. The real time locations of the subject would be available on the map inside the response unit. After the case is serviced, the police officer of the response unit would write the crime report of the case. The crime reports are saved in a database. Simultaneously, other response units which are not servicing an emergency request keep patrolling in specific areas assigned by the administrator. The real time locations of the response units are continuously stored in the database and the patrol accuracies are found in order to regulate the accuracy with which the patrolling is performed. The data mining agent at the server predicts and detects regions and types of victims which are mostly subjected to specific types of crime at specific times of the day. On the basis of the detected crime trends the administrator assigns the patrolling schedules for the different response units.

Math

As outlined in section "System Architecture of ERS", ERS requires the integration of different components. In this section, the conceptual design models of the different components are presented.

Conceptual Model of Server

The server is the central element of the system. It performs all the business logic of the application. All the computations and distribution of different resources to different clients are done by the server. The primary task of the server is to coordinate the actions of the subjects, administrator and the different response units at times of emergency. Figure 3 shows the computation logic (in the shape of flow chart) which the server uses to send the nearest response unit to the subject.

Once an emergency request is obtained, the server retrieves the subject location from the database along with the locations of the response units. It computes the nearest response unit to the subject location and sends the emergency request to that response unit. Once the response unit accepts the request, the location of the subject is continuously sent to the response unit. The response unit can request for extra units at any time. The server processes this request and assigns more re-sponse units for the specific request. Each response unit then can view the locations of other response units and the subject on the map. If the request for number of extra units becomes greater than four, the server stops assigning any more response units and the administrator is alerted. This is because any situation which requires more than four response units is a very high level emergency situation which needs to be directly coordinated by the administrator.

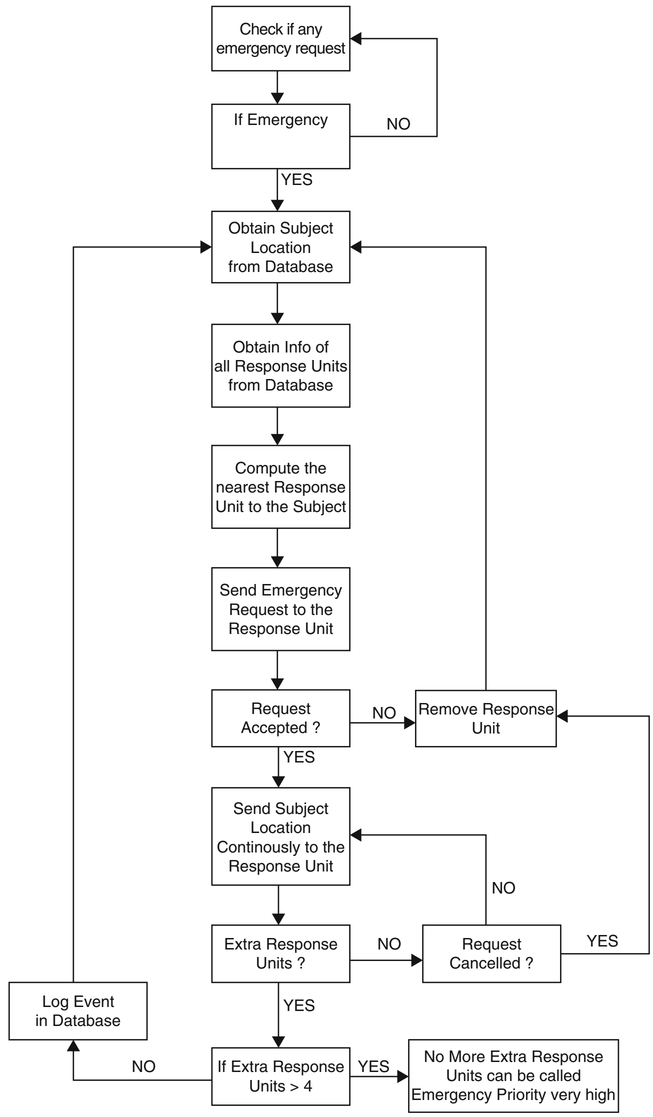

Fig. 3 Flowchart for emergency request processing

Data Mining Agent

In ERS, data mining is used to detect and predict where and when and against whom certain types of crimes have a greater probability of being committed. Predictive crime analysis employed by the Santa Cruz Police department in California [7] resulted in 27 % reduction in burglaries. Felson and Poulsen [8], Harries and Thomas [9] confirm that crimes vary by the time of the day. Petherick [10] mentions that each victim is chosen to be a victim by the perpetrator of crime because of certain unique characteristics. Hence, there are many factors which influence crimes in a region. Therefore, in ERS the following factors are taken into account to analyze the crimes.

- The time of the day the crime has taken place.
- The type of victim involved in the crime whether he/she belongs to upper/lower caste, is old or young or is a male or a female.
- The type of crime, whether it is a murder, rape, kidnapping or a robbery.
- The location where such a crime has taken place.

Design of Data Mining Agent

Apriori data mining algorithm [11] has been used to unravel hidden relations between input variables in large databases. It detects frequent items that occur together in the crime data. Each input variable is considered as a single item in the item set. It employs a breadth first search method to count all the different candidate item sets. The algorithm generates all candidate item sets of length k from an item set of size k-1. All those candidates which have an infrequent pattern are pruned. The resultant candidates contain all the frequent k-length item sets. Hence, after the entire database is scanned, the final set would contain those items that occur most frequently together and satisfy the minimum support threshold.

The role of the data mining agent is to detect patterns of crimes from the generated crime reports. The data mining agent at the server mines for special correlations among the data in the crime reports. The discovered associations between the items sets are analyzed further to find the correlation rules. The outcome of the mined data is used by the administrator to set the patrolling schedules for the different response units. The main role of the data mining agent is to detect those input patterns which occur together most frequently. This would provide information about which kinds of events give rise to certain specific types of crimes.

Inputs

To mine for crime data, various input variables have been chosen. These are the time of day, type of crime, type of victim, and region of crime. Each day has been divided into four time slots of six hours each. Each time slot of six hours is further divided into two time slots overlapping the next time slot. For simulation purposes twelve different regions are considered where crimes are committed. Each region is considered as a separate item. Similarly, each of the four different crime types namely murder, rape, kidnapping and robbery are considered a separate item. Six different input variables have been chosen to represent the characteristics of the victims namely, upper caste, lower caste, old, young, child, male and female. Each of these variables is a separate item.

Results

In this section, the results of testing the proposed ERS would be presented. Through these results it would be shown that ERS would function much more effectively and efficiently as compared to the current system of emergency response in India.

In ERS, the crime data mining agent running at the server detects crime patterns from the crime reports written after each case is serviced. These detected crime patterns provide a detailed description of the type of the crime and the types of victims against which crime take place at certain times of the day. It should be noted that, since traffic modeling is outside the scope of the paper in order to measure the performance of police vehicles to emergencies, the response times are not considered. Instead, the distance between the location of the subject and the nearest police vehicle is used as a performance measurement factor. The results from these tests would further prove that the distances between the police vehicles and the subject at different times of the day would be very high before crime data mining is performed. However, after the data mining agent performs data mining and the administrator assigns patrolling schedules based on these detected crime patterns, the results would show that distance between the nearest police vehicle and the subject would be significantly reduced. These detected crime patterns would be plotted in a graph which would provide a visual description of the crimes and would enable us to get a much better understanding of the crime statistics in a particular area. Finally, the crime reports of all the simulated cases would be plotted in a graph to illustrate the crime patterns in a particular region.

Simulation Techniques

To simulate the ERS, the following factors have been taken into account:

A total of 12 different regions within a city have been considered. All these regions have been selected at a reasonable distance away within the city of NOIDA in Uttar Pradesh. In each of these regions, four different coordinates have been chosen to simulate the patrolling of the police vehicles.

A day has been divided into 4 different time intervals of 6 h each, namely 12 am–6 am, 6 am–12 pm, 12 pm–6 pm, 6 pm–12 am and four different response units were simulated. A total of 100 cases before data mining and 100 cases after data mining were simulated.

The coordinates for simulation of patrolling of the different response units and the subjects were obtained through Google maps.

Data Mining Analysis

For simulation 100 cases were run before performing crime data mining and 100 cases were run after performing crime data mining. In each case, the subject sent a help request from a different location. After servicing of each case crime reports were written to generate crime data. These crime reports were persisted into a database which then was used by the data mining agent to detect crime patterns. These detected crime patterns then were used to enter the new patrolling schedules. Once, the new patrolling schedules were entered 100 more cases were run. The distance between the nearest response unit and the location of the subject before and after data mining are plotted in Fig. 4.

The blue markers represent the cases before data mining was performed. The red markers represent the cases after data mining was performed. From the concentration of the grouping of the red and the blue markers it can be clearly inferred that before data mining was performed in most cases the distance of the nearest response unit lay in the interval 2–4 km. However, after data mining this distance lay in the interval 0.2–3 km. This shows that the patrolling of the response units after the crime data mining was significantly more efficient which resulted in more cases where the response units were near to the scene of the crime.

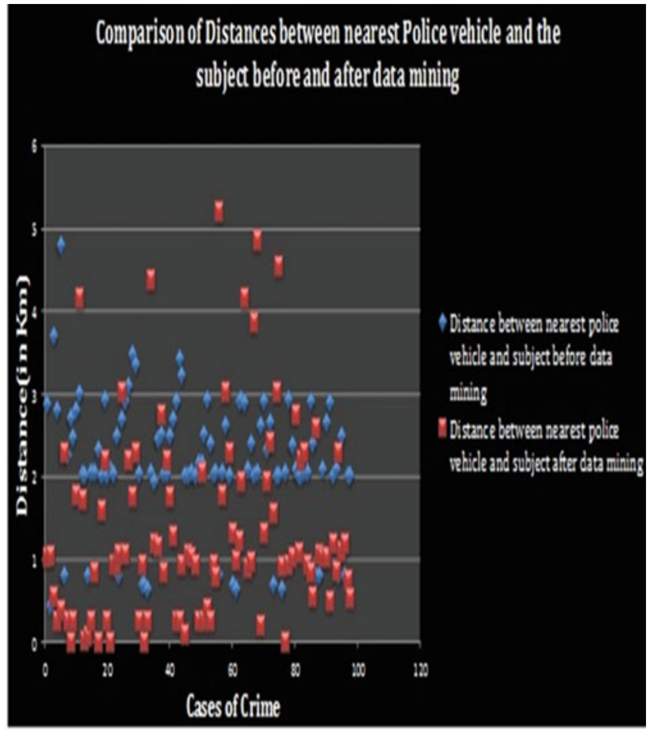

Fig. 4 A plot of the distance between the nearest response unit and the subject before and after data mining vs. the different cases of crime

Figure 5 shows a plot of the average distances of the response unit to the subject before and after data mining against the total simulated cases. We can clearly see that the average distance between the nearest response units to the subject before data mining was higher compared to the distance after the data mining. The average distances are listed in Table 1.

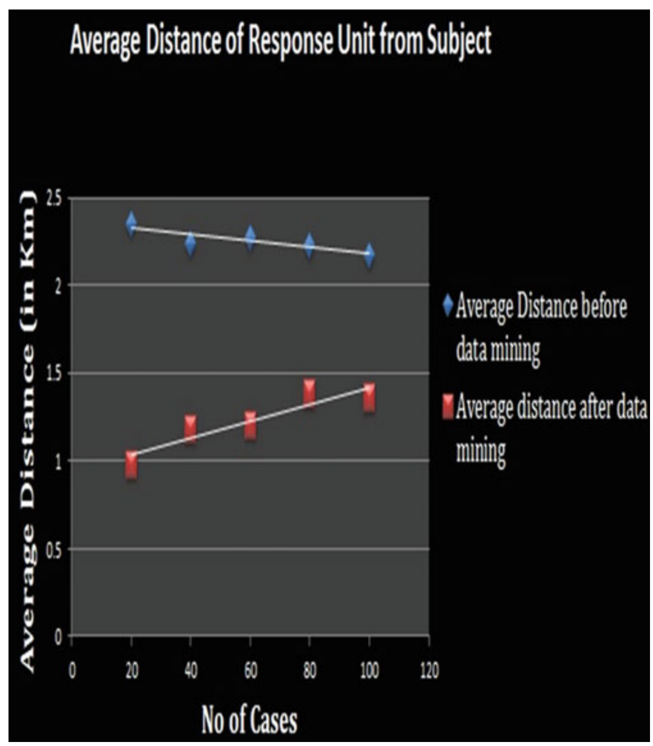

Fig. 5 A plot of the average distance between the nearest response unit and the subject before and after data mining vs. the total cases of crime

From the experiments it is clear that after data mining the average distances between the response units and the subjects have been significantly reduced.

As mentioned, the crime reports were written after each case was serviced. In these crime reports only three types of crimes were considered for simulation purposes namely rapes against upper caste females, murder against lower caste males and kidnapping of lower caste females. It should be noted that at the end of 100 cases using the crime statistics a graph is plotted as shown in Fig. 6 which clearly demonstrates the crime patterns in a particular area. Figure 6 shows that the rapes against upper caste females are on the peak between 6 am and 12 pm and the murder against lower caste males are the maximum between 12 pm and 6 pm. It would beneficial for the police authorities to see the crime statistics and can get a clear indication of the crime patterns in a particular area in order the patrolling schedules of the police vehicles can be set more efficiently.

As per our discussions with the experts, it is witnessed that the results produced show that the proposed ERS has a significant enhancement in comparison to the present system of emergency response in India.

Table 1 List of average distance between nearest response unit and the subject before and after data mining

Total number of cases	Average distance between response unit and subject before data mining (in km)	Average distance between response unit and subject after data mining (in km)
100	2.17	1.36

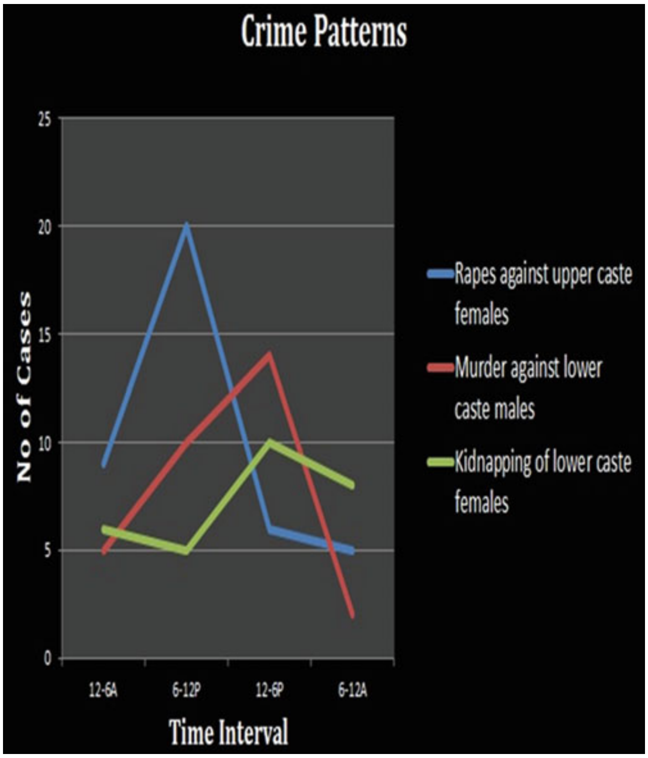

Fig. 6 A plot of the no of cases of certain types of crimes against the different time intervals

Conclusion

The results presented in section "Results" suggest that our approach using Apriori data mining algorithm for the development of an alternate mechanism of emergency response for police vehicles in India is feasible and effective for the whole life cycle of an emergency. The results obtained from the prototypical development prove that such a system is indeed practically and technologically feasible. The conceptual model of the proposed system is a significant improvement over the current state of the art of the emergency response in India. The results further show that a crime pattern based patrolling of police vehicles could drastically reduce the distance between the nearest police vehicle and the scene of the crime. The main theme of this research is to set the solutions for the problems of crimes in India into a software technological perspective and propose an alternate system which could improve upon the current system and eventually help in reducing the crime rates in India.

To address modeling the patrolling of the police vehicles on the basis of the traffic models of that city, we plan to investigate: (1) the traffic patterns in a city at various times of the day (2) the current patrolling locations of the police vehicles based on these traffic patterns (3) the difference in the response times of the police vehicles in the current system and the proposed ERS based on these traffic patterns. In addition we plan to integrate hardware components and test the system in a real time environment.

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Adaptive Instantaneous Traffic Signal Management Through Cascade Object Detection

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Abstract

Road traffic controlling is one of the prime issues of any busy town in the world. The situation is worst in the junction point of multiple roads. It is not always possible to predict upcoming traffic pressure intensity and direction with prior information. An adaptive approach is needed to solve this problem in real time. In this paper an efficient method of controlling traffic signal duration is proposed. Here cascade object detection technique is used to recognize and count the number of moving vehicles towards a junction from various directions through static video camera installed in each direction. Using the static camera, the proposed system senses the current traffic pressure of each connecting road towards the junction and thereby takes sensible decision of setting traffic signal duration in each road.

Keywords

Computer vision • Traffic management • Vehicle management • Traffic signal • Image processing

Introduction

Road traffic controlling is one of the prime issues of any busy town in the world. Traffic volume is not uniform throughout the whole day. Road junctions suffer huge traffic congestion due to change of traffic pattern and intensity of its direction to the junction. In multiple-way junction traffic signal duration is designed considering traffic flow demand, which is generally measured before setting it up.

Research has shown that local administrative authorities do study of traffic at local regional level to make smooth movement of traffic and accordingly these authorities make traffic signals with static time limit to pass [1]. More time is given for that intersection traffic where more traffic load is observed. But this solution is not enough to handle traffic adaptively. In this paper a method is proposed to control traffic signal duration adaptively with the help of computer vision. This method calculates traffic intensity from different direction to the junction point in real time and setup necessary traffic signal duration adjustment for different road connection. Background subtraction and thresholding are performed to produce difference images. Background subtraction has been used by many researchers to extract moving objects in the scene [2–5]. Change detection algorithm that compares successive frames [6, 7] can also be used to extract motion. The choice of the threshold is critical. Many thresholding techniques [8, 9] work very well when there is an object in the scene. This is because these techniques assume that the image to be thresholded, contains two categories of pixel values and they try to separate the two [10]. Here static threshold value is used. The value was obtained by examining an empty background for a short while and measuring the maximum

The paper was submitted for review on 3rd November, 2013.

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DOI 10.1007/978-3-319-06764-3_9, © Springer International Publishing Switzerland 2015

fluctuation of pixel values during this training period. The threshold is set to a little above of that value. The techniques used in this approach are: Gaussian mixture models, Morphological operations, blob analysis and Kalman filter [11, 12].

Existing Approaches

There are many existing methods to control traffic flow efficiently. Every method has some advantages along with some shortcomings.

Goel et al. [1] proposed an intelligent traffic signal system to prioritize traffic clearance for emergency vehicles based on a wireless sensor network. They proved Wireless Sensor Network (WSN) to be very beneficial in the design of an adaptive and dynamic traffic signal intersection system that minimized the waiting time of emergency vehicles and also managed the traffic load at the intersection adaptively.

Findler and Stapp [13] proposed a distributed approach to optimize control of street traffic signals by a network of distributed processors situated at street intersections. In their method every processor ran an identical expert system and communicated directly with the four adjacent processors. Messages could also reach indefinitely distant processors, modulated by the needs of intervening ones. The rule-base of the expert systems had a natural segmentation, corresponding to different prevailing traffic patterns and the respective control strategies. Multidimensional learning programs optimized both the hierarchy of the rules and the parameters embedded in individual rules. Different measures of effectiveness could be selected as the criterion for optimization.

Gartner [14] proposed a demand response strategy for traffic signal control. He described Optimization Policies for Adaptive Control (OPAC) as a computational strategy for real-time demand-responsive traffic signal control. It had the following features: (a) provided performance results that approach the theoretical optimum, (b) required on-line data that could be readily obtained from upstream link detectors, (c) it was suitable for implementation on existing microprocessors, and (d) formed a building block for demand-responsive decentralized control in a network. Studies undertaken in the development of this strategy and the testing of its performance were done via the NETSIM simulation model.

There are also some existing methods of computer vision to detect an object and its movements. These are as follows:

Cheriyadat et al. [12] proposed an object detection system that uses the locations of tracked low-level feature points as input, and produces a set of independent coherent motion regions as output. As an object moves, tracked feature points on it span a coherent 3D region in the space-time volume defined by the video. In the case of multi-object motion, many possible coherent motion regions can be constructed around the set of all feature point tracks. They solve the problem of finding the best set of coherent motion regions with a simple greedy algorithm, and show that their approach produces semantically correct detections and counts similar objects moving through crowded scenes. From the earlier research of Cucchiara et al. [15], we observe that video-surveillance and traffic analysis systems could be heavily improved using vision-based techniques to extract, manage and track objects in the scene. However, problems arise due to shadows. In particular, moving shadows can affect the correct localization, measurements and detection of moving objects. Wedel et al. [16] presented an approach for identifying and segmenting independently moving objects from dense scene flow information, using a moving stereo camera system. Using error propagation and scene flow reliability measures, they assigned dense motion likelihoods to every pixel of a reference frame. These likelihoods were then used for the segmentation of independently moving objects in the reference image. Cucchiara et al. [17] worked to present a general-purpose method for segmentation of moving visual objects based on an object-level classification, ghosts and shadows. Background suppression needs a background model to be estimated and updated: they used motion and shadow information to selectively exclude from the background model moving visual objects and their shadows, while retaining ghosts.

In our proposed method we extracted our moving object from the background using cascade object detection. As our moving object moves for a good span of time, background subtraction is relatively easy for us. We did not give too much effort on accuracy in instant subtraction of background.

Data Collection

The proposed data collection is based on some principles including prescribed methods of collecting data. Violation of following proper method in data collection may produce nebulous results. In this section procedure is described elaborately for proper collection of data. Here a junction point with four inbound roads is considered. In this proposed method stationary video cameras are to be setup in all connecting roads towards the junction point. The stationary cameras, capturing the continuous flow of traffic are to be installed over the inward roads to the junction point. The cameras have to be placed far from the road junction. The distance is to be decided from the traffic density, traffic speed and traffic pattern. It should be placed well above the road from where it is possible to capture vehicles in bird's eye view. Most important part of installing static camera is to keep it in a position where no shaking

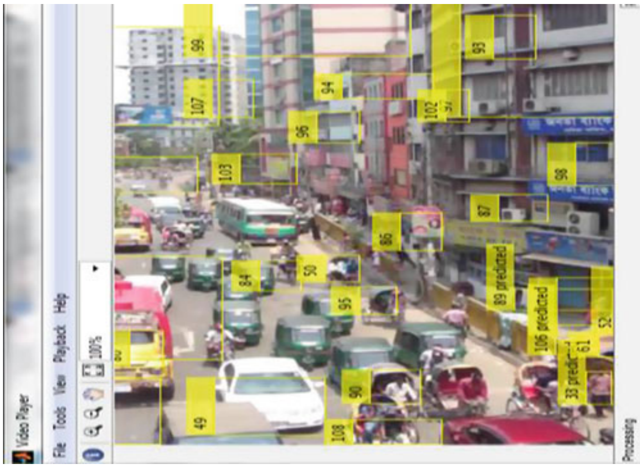

Fig. 1 Unstable camera can give un-expected result. Camera shake can predict static object like building as moving object

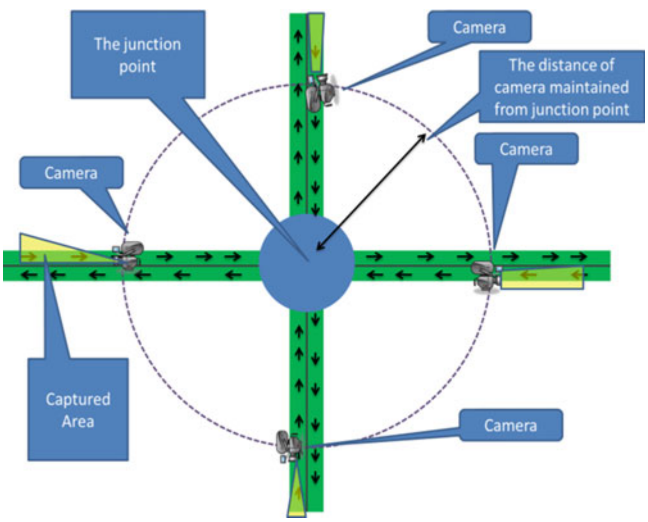

Fig. 2 The position of camera for four road junction

movement is possible. The base and the camera are to be very stable. Shaky camera can produce erroneous video. This video may predict stable objects like buildings as moving object as shown in Fig. 1.

If the connecting roads have both way traffic, the camera is to be directed towards only the incoming traffics. Figure 2 illustrates the positions of the cameras, which indicate distances from the junction point and Fig. 3 illustrates the video frame to be captured.

From the captured frames for equal stipulated time period for different connecting roads frames will be analyzed and get moving objects recognized using techniques described in the next section. Next, the total number of moving vehicles passed through the framed area towards junction is counted. Analyze part use the count of vehicles to take decision for signal duration of different connecting roads.

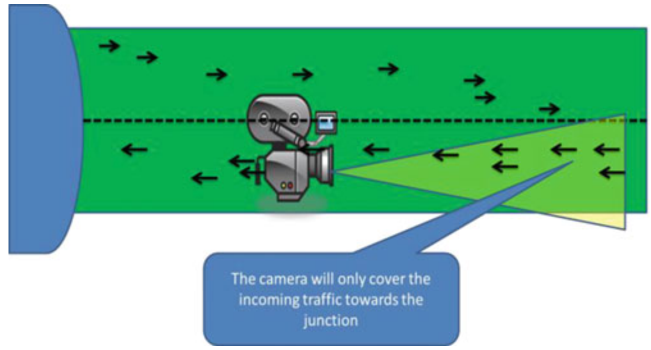

Fig. 3 Camera frame covering only the incoming traffic towards the junction

Proposed Method

The proposed method is divided into two parts. In first part the vehicles are detected, tracked and understood as moving objects. Afterwards the total number of detected vehicles is calculated from captured video frames taken by stationary camera placed in the roads which end up in the same junction [18]. This part is totally based on implementation of computer vision techniques. This part is referred as "Traffic density understanding". In the second part, to calculate signal duration towards a junction point, total number of vehicles (which was counted in previous step) of each road is used. This part is referred as "Signal duration calculation". In this paper the term "roundup time" is used as one complete cycle consisting of vehicle detection, tracking, understanding, counting and signal duration calculation. The resulting signal duration of one roundup time is used in the next roundup time. Figure 4 shows the flow of whole process.

Traffic Density Understanding

The recognition of moving objects uses a background subtraction algorithm based on Gaussian mixture model. The foreground detector is used to segment moving objects from the background. It outputs a binary mask, where two different pixel values correspond to the foreground and the background. Connected groups of foreground pixels are likely to correspond to moving objects. A Gaussian Mixture Model (GMM) is a weighted sum of M component Gaussian densities as given by the Eq. (1) [19]:

$$p(x|\lambda) = \sum_{i-1}^{M} \omega_i g\left(x|\mu_i, \sum i\right) \tag{1}$$

where, x is a D-dimensional continuous-valued data vector (i.e., measurement or features), w_i are the mixture weights,

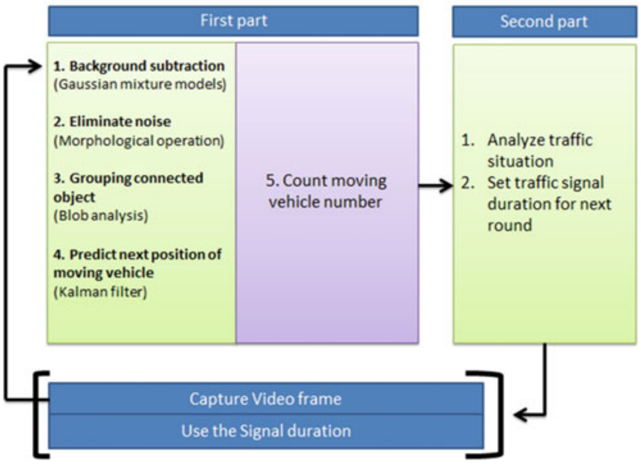

Fig. 4 Workflow of proposed method

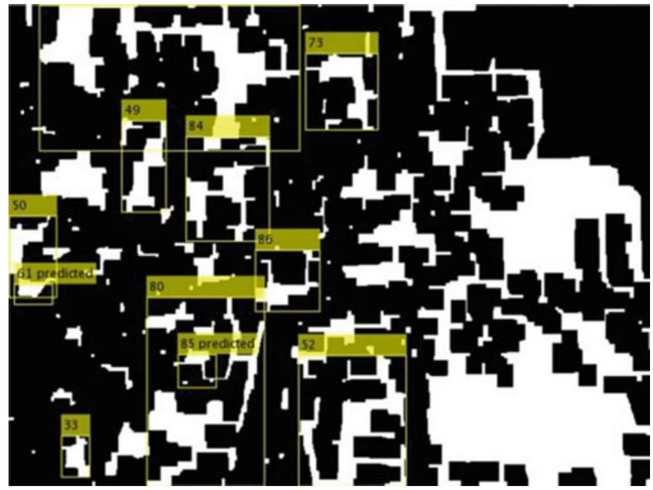

Fig. 6 Middle stage of foreground segregation

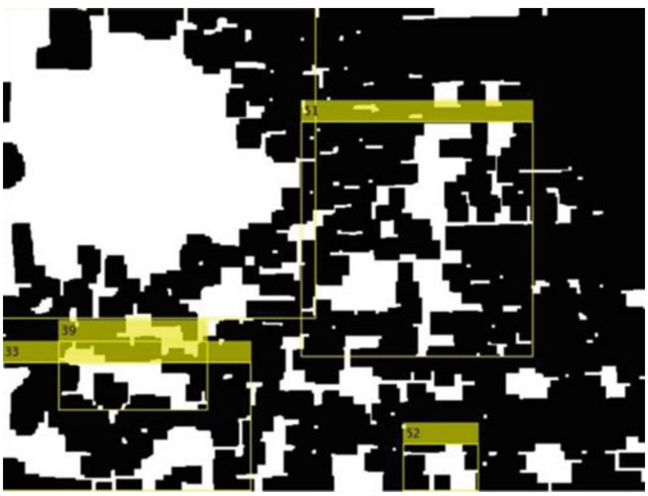

Fig. 5 Initial stage of background-foreground detection

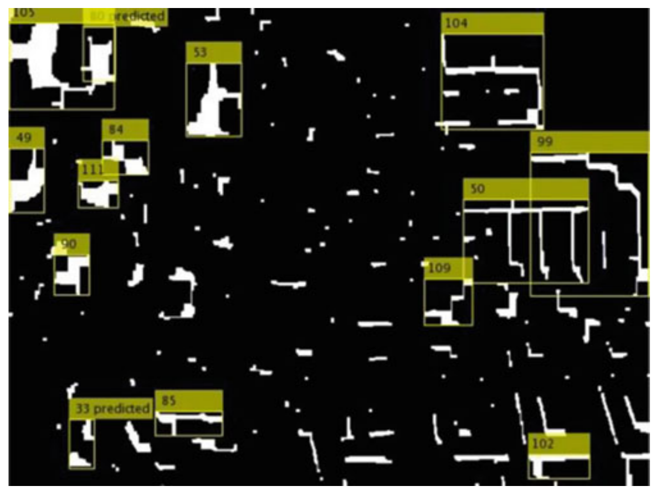

Fig. 7 Moving objects are detected using Gaussian mixture model

and $g(x|\mu_i,\Sigma_i)$ are the component Gaussian densities where $i = 1, \ldots, M$ [19].

Figure 5 shows the initial stage of foreground detection from background. MATLAB R2013a is used as platform. However after analyzing vehicle movements some part of the frames begins to be considered as foreground. With the passage of time the detection tends to be perfect. Figure 6 depicts the improvement of foreground detection from Fig. 5. This is the middle stage to see the result of Gaussian mixture model. In Fig. 7 almost perfect moving objects detected as foreground.

Noisy detections result in short-lived tracks.

Figure 8 shows lots of tracks, which are denoted as "predicted". After observing for a period of time some of the noises qualify to be tracked as moving object. This happens when total visible counts exceed a specified threshold. Figure 9 shows noise covering the full frame however only the portion which is stable for a stipulated period of time is counted as potential object.

When no detections are associated with a track for several consecutive frames, it assumes that the object has left the field of view and deletes the track. A track may also get deleted as noise if it was tracked for a short time, and marked invisible for most of the frames. In Fig. 10 the existence of object number 234 cannot be seen which was in the frame several seconds ago (Fig. 9) because it stayed invisible in the frame for a defined time which makes the system understand, the object left the frame.

Morphological operations are applied to the subsequent foreground mask to eliminate noise [20].

The morphological dilatation is characterized by the following properties [21]:

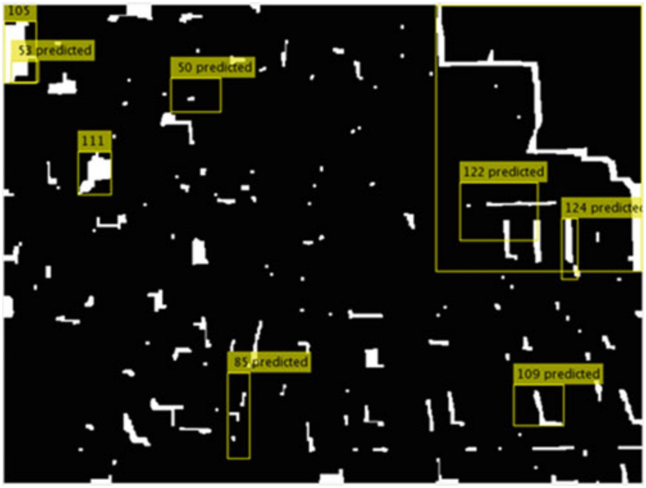

Fig. 8 All tracks are initially considered as predicted moving object. If the track moves for next few numbers of frames, it is detected as moving object

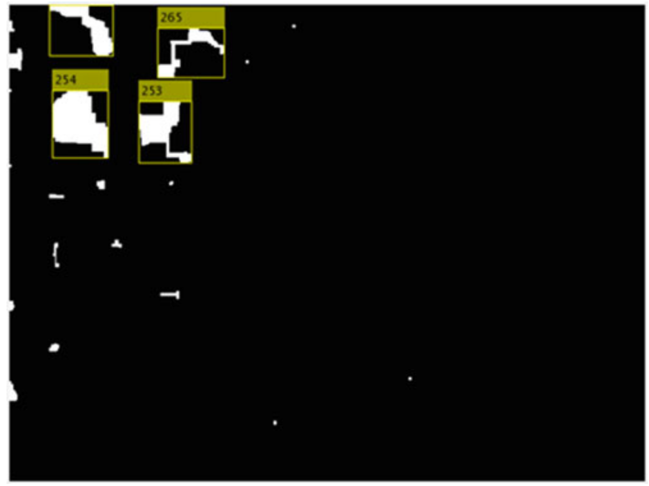

Fig. 10 Noise is detected and discarded from the frame

Fig. 9 Noise is covered through the whole frame

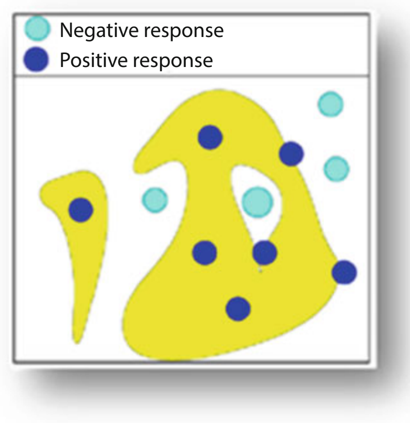

Fig. 11 Morphological dilatation [22]

- Fills holes smaller than specific size (Fig. 11) considering the detection as noise.
- Fills narrow channels.
- Small details disappear.

Blob analysis detects groups of connected pixels. Blob analysis, or Blob detection is used to detect and analyze connected region in the specified image, and this connected region is called Blob. Blob analysis detect the amount, position, shape and orientation of the Blob in the images for machine vision application, even topological relation between objects could be obtained from Blob analysis [23]. The basic work flow of Blob analysis is indicated in Fig. 12.

Signal Duration Calculation

There are differences in traffic signal system in different countries. Some countries use their own convention of traffic signaling system. Some of them use more lights to describe situation more specifically. However in most countries three lights are used in traffic control [24]. Generally three lights used for traffic control are Green, Yellow/Amber and Red. These three lights indicate three different instructions to the vehicles. Some countries use blinking of them to create new instruction. Some countries even use combination of blinking lights. Here mostly used convention is used. Green light allows traffic to proceed in the direction denoted. Yellow light denotes, prepare to stop short of the intersection. Red signal prohibits any traffic from proceeding. In the proposed method, Green means "Go", Red means "Stop"

Fig. 12 Basic concept of Blob analysis

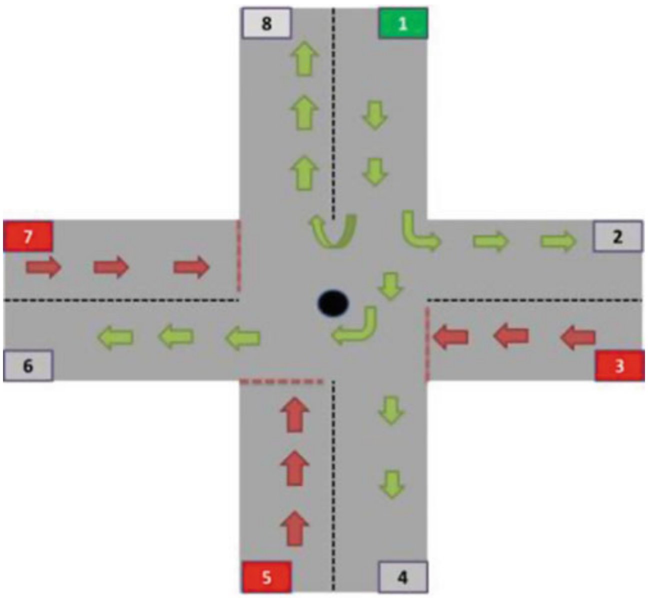

Fig. 13 Traffic flow in a junction of four inbound roads

and Yellow means guard time, which comes in between green and red light and vice versa.

In Fig. 13 there is a junction point of four roads, which creates a scenario of four inbound and four outbound traffic carrying roads. This method only considers regulating traffic of inbound roads. Outbound roads will automatically be maintained while maintaining inbound traffic. In this scenario it is to install four traffic signals in road number 1, 3, 5 and 7. When any of the light turns in to Green three other lights will be turned into Red which means only one inbound road traffic is allowed while restricts other three inbound road traffics. Inbound traffic of one direction can go in three different ways while crossing the junction. Calculation of traffic signal duration has been discussed below, where only the green signal duration of the roads are calculated as all other roads with signal light, will automatically turned in to Red light state. Yellow signals are of fixed time duration.

Let us consider t_1, *t2, t3 and t4* be the duration of green signal for roads *1, 3, 5 and 7* respectively. D be the duration of yellow light, C be the duration of common minimum signal, which is fixed. Minimum signal duration is required as all the roads must have some period of green light even if there is no traffic in the previous roundup time in any road.

We assume the junction is made up with n number of inbound roads and the roundup time is R. In case of n = 4, the expression is:

$$\begin{aligned}
R &= (t_1 + C + D) + (t_2 + C + D) + \\
&\quad (t_3 + C + D) + (t_4 + C + D) \\
R &= t_1 + t_2 + t_3 + t_4 + 4(C + D) \\
R &= T + 4(C + D) \\
\text{Where, } T &= t_1 + t_2 + t_3 + t_4 \\
T &= R - 4\,(C + D)
\end{aligned} \tag{2}$$

Now, if number of vehicles passed in t_1 be m_1, in t_2 be m_2 and so on. It can be written for general,

$$t_k = \frac{T \times m_k}{M} \tag{3}$$

Where, $M = m_1 + m_2 + \ldots + m_n$ total number of vehicles passed in T (time) and k =1, 2, . . ., n.

From Eqs. (2) and (3),

$$t_k = \frac{(R - 4\,(C + D)) \times m_k}{M} \tag{4}$$

For n inbound and outbound roads the Eq. (4) can be written as follows.

$$t_k = \frac{(R - n\,(C + D)) \times m_k}{M} \tag{5}$$

To check the accuracy rate of the proposed system 10 different videos were selected to run with the system. At first vehicles were counted manually, afterwards it was counted by proposed code. The accuracy rate is presented in Table 1. Same video footage was tested for several times with the proposed system and found identical result. The system works best when vehicles have space difference between them. In the test accuracy rate was ~85 % for these cases. When vehicles have very little space difference in between them, accuracy rate decreases. The test performs well also in night or low light condition while vehicle have their head lights on.

Conclusion and Future Work

Our method is concerned with a motion-based system for detecting and tracking multiple moving objects. Here the traffic load of different inward connecting roads to a junction was considered to make efficient traffic signal duration. Minimum signal duration for all the roads is also proposed, so that low traffic carrying road can at least have some time to pass over this traffic through junction point. In this method

Table 1 Accuracy rate of moving vehicle detection

Video serial	Actual count (manual count)	System output	Accuracy (%)
1	38	32	83
2	67	56	83
3	94	74	78
4	76	63	82
5	44	40	90
6	66	63	95
7	79	70	88
8	98	77	78
9	28	26	90
10	34	28	82
		Average accuracy	84.9

leverage in selecting speed level of vehicles by setting up parameters exists. It is possible to discard slow speed or non-motorized vehicles by this parameter if necessary. The success of this method heavily relies on camera position and steadiness of it. The method is tested in densely populated roads and lightly populated roads. In both cases the method worked seamlessly. Here all type of vehicles was considered as a unit. Any discrimination between small and big vehicle was not considered. Some adjustment with this issue can be done as future work of this proposed method.

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Analysis of the Lifetime of Wireless Sensor Networks for Underground Communications in Wet Sand

Abdelrahman Elleithy and Gonhsin Liu

Abstract

In this paper we present an advanced model for power consumption in communications and study its effect on the lifetime of the wireless sensor networks in underground communications for wet sand. The power communications model is incorporated in the life time model of wireless sensor networks. The life time model takes into consideration several parameters such as the total number of sensors, network size, percentage of sink nodes, location of sensors, the mobility of sensors, power consumption when nodes move and the power consumption of communications. The new model for power consumption in communications shows more accurate results about the lifetime of the sensor network in comparison with previously published results.

Keywords

Wireless sensors networks • Network lifetime • Models • Simulation

A Model for Lifetime of Wireless Sensors Networks

Wireless sensors have received increased attention in the past years due to their popularity and cost effectiveness when they are used in harsh environments. They have been used in many applications including military applications, environmental applications, health applications, and home applications. Although they are very cost effective and easily deployed in harsh environments, they are limited by the power available through their life cycle. Sensors are usually deployed with limited power which is depleted over their life cycle. Once their power is depleted, the sensors become dead and they are no more useful. An evaluation of the life cycle of a wireless sensor network is very essential to estimate how long a network can live and when the network and its sensors might be replaced or recharged if possible.

In this section we present a model for the lifetime of Wireless sensor networks based on a paper by [1]. The model takes different parameters that are used in literature. The following parameters are considered:

1. The time until the first sensor is drained of its energy [2];
2. The time until the first cluster head is drained of its energy [3];
3. The time there is at least a certain fraction β of surviving nodes in the network [4];
4. The time until all nodes have been drained of their energy [5];
5. K-coverage: the time the area of interest is covered by at least k nodes [6];
6. 100 % coverage
   (a) The time each target is covered by at least one node [7];
   (b) The time the whole area is covered by at least one node [8];
7. α-coverage

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_10, © Springer International Publishing Switzerland 2015

(a) The accumulated time during which at least α portion of the region is covered by at least one node [9];
(b) The time until the coverage drops below a predefined threshold α (until last drop below threshold) [10];
(c) The continuous operational time of the system before either the coverage or delivery ratio first drops below a predefined threshold [11];

8. The number of successful data-gathering trips [12];
9. The number of total transmitted messages [13];
10. The percentage of nodes that have a path to the base station [11];
11. Expectation of the entire interval during which the probability of guaranteeing connectivity and k-coverage simultaneously is at least α [6];
12. The time until connectivity or coverage are lost [14];
13. The time until the network no longer provides an acceptable event detection ratio [5];
14. The time period during which the network continuously satisfies the application requirement [15];
15. The minimum of t_1, t_2, and t_3 with t_1: time for cardinality of largest connected component of communication graph to drop below $c_1 \times n(t)$, t_2: time for n(t) to drop below $c_2 \times n$, t_3: time for the covered volume to drop below $c_3 \times ld$ [16].

Parameters Used in the Model

In this section we address parameters that were introduced in literature that can be used in a complete model for a wireless sensors networks lifetime. In earlier version of this model was introduced by Elleithy and Liu [17]. The following parameters are introduced:

1. The total number of available sensors
2. The set of all nodes those that are alive at time t
3. The set of nodes those that are active at time t
4. The set of nodes those that are active at any time in the time interval $[t - \Delta t, t]$
5. The set of sink nodes or base stations B(t) is defined to be a subset of the existing nodes SY
6. The ability of nodes m1 and m_n to communicate at a time t
7. The ability of two nodes to communicate in the time interval $[t - \Delta t, t]$ such that the links between consecutive hops become available successively within the time interval (support for delay tolerant networking)
8. The set of target points to be sensed by the network
9. The area that is covered by all sensors of a certain type y, at a time t.

A New Model of Power Consumption in Communications of Wireless Sensor Networks

In this section we use the power consumption of the communications in wireless sensor networks presented by Elleithy et. al. [18].

Simulation Setup

The following parameters values are used in the simulation carried in this paper:
$k = 3.5$
$G_1 = 1{,}000$ W
$B = 10$ KHz
$P_{mix} = 30.3$ mW
$P_{LNA} = 20$ mW
$P_{maxt} = 250$ mW
$T_{tr} = 5$ μ-s
$T = 0.1$ s
drainEfficiency $= 0.35$
$L = 2$ Kbit
$P_{syn} = 50$ mW
$P_{IFA} = 3$ mW
$P_{filt} = 2.5$ mW
$P_{filr} = 2.5$ mW
$M_l = 10{,}000$ W
$P_b = 0.001$
$V_{dd} = 3$ V
$L_{min} = 0.5$ μm
$n_1 = 10$
$n_2 = 10$
$f_{cor} = 1$ MHz
$I_0 = 10$ μA
$C_p = 1$ pF
psdensity $= -83$ dBm
beta $= 1$;

Simulation Results

In this section we show the results of the power consumption in communications where the total energy per bit is calculated versus time spent on mode for different parameters.

In Fig. 1, the Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different distances at a packet size of 2,000 bits. The figure shows the energy consumption from 1 to 5 m. As the distance of transmission increases, the total energy per bit increases. Also, as the time spent in the on mode/Total Time increases, the energy per bit decreases.

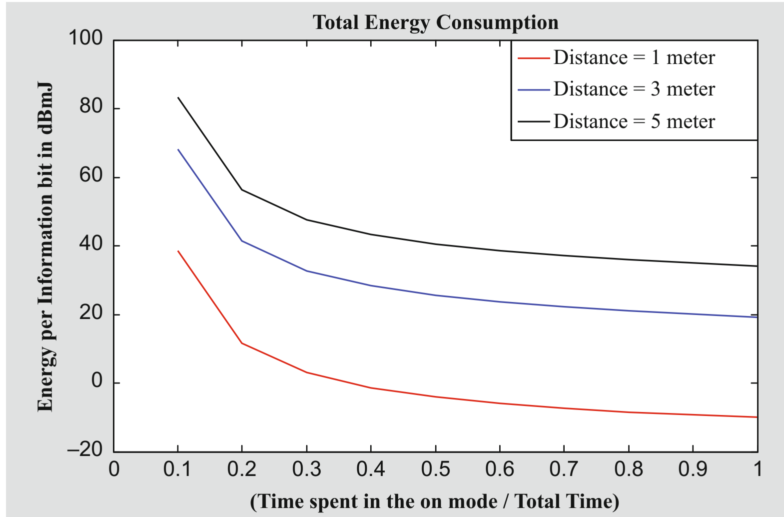

Fig. 1 Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different distances at a packet size of 2,000 bits

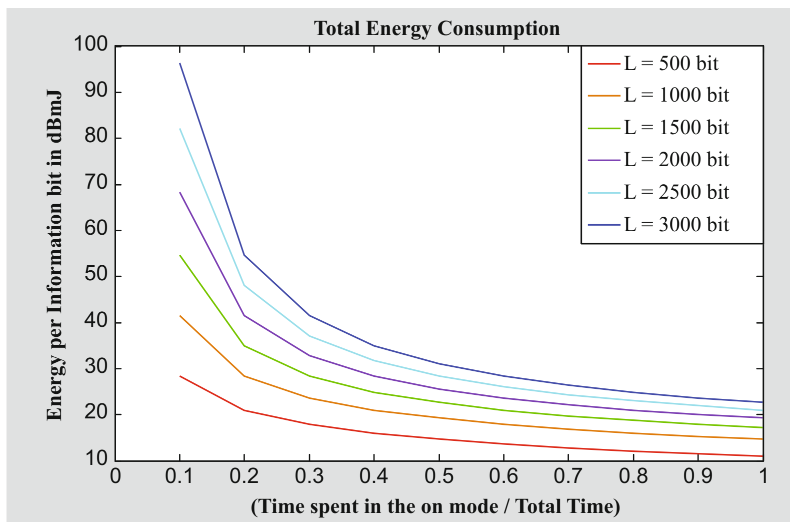

Fig. 2 Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different packet sizes at a distance of 3 m

In Fig. 2, the Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different packet size for a path loss exponent of 3.5 and a distance of 3 m. As the packet size increases, the total energy per bit increases. Also, as the time spent in the on mode/Total Time increases, the energy per bit decreases.

In Fig. 3, the Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different path loss exponent (K) at a packet size of 2,000 bits and a distance of 3 m. As the path loss exponent increases, the total energy per bit increases. Also, as the time Spent in the On mode/Total Time increases, the energy per bit decreases.

In Fig. 4, the Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different bandwidth at a packet size of 2,000 bits and a distance of 3 m. As the bandwidth decreases, the total energy per bit increases. Also, as the time spent in the on mode/Total Time increases, the energy per bit decreases.

In Fig. 5, the Total Energy Consumption per bit versus Time Spent in the on mode/Total Time for different drain efficiency at a packet size of 2,000 bits and a distance of 3 m. As the drain efficiency decreases, the total energy per bit increases. Also, as the time spent in the on mode/Total Time increases, the energy per bit decreases.

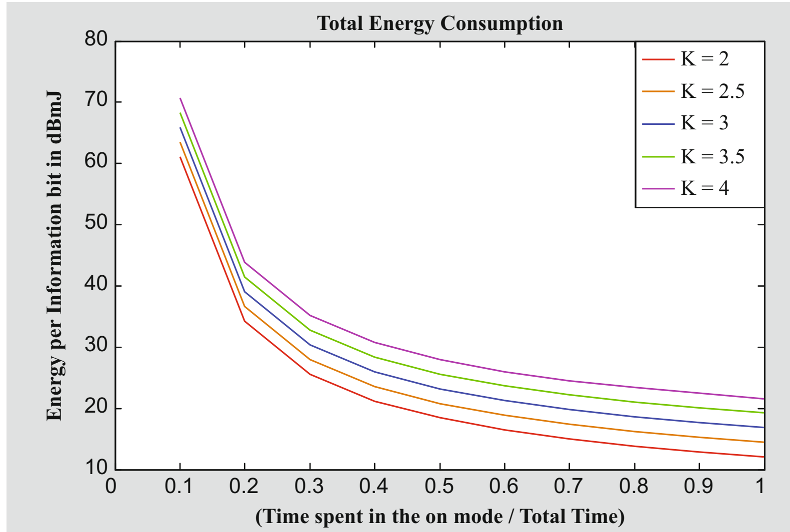

Fig. 3 Total Energy Consumption per bit versus Time Spent in the On mode/Total Time for different path loss exponent (k) at a distance of 3 m

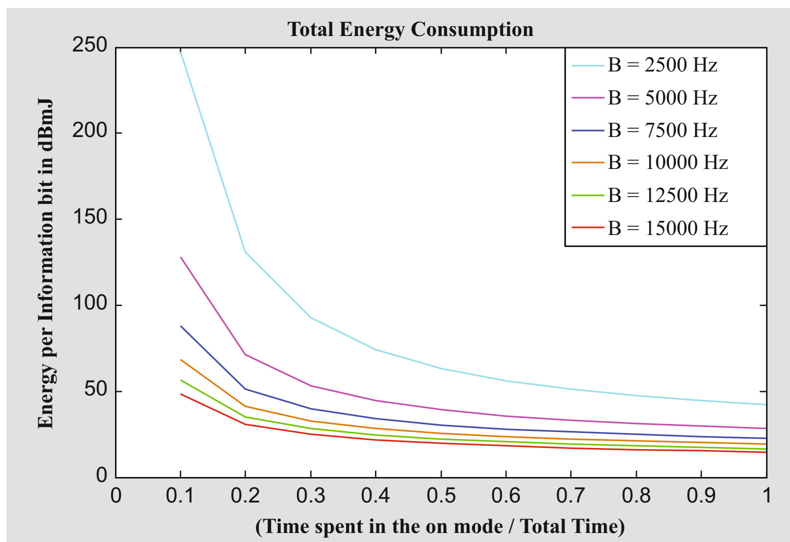

Fig. 4 Total Energy Consumption per bit versus Time Spent in the On mode/Total Time for different bandwidth at a distance of 3 m

Simulation of the Lifetime of WSN

In this section we present the following model for simulating the life time in a WSN. Although the Matlab code is developed with the following default parameters, it can be modified for other values.

Assumptions:

1. Total number of sensors: Input parameter
2. Network size: Input parameter in meters
3. Percentage of sink sensors: Input parameter (between 0 and 1)
4. Location of sensors: Randomly generated over the network
5. Initial power per sensor: Random between 0 and 100 units
6. Movement of sensors:
   (a) Sensors can move in the x direction for a random value between −5 and +5
   (b) Sensors can move in the y direction for a random value between −5 and +5
   (c) The total value a senor moves is: $d = \sqrt{x^2 + y^2}$
7. Power Consumption:
   (a) Communications: As given in [18]
   (b) Movement: d unit for the moving sensor as calculated in 6
8. Stopping criteria (we consider the network dead if one of the following conditions satisfied):

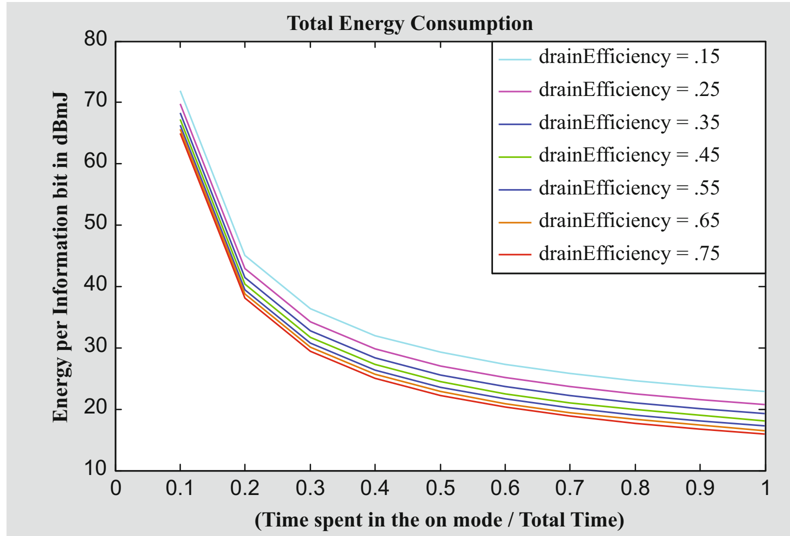

Fig. 5 Total Energy Consumption per bit versus Time Spent in the On mode/Total Time for different drain Efficiency at a distance of 3 m

(a) Percentage of available power to total power: less than 25 %
(b) Percentage of alive sensors to total sensors: less than 25 %
(c) Percentage of alive sink sensors to total sink sensors: less than 5 %

Scenario Number 1

In this scenario the following parameters are considered:

1. Total number of sensors: 150
2. Network size: 3,000 m × 3,000 m
3. Percentage of sink sensors: 15.6 %
4. Maximum Communications power: 12.98 units
5. Packet size (k): 32

Figure 6 shows the status of Initial and final network. Figures 7, 8, 9, and 10 show different network performance parameters over the life cycle of the network. In this scenario the network was considered dead because after 20 cycles the following status is reached:

1. Percentage of available power to total power: 1,757/7,233 = 24.3 %
2. Percentage of alive sensors to total sensors: 75/150 = 50 %
3. Percentage of alive sink sensors to total sink sensors: 14/20 = 70 %

The network is considered dead because condition number (1) is satisfied where the percentage of available power to total power is 24.3 % which is less than the 25 % criteria.

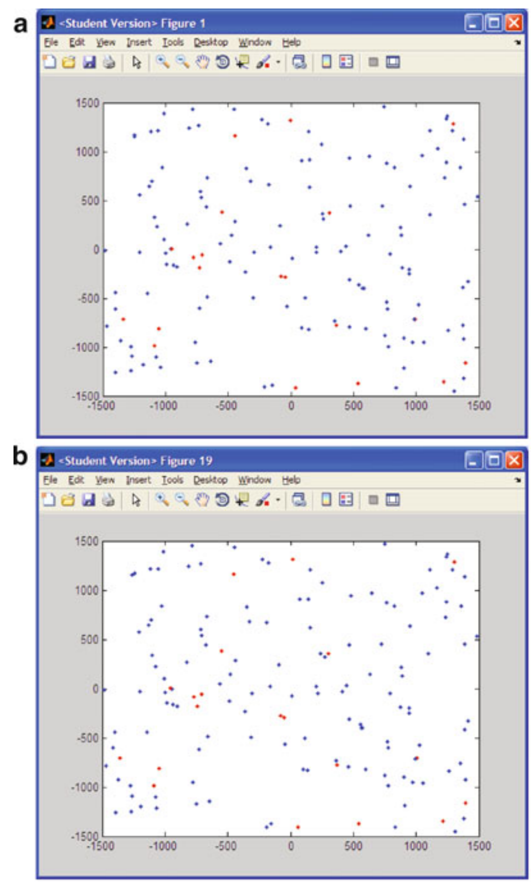

Fig. 6 Status of Initial and final network. (**a**) Initial Network. (**b**) Final Network. *Note*: sink nodes are *red*

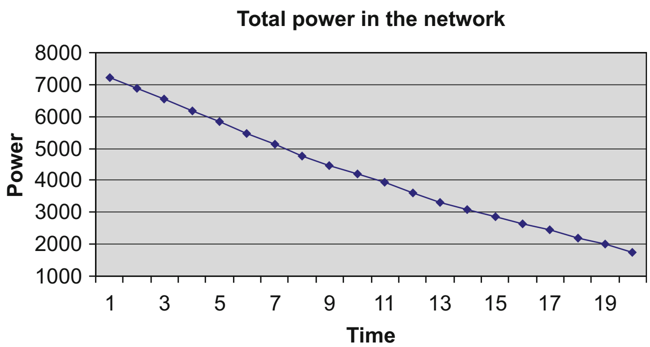

Fig. 7 Total Power in the network over the life cycle

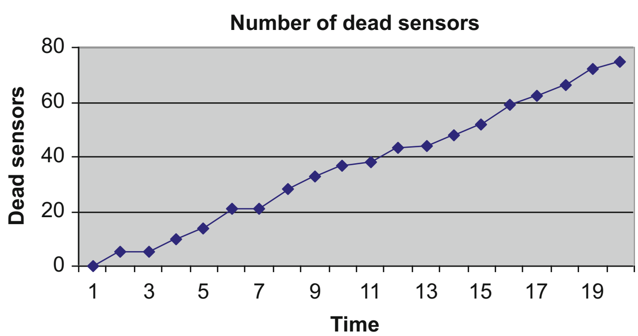

Fig. 8 Numbers of dead sensors over the life cycle

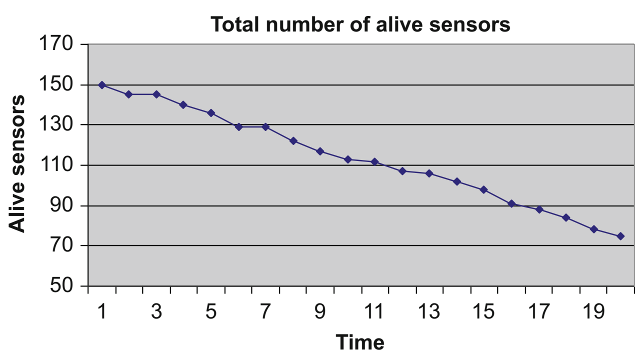

Fig. 9 Total number of alive sensors over the life cycle

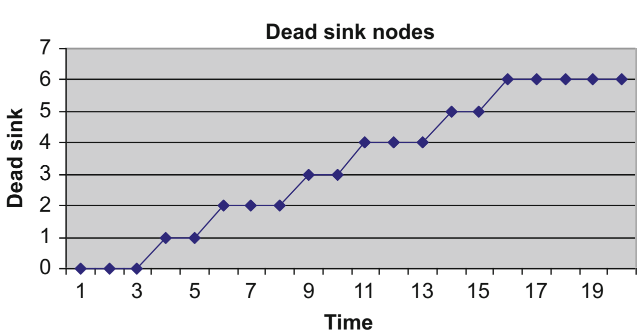

Fig. 10 Total number of dead sinks nodes over the life cycle

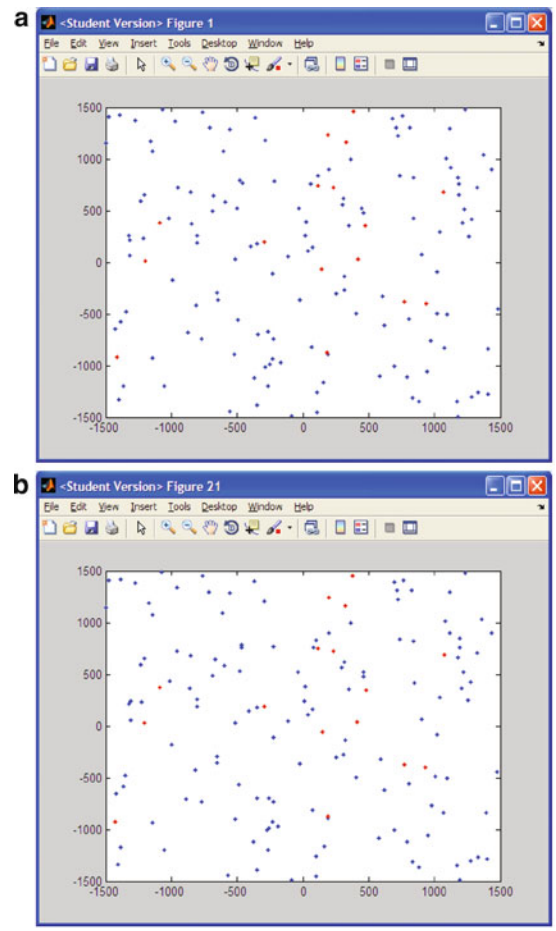

Fig. 11 Shows the status of Initial and final network. (**a**) Initial Network. (**b**) Final Network. *Note*: sink nodes are *red*

Scenario Number 2

In this scenario the following parameters are considered:

1. Total number of sensors: 150
2. Network size: 3,000 m × 3,000 m
3. Percentage of sink sensors: 15.6 %
4. Maximum Communications power: 11.73 units
5. Packet size (k): 32

Figure 11 shows the status of Initial and final network. Figures 12, 13, 14, and 15 show different network performance parameters over the life cycle of the network. In this scenario the network was considered dead because after 22 cycles the following status is reached:

1. Percentage of available power to total power: 1,641/ 6,880 = 23.9 %
2. Percentage of alive sensors to total sensors: 61/ 150 = 41 %
3. Percentage of alive sink sensors to total sink sensors: 11/ 16 = 68.8 %

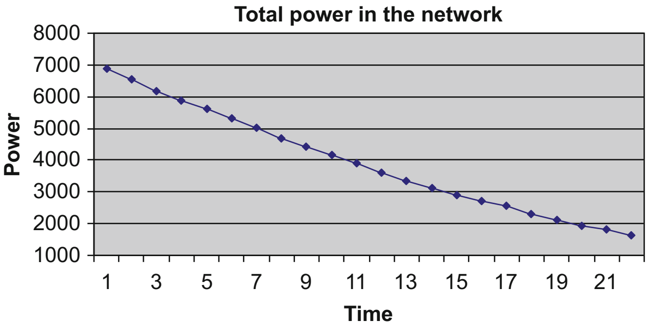

Fig. 12 Total Power in the network over the life cycle

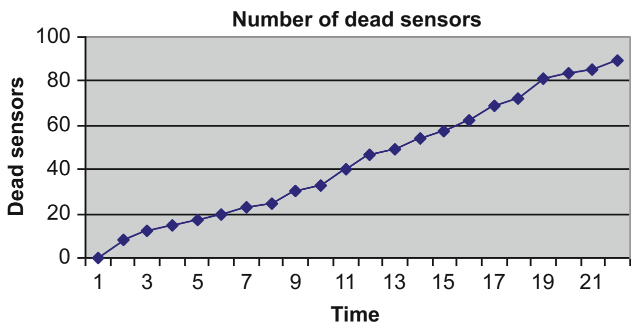

Fig. 13 Numbers of dead sensors over the life cycle

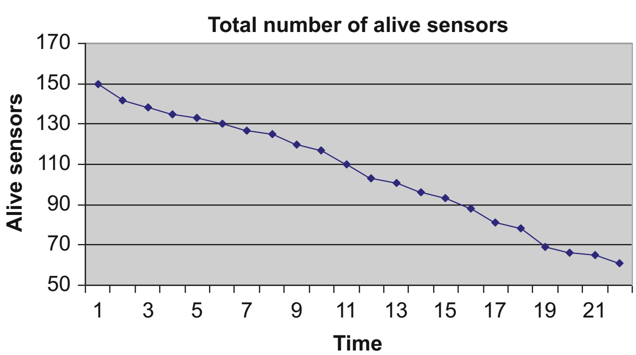

Fig. 14 Total number of alive sensors over the life cycle

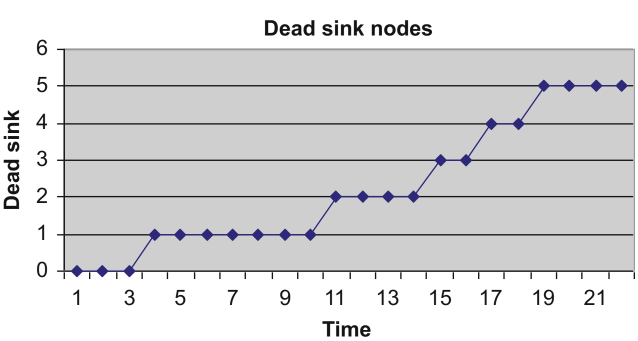

Fig. 15 Total number of dead sinks nodes over the life cycle

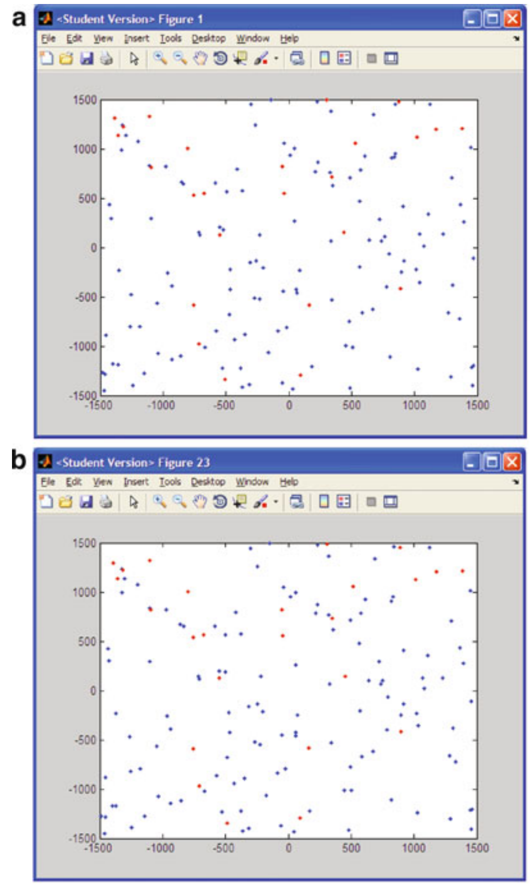

Fig. 16 Status of Initial and final network. (**a**) Initial Network. (**b**) Final Network. *Note*: sink nodes are *red*

The network is considered dead because condition number (1) is satisfied where the percentage of available power to total power is 23.9 % which is less than the 25 % criteria.

Scenario Number 3

In this scenario the following parameters are considered:

1. Total number of sensors: 150
2. Network size: 3,000 m × 3,000 m
3. Percentage of sink sensors: 15.6 %
4. Maximum Communications power: 6.74 units
5. Packet size (k): 32

Figure 16 shows the status of Initial and final network. Figures 17, 18, 19, and 20 show different network performance parameters over the life cycle of the network. In this scenario the network was considered dead because after 24 cycles the following status is reached:

1. Percentage of available power to total power: 1,928/7,897 = 24.4 %

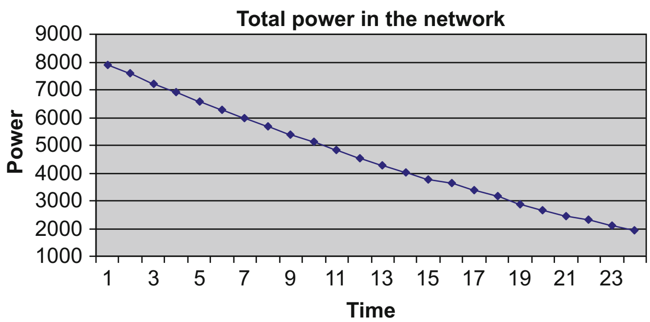

Fig. 17 Total Power in the network over the life cycle

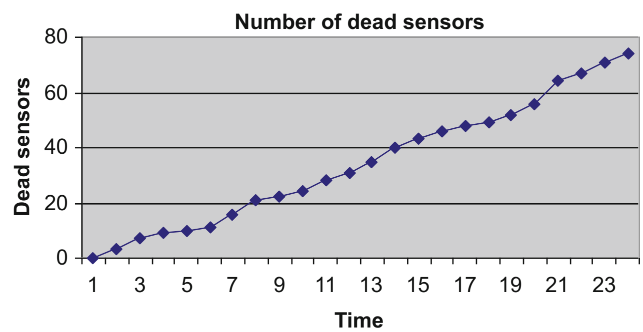

Fig. 18 Numbers of dead sensors over the life cycle

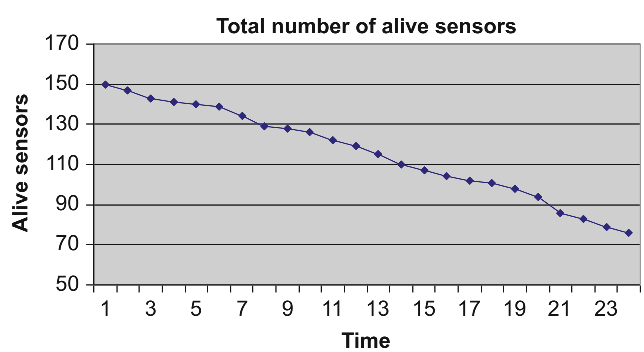

Fig. 19 Total number of alive sensors over the life cycle

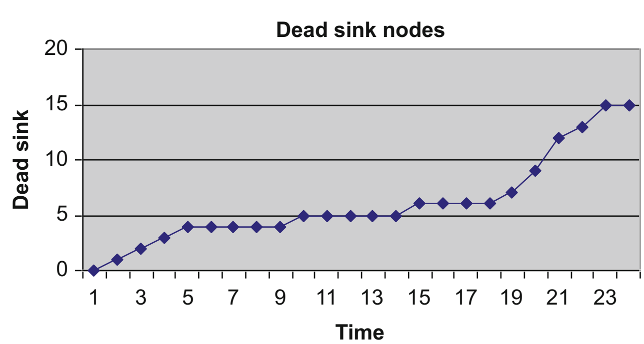

Fig. 20 Total number of dead sinks nodes over the life cycle

2. Percentage of alive sensors to total sensors: 76/150 = 51 %
3. Percentage of alive sink sensors to total sink sensors: 10/25 = 40 %

The network is considered dead because condition number (1) is satisfied where the percentage of available power to total power is 24.4 % which is less than the 25 % criteria.

Scenario Number 4

In this scenario the following parameters are considered:

1. Total number of sensors: 150
2. Network size: 3,000 m × 3,000 m
3. Percentage of sink sensors: 15.6 %
4. Maximum Communications power: 1.75 units
5. Packet size (k): 32

Figure 21 shows the status of Initial and final network. Figures 22, 23, 24, and 25 show different network performance parameters over the life cycle of the network. In this scenario the network was considered dead because after 27 cycles the following status is reached:

1. Percentage of available power to total power: 1,754/7,533 = 23.3 %
2. Percentage of alive sensors to total sensors: 67/150 = 45 %
3. Percentage of alive sink sensors to total sink sensors: 14/24 = 58.3 %

The network is considered dead because condition number (1) is satisfied where the percentage of available power to total power is 23.3 % which is less than the 25 % criteria.

Scenario Number 5

In this scenario the following parameters are considered:

1. Total number of sensors: 150
2. Network size: 3,000 m × 3,000 m
3. Percentage of sink sensors: 15.6 %
4. Maximum Communications power: 0.50 units
5. Packet size (k): 32

Figure 26 shows the status of Initial and final network. Figures 27, 28, 29, and 30 show different network performance parameters over the life cycle of the network. In this scenario the network was considered dead because after 27 cycles the following status is reached:

1. Percentage of available power to total power: 1,850/7,591 = 24.4 %
2. Percentage of alive sensors to total sensors: 76/150 = 51 %

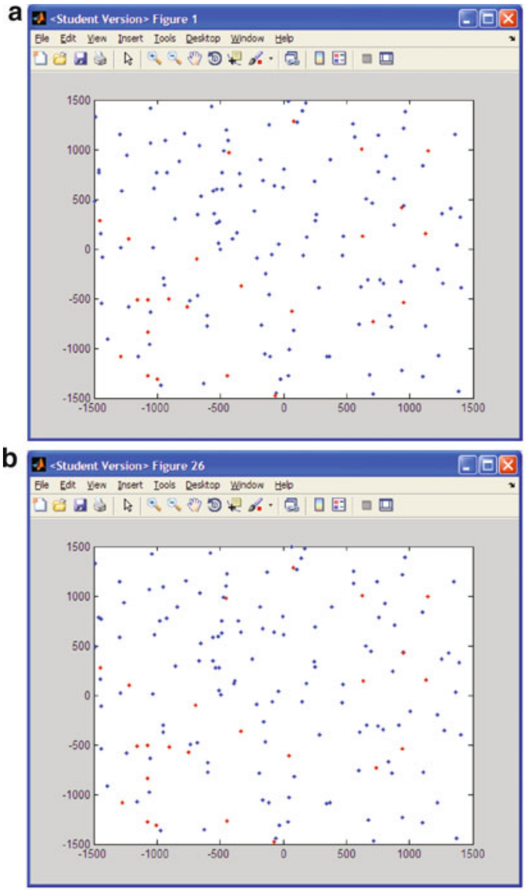

Fig. 21 Status of Initial and final network. (**a**) Initial Network. (**b**) Final Network. *Note*: sink nodes are *red*

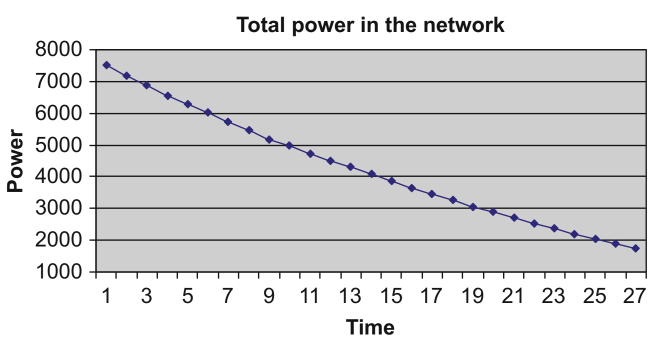

Fig. 22 Total Power in the network over the life cycle

3. Percentage of alive sink sensors to total sink sensors: 9/23 = 39 %

The network is considered dead because that condition number (1) is satisfied where the percentage of available power to total power is 24.4 % which is less than the 25 % criteria.

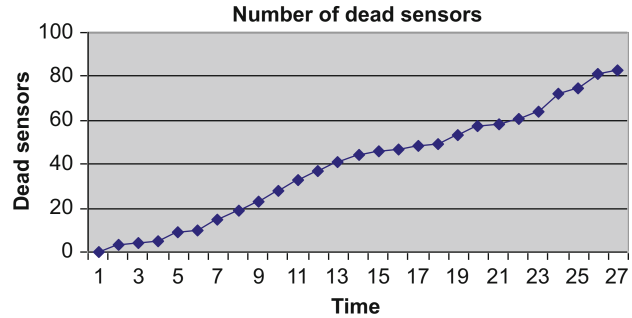

Fig. 23 Numbers of dead sensors over the life cycle

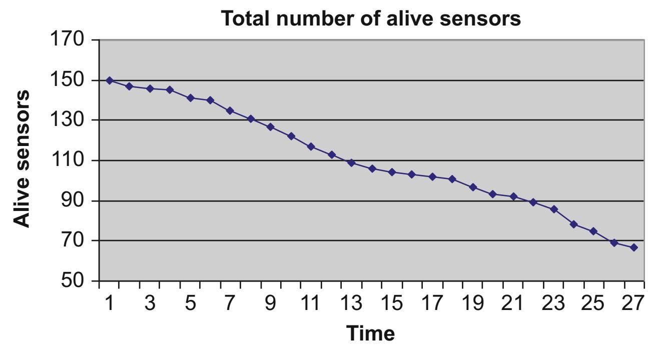

Fig. 24 Total number of alive sensors over the life cycle

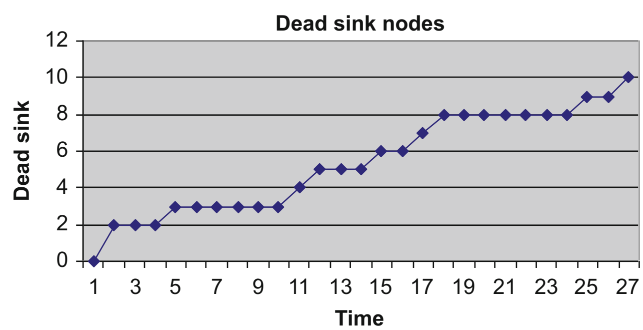

Fig. 25 Total number of dead sinks nodes over the life cycle

Conclusions

A model to estimate the lifetime of wireless sensors networks is necessary to help the designers of the network to design their network by adjusting important parameters such as initial power, number of sensors, number of sink sensors, etc.

In this paper we present a model for the power consumed in communications of the wireless sensors in underground communications for wet sand. The presented model

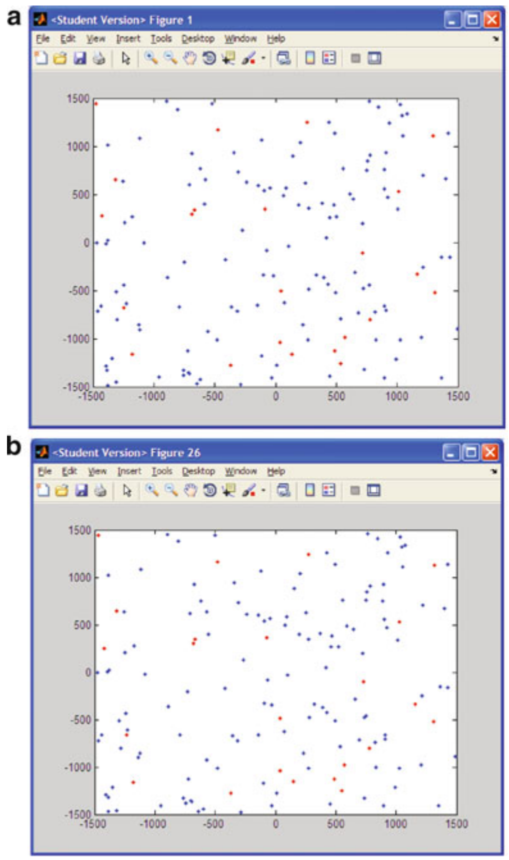

Fig. 26 Status of Initial and final network. (**a**) Initial Network. (**b**) Final Network. *Note*: sink nodes are *red*

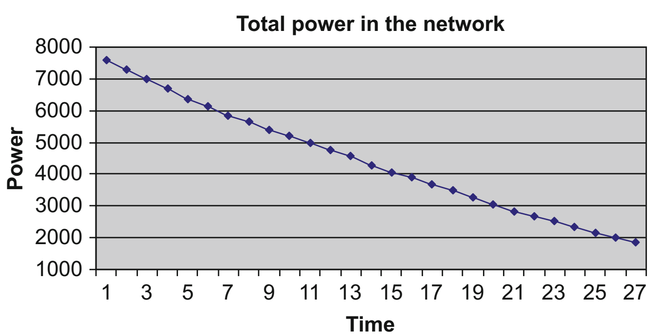

Fig. 27 Total Power in the network over the life cycle

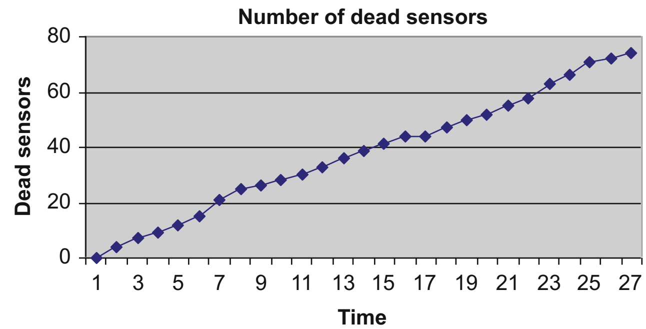

Fig. 28 Numbers of dead sensors over the life cycle

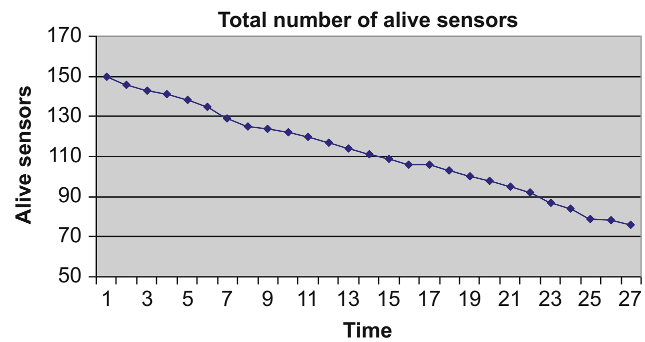

Fig. 29 Total number of alive sensors over the life cycle

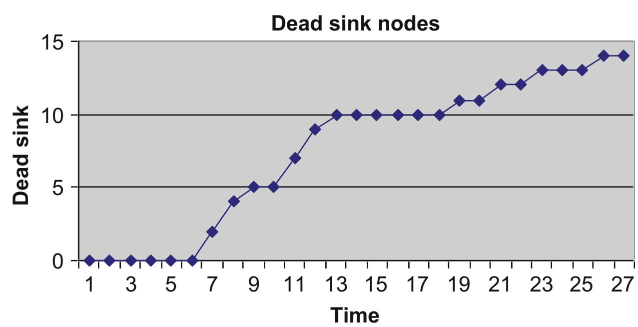

Fig. 30 Total number of dead sinks nodes over the life cycle

for power communications takes into consideration parameters such as power consumption for the active mode, power consumption for the sleep mode, power consumption for the transient mode, transmission period, transient mode duration, sleep mode duration, and active mode duration.

In order to examine the validity of our model, we have tested it for many lifetime scenarios. In this paper we are presenting five of these scenarios. The following parameters are used: total number of sensors, network size as defined by its width and length, and the percentage of sink sensors. In each scenario, we have evaluated both the total power in the network over the life cycle, number of dead sensors over the life cycle, total number of alive sensors over the life cycle, and number of dead sinks nodes over the life cycle.

The results presented in this paper show the importance of such a simulator from the designer perspective. The model can be used as a design tool as well as a research tool to evaluate the network performance. In the future, we would expect to extend the work presented in this model to include other parameters and modes of operations for underwater and underground wireless sensor networks.

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About One Way to Discover Formative Assessment Cheating

Jan Genci

Abstract
Massification of Slovak higher education system requires new ways how to organize educational process. That was why we decided to introduce formative assessment in our course. We immediately registered attempts to cheat during this type of assessment. This inspired us to find effective ways how to discover such attempts. Using timestamps associated with each activity carried out in the Moodle tests helped us to gather relevant data and analyze them. Paper presents approaches we proposed to recognize students suspect for cheating.

Keywords
Assessment • Data analytics • e-learning • Formative assessment • Moodle • Moodle test • Test statistics

Introduction

Two types of assessment are widely presented in a pedagogical practice—formative and summative. While a goal of the summative assessment is to measure performance of students, the formative assessment, according to [1] (based on [2]) represents "all those activities undertaken by teachers, and/or by students, which provide information to be used as feedback to modify the teaching and learning activities in which they are engaged."

Enormous changes in Central European higher education area during last 15–20 years shifted education from an elite to mass or, may be, even universal type of education [2, 3]. According [4]: "The number of students in public . . . and private colleges . . . increased from 62,103 in 1990 to 196,886 in 2006, which is about three-fold increase".

Trow [3] and Prudký [5] claim, that transition period is specific for attempts to run the new system using old methods. It seems to us, the claim is true not only regarding management methods, but is valid even when dealing with methods used in the education process. We have discussed this topic in more details in [6]. Just to summarize [6] shortly—average intellectual potential of students is reduced from around IQ 125 to IQ 105; correlated self-motivation to study is lowered as well. At the present, self-motivation of our students seems to be very low.

Moreover, cheating in our region was always quite high. Advent of new technological achievement (e.g. mobile devices) and wide spread of social networks cause further increase of cheating.

To reduce some of the above mentioned problems, we decided to implement some new approaches—strict application of credit transfer system. We oblige students to earn their credits step-by-step during the term by set of activities strictly defined at the beginning of the term (which was not usual in the past and is still not usual in all courses). One type of these activities is represented by preparatory tests (formative assessment), which should be carried out by students during the week before seminar.

The goal of the test is to motivate students to prepare for seminars by filling open item (questions) forms/tests tied with related topics. Students are required to prepare themselves for seminars by answering questions specified

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_11, © Springer International Publishing Switzerland 2015

in the test item. Very quickly, however, students published correct answers for test item using their social networks. At the first stage, we were able to identify cheating students by the average completion time. Moreover, monitoring students' social network gave us possibility to recognize students, who published correct answers (not directly). Of course, they learnt and realized all this facts very quickly. That was the reason why some of the students decided to cheat in a more sophisticated way and students, as community, closed their network for public. Based on these facts we decided to evaluate time based data tied with each student's quiz carried out in the Moodle LMS.

Related Works

Google search on the term "moodle assessment statistics" provides about 755 thousand results, while for the term "moodle assessment cheating" the result reaches up to about ten millions. Scopus search on the first term provides four results only, on the second one only two. Searching term "moodle assessment cheating detection" provides about three millions results on Google, however in Scopus, no results were provided.

The most of the search results were irrelevant, relevant result were mostly devoted to topics related to cheating prevention; in the case of the term "... cheating detection" relevant results were mostly about plagiarism detection.

The most valuable source of information related to interpretation of test data seems to be Moodle documentation related to quiz statistics [7, 8, 9].

Because we were not able to find any similar approach, based on the results of our brief search, we dare to claim that our approach is quite unique, perhaps because of the specific use of Moodle tests for formative assessment.

Description of a Problem

By formative assessment we tried to address very low motivation of our students regarding preparation for seminars in the Operating Systems subject. Seminars of the course are devoted to the topics oriented to System Programming in the UNIX/Linux environment—starting from file manipulation and device handling, continued by process management and inter-process communication (pipes, signals, shared memory and semaphores) and ended with network communication using TCP/IP and TCP/UDP protocols (sockets). Our students have at their disposal a detailed lab manual for each seminar, which should guide them through the particular topic as they revise. For each topic we designed a set of open answer quiz items based on study materials (lab manual, LINUX man pages, books) which cover relevant parts of the topic we work on during seminar. Students are required to complete quiz 'at home' during the week before seminar. They are not limited nor as to the way how to complete it or the time when to start and when to proceed; the only limitation is the deadline of quiz, which has to be filled in until midnight of the day before the seminar. It means that students can proceed with quiz in accordance with their preparation for the topic. Every quiz item is provided in the adaptive mode without or with very small penalty, so student can try to find the correct answer typing the answer several times. Because of logical sequence of terms, concepts and principles of quiz items are not shuffled. We require 75 % for passing the quiz.

Because the items in the Moodle LMS are processed by string matching, open answer items are 'sensitive' to exact answer—which requires that all relevant synonyms are marked as correct. We overcome this by monitoring students' answers in the quiz items statistics and adding relevant new answers as correct answers (however, this was an issue mainly at the first stage of the project).

When we started with formative tests, students realized very quickly that the test is common for all students and some of them published all answers at their social network webpage. Just to be honest, we have to say that the discussion regarding particular items was, then, welcomed by us and we did not limit it.

Because of the open nature of the quiz items, to find a person who published answers was quite easy, based on the set of answers. Moreover, students who were cheating, did not realize, that they can be identified based on time spend to complete test (some of them completed test in several minutes). Of course, students very quickly realized it and started to interrupt the quiz for several hours, even days (what was fully legal, according to rules defined by us) in between quiz items.

At that time we realized/noticed time information, which is associated with every student's attempt (Fig. 1). Every answer, send to Moodle, is saved and timestamped. That means that in the systems there is a detailed history of activities carried out by the student during the quiz completion. We were limited only by fact, that in some quizzes items were grouped by 5 or 10 per page and students were able to send them to the system at once—all 5/10 items were timestamped with the same value (which was also a bit suspicious, because more convenient way seems to be to deal with each item independently).

Mining the Data

Moodle stores data in relational database. Getting data directly from the database requires authenticated access to the database, which is usually granted to administrator of the


Hodnotu ktorej zložky i-nodu reprezentovanú záznamom (štruktúrou) "struct stat" ovplyvní úspešné vykonanie služby link()?

Odpoveď: st_nlink

Vložte komentár, alebo prepíšte body

Správny

Známky za odoslaný test: 1/1. S predošlými trestnými bodmi je výsledok 0.6/1.

História odpovedí:

#	Akcia	Odpoveď	Čas	Hrubé skóre	Známka
1	Oznámkovať	ino_t	17:18:51 dňa 20/02/13	0	0
2	Oznámkovať	stino	17:19:19 dňa 20/02/13	0	0
3	Oznámkovať	ino t	17:19:40 dňa 20/02/13	0	0
4	Oznámkovať	nlink t	17:20:24 dňa 20/02/13	0	0
5	Oznámkovať	st_nlink	12:41:37 dňa 24/02/13	1	0.6
6	Zatvoriť	st_nlink	12:41:37 dňa 24/02/13	1	0.6

Fig. 1 Timestamps associated with stundet's attempts

```
<ATTEMPTS>
   <ATTEMPT>
   <ID>72431</ID>
   <UNIQUEID>72431</UNIQUEID>
   <USERID>9516</USERID>
   <ATTEMPTNUM>1</ATTEMPTNUM>
   <SUMGRADES>65.1</SUMGRADES>
   <TIMESTART>1361182737
     </TIMESTART>
   <TIMEFINISH>1361623146
     </TIMEFINISH>
...
  <STATES>
    <STATE>
      <ID>2825275</ID>
      <QUESTION>15518
        </QUESTION>
      <SEQ_NUMBER>0
        </SEQ_NUMBER>
      <ANSWER>duplication
        </ANSWER>
      <TIMESTAMP>1361182737
      </TIMESTAMP>
      <EVENT>0</EVENT>
      <GRADE>0</GRADE>
      <RAW_GRADE>0</RAW_GRADE>
      <PENALTY>0</PENALTY>
    </STATE>
```

Fig. 2 Content of XML File—data about answers

LMS only and is beyond the rights granted to ordinary user of LMS Moodle. That was the reason why we used Backup functionality provided by Moodle to each course administrator. The backup data is provided as an XML file with all data specified by the course administrator during the backup procedure. Fragment of XML file related to the data about answers is presented in Fig. 2.

The data we obtained in the XML file was transformed to an ordinary text file in CSV (comma separated values) format using XSL transformation. Final data was stored in SQLite database. That gives us possibility to manipulate the data using standard SQL statements.

Last step in the process consists of producing of CSV files which serve as input to EXCEL, where we made final data evaluation. This process was accomplished by simple BASH scripts in the Linux environment, with an access to SQLite database and some additional processing using program 'awk'.

Database Schema

Data extracted from Moodle database is stored in a local SQLite database using following database schema:

```
CREATE TABLE mintime(
  IDST INT, MinTime INT);
CREATE TABLE responses(
  ID INT, question INT, Time INT,
  Attempt INT, Start INT, End INT,
  Duration INT);
CREATE TABLE Questions(
  ID, Cathegory, Name);
CREATE TABLE groups(
  GroupID INT, Name VARCHAR, MembID);
CREATE TABLE students(
  ID INT, FirstName, LastName);
```

Data stored in the database is queried by standard SQL SELECT statements according to required reports.

Evaluation of Data

We evaluated several outputs, from which the most interesting seems to be the data ordered by students and timestamp (Fig. 3). We calculated difference between each timestamp and the beginning of the test (column TM_delta) and the difference between two consequent attempts (column TM_diff).

Figure 3 provides random values in column L. In the Fig. 4 we can see (column TM_diff, again) another set of

IDQ	CategorQ	NameQ	TMPSTMP	Attempt	Duration	TM delta	TM diff.
15516	01-Services-All	UND-01-02	1361813562	1	6435	167	167
15517	01-Services-All	UND-01-03	1361813583	1	6435	188	21
15518	01-Services-All	UND-01-04	1361813600	1	6435	205	17
15519	01-Services-All	UND-01-05	1361813619	1	6435	224	19
15520	01-Services-All	UND-01-06	1361813884	1	6435	489	259
15521	01-Services-All	UND-01-07	1361814062	1	6435	667	178
15522	01-Services-All	UND-01-08	1361814316	1	6435	921	248
15524	01-Services-All	UND-01-10	1361814402	1	6435	1007	86
15525	01-Services-All	UND-01-11	1361814432	1	6435	1037	30
15526	01-Services-All	UND-01-12	1361814571	1	6435	1176	81

Fig. 3 Timestamp data of particular student and consequent derived data

IDQ	CategorQ	NameQ	TMPSTMP	Attempt	Duration	TM delta	TM diff.
18648	k-02-Files-Understn	Post-Q01-001	1361911487	1	42594	728	32
18649	k-02-Files-Understn	Post-Q01-002	1361911893	1	42594	1134	406
18649	k-02-Files-Understn	Post-Q01-002	1361912791	1	42594	2032	898
18649	k-02-Files-Understn	Post-Q01-002	1361952259	1	42594	41500	39468
18650	k-02-Files-Understn	ost-Q01-003-	1361952259	1	42594	41500	0
18651	k-02-Files-Understn	ost-Q01-004-	1361952259	1	42594	41500	0
18652	k-02-Files-Understn	Post-Q03_001	1361952259	1	42594	41500	0
8464	k-02-Files-Understn	ost-Q04-1_1	1361952259	1	42594	41500	0
18653	k-02-Files-Understn	ost-Q09-1_0	1361952259	1	42594	41500	0
18654	k-02-Files-Understn	ost-Q10-V2-0	1361952259	1	42594	41500	0
14875	Week-02-Files-II	FL2-01	1361952259	1	42594	41500	0
14876	Week-02-Files-II	FL2-02	1361952259	1	42594	41500	0
14877	Week-02-Files-II	FL2-03	1361952318	1	42594	41559	59
14878	Week-02-Files-II	FL2-04	1361952318	1	42594	41559	0
14879	Week-02-Files-II	FL2-05	1361952318	1	42594	41559	0
14880	Week-02-Files-II	FL2-06	1361952399	1	42594	41640	81
14881	Week-02-Files-II	FL2-07	1361952399	1	42594	41640	0
14882	Week-02-Files-II	FL2-08	1361952399	1	42594	41640	0
14883	Week-02-Files-II	FL2-09	1361952399	1	42594	41640	0
14884	Week-02-Files-II	FL2-10	1361952399	1	42594	41640	0
14885	Week-02-Files-II	FL2-11	1361952399	1	42594	41640	0
14886	Week-02-Files-II	FL2-12	1361952399	1	42594	41640	0
14887	Week-02-Files-II	FL2-13	1361952470	1	42594	41711	71

Fig 4 Suspicious data (column TM_diff)

values (data for another student). The value—39,468—means more than 10 h break in the student's activity (we tolerate it). Next values—block of zeroes—means, that student filled in nine items during 59 s, what means 6 s per answer in average. Students with such data pattern were put on the list of suspicious students.

Figure 5 shows six variants of time difference profiles (intervals between attempts) in another test. Profiles A and B (columns A, B) seem to be regular profiles. The only interesting value in profile A is the row 37, which shows, that student interrupted the test for approx. 18 h. Values in profile C seems to be correct also, except for values in rows 31–35—they seem to be too low to read the question, find the correct answer and type it on the keyboard. On the other hand, we consider profile F to be very suspicious. The first answer was provided in 30 min, after that, the next three

	A	B	C	D	E	F
1	TM diff.	TM diff.	TM diff.	TM diff.	TM diff.	TM diff.
2	448	127	537	29	0	1800
3	196	66	260	39	0	0
4	276	176	28	20	0	0
5	51	539	173	13	0	30
6	324	115	76	48	302	17
7	569	79	30	14	0	19
8	193	48	26	17	0	22
9	346	56	22	11	84	9
10	57	35	22	16	66	6
11	39	41	31	12	38	5
12	35	39	5	11	92	5
13	82	41	120	12	53	5
14	14	15	30	11	12	205
15	27	33	4	4	21	3
16	6	101	35	5	11	2
17	9	47	110	8	12	87
18	76	35	63	9	18	7
19	58	40	31	14	16	9
20	64	33	7	10	4	7
21	100	34	21	18	17	7
22	28	65	7	7	90	38
23	16	22	40	2	0	0
24	12	28	55	6	0	0
25	21	21	22	7	0	0
26	6	32	28	5	0	0
27	35	34	39	9	82	30
28	102	41	53	9	0	0
29	31	40	7	10	0	0
30	58	14	37	7	0	0
31	17	37	5	8	0	0
32	10	21	6	6	88	94
33	14	31	8	8	0	0
34	12	30	5	7	0	0
35	6	23	5	5	0	0
36	32	13	15	9	0	0
37	65370	32	84	13	65	1006
38	78	68	38	31	0	0
39	15	48	36	7	0	0
40	33	43	124	14	0	0
41	73	21	25	7	0	0
42	17	23	106	12	157	18
43	74	50	46	11	0	0
44	85	171	93	11	0	0
45	51	71	21	10	0	0
46	27	14	33	9	32	0

Fig. 5 Various profiles for intervals between attempts

	D	E	F	G	H	I	J	K	L	M	N	O	P	Q	R	S	T	U	V	W	X	Y	Z	AA	AB	AC	AD	AE	AF	AG	AH	AI
1	Group	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31
5	2013_KK_UT_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
6	2013_KK_UT_1050	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
7	2013_JL_UT_1650	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
8	2013_JL_UT_1510	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
9	2013_JL_UT_1650	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
10	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
11	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
12	2013_JL_UT_1650	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
13	2013_JL_UT_1510	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
14	2013_KK_STR_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	6	7
15	2013_JL_UT_1510	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	2	3	4	5	5	5
16	2013_JL_UT_1510	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	10	12	15	18
17	2013_JL_UT_1650	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	9	10	15
18	2013_KK_UT_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	24	26
19	2013_JL_UT_1650	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	2	3	5	5	6	6	6	7	8	8	8	9	12
20	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	13	16
21	2013_MV_UT_0730	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	7	13	19	24
22	2013_JG_PO_0910	8	8	9	10	11	12	13	13	13	14	15	17	17	17	19	19	20	21	24	25	29	31	31	32	33	35	36	40	40	41	55
23	2013_MV_STV_1555	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	4	6	6	7	8	9	12	12	12	13	13	14	15
24	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
25	2013_KK_STR_0910	0	0	0	0	0	2	2	4	4	4	5	5	5	6	6	6	6	6	6	7	7	7	7	7	8	8	8	8	8	8	8
26	2013_KK_STR_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	7	8	8	10	10	15	16	21	22	23
27	2013_KK_PO_1050	0	0	0	0	3	3	3	3	4	4	4	4	4	4	5	5	5	5	5	5	5	5	6	6	6	6	7	7	7	7	7
28	2013_KK_STR_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	4	7	10	11	13	13	15	16	16	17	17	17
29	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	4	4	6	8	8	11	13	14	15	17	20	20	21	24
30	2013_JL_UT_1510	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	9	14	14	15	22	23	30	32	37
31	2013_KK_UT_1050	0	0	0	0	0	0	0	0	0	0	0	0	0	0	4	5	6	7	8	9	9	10	12	12	13	13	13	14	15	16	17
32	2013_KK_STR_1330	0	0	0	0	0	0	0	0	4	4	5	5	5	6	6	6	7	8	8	8	9	9	9	9	9	11	11	12	13	13	15
33	2013_KK_PO_1050	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	7	8	10	13	13	16	16	16	26	29	34	34
34	2013_KK_PO_1050	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	15	23	25	27	30	37	40
35	2013_KK_UT_0910	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	3	3	3	3	5	8	8	9	11	11	11	11	12	14	14	14
36	2013_KK_STR_0910	0	0	0	3	4	5	5	5	5	5	5	5	6	6	6	6	6	7	7	7	7	7	7	7	8	8	8	8	8	9	9
37	2013_MV_UT_0730	0	0	0	0	2	2	4	4	4	5	5	5	6	6	7	7	7	8	8	8	8	8	9	9	9	10	10	11	11	12	12
38	2013_KK_UT_1050	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	26	39	47	51	56	58	59	75
39	2013_MV_UT_0730	0	0	0	0	0	4	4	4	4	4	5	6	6	6	7	7	7	8	8	8	9	9	9	10	10	13	13	16	18	19	21
40	2013_KK_STR_1330	2	4	5	5	5	6	6	6	7	7	7	7	7	7	8	8	8	9	9	9	9	9	9	10	10	10	10	11	11	11	11
41	2013_KK_STR_1330	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	5	5	8	10	14	14	15	16	16	18	18	18	19	21	21

Fig. 6 List of users (hidden) with sorted intervals spend on items

answers were provided in 30 s. Time for the next three answers seems to be okay, but most of the following items were answered in too short time intervals. Starting from row 23, student change the way how he/she submitted the test—he/she sent to Moodle 5 items at once. We consider profiles D and E also as suspicious for similar reasons.

Based on the data presented above (Figs. 3 and 4) we formed several another reports. Interesting for us seems to be the list of users with sorted intervals spent on items (Fig. 6). This report highlights students with too many zero values for intervals (or too low interval values). Even though the report penalizes students who filled the test properly but submitted all answers at the webpage at once, it seems, that the provided piece of information can be taken into account also in the final decision process.

The number of attempts on each item in the quiz (Figs. 7 and 8) seems to be very good indicator of cheating. Some students' data exhibits randomness (Fig. 7), on the other hand, we identified student(s), whose every item was answered correctly on the first attempt (Fig. 8, extreme case; value 2 in the graph is a result of implicit Moodle additional write operation per item to database for quiz completion).

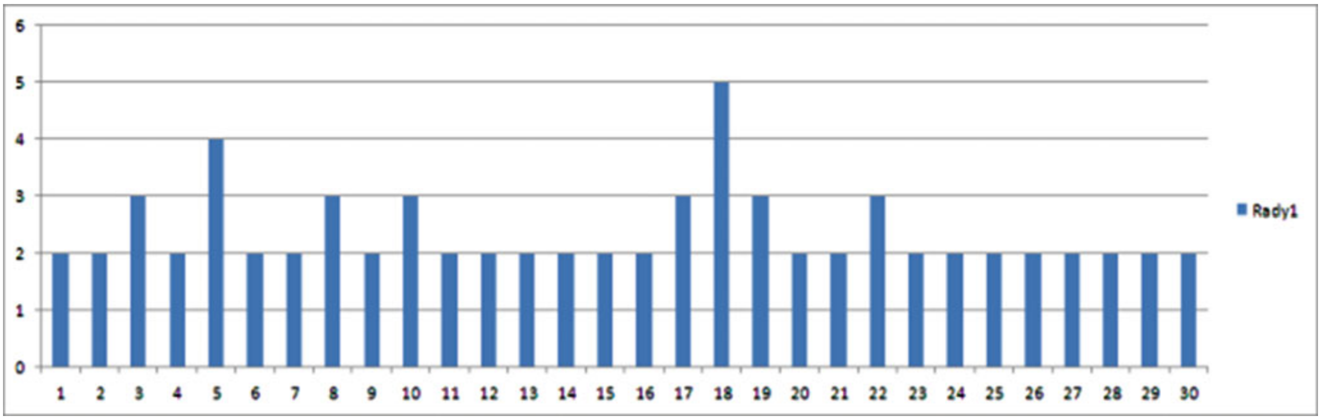

Fig. 7 Number of attempts per quiz item (question)

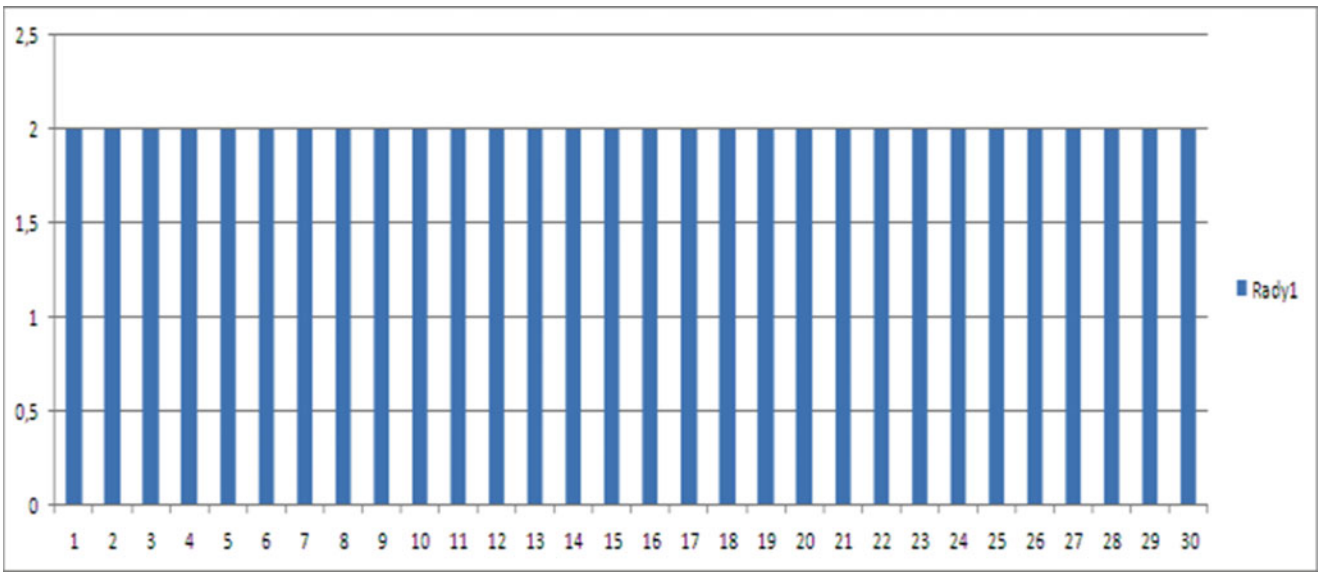

Fig. 8 Number of attempts per quiz item—suspicious data

Conclusion

The presented approach was applied for all formative tests in the Course of Operating Systems. Suspicious students were identified within all above mentioned criteria. There were cases, when a student was suspicious based on number of attempts per quiz item (Figs. 7 and 8), but suspicion was canceled by analysis of other data—mainly detailed data about test (Figs. 3 and 4). However, major part of the suspicions was confirmed. Suspicious students were notified about suspicion and they were given chance to defend themselves. At the end, nobody of them used the presented chance, although initially some of them claimed that he/she did not cheat.

We realize drawbacks of presented approach. Mainly, it is manual evaluation of data—instructor has to browse data manually (which is not so complicated for last approach—number of attempts for item, but quite laborious for first one—typically, there is tens of thousands records in the database for each quiz).

In the future, we plan to introduce a more advanced data processing, which will provide categorization of students—cheater/non-cheater—which, we believe, eliminates necessity of manual investigation of data.

Acknowledgment This work has been supported by the KEGA grant 062TUKE-4/2013, granted by Cultural and Education Grant Agency (KEGA), Ministry of Education of Slovak Republic.

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Application of a Feed-Forward Control Structure

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Abstract

A feed-forward control structure adapted from Vilanova and Serra (IEEE Proc- Control theory for Electrical Engineers 144 (6):103–104, 1997). was investigated through both simulation and application to demonstrate how it could enhance the performance of closed-loop systems. In addition its ability to handle with ease right-half plane (RHP) zeros, unstable poles and dead-time that may exist in the plant will be investigated. Design of the components of structure is shown to be both non-complex and simple to implement.

The novel feed-forward control loop and the traditional PID feedback will be compared using Pareto Fronts to quantify the performance trade-offs. An application will be used to validate the simulation results and conclusions.

Keywords

Feed-forward control • Uncertainties • Independence • Pareto front • Zero error

Introduction

Feed-forward controllers have been used to eliminate process disturbances since the 1950 [1] and these structures will be termed traditional feed-forward control in this paper. Practical experience showed that the success of such feed-forward control depended on model accuracy.

By way of comparison, the feedback loop with PID control has been the most popular control technique in industry since it is simple and well-understood by practicing engineers.

This paper will demonstrate a design procedure for novel feed-forward controllers that were adapted from [2], as well as investigate their performances in various situations that challenge feedback control. These include instances when the plant contains unstable poles, RHP zeros and/or dead-time.

The paper is organized as follows: The setup of system structure will be given in Sect. "The Setup of System Structure". Section "Feed-Forward Controller Design" shows a design method of feed-forward position control to counter various difficulties, and speed control will be discussed in Sect. "Motor speed control". Pareto Front analysis of competing systems is demonstrated in Sect. "Pareto Front Analysis". Section "Application to Motor Speed Control" contains the application of motor speed control, and Sect. "Conclusion" concludes the document.

The Setup of System Structure

Control Loop Structure

The new realization of a 2-degree-of-freedom (DOF) controller that was proposed by [2] is shown in Fig 1. Its unique property is the separation of design of set-point tracking and disturbance rejection. The separation principle is proved in [2–5].

The simple interpretation of this control loop considers the signal pair (u_a and y) associated with the feed-forward

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_12, © Springer International Publishing Switzerland 2015

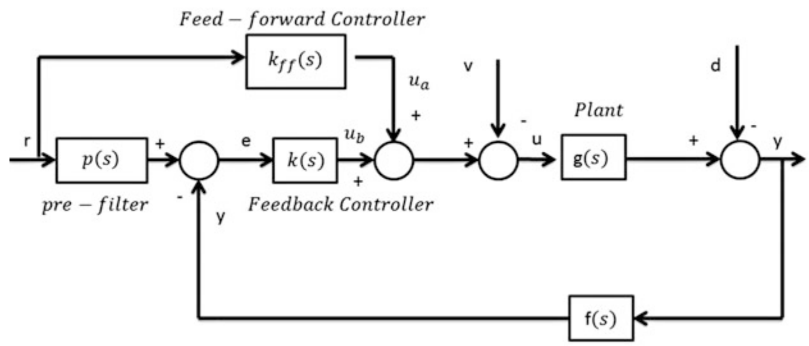


Fig. 1 Feed-forward structure adapted from [2]

controller, combined with a feedback loop around f(s) and k(s). When a command signal is injected into the system at r, the system response is shaped by the feed-forward loop while disturbance rejection is effected by the feedback controller k(s) [3].

The design aim is to achieve fast command tracking and good disturbance rejection by means of a 2 DOF structure that separates the closed loop and open loop characteristics. This allows independent design of the feedback and the feed-forward controllers.

The strength of this loop configuration is that the signal coming out from the feed-forward controller acts as a "correct injection" to the system input that speeds up command tracking. Unstable poles of $g(s)$ are included in the feed-forward controller $k_{ff}(s)$ while RHP zeros and dead-time in $g(s)$ are included in the pre-filter component $p(s)$.

Specifically, assume that the plant model is factored into two (causal or non-causal) transfer functions:

$$g(s) = m(s).n(s) \tag{1}$$

where

- $m(s)$ contain all the (stable and unstable) poles of $g(s)$ as well as its LHP zeros and gain,
- $n(s)$ contains the dead-time and RHP zeros of $g(s)$

Then the pre-filter and feed-forward controllers are given by:

$$\begin{aligned} p(s) &= h(s).n(s) \\ k_{ff}(s) &= h(s).m(s)^{-1} \end{aligned} \tag{2}$$

Where

- $h(s)$ is the transfer function desired for the closed loop dynamics

The System Output Signal

The output signal $y(s)$ can be derived as:

$$y = g(s) \cdot \frac{h}{m}(s) \cdot r + k(s) \cdot g(s) \cdot e \tag{3}$$

Under ideal conditions the feed-forward loop will inject the exact signal u_a into the plant actuator to ensure that $e = 0$ and the system response becomes:

$$y = g(s) \cdot \frac{h}{m}(s) \cdot r \tag{4}$$

thus the system is in open loop during command tracking.

Also if $n(s) = 1$ then $m(s)$ models the plant dynamics $g(s)$ perfectly and the output response reduces to:

$$y = h(s) \cdot r \tag{5}$$

A non-zero error changes the result significantly in that:

$$y = \frac{h(s) \cdot g(s)}{m(s) \cdot (1 + g(s) \cdot k(s))} \cdot r + \frac{g(s) \cdot k(s)}{1 + g(s) \cdot k(s)} \cdot p(s) \cdot r \tag{6}$$

This derived transfer function is defined differently to that stated in [2] that was the basis for the feed-forward loop in this paper.

Thus to design the feed-forward control the system will be assumed to have no uncertainties and no disturbances. In practice this is seldom the case and feedback is necessary to reduce the effect of these effects.

System Input Signal

The input signal to the plant shown in Fig. 1 is calculated as:

$$u = \left(\frac{h(s)}{m(s)} + k(s) \cdot p(s) \right) \cdot S(s).r \tag{7}$$

where $S(s)$ is the standard sensitivity function.

The feed-forward component of the input signal can be interpreted as an ideal signal injected into the loop to ensure set-point tracking. The difference between desired output and the actual output then gets corrected by the feedback controller.

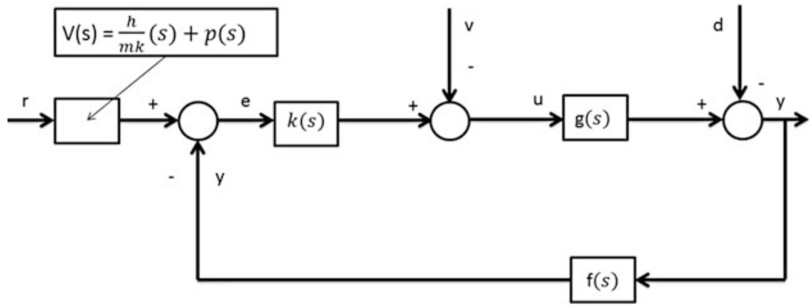


Fig. 2 Feed-forward structure adapted from [2] is transformed into a traditional feedback loop, where feed-forward components are moved into the pre-filter component

Under the assumption that system is linear and time invariant, with no model uncertainties and no disturbances, the input signal into the plant will then solely be generated by the feed-forward controller:

$$u = k_{ff}(s) \cdot r = h(s) \cdot m(s)^{-1} \cdot r \tag{8}$$

The output signal becomes:

$$y = g(s).u = g(s).h(s).m(s)^{-1} \cdot r = h(s).r \tag{9}$$

and tracks the set-point according to the expected dynamics.

Causality

To ensure that the modules in the control system are causal the design engineer needs to select the desired closed loop model, h(s), so that the block $\frac{h}{m}(s)$ satisfies the following constraint:

$$P_{h/m} \geq Z_{h/m}$$

This means that

$$P_h \geq Z_h + P_m - Z_m$$

and since Z_h is most likely to be set to zero:

$$P_h \geq P_m - Z_m$$

Thus the number of poles in the desired closed loop model must equal or exceed the pole excess of the non-minimal phase dynamics of the process model.

An Equivalent Structure

The feed-forward structure in Fig. 1 can be transformed by block diagram algebra to a traditional loop like that in Fig. 2 that contains a pre-filter, V(s), acting on the set-point.

The feed-forward structure will change the definition of the error signal in Fig. 1 to become that shown in Fig. 2.

As an example consider the following case, for no dead-time and RHP poles and zeros in a plant that models a motor position control system:

$$g(s) = \frac{2.18}{s(2.96s+1)} = \frac{2.18}{s(2.96s+1)} \cdot \frac{1}{1} = m(s) \cdot n(s)$$

The feedback controller designed for the loop in Fig. 1 was set to type 1 to track constant set-points while ensuring a damping factor of 0.707 and a faster transient response.

$$k(s) = \frac{9.002s^2 + 9.3023s + 1.5}{s^2 + 7.5s}$$

Given that the desired closed loop model is:

$$h(s) = \frac{2}{s^2 + 2s + 2}$$

the feed –forward Controller becomes:

$$k_{ff}(s) = \frac{h}{m}(s) = \frac{5.92s^2 + 2s}{2.18s^2 + 4.36s + 4.36}$$

and the pre-filter is given by:

$$p(s) = h(s) \cdot n(s) = \frac{2}{s^2 + 2s + 2}$$

The pre-filter for the new system was found to be:

$$V(s) = \frac{5.92s^4 + 46.4s^3 + 54.248s^2 + 40.56s + 6.54}{19.624s^4 + 59.528s^3 + 83.078s^2 + 47.1s + 6.54}$$

Feed-Forward Controller Design

As mentioned earlier feed-forward controller components need to be adjusted when RHP items and dead-time are present in the plant model, but the fundamental transfer

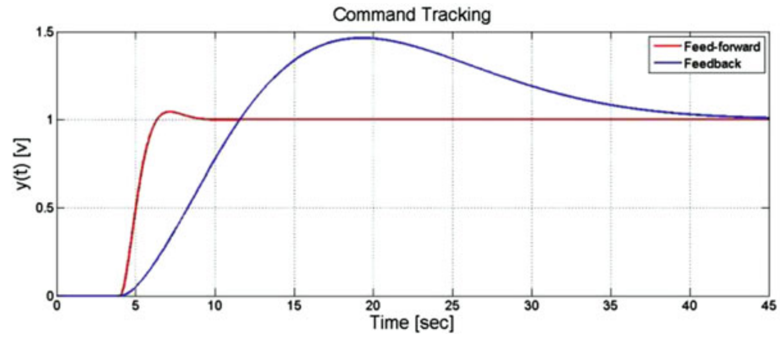


Fig. 3 Command tracking comparison of feed-forward and feedback control. Evidently feed-forward maintained its superior performance even though the dead-time of 3 s remains

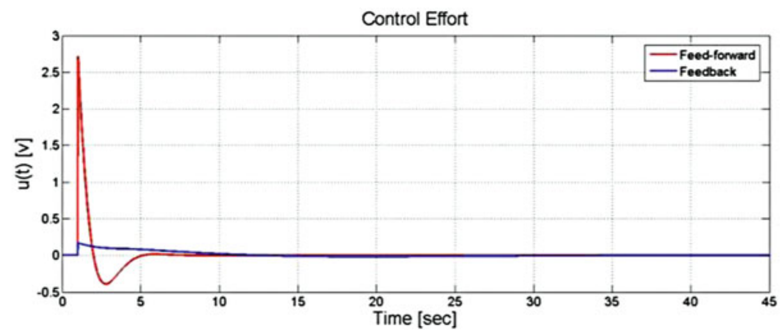


Fig. 4 Control effort comparison of feed-forward and feedback control

function will remain constant. In this section the versatility of the feed-forward structure is demonstrated through simulation of the two plant models that include such problematic dynamics.

The first plant contains dead-time:

(a)

$$g(s) = \frac{2.18}{s(2.96s+1)}e^{-s\tau} = \frac{2.18}{s(2.96s+1)} \cdot e^{-s\tau} \\ = m(s) \cdot n(s)$$

while the second has an unstable pole and a RHP zero:

(b)

$$g(s) = \frac{2.18(s-1)}{2.96s^2+s-1} = \frac{2.18}{2.96s^2+s-1} \cdot (s-1) \\ = m(s).n(s)$$

In each case the maximum allowed input voltage was constrained to be $|10|volts$.

The design ensures command signal tracking, settling time, damping factor, percentage overshoot and control effort. These characteristics can be achieved by pole (and zero) placement.

Feed-Forward of Plant A

Plants that contain the dead-time are difficult to control in feedback due to the increased phase angle.

By employing the feed-forward method, the pre-filter is required to include the system dead-time shown below:

$$p(s) = h(s).e^{-s\tau} = \frac{2}{s^2+2s+2}.e^{-3s} \\ k(s) = \frac{0.2238s^2+0.0827s+0.00452}{s(1+1.2s)} \tag{10}$$

Simulation yielded the closed loop responses: (Figs. 3 and 4)

Feed-Forward of Plant B

Right half plane poles and zeros make the design of feedback controllers difficult. In this scenario, plant model (B) above contained both these and a type one feedback controller could not be designed for the plant. Instead a type zero feedback controller was designed:

$$k(s) = \frac{-0.64884 \cdot (1+1.5s)}{(1+0.34s)}$$

consequently a finite position error was expected.

However, feed-forward control avoids the problem of set-point tracking by including a gain that is the inverse of the plant gain. The RHP zero is put in the pre-filter and the unstable pole becomes a RHP zero in the feed-forward controller. This yield:

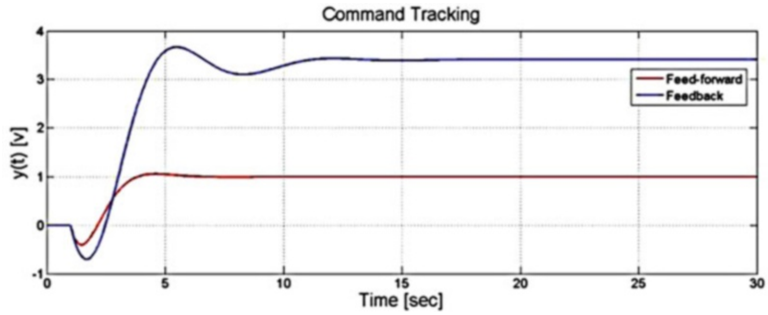


Fig. 5 Command tracking comparison of feed-forward and feedback control. Once again feed-forward control didn't get affected much, but feedback control couldn't track the set-point

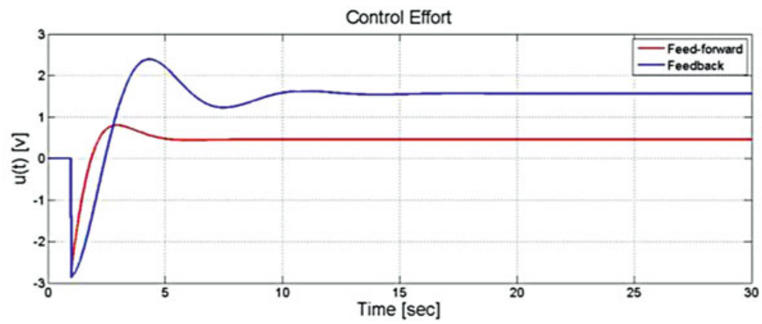


Fig. 6 Control effort comparison of feed-forward and feedback control

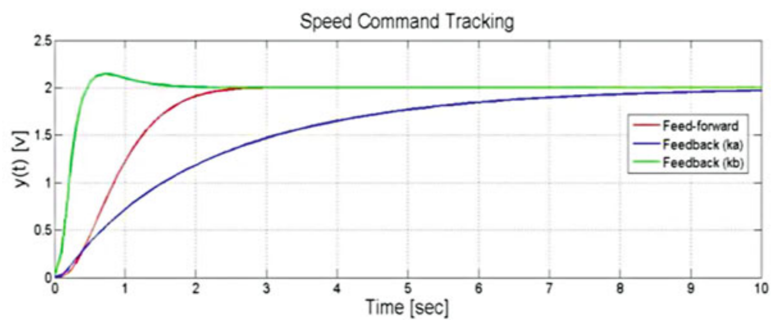


Fig. 7 Command tracking Comparison of feed-forward control and two different feedback controllers designed

$$p(s) = \frac{2(s-1)}{s^2 + 2s + 2}$$

$$k_{ff}(s) = \frac{5.92s^2 + 2s - 2}{2.18s^2 + 4.36s + 4.36}$$

The simulation results yielded the responses: (Figs. 5 and 6)

Motor Speed Control

Consider the plant model for speed control, given as

$$g_\omega(s) = \frac{8.17}{1 + 1.04s}$$

Two controllers $k_a(s)$ and $k_b(s)$ were designed for this plant model. This first (a) ensured a low speed of response, while (b) achieved a fast settling time by permitting some overshoot.

(a)

$$k_a(s) = \frac{0.34 \cdot (1 + 1.2s)}{5.79s}$$

(b)

$$k_b(s) = \frac{1.9022 \cdot (1 + 0.45s)}{s}$$

The feed-forward control improved the undesired performance, giving both speed and damping as shown in Figs. 9 and 10. This set up will also be implemented on the physical DC motor to compare simulation and practical results. (Figs. 7 and 8)

Control effort required from feed-forward controllers was much less than feedback controller $k_b(s)$. Also the percentage overshoot was nearly zero but as a trade-off, it has slightly

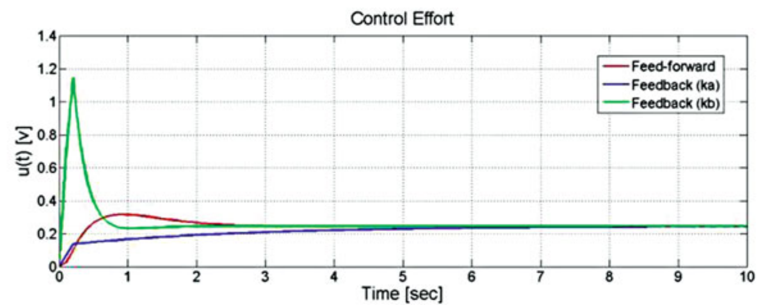


Fig. 8 Control Effort comparison of feed-forward control and two different feedback controllers designed

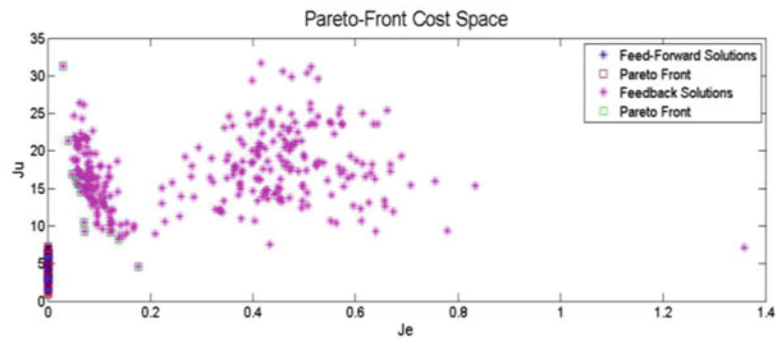


Fig. 9 Pareto front analysis of plant with no dead-time and right half plane pole or zero

slower command-tracking speed. Controller $k_a(s)$ has unacceptable speed of command tracking (as expected) and demonstrates the improvements due to feed-forward control.

Pareto Front Analysis

Pareto Front analysis is a powerful method for solving multi-objective optimization [6]. Furthermore [7] Pareto Front is useful for analyzing controller design methodologies, as it addresses all significant trade-offs simultaneously.

A Pareto-Front represents the best possible solution formed by the non-dominated solution. These multiple solutions are all optimal in the sense that there are no other points in the entire solution domain or search space that are superior to them when all objectives are considered simultaneously [8].

Results generated by a Pareto-Front provide a fair comparison platform between objectives. One can see how the different loop configurations differ from one another by the cost functions that the user defines.

Pareto Differential Evolution proposed by [1] which serves as a Multi-Objective Optimization tool is the evolutionary algorithm used in this paper.

The cost functions on the error and the input used in the Pareto Front were adapted from [9] and defined as:

$$J_e = \frac{1}{T}\int_0^T e(t)^2 dt \quad \text{and} \quad J_u = \frac{1}{T}\int_0^T u(t)^2 dt \qquad (11)$$

Pareto Front Analysis 1: Position Control

When g(s) does not contain RHP zeros, unstable poles, and dead-time, the result that Pareto Front displayed shows feed-forward as the better option. As expected the error cost function (J_e) of feed-forward system remained zero, when the parameters are generated randomly by Gaussian distribution.

This is expected as the feed-forward output injects the desired signal straight into the plant input, before any feedback correction was required. Consequently the error is zero (Fig. 9).

Pareto Front Analysis 2: Position Control

When dead-time is present the feed-forward loop is affected by means of the error development, but fortunately the changes are minor. The nonzero error results in the system taking longer to settle than when no dead-time is present (Fig. 10).

Pareto Front Analysis 3: Speed Control

The aim of this section is to prove that the addition of a feed-forward control on an existing feedback loop, can shortens the settling time and decreases the percentage overshoot.

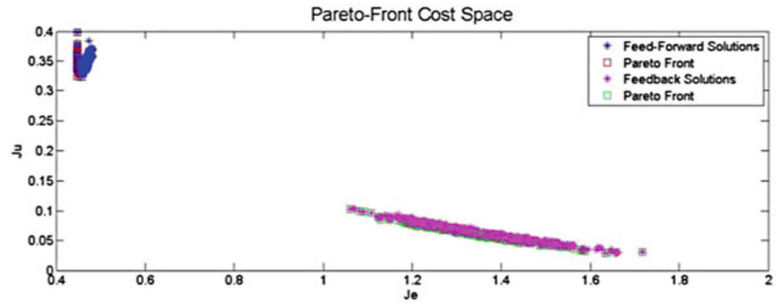


Fig. 10 Pareto front analysis of plant with dead-time

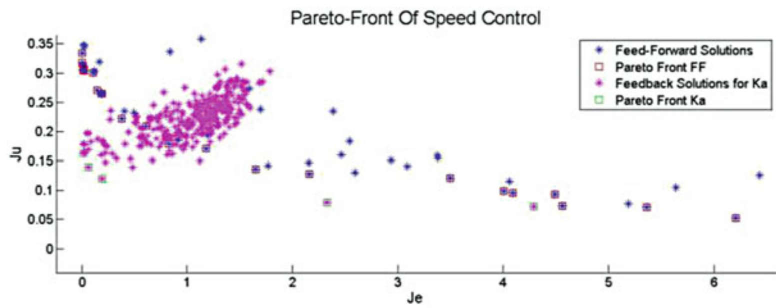


Fig. 11 Pareto analysis with 5 % plant parameter variation, feedback controller was designed to have slow speed of response

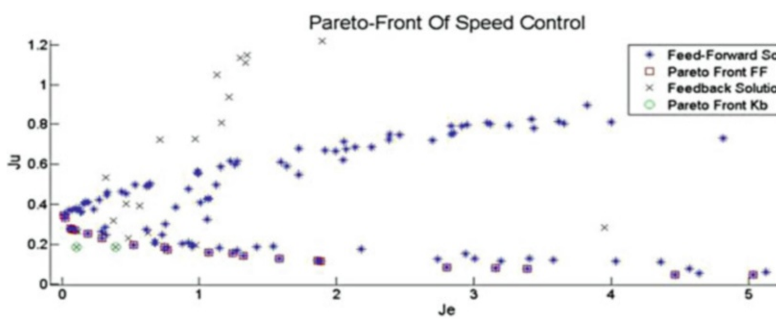


Fig. 12 Pareto analysis with 5 % plant parameter variation, feedback controller was designed to have fast speed response but high percentage overshoot

In contrast to the position control case, the Pareto Front analysis for speed control will include uncertainty and disturbances that would act on the physical model. The aim is to get more realistic results which can be used to predict the behavior of an actual motor system.

Physical disturbances and model uncertainty will be approximated as standard deviations on plant parameters. The assumption is a parameter variation of 5 % peak to peak; hence the standard deviation will be 2.5 %.

Feed-forward control is sensitive to plant uncertainties and disturbances that the plant system may have (Figs. 11 and 12).

Application to Motor Speed Control

A physical DC motor system was used to test the control structure simulated previously (Fig. 13).

Continuous feed-forward and feedback controllers are implemented on the servo motor set shown in Fig. 15.

The continuous controllers are converted into digital domain via z-transform, and coded into a VB program. The speed and position of motor were read at a sampling time of 200 milliseconds from analogue to digital converter (ADC).

The results of applying each control structure to the DC motor are now considered.

Experiment 1: Initial Speed Control

In this case only the feedback control law was present. It was designed to ensure set-point tracking with slow dynamics (Figs. 14 and 15).

When feed-forward was added the response improved: (Figs. 16 and 17)

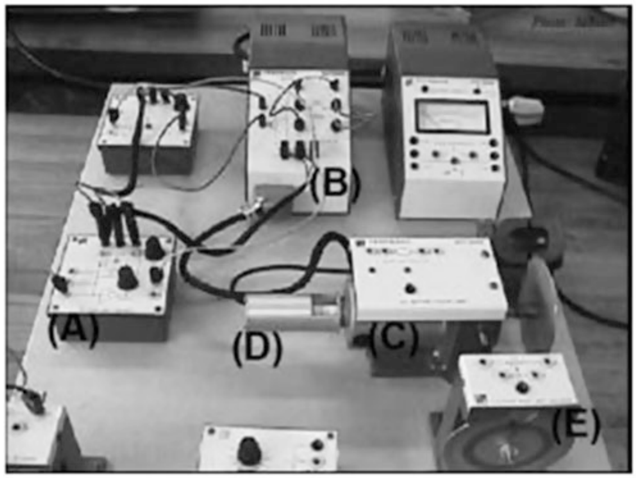


Fig. 13 The servo motor used to test the feed-forward structure, where A is the operational amplifier circuit, B is the Power Amplifier, C is the direct current motor, D is the tachometer that's attached to motor, E is the 360° potentiometer

Fig. 14 Practical result of feedback controller $k_a(s)$ (Output signal)

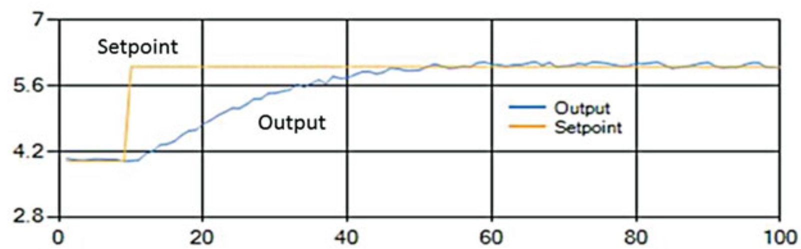


Fig. 15 Practical result of feedback controller $k_a(s)$ (Control effort),where u(t) = utb and alpha = uta = 0

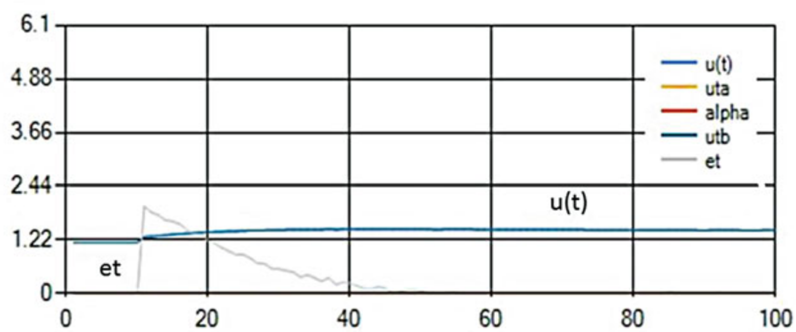
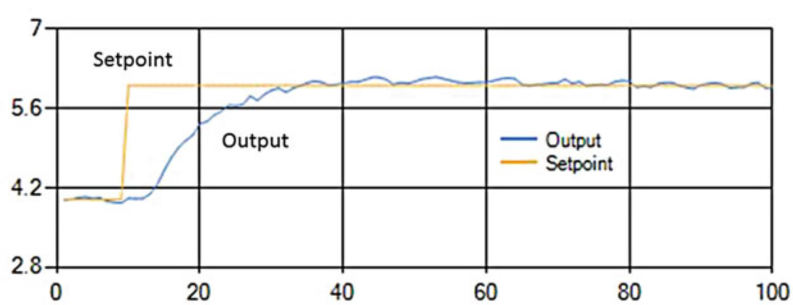


Fig. 16 Practical result of feed-forward control (Output signal)



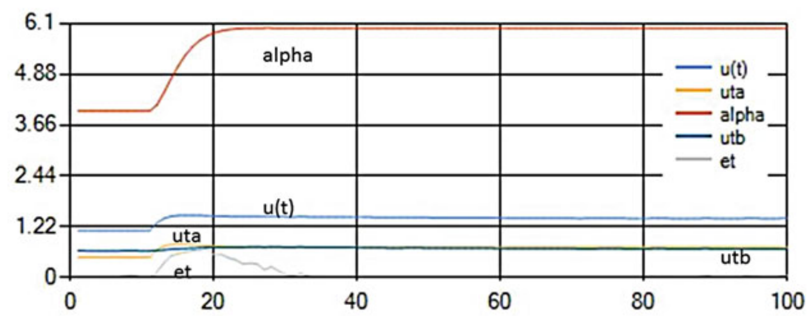


Fig. 17 Practical result of feed-forward control (Control efforts)

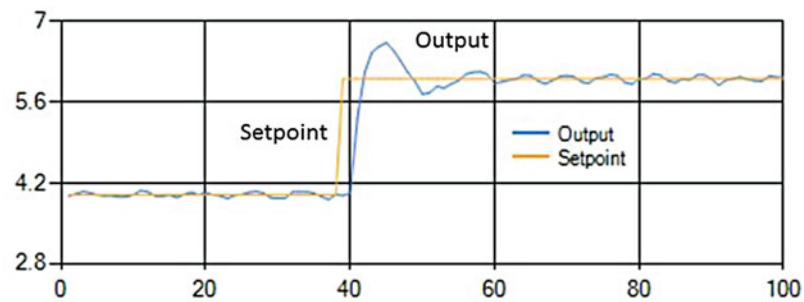


Fig. 18 Practical result of feedback controller $k_b(s)$ (Output signal)

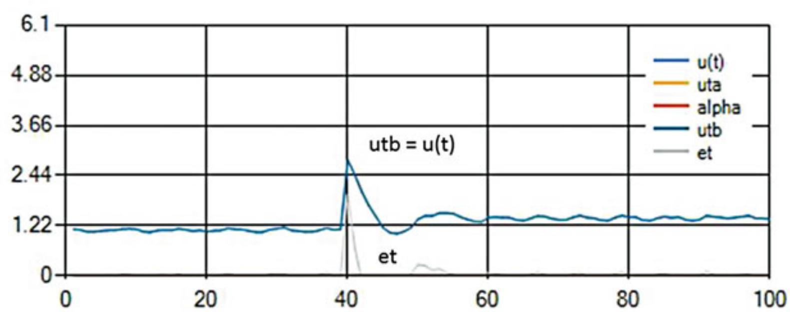


Fig. 19 Practical result of feedback controller $k_b(s)$ (Control effort utb = u(t))

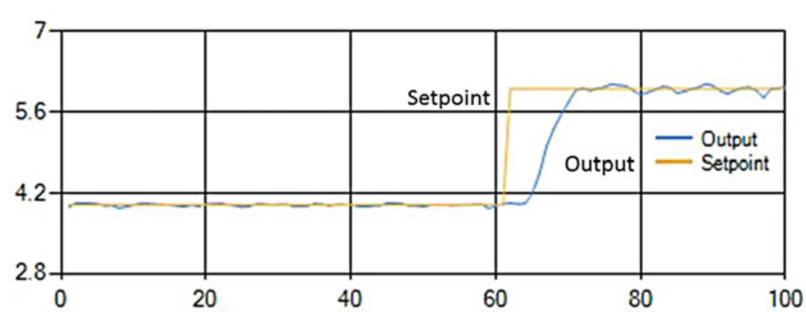


Fig. 20 Practical result of the same feed-forward control (Output signal)

Experiment 2: Faster Control with Overshoot

The second feedback controller improved significantly on speed but at the cost of more overshoot: (Figs. 18, 19, 20, and 21)

Once again the addition of feed-forward control improved significantly on the response that was now both fast and damped:

The results obtained from the motor control applications are summarized in Tables 1 and 2.

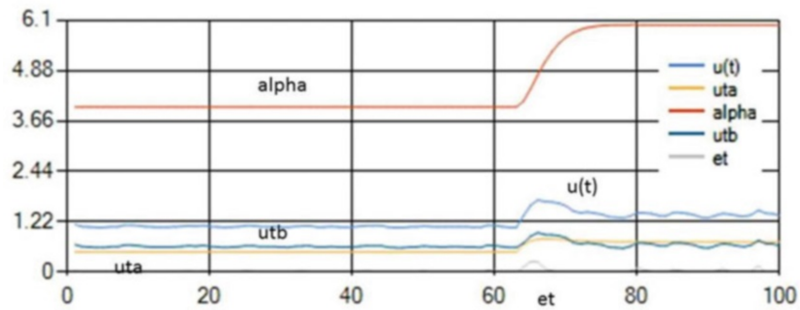


Fig. 21 Practical result of feed-forward control (Control efforts)

Table 1 Practical results of experiment 1

Practical results	Feed-forward	Feedback
Settling time	4.1 s	6.7 s
High frequency gain	1.36 V	1.3 V
Error peak	0.7 V	1.83 V

Table 2 Practical results of experiment 2

Practical results	Feed-forward	Feedback
Settling time	1.9 s	3.8 s
High frequency gain	1.83 V	2.7 V
Error peak	0.3 V	2 V

The enhanced performance introduced by the feed-forward control structure is evident in the reduced settling time, the absence of overshoot and the consequent improvement in tracking error.

Conclusion

From simulation and implementation results it was noted that the design method under investigation provides excellent performance in tracking step commands.

Practical results demonstrated that feed-forward control in general produced faster command tracking and less oscillation in speed control. As anticipated the feed-forward control was sensitive to parameter changes. These results thus support the claim that feedback and feed-forward controllers can be designed separately to handle open and closed loop properties to maximize control efficiency.

Derived transfer functions in theory development proved to be correct. As seen from the practical results control effort from the feedback controller is needed to correct unexpected error in simulation due to the exclusion of uncertainties and electrical time constant.

In position control simulation, the robustness of handling dead-time and RHP poles by this novel feed-forward loop structure can be regarded as a direct non-complex approach.

Acknowledgment I would like to thank Prof. Braae for introducing the area of control engineering to me, as well as all the support and positive criticism.

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Reactive Planning to Compose Learning Routes in Uncertain Environments

Ingrid-Durley Torres and Jaime Guzmán-Luna

Abstract

This paper presents on a reactive planning model, to organize, design and adapt from composing of learning objects (LO), learning routes in virtual environments and real time. The characteristics of these environments include not being controlled by just one participant; hence, compositions here proposed, used a planning algorithm to handle uncertain events on the building of learning routes must face uncertain changes originated in an environment in base to three cases: i) when a LO is no longer available, ii) when a LO is modified, or iii) when the goal state of learning changes depending on the original requirements of the problem. This solution ensures finding a long learning route. Although the first option is the convenience LO student profile, if one is not available replaced by another or others which ensure achieving the desired knowledge.

Keywords

Reactive planning • Learning objects • Learning routes • Virtual environments

Introduction

Although various pieces of literature have concentrated on composing learning routes with the application of Artificial Intelligence (AI) planning techniques to fulfill the best way possible the learning objectives that an user has formulated, those works have been focused personalization or adaptability techniques [1–5] without yet involving considerations inherent to their own virtual environment as problems involving the changing state of the world in matters concerning insertion, elimination, update of LO, or even a change in the formulation of a user's own objectives, when a learning route is being built.

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This paper initially considers the construction of a learning route as a dynamic composition of learning objects. The idea behind this composition of learning objects is that it is essential for the objects themselves be the ones to interact among each other, not through a user. This proposal used AI planning technique [2, 3, 5], as composition mechanism, to form a path which may achieve the user's specific knowledge needs. The purpose of planning is then, the creation of an unanticipated and subordinated sequence of learning objects (i.e. plan or learning route), which is attained from other previously existent objects, (i.e. composition). Because the context corresponds to a virtual environment, is very important perceive continuously the environment to include the actual state and to make updated decisions which consider almost immediately any uncertain change in the state of the world. In order to address this uncertainty, we propose to apply a specific planning technique known as reactive planning [6], which in each of its stages is able to bear in mind information environment to include it and to make updated decisions which consider almost immediately any uncertain change in the state of the world.

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_13, © Springer International Publishing Switzerland 2015

The purpose described herein consists then in generating sequences of actions dynamically (i.e. plan or learning route) were each action is represented by LO; the convenience to select a LO is performed depending on which is most appropriate to the profile of the student as learning theories [Honey-Alonso] indicating that a LO is more or less appropriate, depending on the student's learning style; in such a way that if a user cannot correctly invoke one of those actions (LO), or cannot visualize, it due to: i) an LO is no longer available, ii) the metadata identifying an LO has been modified, iii) a user's objective changes. This action (LO), may be automatically replaced by another action (LO) or by other actions (LOs), that they are deem adequate (in other words, they actions (LO) must fulfill the same learning objectives) for end-users. The mechanism of composition solution ensures finding a long learning route.

To provide a more detailed vision of this paper, we presented it as follows: section "Frame of Reference" defines the main concepts related to this proposal's framework. Section "Mechanism of Learning Route Composition", presents all the details of the composition mechanism of learning routes, with a brief description of the functionality of each one of the component modules. Section "Characterization of the Uncertain Behavior of the Environment", defines characterization of uncertain behavior of environment. Section "Planning Algorithms for Uncertain Events" presents the operation of the planning algorithms to uncertain events. Section "Experimental Case", present an experimental case with its respective test results. Finally, section "Conclusions and Future Work" compiles conclusions and future work arising from the development proposed herein.

Frame of Reference

Description of the LOs

Formally, there is not just one definition of the concept of an LO [7–9]. Nonetheless, it is convenient to consider it as an attempt to unify; the following definition. An LO will be understood as [7], all the material structured in a significant way, and it must be related to a learning objective which must correspond to a digital resource that can be distributed and consulted online. An LO must also have a registration form or metadata that includes a list of attributes which not only describes the possible attributes of an LO, but also allows to catalogue and exchange it. In this setting, standardization is a notably recurrent topic since when one handles various types of resources for different applications and with different technologies, it becomes a key topic to continue operating current applications and even to make them grow. Among these initiatives it is worth highlighting: an LOM [9]; this standard specifies the syntax of a minimum set of metadata required to complete and to adequately identify, administrate, locate and evaluate an LO. Its purpose is to facilitate the task of searching, sharing and exchanging LOs for authors, students and automatic systems.

Reactive Planning

Planning is a process of explicit and abstract deliberation which selects and organizes actions anticipating expected results [10]. This deliberation aims to fulfill some pre-established objectives the best way possible for it is a problem of searching which requires finding an efficient sequence of actions leading to a system from its initial state to an objective state [11]. There is a large variety of planners and of ways to classify them.

For this paper, one of the most commonly used forms of reactive planning will be defined herein. Reactive Planning systems also named planning and execution systems or on-line systems alternate or overlap planning and execution stages to tackle this problem efficiently [10]. Thus, online planning systems may adapt the plan during execution to possible unexpected situations which may arise. Moreover, they may use all the information collected during execution to improve initial plans. One of the techniques used to adapt plans to new situations is replanning. This technique may be found as part of on-line planning systems. Under this technique, instead of calculating an entirely new plan, the aim is to modify an initial plan when it ceases to have validity. This avoids the repetition of a planning effort since normally the original problem resembles the new problem proposed. Although replanning is not an easy task, and many times it may be as expensive a process as planning from scratch [10, 12], in many occasions, there are important time savings, and so there are swift replies to unexpected situations.

Composition of Learning Routes

In essence, in the compositions of learning routes, it is the LO themselves which interact among themselves to form a road perfectly adapted to the learning needs a student has formulated. Then, the purpose of composition of learning routes consists on the dynamic creation (unanticipated) of an LO sequence obtained from other pre-existing LO. This route is one which must follow in strict order a student to reach learning objectives.

In general, a learning route composition problem for the scope of this proposal involves the representation of the need to learn, formulated by a student. Under this consideration, one formal definition are generated inspired on the LRN

Planner [5] for the composition of learning routes, which are stated as follows in their corresponding order.

Definition 1. More formally, a learning route composition problem from the representation of the modeling of a need to learn may be described as a tuple (*S*,*So*,*G*,*A*,*R*), in which S is the set of all possible states of knowledge. *So ε S* denotes a user's initial state of knowledge (student), *G ε S* denotes the objective state of a user's knowledge, expressed by a set of concepts related to a formative intentionality in which the composition system will try to search. A is the set of actions which represent the LO, the composer must consider to change from a state of a user's knowledge to another state of knowledge, and the translation relation *R ε S x A x S* defines the precondition and effects to perform each action (LO).

Mechanism of Learning Route Composition

From User Requirement

The composition of a learning route from the point of view of a user's learning requirement is directly related to the formal specification of a planning problem, which includes collections of actions with pre-requisites and results (preconditions and effects respectively). Inspired in [3, 5]. When an action supports another this generates a causal link between them, meaning that the preceding action is finished before starting a succeeding action. With these actions, it is possible to establish a plan of the actions which will be transferred to users (students) from an initial state of knowledge of a course to a state in which formulated objectives are achieved by users (students) themselves. Expressed analogically, with a virtual learning environment, users (students) express their learning requirement depending on one or various concepts of domain specific knowledge, these concepts are part of a course's contents, and they lead to one or more associated activities. This requirement must be achieved from a state of a student's initial knowledge of that domain (which is even considered null, for the specific case in which an individual knows nothing of a domain); also, including at the same time considerations regarding learning preference and style. Thus, the learning route for this case constitutes a (plan) sequence of learning activities associated to knowledge domain, and customized for a student that the student must follow to fulfill that student's objectives. In this case, the building of a planning problem is done by means of an interpretation that can be conducted computationally on both types of defined relations which may be interpreted by a planning mechanism as AND relations and OR relations; they refer to *RequiredBy* and *IsBasedFor* respectively. Thus, all the concepts defined as AND, must be totally completed before seeing the next, while at least one of the OR relations must have been completed to see what is next.

Then transfer the Definition 1, in to PDDL (Planning Domain Definition Language), which was expressed by the (*S*,*So*,*G*,*A*,*R*) tuple, where *S* is a set of all the possible states of knowledge defined by the course; *So εS* denotes a (student) user's initial state of knowledge within the state of the course, *G ε S* denotes the state of the objective also of a (student) user's knowledge within the state of the course, expressed by a set of concepts associated to the formative intentionality, in which the composition system will try to search (SVC + LOD).

A is a set of actions which represent the LOs which the composer must consider to change from a user-student's state of knowledge, to another state of knowledge, and the translation relation *R ε S x A x S* defines the precondition and effects to execute each action (LO) (See Table 1).

As it is deduced from the tuple, states *S* are defined depending on the course model; this is why, this stage can only be achieved once that, at least one of the learning routes of the course modeling has been defined and it only works on one course at a time; then, there is an attempt to define: (i) Which assignments best adapt to a given student's profiles, and which are the resources required to achieve an objective of knowledge expressed by a student depending on the concepts of the course already instantiated in a domain of specific knowledge (LOD)? (ii) Which LOs must follow a student? To do so, the set of LOM mega data associated to each LO is translated to a planning domain as an action, where preconditions will be defined as two types (i) some of knowledge directly associated to LOD concepts and to the assignments; y (ii) some of precedence associated to represented by *is required* and *isBasisFor*. The effects themselves correspond to the statement that says that the LOD concept is already known and that LO's specific activity has already been carried out. (iii) The convenience of the LOs in accordance with a student's profile is achieved by building a function called *fluent* (datum fluctuating within a planning process called "*reward*"), which will be in charge of incrementing a determined value of 1 if the learning LO is convenient for that student's profile or 0 in the opposite case.

Reactive Planning Schema

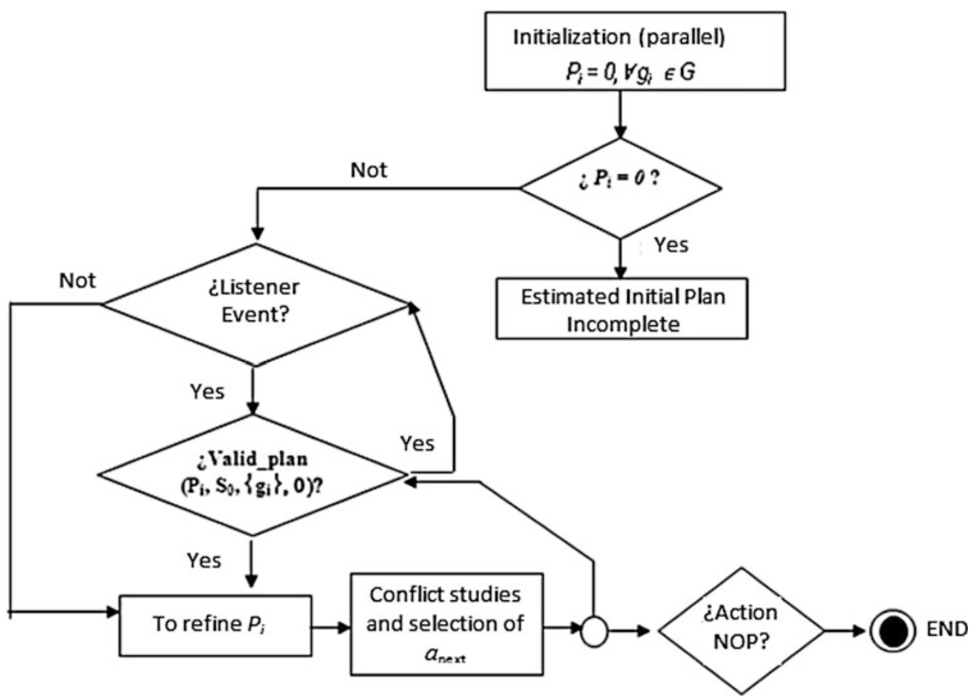
Generally speaking, the function of this model follows a reactive planning schema which starts from an initial state trying to find out within a limited interval of time which action may be the most adequate to reach their objectives.

This schema which has been followed by other proposals in literature [4] to deal with highly dynamic dominions different from the Web is very flexible in the setting of semantic LOs, for it allows one to take advantage of

Table 1 LOM to PDDL transformation

OWL (SLO, SIMS, LOD)	Domain PDDL
LOM:General: LOM: Identifier *LOM*: Title *LOM*: Language *LOM*: Description	***:action name_OP*** :precondition (*language-level languageOA ?who*)
LOM:Technical: LOM:Format LOM: Size LOM: Location LOM:Requirement LOM: Duration	: precondition (*format ?who formatOA*) (*size ?who sizeOA*) (location *?who LocationOA*) (requirement *?who multimedianOA*) (duration ?who durationOA)
LOM Educational: LOM: Interactive Type [Felder, 1996] LOM: Learning Resource Type LOM: Difficulty	: precondition (learning_style ?who styleOA) (user_favoriteformat ?who ResourceOA) (user_difficulty ?who diffcultyOA) : effect (task_OA_OP done ?who)
OWL:Class LOD:Concept	:precondition (user_knows ?who ConceptLOD) :effect (user_knows ?who Concept)

Fig. 1 Schema planning algorithm



execution time to make the best decisions regarding the following action of the plan to execute and it limits reaction time after an unexpected event, in other words, when the state of the world differs from the expected state.

This mode of on-line operation may be seen in the diagram in Fig. 1, in which time is established by the petitions of action the execution model requests. Then, the functional schema of this reactive planning model is based on the idea of decomposing the original planning problem (*PP*) into independent sub-problems. Hence, the planning algorithm calculates concurrently a *Pi* plan separately for each one of objectives of problem "g_i" ε *G*.

The calculation of each P_i plan is done incrementally, when a possibly incomplete initial P_i plan is constructed (or a skeleton), constantly monitored by an event monitor, being refined as the planner has available time. Finally, existing conflicts between said plans are studied to decide which action from which plan will be the first to be executed (a_{next}). The planning model sends a_{next} to the execution model when it requests it, and then updates its beliefs of the world assuming that the action sent will have a successful execution. In case an action fails, the execution model notifies the planning model so that it may once again calculate the plans on the previous state to the a_{next} sent. All of the above process repeats continuously until the user decides to stop the planner's execution, or until when the planning component has reached all objectives; to do so, it returns a special action: *NOP* (no operation). *NOP* is an action without preconditions or effects which allows the planner to wait some time without performing any task. During this latter time, if a user introduces new objectives, the planner will calculate new plans again. This last objective modification characteristic also applies to any point in time in which the planning process is conducted.

Characterization of the Uncertain Behavior of the Environment

In a virtual learning environment, once an LO has been chosen as part of a sequence which reaches an objective, events may occur as for instance that there has been a change in the metadata of an LO which is part of the current learning route, or a new LO may have appeared in the environment, or that an old LO may have been eliminated, or that the link to an LO's own resource is broken. These events inevitably affect a current composition plan, enabling the possibility of invalidating totally or partially the route currently constructed as a solution to a student's learning need.

Then, the idea is to review each step of planning, and see if there has been any change in the environment which may affect the composition plan (learning route) in each step of the planning, and if it happens, an immediate warning must go off allowing one to evaluate if the actions that have already been considered have been affected. In case the answer is yes, the plan is discarded and a new dominion is constructed; on the other hand, the type of event is reviewed and acts depending on an event. An important aspect to highlight is to remember that in this case LOs are available in a repository and that their existence, elimination or updating is measured depending on the metadata recorded in it since the model adopted does not include the complete download of an LO's own content, because what is done is to record the link of the resource itself. Thus, the idea is to try to manage events from their origin in the repository, so that they may be communicated to the planning mechanism almost instantaneously. External changes in the state of the world potentially affect the planning processes in their actions, statements, facts and objects in the description of the domain's partially-generated plans. To monitor these events, adding an event listener that distinguishes the various types of events is necessary (see Fig. 2).

In each step of the i plan before invoking a chosen action in an S_i state, the system must listen to the events to validate if there has been one that has altered the state of the world. S_i means there are no events on the event line; then, *a* is applied

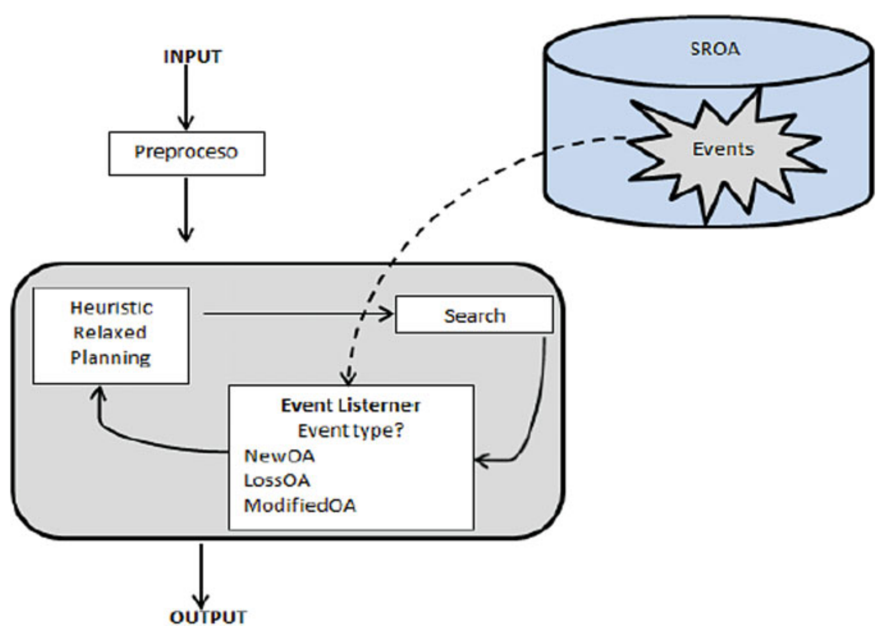


Fig. 2 Mechanism to manage the events

to a S_i state followed by the $i + 1$ step, assuring that the fragment of actions from S_0 to S_i are correct. Nevertheless, the system must replan in the following types of events:

A New Operator's Case

If a new operator (LO) is made available, first review if that operator's instance is associated to at least one of the concepts of the course. This is done to compare if it affects the execution plan; in the event of a negative response (it is not associated) that event is ignored. On the other hand, if it is associated to those concepts, an evaluation is performed which consists in a replanning decision. First, it uses the same initial state of the original partial plan adding the new operator, to construct a relaxed *RP*". Second, we estimate length *h*(*RP*"), with all the actions as a route to the solution of the initial plan.

Lost Operator Case

If a planned operator is eliminated from the current plan, this plan considers that operator appears, and it will try to substitute it for another operator that produces the same effects, and which is currently registered in the dominion (named alternative actions). The replanning decision, begging when all the actions of the plan which may have been affected related to the operator that is going to stop existing are marked off. If they are actions in the plan associated to that operator, the process will continue normally. In case replanning is necessary, the plan shall be evaluated from the state in which the fault was reported so that in case that this case is called to become part of the plan it may be considered failed beforehand and it will obligate the search for alternative action that will achieve the same effects starting from a given state. If there is no action, the system will return a message of impossibility to create a route to the user.

An Objective Changes

This specific case forces one to consider a current state, *S*, the same as the initial state, *So*, and starting there, one is obligated to construct a new planning problem including the objectives that were not fulfilled (or already fulfilled), new objectives. A corresponding RPG will be constructed for them. If there are already actions (LO) which are part of the solution plan (learning route), they will be marked as valid actions of the plan and after them it adds the other actions which may or not have been fulfilled yet, and which were part of the original problem, or which may correspond to the solution to these objectives.

Execution of an LO

There is another type of fault related to the evaluation of the availability of an LO as a resource. This type of uncertainty must be calculated when an action (LO) is part of a valid plan, but it fails, when your web address (defined in the metadata) is invoked. Many times those addresses lose connectivity, or because of web errors they do not allow the download of the resource. This is why each action that is part of the solution plan must be invoked before assuming the task of requesting the following action. This is done so that the state may be reviewed to see if it is possible to access a resource; in case, it is possible to access the resource it will be downloaded and saved in its corresponding sequence order and the state will be updated with the effects of that action. On the other hand, it is necessary to try to find an action that will allow it to replace the previously failed action.

Planning Algorithms for Uncertain Events

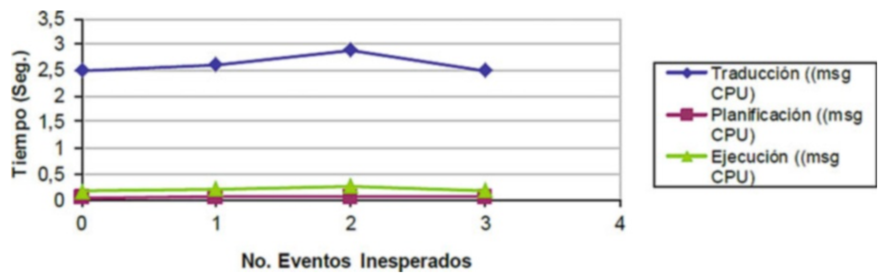
The general behavior of a reactive planning process must be connected to the event listener mechanism, and to do so we follow the procedure described as follows: a first updating relation surges at the beginning of the planning process, in which the current state is made to coincide with initial state of the problem ($So = I$). A second updating relation is the one that takes place once the a_{next} is sent to the execution layer, and updates the model of its S_o environment with the expected effects of the $a_{next,}$ for which the planning model does the following: i) a copy of the current environment state is made as a previous state: $S_{prev} = S_o$, ii) The new current state, $S_{0,}$ of the planning component is calculated as follows: result (a_{next}, S_o, ϕ); The above implies that it saves the trace of the previous state and it modifies the current state, S_0, in which the latter process is carried out considering that the execution of a_{next} will have expected effects, in other words assuming that there are no external factors (ϕ). This operating model is named planning based on assumptions [6].

A third relation that arises is when the listener mechanism of events reports an unexpected event. In this case, the type of event is evaluated as follows: i) If it corresponds to an addition of an LO to the repository and it affects the currently constituted plan (an LO is classified and saved under one of the concepts and the task of the course), the process of sending an a_{next} must be stopped, a copy of the existing state of the environment will be made as a previous state: $S_{prev} = S_o$, and a new operator is included in the dominion to proceed with the calculation of a new RPG starting from a new S_o. ii) If it corresponds to the elimination of an LO from the repository and it affects the currently constituted plan (the LO is classified and saved under one of the course concepts and it corresponds to one of its own activities),

Table 2 Example of uncertain in the composer

Case	LRM1_a124_OP2_task_exam	LRM1_a1234_task_narrative_text	No. of LOs executed	Fluent	Length of plan	Learning route
1	LO available	LO available	3	1	3	1. LRM1_a1234_task_narrative_text 2. LRM1_a1234_task_diagram 3. LRM1_a12_task_diagram
2	LO not available	LO available	3	1	3	1. LRM1_a1234_task_narrative_text 2. LRM1_a1234_task_diagram 3. LRM1_a12_task_diagram
3	LO not available	LO not available	0	0	0	No solution
4	LO available	LO not available	3	2	3	1. LRM1_a124_OP2_task_exam 2. LRM1_a1234_task_diagram 3. LRM1_a12_task_diagram

Fig. 3 Time uncertain events



the process of sending the a_{next} must be stopped, and we make a copy of the existing state of the environment as a previous state: $S_{prev} = S_o$, and we evaluate: if the LO is already part of the solution plan, and if it has already been executed, we proceed to make a request for the following a_{next}, starting from the current state, S_o, or if the action is not part of any solution plan, or if it must be invoked again, we locate the action that represents the eliminated LO and the RPG is constructed again, but this time from the new S_o. iii) If the case that takes place corresponds to the updating of metadata, one must review to see that updated LO represents one of the actions in the plan, and to do so it was evaluated as follows: a given LO is part of the solution plan, in case it is already part of the solution plan and it has been executed, we proceed with the request for the following a_{next}; in case it is part of the solution plan up to now and it has not been executed yet we make $S_{prev} = S_o$ and once again the domin-ion and the RPG are constructed , but this time starting from a new S_o.

Experimental Case

In this Table 2 we show how we manipulate the model, so that while building the learning route, deleting one or more LO, in order to evaluate all the solutions. In these cases we built Learning Route for a student with learning style "active" where an "exam" is more recommended than "text".

As an example from Table 2, explained first, that the planner executes the plan without any problem (all LO are available). in this case, the composition is whatsoever just guided by the convenience of choosing an LO which best adapts to a student's profile; nevertheless, in cases 2 and 4, when a message with the log of events is received in which the most convenient LO has been eliminated from the repository, the system proceeds to eliminate from the domain the action which represented it and consequently, it is obligated to include a new plan. And even though the LO now include is not the most convenient to student, it does sure to he will achieve the same objective of knowledge allowing the continuation of the rest of the plan until the objective is fulfilled. In case 3, the example corresponds to the elimination of several LO, this which impedes at planner the constructing a learning route in accor-dance with a student's needs.

Another important aspect to consider is when we face the problem of an uncertain behavior, we find it represented in the time the composer mechanism uses to react before events, for the exemplified LO elimination case (see Fig. 3). As we can see, the costs of time are very low since the behavior curve is constant.

Conclusions and Future Work

This article provides a brief analysis of the characterization process of a learning route composition problem under the application of a reactive planning technique, seen from the perspective of a student's role. How the reactive planning model along with the event listener allows three types of uncertain events (an LO is available, an LO of a plan is no longer available or a changing objective learning state) was shown in detail and demonstrated online and entirely done automatically. This solution ensures finding a long learning path. Although the first option is the convenience LO student profile, if one is not available replaced by another or others which ensure achieving the desired knowledge.

Acknowledgment The authors thank Colciencias and Universidad Nacional de Colombia for the co financing of the Project "An Infrastructure to Generate Educational Applications Based on Digital Television using Semantic Learning Objects", coded 1118-569-33723. Contract number 0204-2013.

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Biodiesel Production in Stirred Tank Chemical Reactors: A Numerical Simulation

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Abstract

The biodiesel production was performed in stirred tank chemical reactor by numerical simulation. The main results are that the percentage of conversion from triglyceride to biodiesel is approximately of 82 % when the molar flow ratio between triglyceride/alcohol is 1:5. This system displays only one equilibrium point. Since there are imaginary eigenvalues in the Jacobian matrix analysis, the equilibrium point is unstable. The biodiesel production in stirred tank chemical reactor is good because the settling time is short, and has higher conversion.

Keywords

Biodiesel • Equilibrium point • Numerical simulation • CSTR • Phase map

Introduction

Among the preoccupations that concern the world are the reduction of petroleum reserves due to the increased use of diesel fuel, and the environmental issues that caused by the climate change [1]. This situation brings significant attention to new alternatives that secure the future energy supplies. For instance, researchers have focused their investigations to explore plant-based fuels, plant oils, and fats as promising biofuel sources [2]. As a result, biodiesel derived from vegetable oils, can be an option to replace diesel. Biodiesel, consisting of methyl esters of fatty acids [3], is commonly obtained by transesterification of vegetable oils that compromise triglycerides with methanol in the presence of a catalyst [4]. This clean renewable fuel is superior to diesel oil in terms of sulfur and aromatic content. Also, it is environmentally safe, non-toxic, and biodegradable.

The esterification reaction to generate biodiesel can be conceptually represented by the following in (1):

$$Alcohol + Fatty\ Acid \Leftrightarrow Ester + Water \tag{1}$$

The transesterification reaction can be carried out in a continuous stirred tank chemical reactor (CSTR). CSTR are the most common process unit in chemical industries process [5]. Since CSTR are the central part of a whole chemical process the stability study is a relevant topic [6, 7].

The purpose of this work is to investigate the dynamic behavior of CSTR in biodiesel production. We consider the case where vegetable oil reacts with methanol to form methyl esters (biodiesel) and glycerol in the presence of a homogeneous alkaline catalyst (NaOH). This case is investigated because alkali-catalyzed transesterification is the most often used in industry. First, the simulations are performed at different feed ratios of alcohol to oil. Second,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_14, © Springer International Publishing Switzerland 2015

the equilibrium points will be computed. Finally, the bifurcation and stability analysis is also explored.

The present article is arranged in the following order. In section "Theory", shows theoretical background to develop this investigation. Section "Case Study", the continuous stirred tank chemical reactor model is explained for the case of biodiesel production. Section "Results" shows the results and discussion based on the case study.

Theory

Dynamic Model for CSTR

The dynamics of **CSTR** in which m reactions take place involving $n\,(n > m)$ chemicals species can be described by (2):

$$\begin{aligned}\dot{C} &= \theta(C_{in} - C) + Er(C,T) \\ \dot{T} &= \theta(T_{in} - T) + Hr(C,T) + \gamma(u - T)\end{aligned} \tag{2}$$

where:

- $C \in \mathbb{R}^n$ is the vector of concentrations of chemical species.
- $C_{in} \in \mathbb{R}^n$ is the vector non-negative and constant feed concentrations.
- $T \in \mathbb{R}$ is the vector temperature.
- $T_{in} \in \mathbb{R}$ is the vector feed temperature.
- $r(C, T) \in \mathbb{R}^m$ is the smooth, non-negative, bounded vector of reaction kinetics, with $r(C, T) = 0\ \forall\ t \leq 0$.
- $E \in \mathbb{R}^n \times \mathbb{R}^m$ is the stoichiometric matrix.
- $H(C, T) \in \mathbb{R}^m$ is the smooth, bounded row vector of reaction enthalpies, with $H(C, T) = 0\ \forall\ t \leq 0$.
- θ is the reactor dilution rate (i.e. flow rate/volume).
- γ is the heat transfer parameter.
- u is the jacket or wall temperature, which is taken as the *control input*.

Non-linearity in models of equations (2) are introduced by the reaction kinetics *r(C, T)*. Commonly, *r(C, T)* has a polynomial or rational dependency on C and has an Arrhenius dependency on T. Due to this kinetics, CSTRs can display a great variety of dynamic behaviors from multiplicity of steady states to sustained oscillations, including odd attractors [8]. Figure 1 sketched a CSTR.

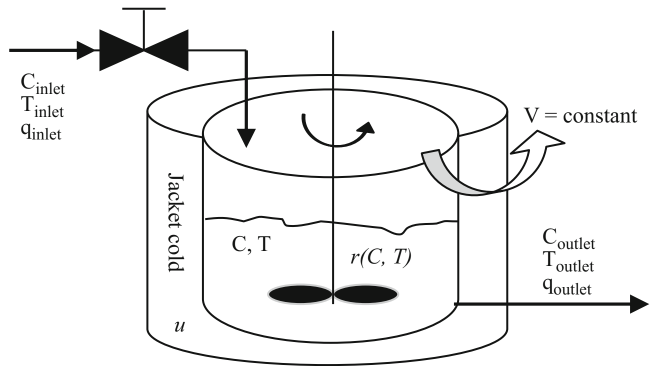

Fig. 1 Continuous stirred tank chemical reactors (based on [9])

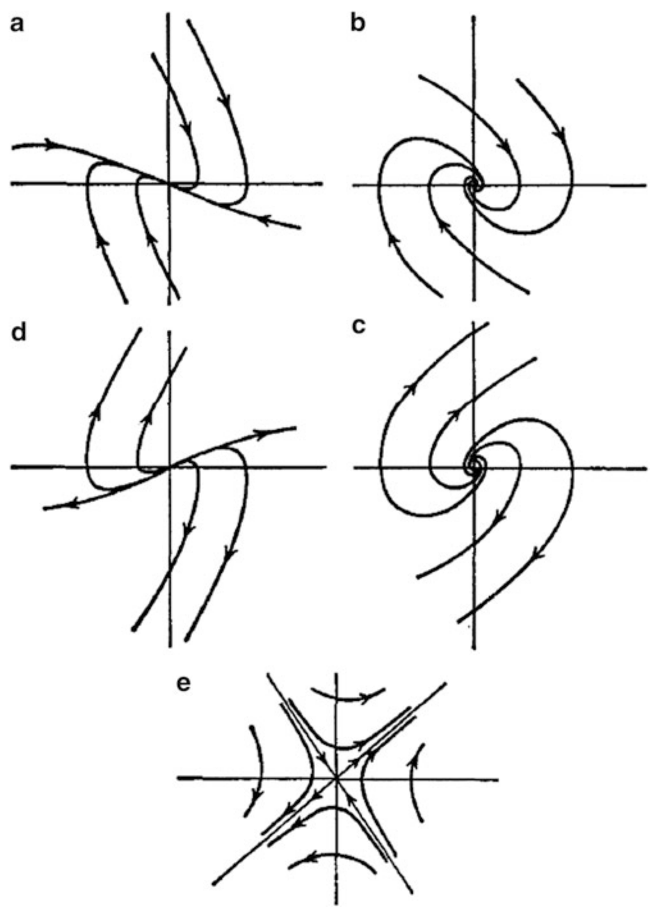

Fig. 2 (**a**) Stable node, (**b**) stable focus, (**c**) unstable focus, (**d**) unstable node, (**e**) saddle point [15]

Bifurcation and Stability Analysis

Bifurcation theory provides tools for a system stability analysis under its parametric changes [10]. As the parameters undergo changes, the existence of multiple steady states, sustained oscillations and traveling waves might occur for highly nonlinear processes [11].

Nonlinear system theory states that if all eigenvalues of the Jacobian matrix lie in the open left half of the complex plane, the system is stable [12]. Conversely, the steady state is unstable if the Jacobian matrix has at least one eigenvalue in the open right half of the complex plane [13, 14].

Figure 2 shows the distinction between stable and unstable nodes or foci is made to indicate that the trajectories move toward the stable type of critical point and away from the unstable point. The saddle point arises when the roots of the characteristic equation are real and have opposite sign. In this case there are only two trajectories that enter the critical point, and after entering, the trajectories may leave the critical point (permanently) on either of two other trajectories. No other trajectory can enter the critical point, although some approach it very closely [14, 15].

Case Study

Triglyceride (oils/fats) reacts with alcohol in the presence of the catalyst to give fatty acid alkyl esters and glycerol. The reaction proceeds in three steps as shown below [16]:

$$\begin{aligned} &Triglyceride + ROH \rightleftharpoons Diglyceride + RCOOR_1 \\ &Diglyceride + ROH \rightleftharpoons Monoglyceride + RCOOR_2 \\ &Monoglyceride + ROH \rightleftharpoons Glycerol + RCOOR_3 \end{aligned} \tag{3}$$

where r_1, r_2, and r_3 are the fatty acid chains associated with glycerol in triglyceride. K_1, K_2, K_3 are equilibrium constants, $K_1 = k_1/k_2, K_2 = k_3/k_4$, and $K_3 = k_5/k_6$ where $k_1, k_2, \ldots, k_6$ are forward and backward rate constants. The reaction rates of each reaction are as follows:

$$\begin{aligned} r_1 &= k_1 C_{TG} C_{ROH} - k_2 C_{DG} C_{AE_1} \\ r_2 &= k_3 C_{DG} C_{ROH} - k_4 C_{MG} C_{AE_2} \\ r_3 &= k_5 C_{MG} C_{ROH} - k_6 C_G C_{AE_3} \end{aligned} \tag{4}$$

where $C_{TG}, C_{DG}, C_{MG}, C_G, C_{AE}$ *and* C_{ROH} represent the concentration of triglyceride, diglyceride, monoglyceride, glycerol, alkyl esters (biodiesel), and alcohol respectively. The temperature dependency of the rate constant (k_i) is expressed by Arrhenius' law [17]:

$$k_i = k_i^o \exp(-E_{A,i}/RT) \tag{5}$$

The kinetic parameters are shown in Table 1. The heat of the reaction is $\Delta H_R = -5.07 \times 10^3$ J/mol for each reaction, calculated from the heat of the formation data [16].

Table 1 Kinetic parameters [18]

	Symbol	Units	Symbol	Units
Component	K^o	L/ mol min	E_A	J/mol
TG	1.469×10^8		58,782.6720	
ROH	105,100		44,962.0452	
DG	1.19×10^{10}		67,193.9532	
AE	1.725×10^8		58,225.8276	
MG	24,940		30,031.9164	
G	1.469×10^8		46,042.2396	

Results

The numerical simulation of biodiesel production in a CSTR has been performed in MATLAB® version 2012b.

Figure 3 shows the concentration profile for the five molor flow ratios between triglyceride/alcohol in the case of $\theta = 0.06\ \text{min}^{-1}$, $u = 5, T_c = 400\ K$, the set of ODEs are integrated using the explicit Runge-Kutta method. All shapes are asymptotic. The settling time diminish when the molar flow ratio between triglyceride/alcohol increase. The relationship between settling time and the molar flow ratio is the form $\tau_A^{1:5} < \tau_A^{1:4} < \tau_A^{1:3} < \tau_A^{1:2} < \tau_A^{1:1}$. In addition, we can see that concentration of biodiesel ($C_{Biodiesel}$) in steady state increase when molar flow ratios between triglyceride/alcohol increase.

Hence, the molar flow ratio between triglyceride/alcohol is too sensitive for biodiesel production.

Figure 4 shows reactor temperature profile for the five molor flow ratios between triglyceride/alcohol. In addition, be seen that reactor temperature (T) in steady state diminish when the ratios of triglyceride/alcohol increase.

In order to find the equilibrium points of CSTR, the model of CSTR in steady state was solved by Newton method for nonlinear system equation. The vector of equilibrium points for five molor flow ratios between triglyceride/alcohol are given in Table 2. Notice that the maximum production of biodiesel is given when the molar flow ratio between triglyceride/alcohol is 1:5. It is in agreement with [19] since there is excess of alcohol. Notice that, there is only one equilibrium point for each molar flow ratio of triglyceride/alcohol. Many studies about characterization and stabilization of equilibrium points show that when

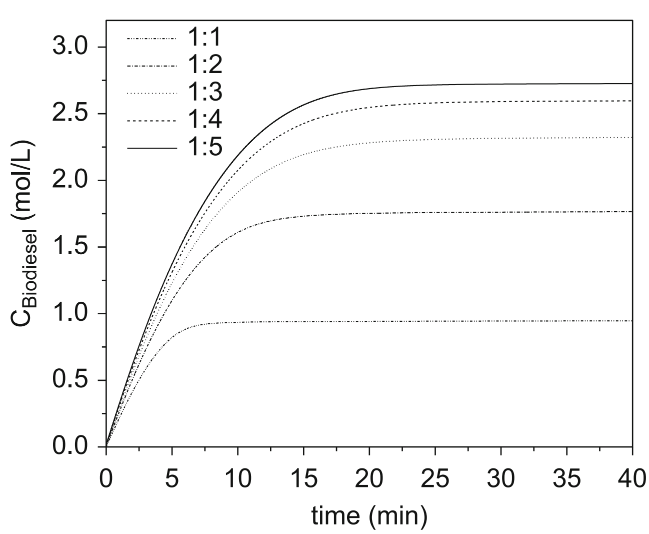

Fig. 3 Concentration profiles at different molar flow ratios of triglyceride/alcohol

there is only one equilibrium point in the dynamic system it should be an unstable point. Aforementioned is in agreement with [6, 8, 20].

Figure 5 illustrates the bifurcation map for 5 M flow ratios between triglyceride/alcohol. All shapes diminish asymptotically and the shape with more pronounced slope is when the molar flow ratio between triglyceride/alcohol is 1:5. In addition, the biodiesel concentrations increase when the dilution rates diminish because at less value of dilution rate the behavior of CSTR is like a batch reactor. This is not desirable because there are not advantages of continuous process.

Hence, the molar flow ratio 1:5 of triglyceride/alcohol is chosen to perform the phase maps, concentration behavior and temperature behavior because this relationship leads to the maximum biodiesel concentration.

Figure 6 shows phase maps in steady state without control in the case of molar flow ratio between triglyceride/alcohol is 1:5. These maps give an idea of the maximum reference point that could be fed in some control law. Also, in Fig. 6 we observe that the maximum biodiesel composition approximately is 2.72 mol/L. It is important know that because the conversion could not be reached for higher value in the input. In addition, the maximum conversion is reached when cooling jacket temperature is 435 K.

Figure 7 presents concentration profiles for the most important components in the case of molar flow ratio between triglyceride/alcohol is 1:5, $\theta = 0.06\ \mathrm{min}^{-1}$, $u = 5$, $T_c = 400\ K$. The purpose of this figure is to show that as the reaction precedes, the concentration of the desired product, biodiesel, increases asymptotically. However, after 26 min there is no significant change in these values. This situation also happens with the rest of the components which no further change occurs after reaching this time. In addition, triglyceride concentration in steady state ($\tau_A \simeq 26$ min) is around of 0.18 mol/L, and the conversion of triglyceride is compute by $(1 - 0.18)/1 \times 100 = 82\ \%$.

Figure 8 depicts the dynamic behavior of reactor temperature. Also, the settling time of the reactor (τ_A) and the steady state reactor temperature are 26.3 min and 350.52 K respectively. In addition, there is the relationship between settling time and dilute rate as follows $\tau_A \approx 4\theta^{-1}$ because $4\theta^{-1} = 4(0.15\ \mathrm{min}^{-1})^{-1} = 26.67\ \mathrm{min} \approx \tau_A$, this is according with [20, 21].

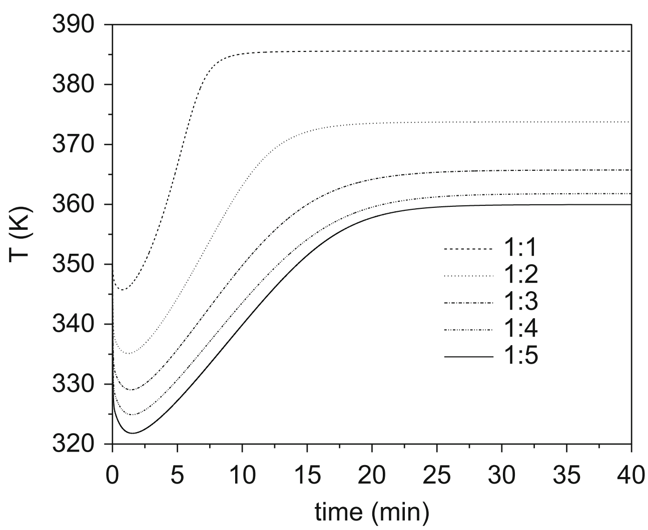

Fig. 4 Temperature profiles at different molar flow ratios of triglyceride/alcohol in the case of $\theta = 0.06\ \mathrm{min}^{-1}$, $u = 5$ and $T_c = 400$ K

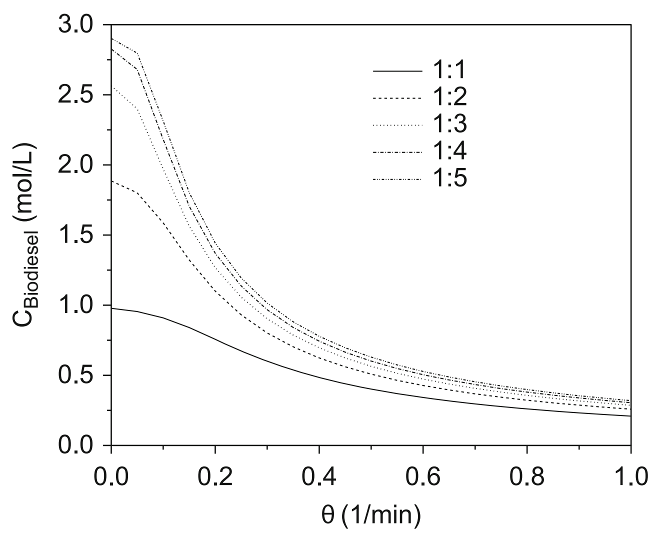

Fig. 5 Bifurcation map at different molar flow ratios between triglyceride/alcohol in the case of $u = 5$ and $T_c = 400$ K

Table 2 Equilibrium points with $\theta = 0.06\ \mathrm{min}^{-1}$, $u = 5$ and $T_c = 400$ K

Variable	1:1	1:2	1:3	1:4	1:5
C_{TG}	0.485757	0.354256	0.204793	0.118279	0.060609
C_{ROH}	0.052477	0.375401	0.820382	1.491553	2.278931
C_{DG}	0.225128	0.108635	0.055797	0.030556	0.018673
C_{MG}	0.144949	0.156599	0.108546	0.076785	0.059503
C_G	0.144166	0.421412	0.640027	0.775062	0.861034
C_{AE}	0.947523	1.627004	2.179937	2.508459	2.721064
T	385.5553	363.0733	360.0601	358.7252	359.6604

C_i concentration of component i, mol/L; T temperature, K

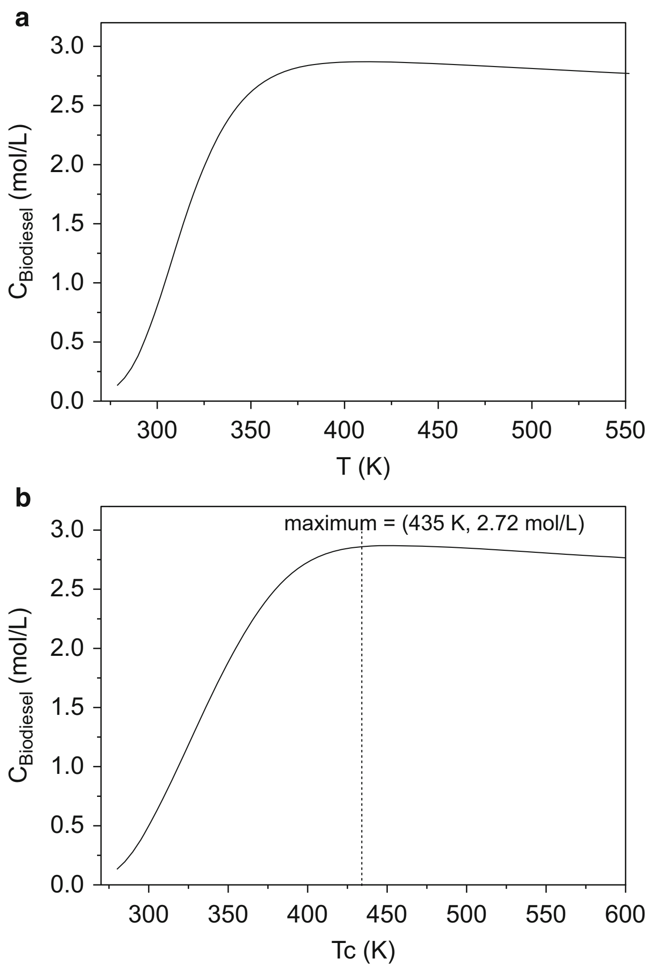

Fig. 6 (**a**) Phase map in steady state for $C_{Biodiesel}$ versus reactor temperature. (**b**) Phase map in steady state for $C_{Biodiesel}$ versus cooling jacket temperature. Both cases with $\theta = 0.06\ \text{min}^{-1}$, $u = 5$ and $T_c = 400$ K

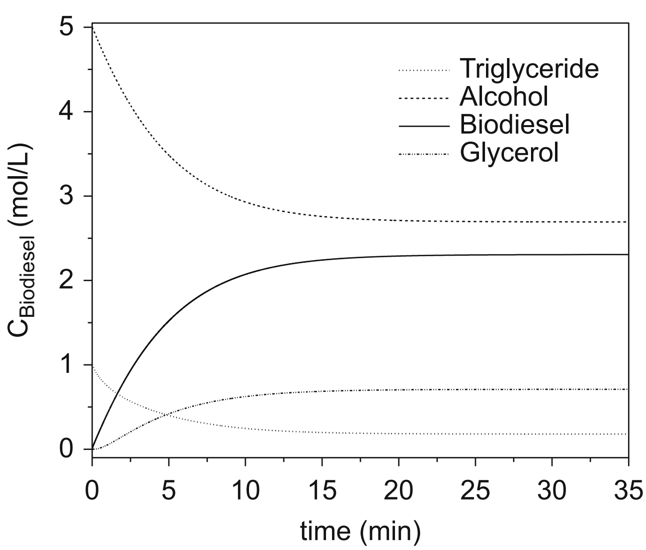

Fig. 7 Concentration behavior of principal components carryout in CSTR for biodiesel production

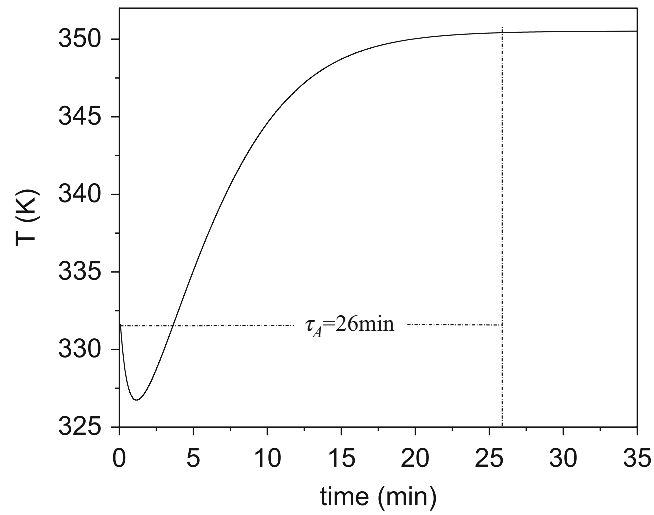

Fig. 8 Temperature behavior in the case of $\theta = 0.15\ \text{min}^{-1}$, $u = 5$ and $T_c = 435$ K

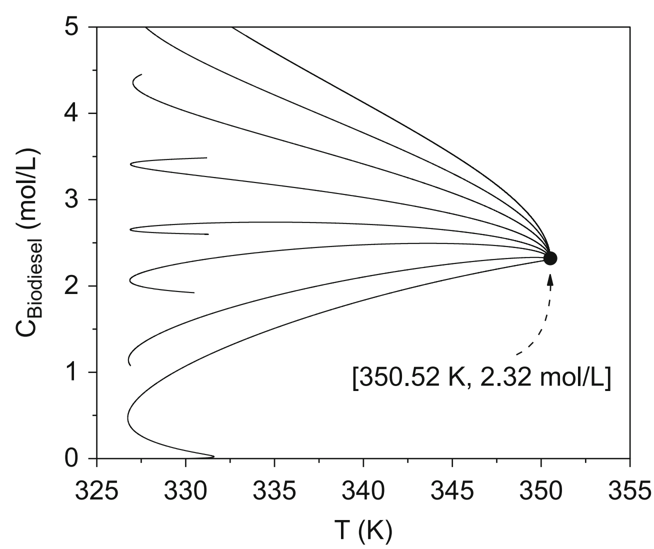

Fig. 9 Phase diagram in the case of $\theta = 0.15\ \text{min}^{-1}$, $u = 5$ and $T_c = 435$ K

The phase-plane plot has been constructed by performing simulations for a large number of initial conditions when the molar flow ratio between triglyceride/alcohol is 1:5. Figure 9 sketches the phase portrait in steady state without control. One steady state value is clearly shown. Also, all shapes converge to equilibrium point (350.52 K, 2.32 mol/L). In addition, the phase map is similar to Fig. 2c, thus the equilibrium point can be characterized as an unstable focus. To confirm the characterization of equilibrium point was perform an eigenvalue analysis for the Jacobian matrix at steady state. Therefore, there are two complex root there the equilibrium point is an unstable point. Detail about this stability analysis is available in the appendix.

Conclusion

The CSTR is a good continuous process to carry out biodiesel production because the settling time is short, and acceptable conversion of triglycerides to biodiesel is achieved. However, this process is expensive because is needed excess of alcohol.

Acknowledgment The author is grateful to Dr. Ever Peralta for his revisions in the text. This work was support in part by COMECYT.

Appendix

The particular model for the case study is governed by the next set of ordinary differential equations.

$$\begin{aligned}
\frac{dx_1}{dt} &= \gamma_1 - \theta x_1 - \beta_1 x_1 x_2 + \beta_2 x_3 x_4 \\
\frac{dx_2}{dt} &= \gamma_2 - \theta x_2 - \beta_1 x_1 x_2 + \beta_2 x_3 x_4 \\
&\quad - \beta_3 x_3 x_2 + \beta_4 x_5 x_6 - \beta_5 x_5 x_2 + \beta_6 x_7 x_8 \\
\frac{dx_3}{dt} &= \gamma_3 - \theta x_3 + \beta_1 x_1 x_2 - \beta_2 x_3 x_4 \\
&\quad + \beta_3 x_3 x_2 - \beta_4 x_5 x_6 \\
\frac{dx_4}{dt} &= \gamma_4 - \theta x_5 + \beta_1 x_1 x_2 - \beta_2 x_3 x_4 \\
\frac{dx_5}{dt} &= \gamma_5 - \theta x_5 + \beta_3 x_3 x_2 - \beta_4 x_5 x_6 \\
&\quad + \beta_5 x_5 x_2 - \beta_6 x_7 x_8 \\
\frac{dx_6}{dt} &= \gamma_6 - \theta x_6 + \beta_3 x_3 x_2 - \beta_4 x_5 x_6 \\
\frac{dx_7}{dt} &= \gamma_7 - \theta x_7 + \beta_5 x_5 x_2 - \beta_6 x_7 x_8 \\
\frac{dx_8}{dt} &= \gamma_8 - \theta x_8 + \beta_5 x_5 x_2 - \beta_6 x_7 x_8 \\
\frac{dx_9}{dt} &= \gamma_9 - \theta x_9 + \Delta H_{rxn}\left[\beta_1 x_1 x_2 - \beta_2 x_3 x_4\right. \\
&\quad \left. + \beta_3 x_3 x_2 - \beta_4 x_5 x_6 + \beta_5 x_5 x_2 - \beta_6 x_7 x_8\right] + u(T_c - x_9)
\end{aligned} \tag{6}$$

Where: $\beta_i = k_i^o \exp(-E_{A,\,i}/Rx_9)$ *and* $\gamma_i = \theta x_{i,0}$

Using the expansion Taylor series the dynamic system governing by (6) can be represented by (7).

$$\frac{d\underset{\sim}{x}}{dt} = \underset{\sim}{f}$$

$$\underset{\sim}{f} = \underset{\sim}{f}\,\underset{\sim}{x}_e + \underset{\sim}{A}\Big|_{\underset{\sim}{x}_e}\left[\underset{\sim}{x} - \underset{\sim}{x}_e\right]$$

Reordering the before equation

$$\frac{d\underset{\sim}{x}}{dt} = \underbrace{\left.\begin{bmatrix} \frac{\partial f_1}{\partial x_1} & \cdots & \frac{\partial f_1}{\partial x_9} \\ \vdots & \ddots & \vdots \\ \frac{\partial f_9}{\partial x_1} & \cdots & \frac{\partial f_9}{\partial x_9} \end{bmatrix}\right|}_{\underset{\sim}{A}\qquad \underset{\sim}{x}_e} \underbrace{\begin{bmatrix} x_1 \\ \vdots \\ x_9 \end{bmatrix}}_{\underset{\sim}{x}} \underbrace{+ \underset{\sim}{f}_{\underset{\sim}{x}_e} + \underset{\sim}{A}\Big|_{\underset{\sim}{x}_e} \underbrace{\begin{bmatrix} x_{1,e} \\ \vdots \\ x_{9,e} \end{bmatrix}}_{\underset{\sim}{x}_e}}_{B} \tag{7}$$

In deviation variables $\underset{\sim}{\xi} = \underset{\sim}{x} - \underset{\sim}{x}_e$ (7) have the next form:

$$\frac{d\underset{\sim}{\xi}}{dt} = \underset{\sim}{A}\Big|_{\underset{\sim}{x}_e} \underset{\sim}{\xi} + \underset{\sim}{f}\,\underset{\sim}{x}_e \tag{8}$$

For the case study the function *Jacobian* matrix values are:

$$\underset{\sim}{f}\,\underset{\sim}{x}_e = \begin{bmatrix} -19.43 \\ -166.33 \\ -27.48 \\ 19.44 \\ -68.06 \\ 46.93 \\ 99.97 \\ 99.97 \\ -194461.33 \end{bmatrix} \times 10^{-15} \cong \underset{\sim}{0} \tag{9}$$

$$\underset{\sim}{A}\Big|_{\underset{\sim}{x}_e} = \begin{bmatrix} -872.98\times10^{-3} & -45.97\times10^{-3} & -17.29\times10^{-3} & -927.18\times10^{-6} & 0 & 0 & 0 & 0 & 1\times10^{-6} \\ -692.98\times10^{-3} & -06.97\times10^{-3} & -3.15 & 927.18\times10^{-6} & -1.97 & 26.09\times10^{-3} & 61.28\times10^{-3} & 61.28\times10^{-3} & -1.23\times10^{-6} \\ 692.98\times10^{-3} & 97.16\times10^{-3} & 3.12 & -927.18\times10^{-6} & -283.93\times10^{-3} & -26.09\times10^{-3} & 0 & 0 & 1.95\times10^{-9} \\ 692.98\times10^{-3} & 3.77\times10^{-3} & -17.29\times10^{-3} & -927.18\times10^{-6} & -150\times10^{-3} & 0 & 0 & 0 & 1\times10^{-6} \\ 0 & 0 & -3.14 & 0 & 1.82 & -26.09\times10^{-3} & -61.28\times10^{-3} & -61.28\times10^{-3} & 198.43\times10^{-9} \\ 0 & 51.19\times10^{-3} & 3.14 & 0 & -283.93\times10^{-3} & -26.09\times10^{-3} & 0 & 0 & 957.55\times10^{-3} \\ 0 & 59.82\times10^{-3} & 0 & 0 & 59.82\times10^{-3} & 0 & -61.28\times10^{-3} & -61.28\times10^{-3} & 960.03\times10^{-3} \\ 0 & 59.82\times10^{-3} & 0 & 0 & 2.26 & 0 & -61.28\times10^{-3} & -61.28\times10^{-3} & 960.03\times10^{-3} \\ -839.72 & -190.22 & -3.78\times10^{3} & 1.12 & -2.39\times10^{-3} & 31 & 74.26 & 74.26 & .5.15 \end{bmatrix} \quad (10)$$

The eigenvalues of *Jacobian* matrix can be find by (11).

$$\det\left(\underset{\sim}{A}\Big|_{\underset{\sim}{x}_e} - \lambda \underset{\sim}{I}\right) = 0 \quad (11)$$

The eigenvalues are displays in (12).

$$\underset{\sim}{\lambda} = \begin{bmatrix} -5.15 \\ 3.57 \\ 481.58\times10^{-3} + 2.44i \\ 481.58\times10^{-3} - 2.44i \\ -834.84\times10^{-3} \\ -59.79\times10^{-3} \\ -1.082\times10^{-3} \\ -516.2\times10^{-6} \\ 99.54\times10^{-18} \end{bmatrix} \quad (12)$$

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Evaluation of Optimal Control-Based Deformable Registration Model

Naleli Jubert Matjelo, Fred Nicolls, and Neil Muller

Abstract

This paper presents an evaluation of an optimal control-based deformable image registration model and compares it to four well-known variational-based models, namely, *elastic, fluid, diffusion* and *curvature* models. Using similarity and deformation quality measures as performance indices, Non-dominated Sorting Genetic Algorithm (NSGA-II) is applied to approximate Pareto Fronts for each model to facilitate proper evaluation. The Pareto Fronts are also visualized using Level diagrams.

Keywords

Deformable image registration • Similarity measure • Quality measure • NSGA-II • Pareto Fronts • Level diagrams

Introduction

DEFORMABLE image registration aims to find a physically realizable deformation that optimizes a chosen similarity measure between two images. That is, image registration can be thought of as a bi-objective minimization problem whereby two key objectives are *similarity measure* and *deformation quality measure*.

Similarity measure provides a quantitative measure of correspondence between two image features and it is upon this quantifiable correspondence measure that optimization problems can be formulated such that alignment of features is maximized. *Sum of Squared Differences* (SSD) is one of the popular similarity measures in the literature (e.g. [1–3]) and it is adopted in this paper.

The deformation quality measure is perhaps the most vital component in the registration process since without it arbitrary deformations may lead to grid tangling and other undesirable solutions. The necessity of good quality deformations has led to development of various kinds of nonrigid/deformable models including variational-based ones like *elastic, fluid, diffusion* and *curvature* models [1, 2]. Each of these variational-based models is equipped with a unique regularizer which penalizes physically unrealizable deformations. Due to different regularizers one model may perform best for a particular application and worse in other applications, so evaluation of these models can serve as a useful guide in choosing a model suitable for a particular application during design, based on these two objectives.

Closely related to variational-based models is an optimal control approach to deformable image registration, whose formulation is based on Grid Deformation Method (GDM) [4, 5]. GDM is a well-established mathematical framework with interesting properties including the following:

- It is able to generate a grid with a desired grid density distribution that is free from grid tangling. This is achieved through a positive monitor function as a control input.
- It gives direct control over the cell size of the adaptive grid and determines the node velocities directly.

Elastic, fluid, diffusion and curvature models and their qualitative evaluations can be found in [2, 6]. This paper presents a comparison of the optimal control approach to

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_15, © Springer International Publishing Switzerland 2015

deformable image registration and these four variational-based methods. A Pareto Front scheme is adopted as a method of performing evaluations here and an elitist multi-objective evolutionary algorithm, Non-dominated Sorting Genetic Algorithm (NSGA-II), is used for approximating the Pareto Fronts for all models based on the two objectives, similarity measure and deformation quality measure.

The rest of the paper is organized as follows: *section* "*Deformable Image Registration*" summarizes the four variational-based models and the optimal control-based deformable registration model as well as their implementations. *Section* "*Evaluation in Fitness Space*" discusses the evaluation procedure in multidimensional fitness space. *Section* "*Results and Interpretation*" presents observations and their interpretations and finally *section* "*Conclusion and Recommendations*" concludes this work and suggests some possible future work.

Deformable Image Registration

Many methods have been suggested to solve the deformable image registration problem, however relatively less work has been published on the subject of their evaluation. The methods considered for registering a template image $T(\mathrm{x})$ to a reference image $R(\mathrm{x})$ in our work are variational-based models and an optimal control-based model. A brief summary of these models is given in the subsections below. Full details of these models can be found in [2, 4–6], respectively.

Variational-Based Models

Elastic, Fluid, Diffusion and Curvature models are variational-based models which all minimize the following objective functional with respect to a deformation field u(x) as outlined in [2]:

$$J[\mathrm{u}(\mathrm{x})] = \frac{1}{2} \parallel T(\mathrm{x} - \mathrm{u}(\mathrm{x})) - R(\mathrm{x}) \parallel_{L_2}^2 + \alpha S[\mathrm{u}(\mathrm{x})] \qquad (1)$$

where α is the regularization constant and $S[\mathrm{u}(\mathrm{x})]$ is the regularity/smoothness measure of u(x), which could be one of the following:

$$\begin{cases} S^{elas}[\mathrm{u}(\mathrm{x})] & = \int_\Omega \left\{ \frac{\mu}{4} \sum_{j,k=1}^{2} \left(\partial_{x_j} u_k + \partial_{x_k} u_j\right)^2 + \frac{\lambda}{2} (\nabla \cdot \mathrm{u})^2 \right\} d\mathrm{x} \\ S^{fluid}[\mathrm{u}(\mathrm{x})] & = S^{elas}[\mathrm{v}(\mathrm{x})] \\ S^{diff}[\mathrm{u}(\mathrm{x})] & = \frac{1}{2} \sum_{l=1}^{2} \int_\Omega \|\nabla u_l\|^2 d\mathrm{x} \\ S^{curv}[\mathrm{u}(\mathrm{x})] & = \frac{1}{2} \sum_{l=1}^{2} \int_\Omega (\Delta u_l)^2 d\mathrm{x} \end{cases} \qquad (2)$$

with the velocity v(x) of the displacement field. Parameters λ and μ are called *Lame-constants* which reflect material properties of an elastic body.

1) Elastic Model: Due to its local nature, the elastic model only allows small deformations and therefore makes it stiffer than other models like the fluid model [2]. This means it can be preferred in small deformation registration such as human brain alignment, but not in large deformation registrations like abdominal organ registration. This model penalizes affine transformations in its formulation, therefore, it requires a preregistration step. The elastic model is more likely to result in better deformation quality than the fluid model due to its relatively high resistance to deformation and therefore it can act as a reference for marking deformation quality of other models.

2) Fluid Model: This model is very similar in structure to the elastic model with only one exception that it minimizes the elastic potential of the velocity field, $S^{elas}[\mathrm{v}(\mathrm{x})]$. Consideration of the velocity field makes fluid models less stiff, hence they are capable of handling larger deformations better than elastic models. This model is reported to be time demanding, thus lighter alternatives like demon registration are preferred in the face of convergence time [2]. However, demon registration will not be part of this work since convergence time has not been chosen as one of the performance indices. Similar to the elastic model, this model also requires the pre-registration step.

3) Diffusion Model: The diffusion model is reported to be very fast and flexible [2], therefore it is very attractive for high-resolution applications such as three-dimensional magnetic resonance imaging (MRI) in conjunction with breast cancer surgery. It can therefore serve as a good reference with regard to flexibility and convergence speed. Similar to fluid and elastic models, the diffusion model penalizes affine transformations, hence the pre-registration step is theoretically necessary.

4) Curvature Model: Even though most deformable models require the pre-registration step, the degree of dependency varies. For instance, the curvature model does not penalize affine and perspective transformations, so it can do without the pre-registration step, as argued in [2]. This contention was further explored here by comparing the performances of all models with and without pre-registration. In this way, relative model dependency on pre-registration was investigated.

Optimal Control-Based Model

An optimal control formulation of the image registration problem based on GDM [5] seeks control inputs $f(\mathrm{x}) > 0$ and $g(\mathrm{x})$, as well as states $\phi(t, \mathrm{x})$ and u(x) such that the following performance index is minimized:

$$J(\phi(1,\mathrm{x}),f,g)=\frac{1}{2}\parallel T(\phi(1,\mathrm{x})-R(\mathrm{x})\|_{L_2}^2+\frac{\alpha}{2}\parallel f(\mathrm{x})\parallel_{H^1}^2 + \frac{\beta}{2}\parallel g(\mathrm{x})\parallel_{H^1}^2 \tag{3}$$

subject to:

$$\begin{cases} \nabla\cdot \mathrm{u}(\mathrm{x})=f(\mathrm{x})-1 & \forall \mathrm{x}\in\Omega \\ \nabla\times \mathrm{u}(\mathrm{x})=g(\mathrm{x}) & \forall \mathrm{x}\in\Omega \\ \mathrm{n}\cdot \mathrm{u}(\mathrm{x})=0 & \forall \mathrm{x}\in\Gamma \\ \dfrac{\partial\phi(t,\mathrm{x})}{\partial t}=\mathrm{u}(t,\phi(t,\mathrm{x})) & \forall t,\mathrm{x}\in(0,1]\times\Omega \\ \phi(0,\mathrm{x})=\mathrm{x} & \forall \mathrm{x}\in\Omega \\ \int_\Omega (f(\mathrm{x})-1)d\mathrm{x}=0 & \forall \mathrm{x}\in\Omega, \end{cases} \tag{4}$$

where α and β are penalty parameters and ϕ, Ω and Γ are the deformed grid, the image domain and its boundary respectively. It is also necessary that $\phi(t,\mathrm{x})\in(0,1]\times\Gamma,\forall\mathrm{x}\in\Gamma$.

Unlike the variational-based models mentioned above, the optimal control-based model is able to offer direct control of deformation grid size and velocity, therefore deformation quality can be monitored and controlled with ease.

Model Implementation

1) Variational and Optimal Control Problem Solvers: Direct methods for solving variational and optimal control problems require two fundamental tools: *(a) method for solving differential equations and integral functions and (b) a method for solving nonlinear optimization problems*. In this paper the *Gradient Algorithm* was used as an optimization method, the *Finite Element Method* was used for solving partial differential equations and the *Gauss-Legendre Quadrature Scheme* was used to handle function integrations.

2) Pre-Registration: Deformable models, presented above, are solved iteratively and thereby often suffer from convergence problems whenever good initial guesses are not available. In such cases, reliable global parametric registration methods, like the affine method, come in handy in the preregistration step. Simplex (Nelder-Mead) search method [7] and Levenberg-Marquardt [8] were used to solve the affine pre-registration problem below:

$$\min_A \parallel T(A(\mathrm{x})-R(\mathrm{x})\parallel_{L_2}^2 \tag{5}$$

where $A\mathrm{x}$ is an affine transformation of x:

$$A\mathrm{x}=\begin{bmatrix} a_1 & a_2 & a_3 \\ a_4 & a_5 & a_6 \\ 0 & 0 & 1 \end{bmatrix}\begin{bmatrix} x \\ y \\ 1 \end{bmatrix}.$$

The scalar variables a_i in matrix A describe translation, rotation, anisotropic scale and shear of the transformation.

Evaluation in Fitness Space

Motivated by Everingham et al. [9] on evaluation of image segmentation methods using Pareto Fronts, we adopt a general form of aggregate fitness function:

$$H(a_\mathrm{p},I)=\Phi(h_1(a_\mathrm{p},I),\ldots,h_n(a_\mathrm{p},I)),$$

where a_p represents the registration algorithm a with parameters p, I is a set of images and $h_i(a_\mathrm{p},I)$ are individual fitness functions defined to increase monotonically with the fitness of some particular aspect of the algorithm's behavior. The behavior of an algorithm a applied to a set of images I with a particular choice of parameters p can be characterized by a point in the n–dimensional space defined by evaluation of the fitness functions. Variation of parameters p of algorithm a produces a new point in the fitness space and this makes it possible to compare algorithms having different parameter sets. This representation decouples the fitness trade-off from the particular parameter set used by an individual algorithm.

Pareto Front Approximation

1) Pareto Front: Points generated by variations of algorithm parameters are plotted on the fitness space and the set of non-dominated points on the fitness space is given by:

$$\left\{\langle a_\mathrm{p}\in P_a,H(a_\mathrm{p},I)\rangle\big|\tilde{\exists} a_\mathrm{q}\in P_a:H(a_\mathrm{q},I)>H(a_\mathrm{p},I)\right\},$$

where P_a is the parameter space of algorithm a, H is a vector of fitness functions $\langle h_1(a_\mathrm{p},I),\ldots,h_n(a_\mathrm{p},I)\rangle$ and a partial ordering relation on H is defined as:

$$\begin{gathered} H(a_\mathrm{q},I)>H(a_\mathrm{p},I)\Leftrightarrow\forall i:h_i(a_\mathrm{q},I)\geq h_i(a_\mathrm{p},I) \\ \textit{and} \\ \exists i:h_i(a_\mathrm{q},I)>h_i(a_\mathrm{p},I). \end{gathered} \tag{6}$$

This construction, referred to as "Pareto Front", is extendable to the case of multiple algorithms and this has the natural consequence that any algorithm which does not contribute to the front can be considered a bad choice for any monotonic aggregate fitness function Φ [9].

Representation (6) above is also abbreviated as p $\prec$ q, which means solution p dominates q.

2) Approximation by NSGA-II: Genetic algorithms are commonly used to approximate the Pareto Front with a finite number of points, and have an advantage in that they require less prior knowledge about the behavior of an algorithm with respect to its parameters [9]. NSGA-II is an elitist multi-objective evolutionary algorithm introduced by Deb and Agarwal as an improved version of the NSGA and a brief description of it is given below. More details about NSGA-II can be found in [10] and the Matlab implementation by Seshadri [11] was used in this paper.

In NSGA-II, for each solution p $\in P$ one has to determine the following:

- The set, S_{p} of solutions q dominated by solution p. i.e. $S_{\mathrm{p}} = \{\mathrm{q} : \mathrm{p} \prec \mathrm{q}\}$
- The number n_{p} of solutions dominating solution p.
- The number p_{rank} indicating the *rank* of solution p. i.e. the front index.
- *Crowding-distance* of solution p which gives the density of solutions surrounding solution p with respect to each objective function.

Binary tournament selection is performed using *crowding-comparison operator* which takes the following into account:

- Non-domination *rank*, p_{rank} of individual p: In the tournament, during selection, non-dominated solutions with lesser *rank* are preferred.
- *Crowding-distance* of solution p: If competing solutions in the tournament share the same *rank* then the solution with high *Crowding-distance* is preferred.

The following *genetic operators* are used to produce offspring:

- *Simulated Binary Crossover (SBX)*:

$$\begin{cases} c_{1,k} = \frac{1}{2}\left[(1-\beta_k)p_{1,k} + (1+\beta_k)p_{2,k}\right] \\ c_{2,k} = \frac{1}{2}\left[(1+\beta_k)p_{1,k} + (1-\beta_k)p_{2,k}\right], \end{cases}$$

  where $c_{i,k}$ is the i^{th} child with k^{th} component, $p_{i,k}$ is the selected parent and β_k (≥ 0) is a sample from a random number generated, having the density:

$$\begin{cases} p(\beta) = \frac{1}{2}(\eta_c + 1)\beta^{\eta_c}, & 0 \leq \beta \leq 1 \\ p(\beta) = \frac{1}{2}(\eta_c + 1)\frac{1}{\beta^{\eta_c+2}}, & \beta > 1, \end{cases}$$

  where η_c is the distribution index for crossover. This distribution can be obtained from a uniformly sampled random number in the interval (0,1).

- *Polynomial Mutation*:

$$c_k = p_k + \left(p_k^u - p_k^l\right)\delta_k,$$

  where c_k is the child and p_k is the parent with p_k^u and p_k^l being the upper and lower bounds on the parent component respectively. δ_k is a small variation which is calculated from a polynomial distribution by using:

$$\begin{cases} \delta_k = (2r_k)^{\frac{1}{\eta_m+1}} - 1, & if \;\; r_k < 0.5 \\ \delta_k = 1 - [2(1-r_k)]^{\frac{1}{\eta_m+1}}, & if \;\; r_k < 0.5, \end{cases}$$

  where r_k is a uniformly sampled random number in the open interval (0,1) and η_m is mutation index.

The offspring population is combined with the current generation population and a selection is performed to set the individuals of the next generation. Since all the previous and current best solutions are preserved in the population, elitism is ensured.

Fitness (Objective) Functions

1) Similarity Measure: SSD and Cross-Correlation (CC) are the most popular geometry-based similarity measures [1]. Although restricted to mono-modality applications, SSD is adopted here due to it being simple, intuitive and computationally efficient. The normalized SSD is much more convenient as a well bounded objective function $h_1(a_{\mathrm{p}}, T, R, \phi)$:

$$h_1\left(a_{\mathrm{p}}, T, R, \phi\right) = \frac{\| T(\phi(\mathrm{x})) - R(\mathrm{x}) \|_{L_2}^2}{\| T(\mathrm{x}) - R(\mathrm{x}) \|_{L_2}^2}, \tag{7}$$

where a stands for a registration algorithm for a particular model and subscript p stands for parameters of the model which can be varied as part of the decision variables to investigate optimal tuning of the model in algorithm a. For variational-based models $\phi(\mathrm{x}) = \mathrm{x} - \mathrm{u}(\mathrm{x})$ and $\frac{\partial}{\partial t}\phi(t,\mathrm{x}) = \mathrm{u}(t, \phi(t,\mathrm{x}))$ for the optimal control-based model.

2) Deformation Quality Measure: The quality of a grid is quantified by skewness, shape, size, aspect ratio and orientation [12, 13], thus a proposed quality measure should account for most if not all of these quantities. However, skewness varies almost linearly with shape. Also, aspect ratio is meaningful only if skew is insignificant. Shape is preferred because it contains both skew and aspect ratio [14]. In this paper we adopt two of these measures and they are shown below as formulated for quadrilateral grid elements in [14]:

- Size Measure for each grid element:

$$f^e_{size} = min\left(\frac{\alpha^e_k}{2w}, \frac{2w}{\alpha^e_k}\right),$$

- Shape Measure for each grid element:

$$f^e_{shape} = \frac{8}{\sum_{k=0}^{3} \frac{trace\left(A^e_k T A^e_k\right),}{\alpha^e_k}}$$

where w is the area of a reference element, $\alpha^e_k = det(A^e{}_k) + det(A^e{}_{k+2})$ and A^e_k is the Jacobian matrix of a grid element e computed at node k and given by:

$$A^e_k = \begin{bmatrix} \phi^e_{1_{k+1}} & -\phi^e_{1_k} & \phi^e_{1_{k+3}} & -\phi^e_{1_k} \\ \phi^e_{2_{k+1}} & -\phi^e_{2_k} & \phi^e_{2_{k+3}} & -\phi^e_{2_k} \end{bmatrix}, \quad k\in\{0,1,2,3\},$$

where $\left(\phi^e_{1_k}, \phi^e_{2_k}\right)$ is the coordinate pair at node k of element e in the deformed grid, $\phi(\mathrm{x})$.

Given that the grid element e is not inverted, both f^e_{size} and f^e_{shape} have the same range of [0,1], such that the best and worst qualities correspond to the upper and lower bounds of the range respectively. These *local* measures are the basis for more complicated and effective grid quality measures. To obtain the *global* quality measure from these *local* measures, we consider the infinity norm of shape and size measures computed over the entire grid. The infinity norm is convenient since it is able to capture and expose the worst element in the whole grid. These measures are shown below:

$$\begin{cases} h_2\left(a_{\mathrm{p}}, \phi\right) = \| 1 - f^e_{size}\|_\infty \\ h_3\left(a_{\mathrm{p}}, \phi\right) = \| 1 - f^e_{sharpe}\|_\infty . \end{cases} \tag{8}$$

Since h_2 and h_3 do not give a measure of how many grid elements are degenerate or inverted *(deformation failure*), we impose another *global* quality measure, which is a modification of the one given in [12]:

$$h_4\left(a_{\mathrm{p}}, \phi\right) = \frac{1}{2N}\sum_{e=1}^{N} \frac{\left\{\left|\alpha^e_k - \rho\right| - \alpha^e_k + \rho\right\}}{\left|\alpha^e_k - \rho\right| + \varepsilon}, \tag{9}$$

where N is the total number of elements in the grid, $0 < \varepsilon \ll 1$ is set to avoid division by zero and $0 < \rho < 1$ is a threshold used to isolate inverted elements with zero area. All objective functions h_i above are dimensionless.

Level Diagrams Analysis of Pareto Fronts

Level Diagrams are used for visualizing Pareto Fronts by providing a geometrical visualization of the Pareto Front based on a distance measure from an ideal solution point, which optimizes all objectives simultaneously [15]. Given normalized objectives h_i, as the ones above, a suitable norm is chosen to evaluate the distance from an ideal solution point to a point on the Pareto Front. In this paper we adopt the infinity norm for evaluating distance since it offers a more compact visualization of the Pareto Front which is useful for trade-off analysis, as argued in [15]. With the distances computed, the plot of Level Diagrams proceeds as follows.

The distance of all points from the Pareto Front are plotted against each objective h_i to produce two-dimensional plots with distance along the *y-axis* and objective h_i along the *x-axis*. In this way, a single point from the Pareto Front is plotted at one level (*y-axis*), carrying with it all the information about its performance on the Pareto Front. The smaller the distance measure on the Level Diagrams the better the performance, so a decision maker can use this distance measure to decide which point, corresponding to a particular algorithm, suits their design specifications. More details about Level Diagrams can be found in [15] and [16].

Results and Interpretation

Experiment Setup

In prostate related radiation therapies, there is a need to register prostate pose and shape captured (e.g. as CT data) during planning phase to that observed (e.g. as digital *X-Ray*) during treatment phase. This helps improve precision when applying dose to target tissues during treatment. One way to accomplish this registration is by producing 2D projections of the CT data, called *digitally reconstructed radiographs (DRRs)* and registering them to the digital *X-Ray*.

In this experiment we used only CT slices from different patients to perform prostate registration. CT slices were grouped into template set $T_s = \{T_i(\mathrm{x}) : i = 1,2,\ldots, 25\}$ (representing a *DRR* set) and reference set $R_s = \{R_i(\mathrm{x}) : i = 1, 2,\ldots, 25\}$ (representing a digital *X-Ray* set). The criteria used when pairing $T_i(\mathrm{x})$ with $R_i(\mathrm{x})$ was based on feature similarity and slice level correspondence between the two, however, no $T_i(\mathrm{x})$ was paired with itself.

A copy of T_s was passed through the pre-registration process and kept as set T_p of pre-registered templates. The experiment proceeded further by testing all models with and without pre-registration separately. The results are shown in Fig. 1 and their interpretation is presented in the subsections below together with the Pareto Fronts visualization shown in Fig. 2. Every parameter (decision variable), for all models, was varied in the open interval (0,1) to produce the Pareto Fronts. Decision variables λ and μ in the elastic and fluid models are bulk and shear modulus measured in Pascal units. Every other decision variable is dimensionless.

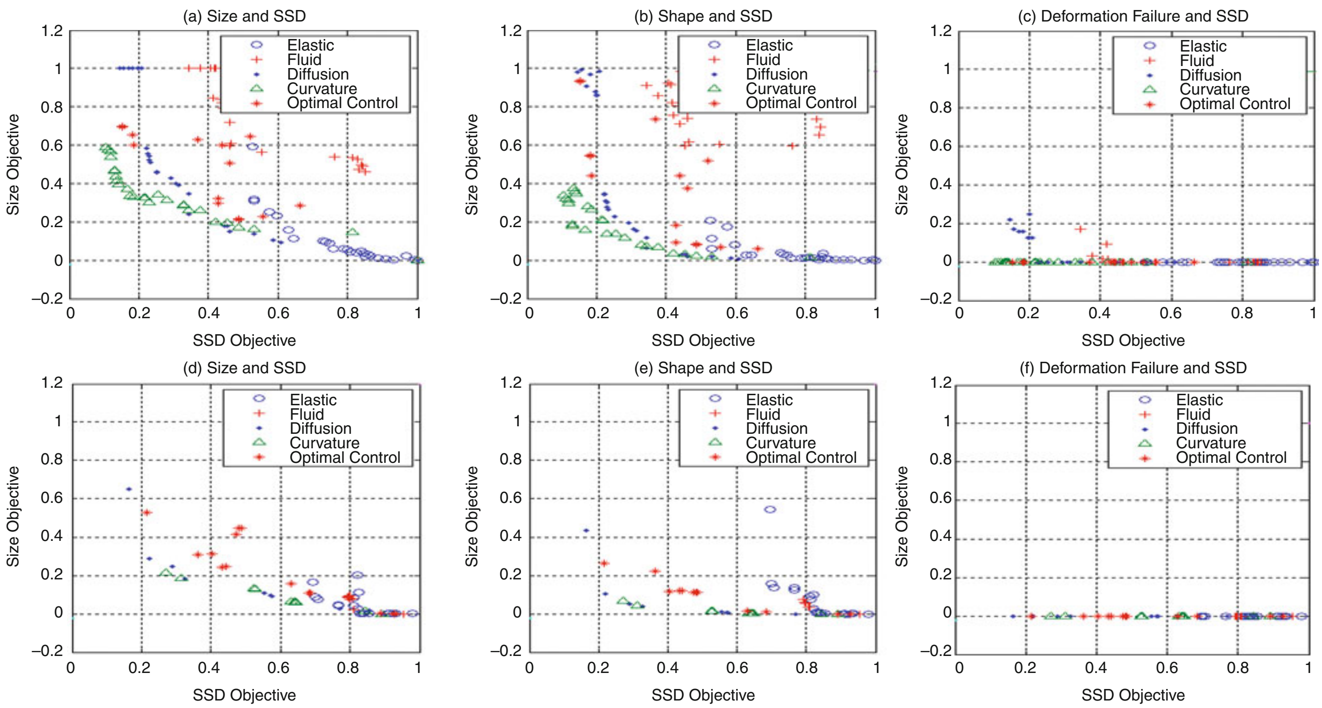

Fig. 1 Side views of Pareto Fronts approximation without pre-registration (**a**)–(**c**) as well as with pre-registration (**d**)–(**f**)

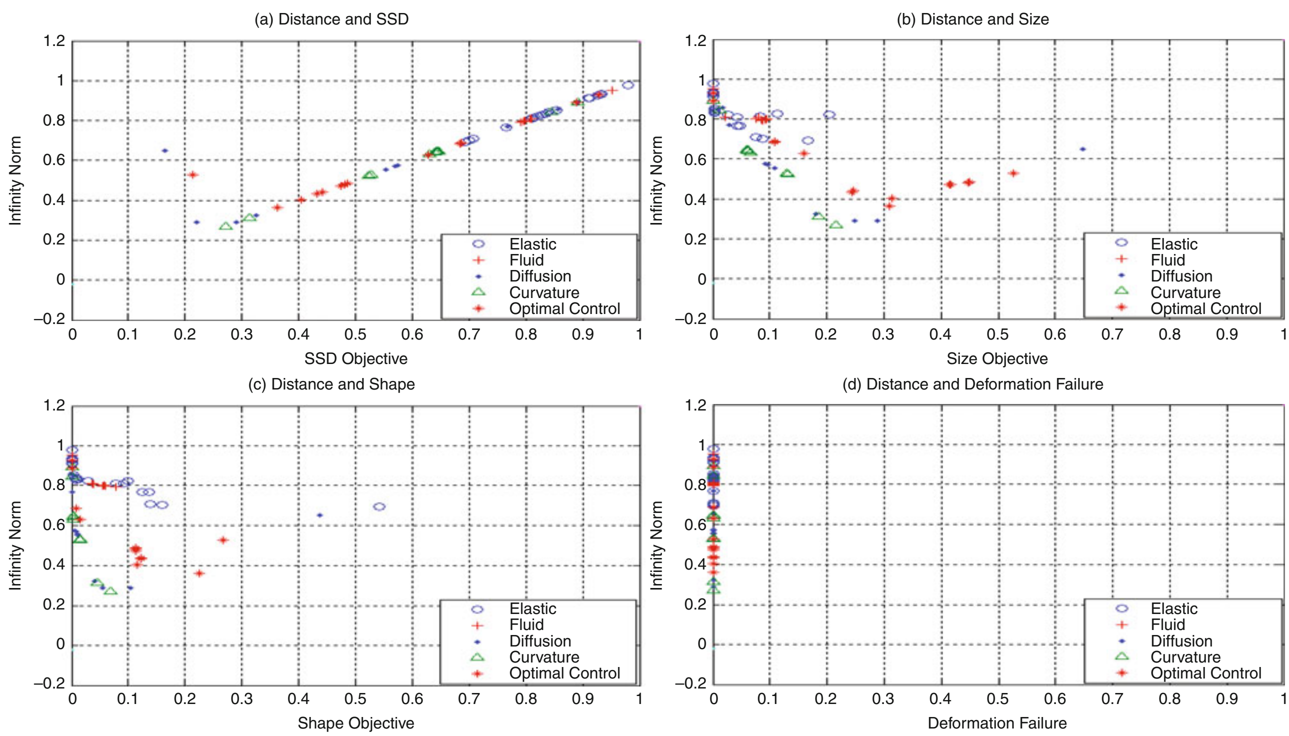

Fig. 2 Level diagrams corresponding to the Pareto Fronts with pre-registration

Pareto Fronts and Comparisons

In this subsection we compare each variational-based model with optimal control-based model using, as reference, the Pareto Fronts shown in Fig. 1. The term *net Pareto Front* refers to the non-dominated set of points contributed by the chosen Pareto Fronts in Fig. 1. As a summary of the Pareto Fronts shown in Fig. 1d–f, and for decision making purposes, Level Diagrams representation of the Pareto Fronts is shown in Fig. 2. Since all objective functions are dimensionless, no units are put in Figs. 1 and 2.

1) Comparison with Curvature Model: From Fig. 1a, b, the curvature model dominates the left side of the net Pareto Front with better SSD as well as deformation quality based on size and shape. It can also be observed from Fig. 1c that the curvature model has no deformation failure throughout. Also its performance did not change much when pre-registration is included except for a slight improvement in size as observed in Fig. 1d. These observations confirm the assertion that the curvature model can do without a preregistration step. It is worth noting also that the curvature model outperformed the optimal control-based model throughout with regard to size, shape and SSD measures, however both performed equally with consideration of deformation failure measure.

2) Comparison with Elastic Model: As expected, the elastic model manifests signs of stiffness as it is dominated by all methods on the left region of Fig. 1a, b, with respect to SSD objective. On their own, elastic and optimal control-based models share their net Pareto Front with elastic performing better in size and shape while optimal control is better in SSD as shown in Fig. 1a, b, d and e. With regard to deformation failure measure the two models are non dominated. Including pre-registration results in the elastic model performing better with respect to size objective but not in shape objective.

3) Comparison with Fluid Model: In the absence of preregistration, the fluid model is dominated by almost all the other models throughout in terms of size, shape and SSD, except for the elastic model, since the only way for it to participate in the net Pareto Front is by allowing it to compete with the elastic model alone. So even though both optimal control and fluid models are not members of the net Pareto Front, the optimal control-based model would participate in the absence of the curvature model while the fluid model still would not. The fluid model also produced a significant number of physically unrealizable elements as shown in Fig. 1c. However, when pre-registration is included the elastic model improves in all deformation quality measures and starts to participate in the overall net Pareto Front.

4) Comparison with Diffusion Model: Excluding the curvature model for the moment, we realize that optimal control and diffusion models dominate the left-center region of the net Pareto Front side by side. This observation is not surprising given that the first variations of these models are expressible by almost similar Poisson equations, with the exception of optimality equations in the case of optimal control-based model. With regard to deformation failure, the optimal control-based model is non-dominated with deformation failure of zero throughout and thus justifying the claim that it gives direct control of element structure. The diffusion model has shown a significant improvement in deformation quality measures, relative to that of the optimal control-based model, when preregistration is included. This demonstrates higher sensitivity of the diffusion model to pre-registration with regard to deformation quality measure. With pre-registration included, the net Pareto Front of diffusion and optimal control-based models is now occupied mostly by the diffusion model.

Visualization Using Level Diagrams

The visualization presented in Fig. 2 is based only on the objectives and distance measures but no decision variables are included. Given Fig. 2, a decision maker who prioritizes the SSD objective over other objectives, for example, needs to use Fig. 2 (a) and to choose the model with the lowest infinity norm at the desired SSD objective value.

Conclusion and Recommendations

In this paper we compared the optimal control-based model with variational-based models for a deformable registration problem and observed that diffusion, curvature and optimal control-based models perform well in all objectives while elastic and fluid models have limited convergence, SSD. We have also shown the relative necessity of the pre-registration step for all models, with curvature model having the lowest while diffusion and fluid having the highest dependency on the pre-registration step. More importantly we showed how Level Diagrams can be used to guide the decision maker who wishes to choose a particular model suitable for their application.

All models were implemented on a framework of Finite Elements, however this results in some models, like the fluid model, being significantly slow due to high computational intensity. In future work, when convergence time is considered as one of the objectives, every model should be implemented in the framework that allows its optimum run-time. Also, evaluation based on different image modalities needs to be considered in future work.

Acknowledgment This work was supported by iThemba LABS, Medical Radiation Group through provision of the CT data set used to facilitate the evaluation procedure. Prof. Braae's support and comments on this paper are highly appreciated.

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The Effect of Mutual Coupling on a Microstrip Printed Antenna Array Operates at 5 GHz Using Three Different Substrates

Ali Elrashidi, Islam Ashry, Khaled Elleithy, and Hassan Bajwa

Abstract

Curvature has a great effect on fringing field of a microstrip antenna. Consequently, the fringing field affects the effective dielectric constant and then all antenna parameters. A new mathematical model for return loss mutual coupling coefficient as a function of curvature for two element array antenna is introduced in this paper. These parameters are given for TM10 mode and using three different substrate materials RT/duroid-5880 PTFE, K-6098 Teflon/Glass and Epsilam-10 ceramic-filled Teflon.

Keywords

Fringing field • Curvature • Effective dielectric constant and return loss (S11) • Mutual coupling coefficient (S12) • Transverse magnetic TM_{10} mode

Introduction

Microstrip antenna array conformed on cylindrical bodies is a commonly used antenna in aircrafts. Furthermore, they are used in millimeter-wave imaging arrays mounted on unmanned airborne vehicles, and antennas for medical imaging applications which may be required to conform to the shape of the human body [1–3]. Low profile, low weight, low cost and their ability of conforming to curve surfaces [4] are some of the advantages of microstrip antennas. Therefore, conformal microstrip structures have witnessed enormous growth in the last few years.

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Some advantages of conformal antennas over the planer microstrip structure include, easy installation, capability of embedded structure within composite aerodynamic surfaces, better angular coverage and controlled gain, depending upon shape [5, 6].

While Conformal Antenna provides potential solution for many applications, it has some drawbacks due to embedding [7]. The two main disadvantages of microstrip antenna arrays are the narrow frequency band and the mutual coupling between the basic elements is higher than in the traditional antenna arrays [8–10]. Mutual coupling between array elements affects the radiation pattern and input impedances. The radiation from one element in the array induces currents on the other elements to a nearby and scatters into the far field. The induced current derives a voltage at the terminals of other elements [11].

In this paper we present the effect of fringing field on the performance of a conformal patch antenna. A mathematical model that includes the effect of curvature on fringing field and on antenna performance is presented. In section "Cylindrical-Rectangular Patch Antenna", we present the cylindrical-rectangular patch antenna. In section "Conformal Microstrip Antenna Array and Mutual Coupling", we discuss conformal microstrip antenna array and mutual coupling. In section "Input Impedance", we introduce a model for

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_16, © Springer International Publishing Switzerland 2015

calculating the impendence. In section "Mutual Coupling", we show how the mutual coupling is calculated. Results are presented and discussed in section "Results".

Cylindrical-Rectangular Patch Antenna

All the reported work in literature for a conformal rectangular microstrip antenna assumes that the curvature does not affect the effective dielectric constant and the extension on the length [12]. The effect of curvature on the resonance frequency has been previously presented in literature [13]. The cylindrical rectangular patch is the most famous and popular conformal antenna (Fig. 1). The manufacturing of this antenna is easy with respect to spherical and conical antennas. Effect of curvature of conformal antenna on resonance frequency been presented by Clifford M. Krowne [13, 14] as:

$$(f_r)_{mn} = \frac{1}{2\sqrt{\mu\varepsilon}}\sqrt{\left(\frac{m}{2\theta a}\right)^2 + \left(\frac{n}{2b}\right)^2} \qquad (1)$$

where *2b* is a length of the patch antenna, *a* is a radius of the cylinder, *2θ* is the angle bounded the width of the patch, *ε* represents electric permittivity and *μ* is the magnetic permeability.

Joseph A. et al. presented an approach to the analysis of microstrip antennas on cylindrical surface. In this approach, the field in terms of surface current is calculated, while considering dielectric layer around the cylindrical body. The assumption is only valid if radiation is smaller than the stored energy [15]. Kwai et al. [16] gave a brief analysis of a thin cylindrical-rectangular microstrip patch antenna which includes resonance frequencies, radiation patterns, input impedances and *Q* factors. The effect of curvature on the characteristics of TM_{10} and TM_{01} modes is also presented in Kwai et al. paper. The authors first obtained the electric field under the curved patch using the cavity model and then calculated the far field by considering the equivalent magnetic current radiating in the presence of cylindrical surface. The cavity model, used for the analysis is only valid for a very thin dielectric. Also, for much small thickness than a wavelength and the radius of curvature, only *TM* modes are assumed to exist. In order to calculate the radiation patterns of cylindrical-rectangular patch antenna, the authors introduced the exact Green's function approach. Using Eq. (4), they obtained expressions for the far zone electric field components E_θ and E_φ as a functions of Hankel function of the second kind $H_p^{(2)}$. The input impedance and *Q* factors are also calculated under the same conditions.

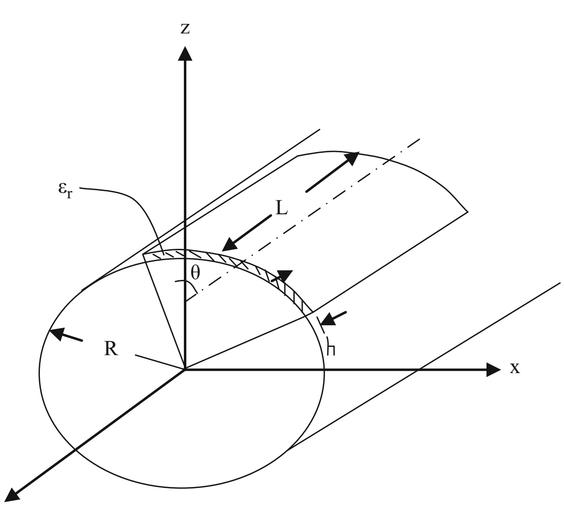


Fig. 1 Geometry of cylindrical-rectangular patch antenna [9]

Based on the cavity model, microstrip conformal antenna on a projectile for GPS (Global Positioning System) device is designed and implemented by using perturbation theory is introduced by Sun L., Zhu J., Zhang H. and Peng X [17]. The designed antenna is emulated and analyzed by IE3D software. The emulated results showed that the antenna could provide excellent circular hemisphere beam, better wide-angle circular polarization and better impedance match peculiarity.

Nickolai Zhelev introduced a design of a small conformal microstrip GPS patch antenna [18]. A cavity model and transmission line model are used to find the initial dimensions of the antenna and then electromagnetic simulation of the antenna model using software called FEKO is applied. The antenna is experimentally tested and the author compared the result with the software results. It was founded that the resonance frequency of the conformal antenna is shifted toward higher frequencies compared to the flat one.

The effect of curvature on a fringing field and on the resonance frequency of the microstrip printed antenna is studied in [19]. Also, the effect of curvature on the performance of a microstrip antenna as a function of temperature for TM_{01} and TM_{10} is introduced in [20, 21].

Conformal Microstrip Antenna Array and Mutual Coupling

Conformal microstrip arrays are used to increase the directivity of the antenna and increase the signal to noise ratio. Better performance is achieved using arrays. The radiation pattern is significantly affected using arrays on a conformal surface to appear as omnidirectional pattern, which is very useful in aerospace systems [22]. The equations of directivity function of the conformal microstrip array on a cylinder and the experimental results of pattern of array of 64 elements are given by M. Knghou et al. [22]. The coupling between the elements is not considered in [22]. The authors calculated the total electric field strength for an array of N elements using Eq. (2).

$$E = \sum_{i=1}^{N} \mathrm{E}_i e^{-j\varphi_i} \tag{2}$$

where, E_i represents the field strength of number i radiator and φ_i is the phase of equivalent transversal magnetic current source of N radiators.

C. You et al. designed and fabricated a composite antenna array conformed around cylindrical structures [23]. The experimental results showed that the radiation pattern is strongly dependent on the cylindrical curvature for the transverse radiation pattern, while the array also exhibits high side-lobes and wider beam width. Problems associated with Ultra Wide Band (UWB) antennas as phased array elements discussed in [24]. The authors introduced various wide bandwidth arrays of antennas that can be conforming. Problems that arise depending on the physical separation of antennas are discussed in the paper. Conformal placement, of an antenna, either as an individual antenna, or as in an array configuration on any arbitrary surface, may require very thin antenna. The authors should be processed preferably on flexible substrates so that the authors will conform to the surfaces without changing the surface geometry.

A. Sangster and R. Jacobs developed a finite element-boundary integral method to investigate the impedance properties of a patch element array for a microstrip printed antenna conformal on a cylindrical body [25]. A mutual coupling between elements is also studied in this paper for its great effect on the impedance properties. Simulation results for mutual coupling coefficient, S12, for a planar and conformal array are compared to a measured values and a good agreement is obtained. A full-wave analysis of the mutual coupling between two probe-fed rectangular microstrip antennas conformed on a cylindrical body is introduced by S. Ke and K. Wong [26]. The authors calculated the mutual impedance and mutual coupling coefficient using a moment of method technique [27, 28]. The numerical results of mutual impedance and mutual coupling coefficient are compared to the measured values for the microstrip antennas conformed on a cylindrical body with different radius of curvatures.

A comprehensive mathematical model for mutual impedance and mutual coupling between two rectangular patches array is introduced by A. Mohammadian et al. [29]. The authors replaced each microstrip antenna element in the array by an equivalent magnetic current source distributed over a grounded dielectric slab. A dyadic Green's function is developed for a grounded dielectric slab, using the rectangular vector wave functions. The active reflection performance and active radiation pattern of two elements in the array of microstrip antenna elements are calculated by S. Chen and R. Iwata [30]. The authors introduced a mathematical derivation of radiation pattern and reflection performance for each element in the array of microstrip antenna. Then by using the introduced model, the mutual coupling between the elements in the array is easily calculated.

N. Dodov and P. Petkov explored the mutual coupling between microstrip antennas provoked by the surface wave [31]. Based on the method of moments, the authors analyze the microwave structure on the microstrip antenna patch surface. N. Dodov and P. Petkov conclude that, the influence of surface wave is not significant in close neighboring resonant elements.

An accurate formula for the coupling between patch elements is introduced by Z. Qi et al. [32]. The classic formula for mutual coupling based on multi-port network theory ignores the impedance mismatching between antenna elements but on the other hand the introduced formula consider this mismatching between antenna array elements. A hybrid method, based on the method of moments, is introduced to analyze a microstrip antenna conformal on a cylindrical body by A. Erturk et al. [33]. The authors introduced three types of space-domain Green's function representations, each accurate and efficient in a given region of space. Input impedance of various microstrip antenna conformed on a cylindrical body and mutual coupling between two elements of the array is introduced and compared to some published results.

Input Impedance

The input impedance is defined as "the impedance presented by an antenna at its terminals" or "the ratio of the voltage current at a pair of terminals" or "the ratio of the appropriate components of the electric to magnetic fields at a point". The input impedance is a function of the feeding position [19]. To calculate the input impedance Z_{in} for the cylindrical microstrip antenna, we need to get the electric field at the surface of the patch. In this case, we can get the wave equation as a function of excitation current density J as follow:

$$\frac{1}{\rho^2}\frac{\partial^2 E_\rho}{\partial \text{ø}^2} + \frac{\partial^2 E_\rho}{\partial z^2} + k^2 E_\rho = j\omega\mu J \tag{3}$$

By solving this equation, the electric field at the surface can be expressed in terms of various modes of the cavity as [19]:

$$E_\rho(z, \text{ø}) = \sum\nolimits_n \sum\nolimits_m A_{nm}\psi_{nm}(z, \text{ø}) \tag{4}$$

where A_{nm} is the amplitude coefficients corresponding to the field modes. By applying boundary conditions, homogeneous wave equation and normalized conditions for ψ_{nm}, we can get an expression for ψ_{nm} as shown below:

1. ψ_{nm} vanishes at the both edges for the length L:

$$\left.\frac{\partial\psi}{\partial z}\right|_{z=0} = \left.\frac{\partial\psi}{\partial z}\right|_{z=L} = 0 \tag{5}$$

2. ψ_{nm} vanishes at the both edges for the width W:

$$\left.\frac{\partial\psi}{\partial \text{ø}}\right|_{\text{Ø}=-\theta_1} = \left.\frac{\partial\psi}{\partial \text{ø}}\right|_{\text{ø}=\theta_1} = 0 \tag{6}$$

3. ψ_{nm} should satisfy the homogeneous wave equation:

$$\left(\frac{1}{\rho^2}\frac{\partial^2}{\partial \text{ø}^2} + \frac{\partial^2}{\partial z^2} + k^2\right)\psi_{nm} = 0 \tag{7}$$

4. ψ_{nm} should satisfy the normalized condition:

$$\int_{z=0}^{z=L}\int_{\text{ø}=-\theta_1}^{\text{ø}=\theta_1} \psi_{nm}\psi_{nm}^{*} = 1 \tag{8}$$

Hence, the solution of ψ_{nm} will take the form shown below:

$$\psi_{nm}(z,\text{Ø}) = \sqrt{\frac{\varepsilon_m\varepsilon_n}{2a\theta_1 L}}\cos\left(\frac{m\pi}{2\theta_1}(\text{Ø}-\text{Ø}_1)\right)\cos\left(\frac{n\pi}{L}z\right) \tag{9}$$

with

$$\varepsilon_p = \begin{cases} 1 & for\ p=0 \\ 2 & for\ p\neq 0 \end{cases}$$

The coefficient A_{mn} is determined by the excitation current. For this, substitute Eq. (9) into Eq. (3) and multiply both sides of (3) by ψ_{nm}^{*}, and integrate over area of the patch. Making use of orthonormal properties of ψ_{nm}, one obtains:

$$A_{nm} = \frac{j\omega\mu}{k^2 - k_{nm}^2}\iint_{dim}^{feed} \psi_{nm}^{*} J_\rho d\text{Ø}dz \tag{10}$$

Now, let the coaxial feed as a rectangular current source with equivalent cross-sectional area $S_z \times S_\text{ø}$ centered at $(Z_0, \text{Ø}_0)$, so, the current density will satisfy the equation below:

$$J_\rho = \begin{cases} \dfrac{I_0}{S_z \times S_\text{ø}} & Z_0 - \dfrac{S_z}{2} \le x \le Z_0 + \dfrac{S_z}{2} \\ & \text{Ø}_0 - \dfrac{S_\text{ø}}{2} \le x \le \text{Ø}_0 + \dfrac{S_\text{ø}}{2} \\ 0 & elsewhere \end{cases} \tag{11}$$

Use of Eq. (14) in (13) gives:

$$A_{nm} = \frac{j\omega\mu l}{k^2 - k_{nm}^2}\sqrt{\frac{\varepsilon_m\varepsilon_n}{2a\theta_1 L}}\cos\left(\frac{m\pi}{2\theta_1}\text{Ø}_0\right)\cos\left(\frac{n\pi}{L}z_0\right) \times sinc\left(\frac{n\pi}{2L}z_0\right) sinc\left(\frac{m\pi}{2a\theta_1}\text{Ø}_0\right) \tag{12}$$

So, to get the input impedance, one can substitute in the following equation:

$$Z_{in} = \frac{V_{in}}{I_0} \tag{13}$$

where is the *RF* voltage at the feed point and defined as:

$$V_{in} = -E_\rho(z_0, \text{Ø}_0) \times h \tag{14}$$

By using Eqs. (4), (9), (11), (14) and substitute in (13), we can obtain the input impedance for a rectangular microstrip antenna conformal in a cylindrical body as in the following equation:

$$Z_{in} = j\omega\mu h \sum_n \sum_m \frac{1}{k^2 - k_{nm}^2}\frac{\varepsilon_m\varepsilon_n}{2a\theta_1 L}\cos^2\left(\frac{m\pi}{2\theta_1}\text{Ø}_0\right)\cos^2\left(\frac{n\pi}{L}z_0\right) \times sinc\left(\frac{n\pi}{2L}z_0\right) sinc\left(\frac{m\pi}{2a\theta_1}\text{Ø}_0\right) \tag{15}$$

Mutual Coupling

Mutual coupling between array elements affects the radiation pattern and input impedances. The radiation from one element in the array induces currents on the other elements to a nearby and scatters into the far field. The induced current derived a voltage at the terminals of other elements [11]. The input terminals of the elements in an array are represented as ports of a microwave network. The equivalent network of two antenna array is shown in Fig. 2. Hence, the mutual coupling is represented as a scattering matrix or S-parameters matrix as illustrated in Eq. (16).

where a_n and b_n represent the forward and reverse voltage wave amplitude at the nth port respectively. The mutual impedance formulation is shown in Eq. (17) [11].

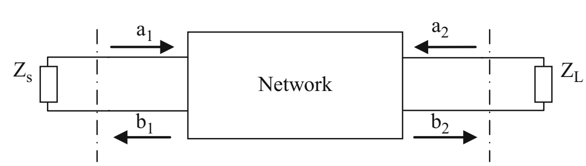


Fig. 2 Equivalent network of two antenna array

$$\begin{bmatrix} b_1 \\ b_2 \end{bmatrix} = \begin{bmatrix} S_{11} & S_{12} \\ S_{21} & S_{22} \end{bmatrix} \begin{bmatrix} a_1 \\ a_2 \end{bmatrix} \tag{16}$$

$$\begin{bmatrix} v_1 \\ v_2 \end{bmatrix} = \begin{bmatrix} Z_{11} & Z_{12} \\ Z_{21} & Z_{22} \end{bmatrix} \begin{bmatrix} i_1 \\ i_2 \end{bmatrix} \tag{17}$$

Hence, the mutual coupling coefficient, S_{12}, can be calculated as in Eq. (18) [26].

$$S_{12} = 20 \log \left(\frac{2z_{21} - z_0}{\left[(z_{11} + z_0)^2 - (z_0)^2 \right]} \right) \quad dB \tag{18}$$

where Z_0 is the characteristic impedance of the feeding coaxial cable (assumed to be 50 Ω in most of the cases).

In case of using identical array elements, same dimension and same feeding position, the values of Z_{11} and Z_{22} will give the same value and Z_{12} and Z_{21} are the same.

Hence, the value of Z_{12} and Z_{21} are given by Eq. (19).

$$Z_{12} = Z_{21} = \frac{1}{I_0} \int_a^{a+h} E_\rho(z_0, \emptyset_0, \rho) d\rho \tag{19}$$

and the value of Z_{11} and Z_{22} are given by Eq. (20).

$$Z_{11} = Z_{22} = Z_{in} \tag{20}$$

By using Eqs. (15) and (18), we can get Eq. (21) for Z_{21} as follow:

$$\begin{aligned} Z_{21} = Z_{12} = j\omega\mu h \sum_n \sum_m \frac{1}{k^2 - k_{nm}^2} \frac{\varepsilon_m \varepsilon_n}{2a\theta_1 L} \\ \times \cos^2\left(\frac{2am\pi}{L} \left(\frac{d+L}{a} \right) \right) \cos^2\left(\frac{n\pi a}{L} \left(\frac{d+L}{a} \right) \right) \\ \times sinc\left(\frac{2am\pi}{L} \left(\frac{d+L}{a} \right) \right) sinc\left(\frac{n\pi a}{L} \left(\frac{d+L}{a} \right) \right) \end{aligned} \tag{21}$$

and by substituting in Eq. (18), the mutual coupling coefficient can be calculated.

Results

The dimensions of the patch are as follows: 20 mm length, 17.5 mm width, 0.8 mm substrate height and the position of a feed point is at 5 mm from the center and parallel to the length. The dimensions of the ground plate are: 52 mm length and the 48 mm width. Those values are used as constant values in this paper for 5 GHz frequency band. The dominant transverse magnetic mode is TM_{10} when a height is less than a width h ≪ W. Three different substrate materials RT/duroid-5880 PTFE, K-6098 Teflon/Glass and Epsilam-10 ceramic-filled Teflon are used for verifying the new model. The dielectric constants for the used materials are 2.2, 2.5 and 10 respectively with a tangent loss 0.0015, 0.002 and 0.004 respectively.

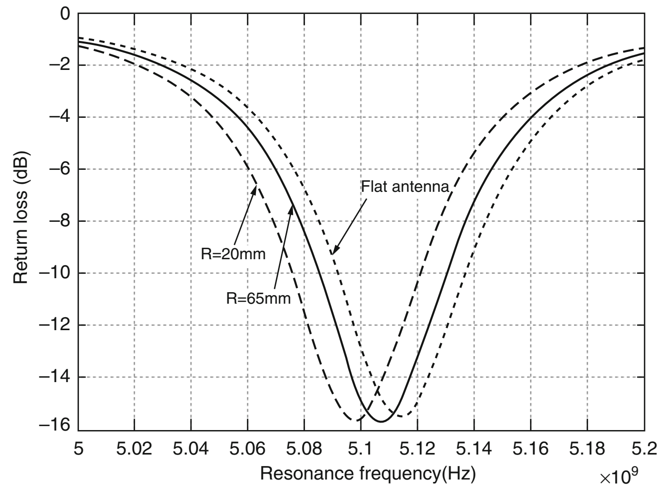


Fig. 3 Return loss (S11) as a function of frequency for different radius of curvatures

RT/Duroid-5880 PTFE Substrate

RT/duroid-5880 PTFE material is a flexible material with a dielectric constant 2.1 at low frequencies and almost 2.02 in the Giga Hertz range and tangent loss 0.0015. Return loss (S11) is illustrated in Fig. 3 [34, 35]. We obtain a return loss, −15.5 dB, at frequency 5.1 GHz for radius of curvature 20 mm, 5.18 GHz at 65 mm and 5.116 GHz for a flat antenna.

Figure 4 shows the mutual coupling coefficient, S12 as a function of resonance frequency for different radius of curvature. The maximum mutual coupling is obtained at the minimum return loss for the same resonance frequency and the peaks are shifted to the direction of increasing frequency with increasing the radius of curvature. The peaks are almost the same at −3 dB, so changing the curvature does not change the mutual coupling value but shift the curve in frequency.

K-6098 Teflon/Glass Substrate

A K-6098 Teflon/Glass material is a flexible material with a dielectric constant 2.5 at high frequency and tangent loss 0.002. Return loss (S11) is illustrated in Fig. 5. We obtain a very low return loss, −8.5 dB, at frequency 4.57 GHz for radius of curvature 20 mm, 4.578 GHz at 65 mm and

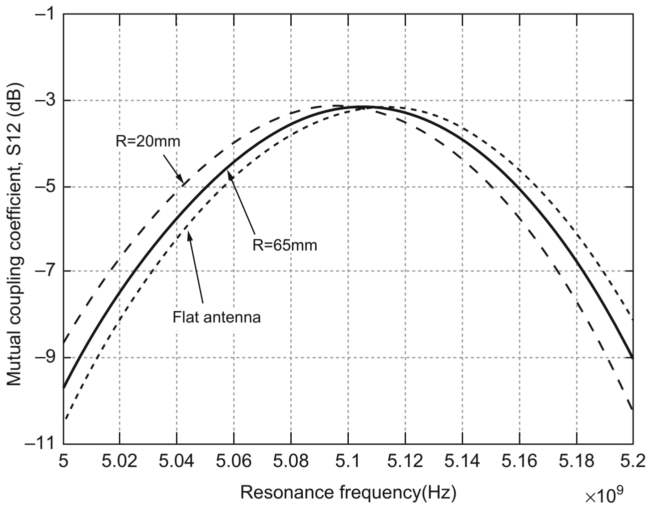


Fig. 4 Mutual coupling coefficient, S12, as a function of resonance frequency for different values of curvatures for TM_{10} mode

4.586 GHz for a flat antenna. The return loss value, −8.5 dB, is obtained for different radius of curvature.

Mutual coupling coefficient is shown in Fig. 6. The maximum mutual coupling is obtained at the minimum return loss for the same resonance frequency.

Epsilam-10 Ceramic-Filled Teflon Substrate

Epsilam-10 ceramic-filled Teflon is used as a substrate material for verifying the new model. The dielectric constant for the used material is 10 with a tangent loss 0.004. Return loss (S11) is illustrated in Fig. 7. We obtain a return loss, −50 dB for all values of radius of curvature, 20, 65 mm and flat antenna.

Figure 8 shows the mutual coupling coefficient as a function of resonance frequency for different radius of curvatures. For Epsilam-10 ceramic-filled Teflon substrate material, the flat antenna has more flat mutual coupling coefficient than the conformal antenna.

Fig. 5 Return loss (S11) as a function of frequency for different radius of curvatures

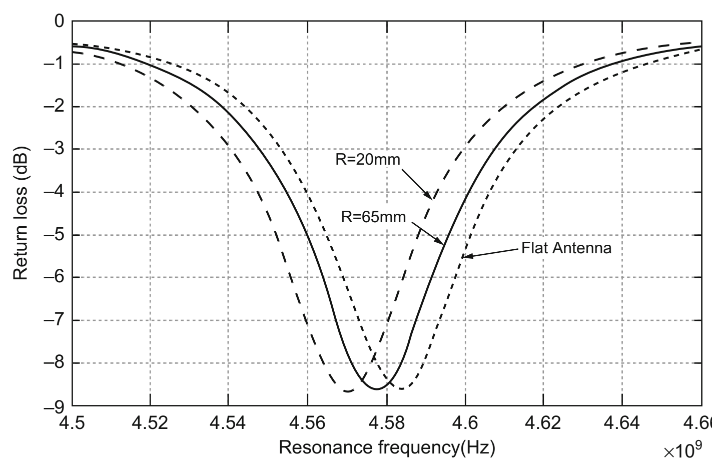
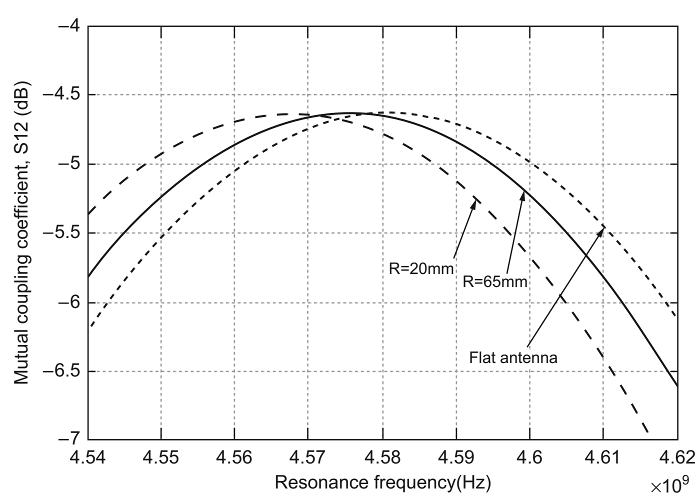


Fig. 6 Mutual coupling coefficient, S12, as a function of resonance frequency for different values of curvatures for TM_{10} mode



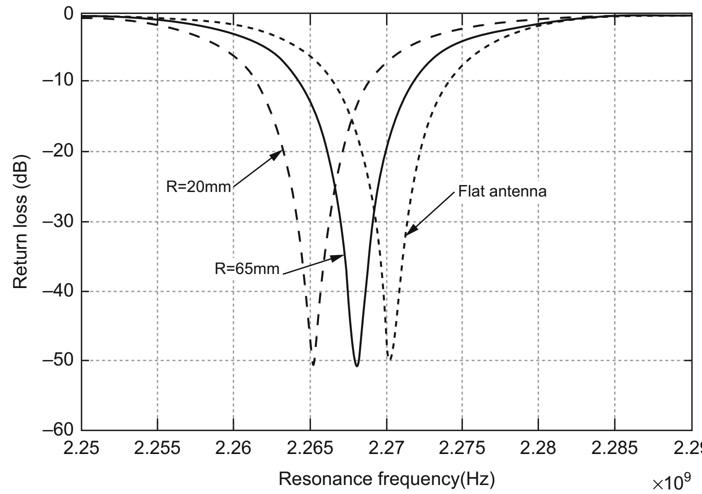


Fig. 7 Return loss (S11) as a function of frequency for different radius of curvatures

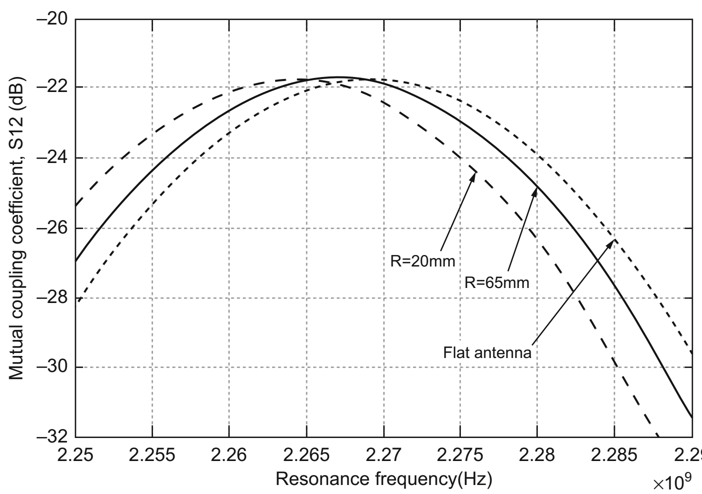


Fig. 8 Mutual coupling coefficient, S12, as a function of resonance frequency for different values of curvatures for TM_{10} mode

Conclusion

The effect of curvature on the performance of conformal microstrip antenna on cylindrical bodies for TM_{10} mode is very important. Curvature affects the fringing field and fringing field affects the antenna parameters. The equations for return loss and mutual coupling coefficient as a function of curvature and resonance frequency are derived.

By using these derived equations, we introduced the results for different dielectric conformal substrates. For the three dielectric substrates, a decreasing in the frequency due to increasing in the curvature is the trend for all materials.

We conclude that, increasing the curvature leads to decreasing the resonance frequency. The return loss peaks do not change for all substrate materials, but the mutual coupling coefficient peaks are changing according to the substrate material used.

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Variable Delay Optical Buffer Using Tunable Fiber Bragg Gratings

Islam Ashry, Ali Elrashidi, Dalia Sallam, Moustafa H. Ali, and Khaled Elleithy

Abstract

A novel all-optical variable delay buffer for next generation optical networks is reported. This buffer can store the contending packets for a relatively long time by using fiber Bragg gratings (FBGs). The proposed design is characterized by the ability to extract the delayed packets as soon as the contention is resolved.

Keywords

Contention resolution • Fiber Bragg grating • Optical buffer • Optical networks

Introduction

The all-optical networks have several advantages such as, high capacity, low insertion loss, and low price [1, 2]. However, they are not commercially used in the current networks. One of the main obstacles is packet contention which happens when two or more packets with the same wavelength are directed to the same destination at the same time. It was proven in the literature that the asynchronous (variable-sized) packet switching has a lower packet blocking probability than the synchronous (fixed-sized) packet switching [3].

Optical buffering is considered as one of the main solutions to resolve packets contention by using optical delay lines [4]. These buffers can delay the contending packets for fixed delay times by changing the lengths of the optical delay lines. Much effort has been done in the literature to design several successful architectures for the optical delay buffers [5, 6]. Though, these designs provide several advantages, most of them have several drawbacks. First, these architectures are bulky because they provide different delay lines and the delay is directly proportional to the length of each one. Second, it is hard to extract the contending packets at any time instant because the variable delay buffers have fixed delay times. Third, the overall design becomes more complicated and expensive as the number of delay lines increases because each delay line requires one switch port.

In this paper, we report a novel architecture to a variable delay optical buffer using tunable fiber Bragg gratings (FBGs). The main advantage of the proposed design over the traditional buffers is its ability to provide variable delay times. Therefore, the contending packets can be extracted at any time instant. Furthermore, using FBGs provide simplicity, low insertion loss, all-fiber geometry, and low cost [7]. To evaluate the efficiency of the reported design, we analyze four performance parameters: power loss, channel dispersion, signal-to-noise ratio (SNR), and delay times.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_17, © Springer International Publishing Switzerland 2015

Description of the Reported Buffer

Figure 1 is a schematic of the reported optical buffer designed for $N \times N$ optical switch fabric, where N represents the dimension of the fabric. This architecture consists of tunable FBGs, three-port optical circulators (OCs), control unit (CU), erbium-doped fiber amplifier (EDFA), gain equalizer, optical coupler, and switch fabric.

In front of each input port, there are two parts. The main part is a standard optical fiber (delay line) of length L_1 used to capture the contending packets. This fiber is terminated by two identical sets of M tunable cascaded FBGs, where M is the number of the channels used in the network. The tuning process of these FBGs is controlled by the CU. The second part is consisting of EDFA and gain equalizer which amplifies the confined packets when needed.

The operation of the designed buffer can be demonstrated as follows; first, the CU detects all of the input packets and specifies the contending packets, their delay lines, and the packets that can be directly served because they do not cause contention. For simplicity, let's assume the number of used wavelengths in the optical networks is four which meet the International Telecommunication Union (ITU) standardization of 100 GHz channel spacing ($\lambda_1 = 1{,}548.5$ nm, $\lambda_2 = 1{,}549.3$ nm, $\lambda_3 = 1{,}550.1$ nm, and $\lambda_4 = 1{,}550.9$ nm). In this case, four FBGs are required to be used in each set such that, their initial Bragg wavelengths are λ_1, λ_2, λ_3, and λ_4 as shown in Fig. 2.

Assume, for example, four packets of different wavelengths arrive simultaneously at the first input port and the CU decides to store the channels of wavelengths λ_1 and λ_2 because they are contending packets. While, its decision for the remaining two packets of wavelengths λ_3 and λ_4 is to serve them directly by the fabric. To perform these decisions, FBGs of Bragg wavelengths λ_3 and λ_4 in both the first and second groups should be tuned to have Bragg wavelengths of 1,550.5 and 1,551.3 nm, respectively, as shown in Fig. 3. This helps the packets of wavelengths λ_3 and λ_4 to be directed to the switch fabric. Additionally, FBGs in the first set of Bragg wavelengths λ_1 and λ_2 should be tuned respectively to have Bragg wavelengths of 1,548.9 and 1,549.7 nm, as shown in Fig. 3. These FBGs should remain in this state till the two packets of wavelengths λ_1 and λ_2 completely pass them, and then they are tuned again to their initial state to capture the contending packets in the delay line, as shown in Fig. 4.

The intensity of the confined packets attenuates due to the propagation through the FBGs and the delay line of length L_1. The CU decidestoamplifytheintensity ofthe confined packets based on the number of round trips through the delay line. When required, the FBGs in the first set of Bragg wavelengths λ_1 and λ_2 are tuned to have Bragg wavelengths of 1,548.9 and 1,549.7 nm, respectively. Consequently, the confined packets will be directed to the EDFA to be amplified. When the amplified packets come back to the delay line, the tuned FBGs should be returned back to their initial states to confine the packets again in the delay line.

After finishing the delay operation, the CU should tune the FBGs of Bragg wavelengths λ_1 and λ_2 in the second set to have respectively Bragg wavelengths of 1,548.9 and 1,549.7 nm. This makes the confined packets free and can be served by the switch fabric.

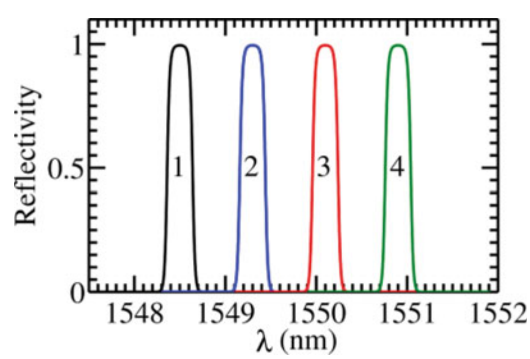

Fig. 2 Reflectivities of FBGs used in both sets 1 and 2 in the initial conditions when their Bragg wavelengths are λ_1 (*black*), λ_2 (*blue*), λ_3 (*red*), and λ_4 (*green*)

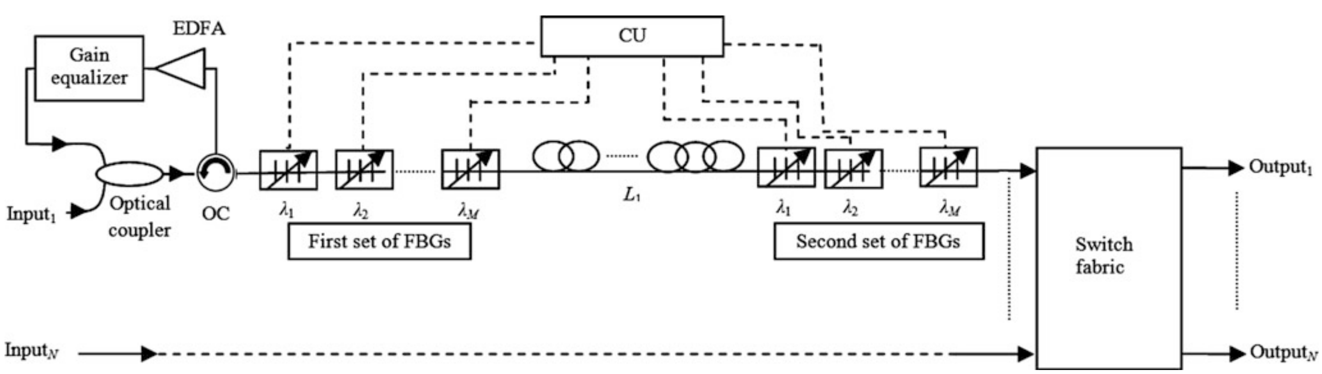

Fig. 1 Schematic of the reported architecture

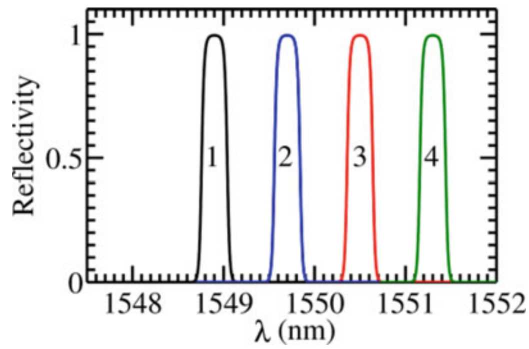

Fig. 3 Reflectivities of FBGs used in the first set when all of the packets enter the delay line

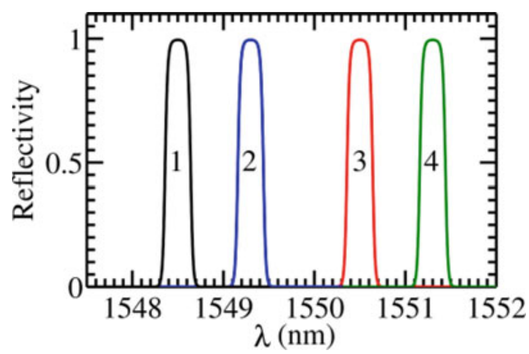

Fig. 4 Reflectivities of FBGs used in the first set when the packets of wavelengths λ_1 and λ_2 are captured by the delay line

The number of round trips that the contending packets can make in the delay line is basically depends on both the number of FBGs and the fiber length. This number is limited by the unacceptable signal attenuation threshold. When the intensity of the confined packet reaches a predefined low value, it is directed to be amplified by the EDFA. The maximum number of amplification by the EDFA is also limited by the unacceptable degradation of the signal to noise ratio (SNR).

Theoretical Performance Analysis

In order to evaluate the advantages and limits of the reported architecture, we measure the different performance parameters such as, delay line length, maximum delay time, signal dispersion, attenuation, and SNR. Assume that the maximum packet size in the optical network is Y and the used bit rate is B. The condition on the length of the delay line to make sure that the packets can completely enter the delay line is:

$$L_1 \geq \frac{cY}{2nB}, \tag{1}$$

where c denotes the speed of light and n is the core refractive index of optical fiber used as a delay line.

The maximum delay time $T_{\max}$ can be represented as follows:

$$T_{\max} = \frac{2nKQL_1}{c} + \frac{nQL_2}{c}, \tag{2}$$

where K is the maximum allowed number of round trips in the buffer before the intensity of a delayed packets falls by attenuation to unaccepted threshold. Q denotes the maximum allowed number of amplifications, and L_2 is the EDFA length.

The packets directed to the switch fabric without delay suffer from maximum attenuation A_1 of:

$$A_1 = A_{OC} + 2MA_{FBG} + \alpha L_1, \tag{3}$$

where A_{OC} and A_{FBG} are the attenuation induced by the OC and FBG, respectively. In this design, it is assumed that the attenuation of FBGs is due to the out of band transmission only. This is because the maximum reflectivity at the Bragg wavelength is high enough to neglect the losses during reflection. Also, it is assumed that the optical coupler has negligible losses. The attenuation per unit length in the delay line is denoted by α.

The maximum allowed attenuation A_2 of a delayed packet before the decision of amplifying it by the EDFA is:

$$A_2 = 2A_{OC} + 2K\alpha L_1 + (M + KM - K + 1)A_{FBG}. \tag{4}$$

Assume that the dispersion parameter of the delay line is D_f and the dispersion resulted from using OC is D_{OC}, the maximum allowed dispersion $D_{\max}$ for a confined packet is:

$$D_{\max} = 2KQL_1 D_f \Delta\lambda + (2Q + 1)D_{OC}, \tag{5}$$

where $\Delta\lambda$ is the spectral width of the light source used in the network.

To accurately evaluate the performance of the reported design, we select some commercial optical components with the following characteristics. The bit rate, maximum packet size, light source spectral width, and number of wavelengths used in the optical network are 40 Gbps, 1,500 byte, 0.4 nm, and 16, respectively [1]. The used OCs are symmetric between their ports and have respectively 0.5 dB and 0.1 ps insertion loss and dispersion [8]. FBGs have 0.3 nm reflection bandwidth, 99.5 % maximum reflectivity, and apodized using Blackman profile to suppress their side lobes [9]. FBG can be written on lithium niobate optical fiber so that, it can be tuned by applying electric field parallel to its axis [10]. This tuning method has a fast tuning speed in order of nm/ns which fits the requirements of the optical networks. The loss of FBG resulted from out of band transmission is 0.05 dB [11]. Furthermore, assume that the maximum allowed attenuation before EDFA amplification is 30 dB, in other words, the EDFA gain in 30 dB, the EDFA length is 30 m, and the gain equalizer offers flattening with peak-to-peak variation of 1 dB [3]. Finally, the optical fiber of the delay line has 0.2 dB/km attenuation loss and 0.1 ps/(nm.km) dispersion parameter [3].

Using Eq. (1), the minimum allowed delay line length is 30.4 m. Its maximum value can be calculated by using Eq. (3) such that, A_1 should not exceed 30 dB. The maximum delay length is found to be 139.5 km. Therefore, let's choose $L_1 = 31$ m to get the packet out of the delay line as soon as the CU declares this. The maximum allowed number of round trips can be obtained by substituting with $A_2 = 30$ dB in Eq. (4) to find K equals 36 round trips. Since the EDFA gain equalizer provides 1 dB peak-to-peak variation, the maximum number of EDFA amplifications that gives acceptable level of SNR is around ten times [3]. Substituting of these values in Eq. (2), the maximum delay time is found to be 116 µs. Finally, Eq. (5) shows the maximum, dispersion of this architecture is 3 ps.

Conclusion

We reported a novel all-optical variable delay buffer based on tunable FBGs. This architecture can be used to delay a large number of contending packets, up to $N \times M$, simultaneously. For an optical network of 16 channels, the reported buffer can store the contending packets up to 116 µs with relatively low dispersion. Finally, this architecture is characterized by its simplicity, low price, and the opportunity to extract the contending packets whenever the CU declares this decision.

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An Exploratory Case Study of Offshore Outsourcing: Problems in Multicultural Settings

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Abstract

Outsourcing means using factors which are external to an organization to perform its functions whereas offshore outsourcing relies on using foreign resources to perform those functions. We focus in this paper on offshore outsourcing of information systems (ISs). We have conducted an exploratory case study of outsourcing IS support services. The services have been outsourced by a supplier organization which has been based in Finland. Main parts of its customer support services have been allocated to so-called production centers, which have been located in three countries; Latvia, India, and the Philippines. We have gathered qualitative data from nine experts via theme-based interviews. The empirical data gathering has focused on revealing problems in the outsourcing practices. There were 74 mentions of different kinds of confronted problems. The differences between the studied countries in terms of the typical outsourcing problems have been noted and the results compared to the most important related works. The presented results on the studied thematic issues can be useful in their part for better understanding typical problems related to initiating and organizing similar IS-related offshore outsourcing activities.

Keywords

Software engineering • Case study • Offshore outsourcing • Theme-based interviews

Introduction

According to Lacity and Hirschheim [1] *outsourcing* means using factors which are external to an organization to perform its functions. According to Lacity and Willcocks [2] *offshore outsourcing* ("offshoring") relies on using foreign resources. Due to globalization there are increasingly better possibilities to apply offshore outsourcing. Due to tightening price competition on free markets there is also a constant pressure to apply it. It has typically been applied to gain bigger cost savings and to reach experts whose skills meet better the requirements set to the business or whose demands on compensations and working conditions are less ambitious than in case of domestic labor markets in the Western world.

This paper presents a case study of *information system customer support services* which have been outsourced offshore into three countries. We focus on studying empirically the situation in which the supplier utilizes foreign production facilities to offer the outsourced services to the customers. The paper is organized as follows. Section "Background" introduces the background; including outsourcing, offshore outsourcing and other relevant main concepts. Section "Empirical case study" describes the context of our empirical study; including descriptions of the selected themes and the applied research method. Section "Results" presents summarized results; related to the identified problems in the outsourcing practice based on

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_18, © Springer International Publishing Switzerland 2015

data as received from interviewed experts and organized based on the studied themes. Section "Discussion" discusses the main results and some of the most related studies and Section "Conclusion" concludes the paper. Appendices provide details of the gathered data.

Background

Outsourcing

Outsourcing has a long history but it has been a widely used strategy for gaining competitive advantage for only a short period [3]. Outsourcing as a concept refers to using of external agents to perform an organizational activity [4]. Outsourcing has considerable relevance for several areas; such as managing information technology (IT); e.g. [5] and business processes; e.g. [4]. IT and information systems (ISs) are both enablers of outsourcing and a major target for outsourcing.

Due to the expensiveness of IT work in the Western world, a lot of it has been outsourced to the areas of lower labor costs. Therefore, it is not a surprise that outsourcing has interested researchers regarding both software development; e.g. [6–9] and software maintenance; e.g. [10–12]. An extensive survey on IS outsourcing has been provided by Dibbern et al. [13]. It has been noted that it is no longer a question of *whether to outsource* but rather of *what and how to outsource* [14]. According to [13] critical questions include: (1) *what to outsource* (business processes vs. IT functions), (2) *where to outsource* (onshore vs. offshore), and (3) *what should be the duration of the outsourcing contract*. In respect to our study the second one of these questions (onshore vs. offshore) comes closest since we are interested in knowing whether there are problems and what is the nature of typical problems in case of offshore outsourcing.

Offshore Outsourcing

Briefly, offshore outsourcing or offshoring refers to overseas outsourcing as an opposite to domestic outsourcing ("onshore outsourcing" or "onshoring"). When onshore outsourcing and offshore outsourcing are compared, it seems that offshore outsourcing tends to be more favorable in a short run, but in a longer run; as the contract duration is long, offshore outsourcing can induce more problems.

Furthermore, with offshore outsourcing companies have to overcome barriers like cultural and language differences [14, 15]. Previous research includes conceptual frameworks [16, 17], conceptual considerations [18–21], and literature reviews; e.g. [9]. Compared to the importance of the issue, empirical studies are relatively rare. Examples of earlier studies on different types of offshore outsourcing include software development [8, 22], software maintenance [23], IT work [24] and IT services [25–27]. We will here take as examples two important empirical studies; Khan et al. [28] and Winkler et al. [8] since they especially contribute to understanding the nature of the problems which may be caused by cultural differences.

Khan et al. [28] reported findings from an interview study of offshore outsourcing in India. They considered the benefits and risks of, as well as strategies for, offshore outsourcing. They concluded that for many companies (i.e. IT suppliers in India) engaging in offshore outsourcing tended to be a low risk and low value business. However, intentions towards businesses of a higher risk and higher income were also observed. Surprisingly, geographical distance or cultural differences were not mentioned among the major risks. Winkler et al. [8] reported cultural differences in German-Indian outsourcing relationships. According to their research framework the quality of the offshoring relationships correlates with offshoring success. The results reveal that cultural differences during the offshore cooperation are rather common, since all of the studied companies had observed such differences to at least some extent. The researchers classified the differences into "grand themes" which were (1) the Indian professionals have difficulty in saying "no", (2) Indian and German IS professionals pursue different development approaches; and (3) Indian and German IS employees differ in criticism behaviour. All of these differences can be explained, at least partially, by *power distance* which is much longer in India as compared to what it is in Germany. Power distance implies a strong social hierarchy. Other explanations include; e.g., differences in values and activity vs. passivity and other studies of outsourcing related to India include [26, 29].

Empirical Case Study

Goal and Focus of the Study

We have conducted an exploratory case study of offshore outsourcing. The case study has focused on the problems of outsourcing information system customer support services offshore. Four themes were used to group the problems: (1) Outsourcing relationship, (2) distributed work, (3) scheduling, and (4) cultural differences. Nine experts from four countries served as data sources. Data were gathered by interviewing them and asking questions related to the selected themes.

Our case can also be characterized as focusing on the *system maintenance phase* of the *outsourced services* from the perspective of the *supplier organization*. That is, we studied the views of the service providers related to supported systems which were matured in a sense that they had evolved to their maintenance phase [10, 11, 30]. Maintenance is typically inevitable due to the changing user requirements, and changes in the technical environments. In this study our focus is on the level of managing the process of providing the service.

Research Method

This is a *qualitative case study*. Qualitative research aims at describing complex phenomena from multiple perspectives in a holistic sense. These phenomena are typically rich in terms of having lots of internal and external relations. The data are also preferably gathered from genuine and natural settings. Qualitative research aims at *revealing* facts rather than proving some preset assertions. In that sense it is an ideal way to investigate weakly structured issues of which there is not yet a clear understanding, to reveal otherwise hidden aspects, and to identify sensitive information. It favors humans, observations, and discussions as means to gather information rather than using measuring devices.

Interviews enable flexibility in the qualitative data gathering, regarding; e.g., the order of the questions, depth of the discussion regarding specific issues, allowing discussion of the emergent issues etc. Interviews also provide more versatile feedback than; e.g., questionnaires, which feedback can be used to direct the interviews accordingly. We apply so-called *theme-based interviews* in this study. When applying theme-based interviews, the themes of the discussion are preset, whereas the exact form of the questions and their order are not. We have selected this approach since we had a prior general understanding of the problems in the area in development projects, whereas there was only a relatively little amount of scientific knowledge on the intended specific focus area; i.e. services in the maintenance phase of offshore outsourcing.

Studied Organizations and Services

We have studied three different kinds of services which have been outsourced offshore. Those services were supplied by one company, which acquired the services from abroad to its customers, which in turn were all located in Finland. The supplier and the customer were large, globally operating Finnish firms. The services were provided by three so-called *production centers*. These centers were located in Latvia, India, and the Philippines (Table 1). The target systems were important to the business of their customers and they were specifically tailored to meet their requirements. Their maintenance had a crucial role. The end users of those systems were either employees of the customer organizations or their prospective customers. Our focus in this study is not on the details of the systems but on the problems in the process of providing service for the customers to deal with the systems.

Table 1 The employees of the involved organizations related to the services

Location of the service	Onshore persons (the supplier in Finland)	Offshore persons (the production centers)
Latvia	2 persons	12 persons
India	3 persons	15 persons
Philippines	3 persons	20 persons

Table 2 The interviewed employees

Location of the service	Onshore persons (the supplier in Finland)	Offshore persons (the production centers)
Latvia	Finnish coordinator (iL1) Finnish chief (iL2)	Latvian group head (iL3)
India	Finnish coordinator (iI1) Indian chief (iI2)	Indian group member (iI3)
Philippines	Finnish group head (iP1) Philippine chief (iP2)	Philippine group head (iP3)

The acronyms iL1, iL2, iL3, iI1, iI2, iI3, iP1, iP2, iP3 refer to the individual persons that were interviewed and will be used in the appendices

The prefix 'i' stands for interviewee, 'L' for Latvia, 'I' for India, 'P' for the Philippines

Interviewed Persons and Their Tasks

We interviewed nine persons—three persons related to each of the services. Six of the interviewed persons worked in Finland in the supplier organization, and three offshore in the production centers. This choice was due to the focus on the perspective of the supplier. The interviewed persons were selected in a such way that related to the service of each production center one served as a chief, another as a practical coordinator or a group head of the delivery of the provided service, and the third as an offshore group head or a group member (Table 2).

These choices were due to the aspiration to get views on different kinds of problems by interviewing persons in different levels of the organizations. The persons within the production centers were locals, i.e. they had the nationality of that country, whereas the persons in the supplier organization (i.e. in Finland) had varying nationalities. The persons in Finland intermediated between the customers and the production centers. *Chiefs* were in charge of making deals, cost surveillance, and general organizing of the activities. *Coordinators* and *group heads* who worked in Finland were in charge of the continuity of the daily action and taking care of reporting, solving problematic situations, and acquiring necessary clearances to the persons working in the production centers. Group heads who were working in the production centers, were in charge of the allocation of the technical work to the work group and of the quality control. The other *group members* did the actual technical service work.

Interviews

The general framework of the theme-based interview was delivered to the interviewees prior to the interview in order to enable them to prepare well to it by considering the themes beforehand. The actual interviews were preceded by a test interview. The Latvian group head who worked in the production center was interviewed during his short visit in Finland. The Indian group member who worked in the production center was interviewed during his long work-task in Finland. The Philippine group head who worked in the production center was interviewed via phone. All other interviewees were located in Finland and were interviewed accordingly. The interviews spanned between 60 and 90 min each. They were all recorded and the central contents were transcribed into textual format.

Results

We will report the identified problems related to the earlier described four themes. Each interviewee provided information regarding the problems which he or she had confronted related to the target system. This section is based purely on the views of the interviewees. It should also be noted that the views of the interviewees are not intended to be generalizations to the discussed nationalities but instead they characterize the work, work tasks, and working environment of the involved persons in the context of the studied services as described earlier. We will provide a succinct summary of the results here in the main text and provide the details of the data in the Appendices A–D, partly due to the posed length limitations of the paper.

Outsourcing Relationship

There were many revealed problems related both to the Latvian and the Indian services, whereas Philippines was an exception in this regard (Appendix A). Most of the revealed problems in all three services concerned the relation between the supplier and the customers. Related to the Indian service the customers especially seemed to have a tendency to have little trust on employees and expect more than could be delivered. Similar problems occurred also as related to the Latvian service but in a much milder form. Both the Indian and the Latvian service suffered also on inadequate control over the information flow between the supplier and the customers.

Distributed Work

Problems concerning distribution related especially to communication and division of work (Appendix B). The main problem related to the Latvian service was work control, whereas in the Indian service the main problems were related to the lack of clear instructions and communication. The Philippine employees revealed most problems, which were varied, concerning especially division of work, task descriptions, and too limited rights to use relevant information systems in some cases. Distributed work was also the clearly most problematic aspect in the Philippine case.

Scheduling

Scheduling is potentially complicated by time zone differences (Appendix C). The problems with the Latvian service were minor and mostly related to exceptions. Most of the problems related to this theme concerned the Indian production center. For that center the identified problems were numerous; including e.g. difficulties related to finding common time, vacations, lunch breaks, and use of substitutes with limited resources and knowledge. The problems in case of Philippines were much less severe but were similarly related to allocating shared time, and allocating vacations.

Cultural Differences

The last studied theme was cultural differences (Appendix D), which may be somewhat sensitive issues. However, most of the problems as identified by the involved experts in this case study clearly related to this category. Therefore, it is important to observe the involved difficulties. The problems in the Latvian case related especially to different roles of the technical and non-technical experts, mismatched expectations concerning proper abstraction levels of technical descriptions, and limited cultural traditions for service-oriented business. Although cultural differences was the most problematic aspect in this case and some of these problems were due to national historical background, some were simply due to the different educational backgrounds of the involved subject groups.

Overall, the problems with the Indian and the Finnish cultural differences was the most prominent individual category. The most conspicuous problems were related to the Indian employees' reserved use of telephone in communication, and limited capabilities related to assessing risks and required effort, meeting the deadlines, and following the approved plans. These were the impressions received from the involved experts and are obviously not intended as generalizations.

Table 3 Frequencies of the mentions of the particular problems related to the studied themes

Involved countries	Latvia 32.4 %		India 47.3%		The Philippines 20.3 %		Total 100.0 %	
Main themes	N	%	N	%	N	%	N	%
(1) Outsourcing relation	8	33.3	9	25.7	1	6.7	18	24.3
(2) Distributed work	4	16.7	3	8.6	8	53.3	15	20.3
(3) Scheduling	2	8.3	10	28.6	2	13.3	14	18.9
(4) Cultural differences	10	41.7	13	37.1	4	26.7	27	36.5
Sum	24	100.0	35	100.0	15	100.0	74	100.0

The main problem related to the Philippine service was that the Philippine employees were clearly afraid of "loosing their faces"; i.e. of being publicly criticized or judged. This and the related phenomena tended easily to lead to a general cover up of actual problems; which could obviously later lead to more serious problems.

Summary of the Problem Types

The frequencies of the identified problems as taken separately for the studied four main themes and three countries in which the offshored services were produced are gathered into Table 3. The main general-level observations are as follows. Overall, cultural differences between the Finnish and the Indian employees was the area which was deemed as the most problematic one, whereas least problems were identified related to the general outsourcing relationship between the Finnish and the Philippine employees.

Discussion

Our case study revealed numerous problems in respect to the relationships of offshoring the customer support services in India, Latvia and the Philippines. It is interesting to note that there is not much earlier empirically validated data on offshore outsourcing of IS services. In this respect, our findings provide a useful addition to the current knowledge. Several aspects of cultural differences (theme 4) have interested IS researchers in general, but so far the results in the context of offshore outsourcing have been either conflicting or inadequate. E.g., Winkler et al. [8] found out that cultural differences are very common and that they may affect the quality of offshoring very much, whereas Khan et al. [28] did not identify cultural differences as any kind of major risk.

In our study, problems related to cultural differences formed the largest category in terms of their frequency. Although problems were revealed in the cases of all the three countries, the Indian case; especially, brought out numerous problems. Most of the problems with the Indian case were related to either managing risks or interpersonal behaviour. With the case of Philippines the major problem was that the Philippines were very afraid of being publicly criticized or judged. We assume that all these problems relate to at least partially to cultural differences in terms of the power distance; which is probably greatest in the Far East and lowest in the USA and in the Northern Europe. This finding is in line; for example, with the finding of Winkler et al. [8].

The Latvian case revealed fewer problems that could be connected to the cultural differences. The major problem with the Latvians was their relatively low proactivity. In general, the Latvian thinking and working seemed to be very similar to the Finnish counterpart. Proper solutions to the cultural problems are probably mainly educational: learning about the other cultures, laws, morals, customs and societal habits. Similar kinds of suggestions have been given; e.g. in [21].

Problems related to the other themes (1–3) were fewer and in average also milder. In respect to the outsourcing relationship (theme 1), the two main problems seemed to be (A) the lack of the customer's trust on the service producer and (B) too high expectations from the customer's side. These problems were identified only related to the Indian and the Latvian cases, and in the latter one they were clearly milder.

It seems that these are problems that can be linked with the cultural problems discussed above, since the lack of trust; for example, can at least partially result from insufficient knowledge about the offshore partner's culture. These problems cannot be resolved easily but the potential resolution requires extended continuous efforts. Solutions to these problems include increasing transparency of the operations giving assurances of the continuity of the service improving the communication between the customers and the supplier.

The study also revealed several kinds of problems induced by distributed work (theme 2). These included problems with work control (the Latvian case), instructions and communication (the Indian case) as well as division of work, task descriptions and lack of necessary facilities (the Philippine case). All of these problems are rather typical to distributed work and cannot be interpreted as characteristics of the offshoring context as such. The solutions might include; e.g., better planning and guidance, better instructions and documents, and improving interpersonal skills and use of modern information technology.

Apparently, improved communication, clearer guidelines and transparent processes are among potential solutions.

Most of the scheduling problems (theme 3) appeared via the Indian case. In Latvian and Philippine cases these problems were lesser and minor. The major scheduling problems related to issues such as finding common time, vacations, lunch breaks, and use of substitutes with limited resources and knowledge. Some of these issues are very much context dependent and difficult to affect. Therefore, one of the best ways of resolving these kinds of problems is probably by improving coordination systems. Like some of the earlier research; e.g. [21] have suggested, time differences can be either an obstacle or an advantage for a successful teamwork. The net effect depends very much on the available coordination capabilities.

Conclusion

We have described in this paper an exploratory case study of offshore outsourcing of information system (IS) services. The supplier organization of the services was based in Finland. The studied production centers were located in three countries; Latvia, India, and the Philippines. Expert data were gathered via semi-structured interviews from nine experts. Four themes were selected related to which problems were investigated.

The experts provided their views on the issues which have been deemed as problematic related to the practice of the offshore outsourcing and the related organization of the provided services. The details of the characteristics of the typical problems in case of the different themes and countries were provided. All the 74 mentions of the problems have been reported. Cultural differences was the largest source of problems in terms of the numbers of the problems. The Indian case was clearly the most problematic one. Obviously, that was at least partly due to the relatively large cultural differences between the involved Finnish and Indian employees.

These results are obviously not intended to be generalizations. The presented observations and knowledge on the identified problems can however be expected to be useful especially for those who are in charge of making decisions regarding the organization of the offshore outsourcing of IS-related activities. Paying sufficient attention to potential problem areas may help in avoiding the most probable complications. Possible important further research directions in the general research area concern conducting empirical studies on the effects of the cultural differences to the typically confronted problems, possible solutions to the most frequent problems, and general success factors of conducting smoothly proceeding offshore outsourcing ventures.

Appendices

The appendices provide the details of the gathered data related to the studied four themes. Due to the posed length limitations the actual transcribed citations cannot be reported here. Instead, they have been digested to a more compact form such that publishing is possible; by using the extra pages option of the target conference.

The prefixes ‘p’ and ‘i’ stand for problem and interviewee respectively in the identifiers to be used. The running numbers differentiate the identified items from each other within each theme and country. ‘O’ stands for Outsourcing, ‘D’ for Distributed work, ‘T’ for Scheduling and Timing, and ‘C’ for Cultural differences. ‘L’ stands for Latvia, ‘I’ for India, and ‘P’ for the Philippines.

Appendix A Outsourcing Relationship

Latvia

Overall, the outsourcing relation worked relatively well in the case of Latvia according to the Finnish coordinator, who regarded the customer relation to be very intimate and spontaneous (iL1). His view was that the relation was not like a normal customer-consultant relation, but a more profound one. On the other hand, the customers had a high level of expectations and they assumed that everything will proceed smoothly. Consequently, the coordinator regarded building trust as the most significant challenge in the beginning of the service (pO-L1,iL1). The Finnish chief noticed that in the starting phase of the service the customer had a feeling of not having enough control (pO-L2,iL2). This was because the customer did not yet know the persons involved in the production center’s service group and was not aware of their working habits.

Another potential problem was that the customer was receiving contradictory messages regarding the fluency of the service (pO-L3,iL2). Occasionally persons who were not communicating face-to-face revealed more readily concerns which were not relevant (pO-L4,iL2). E.g, the problems which were internal to the production center’s service group were necessarily not worth mentioning to the customers. Since the customer was aware of the relatively low expenditures of the offshore production, the customer could impose price pressures to the supplier (pO-L5,iL1). Especially, it occasionally was so that the customer expected and demanded much from the persons who were not being offshored and who thereby were relatively expensive and scarce resources.

One of the problems was the relatively high change rate of the employees (pO-L6,iL1); especially, changing of key-persons tended to cause the customers to have concerns and to raise questions. Nevertheless, overall, the Latvian group head (iL3) regarded the outsourcing relation as being excellent. This stance was positively affected by the fact that the customers' representatives had visited the Latvian production center multiple times. The main problem of the Latvian employees was that they did not always have sufficient clearances and authorizations to work properly (pO-L7,iL3).

The Latvian employees did not regard the customer as being in an important role. Instead, they acknowledged the importance of the end users. Since the end-users were dispersed, the Latvians considered the location from where the service was provided as being inconsequential. The Latvian group head (iL3) neither had noticed any major problems between the involved Finnish persons and the customers. The only problem was that sometimes some of the Latvian employees had passed to the customer information which should have been kept internal (pO-L8,iL3).

India

The problems which were confronted related to the Indian production center included the following: Customers' suspicions regarding whether they were given a truthful description of the production (pO-I1,iI1), customers' prejudices about Indian work having a lower quality than Finnish work (pO-I2,iI2) and that the service which was produced offshore required overall increased time for managing the customer relations (pO-I3,iI2).

Difficulties in knowing what the customers' specific intentions were may have related to the fact that the customers sometimes tried; in this context, to obtain information primarily only in order to compare the supplier to other Asian information technology suppliers (pO-I4,iI1). Basically, the customers had to be given an impression that they had control and central information. This may have; however, also inadvertently affected the situations in which the customer attempted to exceed its authority. E.g., sometimes the supplier did get an impression that the customer had not read the formal contract sufficiently meticulously (pO-I5,iI1).

This phenomenon manifested itself; for example, when the customer showed interest in the supplier's internal matters; such as numbers of the involved persons and their names (pO-I6,iI1,iI2), even though that the supplier had only made a contract that quaranteed that the problems, which the customer confronted (related to the used information system) were solved within the predefined time. A related problem was that the customers sometimes wondered why they were paying for a service also when there were no open; i.e. currently unresolved, defects (pO-I7,iI1). This was inconsistent behaviour since the customer had earlier agreed on the criteria for paying. The paying was expected to be based on the quaranteed availability of the service; instead of being based on the number of the completed given tasks.

The supplier also had much implicit work to be completed related to developing the service. The customer had a tendency of attempting to have also such tasks performed by the supplier which were not part of a contract; even without extra compensation (pO-I8,iI1). The customer also easily did get used to having such extra service. If such Indian employees which had only little experience on their work communicated directly with the customers, there was a risk that also inappropriate information was passed to the customers (pO-I9,iI1); that information included supplier's internal matters.

Philippines

Overall, there were not many problems related to the outsourcing relation as such for this case. All the interviewees related to the Philippine production center (iP1,iP2,iP3) regarded the customer relation as being excellent. For example, the Finnish group head (iP1) regarded the offshore production easy from the view-point of customer relations; the attitude was that the customer relation always had to be formed regardless; e.g., of the locales of the involved persons. One minor problem was that some employees in the production center may have incorrectly assumed that the Finnish persons working for the supplier were customers (pO-P1,iP1).

The Philippine chief (iP2) told that it was typical that the customers were very interested in who were involved in the service and who was doing what in the production center. That attitude and interest was natural since the offshore outsourcing was then still a relatively new model of production (iP2).

Appendix B Distributed Work

Latvia

The primary problem in the distributed work was that even though the formal responsibility of completing the work was in Finland, the Finnish persons did not have full control over those who were doing the actual work (pD-L1,iL1). Controlling the workers too much is obviously not good for their working morals. However, in some cases it was impossible to avoid strict control to ensure progress of the work (pD-L2,iL1). Distributed work was complicated in cases when the offshore group had not been provided with a sufficient clarification of its task (pD-L3,iL2).

Weekly meetings were often considered as a burden and the involved persons could not utilize them in full extent (pD-L4,iL2). However, the Latvian group head (iL3) did not think that there would be any specific problems due to the applied distributed production model, since the end users were in any case dispersed throughout the world.

India

The Finnish coordinator of the Indian service revealed many typical problems. Lack of exact and clear instructions was a central problem (pD-I1,iI1) for correctly performing work tasks. The Finnish coordinator noticed that it is necessary that more work can be constantly transfered to be performed by the Indian employees for that he himself could instead meet new challenges (pD-I2,sD-I4,iI1). The fundamental problem and risk of distributed projects was that communication was complicated (pD-I3,iI2). The Indian group member (iI3); however, did not identify any problems; instead he considered the production center to be able to fulfill all the set requirements. The Indian group members had only very little interaction with the Finnish persons and no knowledge of the future plans of the offshore operation (iI3).

Philippines

The Finnish group head stated that a natural problem related to the distributed work was that non-local persons could not be met (pD-P1,iP1). Consequently, it was harder to recognize problems in the distributed case (pD-P2,iP1). In that case more interpretation of the messages was required to be potentially able to determine whether there were real problems. Customers had sometimes given negative feedback regarding the English language skills of the Philippine workers, but they were nevertheless very good as compared to the skills of the Indian employees in similar tasks (iP1).

One problem was that the persons of the supplier often considered the persons of the production center merely as tools and acted accordingly (pD-P3,iP2). Delocalization introduced some problems which were not met at all in Finland; these included massive natural catastrophes; such as typhoons and earthquakes (pD-P4,iP1). One of the problems for the Philippine group head was that in some cases it was not clear who had the responsibility over specific tasks (pD-P5,iP3). This was a problem especially in the beginning of the service. The production team had limited rights to some of the related information systems because of the data security rules set by the customer (pD-P6,iP3) and consequently; in these cases, the production group had to rely on the help from the Finnish workers. That was problematic since the involved Finnish experts worked only 8 h a day, whereas the Philippine production center operated non-stop in three shifts (pD-P7,iP3). The geographical distribution caused also problems related to the communication channels and performance of the technical systems due to the long distance between the involved servers (pD-P8,iP3).

Appendix C Scheduling

Latvia

The Latvian and Finnish times zones are the same. Therefore, there obviously were no problems due to them. The exact working hours and days had been defined in the contract. Generally, Latvian employees tended to come to the working place later and also to leave from there later than the Finns. This did not cause problems. In matter of fact, this phenomenon has been beneficial since in this way the daily service could be extended. Even though that the contract stated that the service ends at 17.00 o'clock it was common that there was at least one person available until 19.00 o'clock (iL1). On the other hand, during the early morning hours there were not necessarily many workers present (pT-L1,iL2).

Unlike the Finns, the Latvians had their vacations in smaller pieces throughout the year. The Finnish coordinator (iL1) did not consider the Latvian timings of vacations as a problem. According to the Latvian group head (iL3) the national festivals caused problems since after returning to work there may have been 3-4 times more service requests to be handled (pT-L2,iL3). Some of the end users had gotten used to service within a day and the relatively slow service after the national festivals caused some wonder among them.

India

The Indian time zone is +2,5 h as compared to Finland during summer and +3,5 h in other seasons. Therefore, it was sometimes difficult to find common time during the working hours (pT-I1,iI1) for the Finns and the Indians. This problem was increased by the different timings for the lunch breaks (pT-I2,iI1). The Indian chief (iI2), however, regarded timing easy in practice, because the contract set the daily times for the service and there was nevertheless about 5 h common time. Also the vacations were regarded mainly as unproblematic (iI2). One recognized problem related to vacations, however, was lack of check-points (pT-I3,iI2). The Indian group member (iI3) regarded the differing timings for vacations problematic (pT-I4,iI3). The Finnish long summer break caused troubles to the Indian workers.

In those times they did not necessarily know with whom to communicate and how to obtain the needed information (pT-I5,iI3).

Even though that the Finnish persons did have named substitutes, they did not necessarily know all their tasks (pT-I6,iI3) and they did not have access to all critical emails during the previous 11 months (pT-I7,iI3). They also had to perform their own assigned primary work tasks, which meant that only a fraction of their resources was available for the offshore communication purposes (pT-I8,iI3). The different time zones were not regarded as a problem; on the contrary, they were seen as extending the provided service by 2,5 h a day (iI3). A problem for the Finnish coordinator was that the vacations of the Indian employees caused extra work (pT-I9,iI1). Indians do not have long vacations, but they do have several separate public free days and optional vacations. The public free days vary according to the religious groups. This situation called for caution in order to avoid pressing anyone to work in such days (pT-I10,iI1).

Philippines

The Philippine production center had the largest difference to the Finnish time; 6 h. Unlike the other services, that service produced by the center was provided 24 h a day in three shifts. The Philippine group head mentioned; as a former problem, obscurities in service during vacations (pT-P1,iP3). Another problem was that Finns worked only 8 h a day, whereas the Philippines worked 12 h. If the Philippines needed urgent help from the Finns; outside their formal working hours, problems could occur (pT-P2, iP3). The problems related to this main aspect were small.

Appendix D Cultural Differences

Latvia

According to the Finnish chief (iL2), there were very few identified cultural differences between Finland and Latvia. The involved Finns and Latvians had similar cultural mentalities (iL2). However, due to the former Soviet influence, the Latvian business culture was generally not optimally inclined to service (pC-L1,iL2); Latvian employees could sometimes be a bit blunt. Lack of proactivity was sometimes viewed as a weakness of some of the Latvians (pC-L2,iL1). For some it was necessary to define precisely what ought to be done, and when and what was expected. Some of the employees brought forth boldly new solutions, whereas others implemented only what had been specifically requested.

Sometimes, the Latvian employees did give a too positive view of the progress of their work (pC-L3,iL1). Consequently, detailed surveillance was necessary (pC-L4,iL1). Sometimes, some of the Latvians were too timid to tell about the problems (pC-L5,iL1). None of the group members in the production center neither had the best possible English skills (pC-L6,iL2). E.g., they were not always able to make proper summaries. According to the Latvian group head (iL3) there were no actual cultural differences between the Latvians and the Finns, but the organizations differed from each other.

Latvians were in this context usually technical developers, whereas Finns were consultants, coordinators, or chiefs. That phenomenon sometimes made understanding each other hard (pC-L7,iL3). For example, when the Finns asked in a teleconference whether some task would be dealt with, the Latvians started to describe how the related problem can be solved by using many technical details in their descriptions (pC-L8,iL3). That is, the Latvians immersed in this case more into technical language and did not abstract conceptually as much due to their more technically oriented background. The Finns had to tell them precisely what kinds of answers were expected (pC-L9,iL3). That problem, however, did not mean that the Latvians would give too much technical details, but the Finns sometimes did not give them enough (pC-L10,IL3). Nevertheless, overall, the Latvian group head (iL3) felt that the cooperation was easy and was proceeding well.

India

According to the Indian chief the Indians and the Finns had different communication cultures. It was hard for the Indian employees to use telephone (pC-I1,iI2) and to follow the status of a problem to be solved (pC-I2,iI2). They merely sent an email and if they did not receive an answer they tended to forget the whole thing (iI2). This in turn was very frustrating to the involved Finns. Indians also tended to have too much optimism and indifference towards their tasks (pC-I3,iI2). When they noticed a problem they assumed that they were capable of solving it and let time pass (pC-I4,iI2). They usually noticed only after the deadline that they after all were not capable of solving the problem (pC-I5,iI2).

According to the Finnish coordinator, there occurred each week situations in which some issue which had earlier been agreed on had nevertheless not been properly dealt with (pC-I6,iI1). On the other hand, in some cases Indian employees tended to avoid risks excessively (pC-I7,iI1) in which cases they focused heavily on meeting the deadlines. For this reason their own effort estimates contained typically too much slack and consequently the service based on those estimates could not be sold successfully (pC-I8,iI1). They did not necessarily pay attention to the issues which self-evidently should have been checked (pC-I9,iI1). Consequently, there was a need for active work control.

The directness of the Finns raised wonder among the Indian employees; and could even be considered very aggressive by them; especially in the beginning of the service (pC-I10,iI2). The views of the Indian group member (iI3) differed from the views of those who worked in Finland, since according to him there were not many differences between the countries. He also mentioned that the Finns did not master the English language particularly well (pC-I11,iI3).

The policies of the company were followed differently in India and in Finland. In India it was customary to perform quality checks amply and the checks were usually performed by others than the person who had done the work. According to the Indian group member, working only in one service becomes boring (pC-I12,iI3); Indian engineers wanted primarily to be involved in creating something new instead of fixing something old (pC-I13,iI1).

Philippines

The Philippines appeared to have a cultural need to please others; especially the involved Finns (iP2). Thereby, it was hard for them to reveal if they did not like something or if they had a differing opinion (pC-P1,iP2). Practically always, when a Philippine was asked how his or her work was proceeding, the answer was "extremely well". However, this was not always a true statement and a challenge was to get correct information about the actual problems (pC-P2,iP2).

The Philippine employees were afraid of "loosing their faces" which also tended to cause them not to tell all of their problems (pC-P3,iP2). They were also perfectionists. This caused them e.g. to answer to the service requests by any possible means; even though that the service level agreement did not require 100 % performance. If there was a need to analyze those service requests which had not been dealt with within the recommended time limits; they should not be related to individual Philippines, because they may feel themselves being accused of the situation and "go into panic" (pC-P4,iP1). On the other hand, the Philippine group head (iP3) did not feel or admit that there would be major problems which would be due to the cultural differences. The noticed cultural differences, however, included the issue that the Finns involved in this case were extremely straightforward, whereas the Philippines considered meticulously what they themselves were saying related to their work tasks (iP3).

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Comparison of M5' Model Tree with MLR in the Development of Fault Prediction Models Involving Interaction Between Metrics

Rinkaj Goyal, Pravin Chandra, and Yogesh Singh

Abstract

Amongst the critical actions needed to be undertaken before system testing, software fault prediction is imperative. Prediction models are used to identify fault-prone classes and contribute considerably to reduce the testing time, project risks, and resource and infrastructure costs. In the development of a prediction model, the interaction of metrics results in an improved predictive capability, accruing to the fact that metrics are often correlated and do not have a strict additive effect in a regression model.

Even though the interaction amongst metrics results in the model's improved prediction capability, it also gives rise to a large number of predictors. This leads to Multiple Linear Regression (MLR) exhibiting a reduced level of performance, since a single predictive formula occupies the entire data space. The M5' model tree has an edge over MLR in managing such interactions, by partitioning the data space into smaller regions.

The resulting hypothesis empirically establish that the M5' model tree, when applied to these interactions, provides a greater degree of accuracy and robustness of the model as a whole when compared with MLR models.

Keywords

Fault prediction • Interaction • M5' Model Tree • Object oriented metrics • Regression analysis • Quality metrics

Introduction

Software metrics are statistical predictors and estimators used to quantify a specific artifact, at a specific phase of a project [1]. Fault-proneness is an external metric in which internal software metrics are independent variables, and fault-proneness of the module is a dependent variable. A Predictive Model is the outcome of applying a modelling technique to predict the fault-proneness of a software module [2].

In a majority of studies aimed at identifying the fault-proneness of modules the metrics defined in Chidamber and Kemerer's (CK) object oriented metrics suite are most commonly used [3]. D'Ambros et al. reported an enhanced model performance when CK and other Object oriented (OO) metrics were used together and this was endorsed by five independent software modules [4]. However, in this approach of combining metrics, the effect of one metric over another was not considered. For example in the CK metrics suite, Depth of Inheritance Tree (DIT) and Number of children (NOC) are correlated and their interaction effect must be taken into account while developing regression based models. Within the purview of this paper, we use the same dataset created and made publically available by D'Ambros et al. [4]. Furthermore for analysis 6 CK metrics and 11 OO Metrics provided in the dataset are used across three Java based software systems i.e. Eclipse, Mylyn, and Lucene.

These 6 CK metrics, when taken in combination (with two term interaction) result in ($6 + {}^{6}C_2 = 21$) 21 linearly

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_19, © Springer International Publishing Switzerland 2015

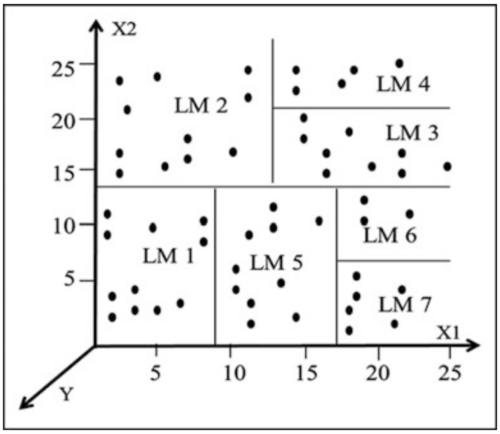

Fig. 1 Illustration of the splitting of input space

interacting metrics. Similarly, the 11 OO metrics in combination return 66 interacting metrics. However, when combined CK and OO metrics give rise to 153 interacting metrics. Such a large number of predictors [metrics] lowers the efficacy of the multiple linear regression and motivates us to explore an alternative modelling technique.

This study expounds our previous work [5, 6] which shows the extent to which a combination of metrics following an interaction results in improved predictions using multiple linear regression (MLR).

However, in this paper, we use the M5' model tree based approach to develop a predictive model, comprised of a large number of predictors [metrics] arised due to the interaction and subsequently infer statistically significant results compared to MLR.

Contrary to the MLR technique, the M5' model tree technique generates piecewise linear models and thereby effectively manages such a multitude of interacting predictors. Figure 1 illustrates how the application of the M5' model tree splits the input space, i.e. X1 x $X2$, into a set of corresponding linear models represented as LM, thus cutting through the entire set of the predictors[7, 8]. This tree-based regression technique has been explained in section "Data description and methods" of this paper.

Applying the M5' model tree to the combination of CK and OO metrics alone, inclusive of interaction, returns better values of statistical measures which otherwise could have achieved by D'Ambros et al. [4] by combining not only CK and OO metrics, but also other information-theoretic code metrics.

The rest of this paper is organized as follows: Section "Related work" presents the review of literature; Section "Data description and methods" describes the data description and methods used, followed by a model methodology utilized in section "Model methodology"; section "Results and discussion" presents the results derived and further interprets the results of this study. Our conclusion is presented in section "Conclusion".

Related Work

The regression tree model for fault prediction was first reported by Gokhale and Lyu [9]. Since then a large number of studies have used these trees-based regression techniques, relevant amongst them are following

Khoshgoftaar et al. illustrated the strength of a regression tree algorithm through the classification of fault-prone modules for a very large telecommunications system [10].

Bibi et al. performed regression via classification by discretizing target variables for training the classification model, and then reversed the process to transform the output generated, back into a numerical prediction [11]. Guo et al. concluded an improved performance of random forest over logistic regression and discriminant analysis using five case studies on a NASA dataset [12]. Chowdhury et al. used techniques like C4.5 Decision Tree, random forests, and logistic regression and compared their prediction performances for fifty-two releases of Mozilla Firefox developed over a period of 4 years [13]. Rodriguez et al. empirically investigated the use of the M5 model tree in effort estimation [14].

Model trees are an extension of regression trees, and the use of a model tree and its variants is not common in software engineering, though other streams of science and technology have reportedly used these techniques, as is briefly highlighted below;

Etemad-Shahidi and Mahjoobi compared M5' model tree and neural networks for prediction of wave height in Lake Superior and concluded the relative advantage of the model tree over an artificial neural network (ANN) both in terms of prediction capability and convenience of use [15]. Bhattacharya et al. used neural networks and M5 model trees to model water level–discharge relationships and established that the ANN- and M5 model tree based models are more accurate than traditional models [16]. Solomatine et al. used model trees as an alternative to neural networks in rainfall–runoff modelling and establish that both models exhibit similar performances [17].

Data Description and Methods

Dataset

The data used in this study is retrieved from a publicly available data set (D'Ambros et al.) [4] for single version CK-OO Metrics. Datasets for three Java based software modules have been utilized (Mylyn, Eclipse and Lucene). Details of the source code metrics considered in this paper are provided in Appendix A.

Multiple Linear Regression With and Without Interaction

Multiple linear regression analysis is carried out to map a numerical response (dependent variable) with two or more independent variables (predictors) (Eq. 1)

$$y = \beta_o + \beta_1 x_1 + \beta_2 x_2 + \ldots + \beta_k x_k + \epsilon \quad (1)$$

Where y is numerical response and x_1, $x_2 \ldots x_k$ are independent variables. β_k determines the contribution of independent variable x_k in the overall fit of the model with βo as the intercept term. ϵ is the random error assuming it to be normally distributed. To find regression coefficients (β_1, $\beta_2 \ldots \beta_k$), the ordinary least square (OLS) method is performed, to minimize the squared distance between the predicted and actual values [18].

More exhaustive linear regression analysis includes higher powers of predictors and interaction between them, to provide synergistic effects of combined predictors (Eq. 2)

$$y = \beta_o + \beta_1 x_1 + \beta_2 x_2 + \beta_{12} x_1 x_2 + \beta_3 x_1^2 + \beta_4 x_2^2 + \epsilon \quad (2)$$

Where β_{12} is the coefficient for the interaction term and interpretation of all other terms is the same as Eq. 1.

In this paper, following equation is used to compute two term interaction (including quadratic terms) between metrics

$$y = \beta_0 + \sum_{i=1}^{P} x_i \beta_i + \sum_{i,j=1;i>j}^{P} x_i x_j \beta_{ij} + \sum_{i=1}^{P} \gamma_i x_i^2 \quad (3)$$

Where P = number of predictors (Metrics)

Henceforth regression expressed through Eq. 3 will be referred to as the "**full interaction model**".

Regression Tree and Its Variants

A classification and regression tree (CART) is a treelike representation of a sequence of decisions taken. Each internal node encapsulates a decision taken to carry out subsequent predictions. In regression trees, the response variable holds a numerical value and differs from other documented classification techniques, like decision trees, which have labels associated with the leaves instead [19].

Model trees are an extension of regression trees which generate decision trees with multivariate linear models associated with each of the leaves [20, 8].

The M5 model tree and its reconstructed and improved implementation, the M5' model tree, are two significant approaches in the implementation of model trees [21, 7].

Standard M5 algorithm adopts a greedy approach to construct a model tree with a flexible structure by applying a predefined stopping criterion. M5 minimizes errors starting at the root and repeating the process recursively at each interior node, considering one node at a time, until all or almost all of the instances are correctly classified [22].

In an M5 model tree, an input set is progressively split into subsets corresponding to outcomes derived from the splitting procedure. The split is based on minimizing the variation within the subsets, each shown as an individual output value along each branch. The standard deviation of output values at each node is taken as a measure of error. Thereby, it becomes possible to estimate the expected reduction in error accruing to testing each attribute for all split values.

The attribute that maximizes the expected error reduction is chosen. The standard deviation reduction (SDR) is calculated by [7].

$$SDR = SD(T) - \sum SD(T_i) \times |T_i|/|T| \quad (4)$$

Where T = set of examples that reaches the node; T_i = subset of examples having i^{th} outcome; SD = standard deviation

Through the further evaluation of the M5 model tree, the M5' model tree was derived to provide a more computationally efficient strategy to construct piecewise linear models, by first constructing a piecewise constant tree and then associating a linear regression model to each leaf node.

Model Methodology

To compare the performance of the models developed, we present R^2, Adjusted R^2 and root mean square error (RMSE) values as statistical measures. The R^2 is the fraction of the variance shared in between a dependent and the independent variables in regression analysis. Its value lies in between 0 and 1, with a value closer to one indicating more variance explained by the model.

The adjusted R^2 value measures how well a model will fit a given data set and takes into account the degree of freedom of the model. Thus, the adjusted R^2 is a preferred metric to determine model fit. It signifies the proportion of total variance explained by the model and hence, its value does not increase merely by adding more predictors to the regression model.

RMSE is the standard deviation of errors, which measures the spread of data around the regression line [23].

It shows the absolute fit of that particular model, thereby making it an important criterion to measure predictive capability. Statistically, the value of RMSE decreases as the value of Adjusted R^2 increases, therefore a lower value of RMSE ensures a better explanatory power of the model.

Both Adj R^2 and RMSE are relevant in the context of a model in isolation. Neither takes into account the accuracy of the model in comparison to other models.

To establish the relative accuracy of a model to predict future data, **Akaike's Information Criterion (AIC)** is used [24]. AIC is the information-theoretic measurement to select a model with an enhanced predictive accuracy from amongst the given set of models [18]. It is defined as follows:

AIC = Sample size × log (Mean Square Error) + 2 × numbers of Predictors

AIC is not an absolute criteria and is typically used for comparing models. Therefore the model with the smallest value of AIC, from within the set of the models being compared will have the highest predictive accuracy. The relative accuracy of i^{th} model in the set is represented as Δi, where

Δi = AIC of i^{th} Model—AICmin; where AICmin is the minimum AIC from the given set

The model with Δ = 0 will outperform all the other models in the set. Further, the relative predictive accuracy of regression techniques applied here is measured by the difference in between the AIC i.e. AICd values of corresponding model.

The above mentioned statistical measure (i.e. R^2, Adj. R^2, RMSE and AIC) have been computed in this study to compare the performance of M5' model tree and MLR to each other.

These have been implemented and simulated in the Matlab 7.9.0 (R2009b) environment [7, 25]:

Results and Discussion

A combination of metrics in isolation, as well as with interacting terms (Refer Eq. 3), is considered in this study to appropriately highlight the importance of M5' model trees in fault prediction. The statistics generated from three software modules (Mylyn, Eclipse and Lucene) of the dataset will be presented here.

The M5' model tree and MLR based models built for each software module is characterized in Tables 1–6.

Table 1 Comparison of statistical predictors of models in Mylyn Software module

Mylyn software module						
	M5' Model tree			MLR		
Metrics	R^2	Adj R^2	RMSE	R^2	Adj R^2	RMSE
CK	0.3017	0.2994	0.5505	0.1186	0.1157	0.5634
CK(WI)	0.5003	0.4984	0.4234	0.2198	0.2084	0.5330
OO	0.5149	0.5120	0.4171	0.1756	0.1708	0.5455
OO(WI)	0.5882	0.5855	0.3844	0.3717	0.3446	0.4850
CK + OO	0.5039	0.4993	0.4218	0.2024	0.1950	0.5374
CK + OO (WI)	0.6164	0.6126	0.3709	0.4881	0.4366	0.4497

Table 2 Comparison of AIC values of models in Mylyn software module

Mylyn software module					
Metrics	AICM5'	AIC mlr	AICd = AICM5'- AIC mlr	Δ M5'	Δ MLR
CK	−2566	−2125	−441	1091	818
CK(WI)	−3187	−2331	−856	470	612
OO	−3234	−2235	−999	423	708
OO(WI)	−3537	−2673	−864	120	270
CK+OO	−3180	−2278	−902	477	665
CK+OO (WI)	−3657	−2943	−714	0	0

Table 3 Comparison of statistical predictors of models in Eclipse. Software module

Eclipse software module						
	Model tree			MLR		
Metrics M5'	R^2	Adj R^2	RMSE	R^2	Adj R^2	RMSE
CK	0.6486	0.6465	0.6169	0.3856	0.3818	0.8186
CK(WI)	0.7671	0.7655	0.5022	0.5649	0.5527	0.6963
OO	0.7409	0.7380	0.5297	0.4129	0.4064	0.8021
OO(WI)	0.7963	0.7939	0.4696	0.6626	0.6344	0.6269
CK + OO	0.8196	0.8165	0.4420	0.4280	0.4190	0.7943
CK + OO (WI)	0.8540	0.8513	0.3977	0.8173	0.7797	0.4887

Table 4 Comparison of AIC values of models in Eclipse Software module

Eclipse software Module					
Metrics	AICM5'	AIC mlr	AICd= AICM5'- AIC mlr	Δ M5'	Δ MLR
CK	−951	−387	−564	852	1007
CK(WI)	−1359	−710	−649	444	684
OO	−1245	−418	−827	558	976
OO(WI)	−1483	−901	−582	320	493
CK + OO	−1594	−425	−1169	209	969
CK + OO (WI)	−1803	−1394	−409	0	0

Table 5 Comparison of statistical predictors of models in Lucene software module

Lucene software module						
	M5' Model tree			MLR		
Metrics	R2	Adj R2	RMSE	R2	Adj R2	RMSE
CK	0.5115	0.5072	0.4148	0.3838	0.3784	0.4682
CK(WI)	0.5694	0.5650	0.3894	0.4308	0.4078	0.4571
OO	0.5134	0.5056	0.4140	0.2458	0.2335	0.5200
OO(WI)	0.6189	0.6121	0.3664	0.5650	0.5104	0.4156
CK + OO	0.6269	0.6174	0.3625	0.4050	0.3899	0.4639
CK + OO (WI)	0.7815	0.7758	0.2774	0.6972	0.5982	0.3765

Table 6 Comparison of AIC values of models in Lucene software module

Lucene software Module					
Metrics	AICM5'	AIC mlr	AICd= AICM5'- AIC mlr	Δ M5'	Δ MLR
CK	−1204	−1037	−167	532	279
CK(WI)	−1289	−1070	−219	447	246
OO	−1197	−882	−315	539	434
OO(WI)	−1364	−1192	−172	372	124
CK + OO	−1368	−1028	−340	368	288
CK + OO (WI)	−1736	−1316	−420	0	0

The statistical measures (i.e. Adjusted R^2) for CK metrics, OO metrics and their combination with and without interactions are compared with the results reported in our previous study [5].

In the following tables, CK refers to CK metrics without interaction and CK (WI) considers CK metrics with interaction. A similar terminology has also been applied to the other metrics considered.

Analysis of the Results of Mylyn Software Module

Predictive modelling statistics were generated for CK metrics, with and without interaction from MLR models.

The data in Table 1 adequately reflect that after considering the interaction with CK metrics, there is a significant improvement in the value of adjusted R^2.

The value of Adj R^2 is 0.2084 which is more favourable than the value obtained without interaction. The dip in the value of the corresponding RMSE from 0.5634 to 0.5330 also affirms the improved value of the AdjR^2.

Similar statistics were generated using the M5' model tree, which resulted in better values of Adj R^2 and correspondingly lower values of RMSE for both CK with and without interaction.

For the same software module i.e. Mylyn, OO metrics were considered, again with and without interaction with both MLR and M5' model Tree. Subsequently, through combining CK and OO metrics, the M5' model tree not only has the highest predictive power, but also causes the most improved value of Adj R^2 (0.6126) whilst conforming to the minimum value of RMSE (0.3709) as depicted in Table 1.

To determine the predictive accuracy of these models, the AIC values were computed and compared to the models developed, using both MLR and M5' model trees. The data in Table 2 clearly specifies that the M5' model tree consistently results in minified AIC values when compared to the corresponding values of the models developed with MLR. The minimum AIC value of the model implemented using M5' model tree and MLR is −3,657 and −2,943 respectively, and hence it directly shows the prevalence of the M5' model tree over MLR. Moreover, consistently negative values reflected in the difference (AIC) column of Table 2 determine the superiority of M5' model trees as a prediction technique.

Subsequently, the Δ values (relative accuracy of a model), within each regression technique are given in Table 2. A lower Δ value indicates better accuracy of a model in the set. The data derived concludes that while considering metrics in combination, lower Δ values for M5' and ΔMLR are consistently returned. Furthermore, similar results are obtained for the combination of CK and OO metrics with interaction. As previously discussed (Refer section "Model methodology"), the model implemented by considering the combination of CK and OO metrics with interaction, results in $\Delta = 0$ endorsing the results derived from Adj R^2 and RMSE values in Table 1.

Analysis of the Results of Eclipse Software Module

Experiments similar to Mylyn were conducted for Eclipse as well and CK metrics alone were considered for MLR and the M5' model Tree. The M5' model tree was found to yield a better value of Adj R^2 (0.6465) and a correspondingly better (decreased) value of RMSE (0.6169). Similarly for OO metrics, the M5' model tree consistently returns the most favorable results. While considering the combination of CK and OO metrics, we get an even greater value of Adj R^2 (0.8513) and the least value of RMSE (0.3977) when using the M5' model tree (Refer Table 3)

For the M5' model tree and MLR, the minimum value of AIC is −1,803 and −1,394 respectively (Refer Table 4). This once again confirms the dominance of M5' model tree technique over MLR and is further endorsed by the value of AICd. Moreover, for the combination of CK and OO with the interaction Δ becomes zero. This is in consonance with the results derived through Table 3.

Analysis of the Results of Lucene Software Module

When applied to the Lucene, the M5' model tree reveals a more proficient explanatory power of CK and OO metrics in isolation.

The combination of these metrics provides even better values of the statistical measure (Adj $R^2 = 0.7758$ and RMSE = 0. 2774) (Refer Table 5). Similar outcomes for the model's predictive accuracy can be inferred by using AIC statistics as shown in Table 6.

Conclusion

We have empirically established the prevalence of the M5' model tree over MLR in the design of a fault prediction model involving interaction amongst metrics.

Model trees, in contrast to MLR, divide the input set into a number of subsets, and for each subset a separate specialized piecewise linear model is built, thereby managing interactions effectively. Moreover, model trees have the additional advantage of comprehensive representation and less complex parameter settings.

This study points towards a means by which conventional empirical analyses can be improved upon and further work will consist of identifying the most influential set of interacting metrics may be using efficacious information-theoretic techniques, to reduce the dimensionality of the large number of predictors. This will assist project managers to take the most relevant set of metrics which contribute the most towards the precise measurement of external quality characteristics of software.

Acknowledgment Corresponding author would like to thank Mr. Tanveer
Oberoi and Ms. Preeti Goyal to copyedit manuscript.

Appendix A

CK Metric (Chidamber and Kemerer 1994)	Interpretation
Weighted Methods per Class (WMC)	Identify complexity of class by finding the weighted sum of the complexity of the methods
Coupling Between Object classes (CBO)	Identify the coupling between classes by considering the dependency of one class with other classes in the design
Depth of the Inheritance Tree (DIT):	Identify the complexity of inheritance hierarchy by calculating the maximum length of a given class to the root class
Lack of Cohesion metric (LCOM)	Identify cohesion with a class by counting the number of method pairs with zero similarity
Number of Children (NOC):	Identify complexity of inheritance hierarchy by counting the number of immediate child classes that have inherited from a given class
Response for the classes (RFC)	Identify the coupling between classes by calculating the sum of the number of local methods and the methods that can be called remotely
OO (Object Oriented)	**Interpretation**
NOM	Number of methods
NOPM	Number of public methods
NOPRM	Number of private methods
NOMI	Number of methods inherited
Fan-in	Number of other classes that reference the class
Fan-out	Number of other classes referenced by the class
NOAI	Number of attributes inherited
NOA	Number of attributes
NLOC	Number of lines of code
NOPRA	Number of private attributes
NOPA	Number of public attributes

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Content Based Image Retrieval System with a Combination of Rough Set and Support Vector Machine

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Abstract

In this paper, a classifier based on a combination of Rough Set and 1-v-1 (one-versus-one) Support Vector Machine for Content Based Image Retrieval system is presented. Some problems of 1-v-1 Support Vector Machine can be reduced using Rough Set. With Rough Set, a 1-v-1 Support Vector Machine can provide good results when dealing with incomplete and uncertain data and features. In addition, boundary region in Rough Set can reduce the error rate. Storage requirements are reduced when compared to the conventional 1-v-1 Support Vector Machine. This classifier has better semantic interpretation of the classification process. We compare our Content Based Image Retrieval system with other image retrieval systems that uses neural network, K-nearest neighbour and Support Vector Machine as the classifier in their methodology. Experiments are carried out using a standard Corel dataset to test the accuracy and robustness of the proposed system. The experiment results show the proposed method can retrieve images more efficiently than other methods in comparison.

Keywords

Classifier • Content Based Image Retrieval system • Rough Set • Support Vector Machine

Introduction

Image Classification is one of the most important aspects in Content Based Image Retrieval (CBIR) systems [1, 2], therefore using the appropriate classifier for CBIR systems is critical. Many past papers used different classifiers for their CBIR systems [3–5], however some problems still remain. Some of the drawback of current classifiers is the lengthy training time [6], high storage requirements [7], did not achieve the required semantic results [7] and cannot deal with incomplete and uncertain data and features [8].

In this paper, we used a combination of Rough Set and 1-v-1 Support Vector Machine (SVM) as the classifier for our CBIR system (which was proposed in [9]). In the 1-v-1 SVM, one SVM is constructed for each pair of classes [10].

The reasons why the proposed Rough Set with SVM as a classifier has better result compared to the conventional SVM classifier due to the fact that conventional SVM has high storage requirements and lack of semantic interpretation of classification process [10, 11]. Rough set can reduce the storage requirements by using upper and lower approximations. This means that the $N \times (N-1)/2$ rules should store in conventional 1-v-1 SVM, however this amount is reduced to $2 \times N$ for classifier include the Rough Set and 1-v-1 SVM [11]. In addition, combination of Rough Set and 1-v-1 SVM can provide a better semantic interpretation of the classification process using properties of the Rough Set boundary region [9].

This paper is organised as follows: section "Rough set method to support vector machine 1-v-1 multi classifiers"

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_20, © Springer International Publishing Switzerland 2015

presents Rough set method to support vector machine 1-v-1 multi classifiers. Experiment setup is given in section "Experiment setup". In section "Experiment results", we show experimental results. Conclusion is made in section "Conclusion".

Rough Set Method to Support Vector Machine 1-V-1 Multi Classifiers

This section describes a rough set method to SVM 1-v-1 multi classifiers as proposed in [9]. First, a nonlinear separable feature space is transformed to a linear separable feature space using a Radial Basis Function Kernel (RBF Kernel). The reason for choosing this kernel is RBF kernel has better results in CBIR systems [12]. The perfect situation is that the SVM can find the hyper-plane by maximizing the margin between two classes and no example are in the margin i.e. after transforming non-linear feature space to linear feature space [13, 14] (see Fig. 1).

However when there are some examples between the margin, applying a method which can deal with vague and uncertain spaces like Rough Set is essential. The margin can be used as the Rough Set boundary region. Using the formulas shown below, Rough Set is applied to SVM, and b_1 and b_2 correspond to the boundaries of the margin in Figs. 3, 4, and 5 (red lines).

b_1 is defined as follows: $y \times [< x, w > + b_1] \geq 0$, for all $(x, y) \in S$, and there exists at least one training example $(x, y) \in S$ such that $y = 1$ and $y \times [< x, w > + b_1] = 0$.

b_2 is defined as follows: $y \times [< x, w > + b_2] = 0$, for all $(x, y) \in S$, and there exist at least one training example $(x, y) \in S$ such that $y = -1$ and $y \times [< x, w > + b_2] = 0$.

The above variables are defined as follows: Assume x is an input vector in the input space X and y is the output in $Y = \{-1, +1\}$. Training set used for supervised classification is $S = \{(x_1, y_1), (x_2, y_2), \ldots, (x_i, y_i), \ldots\}$. $< x, w > = \sum_j x_j \times w_j$ is inner product and x_j and w_j are components of two vector x and w.

According to R1, R2 and R3 rules, a Rough Set based SVM binary classifier can be defined when:

[R1] If $< x, w > + b_1 \geq 0$, classification of x is positive (+1).

[R2] If $< x, w > + b_2 \leq 0$, classification of x is negative (−1).

[R3] Otherwise, classification of x is uncertain.

In the SVM 1-v-1 multi classifier, one binary SVM is constructed for each pair, (i,j), of classes. According to the rules of R1, R2 and R3, three equivalence classes can be defined for each pair. $P_{ij}(POS_i)$, $P_{ij}(POS_j)$ and $P_{ij}(BND)$ are the set of x (or region) that follows the rule R1, R2 and R3 respectively. Lower $\left(\underline{A}(class_i)\right)$ and upper approximations $\left(\overline{A}(class_i)\right)$ and boundary region for class i and j are summarised in Table 1.

Classification problem with three classes i.e. Flower, Elephant, and African people is shown in Fig. 2. Figures 3, 4, and 5 show the Rough Set method to SVM 1-v-1 classification for classes Flower and Elephant, Flower and African people and Elephant and African people respectively.

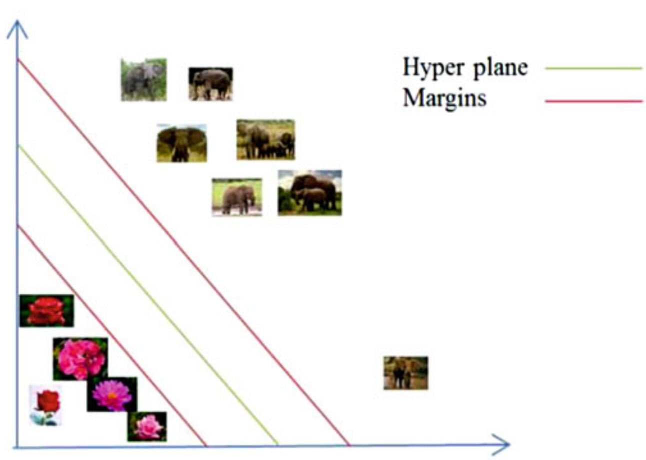


Fig. 1 Maximizing the margin between two classes

Experiment Setup

Our proposed CBIR system in this paper has training and testing phases. For each semantic group, two rules are extracted as we used 10 semantic image groups in our experimental results so that 20 rules are extracted in the training phase ($2 \times N$). In the testing phase, the user feed a query image to CBIR system. According to rules from the training phase, if the query image is related to the positive region, images in that positive region are shown to user as the retrieval results. However, if the query image is related to the boundary region using a threshold, the images which are most likely similar to query image is then shown to user.

From many experimental results, a threshold was defined for each of the semantic image groups. For each semantic group, if distance between image in boundary region and images in the positive region is less than the threshold, the image in the boundary region will be categorized in the positive region group.

Experiment Results

In this section, the results that compare the three retrieval systems with the proposed retrieval system are presented. These three retrieval systems used SVM [7], neural network

Table 1 Lower and upper approximations and boundary region for class i and j

	Lower approximation	Upper approximation
Class i	$P_{ij}(POS_i)$	$P_{ij}(POS_i) \cup P_{ij}(BND)$
Class j	$P_{ij}(POS_j)$	$P_{ij}(POS_j) \cup P_{ij}(BND)$
Over all lower approximation for class i (N is number of classes)	$\underline{A}(class_i) = \bigcap_{j=1, j \neq i}^{N} P_{ij}(POS_i)$	
Over all Boundary region for class i (N is number of classes)	$\overline{A}(class_i) - \underline{A}(class_i) = \bigcup_{j=1, j \neq i}^{N} P_{ij}(BND) - \bigcup_{j=1}^{N} \underline{A}(class_j)$	
Over all upper approximation for class i (N is number of classes)	$\overline{A}(class_i) = \bigcup_{j=1, j \neq i}^{N} P_{ij}(BND) - \bigcup_{j=1}^{N} \underline{A}(class_j) + \underline{A}(class_i)$	
Some rules extract from above formula	$P_{ij}(POS_i) \cap P_{ij}(POS_j) = \emptyset$; $\underline{A}(class_i) \subseteq P_{ij}(POS_i)$; $\underline{A}(class_j) \subseteq P_{ij}(POS_j)$ $\underline{A}(class_i) \cap \underline{A}(class_j) = \emptyset$	

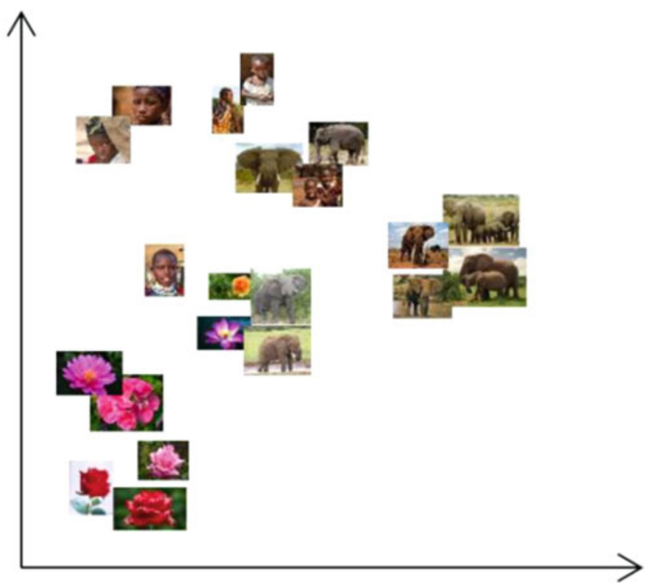


Fig. 2 A classification problem including three classes Flower, Elephant and African people

(NN) [8] and K-nearest neighbour (KNN) [6] as the classifier in their methodology.

To investigate the function of the image retrieval system based on the above mentioned methods, we used the COREL image database containing one thousand images. In this database, images are classified into ten semantic groups. The groups are African people, beach, bus, flower, mountains, elephant, horse, food, dinosaur, and building. We expressed the results of each group with a number e.g. number 1 represents African people; number 5 represents mountains, and etc.

Precision-Recall Graph

Recall equals to the number of the related retrieval images to the number of the related images available in the images database. The precision equals to the number of the related retrieval images to all of the retrieval images [2]. Figure 6 shows the precision-recall graph for ten semantic groups that is used for measuring the efficiency of the proposed retrieval system. In the experiment results, the proposed retrieval system is shown as RSSVM. From the graph, we observed that the proposed retrieval system achieved better results than the other three systems. The reason for this is the proposed method is the ability to handle the uncertain boundaries better, thus able to classify those in the region more accurately.

The Investigation of the Retrieval Precision

To investigate the total precision of the above mentioned retrieval systems, 1,000 images are fed into the system as the queried images. The average of the retrieval precision is calculated for each class. Figure 7 shows the results using different classifiers. As anticipated, the results are better using the proposed system. The average of the retrieval precision is 55.9 %, 59.4 % and 68.1 % for SVM, NN and KNN respectively. It increases to 73.8 % for RSSVM.

The reasons behind the superiority of RSSVM are the:

1. Overlapped region in the classification problem can be described using boundary region in Rough Set more accurately.
2. Optimal separating hyper-plane by maximizing the margin is constructed using SVM effectively.

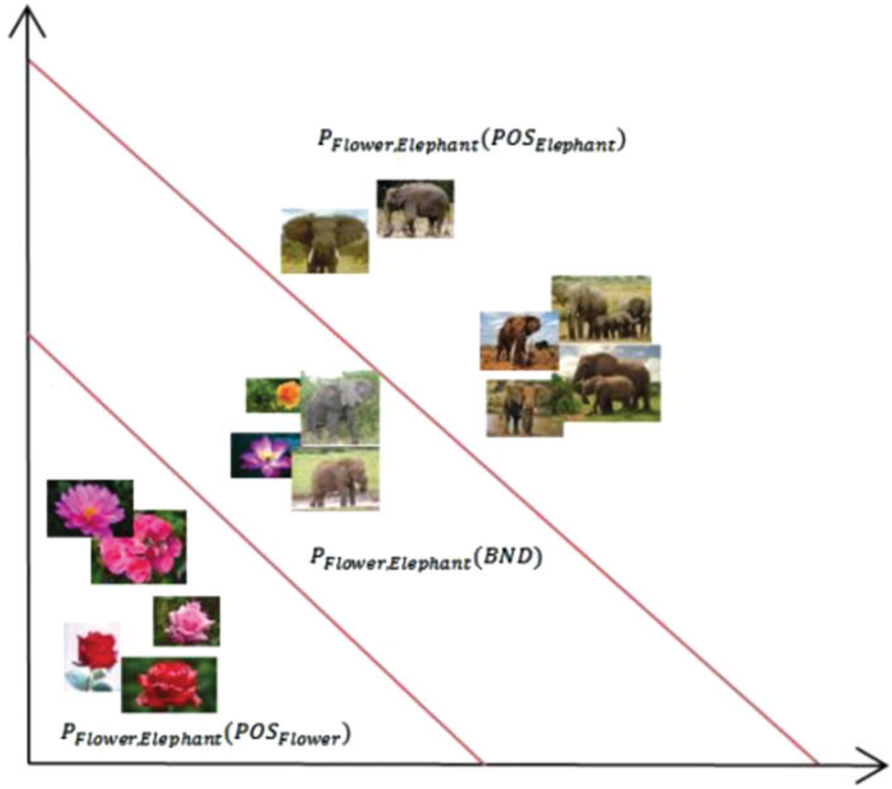


Fig. 3 A rough set method to SVM 1-v-1 classification for classes Flower and Elephant

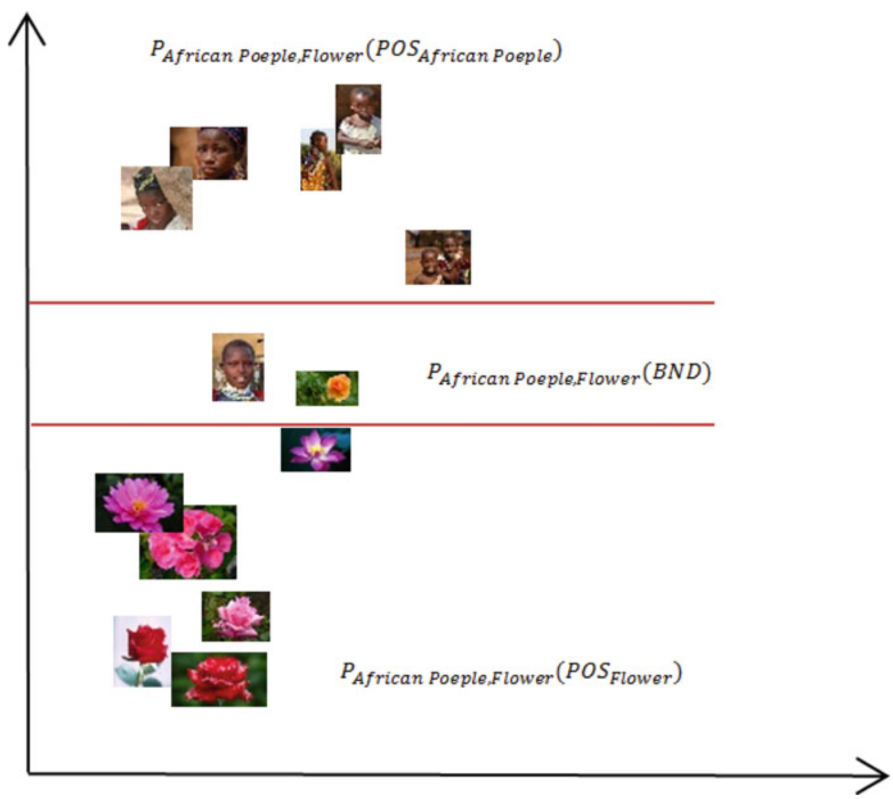


Fig. 4 A rough set method to SVM 1-v-1 classification for classes Flower and African people

Fig. 5 A rough set method to SVM 1-v-1 classification for classes Elephant and African people

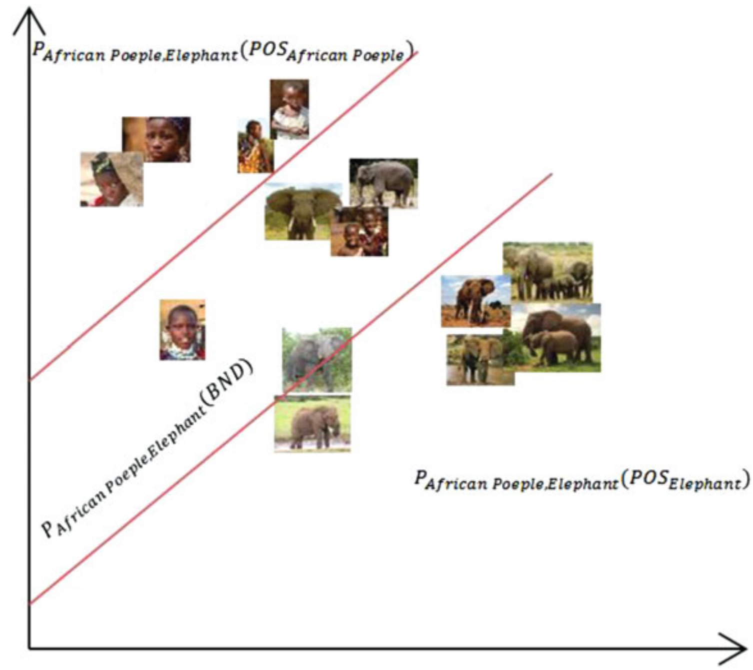
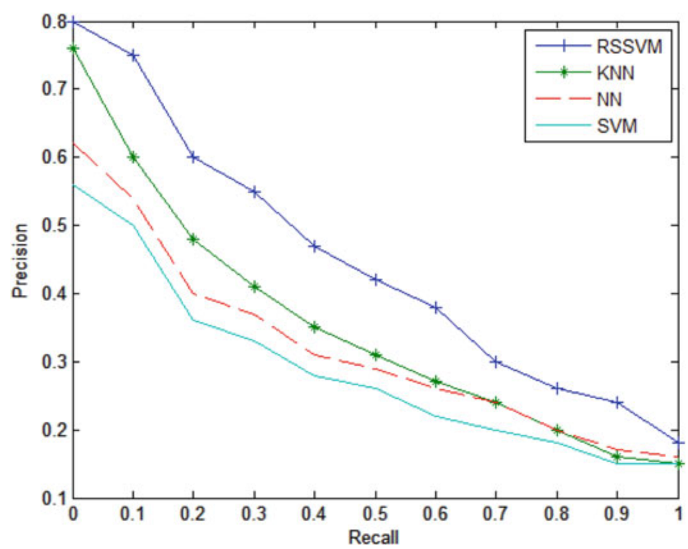


Fig. 6 Precision- recall graph



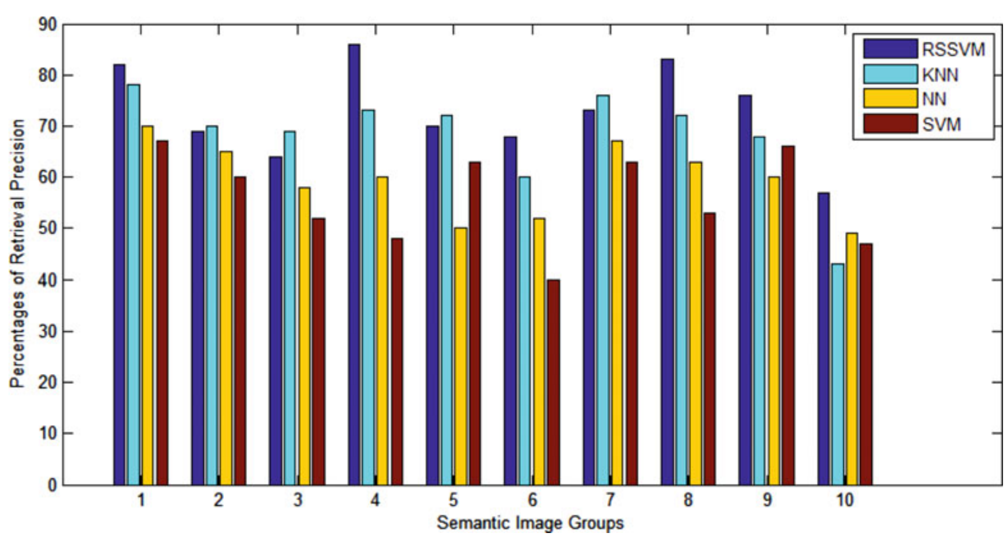


Fig. 7 Precision of retrieval

3. Perfect generation ability is the SVM's properties, however cannot deal with imprecise or incomplete data.
4. Most important properties of rough set is that it can deal with vague and incomplete data efficiently.

In addition, the RSSVM classifier has some advantages compared to the conventional SVM. One of the advantages is RSSVM reduced storage requirements. RSSVM requires to store just $2 \times N$ rules for each class (one rule for lower approximation and one rule for upper approximation), compared to conventional SVM that needs to store $N \times (N - 1)/2$ rules [10]. Another advantage of RSSVM is that it has better sematic interpretation of the classification process compared to the conventional SVM [11].

The Image Comparison of the Retrieval Systems

In the final test, we presented the retrieval results for the queried flower image (Fig. 8). The first, second, and up to the fourth row in Fig. 9 is related to RSSVM, KNN, NN and SVM respectively. Referring to Fig. 9, the retrieval system with the RSSVM classifier represented a more related output images to the user. The first left image in Fig. 9 matched closely to the queried image.

The reason why the proposed method has better results than those in other retrieval systems is that the rules extracted from the RSSVM classifier are semantic and can better classifies images. Consequently, the RSSVM classifier can show more relevant images to user.

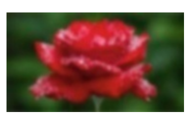


Fig. 8 Query image

Conclusion

This paper proposed a classifier based on a combination of Rough Set and a 1-v-1 Support Vector Machine (SVM) in a Content Based Image Retrieval (CBIR) system. The proposed image retrieval system is compared with other image retrieval systems which used other classifiers such as neural network, K-nearest neighbour and Support Vector Machine in their systems. Based on the experimental results, it can be concluded that CBIR system with Rough Set and SVM classifier has good results and better semantic interpretation of the classification process compared to conventional SVM. In addition, this classifier reduced the storage requirements because it only requires to store a $2 \times N$ rules.

In this paper we focused on a 1-v-1 (one-versus-one) support vector machine for future it is good idea will do some research on 1-v-r (one-versus-rest) support vector machine, and evaluated results using combination of rough set and 1-v-r support vector machine as a classifier in content based image retrieval systems.

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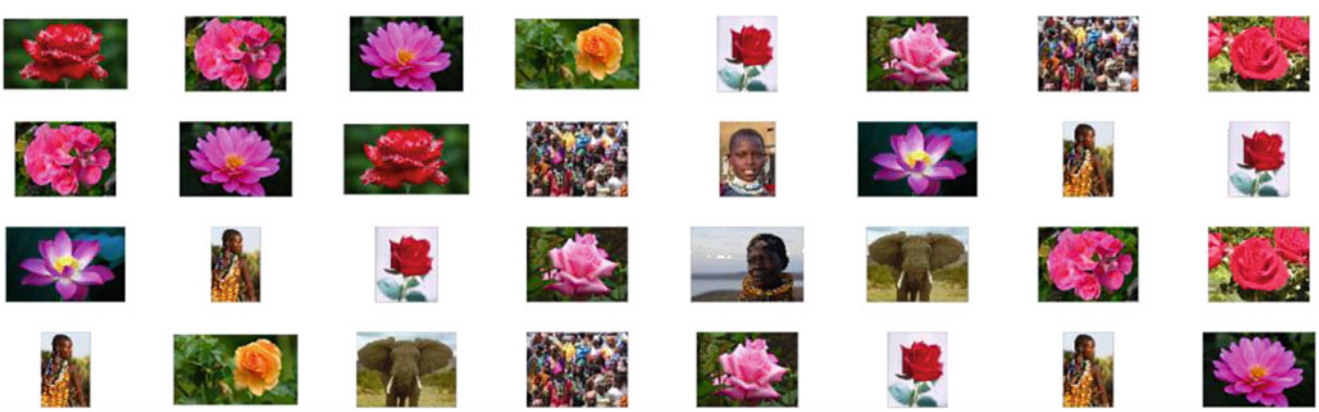

Fig. 9 Retrieved images according to: first raw—RSSVM, second raw—KNN, third raw—NN, fourth raw—SVM

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Cloud Storage Using Matches Algorithms

Zyad Nossire, Julius Dichter, Eman Abdelfattah, and Subrina Thompson

Abstract

The cloud is currently a ubiquitous facility used in global data storage. The cloud technology offers users an ability to store and retrieve data from disparate applications and devices. In order to utilize this data technology efficiently and to reduce the associated costs, we introduce a technique to minimize the space used for data storage. Storing data on a cloud server may not be efficient because multiple uploads of the same file may produce redundant data in the cloud. Currently, this technology is not utilizing matching algorithms which could determine the changes in the new version of a previously uploaded file. Use of such an algorithm could save storage space by only requiring modification of the previous version already stored in the cloud. In this paper we will check several different, and difficult, problems and see what affects the automatic load-balancing and asynchrony have on the speed of resolution of problems.

Keywords

Cloud Storage • Global data storage • Matching algorithms • Load-balancing

Introduction

In this work, we are looking at a disseminated evolutionary calculation trial and the main idea is using a processing node in order to find accessible intends. In order to accomplish this, we may need to access cloud administrations frequently. Such an approach is utilized in cloud storage. Cloud storage is sold as a fog space administration and is free within a certain level of use, as determined by the activity and amount of use. There are other administrations similar to cloud storage but cloud storage was chosen because of its prominence as well as the many potential volunteer clients. There are also other factors about it that make it the right administration to use. Wuala is a service of fog space and gives a customer project where you need to include a document that needs to be saved. But it doesn't give you a smooth coordination of the system. Zumodrive is another service that uses remotely-mounted document frameworks to gain access. Drop box service is more efficient because it uploads data asynchronously. We are interested in this use in the experiments we are doing. When one record in an envelope is seen by Cloud space and updated, it is transferred to the Cloud space servers and afterward sent to all customers who have the same envelope. It is intriguing to note that from the modifying perspective, all envelopes are composed and read as if they were a nearby one. This makes its use easy and transparent.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_21, © Springer International Publishing Switzerland 2015

In past trials we measured if adding numerous workstations to an analysis of this kind brought about an expansion in the amount of concurrent assessments [1]. We will also measure if by an increase in speed, the calculation benefits from the appropriation and asynchrony of the specific task we have executed, or is affected adversely. Keeping in mind the end goal to do that, we picked two complex issues with an alternate level of challenge and measured the time it would have done well to discover the result as well as the amount of assessment. The remaining part of the paper is formed as follows: after a concise area exhibiting the state of the art in voluntary and pool based evolutionary processing, we depict the calculation and the exploratory setup and usage in section "Experiments"; results of these tests will be shown in section "Conclusion", followed by the conclusions and future work.

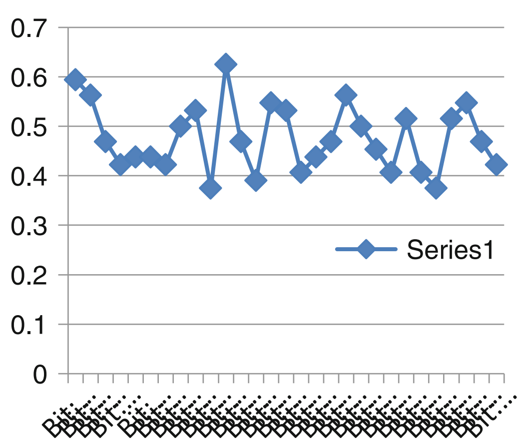
State of Cloud Computing

Cloud computing is a new but quickly developing technology [2, 3]. Research into fog space is basically identified with substance conveyance [4] or planning information repetition plans to guarantee qualified data [5]. However, its use in dispersed processing was not well thought-out beforehand. Regardless of the identified possibility that it is with information matrices [6], in this paper we do the utilization of free mist space as a medium for doing dispersed evolutionary calculation, in the approximation of a parasitic route [7]. This is because of the fact that we make use of the framework laid by the supplier as a major aspect of a movement plan in the evolutionary calculation [8]. Therefore, we will consider pool-based dispersed evolutionary calculations for similar systems to the one we will demonstrate here. In these systems, many nodes or islands offer a pool where the normal informative content is composed and read. Functioning against a solitary pool of results has been the case from the time of the beginning of the exploration in dispersed evolutionary calculations. Non concurrent Teams or A-Teams [9–11] were proposed in the early nineties as a helpful plan for independent operators. The idea is to make a work-stream on a set of results and use many heuristic systems to make them stronger while adding people to finish them. This procedure isn't only for evolutionary calculations; it may also be connected to any populace-based procedure. However for the ease of connection, it might mean making distinctive single-era calculations, which may include some methods that might make another era from the existing pool. The A-Team system does not depend on a solitary execution but rather on concentrating on the method and information stream viewpoints, in the same way as the Meander framework. This framework makes an information stream system, with its own special terminology (called Zigzag), which could be connected, specifically, to evolutionary calculations [12]. Although calculation configuration is very significant, its usage issues are of greater importance. Some papers have thought about managing pool architectures on ground: Satyanarayanan et al. [13] propose an imparted memory multistring structural engineering, in which numerous strings work freely on a solitary imparted memory, having read-access to the entire pool but compose-access to only a part of it. That way, interlock issues could be avoided, and, showing the various strings improved building design of today's processors, they can get very effective, running time-wise, results, with the included algorithm preference of working a nature's turf. Although they don't distribute scaling outcomes, they examine exchange of working with a pool whose size will have a bigger impact on exhibition than on the populace size on a single-processor or the conveyed ease. Bolin and Piasters found the same problem [14], and they presented a configuration example for constant and dispersed evolutionary calculations; despite the fact that their stress is on perseverance, and not exhibition, they tried to make some options to decouple populace space from development itself (customary evolutionary calculations are connected straight on space) [14]. They made an item-situated database administration framework. That way, our old browser-based evolutionary calculation is now additionally comparative, utilizing for determination a minor database accessed by a web interface, with the goal of distributing users around different locations, not just as space for everyone. Few stable changes have been seen currently. That may be because until recently there had been few, if any, freely open online databases. After the rise of mist-registering stages recently, investment in this sort of calculations has bounced back once more, when operation using the public Fluid Data base platform was published [10].

Algorithm and Implementation

A pool based evolutionary algorithm can be described as an island model without topology; the evolutionary algorithm is an accepted unified with binary encoding, other than the two steps inside the cycle that conditionally gain foreigners [13]. A base number of assessments for the entire calculation is executed from the start; we will see later that the best method to control when this base number of assessments is known a priori. Throughout the evolutionary circle, new people are chosen utilizing a three-competition choice and created utilizing touch flip transformation and uniform hybrid. Relocation is presented in the calculation it makes afterward. After the populace is put in, relocation might happen if the number of generations selected to do it can be determined. The best *distinct* is sent to the pool, and the best *single* in the pool (picked around those emitted by a different node) is built up into the populace. Provided that there has been no change in the best single since the final movement, a random distinct is added to the pool, which adds opposite qualities to the populace regardless of the fact that the distinctive fitness is not the clearest one. This has no impact on the consequence in the present, and even later when algorithmic tests are run. Transients, if any, are merged into the populace substituting the worst quality single as well as the posterity of the past era utilizing a generational swap with a one-tip top. Populace was set to 1,000 people for all issues, with base assessment of four million. Many relocation rates were tried to check their effect on exhibition. In addition, we presented a 1-s postponement after relocation so that the workload was decreased and the mist space daemon had enough time to make records for the workstations. This postponement makes one-computer analyses faster when less relocation is made, and must be in tuned sometime later. The outcomes are repaired at the close of the circle to check if the calculation has finished, that is, discovered the (single) result for the issue. One of the advantages of this topology-less arrangement is the independence on the number of computers participating in the experiment, and also the lack of need of a central server (although it can be arranged so that one of the computers starts first, and the others start running when some file is present). If any other PC is included, then it does not suggest making new connotations with the present set of PCs; the only thing that needs to be done is to locate the directory with the migrated individuals that are to be shared. Two agent capacities have been chosen to make the tests; the principle is that they took a step back, but in a way that the experiments did not take a long time. One of them is P-Peaks, a multi-modal problem generator proposed by De Jong in [14]; a P-Peaks instance is created by generating P random N-bit strings where the fitness value of a string by generating P random N-bit string

$$\int \boldsymbol{P-Peaks}\left(\vec{x}\right) = {}^{1}\!/_{N} \max_{1\le i\le P} \left\{ \boldsymbol{N} - \boldsymbol{H}\left(\vec{x}, \boldsymbol{Peaki}\right) \right\} \quad (1)$$



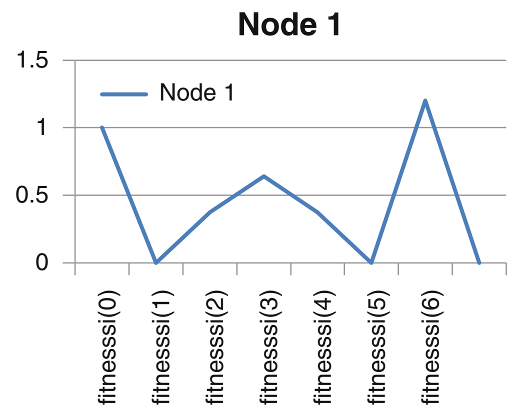
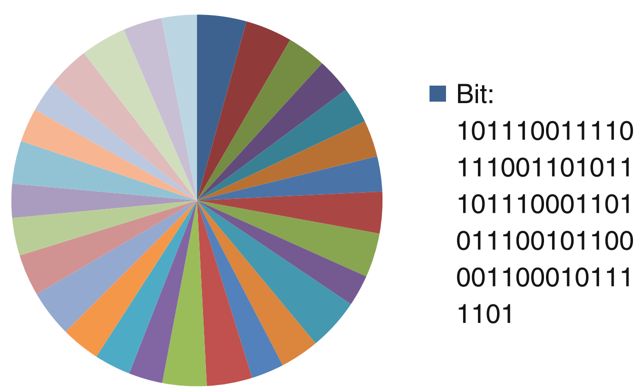


Fig. 1 Graph of the values for a single variable in the MMDP problem (*bottom*)



Where $H\left(\vec{x}, \vec{y}\right)$ is the Hamming distance between binary string $\vec{x}$ and $\vec{y}$, in the experiment made in this paper.

We consider P = 100 and N = 64 and Fitness it will be 1.0 the anther Function is MMDP [14]. This is a deceptive Problem composed of k sub problems of 6 bits each one (S_i).

Values depicted next:

$fitnesss_{S_i}$ (0) = 1.0 $fitnesss_{S_i}$ (1) = 0.0

$fitnesss_{S_i}$ (2) = 0.372384 $fitnesss_{S_i}$ (3) = 0.640576

$fitnesss_{S_i}$ (4) = 0.372384 $fitnesss_{S_i}$ (5) = 0.0

$fitnesss_{S_i}$ (6) = 1.2

The fitness value is defined as the sum of the si sub problems with an optimum of k [Eq. (2)]. Figure 1 represents one of the variables of the function. The number of local optima is quite large (22k), while there are only 2k global solution s2

$$\int \boldsymbol{MMDP}\left(\vec{s}\right) = \sum_{i=1}^{k} \boldsymbol{fitness\, s_i} \quad (2)$$

In this paper we have used k = 20, in order to make it difficult enough to need a parallel solution.

Experiments

Here, the investigations were finished in many varying machines associated in distinctive ways; notwithstanding, PCs were added to the investigation in the same request; some problems were noticed first in a solitary workstation, then in two and finally with four devices. The total time and number of evaluations were measured. Since the end of the analysis is increasing quickly, for example by means of mist space, the amounts of assessments were not taken at the same time like when the result is first discovered. This number also increases with the number of nodes.

The main parameter that was modified in trials was the relocation rate. We were surprised by the results of this because the system behavior is very much affected by the movement rate: band with use (and maybe latency) builds with the reverse of the movement rate. Then again, evolutionary exhibition will expand in the inverse bearing: the greater the relocation rate. Also, evolutionary performance will increase in the opposite directions which make discovering the result easier. Additionally, it will make assorted qualities smaller, making the relationship between movement, rate, and evolutionary/runtime behavior very complex. To keep whatever is remaining of the conditions uniform for one and two machines, all parameters were changed for the populace by transmission to the machines in equivalent extents. Therefore all machines kept up a populace of 1,000, so that introductory differences were the same. Further analyses will be made keeping populace consistent, but this is left for further study. At last, the Cloud space itself was utilized to check for end conditions: a document was made on the organizer showing when the experiment was finished. When the other junction read that record, it completed too; all nodes continued running until the result was discovered or until an extreme number of eras were reached. For this reason, in a few cases, results were not discovered; the amount of eras was registered for the reason that it was made to become easy to understand in a high number of cases to discover the result. The PCs utilized as a part of this examination were laptops connected by wireless, they were different models, and were running different working frameworks and forms of them. The most effective PC was the first; then #2 was the second-best, and numbers 3 and 4 were having minimum capabilities. Since machines run one by one without synchronization checkpoints, load adjusting is immediate, with additional capable workstations making more changes to the mix, and the less capable ones making less contribution. The foremost thing that was checked with the two issues examined (P-Peaks and MMDP) was if including more workstations influenced the result rate. For P-Peaks there was no distinction, freely of relocation rate and number of Pcs, all analyses discovered the result. Then again, there was a distinction for MMDP, demonstrated in Table 1. The development with the relocation rate can also be watched in Fig. 2; as was progressed in the presentation, the relationship is very difficult to understand and increment don't make a faster single-rate change of the victory rate. Indeed, the best rate relates to the very best relocation rate, the best success rate corresponds to the highest migration rate (migration after 100 generations), but the second best corresponds to the lowest one (migration after 450), is about much the same as no movement, since considering that eras do not run concurrently, this may imply that transient from different node is combined into the populace. This outcome is as per the transitional aggravation theory, demonstrated by us at one time [12]. Nonetheless, it is not clear that relocation in 100 eras could be really acknowledged moderate and in 200 too high. Therefore more trials will be needed to learn the ideal movement rate. Accordingly, having demonstrated that

Table 1 Success rate for the MMDP problem with different number of nodes and migration rates

Nodes	Generations between migration	Success rate
1	100	0.83
2		0.95
4		1
1	250	0.70
2		0.88
4		1
1	450	0.80
2		0.90
4		1

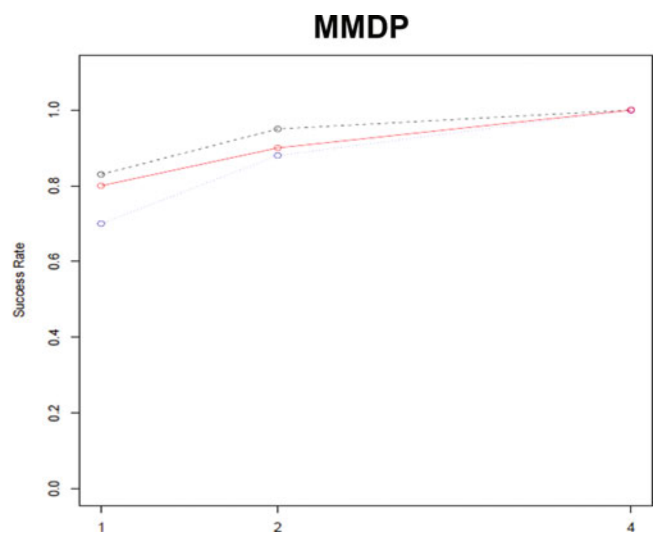


Fig. 2 Variation of the success rate in the MMDP

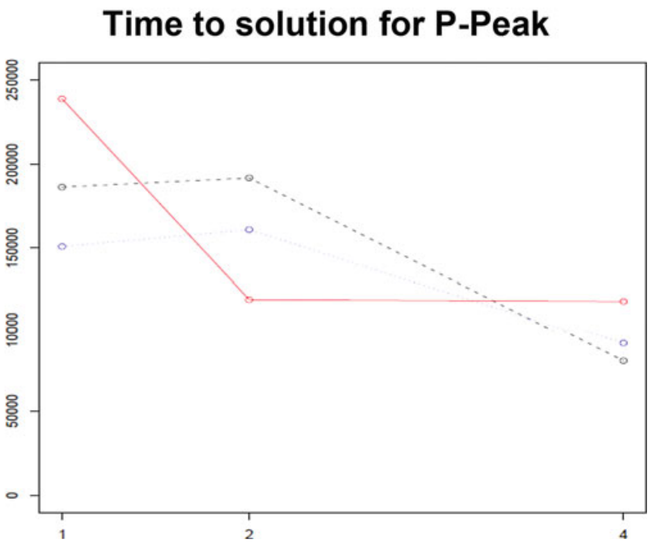


Fig. 3 Variation of the time needed to find the solution in the P-Peaks

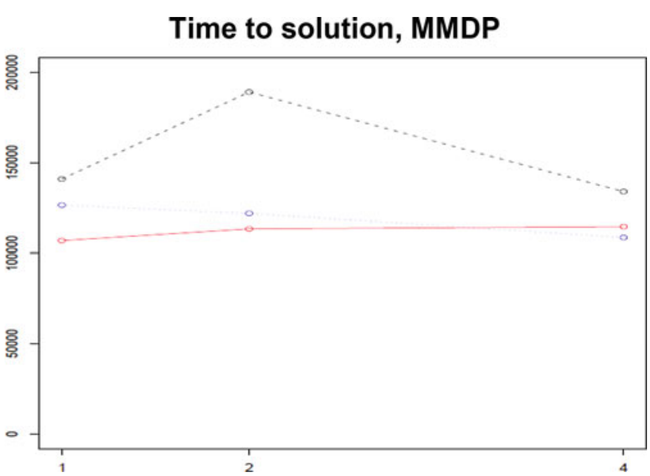


Fig. 4 Variation of the time needed to find the solution in MMDP with the number of nodes

victory rate builds with the amount of node, we will consider how exhibition differs with it. Does the calculation truly discover the result speedier at what point more node are included? We have registered time just for the analyses that really discovered the result, and plotted the outcomes in Figs. 3 and 4. As seen above on account of MMDP, there is not a straightforward relationship between the relocation rate and the time to result; that's why the relationship between the number of machines and time result is very much surprising. If we turn first toward the P-Peaks trial in 3 we see that we acquire small time change when adding more nodes to the mix. Since triumph rate is now 100 % with a solitary Pc, and the result takes around 2 min, the postponement by Cloud storage will show that it is not likely that the moved results are transmitted to the next node. Hence it is the middle of the road relocation rate (each 40 eras) the one and only that gets an enduring abatement of time to result. The best time is acquired for a solitary junction and a relocation crevice of 60; so, the best times are for the most elevated movement crevice as the formed postpone made by movement is also the minimum. These outcomes assume that there must be a sure level of many-sided quality in the issues to show the characteristics in nature's turf. For generally simple issues, which do not require many eras, there is nothing more to add. The scenario is very different for the MMDP, as seen in Fig. 4. Hence, the best outcome is acquired for four nodes and the smallest change crevice (each 100 eras, dashed dark line). It is interesting to watch that pattern change nodes in all cases, either the solution takes more time than for a single node or it takes less than for four nodes. The conclusion is that a wide number of concurrent assessments carried by the amount of node makes results quicker. However, a fine-tuning of the relocation crevice is wanted with a specific end goal to use the parallel assessment in the Cloud storage-based framework.

Conclusion

As a rule, and for complex issues like the MMDP, a Cloud storage-based framework might be arranged further bolstering take good fortune of the parallelization of the evolutionary calculation and get dependably (in a 100 % of the cases) results in less time than a solitary machine might. In addition, it has been demonstrated that it doesn't make a difference if the new Pcs added to the set are give or take effective than the first. By and large, then again, adding more Pcs to a set synchronized by means of Cloud storage has more impact in the triumph rate than in the time would have done well to discover the result, which appears to be harshly interfaced to the populace size, despite the fact that this theory will be tried tentatively. Then again, utilizing moderately straightforward issues like P-Peaks yields no sensible change, because of the deferral in relocation encroached by Cloud storage , which infers that this sort of method might be better left just for issues that are in the meantime challenging from the evolutionary outlook and likewise moderate to assess. numerous issues stay to be mulled over. In the first place, more exact exhibition measures must be taken to measure how the time would have been wise to discover the result in all events scales when new machines are included. We will research how parameter settings, for example populace measure and movement hole (time passed between two movements) impact these measures. This paper demonstrates that this impact is vital, however it is not clear what is the impact on the last outcome. It might be likewise fascinating to test distinctive movement approaches influence last come about, as done in, where it was figured out that relocating the best one may not be the best approach.

A vital issue too is the means by which to connect with Cloud storage for the purpose that qualified information is dispersed optimally and with a negligible dormancy. Thus we needed to stop every junction for a certain time (which was heuristically discovered to be 1 s) to leave time for the Cloud storage daemon to circulate records. In a trial that keeps up for less than 2 min, this can consume 25 % of the aggregate time (for every junction), bringing about an evident drag in exhibition that can take numerous extra node to repay. A deeper examination of the Cloud storage API and a fine-tuning of these parameters will be finished with a specific end goal to settle that. At last this structure opens numerous new potential outcomes for dispersed evolutionary processing: meta-evolutionary reckoning, manufactured life reproductions, and enormous scale recreation utilizing hundreds or even many customers. The sort of issues suitable for this, and also the configuration and usage issues, will be investigated. Other mist space results, rather incorporating open source executions, will be likewise tried. Since they have distinctive models (synchronization daemon or client mounted file systems, fundamentally) idleness and different characteristics will be totally diverse.

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Information Retrieval with the Use of Music Clustering by Directions Algorithm

Adam L. Kaczmarek

Abstract

This paper introduces the Music Clustering by Directions (MCBD) algorithm. The algorithm is designed to support users of query by humming systems in formulating queries. This kind of systems makes it possible to retrieve songs and tunes on the basis of a melody recorded by the user. The Music Clustering by Directions algorithm is a kind of an interactive query expansion method. On the basis of query, the algorithm provides suggestions that may be used to enhance the query. The MCBD algorithm is bases on the Clustering by Directions (CBD) that was designed to support users of web search engines in information retrieval. The MCBD algorithm is used for retrieval of music documents similarly as the CBD algorithm is used with text documents. This paper also introduces a new kind of interface designed to perform interactive query expansion in query by humming systems. The interface is a kind of tag cloud applied to music information retrieval systems.

Keywords

Music Information Retrieval • Query by Humming • Query Expansion

Introduction

Searching for a song is quite easy when a person knows the title of the song and its author. The search becomes more complicated when someone remembers only a part of a melody which occurs in that song. There have been many attempts to create search algorithms which would make it possible to find music on the basis of a sampled recording of a user singing a part of a tune. Systems that use this kind of algorithms are called query by humming (QbH). It is a kind of content based music retrieval method.

One of the main problems in searching for data and information is forming a right query. Search engines designed to search for web pages and documents in multiple ways facilitates user in formulating queries. Popular web search engines provide dictionaries that tells the user when she or he makes a typo or other language error. Apart from that, web search engines use real time query expansion technique suggesting how to finish a query while a user is typing it. Search engines provide also related search queries and similar queries to the one inserted by a user.

Search engines designed to search for songs on the basis of a user's recording of a tune support users in forming queries in a very low extent. However, it would be possible to develop and implement this kind of techniques. They could be available in the search of internet, internet stores and digital libraries. This paper addresses this problem.

This paper introduces a novel algorithm designed to support users in the search for music on the basis of sound samples. The algorithm is called Music Clustering by Directions algorithm (MCBD). It is bases on Clustering by Directions algorithm (CBD), that was also designed by the author of the paper [1, 2]. The CBD algorithm was intended for use in the search for web pages and documents in the Internet. The algorithm is designed to provide an interactive query expansion technique. When user runs a query, the

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_22, © Springer International Publishing Switzerland 2015

CBD algorithm presents suggestions on improving the query and making it more explicit.

This paper presents application of the Clustering by Directions algorithm to the search for tunes, music samples and various kinds of music documents. The algorithm originally processed text files, however as this paper shows it can also be used to process music files.

Related Work

Music Search Engines

There is a large number of query by humming systems. There is also a lot of methods that are used in these systems to provide users with a list of songs relevant to their query. Kotsifakos et al. presented a survey on query by humming methods [3]. They provided a paper with 51 bibliography items describing methods used in query by humming systems. The main problem in query by humming method is the development of a similarity measure which would precisely determine whether songs are relevant to user's humming query or not. Various techniques are used. The most popular takes advantage of dynamic programming [4, 5] and Hidden Markow Models (HMM) [6, 7].

There are many variations of dynamic programming in the area of query by humming systems. One of commonly used is Dynamic Time Warping (DTW) [4]. DTW is used to determine the similarity level between two sequences of sounds. In DTW, a sequence of points in the first melody is associated with a corresponding sequence of points found in the other melody. It is crucial that the time intervals between points in the first melody may differ from time intervals between matching points in the second melody [5].

In comparison with DTW there is some starting point of two sequences. The comparison between two melodies is not realized as a comparison between values of pitches in subsequent points in time. For some point in time in the first sequence a corresponding point in the second sequence is determined. For these points there may be different amount of time which elapsed since the beginning of sequences. Algorithms based on DTW search for corresponding pitches near the point in time of a pitch from the first sequence however it does not need to be the same point of time. The disadvantage of Dynamic Time Warping algorithms is that they have square computational complexity. Thus, these algorithms are rather slow.

Algorithms based on Hidden Markow Models is another class of algorithms designed to realize query by humming systems [6, 7]. It is a very popular model that is used to process uncertain and biased data like a melody recorded by a user. HMM is based on calculating probabilities. Shifrin et al. used HMM and they treated user's query as an observation sequence [6]. They estimated likelihood that such a sequence would be generated on the basis of some song available in the database they used. Unal et al. presented another influential paper on application of HMM to query by humming systems [7]. They used representative human humming data to train their recognition system. They also segmented recordings into notes with the use of HMM.

Apart from dynamic programming and HMM there are also other method used to match humming queries with songs in database. In particular, a concept of n-grams is used [8]. This method is mainly used in string and text processing. N-grams are sequences of some kind of data of a constant size. In the field of query by humming systems, n-grams can be sequences of sounds or notes. A sequence contains a certain number of data. For example such a sequence can consists of four sounds. The set of these sequences i.e. n-grams is used to find similarities between queries and dataset. Dannenberg and Hu described a QbH retrieval system that took advantage of n-grams [8].

Some QbH systems are limited to certain data format of music files. Francu and Nevill-Manning presented the entire structure of a digital library of music with a QbH system. They experimented on MIDI files. Their retrieval system was based on matching contours of tunes. Melodic contour is the sequence of relative differences in pitch between successive notes [9]. Melodic contours are also widely used in other music retrieval systems [4].

There is a large number of Query by Humming systems available for users of Internet. This method of information retrieval is also supported by multimedia content description standard MPEG-7 [10]. This standard includes descriptors that facilitate music retrieval in QbH systems.

Query Expansion

Query expansion techniques are commonly used in the field of web search engines. Query expansion are methods for supporting user in forming queries and acquiring search results that better match users needs. Query expansion methods provide users with additional data that they may use in creating their queries.

Real time query expansion algorithms [11] present a list of suggested queries while users are typing their query. This list of queries is usually created on the basis of statistics concerning queries inserted by various users. There are also method that presents their result after the search for a given query is completed. This kind of methods include interactive query expansion (IQE) techniques and other methods presenting queries related to the one provided by the user [12]. Interactive query expansion algorithms provide users with terms and phrased related to their queries. Users can use these data to make a subsequent query which would more precisely specify the type of data that they are looking for. Terms in IQE and related queries can be generated on the basis of different queries inputted by other users, like in real time query expansion methods. These data can be also generated on the basis of dictionaries [13]. It is also possible to acquire terms and phrases related to user's query on the basis of web pages presented as a result for a given query.

Presenting a tag cloud is a quite popular method for supporting users in searching and forming queries [14]. A tag cloud contains popular words and phrased searched by other uses. These words are displayed with different font sizes. The size of the font indicates the importance of words. More popular and more important concepts are displayed in larger fonts.

There were also attempts to apply query expansion methods in query by humming systems. Su et al. proposed a query expansion method for query by humming systems based on relevance feedback [15]. Relevance feedback is a method used for improving search results when a user has a list of search results acquired for a given query. Users can mark items as relevant or not relevant to their expectations in this list. The search is then performed again and the search engine takes into account user's selections.

Tseng proposed another method [16]. He extended the query by humming system functionality with identification of melodies similar to the one inputted by a user as a query. This function works similarly as a dictionary in web search engines. A user may have recorded a tune with some falsehood. This tune may be similar to other melodies or fragment of melodies used often in songs. The system proposed by Tseng provided a user with melodies similar to the one which was in the user's query.

In general query expansion methods are rarely used in query by humming systems. Music Clustering by Directions algorithm presented in this paper is a new kind of query expansion method which introduces a novel form of supporting users in formulating humming queries. The algorithm can be widely used in query by humming systems.

Clustering by Directions Algorithm

The Music Clustering by Direction algorithm is based on the Clustering by Direction (CBD) algorithm [1, 2]. CBD is design to perform interactive query expansion in web search engines. In order to take advantage of CBD, a user needs to start searching by inserting a query into a search engine. The search engine performs web search for the given query. On the basis of user's query and search result, the CBD algorithm determines suggestions that are relevant to the user's query. Users may use these suggestions to expand their queries. It is up to the user how to modify the query to make it better. The purpose of the CBD algorithm is to present a set of different possibilities.

The results of the algorithm are presented in a form of a tag cloud which was designed particularly for the CBD algorithm. In a tag cloud words are located in radial arrangement.

Clustering by Direction consists of the following steps:

1. Calculation of documents representations
2. Calculation of distances between representations
3. Selection of different directions
4. Assignment of documents to directions and selection of representative words
5. Presentation of terms on user interface

Calculating Representations

The CBD algorithm uses Vector Space Model [17]. It is a popular method of representing text documents. Documents written in natural language are hard to process by computer algorithms. In information retrieval systems, it is required to determine the subject of the document and topics mentioned in it. It can be realized with the use of VSM.

In VSM, documents are represented by a set of pairs consisting of a word and a weight. The set contains most of words used in the document, but some words are omitted e.g. those which occur in documents very often. Weights correspond to the importance of words in document. They are calculated on the basis of the number of occurrences of a word in the document and the number of documents that contain that word. A weight is higher if there are more occurrences of the word in the document for which the weight is calculated. Weights are also higher for words which occur in smaller number of documents. Such words are considered more important as few documents contain them.

The method of calculating weights in CBD algorithm differs from the method used in the original VSM. In CBD, every document is represented by a set of pairs: $\{(t_1, w_1), (t_2, w_2), \ldots, (t_n, w_n)\}$ where t_k is a word and w_k is a weight between 0 and 1. The weights are calculated according to (1).

$$w_k = \left(\frac{p_k}{\log(Q_k)}\right) / \left(\sqrt{\sum_{i=1}^{n}\left(\frac{p_i}{\log(Q_i)}\right)^2}\right) \tag{1}$$

where p_k is equal to the number of occurrences of the word t_k in the document, and Q_k is the frequency of occurrences of word t_k in language.

Calculating Distances Between Representations

The set of pairs can be interpreted as vectors in multidimensional space that have the number of dimensions equal to the number of considered words. Computer algorithms can easily compare such representations of documents and determine which of them are similar to each other. It is also possible to determine the distance between representations in space. It is used for finding documents in information retrieval systems. Document are similar when a representation of one of these documents is located in multidimensional space near the representation of the other document. Different kinds of metrics can be used to calculate distances. In CBD, like in the original VSM, a cosine coefficient is used [17] presented in (2).

$$dist(v_a, v_b) = 1 - \left(\sum_{i=1}^{n} w_{ai} w_{bi}\right) / \left(\sqrt{\sum_{i=1}^{n} w_{ai}^2} \sqrt{\sum_{i=1}^{n} w_{bi}^2}\right) \quad (2)$$

where v_a and v_b are representations and w indicates words.

Selecting Directions

Selecting different directions is the most important part of the CBD algorithm. The algorithm first selects different directions in the multidimensional space and then it determines what kinds of documents are located in space in these directions. This way of selecting directions was design to fulfill the purpose of the algorithm which is showing users various possibilities of modifying queries.

The main problem in this step of the algorithm is the high number of dimensions in the space where representations are located. Selecting different directions in a space with small number of directions is much simpler. The space in VSM has number of dimensions equal to the number of words in language apart from a small group of excluded words. Directions may point out exactly the same representations of documents, although they may seem different.

A special method for selection different directions was developed in Clustering by Directions algorithm [1, 2]. Directions are selected on the basis of representations of documents. First, the number of desired directions, denoted D, is selected. Then, a subset containing D representations is selected from a set of all representations in such a way that the sum of distances between every two representation in this subset is greater than the sum of distances between representations in any other subset with D elements. The representations from the selected subset point out different directions.

A sum of distances between every two representations in this subset can be calculated for every subset with D elements. This sum is equal to the value presented in (3). The selected subset is the one, for which the value defined by (3) is the greatest.

$$f(v_{i_1}, \ldots, v_{i_D}) = \sum_{v_{i_p} \in \{v_{i_1}, \ldots, v_{i_D}\}} \; \sum_{v_{i_r} \in \{v_{i_1}, \ldots, v_{i_p}\}} dist(v_{i_p}, v_{i_r}) \quad (3)$$

where $v_{i1}, \ldots, v_{iD}$ are representations included in a subset, and function $dist(v_{ip}, v_{ir})$ stands for the distance between representations.

Assigning Representations to Clusters

When directions are selected, the algorithm assigns representations to directions. A cluster is made for each direction containing representations located in space near the representation pointing out the direction. A constant number of representations is assigned to each direction. This number is equal to the number of analyzed representations divided by the number of all analyzed representations.

Representative terms are selected from each cluster. In order to select these terms, all representations included in the cluster are summed up. These representations are in fact pairs of words and weight. The sum of these pairs is also a kind of pair. Words with the greatest value of weight in these pairs calculated for clusters are words representative for a cluster. The same number of representative words is selected from every cluster. These words are presented in users interface.

Presenting Results on User Interface

The results are presented in a special form of a tag cloud. The arrangement of words in this tag cloud is such that they are located along lines from the center of the tag cloud to its borders. Words along each line are those selected form the same cluster. Words with greater weight are displayed with a larger font. A user can use these suggested words in order to improve his or her query. A sample tag cloud generated by CBD algorithm for query *car* is presented in Fig. 1.

Music Clustering by Directions Algorithm

The Clustering by Directions algorithm may be applied to support users of query by humming systems. The version of the CBD algorithm designed for music files and data was designed by the author of this paper. This is algorithm was called Music Clustering by Directions algorithm.

The MCBD algorithm support users in creating their humming queries similarly to the CBD algorithm supporting users in forming web search queries. The procedure of using MCBD algorithm is such that first a user creates his or her humming query. This query to some extend specifies the kind of music a user is looking for. The query by humming system retrieves songs that match the query prepared by the user.

Next, the MCBD algorithm calculates suggestions for the user on the basis of the user's query and a list of retrieved songs. These suggestions are fragments of melodies that the

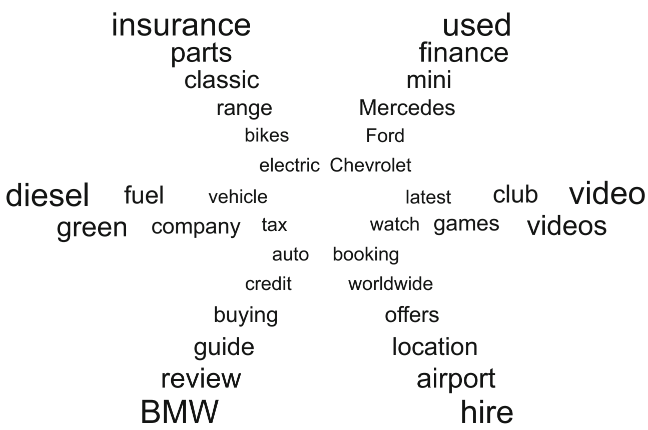

Fig. 1 Tag cloud in Clustering by Directions algorithm

Table 1 Comparison between CBD and MCBD algorithms

Data	The CBD algorithm	The MCBD algorithm
Data set	Web pages, text documents	Music files, sound midi
Data source	Web search results	Results in query by humming systems
Query	Text	Sound recorder by a user
Results	Suggested words	Suggested parts of melodies
Interface	Radial tag cloud	Music radial tag cloud

user may include in the query to improve it. It is up to the user to choose which tunes he or she will include in the renewed query. The purpose of the MCBD algorithm is to show possibilities. The results of MCBD algorithm are presented in a novel form of a musical tag cloud designed by the author of this paper. This kind of interface is described in section "The Interface and Results".

There are major differences in the type of data that are processes by CBD algorithm and MCBD algorithm as presented in Table 1. The Clustering by Directions algorithm process text i.e. web pages and documents. The algorithm generates suggestions for users on how to improve their queries. These queries consist of words. The Music Clustering by Directions algorithm operates on sound. The suggestions for the user are fragments of melodies.

There is also a difference in the source of data on which algorithms CBD and MCBD operate. There are two sources of data for both of these algorithms: the user's query and search results. In case of CBD, the search result is a list of web pages found by a search engine. In case of MCBD, the search result is a list of songs retrieved by a query by humming system on the basis of user's query.

There are multiple operations that are performed on these music files by the MCBD algorithm. Processing music files consists of the following steps:

1. Extraction of the foreground melody
2. Rounding the duration of sounds
3. Creating n-grams of sounds
4. N-grams unification
5. Creating representations based on n-grams

These operations are described below.

Extraction of the Foreground Melody

A song is a combination of many musical elements. There are rhythm, drums, vocal, background instruments, etc. Query by humming systems takes advantage of various data to retrieve music [4, 6]. Nevertheless, the most important is the melody of a song. Regardless if it is a melody of a vocal or some leading instrument most of songs have a main, foreground melody. The users' query is matched with this tune. The main melody is also the basis for generating search suggestions by the MCBD algorithm.

The collection of music needs to be preprocessed in order to retrieve the main melody before the Music Clustering by Directions algorithm is applied. For every song there need to be data on musical notes that comprise the main melody. The main melody needs to be saved in a form that makes it possible to determine every tone of it. In particular, the main melody can be saved in a form of a midi file, where one of voices corresponds to it.

These data are in fact a metadata of a song containing additional information. The data in that form can be easily processed by computer algorithms, unlike data in a form of a complete song that may contain a large number of various sounds. The MCBD algorithm does not interfere with the method of extracting the main melody. It can be retrieved manually or automatically on the basis of sound in the song. The melody can be also acquired on the basis of musical notation of a song [18]. It is also possible that these required data are already embedded in the file containing a song.

Rounding the Duration of Sounds

Sounds have different duration. Apart from differences in the sense of whole notes, half notes etc, musical notes are in fact played with different physical duration. Moreover queries in humming systems will probably not be precise. Users will not sing parts of songs precisely enough to make it possible to detect songs on the basis of small changes in the duration of sounds. In MCBD, the duration of sounds is rounded in different tunes in order to analyze the same kind of music sequences.

Let us suppose that there is some reference sound duration equal to T. It may by an exact duration of a sound denoted by a whole note. When there is also some sound in the song with duration equal to t then this duration is rounded according to T. The value of the rounded duration is presented in (4).

$$t' = \begin{cases} \frac{1}{2^n}T & \text{if } \frac{1}{2^n}T \le \mathrm{t} < \left(\frac{1}{2^n} + \frac{1}{2^{n+2}}\right)T \\ \frac{3}{2^{n+1}}T & \text{if } \left(\frac{1}{2^n} + \frac{1}{2^{n+2}}\right) \le \mathrm{t} < \left(\frac{1}{2^{n-1}} - \frac{1}{2^{n+2}}\right)T \\ \frac{1}{2^{n-1}} & \text{if } \left(\frac{1}{2^{n-1}} - \frac{1}{2^{n+2}}\right)T \le \mathrm{t} < \frac{1}{2^{n-1}}T \end{cases} \quad (4)$$

where T is the exact duration of a whole note, t is the real duration of a sound in a song, t' is the rounded duration and n is an integer value.

For example a rounded value of sound duration can be equal to 1/4T, 3/8T, 1/2T, 3/4T, T etc. This kind of rounding sound duration values makes possible to determine in MCBD algorithm similar sequences of sounds in various sounds and users' queries.

N-Grams of Sounds

The next step of the MCBD algorithm is dividing the melody into n-grams consisting of subsequent sounds. Sequences of three sounds are used in MCBD. These sequences overlap with each other. For every sound, there is an n-gram sequence that starts at this sound. For example, for a tune presented in Fig. 2a, there are three n grams presented in Fig. 2b–d.

N-Grams Unification

Different songs may be composed with different tonality. Moreover different keys and tones may occur in n-grams acquired in the previous step of the algorithm. In the MCBD algorithm tones in n-grams are unified with respect to the first sound in every n-gram.

The pitches of sounds are altered in such a way that the first sound in every n-gram is at the same pitch. Other sounds in n-grams are changed to lower or higher respectively to the change of the first sound of an n gram. It is performed in such a way that differences between subsequent pitches in n-gram remains the same. Figure 3 presents n-grams from Fig. 2 after this kind of unification.

The unification of n-grams is a process similar to making a melodic contour of a tune [9]. However contours are often concerned only with information whether subsequent tone is higher, the same or lower than the previous one. Data about the scale of differences in pitches is neglected. In the MCBD algorithm this information is taken into account.

Fig. 2 Dividing melody to n-grams. (**a**) Melody, (**b–d**) n-grams

Fig. 3 N-grams after tonality unification

Creating Representations Based on n-Grams

After determining n-grams that occur in songs, the MCBD algorithm creates representations of music files. These representations are located in a space similarly to the one used in VSM. In VSM, every dimension corresponds to a different word. In the space used in the MCBD algorithm, every dimension corresponds to a different n-gram. These n-grams are regarded as words.

In order to create representations of songs for every song, the MCBD algorithm counts the number of occurrences of each n gram. This creates a histogram of n grams. The representation of a song is a set of pairs which consists of an n gram and the number of its occurrences in the song. In order to determine distance between representations of different songs the cosine coefficient is used similarly as in CBD.

Selecting Directions and Creating Clusters

Next steps of the Music Clustering by Directions algorithm are the same as in the Clustering by Directions algorithm. The representations based on n-grams have the same function in VSM as representations based on words. The MCBD algorithm selects directions, assigns representations to clusters and select n-grams representative for each cluster in the same way as the CBD algorithm selects representative words.

The Interface and Results

The interface used in MCBD has a novel form of tag cloud containing fragments of musical notation which visualize n grams selected from directions. N-grams with greater weight

Fig. 4 Music tag cloud generated with the use of the Music Clustering by Directions

are showed larger than others. The location of n-grams is similar to the location of words in the interface of CBD algorithm. The concept of the MCBD interface is such that when a user can click on such a fragment of musical notation the melody is played. If a user finds a melody relevant to his or her needs, it can be added to user's query in order to make it more precise.

The interface is presented in Fig. 4. The results presented in this figure were generated for query with one note C in a collection which returned as a result a list of 44 songs recorded by famous ABBA music group. The tag cloud was generated for six directions. Two representative n-grams were selected from each direction. All of fragments of melodies included in this tag cloud are related to users' query and search results as all of them are parts of songs included in search results.

Conclusion

The main purpose of the MCBD algorithm is to show possibilities of modifying humming queries. The interface in a form of tag cloud reached high popularity. The presented kind of interface has also potential to become popular in query by humming systems. Further work in this subject will be focused in performing large scale experiments with the proposed method.

Acknowledgment This work was supported by the National Science Center grant no. DEC1-2011/01/B/ST6/06500.

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Using SQL Queries to Evaluate the Design of SQL Databases

Erki Eessaar and Janina Voronova

Abstract

The system catalog of a database with explicit schemas contains among other things information about the structure of the database. Queries based on the system catalog allow us to search occurrences of database design antipatterns (database design flaws). In this paper, we present the results of an evaluation of a set of SQL databases. We used the queries that were presented in the previous paper on this topic. A goal of the research is to further experimentally evaluate the queries. We present findings about the queries as well as evaluated databases. In addition, we propose more questions about the design of conceptual schemas of SQL databases that can be answered by querying their system catalogs. The use of the queries would allow us to partially automate the process of evaluating structure and constraints of existing databases and detecting design flaws.

Keywords

Antipattern • Database design • Metadata • SQL • System catalog

Introduction

There is a persistent problem that many existing databases have various design flaws. These make it more difficult to maintain and extend the databases as well as develop, maintain, and extend applications and utilities that use the databases. Catalogs of database design patterns or antipatterns that describe good and bad/questionable database design practices, respectively, help the database design community to systemize the information about design practices, teach these more effectively, and take steps towards automating the detection and resolving of database design problems.

The previous paper on this topic [1] proposed a set of SQL queries for detecting *possible* occurrences (instances) of SQL database design antipatterns in the existing SQL databases. Each such antipattern describes violations of some SQL database design principle/heuristic. The paper analyzed 12 database design antipatterns [2] and proposed in total 14 queries for 11 antipatterns. Eight of the analyzed antipatterns were about logical design and four were about physical design. The queries are based on the Information Schema views that are themselves created based on the base tables of the system catalog. We are interested in queries based on the Information Schema because the structure of the views in the *information_schema* schema has been standardized by the SQL standard [3] and hence there is a better chance that the queries are usable in case of different database management systems (DBMSs). PostgreSQL™ 9.2 [4] was used as the basis to write and test the queries because PostgreSQL provides the *information_schema* and we have experience in using the DBMS. The query statements [1] can be found from the file: http://staff.ttu.ee/~eessaar/files/Design_flaws_queries.pdf.

According to the ANSI-SPARC architecture, a database has one or more external views, one conceptual view, and one internal view [5]. All these are described by means of

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_23, © Springer International Publishing Switzerland 2015

schemas. The SQL standard [6] specifies database objects that constitute external and conceptual schemas. Hence, it would be possible to use the Information Schema views to find answers to the questions about the external and conceptual schemas of databases. The antipatterns [2] and their detection queries [1] deal with the design problems of conceptual schemas.

The first goal of the paper is to experimentally evaluate the antipattern-detection queries [1] by using them to evaluate a set of existing databases. This was not extensively done in the previous paper [1] due to space restrictions. *The second goal* of the paper is to propose additional questions about the design of conceptual schemas of SQL databases that can be answered by querying their system catalogs. *The third goal* is to discuss possible application areas of the queries.

Reference [1] and the current paper offer different but complementary approach for evaluating the quality of the conceptual schema of an existing database compared to a previous paper [7] on this topic. Reference [7] suggested a semiotic approach for using a conceptual data model of a database to evaluate the *semantic quality* of its conceptual schema. In the present approach, we do not use conceptual/logical/physical data models but rely on the specification of the database schemas that are stored internally by a DBMS as well as actual data that has been registered based on the conceptual schema. Queries based on the system catalog allow developers to better comprehend the schemas of a database and as such are tools to improve the *pragmatic quality* of the database schemas. By using some of the queries one can identify missing constraints or places where a declarative approach for enforcing constraints can and should be used instead of an imperative approach. Declaratively enforced constraints in databases are the means through which DBMSs get information about the semantics of data in the databases and are also able reflect this information to the users and evaluators of the conceptual schemas. The results of the queries are a basis to improve the semantic quality of the conceptual schema so that there could be a better correspondence between the stakeholders' knowledge about the domain of the database and its conceptual schema. It could increase the *perceived completeness* [7] of the conceptual schema and does not reduce the *perceived validity* [7] of the schema.

The rest of the paper is organized as follows. Firstly, we explain the experiment for investigating existing databases. Secondly, we offer the main findings about the queries and the databases. Thirdly, we present further questions about the design of the conceptual schemas of SQL databases that can be answered by querying their system catalogs. After that, we discuss possible application areas of the queries. Finally, we conclude and point to the future work with the current topic.

A Research Experiment to Evaluate the Antipattern-Detection Queries

In [1] it was already shown that the queries can produce false-positive and false-negative results and hence their results point to *possible* problems in schemas rather than confirm with certainty that there is/is not a design problem. Because the use of the queries does not eliminate the need of experts to review the schemas, we are especially interested in the *frequency* and *reasons of false-positives* in the query results because these clutter the results and decrease the usability of the queries.

We needed databases for the experiment. The first author teaches in the Tallinn University of Technology (Estonia) courses about database design and programming mostly to the bachelor students of informatics and business information technology. Students have to create database projects and finally implement the database by using a server SQL DBMS. One of the possible server SQL DBMSs that students can use is PostgreSQL™ (another is Oracle Database™). Students have a freedom to select the information system for that to design a database. We used a set of students' databases for the experiment. We did not want to evaluate only the best or only the worst presented databases and hence, we evaluated all the PostgreSQL™ databases that were presented at the end of the 2012 fall semester. In total, we evaluated 41 students' databases. Forty-nine projects were presented in the period and only eight projects did not use PostgreSQL™ at all.

The use of student projects has its limitations. The size of the databases was quite small (average number of project-related base tables in the databases was 10.9). The total number of evaluated base tables was 447. During the design process the databases were repeatedly manually reviewed by the academic and many of the design problems were resolved as a result of that. During the study process many of the design problems and their solutions referenced in [2] were explained. The importance of integrity constraints and advantages of enforcing constraints declaratively was stressed and the use of constraints was also required in the project. Then again, the used CASE tool (IBM Rational Rose™ [8]) does not support automatic detection of database design problems. In addition, some students do not pay enough attention to the talk of academics and requirements of projects or misunderstand these. Hence, there is still a possibility that their database designs have problems. Regardless of limitations, the databases allow us to evaluate how many false-positives the queries produce and what are the reasons of the false-positive results.

The system catalog of a SQL database is based on the underlying data model of SQL. Hence, the queries can be used to evaluate the base tables of the system catalog.

We used the queries to evaluate the schemas of PostgreSQL™ 9.2 system catalog. There are 50 and 7 base tables that are in the schemas *pg_catalog* and *information_schema*, respectively. For this task we had to adjust the queries to find information about the schemas other than *public* that have the owner *postgres*.

Of course, one can execute queries one by one in a database by using a general-purpose database management tool. However, it would be much more efficient to be able to select a database, select desired queries, and execute the selected queries based on the selected database. Therefore, we suggest a specialized software tool for executing the queries. An initial version of the tool was implemented for the PostgreSQL™ DBMS. The tool requires a separate PostgreSQL database. The base tables that the tool internally needs could be created based on the conceptual data model in Fig. 1.

It is possible to access and evaluate all the PostgreSQL databases on the server where the tool resides by using the tool. Users would have to log in by using the username/password of a database user who has enough privileges to query the databases that they want to investigate.

Categories and tests are unordered and ordered sets of queries, respectively. An example of a category is “Database design antipatterns”. An example of a test is “Checking of student projects in the course X”. Surely it would be possible to execute all the queries belonging to a category one after another based on the selected database. However, tests allow us to specify sets of queries that belong to different categories. It would be possible to select a database, select a test, and let the system to execute all the queries that belong to the test in the order specified in the test based on the selected database.

The Results of the Experiment

Observations About the Antipattern-Detection Queries

The numerical values in this section are based on the 41 students’ databases. There were 81 cases (triples of a database, an antipattern, and a query) where a query pointed to the presence of one or more occurrences of an antipattern in a database. However, after manual checking of the database specifications only 23 cases (28.4 %) turned out to include real occurrences of the antipattern. In eight cases (out of 23) the query result pointed to the real as well as false-positive occurrences of the antipattern.

In general, the queries are based on the existence or lack of database constraints, data in the base tables that design we evaluate, or names of database objects. Although many queries use a combination of these aspects, it is possible to state the main aspect for each query. In case of mainly constraint-based queries there were five cases that consisted only of false-positive results. In case of mainly data-based queries there were 19 cases that consisted only of false-positive results. In case of mainly name-based queries there were 34 cases that consisted only of false-positive results. There were two queries that resulted with the biggest number of false-positive results.

One of the queries takes into account data in the base tables that design we evaluate. It is used to detect the occurrences of antipattern “Format Comma-Separated Lists”. It detects implementations of multi-valued attributes by searching columns with the VARCHAR or TEXT type that contain values that themselves contain separation characters like “,” or “;”. The query detected possible occurrences of the antipattern in 20 databases. However, only in one of the databases there was a real occurrence of the antipattern. A main reason of the false-positive results is that columns contain textual data that represents objects with complex internal structure. This happened at least once in 18 databases. The main reason was that the identified column contains data about postal addresses. A heuristic is to ignore columns those names contain the word *address*. Another main reason of the false-positive results is that columns contain ordinary sentences like free-form comments. This happened at least once in 15 databases. A heuristic is to ignore columns those names contain the words *description* or *comment*. The third reason of false-positive results is that columns contain names that themselves contain separation characters. It occurred in two databases. In one case a column contained the names of election constituencies and in

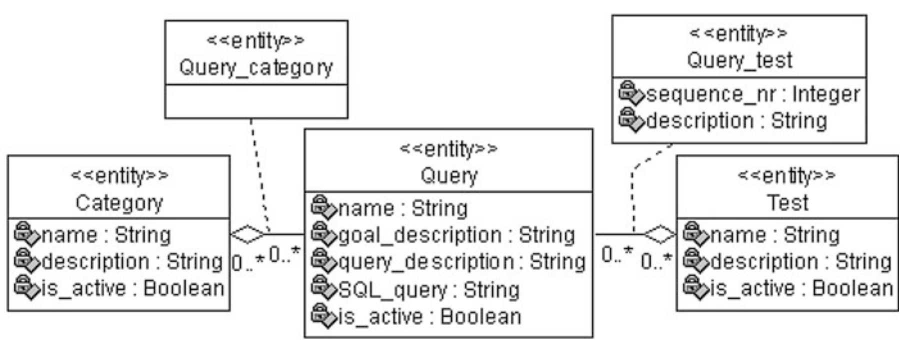

Fig. 1 Conceptual data model of a software system that can be used to execute queries for evaluating the design of a database

another case the names of diagnoses. A heuristic is to ignore columns with the names like *constituency*. The suggested sets of names can be modified if one gets more information about the naming practices in the databases that will be evaluated.

Another query that produced a lot of false-positive results takes into account the column names of the evaluated base tables. It is used to detect the occurrences of antipattern "Leave Out the Constraints". It searches missing referential constraints by finding pairs of columns of different base tables where the names and types of the columns are the same and there is no referential constraint that connects these columns. In each pair, at least one of the columns is the primary key column or a unique column of a base table. The query (query A) detected possible occurrences of the antipattern in 36 databases. However, only in six of the identified databases there were one or more real occurrences of the antipattern. In case of five of these databases there were also false-positive results. Another query (query B) for detecting the antipattern, which searches base tables that are not connected with any other base table through a referential constraint, detected the same occurrences as well as pointed towards two more missing referential constraints in another database. The query A did not find these because referencing (foreign key) and primary key columns had a different type. On the other hand, the query A detected one missing constraint that the query B did not detect because only one, not all/most of the constraints were missing.

One of the main reasons of the false-positive results is the implementation of a generalization hierarchy that occurs in a conceptual data model. The resulting base tables can contain columns with the same name and type if *class table inheritance* or *concrete table inheritance* [2] designs or the PostgreSQL-specific *table inheritance feature* [4] is used to implement the hierarchy. This happened at least once in 25 databases. However, such effect to the results of the query should not be the reason why to avoid these designs. It is difficult to find a good heuristic if the Information Schema does not provide explicit information about the existence of generalizations behind the base tables. Another reason of false-positive results is the existence of columns in different base tables that have the same name and type but are meant to record data with different semantics. This happened at least once in 24 databases. For instance, *Organization* and *Service* do not have a common supertype in the conceptual data model but both could have the attribute *name*. The third reason of false-positive results is the existence of multiple one-to-one or one-to-many relationships out from the same base table resulting with multiple base tables with referencing columns that have the same name and type. However, there should not be referential constraints between the columns. This happened at least once in 13 databases. A heuristic is to ignore pairs of columns that are both referencing columns and in case of both the referenced table is the same base table.

The suggested heuristics would decrease false-positive results but would increase false-negative results.

To summarize, different queries have a different probability of false-positive results. Specialized software for executing the queries could display the level of confidence that the results do not contain false-positives. The confidence information could be stored in the query database (see Fig. 1) by using a classifier that characterizes the queries. Queries that probably cause more false-positives could be hidden from the less experienced users. Moreover, the possibility to sort the results of individual queries based on different columns would be useful in dealing with a large number of false-positive results.

There are many reasons why data in a particular column could resemble data that can be recorded in a column due to an occurrence of a database design antipattern. Declared constraints in a database help DBMSs as well as human investigators to understand the meaning of data in the database. If there are no declared constraints, then it is difficult for everyone to understand the meaning of data. In this case, incorrect data can be easily recorded in the database.

In natural language people can use different terms to refer to the same concept as well as use the same term to refer to different concepts. Finding good and expressive names to database objects is important and certainly simplifies understanding of the database schema. However, the names cannot be the only basis for understanding the semantics of data in the database.

The least susceptible to false-positive results are queries that are only based on the declarations of constraints in databases. However, in case of the query for detecting the occurrences of "Always Depend on One's Parent", we got false-positive results in two databases. It searches referential constraints where the referencing table and the referenced table are the same. PostgreSQL™ allows multiple referential constraints with the same name in the same schema if these constraints are associated with different base tables. In this case the query will give incorrect results because it cannot unambiguously determine the referencing and the referenced table. This information is not available through a single Information Schema view and we have to find the result by joining different views based on the names of referential constraints.

Observations About the Students' Databases

By using the queries, we detected at least one real occurrence of database design antipatterns in 13 databases (out of 41) (31.7 % of databases)—in total eight different antipatterns. The most frequently occurring antipattern was "Leave Out the Constraint" that means missing referential constraints. It manifested in seven databases. The following problems occurred at least once in two databases (not necessarily the same databases): list of permitted values in a column was specified in a check constraint (antipattern “Specify Values in the Column Definition”) and hierarchical structure was implemented in a base table by adding a foreign key that refers to a candidate key of the same table (antipattern “Always Depend on One’s Parent”). In addition, in one database *double precision* type was the declared type of columns (antipattern “Use FLOAT Data Type”) for recording quantities of products and quantities of money. The queries did not detect real occurrences of antipatterns “Assume You Must Use Files”, “Clone Columns”, “Use Generic Attribute Table”, and “Use Dual Purpose Foreign Key” in any evaluated database.

Observations About the System Catalog of PostgreSQL

By using the queries, we found *possible* occurrences of six antipatterns in the PostgreSQL™ 9.2 system catalog [4].

“Clone Columns”: Base table *pg_statistic* of schema *pg_catalog* that stores statistical data about the contents of the database has columns *stakind1*. . .*stakind5* (values “indicate the kind of statistics stored in the *N*th “slot”” [4]), *staop1*. . .*staop5*, *stanumbers1*. . .*stanumbers5*, and *stavalues1*. . .*stavalues5*.

“Format Comma-Separated Lists”: Base table *pg_aggregate* of schema *pg_catalog* that stores information about aggregate functions has column *agginitval* where each value is the external string representation of the initial value of the transition state. These values consist of comma-separated components. Although this is not an occurrence of the antipattern, the query also reveals that *pg_ts_dict* base table of *pg_catalog*, which contains entries defining text search dictionaries, has column *dictinitoption* that contains information about both language and stop words. These values could be put into different columns.

“Leave Out the Constraints”: Referential constraints are not enforced in the base tables of the system catalog. Poorly tested DBMS modules could violate referential integrity. In addition, reverse engineering of the system catalog does not reveal relationships.

“Specify Values in the Column Definition”: Base tables *sql_features*, *sql_packages*, and *sql_parts* of *information_schema* contain column *is_supported* that is based on domain *yes_or_no*. It uses varchar(3) as the base type and has a check constraint that determines possible values “yes” and “no”. PostgreSQL™ supports type *Boolean* that could be used as the base type of *yes_or_no*.

“Use a Generic Attribute Table”: Base table *pg_largeobject* of schema *pg_catalog* for holding “large objects” resembles the key-value structure of the antipattern. The design is caused by the internal design of the DBMS. Table *pg_attribute* of schema *pg_catalog* contains columns *attoptions* and *attfdwoptions* for recording key-value pairs. The query did not refer to the columns but referred to the table due to its name.

“Use FLOAT Data Type”: There are in total four base tables in *pg_catalog* that have in total six columns that have type *real*. The columns could have *decimal* type and column *reltuples* of *pg_class* could in our view even have type *bigint*.

PostgreSQL uses OIDs (Object Identifiers) as the primary keys of system tables [4]. It is an occurrence of “One Size Fits All” antipattern. However, the corresponding query did not reveal that because data about these primary key columns is not provided through the Information Schema.

Some Additional Questions About the Design of Conceptual Schemas of SQL Databases

If we want to find answers to questions about the internal schema of a database, then we have to query unstandardized base tables of the system catalog because the Information Schema does not provide this information. However, there are plenty of tools for advising developers how to change the internal schemas. Moreover, some DBMSs are capable of self-tuning internal schemas [9]. Unfortunately tools for verifying the conceptual schemas of databases are much less common. We are aware of Parasoft™ [10] that offers some commercial tools for the database conceptual schema verification.

Next, we present some additional questions about the conceptual schemas of SQL databases, the answers of which help us to characterize databases and possibly identify their design flaws. The questions can be answered by querying the system catalog. We have created all the queries in PostgreSQL™ 9.2 [4]. We do not present the query statements here due to space restrictions. We do not claim that it is the final list of useful questions and encourage more research in this field.

Which base tables are without any unique constraints and primary keys? The answer identifies base tables permitting duplicate rows that cause various practical problems [11].

The next four questions help us to identify base tables where some integrity constraints are *possibly* missing. The reason could be that some requirements have not been identified or that the requirements have been incompletely translated into database design. Each not null constraint is a check constraint. However, the queries that are created to answer these questions should not take into account not null constraints.

Which base tables are without any associated (directly or through domains) check constraints?

Which base tables have at least two columns of *date* or *timestamp* (with or without time zone) type and have no check constraints involving two or more of these columns? The columns mean that we want to record data about events or processes, which often have a certain order. Hence, the values in these columns must be in a certain order in case of each row of such a table. For instance, the end of a meeting cannot be earlier than the beginning of the meeting. Such constraints cannot be associated with a SQL domain.

Which base table columns are not referencing columns of any referential constraint and are without any associated (directly or through domains) check constraints?

Which base tables have a surrogate key as the primary key and do not have any additional unique constraints? In case of defining the surrogate primary key in a base table it is a common mistake to forget to declare existing natural keys.

What is the number of triggers in the database? Triggers can be used to enforce complex integrity rules. Hence, the number of triggers gives an indication about how many complex integrity rules have been enforced at the database level.

The following three questions help us to identify the extent of using NULLs in SQL base tables. NULLs could lead to wrong answers to certain queries [11].

Which referencing columns of base tables permit the use of NULLs? Among other things one has to check whether such columns are consistent with the conceptual data model.

What is the percentage of optional columns (that permit NULLs) in case of each base table?

Which optional base table columns do not have a default value for preventing NULLs in the database?

What are the pairs of base tables that have at least two columns with the same name and data type? The tables might violate the principle of orthogonal design [5] and hence might facilitate uncontrolled data redundancy over different tables.

In addition, queries based on the system catalog can calculate values of database metrics [12] *Table size*, *Depth of relational tree of a table T*, *Referential degree of a table T*, *Size of a schema*, *Depth of referential tree*, and *Referential degree*. All the table level measures are about base tables.

Reference [1] and the current paper are about evaluating the design of SQL databases with the help of queries based on their system catalog. However, one could use queries based on system catalog in case of DBMSs with different data models. The prerequisite is that it must be possible to explicitly define database (external, conceptual as well as internal) schemas at the database level so that information about these would be stored in the system catalog by the DBMS. If one uses NoSQL systems to create schemaless data stores [13], then one has to review application code to evaluate the implicit schema scattered amongst the code that accesses the data.

Possible Application Areas of the Queries

The queries facilitate partial automation of the evaluation of the quality of the conceptual schemas of existing databases. The bigger is the number of schema elements, the bigger advantage the queries offer. They would be helpful in the context of database evolution because it allows interested parties to quickly evaluate the design of a database and find out what, if any, database refactorings are needed. Such automation would of course be useful in case of large (in terms of schema size) legacy databases with no or limited documentation. The use of the queries would be helpful even if documentation does exist because (a) there could be discrepancies between the models and the implementation and evaluation of the actual database gives the best information about its possible design problems, (b) reverse engineering of the schemas by using a CASE tool is not necessarily needed, (c) agile modelling promotes the use of the simplest tools like whiteboards where the use of such automation at the model level would be impossible, and (d) not all CASE tools allow execution of queries based on models. However, if a CASE tool supports querying database design models, then many of the proposed queries can be translated to queries based on models. An advantage of queries based on an actual database is that they make it possible to take into account data in evaluated tables. However, missing constraints raise the possibility of incorrect data. An advantage of querying models is that mappings between analysis and design models can be used to find out whether the design models are *complete* and *valid* in terms of captured requirements to the system [7].

The queries help us to investigate the quality of the end product (database) instead of the quality of the production of the product (database design process). It is not the most effective way to improve the quality [14] but it is what we can do if we have a ready-made database. We should ensure that in the future development process tries to avoid mistakes that we discover in the existing databases. If we integrate the use of the queries (continuous feedback) to the development process, then it improves the process quality and helps developers to create end products (databases) with fewer design flaws.

The queries are also useful in the context of teaching/learning database design. A survey [15] showed that 80 % of surveyed academics who taught databases used practical work for assessment. It was the most widely used assessment type in case of databases. If the students can use a learning software environment with an intuitive and user-friendly user interface to execute the queries, then it increases the immediacy of feedback about their practical work and allows them to work at their own pace. Preferably the software should be web-based to avoid installation and configuration of the system by students and allow users to access the system regardless of their location and their computing resources. If a learning environment where the queries are executed would be able to store statistics about the query results, then it would give information about the frequency of database design problems. Academics can take this into account and adjust their ongoing and future courses accordingly. In addition, the statistics could reveal whether the work is based on trial and error or whether students consciously apply the design knowledge. A possibility to extend the software is to show information about better design solutions in case of identified possible design problems.

Moreover, if an academic has to review many projects or the results of smaller practical tasks, then the queries will give quick overview of answers and one can concentrate attention to the most problematic parts of the answers. If the number of students increases, then supporting systems like this would allow academics to avoid simplification/ reduction of assessed tasks. There are many systems for automatic assessment of assignments of imperative programming [16] or automatic assessment of assignments to write SQL statements [17]. We are currently not aware of systems for automating assessments of database design assignments. The system of queries cannot determine the grade automatically due to the possibility of false positive and false-negative results of queries. However, it would be a useful assistant to the academics who have to grade the work. In our view, involvement of academics in grading is not something to be abolished because it makes students sense human involvement in the assessment process and hence feel that their work effort is valued.

It is possible to find values of software measures by using queries. These queries would give numeric results that characterize schemas but are less useful in determining the quality of a schema or the grade of a student that should depend on the quality of the schema. For instance, Piattini et al. [18] conclude "that the number of foreign keys in a relational database schema is a solid indicator of its complexity". Increasing complexity reduces maintainability.

However, it could be that a database has been created by using the anchor modeling [19]. Although there are a large number of base tables in the sixth normal form, there are also views based on the based tables that are in a lower normal form, and the conceptual and external schemas have probably been generated based on an anchor model. In this case the maintainability of the schemas is quite good because, for instance, it is easy to extend the conceptual schema—one always has to add new tables instead of altering existing tables.

In the context of assessing students' projects such queries can be used to count the number of different types of database objects. The result would be used if there were requirements to the minimal number of different types of database objects (for instance, a database must contain at least ten base tables).

Conclusions

We presented the results of an experimental evaluation of queries for detecting possible occurrences of SQL database design antipatterns. We discovered various reasons why some of the queries would give false-positive results. Most susceptible are queries that apply heuristics based on the names of database objects or data in the evaluated base tables. Despite that, the queries were able to detect different problems in the evaluated databases and in our view they would be a useful tool in case of creating and managing databases as well as teaching/learning database design. Their use would not hinder implementation of innovative systems but allow developers to be more informed. For instance, the queries allowed us to quickly detect some possible design problems of the quite large PostgreSQL™ 9.2 system catalog. We also proposed more questions about database design that can be answered by querying the system catalogs of SQL databases.

Future work must contain more evaluations of the proposed queries, creation of audience-specific software systems for executing the queries, and identification of additional design problems that occurrences can be detected by querying the system catalogs of databases.

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Improving Higher-Order Learning and Critical Thinking Skills Using Virtual and Simulated Science Laboratory Experiments

Nicole Simon

Abstract
Laboratory experiences are a vital component within science education. This paper is a report on the findings of a study conducted on undergraduate level laboratory science courses. A causal-comparative quantitative study was conducted with 150 learners enrolled at a 2-year community college, to determine the effects of simulation laboratory experiments on Higher-Order Learning, Critical Thinking Skills, and Cognitive Load. The treatment population used simulated experiments, while the non-treatment sections performed traditional experiments. Comparisons were made using the Revised Two-Factor Study Process survey, Motivated Strategies for Learning Questionnaire, and the Scientific Attitude Inventory survey, using a Repeated Measures ANOVA test for treatment or non-treatment.

Keywords
Educational technology • Electronic learning • Physics education • Student experiments • Virtual laboratory

Introduction

Computer simulations and virtual laboratory science experiments have radically changed science education [1–3]. These laboratory experiments offer learners a more inquiry-based approach to science education following less constructed learning experiences that does not promote the prescribed outcomes of traditional science experiments [20]. As per the National Science Education Standards [4], Benchmarks for Science Literacy, the importance of inquiry-based science education throughout the United States is vital to progressive education, underscoring standards that address students devising scientific investigations incorporating Higher-Order Learning and promoting self-directed technology-centered learning. Traditional expository or teacher-centered labs echo, nearly verbatim, the prior learned topical scientific material. These labs require learners to conduct experiments with already known scientific outcomes, thus referred to as rote-memorization [5]. Laboratory goals of rote-based labs do not engage Critical Thinking in students [5] therefore; they do not always identify lab goals or regard the research process. Undergraduate lab science courses play a pivotal role in science curriculum, chiefly within the context of scientific inquiry ability and Critical Thinking [4, 6]. Virtual lab experiences, include science labs conducted via simulations, are intended to develop the skills vital for scientific research [7], as this modality allows for creating learning goals of Higher-Order Learning and Critical Thinking, not always possible in physical labs.

Higher-Order Learning and Cognitive Load

Higher-Order Learning Skills outlined in Bloom's Taxonomy [8] promote synthesis and evaluation of learned material. Critical Thinking Skills involve discerning rationale and

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_24, © Springer International Publishing Switzerland 2015

analysis of material for breadth of knowledge [8]. Research has shown that visual symbolization, such as animations, is crucial to conveying scientific concepts [9]. Educational technology has been designed to coordinate visual cues and auditory procedures that guide learners through the construct of deeper understanding without taxation often associated with non-science learners during labs [4, 6]. Increased Cognitive Load builds learners' capability to comprehend more material at once by relying upon prior information, by using working memory, in union with multimedia technology, to create a learning environment coupling computer use with educational tools. Emerging educational technology, including simulations and virtual labs, can play a pivotal role in science education, offering learners the chance to practice true research methods that model scientific inquiry.

The Study

Review Stage

The purpose of this quantitative quasi-experimental study was to determine how the use of inquiry-based, virtual, and simulated science laboratory experiments increase learner perception of depth of learning via Higher-Order Learning Skills (HOLS), Critical Thinking (CT) skills, and Cognitive Load (CL) among learner participants at a 2-year community college. The predetermined instruments include: the Motivated Strategies for Learning Questionnaire (MSLQ) [10], the Scientific Attitude Inventory (SAI II) [11], and the Revised Two-Factor Study Process Questionnaire (R-SPQ-2F) [12]. The statistical analysis was conducted as to explore the relationships between use of inquiry-based simulated and virtual laboratory experiments and learner perceived increases in Higher-Order Learning Skills, Critical Thinking Skills, and Cognitive Load ability. The intention of this study was to investigate the increases of HOLS and CT skills in online and virtual science laboratory experiments, as compared to traditional face-to-face laboratory experiments. This study served to identify which course instructional delivery method will attain: perceived increased HOLS, CT skills, and increase CL capability within virtual laboratory science experiments.

The learners were separated into two pre-existing groups based on random selection of course section. The analysis was aimed at comparing laboratory format pre/post-lab surveys with learner perceived increases in Higher-Order Learning Skills (dependent variable), Critical Thinking Skills (dependent variable), and Cognitive Load (dependent variable) with respect to the two learner groups being treated or controlled (independent variable) controlling for the mathematical and scientific backgrounds of the learners (covariate) and the possible effects of computer usage skills will performing the simulated labs (covariate). The surveys were administered at the beginning and at the mid-point of the course, and it will determine how learners perceive an improvement or increase in their Higher-Order Learning Skills in science due to completing the course laboratory experiments. The rationale for administering the post-lab survey at the mid-point of the course is based on attrition rates for non-majors in the STEM laboratory courses. By surveying at mid-semester, the researcher had a more heterogeneous participant population. Whereas at the end of the semester, the population would be more homogenous consisting of the academically stronger students, thus biasing the data by showing an unequal mix of learners throughout the course.

Results

The results of the presented study are based on data collected during an academic year. A total of 150 learners participated in the study. Each learner provided informed consent to the collection of this data. All data are anonymized as to avoid any personally identifiable information.

Higher-Order Learning Skills

Higher-Order Learning Skills (HOLS) were assessed as to the extent in which they differed between simulated and virtual versus traditional laboratory experiments when comparing pre- and post-laboratory experiments? Results of the statistical analysis of the repeated measures indicated that the use of the simulation and virtual labs in the treatment group does increase the perception of comprehensive HOLS employment. The use of HOLS with simulations and virtual laboratory experiments consisted of two main subscales, that of motivation and learning strategies. The results showed that learners in the treatment (simulation and virtual laboratory experiment modalities) group had higher levels of HOLS usage and implementation (as measured by the MSLQ survey) than learners in the non-treatment (face-to-face on-campus laboratory experiment modalities) group. The findings from this research are that those learners in the treatment (simulation and virtual laboratory experiment modalities) group performed at a more advanced level than learners in the non-treatment group (face-to-face on-campus laboratory experiment modalities), is consistent with past research on the usage of simulation and virtual laboratory experiment within the science disciplines [13–16]. Metacognition involves the progression of self-correction and self-awareness of one's thinking processes and the application of heuristics [13, 17].

The results are significant in that they indicate the use of HOLS in a metacognitive capacity. The research findings denoted that learners are using more developed methods for scientific analysis through the use of Higher-Order Learning Skills. The inclusion of synthesis and analysis, within experimentation [6, 8], is evident that the progression of high-order processing and evaluation of learner material is ongoing. Furthermore, motivation and learning strategies were employed more often and in more substantial form from the treatment modality than from the non-treatment modality. The MSLQ assessed motivation levels between the treatment groups and the data showed greater use of motivation throughout the experiments and the overall scientific learning. Learning strategies were reflected in the use of Bloom's Taxonomy [8] that utilizes synthesis and analysis in situational learning experiences, such as laboratory courses. The estimated marginal means data (as shown in Fig. 1) shows that the treatment group increased their HOLS post-experiment.

The graph of estimated marginal means represented a comparison of unequal sample sizes between the treatment group and the non-treatment group. The relationship between the dependent variables depicted a computed average across the levels of within and between subject factors. The graph displays an interaction of the variables pertaining to the usage of virtual and simulated science labs' effect on Higher-Order Learning Skills.

These new findings can help to encourage science educators to include more simulation-based experiments and to author more inquiry-based labs. Previously, it was thought [4, 6] that these types of labs were inconclusive in their efforts to maintain strict scientific research principles and lacked the analytical prowess of traditional expository experiments. The data showed that this was not the case, based on the MSLQ ANOVA results that indicated learner motivation and analytical skills being higher than anticipated in the treatment group. The results will bolster the field of science education and educational technology as this new information will allow for more advanced research.

Critical Thinking Skills

Critical Thinking (CT) Skills were assessed as to the extent in which they differed between simulated and virtual versus traditional laboratory experiments when compared pre- and post-laboratory experiments. Results of the statistical analysis of the repeated measures indicated that the use of the simulation and virtual labs in the treatment group does increase the use of CT Skills. The use of CT Skills with simulations and virtual laboratory experiments consisted of two main subscales, deep learning and surface learning. The results showed that learners in the treatment (simulation and virtual laboratory experiment modalities) group had higher levels of CT Skill usage and implementation (as measured by the R-SPQ-2F survey) than learners in the non-treatment (face-to-face on-campus laboratory experiment modalities) group. The estimated marginal means data (as shown in Fig. 2) shows that the treatment group increased their CT Skills post-experiment.

The graph of estimated marginal means represented a comparison of unequal sample sizes between the treatment group and the non-treatment group. The relationship between the dependent variables depicted a computed

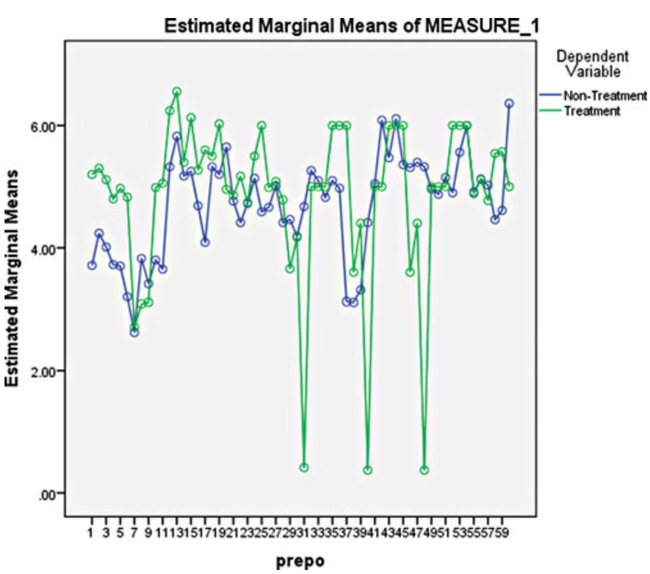

Fig. 1 Estimated marginal means for with-in subjects (pre/post-lab) MSLQ subscales

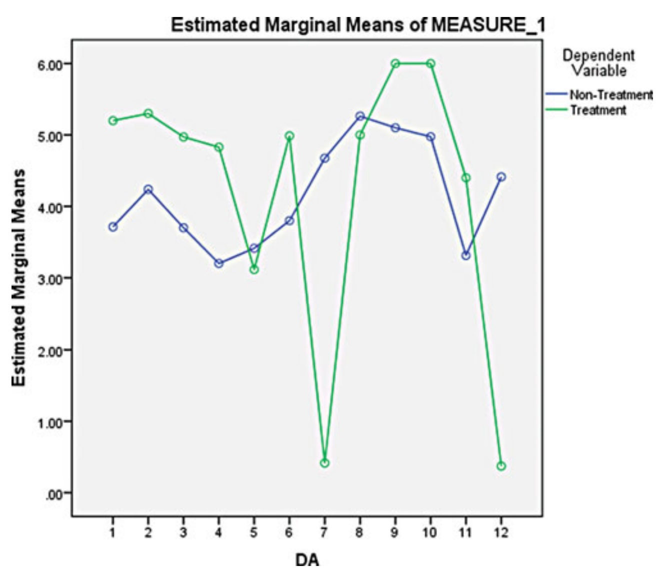

Fig. 2 Estimated marginal means for with-in subjects (pre/post-lab) R-SPQ-2F deep approach subscales

average across the levels of within and between subject factors. The graph displays an interaction of the variables pertaining to the usage of virtual and simulated science labs' effect on Critical Thinking Skills as it relates to deeper levels of learning. The levels of learning, both deep and surface level approaches, correspond to the treatment and non-treatment groups respectively.

The findings from this research are that those learners in the treatment (simulation and virtual laboratory experiment modalities) group thought more critically and analytically than learners in the non-treatment group (face-to-face on-campus laboratory experiment modalities), is consistent with past research on the usage of simulation and virtual laboratory experiment within the science disciplines [13–16]. According to Zoller and Pushkin [16], three components integral in science education within the domain of higher-order cognitive skills development include: problem-solving, Critical Thinking, and laboratory practice. The results are significant in that they indicate the use of CT Skills in a deep and surface learning capacity. The research findings denoted that learners are using more in-depth methods for scientific analysis through the use of Critical Thinking. The inclusion of critical thinking, within experimentation, is evident that the progression of synthesis and evaluation of learner material is ongoing [6, 8]. Furthermore, deep and surface learning strategies were employed more often and in more substantial form from the treatment modality than from the non-treatment modality. The data showed that this was not the case, based on the R-SPQ-2F ANOVA results that indicated learner depth of knowledge was higher than anticipated in the treatment group.

Cognitive Load

Cognitive Load (CL) capacity was assessed as to the extent in which they differed between the use of simulated and virtual laboratory experiments when compared pre- and post-laboratory experiments. Results of the statistical analysis of the repeated measures indicated that the use of the simulation and virtual labs in the treatment group does not statistically show great significance in the Cognitive Load. The effective use of Cognitive Load with simulations and virtual laboratory experiments was somewhat significant, as the within groups showed more efficient use of working memory than that of the between group results. The results showed that learners in the treatment (simulation and virtual laboratory experiment modalities) group had moderately higher levels of Cognitive Load usage and implementation (as measured by the SAI-II survey) than learners in the non-treatment (face-to-face on-campus laboratory experiment modalities) group. The findings from this research are that those learners in the treatment (simulation and virtual laboratory experiment modalities) group used similar levels cognitive load capabilities as did learners in the non-treatment group (face-to- face on-campus laboratory experiment modalities), is consistent with past research on the usage of simulation and virtual laboratory experiment within the science disciplines [13–16].

The results are moderately significant in that they indicate the use of Cognitive Load in a memory-based capacity. The research findings denoted that learners are using more working memory for scientific analysis through the use of Cognitive Load. The inclusion of working memory is evident, due to the amount of recall the learners had within the experiments. Furthermore, CL was employed often in the treatment modality than the non-treatment modality. These new findings can help to encourage science educators to include more simulation-based experiments that rely upon recall ability and the use of inter-connected knowledge from past learned material to form new connections. The data showed that CL ability, based on the SAI-II ANOVA results that indicated learner reliance on previously learned material was moderately higher than anticipated in the treatment group. The estimated marginal means data (as shown in Fig. 3) shows that the treatment group marginally increased their CL abilities post-experiment.

The graph of estimated marginal means represented a comparison of unequal sample sizes between the treatment group and the non-treatment group. The relationship between the dependent variables depicted a computed average across the levels of within and between subject factors. The graph displays an interaction of the variables pertaining to the usage of virtual and simulated science labs' effect on Critical Thinking Skills as it relates to surface levels of learning.

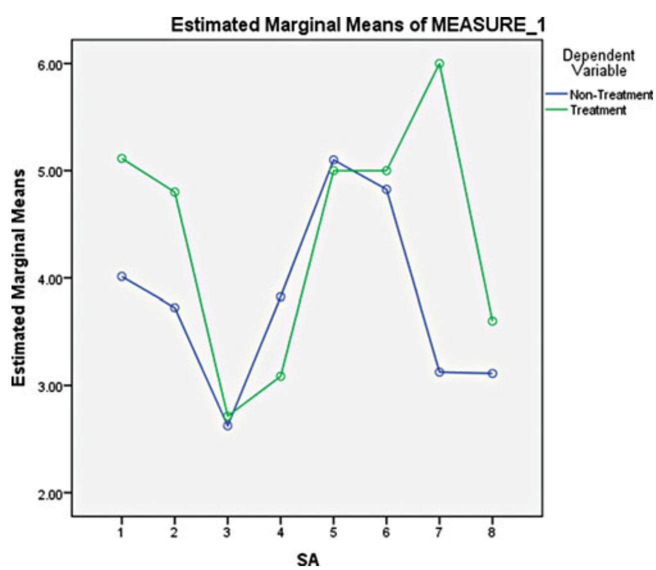

Fig. 3 Estimated marginal means for with-in subjects (pre/post-lab) R-SPQ-2F surface approach subscales

The levels of learning, both deep and surface level approaches, correspond to the treatment and non-treatment groups respectively. The treatment group showed a significant interaction for the with-in subject groups.

Summary and Conclusion

In this paper, the impact of virtual and simulation virtual laboratory experiments, on learning developing, Higher-Order Learning Skills (HOLS), Critical Thinking (CT) skills, and Cognitive Load (CL) capabilities, is studied. A pre/post survey methodology is introduced, as to distinguish learning events during laboratory experimentation. Statistical results are presented that show Higher-Order Learning Skills can be achieved as well as increased Cognitive Load ability in virtual and simulation laboratory learning environments. With the incorporation of these learning environments into General Science Study laboratory courses, learners may increase their knowledge base within defined course content areas more expressly directed at the science disciplines. The research study showed that, when utilized properly, simulation software and virtual laboratory experiments can facilitate an environment for learning that develops and fosters Critical Thinking and Higher-Order Learning. These variables will help to maintain or exceed Cognitive Load abilities; explicitly aimed at the scientific disciplines, scientific technology, and overall scientific awareness. Instructional and educational design of a course aids in the determination of whether a learner utilizes deep or surface learning through Higher-Order Learning Skills and Cognitive Load abilities [18]. Educators need to heed the use of simulations and virtual laboratory experiments in science courses so that their uses are based upon sound instructional theories and best practices [15, 19].

This study offered a discrete perspective for science educators with interests in simulation and virtual laboratory experiments and for educational technologists interested in creating these learning environments. Therefore, it is recommended that science educators and educational technology specialists in higher education fully examine the effectiveness of simulations and virtual laboratory experiments in science education. The requirements for learners in General Science Study laboratory courses is to master scientific concepts and engage in meaningful knowledge through learning approaches that can be used for a multitude of educational and career pathways. For the reason that many of these laboratory experiments will never be used by non-science major learners in their academic futures, the results of this research study may support the use of the educational technology instructional methodologies that are not "wet" laboratory based. The current research findings support the use of simulations and virtual laboratory experiment software in science laboratory experiments as long as science educators use the educational technology in a proper manner when designing and implementing instructional design planning for curricula. The results of this study will allow educators to identify that simulations and virtual laboratory experiments play an integral role in science education. Educational technology will only enhance the learning experience, not distract from the outcomes of the science curricula.

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Handling a Device Changing from 3G to Wi-Fi Without Breaking Established Connections

Richard Sims and Carolin Bauer

Abstract

The vertical handover is the ability for a device, such as a smartphone, to change between cellular data and Wireless LANs without losing any established connections. Previous approaches rely on tunneling to achieve this. This research aims to demonstrate a new approach to the problem, without tunneling, using a new intermediate layer 4 protocol running on top of TCP. The newly developed concept was tested through three scenarios that proved the mechanism works.

Keywords

3G • Mobile broadband • Vertical handover • Wi-Fi

Introduction

The vertical handover is seen as the next problem to overcome in the mobile field. At the moment when changing between different Wireless Local Area Networks (WLANs), or from a 3G (cellular) network to a WLAN and back again, all established connections are dropped and must reconnect. Existing solutions already exist for this, such as the IEEE 802.21 standard, however they require a Home Agent (HA) to route traffic through [1], effectively tunneling all communications. This will add additional latency, the additional cost of a HA and, if the HA is a residential router, bandwidth limits. The HA also creates a single point of failure.

There has been previous work in this area, such as that by Yang and Mi-Jeong [2], which builds on 802.21 however still requires a HA in the solution.

This research aims to solve the problem of connections dropping without using tunneling. The aims of the solution are to be fast at reacting to changes, transparent to the user and to be direct. It must also be compatible with existing systems and firewalls. This also includes Internet Protocol version 4 (IPv4), the most common addressing scheme at time of writing. The solution must be secure and prevent connection hijacking by a third party but without letting the security aspect noticeably slow the system down.

It is important to not try and reinvent the wheel and instead remain true to the standards and protocols we have today. This is why it would be best to look at running another protocol on top of either the Transmission Control Protocol (TCP) or the Universal Datagram Protocol (UDP), encapsulating the data from the main application instead of trying to integrate it straight into TCP/UDP.

The Idea

Problems

There were several potential problems that had to be worked out during the design stage. A few trade-offs were made between time and quality. As this is just a proof of concept, a few exceptions were made.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_25, © Springer International Publishing Switzerland 2015

Complexity

Modern day operating systems are very complex and even though some, such as Linux, are open source it would still take a large amount of time to fully integrate this solution into them. There would also most likely be issues with other applications that are not expecting the changes. Because of this it was decided the final solution would have to be run on a higher level in the userland area, as opposed to kernel level. Randomly changing TCP would have caused a lot of problems.

Compatibility with Existing Systems

As well as problems with existing software there will also potentially be issues with existing hardware or networking devices. Layer 7 firewalls were identified as the major issue here. A layer 7 firewall would not like it if a protocol resumes halfway through, and will flag it as suspicious or block it altogether. To get around this the solution will have to run on layer 7 of the Open Systems Interconnections (OSI) model. This way the layer 7 firewall will hopefully treat it as it would the File Transfer Protocol (FTP) or the Hypertext Transfer Protocol (HTTP), used for websites, and not attempt to interfere. In this regard the first solution will be more of a new protocol design technique that would have to be implemented protocol by protocol, instead of a system wide fix for everything.

It would also be too complex to make it work when both endpoints are behind different Network Address Translated Networks (NATs). A possible extension would be to add UDP hole punching, although this would have to be implemented on an application by application basis due to the need to have a third node to bounce connections.

TCP vs. UDP

TCP has a lot of features that would be quite time-consuming to create, such as the acknowledgement (ACK) packets [3]. This helps to make sure the packet was received on the other side. The ACK packets link in with the packet number of TCP. This helps ensure all packets are in the right order before being passed further along [3]. This is a feature missing from UDP [4]. Both TCP and UDP have checks in place to make sure packets have not been modified between the sender and the receiver [3, 4]. Based on the extra features it made more sense to use TCP for prototyping. Some aspects of TCP, such as ACKs and content lengths needed to be re-created, however. This will make experimenting with UDP easier in the future.

Basic Design

Although a fair amount of the design was left to experimentation and trial and error, a few brief concepts were adhered to throughout:

Packet Formation

All packets must contain a header including the connection ID and the type of packet that follows. Two types of packets will exist; for example data and admin. Data is for when another raw packet is being passed along, whereas admin is for tasks such as establishing a new connection or informing the server that the IP has changed. All raw data will need encapsulating and the protocol needs to handle fragmentation as appropriate. Each encapsulated data packet also needs a length and an ID, like the TCP sequence numbers [3].

When the connection is first initiated, the first packet is to be sent with an ID of 0. The packet type will be admin and will be requesting an ID. Once the server has assigned the client an ID, this will be sent for all future reconnection requests.

Buffering

Upon reconnection any packets sent during the outage need to be re-sent. As the system TCP buffer cannot be used here, another buffer will need to be made on a higher level. Packets will be logged with their ID and removed once an ACK is received. Upon reconnection the packets still left in the buffer will be re-sent.

Connection Management and Security

A challenge response method is to be used for security when changing IP. This is to prevent connection hijacking as hopefully the secret will only be known by the server and the client. A Challenge Handshake Authentication Protocol (CHAP) authentication-like approach was used here as it is a tried and tested method and was more secure than attempting to write one from scratch.

CHAP authentication requires that both sides have a shared secret and that it is not transferred during the challenge response. In order to create a shared key securely, a public key exchange will be used. The server will invent a public key and share it with the client, and from there a secret key will be created.

The Diffie Hellman [5] algorithm is a simple yet secure public key exchange. It relies on some maths functions built into most programming languages making it the easiest to use.

From this point all packets have a type of 0 and data will flow from both sides. The server will use the ID field to work out which client is sending the data. Each side will keep a packet buffer and will hold on to all packets until an ACK is received. Should a client disconnect, the buffers will keep filling and prevent packet loss.

Handling Disconnections

The server will not truly be informed of a client's disconnection until it tries to reconnect, making the buffer quite important. The client's first packet after reconnection will still use the old ID number and will be an admin packet saying it disconnected. The server will then issue a challenge to the client to confirm it is the same client. After the client has been authenticated using the shared secret and a CHAP authentication-like challenge, the buffers are flushed and data will flow once more.

Protocol Design

The new protocol encapsulates everything into a new packet with a new header. This helps with buffering amongst other things. As part of that a new packet structure needs to be adhered to. Although it could have more added, the current version is quite simple, generally featuring a packet type, a packet ID and a packet length. A new system of ACK had to be devised as well, in order to know when to clear data from the buffer. This was especially important due to the unexpected nature of some of the disconnections dealt with.

Every message exchanged was encapsulated into this format, whether it were a control message or just passing data through. The receiving side then read the packet type first and reacted accordingly.

Results

Testing simulated three different scenarios; Local, Virtual Private Network (VPN) and Internet using a prototype client and server coded in C#.NET, a programming language prevalent on Microsoft Windows.

The local test involved the client and the server running on the same machine, and was generally used in ad-hoc testing. The VPN test involved the server running on a remote server connected via VPN to the local LAN. The client computer was disconnected from the LAN and then reconnected (same IP) to see if the application recovered and caught up. The final test involved going via the Internet instead of a VPN. The client machine left the LAN and roamed onto Mobile Data via an Android phone in hotspot mode (different IP) to see if that scenario also worked.

Testing only looked at the solution's ability to recover from a disconnection and the speed in which it could do this. At this stage, the speed of noticing the disconnection and recovering was the most important.

Reference [6] provides the icons used for the diagrams in this section.

Test Network

Figure 1 shows the network layout used during testing. The client could connect to the server either via the VPN, symbolized by the grey pipe in Fig. 1, via the Internet using the wired network or internet only via the wireless network at the bottom of Fig. 1. The wireless connection symbolises an Android phone in hotspot mode. The networks were completely separate and the laptop was acting as if a phone went from home Wi-Fi to Mobile Data.

Local Machine

In Fig. 2 the key exchange is visible and the server has started to send packets to the client. After a while the client looks like the following:

Figure 3 shows the time being successfully sent every 5 s. As this test was being performed locally, the client-side

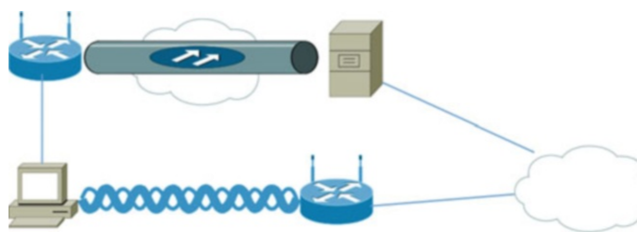

Fig. 1 Network diagram of test network

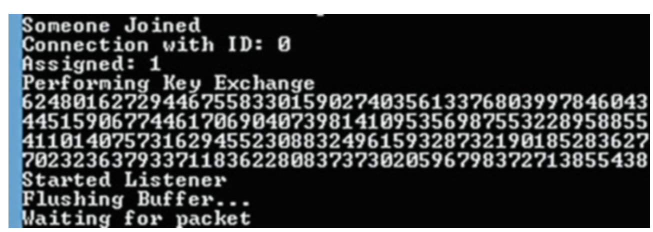

Fig. 2 The server after a connection is made

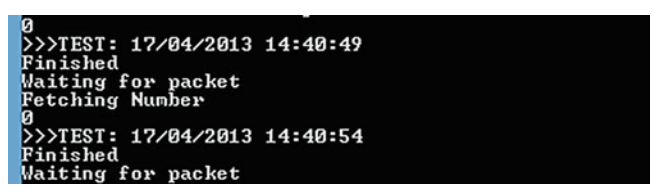

Fig. 3 Application working normally

'disconnect' command was used. This forces the application to disconnect and then attempt to reconnect.

The client states at the top of Fig. 4 that it noticed the disconnection and killed off part of the system. It then immediately tried to reconnect. After succeeding it then performed the challenge, before continuing.

Server via VPN

During this test the server was not running on the local machine. The server itself was in the cloud and was connected to the client machine via VPN at the router level. This meant the client was free to roam around inside the LAN and always had access to the VPN. The VPN also helped ensure that filtering rules at the ISP level would not interfere with the operation, in the event of them having been applied. The network was known, whereas the internet was rather unknown. Figure 5 shows the trace route for the server when going via VPN.

During this test the client was unplugged and switched from a wireless to wired connection simulating a disconnection. The wireless was turned off before the cable was added as that was a closer match to how Android phones handle the hop. Figure 6 shows the client connecting to the server down the VPN. Figure 7 then shows the application running normally, before a disconnection occurs.

In Fig. 7 the client is now running.

Fig. 4 Application disconnecting, then performing a challenge

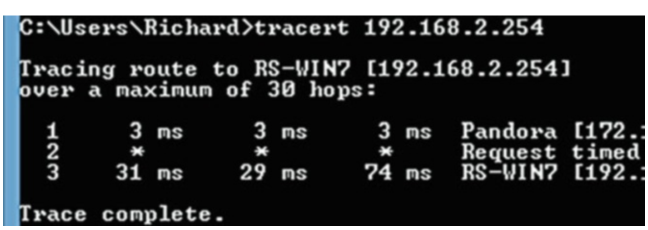

Fig. 5 A trace route. Although the VPN edge does not reply the server is visible as hop 3

Figure 8 shows that although the server registered the lost connection immediately, the client noticed after a delay. This was roughly 1 min.

After the time was up, the cable was reinserted and the system was able to reconnect, challenge and catch up on lost packets. Figure 9 shows the recovery, featuring the challenge upon reconnection.

After further research the delay issue seems to be down to the TCP keep alive being by default 1 min. Artificially setting it to 5 s means it takes 5 s to notice. The keep alive relies on data being sent, in this case the 5 s time bursts.

Fig. 6 The client is started and the server's details are entered

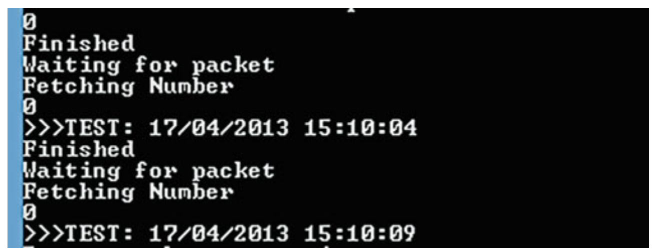

Fig. 7 Application running normally

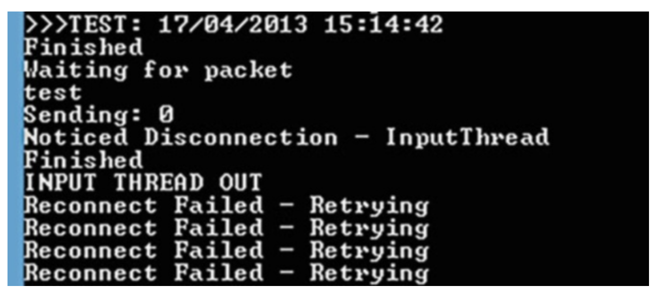

Fig. 8 Disconnection handling

Fig. 9 Reconnection and challenge

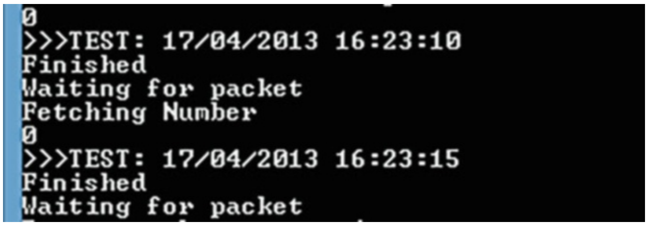

Fig. 10 Stable connection to 37.59.81.180 via the internet

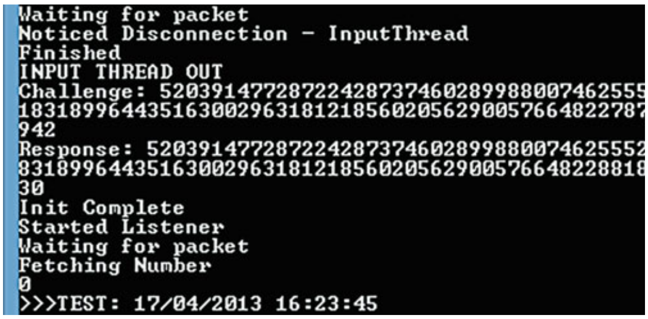

Fig. 11 Reconnection and challenge

Setting the limit to 5 s made disconnection sensing much faster. The problem, however, with using a 5 s keep-alive is that data must be sent more frequently which will cause battery issues on mobile devices. On Android sending data at intervals of less than a minute seriously affects battery performance [7].

Server Over Internet

This test was done by establishing a connection via the internet, not via the VPN, and then changing the wireless network on the PC to a hotspot run off an Android powered smartphone. Figure 10 shows a stable connection between the client and server using the wired network.

Reconnection was almost instant, despite the TCP keep alive still being 1 min. Figure 11 shows the reconnection after the client joined the Android Wi-Fi hotspot. Instead of changing network interface, only the wireless network was altered. Further research should show why this occurs.

Conclusion

The results were very promising with the system, for the most part, functioning as expected with only a few issues. There was no noticeable lag with authentication when connecting for the first time, as well as subsequent reconnects, and during the tests the system was able to reconnect multiple times while changing networks.

There were issues with the time it took to notice a disconnection under some quite specific circumstances. When the test involved changing wireless network, the software detected it instantly, whereas when it was going from wired to wireless and vice versa, across separate networking interfaces, there was a minute-long delay. This is most likely due to Windows hoping the disconnected interface returns before eventually giving up. With wireless this clearly cannot happen.

A potential solution would be to use features built into the operating system (OS) that signal when a networking change has occurred, such as intends in the Android Smartphone OS [8]. Intents allow the OS to send a signal to every app signaling an event. There are system intents to say when wireless networks have changed or the phone has moved onto mobile data from wireless.

Despite the success of this research, it is still possible to improve upon the methods used. UDP should be considered and tested. Due to its stateless nature [4], it will instantly react to changes in networking without any additional code.

As there is already a key exchange in place, the entire packet could be encrypted with trivial code changes, adding an additional layer of security. As this research considered things like public hotspots, this would be a very worthwhile follow-up.

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Implementation and Analysis of a Morphological Algorithm for the Attributes Opening and Closing

D. Bachurin and A. Lakhtin

Abstract

In this paper we present the implementation and analysis of a morphological algorithm for the attributes opening and closing on grayscale images. We investigated the application of this algorithm to different types of images. As a result we determined the range of the algorithm applicability and obtained recommendations for choosing it's parameters.

Unlike most morphological algorithms this one solves both tasks: removing grayscale peaks and retaining image structure.

This makes the image more simple for subsequent processing. At the same time we retained all significant image features for solving the segmentation problem.

Keywords

Morphological operations • Image processing • Tree data structures

Introduction

In this paper we present the implementation of fast morphological algorithm for area opening and closing on grayscale images.

We are based on a research proposed by J. Darbon and C. Ul in [1]. Their paper contains well-described idea of the algorithm and is more theoretical than applied. It does not disclose all details of the algorithm and sometimes even contains some indeterminacies and inconsistencies. We have implemented the algorithm to test its efficiency on different types of images, to evaluate it's speed and memory requirements. The main task is to determine the range of applicability of the algorithm and obtain recommendations for it's using for different types of images.

In section "Algorithm" we discuss main ideas of this algorithm and represent details of it's implementation. Section "Applicability of the Algorithm" contains examples of using implemented algorithm in different conditions. In the conclusion we discuss the fields where the algorithm may be useful.

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Algorithm

Darbon and Ul presented [1] their own algorithm for area opening and closing on grayscale images. They claim that their algorithm exhibits considerable performance compared to other modern algorithms in morphological image processing, such as the max-tree algorithm [2] and the union-find approach [3]. Of course, these assertions call to pay close attention to the proposed algorithm.

Their article contains a new technique to realize an attribute opening (closing). The algorithm is based on the well-known method of Salembier's max-tree construction [2]. It uses a tree-based image representation where each node in the tree stands for a flat region or a connected component. The tree itself is oriented towards the maxima of the grayscale image and hence is called a max-tree. The max-tree is constructed using a hierarchical queue data structure. In order to filter a grayscale image one needs to measure a specific attribute associated with the connected component of each node and to decide whether to keep the component or merge it to it's father.

The decision is based on comparing the value of the attribute against a prescribed threshold. The main advantage is based on the tree pruning strategy, using the area as the

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_26, © Springer International Publishing Switzerland 2015

attribute. This allows to the algorithm not to build any tree in contrast to those that first explicitly build the tree and only after that prune it for filtering. The primary part of the algorithm is tree creation using a recursive flooding procedure.

Before proceeding to a detailed analysis of the algorithm, let's consider the data structures that are used in the algorithm. We shall use the same names and data types as in [1].

Parameters and Variables

The image size N and the gray levels L are determined. An array *status* of length N is used to store pointers to the points, encoded by positive numbers. The array becomes a max-tree, coding references child-to-parent. It is initialized by a special value *NOT_ANALYZED*, which means that there is no connection yet. The gray levels determine the lengths of the following three arrays: *last*, *representative* and *area*.

1. *Last*—to store the last element added to the queue.
2. *Representative*—to store the unique representative of gray level components. The value of *representative[h]* keeps the first point added to the current gray level h of the component C_h^k the point is the unique representative of this component. Each time a new component is being extracted, it's first encountered element is stored in this array.
3. *area* is an array to store the image size. The value *area[h]* stores number of elements of the gray level h of the component C_h^k.

Arrays *last* and *representative* are initialized by a special value *NONE*, which means that the gray-level queue is empty and there is currently no points to represent the component. Array *area* is initialized by *0*, which means that there is no point of this gray level yet.

Usually we have L much less than N, therefore, the algorithm requires only one memory array of length N.

Preflood Procedure

Considering a pixel p of a gray level h, which was not analyzed yet, we perform some actions that were not available in [1].

We have identified this stage of the process, as *preflood*. For each gray level, starting from the maximum for a closing operation (from the minimum one for opening operation), we call *preflood(h)* procedure, where h—is the considered gray level. The procedure processes the points of gray level h in an arbitrary order.

The Preflood procedure

Require: int $h \geq 0 \vee h \leq 255$

```
for all pixel p from level h do
  {check if the point is new, add it in the queue of the
  level h and make it a representative of this level}
  if status[p] is NOT_ANALYZED then
    status[p] = NONE;
    last[h] = p;
    representative[h] = p;
    flood(h);
  end if
end for
```

For the selected point p, it is verified that the point has not been considered yet. If so, its status is set to *NONE*, the point is added to this level queue (*last[h]* $= p$), and is made a representative of this gray level (*representative[h]* $= p$). Only after that, the construction of a maximal tree for the selected level begins in this procedure.

If the point was already considered earlier, we select the next point of the current level h. After consideration of all points of gray level h, the procedure *preflood* is called for the next gray level.

Flood Procedure Implementation

From the procedure *preflood*, the procedure *flood* is called with a single parameter which is the gray level h.

At the propagation stage trees are built. We shall explain the procedure in terms of the closing operation. The opening procedure can be described in analogous way.

Let's consider a pixel and its neighbours. Neighbour pixels with the same level form one connected component. Neighbour pixels with lower intensities are added to this component if its size is less than a threshold value now. This stage is detailed lower. Just note that in the frame our implementation is shown and it differs from the original in [1].

The Flood procedure (part 1)

Require: int $h \geq 0 \vee h \leq 255$

Ensure: int

```
1. while last[h] ≠ NONE do
2.    {Take the point from the queue. It can't be added
      in a queue again}
3.    p = last[h];
4.    last[h] = status[p];
5.    status[p] = representative[h];
6.    for all neighbour n of pixel p do
7.       if status[n] is NOT_ANALYZED then
8.          {if a neighbour point n was not considered
      yet.}
9.          m = luminance[n];
10.         {if there is no representative for the level m }
11.            if representative[m] is NONE then
12.               representative[m] = n;
13.            end if
14.            {add the point to the queue of the level m }
15.            status[n] = last[m];
16.            last[m] = n;
17.            while m < h do
18.               m=flood(m);
19.            end while
20.         end if
21.      end for
22.      area[h]++;
23.   {continue in part 2}
24.   end while
```

At the first step, a loop begins with the condition "while the queue for the current gray level is not empty." This is so, because earlier in procedure *preflood*, we have put a pixel in the queue. This would be the first difference from [1], if we ignore the actions that we have made in the procedure *preflood*.

In the paper [1] on the first line, there is a while loop "while queue for this gray level is empty". However, the next step must take an element from the queue. And there are no comments from authors about this collision. The second difference of our implementation is on 9–13 steps. Here we must define the gray level representative and build a tree. In the original paper, this block was as following:

Algorithm error from [1]

```
1. if representative[h] is NONE then
2.    representative[h] = p;
3. end if
4. status[p] = last[h];
5. last[h] = p;
```

Likely, at this point there is a typo, because the authors write approximately the following comments to these lines: "If a pixel in the neighbourhood of *p*, which has not been processed, is encountered, then it is added into the hierarchical queue (lines 13–14); and if it is a new component, then we update the array representative". However, the difference between the text and the code can confuse a reader. After obtaining an empty queue of the level *h* and the trees are constructed, it is time to merge different level trees into one.

The Flood procedure (part 2)

```
1. while last[h] ≠ NONE do
2.    {part 1}
3.    m=h+1;
4.    while m ≤ 255 ∨  representative[m] is NONE do
5.       m++;
6.    end while
7.    if m ≤ 255 do
8.       if area[h] < λ do
9.          status[representative[h]]=representative[m];
10.         area[m]+=area[h];
11.      else
12.         status[representative[h]]=-h-1;
13.      end if
14.   else
15.      status[representative[h]]=-h-1;
16.   end if
17.   area[h]=0;
18.   last[h]=representative[h]=NONE;
19.   return m;
20. end while
```

For the opening operation, the variable is changed in opposite direction that is from $h - 1$ to the minimal gray level. When the procedure *flood* is finished, each pixel in the array *status* keeps a pointer to his host representative or gray level (in the negative form). Therefore the stage of restitution is performed in such a way that all pixels are stored together with associated levels. This process is illustrated below.

Restitution stage

```
1. for all pixel p do
2.    int root=status[p];
3.    if root<0 then
4.    root=p; {case of the tree root }
5.    end if
6.    while status[root] ≥ 0 do
7.       root = status[root];
8.    end while
9.    int value = status[root];
10.   while p ≠ root do
11.      tmp = status[p];
12.      status[p] = value;
13.      p = tmp;
14.   end while
15. end for
```

Note that in the paper [1] there was no checking that the considered point is the tree root (the step 3). This could lead to several errors: incorrect determination of the gray level of the tree (variable *val*) and an infinite loop in the step 6.

Applicability of the Algorithm

In order to determine the range of applicability of the implemented algorithm, it should be tested on different types of images. In our study we have used the following image types:

1. Satellite images containing a road network;
2. Portraits;
3. Images from medical and biological researches.

Thus a wide range of image processing tasks was covered.

In most applications of image processing, the algorithm complexity increases due to large number of small minor details. At the same time if we want to get rid of those unnecessary details then we would have another task, namely to keep essential features of images such as borders of areas.

To solve the problems and achieve the goals, we apply morphological algorithms with an attribute of opening and closing. The resulting image data operation is called the domain closing and opening, respectively. If we consider the closing (opening) grayscale image then in general case it allows us:

1. To get rid of the peaks with low (high) level of brightness, which do not satisfy a criterion;
2. To keep peaks with a high brightness level satisfying a criterion;
3. To define a criterion and set parameters. For example, the parameter may be equals to the size of the area. Then the criterion for selection is to verify whether the size of the area is less than the given parameter; if so, one should be gotten rid from the area, otherwise, it is kept.

When we get rid of an area, it means that this area is joined to the area with a higher gray level in the case of closing, and to the area with a less intensity in the case of opening the image. Figure 1 illustrates the image before processing, and after closing or opening. The parameter is the size of the area. We can see how the result of image processing depends on the parameter.

Further let's consider how the implemented algorithm works with specific types of images.

Analysis of Satellite Images

There is a large class of problems associated with the analysis of satellite images [4]. One of them is to highlight some features such as a road network or borders of areas. Figure 2 shows an example of such an image, as well as it's opening and closing.

The image is large enough, so we consider a large view of a small portion of the image. Figure 3 shows how the closing of the image will be painted with small dark areas in a monotone color. With that the body shape is similar to a triangle, and the adjacent lines (roads) stay as bright as in the original image.

Opening the image with a current parameter value has eliminated all differences of small details. After conducting a series of test over images with road network we concluded that:

1. The algorithm is resistant to this type of images;
2. Opening and closing of the images remove unnecessary details, and the road network retains it's structure. The resultant image contains more uniform areas and less local differences. This should help in the problem of recognizing a road network;
3. Closing of the image makes the road network more contrast in comparison with the adjacent areas.

Fig. 1 Illustration of the effect of an area closing and area opening. Images from *left to right*: original image, result of area closing, result of area opening

Fig. 2 Illustration of the effect of an area closing and area opening to satellite images. Images from *left to right*: original image, result of area closing, result of area opening

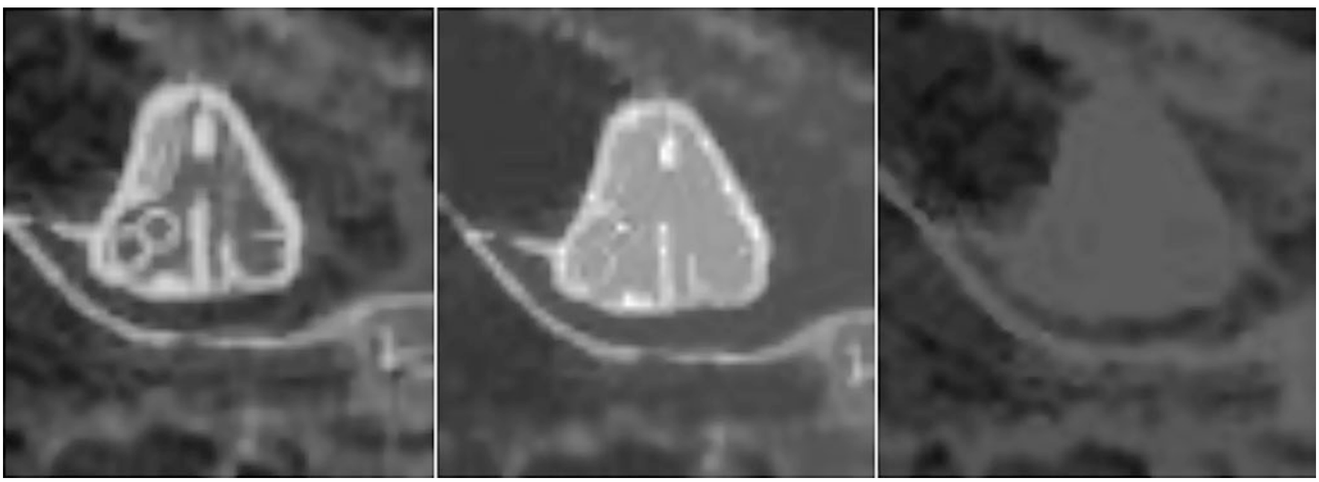

Fig. 3 Illustration of the effect of an area closing and area opening in details. Images from *left to right*: original image, result of area closing, result of area opening

Portraits

Face detection [5], landmarks and pose identification [6–8] and recognition [9] are well studied problems and not the contribution of this paper. However, it is interesting which results will be obtained after applied this algorithm to processing such images. An example of applying the algorithm to a portrait is shown on Fig. 4. Obviously, the closing of the image makes the face contours less contrast with the adjacent areas. Note, however, that the girl's hair becomes more monotonous, that is, contains far fewer small drops. Opening images also clears swings of another character: with a high level. In this case, the contours of the face and the eyes become a bit clearer.

We can make the following conclusions about using of the implemented algorithm with portraits:

1. The algorithm is applicable to this type of images;
2. Closing of the image becomes more homogeneous;
3. The value of the parameter should be selected carefully to keep the important features of the image;
4. Opening image also leads to a reduction of small features; important details are kept and become more clear, that is useful for further processing.

Images in Medicine and Biology

These images have a lot of applications in single photon emission computed tomography (SPECT) [10] and confocal microscopy [11]. An example of applying the algorithm to the medical images is shown on Fig. 5. The feature of most medical images is the presence of large noise.

After analyzing result of application of the implemented algorithm to a series of images of this type, we can conclude that:

Fig. 4 Illustration of the effect of an area closing and area opening to a portrait. Images from *left to right*: original image, result of area closing, result of area opening

Fig. 5 Illustration of the effect of an area closing and area opening to in medicine. Images from *left to right*: original image, result of area closing, result of area opening

1. The algorithm can be partially applied to this type of images;
2. Closing of the image highlights features that were difficult to see in the original image;
3. Image throws away minor features and keeps only the big ones.

Algorithm shows similar results on biological images. An example of applying the algorithm to the pictures related to biological researches is presented in Fig. 6. Note that the closing is almost completely cleared the dark noise, and the opening the bright noise.

Conclusion

The implemented algorithm is resistant to various types of images and can be applied to a wide range of image processing tasks.

By applying the algorithm to an image, we enhance image significantly in terms of convenience of further processing.

For many types of images, it will be useful to apply not only the morphological closing but opening too. Moreover, in some situations, the combination of both operations can provide a more useful result than using them separately.

In the algorithm, we have the ability to change the criterion of merging areas. This may not necessarily be the size of the region, as in our case. One can, for example, use the acceptable dispersion of pixel brightness value.

The algorithm has the only parameter, which can be customized for a specific image.

The algorithm has shown a good speed of image processing, comparable to the results of [1], and low memory requirements. The algorithm may be applied in such areas as machine vision, geographical information systems, segmentation tasks and processing of images in medicine and biology.

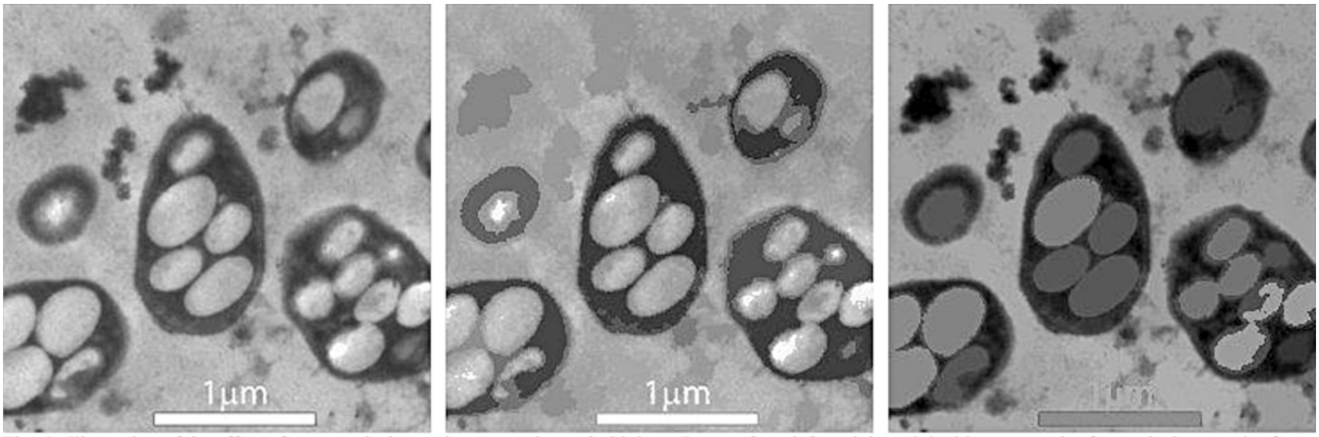

Fig. 6 Illustration of the effect of an area closing and area opening to in biology. Images from *left to right*: original image, result of area closing, result of area opening

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Document Classification Using Enhanced Grid Based Clustering Algorithm

Mohamed Ahmed Rashad, Hesham El-Deeb, and Mohamed Waleed Fakhr

Abstract

Automated document clustering is an important text mining task especially with the rapid growth of the number of online documents present in Arabic language. Text clustering aims to automatically assign the text to a predefined cluster based on linguistic features. This research proposes an enhanced grid based clustering algorithm. The main purpose of this algorithm is to divide the data space into clusters with arbitrary shape. These clusters are considered as dense regions of points in the data space that are separated by regions of low density representing noise. Also it deals with making clustering the data set with multi-densities and assigning noise and outliers to the closest category. This will reduce the time complexity. Unclassified documents are preprocessed by removing stops words and extracting word root used to reduce the dimensionality of feature vectors of documents. Each document is then represented as a vector of words and their frequencies. The accuracy is presented according to time consumption and the percentage of successfully clustered instances. The results of the experiments that were carried out on an in-house collected Arabic text have proven its effectiveness of the enhanced clustering algorithm with average accuracy 89 %.

Keywords

Clustering • K-means • Density based clustering • Grid based clustering • TFIDF

Introduction

Pattern recognition is generally categorized according to the type of learning procedure used to generate the output value. One of these types of learning is ‘Unsupervised Learning’.

An example of unsupervised learning is called ‘Clustering’ based on the common perception of the task as involving no training data to speak of, and of grouping the input data into clusters based on some inherent proximity measure, rather than assigning each input instance into one of a set of pre-defined classes. One of the important applications on clustering is ‘Text Clustering’. It is the method of partitioning unlabelled data into disjoint subsets of clusters. It is developed to improve the performance of search engines through pre-clustering the entire corpus.

Arabic language is one of the six international languages that are used by most of the humans all over the world. Most of the Arabic words are found from list of Arabic language roots which could be roots of three, four, five or six letters. Sentences in Arabic language consist of nouns, verbs, pronouns, preposition and conjunction. The noun and the verb have a root while the others don’t have so it is cancelled in the preprocessing phase.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_27, © Springer International Publishing Switzerland 2015

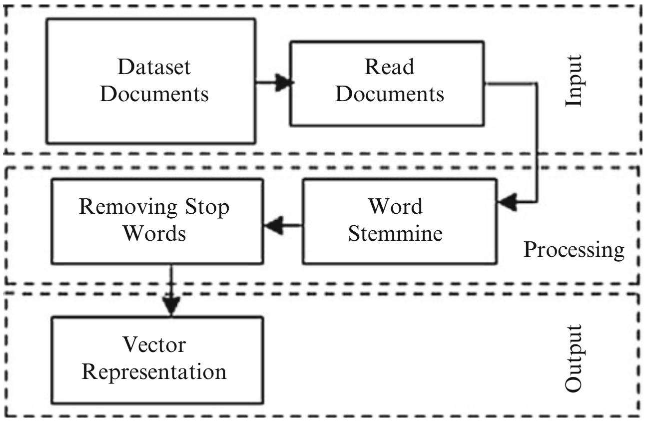

Fig. 1 Documents data preprocessing

One of the important steps in this research is document preprocessing after getting them from the internet or any other resources. On this spot two research questions could be raised about how to select the appropriate stemming methodology and how to allow scaling to large datasets and reducing the computational complexity in such applications.

Figure 1 shows a data block diagram of the data documents preprocessing phase.

The rest of this research is structured as follows. In section “Related Works”, brief description of some related works in this area of research is presented. Section “Document Preprocessing”, discuss how to extract data from the preprocessed documents and be ready for training. Also this research presents one of the methods of feature selection and weighting algorithm called TF-IDF. The research presents three of the text clustering algorithms K-means, DBSCAN improved with K-mediods and Grid Based Multi-DBSCAN using representative points. Section “Experimental Results”, evaluates experimental results. Finally, conclusion will be presented in section “Conclusions”.

Related Works

An efficient density based K-medoids clustering algorithm has been proposed to overcome the drawbacks of DBSCAN and K-medoids clustering algorithms [1].

Another related research deals with the problem of Text Categorization [2]. It concentrates on the filter approach to achieve dimensionality reduction. The filter approach consists of two main stages; feature scoring and thresholding.

A new algorithm based on DBSCAN is proposed in [3]. It presents a new method for automatic parameters generation that create clusters with different densities and generates arbitrary shaped clusters.

To improve the performance of DBSCAN algorithm, a new algorithm combined Fast DBSCAN Algorithm [3] and Memory effect in DBSCAN algorithm [4] to speed up the performance as well as to improve the quality of the output. As the Region Query operation takes long time to process the objects, only few objects are considered for the expansion and the remaining missed border objects are handled differently during the cluster expansion.

According to the for mentioned related works [1], used K-mediods to improve the performance of DBSCAN. Whereas [2] used a popular algorithm which is SVM, one of the drawbacks of this algorithm is the long training time. Finally [3, 5] had used DBSCAN approach. There are two parameters required that must be specified manually **Eps** and **MinPts**. DBSCAN cannot cluster data sets well with large differences in densities since the minPts-epsilon combination cannot be chosen appropriately for all clusters. Therefore, a new approach is proposed into order to reduce the time complexity, deal with making clustering of dataset with multi-densities and the density parameters are not specified manually.

Document Preprocessing

Before starting the feature selection procedure, there are five proposed steps for preprocessing the online datasets. Finally, clustering phase is proposed to implement the recognition task.

Preprocessing Phase

The common model for text presentation is Vector Space Model [2, 6–8]. Its main idea is to make the document become a numeral vector in multi-dimension space. The preprocessing phase includes HTML parsing and tokenization, word rooting, removing stop words, vector space representation, feature selection and weight calculation. Figure 2 shows the steps for preparing the datasets.

HTML Parsing and Tokenization

HTML Token Parser application could be used in order to extract information from the inner body of HTML files by removing HTML tokens such as tags, newlines . . .etc.

Arabic Datasets Documents

This research uses Arabic datasets which are online and dynamic datasets abstracted from the internet in form of HTML text files which needs preprocessing to remove the tags and get the main body that contains the data, as shown below in Table 1.

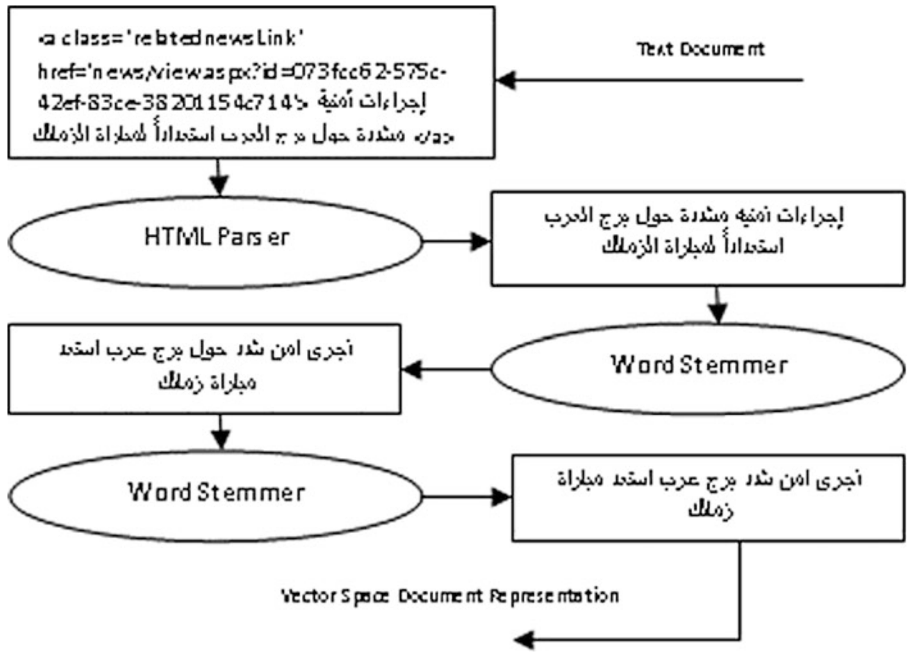

Fig. 2 Text processing with root extraction and removing stop words

Table 1 Arabic dataset sample

Category	Document number	Sample data
Economy	1500	تراجعت إيرادات عبور قناة السويس، لتصل إلى 2.6 مليار دولار، خلال الفترة من يوليو إلى ديسمبر.
Culture	1500	رفعت الرئاسة العامة لشئون المسجد الحرم والمسجد النبوى الجزء السفلى من كسوة الكعبة المشرفة بمقدار نحو ثلاثة أمتار.
Religion	1500	رفعت الرئاسة العامة لشئون المسجد الحرم والمسجد النبوى الجزء السفلى من كسوة الكعبة المشرفة بمقدار نحو ثلاثة أمتار.
Sport	1500	خطا برشلونة خطوة جديدة نحو استعادة اللقب اثر فوزه على ضيفه رايو فايكانو بثلاثة أهداف مقابل هدف وحيد

Word Rooting (Stemming)

Arabic language belongs to the Semitic group of languages. Arabic language consists of 28 letters as follow:

أ, ب, ت , ث , ج , ح , خ , د , ذ , ر , ز , س , ش, ص

ض, ط , ظ , ع , غ , ف , ق , ك , ل , م , ن , ه , و , and ي . The letters (ا و ي) are vowels; the others are consonants.

The algorithm presented in [9] is preferred for root extraction for its simple manipulation. The simplicity of this algorithm comes from employing a letter weight and order scheme. This algorithm extracts word roots by assigning weights and ranks to the letters that constitute a word. Weights are real numbers in the range 0–5. The order rank of letters in a word depends on the length of that word and on whether the word contains odd or even number of letters.

Removing Stop Words

This step removes stop words from the documents in order to generate the frequency profile of a certain list of words. Theses words might be words like prepositions, Arabic particles, and special characters.

Vector Space Document Representation

Now, Vector space model with those keywords and documents is formed [6]. Here every element in the vector space model indicates how much number of times a word occurs in the document. For example consider the following diagram which shows the term and document matrix. Rows represents words in each document, columns represents documents. Each cell represents the word count in the particular document. If the number of documents is increasing the size of the matrix will also be increased.

Feature Selection and Weighting Phase

As mentioned in [7, 8], TF-IDF for term weighting is used for weighting phase. Essentially, TF-IDF works by determining the relative frequency of words in a specific document compared to the inverse proportion of that word over the entire document corpus. Intuitively, this calculation determines how relevant a given word is in a particular document. Words that are common in a single or a small group of documents tend to have higher TFIDF numbers than common words such as articles and prepositions.

$$TFIDF(t_k, d_j) = TF_{kj}.\log\left(\frac{T_r}{df_k}\right) \quad (1)$$

Clustering Phase

It deals with finding structure in a collection of unlabeled data. There are many types of unsupervised linear clustering algorithm such as:

1. K-means clustering algorithm.
2. Fuzzy c-means clustering algorithm.
3. Hierarchical clustering algorithm.
4. Threshold based clustering algorithm.
5. Density Based clustering algorithm.

K-means is a simple partitioning algorithm as it keeps track of the centroids of the subsets, and proceeds in simple iteration the initial partitioning is randomly generated, randomly initialize the centroids to some points in the region of the space [10]. The K-means needs to perform a large number of "nearest-neighbour" queries for the points in the dataset. If the data is 'd' dimensional and there are '**N**' points in the dataset, the cost of a single iteration is O (kdN). As one would have to run several iterations, it is generally not feasible to run the naïve K-means algorithm for large number of points. So, a modification for K-mean is essential as concluded in [1] for the following reasons:

1. DBSCAN does not perquisite to know the number of clusters in the data, as opposed to k-means.
2. DBSCAN can find arbitrarily shaped clusters. It can even find clusters completely surrounded by a different cluster.
3. DBSCAN has a notion of noise.
4. DBSCAN requires just two parameters and is mostly insensitive to the ordering of the points in the database.

But the main disadvantage of DBSCAN that it does not respond well to data sets with varying densities. Also DBSCAN is sensitive to two parameters, **Eps** and **MinPts** must be required determined manually. Also the number of cluster, **K**, must be determined beforehand.

In order to solve these drawbacks of the two algorithms mentioned above, a grid based algorithm based on multi-density DBSCAN using representative points is proposed.

Experimental Results

Datasets Outlines

The selected datasets are online dynamic datasets that are characterized by its availability and credibility on the internet. Arabic text corpus was collected from online magazines and newspapers. Datasets are collected from EL-Watan News [11] and EL-Jazeera News [12]. 5,000 documents that vary in length and writing styles were collected. These documents fall into four pre-defined categories. Every category contains 1,500 documents. The set of pre-defined categories include: Sports, economic, science, religion and politics. Every document of these collected documents is automatically categorized to only one category according to human categorizer's judgment.

Proposed Algorithms

In this research, three algorithms are used. The traditional K-means algorithm, DBSCAN using K-mediods and Grid Based Density algorithm.

K-Means Algorithm

As presented in [10], the traditional K-means algorithm

1. Input **k**, number of suggested clusters
2. Select **k** centers randomly.
3. Determine numbers of trials to test Clusters,
4. Assign each document to the closest cluster based on the distance between centers and the document. Compute square error between centers and selected documents.
5. Store the results in the database.

Density Based Algorithm

Basic DBSCAN algorithm

The steps involved in DBSCAN algorithm are as follows:

1. Arbitrary select a point **P**.

2. Retrieve all points density-reachable from **P** w.r.t **Eps** and **MinPts**.
3. If **P** is a core point, cluster is formed.
4. If **P** is a border point, no points are density-reachable from P and DBSCAN visits the next point of the database.
5. Continue the process until all of the points have been processed.

Enhanced DBSCAN with K-Mediods Algorithm

As stated in [1], the algorithm could be described as follows:

1. For each unvisited point **p** in dataset **D** get neighbors w.r.t **Eps** and **MinPts**.
2. Now will have **m** clusters, find the cluster centers and the total number of points in each cluster.
3. Join two or more clusters based on density and number of points and find the new cluster center and repeat it until achieving **k** clusters.
4. Else split one or more clusters based on density and number of points using K-mediods clustering algorithm and repeat it until achieving **k** clusters.

Proposed Grid Based Multi-density Using Representative Points Algorithm

The main purpose of this algorithm is to divide the data space into clusters with arbitrary shape. These clusters are considered as dense regions of points in the data space that are separated by regions of low density representing noise. Also it deals with making clustering the data set with multi-densities and assigning noise and outliers to the closest category.

The proposed clustering algorithm consists of seven steps as shown in Fig. 3:

Here we will illustrate the algorithm implementation steps in more details:

```
If each cell has been clustered
        Then deal with noises, outliers
Else
        While there is a cell in the data space not clustered
                Select half of the cell points and put them in
                the daraset_Representative

                Put the other half of the cell points and put
                them in dataset_Remainder and remove
                them from the corresponding cell

                Select cell whose cell density is max and has
                not been clustered

                Compute MinPts = [factor * CD]
                For each data in cell
                        Cluster with DBSCAN algorithm
                If data belong to other sub-cluster
                        Then if gds≥ similar then merge
                        sub-cluster

                        Else assign it to the sub-cluster
                        whose central point is most nearest
                        from this point
                        End;
                Else tag the data as a new cluster
                End;
End while
For each point in dataset_Remainder
        Search for the nearest point in all resulted clusters
        and assign this point to the cluster contains the
        nearest point.
End;
For each resulted cluster
        Apply the remerge method
End
End;
```

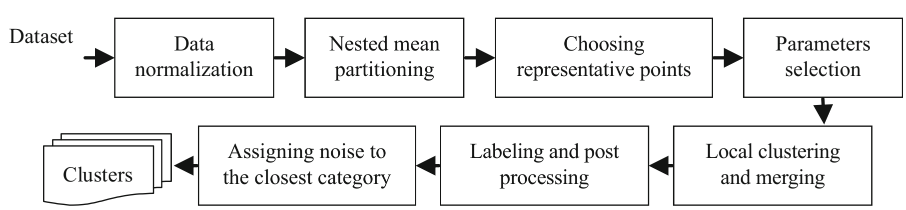

Fig. 3 proposed clustering algorithm milestones

Table 2 EL-Watan news dataset

Scenario	No. of categories	No. of files	No. of doc. per each category
Scenario 1	4	400	100 doc
Scenario 2	4	1000	250 doc
Scenario 3	4	2,000	500 doc
Scenario 4	4	3,000	750 doc
Scenario 5	4	4,000	1,000 doc
Scenario 6	4	5,000	1,250 doc

Performance Measures

In the proposed research, performance is measured according to the percentage of success rate, the number of incorrectly clustered instance and the time consumption for each algorithm to cluster the data space.

Results Evaluation

The evaluation of results is conducted via six scenarios. Each scenario has a number of files and number of categories. These categories refer to the type of the news as shown in Table 2.

Data Space Grid Distribution

As mentioned before that there are some factors that affect the effectiveness of the clustering such as the preprocessing phase, the proposed algorithm also has a factor for increasing the performance of the clustering. Below there is a comparative results between two types of dividing data space into grid of cells.

Sparse Tree

Sparse tree divides a dimension into number of intervals, each of which has the same length. This algorithm does not consider the distribution of the data. Although it can effectively locate strong clusters, it often results in an extremely uneven assignment of data items to cells and fails to examine detailed patterns within a dense area. Extreme outlier values can also severely affect the effectiveness of the sparse tree algorithm.

As shown in Fig. 4, with the sparse tree algorithm the two smaller but much denser clusters fall in a single cell. Therefore these two clusters can never be distinguished in further analysis with those cells extreme outlier values can also severely affect the effectiveness of the algorithm.

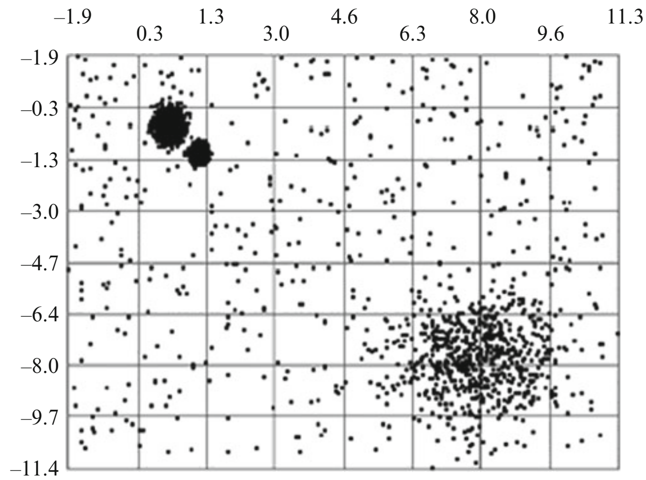

Fig. 4 sparse tree data grid distribution

Nested Mean

The nested mean divides a dimension into a number of intervals and can adapt well with the data distribution and is robust with outlier values and noisy points. It recursively calculates the mean value of the data and cut the data set into two halves with the mean value. Then each half will be cuts into halves with its own mean value.

This recursive process will stop when the required number of intervals is obtained. It can examine detailed structures within a dense region. Although it tends to divide a cluster into several cells, those cells that constitute the cluster are always denser than neighboring cells. The distance among those cells of the same cluster are very small and the clustering procedure can easily restore the cluster by connecting them.

As in Fig. 5 the two smaller but denser cells now fall in eight cells, each which is still denser than cells in a sparse area. Thus these two clusters are distinguishable in further analysis. The distance among those cells of the same cluster are very small and the clustering procedure can easily restore the cluster by connecting them.

The two data grid distribution algorithms are applied on El-Watan news dataset and the results of the proposed algorithm through three scenarios are represented as shown in Table 3.

Table 4 shows the performance measures when applying K-means algorithm to each scenario.

Table 5 shows the performance measures when applying DBSCAN using K-mediods algorithm to each scenario.

Table 6 shows the performance measures when applying Grid Based Multi-density DBSCAN using representative points algorithm to each scenario.

As shown in Table 7, it is concluded that the degradation of the accuracy results of the three algorithms is due to the

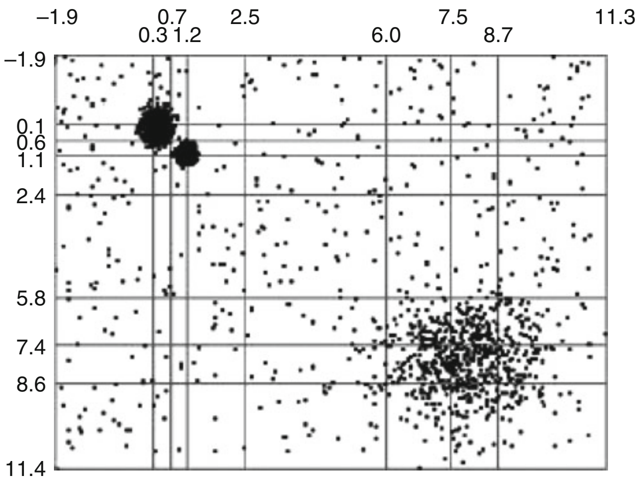

Fig. 5 Nested mean data grid distribution

Table 3 Comparative accuracy sparse tree and nested mean (%)

Scenario	No. of categories	No. of files	Sparse tree	Nested mean
Scenario 2	4	1,000	85.30	92.30
Scenario 3	4	2,000	80.80	89.10
Scenario 4	4	3,000	77.50	86.80

Table 4 K-means algorithm

Scenario	Time (Sec)	Incorrectly Instance	Success rate (%)
Scenario 1	87.03	62	84.60
Scenario 2	122.20	272	72.80
Scenario 3	253.54	698	65.10
Scenario 4	220.94	1,131	62.30
Scenario 5	428.20	1,680	58.00
Scenario 6	495.40	2,280	54.40

Table 5 DBSCAN using K-mediods algorithm

Scenario	Time (Sec)	Incorrectly instance	Success rate (%)
Scenario 1	72.20	58	85.60
Scenario 2	110.30	160	84.00
Scenario 3	229.58	443	77.85
Scenario 4	218.15	712	76.27
Scenario 5	313.92	1,108	72.30
Scenario 6	430.50	1,469	70.62

increase of degree of complexity of scenarios according to the increase of the data set samples and also another factor for this difference is the data set preprocessing. In order to measure the performance of the three algorithms a suitable metric is needed, so F-measure is used as measure for the accuracy of clustering of each algorithm for this type of case study used in the research [8, 10].

Table 6 Proposed grid based multi-density using representative points algorithm

Scenario	Time (s)	Incorrectly instance	Success rate (%)
Scenario 1	68.32	50	94.40
Scenario 2	97.60	77	92.30
Scenario 3	219.34	218	89.10
Scenario 4	209.00	396	86.80
Scenario 5	285.30	580	85.50
Scenario 6	378.20	778	84.45

Table 7 Percentage of F-measure (%)

Scenario	K-means	DBSCAN using K-mediods	Proposed algorithm
Scenario 1	84.60	85.60	94.40
Scenario 2	72.80	84.00	92.30
Scenario 3	65.10	77.85	89.10
Scenario 4	62.30	76.27	86.80
Scenario 5	58.00	72.30	85.50
Scenario 6	54.40	70.62	84.45
Average	66.20	77.77	88.76

Table 8 AL-Jazeera news dataset

Scenario	No. of categories	No. of files	K-means (%)	DBSCAN using K-mediods (%)	Proposed algorithm (%)
Scenario 1	4	1,000	77.80	89.00	97.30
Scenario 2	4	2,000	70.10	82.85	94.10
Scenario 3	4	3,000	67.30	81.27	91.80

Also it is realized that these three approaches are sensitive to the dataset, because according to the visualization of the clustering results of the selected dataset there are two categories (Economy and Culture) that are closely related to each other. So something like that will affect the effectiveness of the clustering.

The proposed clustering approach is also applied on AL-Jazeera news dataset and Table 8 shows a comparative result between the three approaches after passing the dataset through the same preprocessing phase. The same categories are used (Culture, Economy, Sports and Religion) and each scenario with different number of files.

The three algorithms shows better accuracy results here when applied on AL-Jazeera news dataset. This proves the theory that the effectiveness of clustering is sensitive to the dataset although this dataset has the same categories as AL-Watan news dataset.

In addition the proposed algorithm is compared with an algorithm proposed by [2]. The algorithm proposed is support vector machine. The algorithm is applied on a dataset collected from AL-Jazeera news categorized into five categories, science, arts, economy, religion and sports.

Table 9 Comparative accuracy

	Reference [2]	Reference [10]	Proposed algorithm
No. of categories	4	4	4
No. of docs	600	1,000	1,000
Accuracy (%)	93.00	65.00	95.00

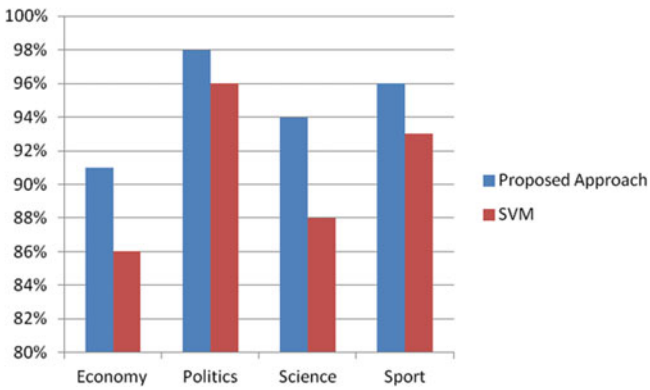

Fig. 6 Difference in accuracy between the two algorithms for each category

Table 9 below shows a comparative accuracy results between the proposed algorithm and the algorithms proposed by [2, 10] where Arabic corpus is the common case study of each research.

The two algorithms are applied on the dataset case study used by [2] to test the clustering accuracy of each approach and how accurate documents are assigned to its right category as shown below in Fig. 6. The categories used are science, economy, sports and politics, each contains 150 documents.

Conclusions

This research has shown the effectiveness of unsupervised leaning algorithms by using K-means, DBSCAN using K-mediods and Grid Based Clustering algorithm.

Arabic language is a challenging language when applied in an inference based algorithm for number of reasons. First, orthographic variations are common in Arabic language; certain combinations of characters can be written in different ways. Second, Arabic has a very complex morphology. Third, Arabic words are often ambiguous due to the trilateral root system. In Arabic, a word is usually derived from a root, which usually contains three letters. Fourth, short vowels are omitted in written. Fifth, synonyms are widespread. That is why this proposed solution is customized for Arabic language.

Selecting the appropriate dataset is an important factor in such research. The chosen datasets are dynamic, robust and applicable. This extends the range of applications that could follow the proposed algorithm.

Grid Based Multi-Density using representative points algorithm has shown better results than the other two algorithms, K-means and DBSCAN improved with K-mediods algorithm due to the following reasons: first, it takes into account the connectivity of clusters that are closely related. Second, it operates successfully on data sets with various shapes. The third reason, it doesn't depend on a user supplied model. Also, the fourth reason, it divides data space into grid using nested mean which means allows scaling to large datasets and reduces the computational complexity. Finally, it is very sensitive to two parameters, **MinPts** and **Eps** changing dynamically according to a certain factor for each cell and could handle the issue of noise and outliers.

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Improving Identity Privacy in 3GPP-WLAN

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Abstract

The mobile telephony system has become popular due to its wide coverage, resulting in near-universal roaming service to its subscribers. However, when it comes to data transfer rates, the WLANs lead the way. WLANs have much restricted hot-spot coverage compared to mobile telephony systems, but provide better data rates at lower costs. Combining the two, results in best of both the worlds. Thus, 3GPP has proposed 3GPP-WLAN architecture to bring in this synergy between the two systems. From the subscriber's perspective, one of the issues to be taken care of is "identity privacy"—how the identity of the subscriber be hidden from the eavesdroppers wanting to track the subscriber. 3GPP has devised a scheme for taking care of this in 3GPP-WLAN, which is different from that followed in other 3GPP systems. However, the said scheme does have certain vulnerabilities, and cannot guarantee the desired privacy. In this paper, we put forward an extension to the existing scheme to take care of these vulnerabilities. Also, this can be implemented without changes to intermediary networks or components, allowing for an easier transition.

Keywords

Anonymity • Privacy • 3GPP • WLAN • Anonymous roaming

Introduction

3GPP system, as standardized by the Third Generation Partnership Project (3GPP), is a telecommunication system comprising of a wired core network and a wireless radio access network [1]. Some examples of 3GPP system include Enhanced Data Rates for GSM Evolution (EDGE), Universal Mobile Telecommunication System (UMTS), etc. These provide a wide coverage resulting in near-universal roaming service, high speed mobility, efficient management and billing. Wireless Local Area Network (WLAN) technologies like IEEE 802.11 standards provide limited hot-spot coverage, but a considerably higher data rate at lower cost as compared to the 3GPP systems. An interworking of the two systems will provide the subscribers with an anytime, anywhere service (using the 3GPP systems), and a higher data rate at a reduced cost (using WLAN, wherever available). 3GPP has proposed the 3GPP System to WLAN Interworking (3GPP-WLAN) specifications to bring about this synergy [2].

Interworking of diverse systems does bring in certain concerns. One such concern is maintaining the privacy of the users visiting these diverse networks on the face of threats like location tracking and comprehensive profiling—wherein data on usage, movement, etc. of a subscriber is collected over a period of time and linked to his/her identity. This might allow an eavesdropper to collect sensitive information

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_28, © Springer International Publishing Switzerland 2015

about a subscriber. 3GPP had taken measures to address this issue of ‘identity privacy’ in 3GPP-WLAN, which is different from the scheme to protect the same in other 3GPP systems. While the scheme for ensuring identity privacy in 3GPP-WLAN seems to be more robust as compared to that in 3GPP systems [3], it has introduced new complexities, trust issues and vulnerabilities.

The Authentication and Key Agreement (AKA) protocol providing access security to a subscriber in 3GPP-WLAN is based on the Extensible Authentication Protocol (EAP) [4]. It is called the EAP-AKA protocol for cases where a WLAN User Equipment (WLAN-UE) is equipped with a Universal Subscriber Identification Module (USIM); and EAP-SIM protocol in cases where the WLAN-UE is equipped with a Subscriber Identification Module (SIM) [5]. WLAN-UE is a devise that the subscriber carries to access the service—it may be a laptop, a tablet, a smart phone, etc. The scheme to ensure improved identity privacy during execution of EAPAKA/EAP-SIM procedure expects the WLAN-UE to have unconditional trust on the WLAN Access Network (WLAN-AN), so as to share its permanent identity with the latter. With two diverse systems (3GPP and WLAN) working in tandem, and not all networks owned by the same service provider (even in case of 3GPP networks), there are scenarios where identity privacy of the subscriber might get compromised.

This paper presents an extension to EAP-AKA protocol, which can also be adapted to EAP-SIM, to reinforce the identity privacy of the subscriber. It also relaxes the requirement of the subscriber to trust the visited WLANs: this will work as an enabler for 3GPP based service providers to extend their services to any participating WLANs—trustworthy or untrustworthy.

The rest of the paper is organized as follows: section “Security Architecture” explains the security architecture of 3GPP-WLAN; section “Authentication and Key Agreement” elucidates the EAP-AKA protocol; section “Identity Privacy in 3GPP-WLAN” provides an analysis of identity privacy in 3GPP-WLAN; section “Related Work” carries a review of the literature on related work; section “Improving Identity Privacy” contains our proposed extension to achieve reinforced identity privacy in 3GPP-WLAN; section “Key Features” carries the key features of our proposal; section “Formal Analysis” presents formal analysis of the proposed extension and section “Conclusion” concludes this paper.

Security Architecture

Figure 1 shows a simplified model for accessing WLAN services by a 3GPP subscriber roaming into a WLAN hot spot. Only the key elements associated with EAP-AKA

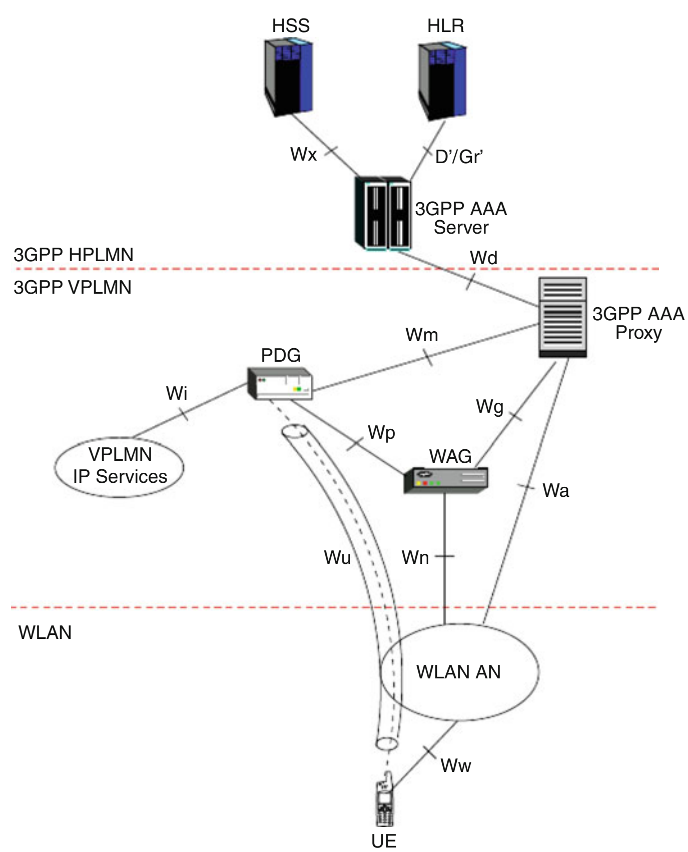

Fig. 1 Simplified roaming architecture of 3GPP-WLAN

protocol are shown. The WLAN-UE desiring to access the WLAN services will be equipped with a WLAN card, Universal Integrated Circuit Card (UICC) and necessary software applications. The permanent identity—International Mobile Subscriber Identity (IMSI)—and other long term security credentials of the subscriber are stored in the UICC. The IMSI consists of two parts—the Mobile Country Code (MCC) and a Mobile Network Code (MNC) that identifies the subscriber's home network; and the Mobile Subscription Identification number (MSIN) which uniquely identifies the subscriber within the home network.

$$\mathrm{IMSI} = \mathrm{MCC}\|\mathrm{MNC}\|\mathrm{MSIN} \tag{1}$$

Where, '||' indicates concatenation. Whenever the WLAN-UE connects to a network, the subscriber needs to be authenticated before he/she can access the regular services. The 3GPP Authentication Authorization Accounting Server (3GPP-AAA Server) located at the subscriber's Home Public Land Mobile Network (HPLMN) authenticates the subscriber using the authentication information retrieved from the Home Subscription Server (HSS) or the Home Location Register (HLR). The authentication signaling, in its passage from the home network to the visited WLAN, may be relayed through several AAA proxies, which may reside in the intermediate networks.

The subscriber may opt for two types of IP accesses: WLAN 3GPP IP access and WLAN Direct IP Address. In case of the former, the access to an IP network is provided via the 3GPP system; in case of latter, the access to IP network is provided directly by WLAN-AN. 3GPP IP based services are accessed via a Packet Data Gateway (PDG) located in the 3GPP network either at the home network (i.e. the HPLMN) or at the last visited 3GPP network, i.e. the Visitor Public Land Mobile Network (VPLMN) which the WLAN-UE had connected to before moving into the WLAN. Since the focus of this paper is on identity privacy in 3GPP-WLAN, we concentrate only on WLAN 3GPP IP access.

In WLAN 3GPP IP access, a tunnel is established between WLAN-UE and PDG to protect the user data packets transmitted between them. The PDG authorizes this tunnel using information retrieved from the 3GPP-AAA server. A gateway WLAN Access Gateway (WAG) routes the data to/from the WLAN via a Public Land Mobile Network (PLMN) to provide the WLAN-UE with 3GPP IP based services. The WAG might reside in HPLMN or VPLMN depending on whether the subscriber was in the home network, or he/she was roaming in another (visitor) 3GPP network before moving into the current WLAN. D, Gr, Wd, Wu, Ww, Wn, Wp, Wx, etc., are reference points that define the interfaces between the various network components.

Authentication and Key Agreement

The WLAN-UE undergoes two rounds of authentications with the 3GPP-AAA server to get IP access through HPLMN/VPLMN. First, it executes the EAP-AKA protocol to register with the WLAN-AN. Next, it executes EAP-AKA within Internet Key Exchange version 2 (IKEv2) protocols [6] to establish a tunnel with the PDG, which registers it, to the PLMN—the 3GPP core network. Following is an overview of EAP-AKA and the two rounds of authentications.

EAP-AKA for Registration to WLAN-AN

When the WLAN-UE visits a WLAN hot spot, it connects to the WLAN-AN using WLAN technology specific procedure. The WLAN-AN then initiates the EAP-AKA procedure to register the WLAN-UE by sending an EAP Request/Identity message to the latter. The WLAN-UE responds by sending back an EAP Response/Identity message containing the subscriber's identity in Network Access Identifier (NAI) format [7]. This identity may either be a temporary identity assigned to the subscriber's WLAN-UE in the previous authentication or, in case of first authentication, the IMSI (temporary identities are used to restrict the transmission of IMSI to provide identity privacy). The message is then routed to the 3GPP-AAA server via one or more AAA proxies using the realm part of the NAI.

The 3GPP AAA server, on receiving the message, acquires a new Authentication Vector (AV) from the HSS by producing the IMSI. In case the NAI contains a temporary identity, it extracts the corresponding IMSI from its database. The AV, which is based on UMTS AV [3], contains a random part (RAND), an authenticator token (AUTN) for authenticating the WLAN-AN to the WLAN-UE, an expected response part (XRES), a 128-bit session key for integrity check (IK), and a 128-bit session key for encryption (CK). The AUTN contains a sequence number (SQN) to ascertain the freshness of AV.

$$\mathrm{AV} = (\mathrm{RAND}, \mathrm{AUTN}, \mathrm{XRES}, \mathrm{IK}, \mathrm{CK}) \tag{2}$$

The 3GPP-AAA server then derives a new keying material Master Key (MK) using the IK and CK from the received AV. Fresh temporary identities, to be used instead of IMSI in future authentications, may also be chosen at this stage. Temporary identities are generated by mechanism explained in section "Temporary Identity Generation". These identities are protected using MK. The 3GPP-AAA server sends RAND, AUTN, a Message Authentication Code (MAC) and the protected identities to the WLAN-AN using an EAP Request/AKA-Challenge message. On receipt, WLAN-AN forwards these RAND, AUTN, MAC and the protected identities to the WLAN-UE.

The WLAN-UE runs the UMTS algorithm [3] to verify whether the received AUTN is correct. On positive verification, it computes RES, IK and CK; and derives MK to check the received MAC. If temporary identities were received, the same are stored for future authentications. It then computes a new MAC, protects it using MK, and sends EAP Response/AKA Challenge to WLAN-AN containing the newly calculated RES and MAC. The WLAN-AN forwards these to the 3GPP-AAA server in the form of EAP Response/AKA Challenge message.

The 3GPP-AAA server, on receiving the EAP Response/AKA Challenge message, verifies the contents using the UMTS algorithm. On positive verification, the 3GPP-AAA server send an EAP Success message to WLAN-AN confirming the authenticity of the WLAN-UE. Additional keying material Master Session Key (MSK) derived from MK for WLAN technology specific confidentiality and/or integrity protection may also be included in this message. The WLAN-AN uses this MSK to securely communicate with the authenticated WLAN-UE; the latter can derive MSK from its previously computed value of MK. WLAN-AN informs WLAN-UE about the success of the authentication procedure. This completes the EAP-AKA protocol which registers the WLAN-UE with the WLAN-AN, and enables secure communication among them.

EAP-AKA for Registration with 3GPP Core Network

Post successful registration, the WLAN-UE and the PDG exchange a pair of messages to establish an IKEv2 channel, in which the PDG and WLAN-UE negotiate cryptographic algorithms, exchange nonce and perform a Diffie Hellman exchange. In the remaining part of the authentication process, EAP-AKA (as explained in section "EAP-AKA for Registration to WLAN-AN") is executed through this channel. The WLAN-UE sends its identity (in NAI format, containing the IMSI or temporary identity) to the PDG via the IKEv2 secured channel. This channel can be decrypted and authenticated only by the end points (i.e., the WLAN-UE and the PDG). The PDG then sends an authentication request message to the 3GPP-AAA server, containing the user identity as received from the WLAN-UE. The 3GPP-AAA server fetches the AVs from the HSS and initiates the authentication challenge by sending an EAP message containing RAND, AUTN, MAC and protected identities to the PDG. The PDG in turn forwards the challenge along with its identity and a certificate to the WLAN-UE. The WLAN-UE checks the authentication parameters and, if found to be in order, responds to the authentication challenge. The PDG forwards this response to the 3GPP-AAA Server. When all checks are successful, the 3GPP-AAA Server sends an EAP success and a key material to the PDG. The EAP success message is forwarded to the WLAN-UE over IKEv2 channel. This completes EAP-AKA exchange for registering the WLAN-UE with the PDG (thereby with 3GPP core network), resulting in the UE and the PDG sharing a keying material derived during the exchange.

Identity Privacy In 3GPP-WLAN

In order to ensure identity privacy of the subscribers, their permanent identities the IMSI—should be adequately secured. Ideally, the knowledge of IMSI should be restricted only to the WLAN-UE and the HPLMN. In order to achieve this goal in 3GPP-WLAN, the 3GPP-AAA server generates and allocates temporary identities to the WLAN-UE in a secured way (as discussed in section "EAP-AKA for Registration to WLAN-AN"). For identity presentation, these allocated temporary identities are transmitted by the WLAN-UE instead of the permanent identity. The following subsection explains the generation of temporary identities.

Temporary Identity Generation

The temporary identities are generated from the IMSI using Advanced Encryption Standard (AES) in Electronic Codebook (ECB) mode of operation. A 128-bit secret key is used for the encryption. A particular key is, however, used only for a given period of time to generate the temporary identities. Once this period expires, a new key is used. The different keys are identified by a key indicator value. When WLAN-UE presents its temporary identity during the registration process, it also includes this key indicator value; this allows the 3GPP-AAA server to identify the key to be used for finding out the corresponding IMSI. The 3GPP-AAA server stores a certain number (set by the operator) of old keys. If a particular key is discarded, all temporary identities generated using that key stands expired.

Identity Privacy Vulnerabilities During Registration to WLAN-AN

In-spite of the aforesaid security measures, there are situations during execution of EAP-AKA protocol where the identity privacy of a subscriber may get compromised. Some such situations that arise during the WLAN-UE's registration with the WLAN-AN are as follows:

1. During the very first authentication, the IMSI is transmitted in clear text over the wireless link.
2. The temporary identities stand expired once the corresponding key at the 3GPP-AAA server is discarded. The 3GPP-AAA server will fail to identify a subscriber using an expired key. The server will then ask the WLAN-UE to furnish some other temporary identities (re-authentication identity and pseudonym [5]), and if even those fail to identify the subscriber to the server, the server will ask for the IMSI, which the WLAN-UE will comply by sending the same in clear text over the wireless link.
3. A WLAN-AN may also turn hostile and use the privacy information received by it for malafide purposes. A fake WLAN-AN may transmit beacons to WLAN-UEs in the form of spurious EAP Request/Identity message requesting the latter's IMSI to be transmitted in plain text over the wireless link.

Identity Privacy Vulnerabilities During Registration with 3GPP Core Network

During WLAN-UE's registration with PDG, the EAP-AKA message exchanges use an IKEV2 protected channel providing encryption and integrity protection. Thus, threats against identity privacy from passive attackers like eavesdroppers are significantly reduced. However, the aforesaid situations where the IMSI, instead of the temporary identities, need to be transmitted to the 3GPP-AAA server pose certain threats:

1. The protected channel, while being encrypted, is not authenticated when it receives the IMSI. The authentication happens only at the end of the EAP-AKA process. An imposter can thus pose as a genuine PDG and may request for the IMSI. While this attack will fail the authentication, the permanent identity is however, compromised.
2. The PDG, which may be situated in the visited network (VPLMN), might see the IMSI, if the same is being sent instead of the temporary identity. This allows visibility of IMSI beyond the home network.

Related Work

In mobile networks, the need to protect the identity privacy of a subscriber even from intermediary networks like the access network is well established. Herzberg et al. [8] pointed out that in an ideal situation no entity other than the subscriber himself and a responsible authority in the subscriber's home domain should know the real identity of the user. Towards this, several schemes have been proposed with each of them following a varied approach.

Many of the proposed schemes employ public key infrastructure [9, 10], but due to their processor intensive nature, such solutions are not the best of solutions for 3GPP based mobile systems, as the UE may not have high processing and power capability.

Off late, several other schemes to protect identity privacy of the subscriber from visited access network were proposed by various researchers [11–18]. However, none of these schemes are in the general lines with the AKA protocols used in mobile systems developed by 3GPP. Therefore, if any of these schemes are to be adopted, they will bring about major changes to the existing security arrangement.

For a mobile operator that already has a big subscriber base, changing over to a completely new AKA protocol is a big challenge. Moreover, the AKA protocols used in mobile systems standardized by 3GPP have already proven their efficiency with respect to other security features, through their successful real time implementations over the years. Therefore, an ideal scheme for enhanced identity privacy in a 3GPP defined mobile system would be the one that can be easily configured into the existing AKA protocol. At the same time, implementation of such a scheme should be restricted only to the operator. Intermediary networks that may even belong to third party operators should not be expected to participate equally. To the best of our knowledge, none of the previous proposals to improve identity privacy of the subscriber looks at the problem from this perspective.

Improving Identity Privacy

In this section, we propose an extension to EAP-AKA that overcomes the aforesaid vulnerabilities, and ensures end-to-end subscriber identity privacy. This extension is based on our earlier proposal for Universal Mobile Telecommunication System (UMTS) [19] and Long Term Evolution (LTE) [20]. Here the IMSI is known only to the WLAN-UE and to its HSS; it is not revealed to any third party at any point in time. We propose to replace the transmission of IMSI with a Dynamic Mobile Subscriber Identity (DMSI).

In this scheme, the HSS stores a large pool of numbers as a set of Random Number for Identity Confidentiality (RIC). Out of the pool of these RICs, some will be available for use, while the other will not (explained later). From this very big pool of numbers, any one of these available RICs is randomly picked up and securely transferred to WLAN-UE during each run of a successful EAP-AKA procedure, whereby the WLAN-UE registers (or renews it's registration) with WLAN-AN. The process of selection of RIC is sufficiently random such that there is no correlation between the current and the previously selected value. The RIC selection process at HSS takes place when it receives an AV request from 3GPP-AAA server for a particular WLAN-

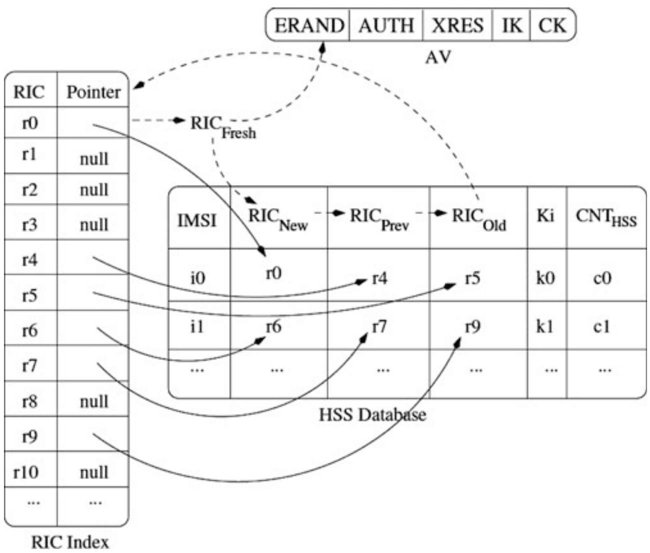

Fig. 2 HSS Database and the RIC Index

UE. A mapping between the selected RIC and IMSI of corresponding WLAN-UE is maintained at the HSS (Fig. 2). This allows unique identification of WLAN-UE through the RIC value.

DMSI, which has a transient lifetime, is generated at WLAN-UE and is a function of the value of RIC received during the last successful authentication procedure. A fresh DMSI is assembled and used whenever, as per section "Authentication and Key Agreement", IMSI needed to be transmitted. Being generated from a random number, the value of a fresh DMSI will not have any correlation with a previously generated one. Due to this, transmission of these short-lived DMSIs by any subscriber will not compromise his/her permanent identity.

The exact size of RIC is determined by the operator, and should be less than 128 bits. For an n bit RIC, there will be 2^n different possible RIC values. The temporary identities used instead of IMSI in UMTS, for example, have 32 bits. If the RIC is kept to this same size, there will be a pool of 4.29 billion unique RIC values.

A RIC-index is maintained at the HSS, storing all the available values of RIC in a sorted order. Against each value of RIC, there is a pointer which either points to an IMSI in the HSS database or is null. If this pointer against a given RIC is null, then that RIC is free and is available for use. If the pointer contains non-null value, then the RIC is not available for use, and is currently being used by some WLAN-UE.

Generation of DMSI by WLAN-UE

During each run of the EAP-AKA procedure for registering a WLAN-UE, one of the free RICs $\mathrm{RIC_{Fresh}}$—is chosen randomly from the pool for 'free to use' RICs at the HSS. The pointer against this $\mathrm{RIC_{Fresh}}$ is made to point to that record at the HSS's database which stores the IMSI of the given WLAN-UE. The $\mathrm{RIC_{Fresh}}$ is then cryptographically embedded into the RAND portion of the newly generated AV [Eq. (2)] using Ki the long term key shared between the WLAN-AN and its HSS. We call this resultant random number as Embedded RAND (ERAND). A method for embedding a RIC into a RAND is proposed in [21].

$$\mathrm{ERAND} = \mathrm{f_{e\ Ki}}(\mathrm{RIC_{Fresh}}, \mathrm{RAND}) \tag{3}$$

The AV sent to the 3GPP-AAA server has its RAND value [Eq. (2)] replaced by ERAND value:

$$\mathrm{AV} = (\mathrm{ERAND, AUTH, XRES, IK, CK}) \tag{4}$$

The 3GPP-AAA server uses this ERAND (instead of RAND in section "Authentication and Key Agreement") in the EAP Request/AKA-Challenge message sent to the WLAN-UE. As the size of ERAND is same as RAND, and it takes the position of RAND in AV, the 3GPP-AAA server will not perceive the change of RAND to ERAND, and will continue to function as before. The WLAN-UE, having the knowledge of the long term key Ki, will be able to extract RIC from ERAND to generate a fresh DMSI:

$$\mathrm{RIC} = \mathrm{f_{x\ Ki}}(\mathrm{ERAND}) \tag{5}$$

To ensure minimum computational overhead at WLAN-UE, extraction of RIC is done only when the former needs to present its identity using DMSI. In all other cases, the ERAND is used just as RAND. It may be noted that DMSI is generated only in cases discussed in section "Authentication and Key Agreement" where the IMSI is needed to be presented. The DMSI is composed as:

$$\mathrm{DMSI} = \mathrm{MCC} \| \mathrm{MNC} \| \mathrm{RIC} \| \mathrm{ERIC} \tag{6}$$

Where ERIC is generated by encrypting a padded RIC ($\mathrm{RIC_{padded}}$) with Advanced Encryption Standard (AES) algorithm using long term shared key Ki. Thus,

$$\mathrm{ERIC} = \mathrm{f_{n\ Ki}}\left(\mathrm{RIC_{padded}}\right) \tag{7}$$

And

$$\mathrm{RIC_{padded}} \mathrm{RIC} \| \mathrm{CNT_{UE}} \| \mathrm{RN} \tag{8}$$

$\mathrm{CNT_{UE}}$ is 32 bit counter at WLAN-UE which gets incremented every time a fresh DMSI is generated by the WLAN-UE, RN is 128 (32 + n) bit value, n being the number of bits in RIC. While $\mathrm{CNT_{UE}}$ is used to prevent replay attack (explained later), inclusion of RN completes the block size of 128 bits necessary to feed into the AES cipher and ensure sufficient amount of randomness to harden cryptanalysis of the cipher text. A fresh RIC reaches the WLAN-UE during every successful authentication. To take care of the stage before the very first authentication, an $ERAND_{First}$ (allowing extraction of RIC_{First}) is stored in the flash memory of USIM, before the same is handed over to the subscriber. A pointer to the corresponding IMSI is also added against this RIC_{First} value in the RIC-Index.

Extraction of IMSI from DMSI at HSS

Under this extension, the Response/Identity message received from WLAN-UE containing DMSI (instead of IMSI) is forwarded to 3GPP-AAA server by the WLAN-AN. The 3GPP-AAA server that cannot distinguish a DMSI from an IMSI forwards the DMSI along with a request for AV to the HSS. The HSS, on receiving the DMSI, extracts RIC and CNT_{UE} from RIC_{padded}, the last one being decrypted from ERIC part of DMSI using AES and the key Ki:

$$RIC_{padded} = f_{d\ Ki}(ERIC) \tag{9}$$

HSS searches this RIC in the RIC-Index to identify the IMSI of the WLAN-UE. The sorted RIC-Index allows use of efficient binary search algorithm. Once IMSI is derived, the usual EPS-AKA process at HSS, incorporating the changes suggested in Eqs. (3) and (4), can follow.

Protection Against Replay Attack

To prevent replay attack using old DMSI values, the HSS maintains a field CNT_{HSS} against every IMSI in its database (Fig. 2). HSS checks for replay attack by comparing the stored value of CNT_{HSS} with that of CNT_{UE} it just decrypted. $CNT_{UE} > CNT_{HSS}$ assures the freshness of DMSI, else the AV request from 3GPP-AAA server is rejected. Once assured of freshness of DMSI, HSS updates the CNT_{HSS} value with that of CNT_{UE}, and goes on to generate a fresh AV.

Protection Against Message Loss

To ensure robustness of our extension in the face of message losses (when a fresh RIC may not get delivered to the WLAN- UE), a given number m of previously used RICs are maintained against each IMSI. Value of m is decided by the operator based on channel quality. For instance, if $m = 3$, we store three values of RIC for every IMSI (Fig. 2). When a RIC_{Fresh} is generated, its value is stored against the corresponding IMSI in the RIC_{New} field of the database. The original value stored at RIC_{New} is moved to RIC_{Prev}, and that stored in RIC_{Prev} is moved to RIC_{Old}. The value stored at RIC_{Old} is released to the pool of unused RIC values, i.e., the pointer against that particular RIC value in the RIC-Index is replaced to null. Thus, the given IMSI, and thereby the WLAN-UE, can be identified through RIC-Index using any of the RIC_{New}, RIC_{Prev}, or RIC_{Old} values.

Key Features

Following is a summary of the key features of our extension:

1. *End to end user identity privacy*: IMSI is known only to WLAN-UE and it's HSS; it is never transmitted.
2. *Relaxed trust requirement*: As WLAN-AN or PDG are not entrusted with IMSI, the trust requirement between WLAN-AN and HSS, and between PDG and HSS is relaxed.
3. *No overhead at the intermediary network*: No computations/changes are introduced to the intermediary components like WLAN-AN, AAA Servers, PDG, etc. This scheme can be rolled out at the operator's level with only changes to be incorporated in HSS and WLAN-UE.
4. *Extension of EAP-AKA*: It can easily be implemented as an extension to EAP-AKA.

With IMSI never being transmitted at any stage of the EAP-AKA extension, all the vulnerabilities listed in sections "Identity Privacy Vulnerabilities During Registration to WLAN-AN" and "Identity Privacy Vulnerabilities During Registration with 3GPP Core Network" are eliminated.

Formal Analysis

We performed a formal analysis of the proposed scheme through an enhanced BAN logic [22] called AUTLOG [23]. Through this analysis, the following security goals are proven to be achieved.

1. IMSI should be a shared secret between the WLAN-UE and the HSS. The same should not be disclosed by the WLAN-UE to any third party including the WLAN-AN.

$$G1: \textit{WLAN-UE believes WLAN-UE} \overset{IMSI}{\leftrightarrow} \textit{HSS}$$

$$G2: \textit{WLAN-UE believes} \neg (\textit{WLAN-AN sees IMSI})$$

2. During every successful run of the EAP-AKA protocol, the WLAN-UE should receive a fresh RIC.

$$G3: \textit{WLAN-UE believes WLAN-UE has RIC}$$

$$G4: \textit{WLAN-UE believes fresh}(\textit{RIC})$$

3. It should not be possible for anyone except the HSS to map a DMSI with its corresponding IMSI.

$$\text{G5}: \textit{WLAN-UE believes} \neg(\textit{DMSI} \equiv \textit{IMSI})$$

Conclusion

In this paper, we looked into the issue of subscriber identity privacy in 3GPP-WLAN. We then proposed an extension for EAP-AKA to improve subscriber identity privacy in 3GPP-WLAN. The main contributions of this extension are: it proposes to provide end-to-end identity privacy with the real identity known only to WLAN-UE and HSS; and it proposes no change in any component present between WLAN-UE and HSS (making it easy to implement on the part of an operator). To the best of our knowledge, there is no other work which proposes to provide end-to-end identity privacy without changes in the intermediate networks and components therein.

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Performance of BFSA Collision Resolution: RFID Including Non-unique Tag IDs

Kirti Chemburkar, Zornitza Genova Prodanoff, Kenneth Martin, and Susan Vasana

Abstract

As RFID technology becomes ever more affordable, its large scale implementation has been a growing trend in recent years. While current protocols allow for non-unique tag IDs, most existing implementations largely optimize RFID system's performance based on the assumption of unique tag IDs and treat the existence of non-unique tag IDs within reader's range as a rare occurrence. Nevertheless, unless formally evaluated, it is not clear, what is the degree of performance degradation in the presence of non-unique tag IDs . We evaluate the behavior of Basic Frame Slotted ALOHA (BFSA) collision resolution for an RFID system using OPNET Modeler 14.5 as both simulation, as well as, analytical results visualization platform. The system is built assuming muting of tags by the reader and contains a mix of unique and non-unique tag IDs. Our findings are compared with results obtained from the evaluation of a similar model for a system consisting solely of unique tag IDs . The comparison of total census delay and throughput under variable frame sizes showed an increase in total census delay with an increase in number of tags and a decrease in network throughput with increase in the number of tags for the system allowing non-unique IDs.

Keywords

ALOHA collision resolution protocol • BFSA • RFID networks • Wireless networks

Introduction

Over the last few decades, supply chain management and logistics companies have successfully implemented large scale Radio Frequency Identification (RFID) systems all over the world. RFID networks are implemented to facilitate transfer of data between a reader and set of tags in order to identify, track, or locate items. The advantage of RFID technology over barcode and smart card techniques is that the reader can read tags without direct contact and beyond line-of-sight. Large scale applications are hence feasible, even when tags are read through a variety of challenging conditions such as snow, ice, fog, paint, grime, inside containers and vehicles, and while in storage [1].

RFID tags can be either passive or active. A passive tag does not require an internal power source, where active tags are battery powered. Passive tags are accessed by a reader through inductive coupling. After converting the received signal to electrical current, the on-board capacitor stores the electric current. In the case of near-field coupling, this technique is called load modulation and for far-field coupling it is referred to as back scattering [2]. Due to the versatile applicability of RFID and declining cost of implementation, companies are expected to continue implementation of this technology in large scale deployments. This trend warrants further investigation of the related protocol performance.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_29, © Springer International Publishing Switzerland 2015

Since transmission of tag IDs by multiple tags occur at the same time, a coordination scheme is required to resolve collisions. Time Division Multiple Access (TDMA) is the most frequently used technique to identify multiple tags simultaneously. In this technique, each tag uses an uplink to send data during a read cycle. TDMA for RFID has two specifications [3]:

- ALOHA protocol: this is a tag-driven stochastic TDMA protocol. In this protocol, multiple tags transmit data packets at random intervals. If data packet collision occurs, the tags wait for a random time before resending a data packet.
- Binary tree protocol: this is a reader-driven deterministic TDMA protocol. If data packet collision occurs, the reader resolves each collision one bit at a time with a binary search tree algorithm. Each tag has an ID associated with it. A reader specifies the range of IDs that must reply, while other tag IDs must not respond. If two tags transmit their ID at the same time, which results in a collision, then the reader can detect the exact bit at which the collision occurred. The reader is able to read every tag by using a sophisticated binary search tree algorithm [4].

In this paper, we address the framed slotted ALOHA protocol as it is specified or adopted in current standards that are developed and promoted by the most recognized global standardization organizations: International Organization for Standardization (ISO) [3], the International Electrotechnical Commission (IEC) [3], ASTM International [5], the DASH7 Alliance [6] and EPCglobal [7].

Each read cycle for an RFID system begins with an allotment of time for timer synchronization, followed by frame time that is divided into slots. A slot is a discrete interval of time that allows a tag to transmit its ID number along with a CRC code. The reader synchronizes tags timers to ensure no collisions are partial. A read cycle is typically defined as the time interval between two REQUEST commands and is repeated until identification of all tags in the interrogation range is completed. A REQUES message sent from the reader to all tags synchronies and prompts the tags to transmit their IDs to the reader during the specified in the REQUEST message timeslot.

FSA protocols are classified as Basic FSA (BFSA) and Dynamic FSA (DFSA) based on whether fixed or variable size frames are assumed. BFSA and DFSA are further classified as muting or early end protocols. When muting is implemented, the reader has to "silence" successfully identified tags by not allowing them to transmit their ID during the next frame. If early end is implemented, the reader closes the idle slots or the no response slots early [8].

This paper is organized as follows: a brief review of previously published results on network throughput and total census delay is presented in section "Previous Work"; section "Formal Analysis" introduces our proposed extension of the algorithms proposed in [8]; the methodology we use, model validation, and evaluation are presented in the last three sections, respectively.

Previous Work

Delays Incurred by BFSA Non-muting

Equation (1) calculates the probability of r tags responding in a slot in the ith read cycle, where N is the given frame size (slots) and n is the number of tags to be read in the ith read cycle [2].

$$p_r(i) = \binom{n}{r}\left(\frac{1}{N}\right)^r\left(1-\frac{1}{N}\right)^{n-r} \tag{1}$$

From Eq. (1), the probability of having one or more idle slots in the ith read cycle ($p_0(i)$) is calculated in Eq. (2), the probability of successful slots in the ith read cycle ($p_1(i)$) is calculated in Eq. (3), and the probability of collide slots in the ith read cycle ($p_k(i)$) is calculated in Eq. (4).

$$p_0(i) = \left(1-\frac{1}{N}\right)^n \tag{2}$$

$$p_1(i) = \frac{n}{N}\left(1-\frac{1}{N}\right)^{n-1} \tag{3}$$

$$p_k(i) = 1 - p_0(i) - p_1(i) \tag{4}$$

Then the expected number of the successful transmissions in the ith read cycle becomes $Np_1(i)$, since a read cycle has N slots [2]. Equation (5) calculates the probability of having an unread tag after R read [8].

$$p_{miss}(i) = \prod_{i=1}^{R}\left(1-\frac{Np_1(i)}{n}\right) = 1-\alpha \tag{5}$$

R represents the number of required read cycles to identify a set of tags with a confidence level α. Because the number of tags n and the frame size N are the same for all read cycles, $p_1(i)$ remains constant. As a result, the equation transforms into Eq. (6).

$$\left(1-\frac{Np_1}{n}\right)^R = 1-\alpha \tag{6}$$

Solving Eq. (6) for R, Eq. (7) provides the condition of [8]

$$R \geq \left\lceil \frac{\log(1-\alpha)}{\log\left(1-\frac{Np_1}{n}\right)} \right\rceil \tag{7}$$

The ceiling function represents the integral value of R. By using R, Eq. (8) provides the theoretical delay of success (D_{Succ_BFSA}), idle (D_{Idle_BFSA}), and collision (D_{Coll_BFSA}), where N is a frame size and T is slot duration [8]. The summation of the following three delays is the total census delay:

$$D_{Succ\_BFSA} = NRT \tag{8}$$

$$D_{Idle\_BFSA} = Np_0RT \tag{9}$$

$$D_{Coll\_BFSA} = NRT(1 - p_0 - p_1) \tag{10}$$

Delays Incurred by BFSA Muting

During muting, after the reader successfully identifies the tags, these identified tags will not respond to the reader. As a result, the number of responses reduces after each identification cycle. Equation (11) calculates the number of tags in the (i + 1)th round [2]:

$$n(i+1) = n(i) - p_1(i) \times N(i) \tag{11}$$

In Eq. (11), $p_1(i) \times N(i)$ is the number of tags identified in a read round. Based on Eq. (11), the algorithm shown in Fig. 1 calculates the total delay, collision delay, and idle delay [8].

```
BEGIN
Initialize unread tags = actual number of tags ;
Initialize non-unique tags = 0;
While true do
    Perform a read cycle for unread tags;
    Store the number of identified tags;
    Store the number of slots filled with collisions;
    Store the number of slots filled with idle responses;
    Store current frame size;
    if ( non-unique tags are identified ) then
        Non-unique tag IDs = Non-unique tags ID-ed in read cycle;
        Store non-unique tag IDs;
    end
    if (no collisions ) then
        Break;
    else
        Unread tags = actual – identified tags;
    end
end
total_delay = T × ∑stored_frame_sizes;
collision_delay = T × ∑stored_collision_slots;
idle_delay = T × ∑stored_idle_slots;
total_nu_tags = ∑non-unique_tag_IDs;
END
```

Fig. 1 Pseudo-code for non-unique BFSA muting

Delays Incurred by BFSA Non-muting Early End

If the reader does not detect any response from the tags, then the reader closes the slot in BFSA Non-Muting early end. Time t denotes the time after which the reader closes a slot and N_{Idle_early} denotes the number of no response slots. Equations (13) and (14) calculate the success delay (D_{Succ_early}) after substituting Eq. (8) in Eq. (12), and the idle delay (D_{Idle_early}) for BFSA early end [8].

The collision delay remains unchanged in BFSA-non muting early end, as the probability of collision is not dependent on slot duration.

$$D_{Succ\_early} = D_{Succ\_BFSA} - (T - t)N_{Idle\_early} \tag{12}$$

$$D_{Succ\_early} = NR(T - (T - t)p_0) \tag{13}$$

$$D_{Idle\_early} = Np_0Rt \tag{14}$$

BFSA Muting Protocol Simulation Using OPNET

A model simulation of the BFSA Muting Protocol with OPNET IT Guru 14.0 [9] measures the total census delay and network throughput. Minimum total census delay is shown to increase linearly with the number of tags. A comparison of the model simulation and analytical results from the algorithm by [8] show that results are in general agreement. Network throughput comparison for BFSA muting for analytical and simulation results has been shown to decrease slightly with the increase in the number of tags [9].

Formal Analysis

RFID Networks Including Non-unique Tag IDs

Ports and airport terminals integrate RFID technology to increase the efficiency and effectiveness of container systems. Ports use RFID technology for container tracking, wherein the reader can read tags from multiple vendors [10]. At any port/terminal, hundreds of containers arrive from around the globe with RFID tags manufactured by various vendors located around the world. It is quite possible that RFID tags manufactured by various vendors have the same tag ID, because vendors' ID code data sets are not synchronized. Existing implementations of RFID systems mostly use unique tag IDs; hence, most multi-tag anti-collision protocols do not support non-unique tag identification. For instance, Philips I-Code1 cannot identify some RFID tags, if non-unique tag IDs are present, as non-unique tag IDs will always occupy the same timeslot in every read cycle [11].

Consecutive Timeslot Number Calculation

According to the Philips I-Code1 standard [11] each tag chooses in what timeslot to transmit based on the tag ID itself, as result of which non-unique tag IDs will always generate the same timeslot index number and corresponding collisions cannot be resolved [11]. The main purpose of pseudo-random number generation in ISO 18000-6 [3] or a 16-bit slot identifier (RN16) is to increase the probability of avoiding a collision (on consecutive rounds/frames) between two or more tags whose responses contain the same tag ID (e.g. same 96-bit GID). That is, the value of the tag ID and the 16-bit random number are independent of each other. Hence, the ISO 18000-6 protocol allows the use of non-unique tag IDs [3].

Extending Existing Algorithms

This study analyzes the performance of BFSA muting for non-unique tag IDs by extending the algorithms proposed in [8] in order to determine total delay, collision delay, idle delay, and the total time to identify all tags in the system. Without extending latest protocols (e.g. [3]) to keep track of multiple instances of the same tag ID, an RFID system using the BFSA non-muting technique is not practical, as the reader cannot determine whether it is the same tag that is responding in consecutive cycles or if there exists another tag with the same tag ID.

Figure 1 shows pseudo-code for the extended algorithm to determine the total time required to find non-unique tag IDs using the BFSA muting technique. Total census delay is computed by summing up the frame sizes from each round during the duration of the census and multiplying them to the slot duration. Total census delay is computed by not factoring in the time it takes to keep track of multiple instances of tag IDs. Time T corresponds to the duration of a slot. The time complexity depends on the number of read cycles it takes to read all n tags. For example, if R is the duration of a single read cycle and m is the number of cycles required to read n tags, then the time complexity for the first algorithm is the total number of read cycles $O(mR)$. For the same RFID system, R will be a constant, but m will vary with number of RFID tag IDs.

Methodology

Evaluation of BFSA with Muting

The analytical model built as part of this study with OPNET Modeler 14.5 analyzed the Basic Frame-Slotted ALOHA protocol supporting non-unique tag IDs of a passive RFID network system. This model is a limited simulation of a real-world RFID system, as we assume no tags or readers ever fail. We also assume there are no signal anomalies caused by the environment, such as attenuation and distortion. The sample size for the BFSA muting RFID system varied from 10 to 100 tags. The model simulated the behavior of non-unique tag IDs , by implementing the BFSA protocol to predict the total census delay and network throughput. Our simulation results were compared against a previous findings assuming unique tag IDs [9].

Evaluating Delays

The total census delay consisted of success delay, collision delay, and idle delay [12]. Equation (15) calculated the slot delay duration T in seconds (s), where ID (bits) is the size of the packet containing the tag's ID and data_rate (bps) is the rate of data transfer from the tag to the reader.

$$T = \frac{\text{ID (bits)}}{\text{data\_rate (bps)}} \tag{15}$$

Equation (16) calculated the total census delay [8], where T is slot duration defined by response packet size by data rate and $\sum$ stored_frames is the sum of all frame sizes.

$$\text{Total delay} = \text{T} \times \sum \text{stored\_frames} \tag{16}$$

Evaluating Network Throughput

A ratio between the number of successfully transmitted packets (one per tag) and total number of packets sent by tags during the census defines network throughput [13]. The model measures the number of required read cycles and the number of unread tags. These parameters determine the total number of packets (tag IDs) transmitted during the census. Equation (17) helps calculate network throughput, where $S[n]$ is network throughput, n is total number of identified tags, α is the assurance level, and $P[n]$ is the total number of packets sent by tags during the census.

$$S[n] = \frac{\alpha n}{P[n]} \tag{17}$$

The assurance level is the probability of identifying all tags in the reader's interrogation range [14]. A value of 0.99 for α in the model indicates less than 1 % of the tags are missing.

Verification of the Model

BFSA Muting with Non-unique Tag IDs

Our model is verified by recording execution steps and corresponding parameters for a small set of tags and a single reader for the sake of visualization simplicity. At the beginning of the census, the number of unread tags is initialized to the actual number of tags in the range. Values that are stored include: the number of identified tags, the slots with collision (s), the idle slots, the current frame size, and the number of non-unique tag IDs (if any). If there are no more collisions in a given frame, the census is assumed to have completed and the last steps in the algorithm are used to compute total census delay, collision delay, and idle delay.

Figure 1 shows the pseudo code for the RFID system with non-unique tag IDs in BFSA muting. The time required for frame transmission can be calculated using a given frame size and data rate as shown in Table 1.

Table 1 List of model parameters

Parameter	Value
Assurance Level	0.99
Data rate (between reader and tags)	500,000 (bps)
Frame size (number of slots in a read cycle)	8
Duration of single slot	0.00016 s
Duration of read cycle	0.00128 s
REQUEST packet size (from reader to tag)	88 bits
SELECT packet size (from reader to tag)	72 bits
RESPONSE packet size (from tag to reader)	80 bits

Figures 2, 3, and 4 show the sequence of events during the BFSA muting for non-unique tag IDs . A frame consists of eight slots. At the beginning of the census, the reader broadcasts the REQUEST packet to all the tags. As seen from Table 2, the transmission delay for a REQUEST packet is 0.000176. The propagation delay is assumed to be negligible.

On receiving the REQUEST packet, the tags synchronize their timers to avoid partial collisions. Tags are allowed to transmit only once per read cycle, in accordance with a classical implementation of the FSA protocol. Tags randomly select one of the slots to transmit a RESPONSE packet. Each tag has an ID, which could be non-unique or same as the IDs of one or more other tags. The label "unique ID" is not used by the collision resolution protocol, but is merely assigned to tags to facilitate visualization. When multiple tags transmitted their tag ID's to the reader within the same slot, a collision occurred and the reader could not identify any tags.

Figure 2 shows that two collisions occur in the first read cycle. Three tags (IDs: 1, 2, and 6) (Unique ID: 2, 3, 13) transmit their IDs by occupying the second slot and two tags (IDs: 1 and 7) (Unique ID: 4, 17) transmit by occupying the third slot. Because of the collision, the second slot and third slot are rejected. As can be seen in the fourth, sixth, and seventh slots, the reader successfully identifies a single tag transmission without a collision. The first, fifth, and eighth slots are idle slots in the first read cycle. After the end of a read cycle, the reader computes and stores the number of identified tags, number of non-unique tag IDs , slots with collisions, and idle slots. If there are no collisions in the read cycle, then the reader completes the census.

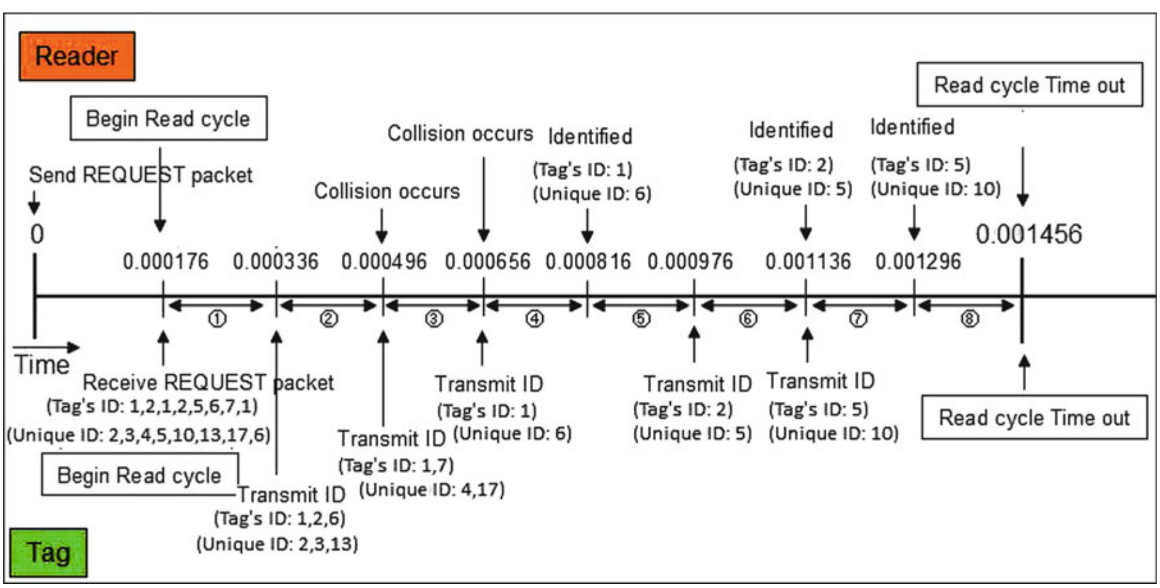

Fig. 2 First read cycle

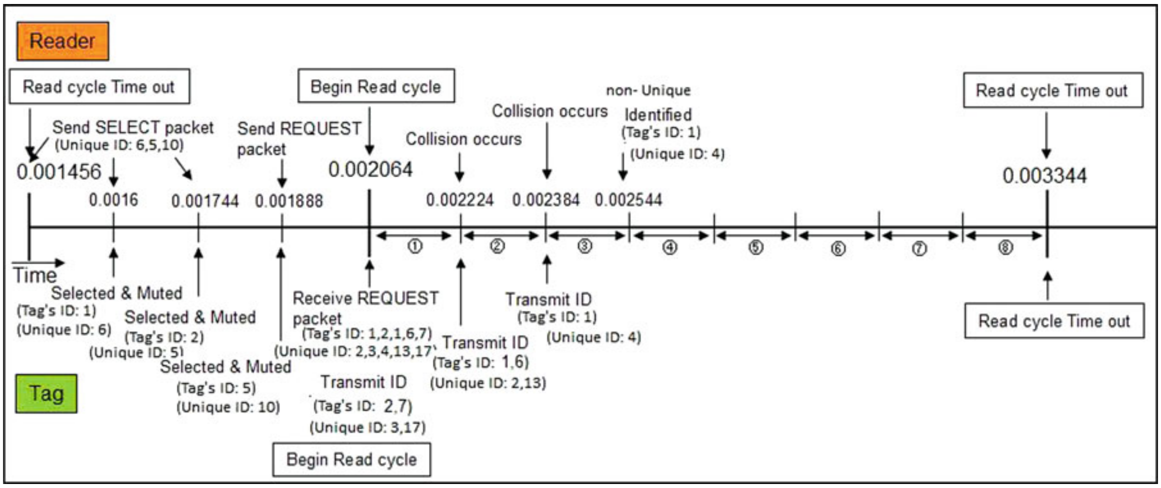

Fig. 3 Second read cycle

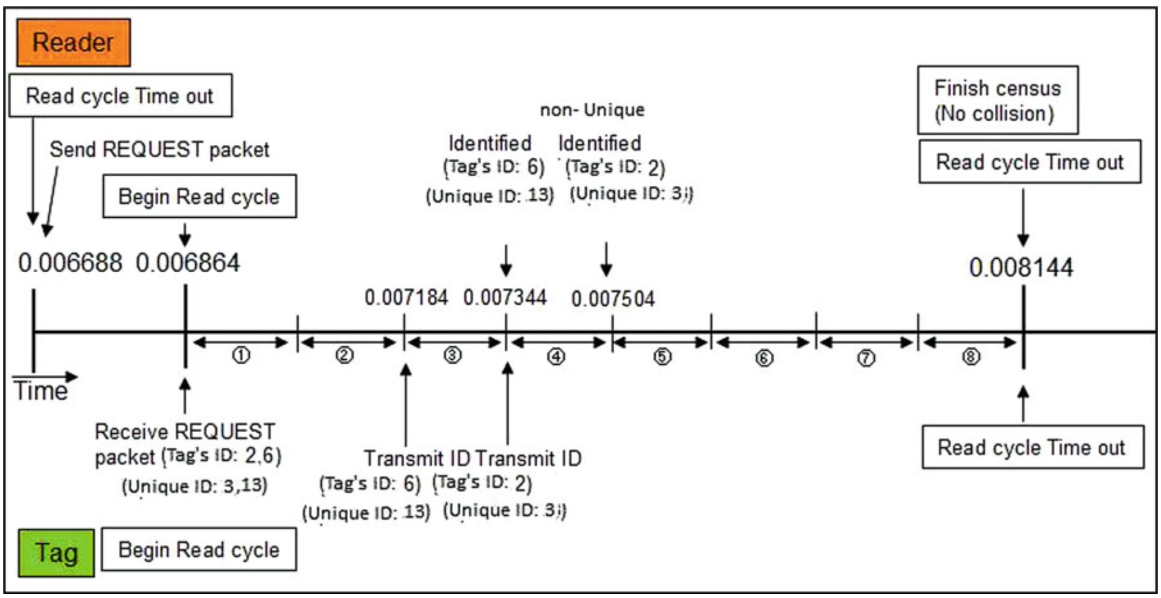

Fig. 4 Fifth read cycle

Table 2 Packet transmission time

List	Value
REQUEST packet	0.000176 s
SELECT packet	0.000144 s
RESPONSE packet	0.00016 s

For the identified tags, the reader transmitted SELECT packets with unique IDs as shown in Fig. 3. As soon as a tag with a matching unique ID receives the SELECT packet, the reader stops the tag from transmitting the tag ID by muting the tag. The reader transmits the SELECT packets to three identified tags from the previous read cycle, as shown in Fig. 3. Then, the reader broadcasts the REQUEST packet to tags. Only, unread tags will respond to the REQUEST packet.

Figure 3 shows two collisions occurred in the second read cycle. Two tags (IDs: 2 and 7) (Unique ID: 3 and 17) transmit their IDs by occupying the first slot and two tags (IDs: 1 and 6) (Unique ID: 2 and 13) transmit by occupying the second slot. Because of the collision, the first slot and second slot are rejected. In the third slot, the reader successfully identifies a single tag transmission without a collision. As the tag identified in the third slot has a non-unique tag ID to a tag previously identified, the reader updates the non-unique tag IDs count.

In the third, fourth, and fifth read cycles, on delivery of a REQUEST packet to the tags, the read cycle begins. The procedure for transmitting tag ID's, the detection of a collision, and the identification of a non-unique or unique tag is the same as in the previous read cycle. Figure 4 shows the fifth and last read cycle where no collisions occur so the census finishes after all tags are successfully identified.

Evaluation

This study compares simulation results with the results from a previous study for unique tag IDs [9]. We perform the evaluation under the premise that the scope of this work is theoretical, assuming "ideal conditions." For the purposes of this study "ideal conditions" imply a fixed frame size and fixed slot duration for the duration of the census; no failing tags or readers; no signal anomalies caused by the environment, such as attenuation, and distortion. We relax the assumption for tag ID uniqueness by having a (rather pessimistic) mix of 40 % vs. 60 % non-unique vs. unique tag IDs, respectively. Since there are no known closed form solutions for the computations in this section, we arrive at the results iteratively, using the algorithm presented in section "Formal Analysis".

Total Census Delay

In this evaluation we compute census delay for the case of BFSA with tag muting. The total census delay depends on frame size and the actual number of tags. The total census delay would be longer, if the frame size is larger or smaller than the optimal. If the frame size is larger, there would be too many idle slots resulting in a longer total census delay. Alternatively, if the frame size is too small, there would be too many collisions resulting in a longer total census delay [15]. The calculation of minimum total census delay requires execution of a census assuming implementation of optimal frame size. As stated in section "Formal Analysis" already, unlike the case of non-muting operation, when muting is implemented, there is no closed form solution and optimal frame size has to be determined iteratively. The results that we obtain are hence suboptimal. In order to determine the optimal frame size for a given number of tags, we vary the frame size from 10 to 120 slots, in increments of five, and execute the algorithm for ten censuses with each frame size value [9]. Tag sets are with sizes from 10 to 100 tags, using an increment of five.

Since variability in the values of performance evaluation metrics can have a great impact on network performance, we compare the variability between the two populations of non-unique and unique tag IDs in all tests described in this section. We first performed a two-sample F-test, a statistical test for variance that compares the variance between the minimum total census delay for non-unique tag IDs and unique tag IDs We chose to perform a one-sided *p*-value

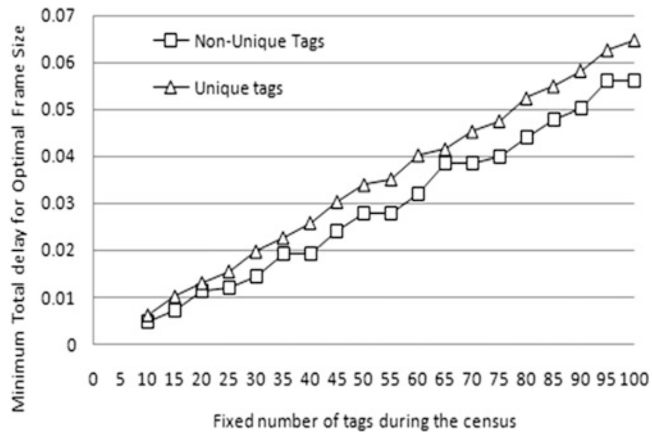

Fig. 5 BFSA muting: min. total census delay

Table 3 F-test for total minimum census delay for non-unique and unique tag IDs

	Non-unique	Unique
Mean	0.03006316	0.035869
Variance	0.00027145	0.000334
Observations	19	19
df	18	18
F	0.81279457	
P(F ≤ f) one-tail	0.33241392	
F Critical one-tail	0.45101989	

test, since the expectation is that the non-unique tag IDs variation in results would have a larger mean. We assumed two independent random samples and Gaussian populations (normally distributed). The reader should note that if this assumption of normality does not hold, they might need to perform other statistical tests.

Figure 5 shows the minimum total census delay increases linearly with the number of tags for both BFSA muting with non-unique tag IDs and BFSA muting tags with unique tag IDs. Our results are in general agreement with the results from [9] and the slight difference in delay could be attributed to the choice of suboptimal frame size.

In all tests described in this section, the null hypothesis states that the two population variances are equal and the alternative hypothesis is non-unique tag IDs have greater variance in results. Assume the level of significance α as 0.05. The F-test calculates the F-statistic, F-critical value, and p-value. According to the F-test results, when F-statistic < F-critical and p > α, then the variance are equal for the two samples.

Table 3 shows the F-test for variance results for minimum total census delay for non-unique tag IDs and unique tag IDs. Since F-statistic > F-critical (0.81279457 > 0.45101989), using a lower tail critical value, and p > α (0.33241392 > 0.05), we fail to reject the null hypothesis.

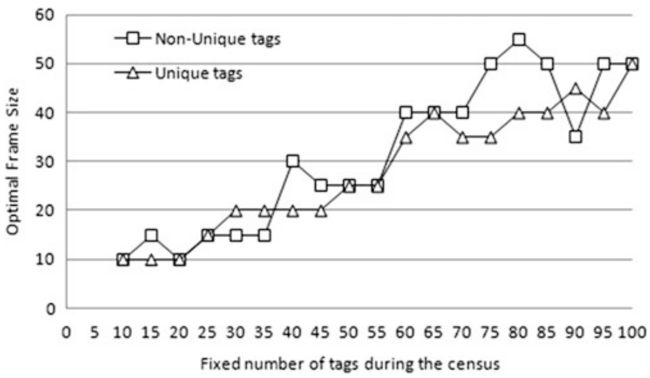

Fig. 6 Optimal frame size for BFSA muting

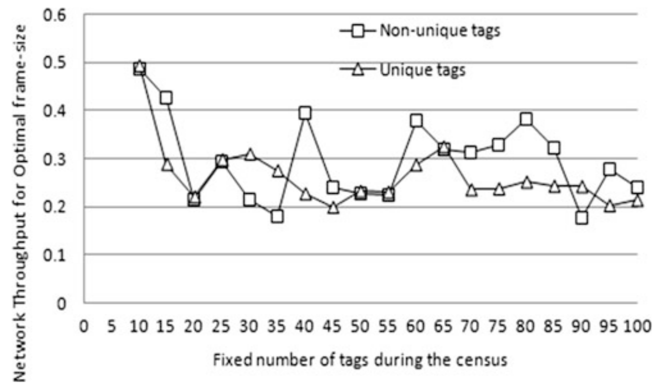

Fig. 7 Network throughput when frame size is optimal for BFSA muting

Table 4 F-test for optimal frame size for non-unique and unique tag IDs

	Non-unique	Unique
Mean	31.3157895	28.15789
Variance	238.450292	161.6959
Observations	19	19
df	18	18
F	1.47468354	
P(F $\leq$ f) one-tail	0.20892521	
F Critical one-tail	2.21719713	

Table 5 F-test for network throughput for non-unique and unique tag IDs

	Non-unique	Unique
Mean	0.29680101	0.263573
Variance	0.00757878	0.004417
Observations	19	19
df	18	18
F	1.71574161	
P(F $\leq$ f) one-tail	0.13077254	
F Critical one-tail	2.21719713	

With a p-value of 0.33, we cannot reject the assumption that the variance in a population of unique tag IDs is equal to the variance in a population of non-unique tag IDs. Figure 6 shows a comparison of optimal frame size for BFSA muting allowing non-unique tag IDs and BFSA muting with unique only tag IDs .

Figure 6 shows that optimal frame size increases approximately linearly with the number of tags, irrespective of the non-unique tag IDs or unique tag IDs type of protocol.

Table 4 contains F-test results on variance assuming optimal frame size and a census including non-unique tag IDs vs. a census assuming only unique tag IDs. Since F-statistic < F-critical (1.47468354 < 2.21719713) and p > α (0.20892521 > 0.05), we failed to reject the null hypothesis. With a p-value of about 0.21, we cannot reject the assumption that the variance in optimal frame size in a population of unique tag IDs is equal to the variance in a population of the non-unique tag IDs .

Network Throughput

This sub-section describes the evaluation of network throughput when frame size is optimal: maximum throughput, minimum throughput, and average throughput.

Figure 7 shows network throughput when frame size is close to optimal for runs including non-unique tag IDs (as chosen iteratively) and optimal when assuming unique only tag IDs (as calculated based on results from [15]). Network throughput decreases with the increase in number of tags for non-unique tag ID and unique tag ID versions of the protocol. Table 5 presents the F-test for variance results for network throughput when frame size is optimal for unique and non-unique tag IDs . Since F-statistic < F-critical (1.71574161 < 2.21719713) and p > α (0.13077254 > 0.05), we, failed to reject the null hypothesis. With a p-value of 0.13, we cannot reject the assumption that the variance in throughput when frame size is optimal in a population of unique tag IDs is equal to the variance in a population of non-unique tag IDs .

Network throughput depends on the total number of read cycles and the number of tag IDs transmitted. When frame size varies for a fixed number of tags, the number of read cycles required to complete the census varies, which results in fluctuation of throughput. Hence, there is a need to evaluate the minimum throughput, the mean throughput, and the maximum throughput. Figures 8, 9, and 10 show minimum throughput, mean throughput, and maximum throughput for BFSA muting with non-unique tag IDs and BFSA muting with unique tag IDs with frame size varying from 10 to 120. Figures 8, 9, and 10 show throughput gradually decreases with an increase in the number of tags for BFSA muting with non-unique tag IDs , while in the case of unique tag IDs there

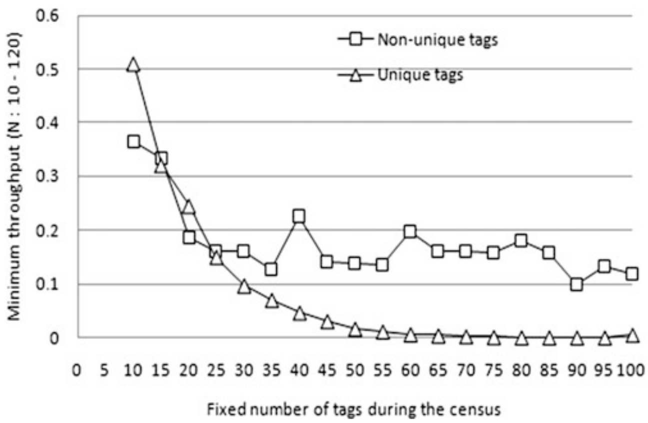

Fig. 8 BFSA muting: minimum network throughput

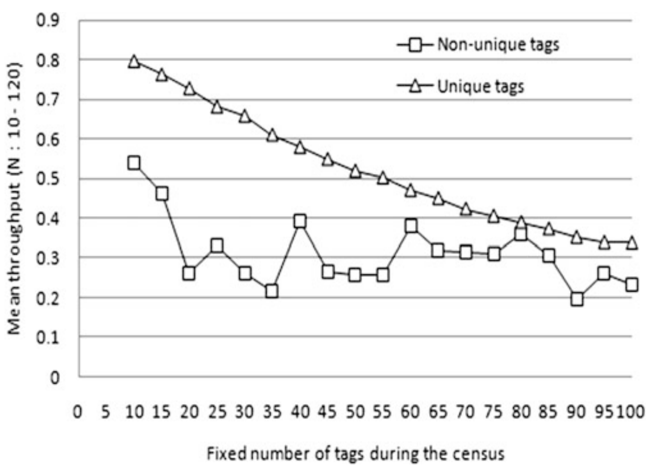

Fig. 9 BFSA muting: mean network throughput

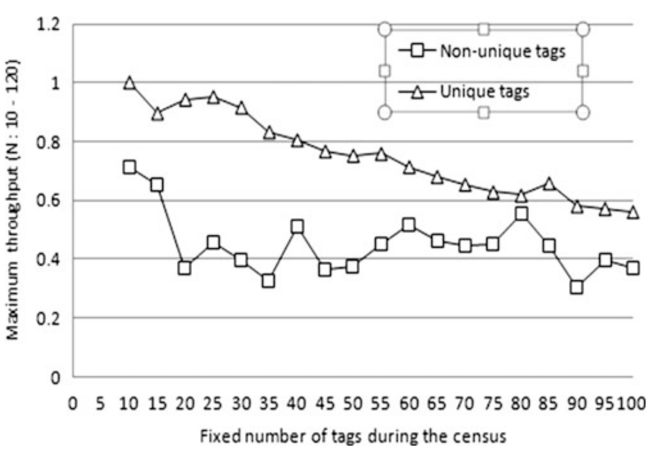

Fig. 10 BFSA muting: max. network throughput

is a rapid decrease in throughput with an increase in the number of tags.

Table 6 shows the F-test for variance results for minimum network throughput for non-unique tag IDs and unique tag IDs. Since F-statistic < F-critical (0.24034048 < 0.45101989) and p < α (0.00204818 < 0.05) the variance of non-unique tag IDs differs from unique tag IDs for minimum network throughput, we can reject the null hypothesis. The two populations show a statistically significant difference in the variance in minimum network throughput.

F-test results for mean network throughput for non-unique tag IDs and unique tag IDs are summarized in Table 7. We can reject the null hypothesis, since F-statistic < F-critical (0.33422389 < 0.45101989) and p < α (0.01255299 < 0.05). The two populations show a statistically significant difference in the variance in mean network throughput.

The F-test for variance results for maximum network throughput for non-unique tag IDs and unique tag IDs are presented in Table 8. Since F-statistic > F-critical (0.5693935 > 0.45101989) and p > α (0.1208633 > 0.05) we failed to reject the null hypothesis. With a p-value of 0.12, we cannot reject the assumption that the variance in maximum throughput in a population of unique tag IDs is equal to the variance in a population of non-unique tag IDs.

Table 6 F-test for minimum network throughput for non-unique and unique tag IDs

	Non-unique	Unique
Mean	0.17569229	0.080122
Variance	0.00457054	0.019017
Observations	19	19
Df	18	18
F	0.24034048	
P(F ≤ f) one-tail	0.00204818	
F Critical one-tail	0.45101989	

Table 7 F-test for mean network throughput for non-unique and unique tag IDs

	Non-unique	Unique
Mean	0.31175584	0.521476
Variance	0.00741211	0.022177
Observations	19	19
df	18	18
F	0.33422389	
P(F ≤ f) one-tail	0.01255299	
F Critical one-tail	0.45101989	

Table 8 F-test for maximum network throughput for non-unique and unique tag IDs

	Non-unique	Unique
Mean	0.45136788	0.752328
Variance	0.01102014	0.019354
Observations	19	19
df	18	18
F	0.5693935	
P(F ≤ f) one-tail	0.1208633	
F Critical one-tail	0.45101989	

Conclusion

We evaluated the performance of the BFSA protocol for RFID, assuming support for non-unique tag IDs . Our model is implemented using OPNET Modeler 14.5. Its accuracy is verified using a detailed simulation log. To study the effect of non-unique tag IDs on RFID performance, we compared the simulation results of the BFSA muting protocol supporting a mix of non-unique and unique tag IDs with BFSA muting protocol assuming unique tag IDs only [9].

Comparing minimum network throughput, mean network throughput, and maximum network throughput for variable frame sizes showed a general agreement with the results from [9]. The statistical analysis of the results showed a significant difference between the mix of non-unique and unique tag IDs population vs. unique only population in the results obtained for minimum network throughput and mean network throughput.

The mean network throughput and maximum network throughput is slightly worse for census procedures assuming a mix of non-unique and unique tag IDs in comparison to querying sets of tags consisting solely of unique IDs.

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Assessing Software Project Management Complexity: PMCAT Tool

Vyron Damasiotis and Panos Fitsilis

Abstract

Software projects are complex endeavors that quite often fail to satisfy their initial objectives. As such the need to systematically study and assess the complexity of software projects is quite important. This study presents a systematic framework for assessing complexity of software projects that is based on the study of project management subject areas as defined in Project Management Body of Knowledge (PMBOK). The presented framework is based on a model that combines the concepts of project, complexity model, complexity factor etc. is an attempt to systematically assess and compare the complexity of software projects. The whole concept has been implemented within Project Management Complexity Assessment Tool (PMCAT) and it is available as a software service over the web.

Keywords

Project management • Software projects • Project complexity • Complexity measurement

Introduction

Even though Project Management (PM) has received a lot of attention from industry and academia, a great number of projects still fail to meet their requirements in terms of time delay, cost overrun and quality restrictions [1–3]. Most of these failures have been attributed to the complexity of the projects. Many studies have been undertaken the last years in order to understand, define and determine the concept of project complexity [4–9].

Software projects are among the most complex ones. Many studies on various types of software project have proven that their outcomes are far from the complete fulfillment of the initial requirements [10–12]. Most studies measure complexity either by measuring the software project product based on its attributes such as size, quality, reliability or the characteristics of software project process using attributes such as performance, stability, improvement [13–15]. As such the need to establish a systematic way to evaluate the software project complexity is important.

The structure of this paper is as follows. Section "Literature Review" presents a short introduction to PM, the structure and the typology project complexity. In section "Software Project Complexity", the focus is moved on software projects and an attempt is made to establish a connection between project management complexity and software project failures and to prove need for complexity assessment in these projects. In section "Modeling Project Management Complexity and PMCAT Tool" our approach towards to modeling the project complexity is presented. Further a short presentation of PMCAT tool is given, along with the description of the requirements that was set and the design approach was followed. Finally, conclusions and implications for future work are discussed.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_30, © Springer International Publishing Switzerland 2015

Literature Review

Project Management

Several project management frameworks have been developed so far three of which gained globally acceptance from industry and academia, Project Management Body of Knowledge (PMBOK) by Project Management Institute (PMI) [16], IPMA Competence Baseline (ICB) by International Project Management Association (IPMA) [17] and Project IN Control Environments (PRINCE2) by the Office of Government Commerce (OGC) of British government [18].

ICB proposed by IPMA describes in detail the competences that are required for project management. These competencies are classified in three main categories, which are the Technical competencies, the Behavioral competences and the Contextual competencies [16]. ICB is not process based and is focused on skills, tasks and activities.

PRINCE2 is used widely in UK and was introduced by OGC in 1996. It was fundamentally revised in 2009 in order to adapt to changes in the project and business environment, address weakness and adjust with other OGC methods. PRINCE2 is process driven PM framework which is very prescriptive and provides the necessary techniques and templates for project manager to apply.

PMBOK is globally accepted as the main standard for PM both from companies and from organizations such as IEEE and ANSI [19–21]. PMBOK identifies nine areas in project management which are: Project Integration Management, Project Scope Management, Project Time Management Project Cost, Project Quality Management, Project Human Resource Management, Project Communication Management, Project Risk Management, and Project Procurement Management [16].

In terms of PM context, PMBOK and PRINCE2 are focusing on "hard" skills such as processes, procedures and techniques, while the ICB is focused on "soft" skills which are related to human behavior such as leadership, motivations etc. Winter et al. [22] states that project management thinking is focused on "hard" aspects which emphasizing more in planning and control rather than in "soft" skills and this is represented and in dominant PM frameworks. On the other hand the importance of "soft" skills such as leadership, social conduct and interaction between project stakeholder and active participation and accountability are revealed in researches [23–25]. PMBOK is chosen as the base PM framework in this research as although it provides project managers with a pool of procedures and techniques, does not provide any templates on them, which means that leaves the freedom in project managers to apply them according to their experience, hence, leave a port open and for PM "soft" skills.

Project Complexity

Complexity is part of our environment and appears in different domains. Many scientific fields have dealt with complex systems and have attempted to define the term complexity according to their domain. This implies that there is a different definition of complexity in computational theory, in information theory, in business, in software engineering etc. and many times there are different definitions inside the same domain [26], Schlidwein and Ison [27] states that are two major approaches of complexity. The first one describes the complexity as a property of a system, called descriptive complexity. The other approach describes complexity as perceived complexity and translates it as the subjective complexity that someone experiences through the interaction with the system.

Complexity in PM attracted significant attention from researchers during the last decade [4, 8, 28–30]. However there is no consensus on what really complexity is [26, 31]). Many project managers cannot distinguish the difference between complex and complexity [32]. However a definition of project complexity should at least contain interaction, structural and dynamic elements [32].

Complexity Typologies

This lack of consensus in defining what project complexity is has resulted in a variety of approaches on classifying project management complexity. One of the first researches that deal with the concept of complexity was Baccarini [33]. He considers complexity as something "consisting of many varied and interrelated parts" and operationalized them in terms of "differentiation" the number of varied elements (e.g. tasks, components) and "interdependency" the degree of interrelatedness between these elements. Finally he describes four types of complexity: a) organizational complexity by differentiation and b) organizational complexity by interdependency c) technological complexity by differentiation and d) technological complexity by interdependency.

Extending the work of Baccarinni, Williams [34] added the dimensions of uncertainty in projects and the multi-objectivity and multiplicity of stakeholders. The definition of project complexity according to Williams is divided in structural complexity sourcing from number and interdependence of elements and uncertainty sourcing from uncertainty in goals and methods.

Xia and Lee [29] focused on Information System Development Projects (ISDP) and portrayed complexity of these type of projects by defining two types of complexity: a) organizational complexity and b) technological complexity described under two dimensions the structural dimension and dynamic dimension. They proposed an ISDP complexity model in order to measure complexity in IS development projects.

Geraldi and Adlbrecht [5] and Geraldi [35, 36] based on structural complexity and uncertainty defined three types of complexity, the complexity of faith (CoFaith), Complexity of Fact (CoFact) and Complexity of Interaction (CoInt).

Maylor et al. [7] focused on perceived managerial complexity under two dimensions structural and dynamic and identified five aspects of complexity. They defined a complexity model that is based on Mission, Organization, Delivery, Stakeholders and Team (MODeST).

Vidal et al. [30], studied project complexity under the organizational and technological dimensions and identified four aspects for studying project complexity: project size, project variety, project interdependence and project context.

Bosch-Rekveltdt et al. [4], proposed a framework for "Grasping complexity in large engineering projects", in which they identify 3 categories of project complexity the technical complexity, organizational complexity and environment complexity (TOE) and 14 subcategories.

Sedaghat-Seresht et al. [37] identified five dimensions of complexity the environmental, organizational, objective, stakeholder, task, technology and information systems complexity. They also used "DEMATEL" method to identify the relationships between project complexity dimensions.

A step further is made in the studies of Xia and Lee [29], Vidal et al. [30], Bosch-Rekveltdt et al. [4], Sedaghat-Seresht et al. [37] who not only attempted to identify project complexity dimensions but as well to propose models for measuring project complexity.

Software Project Complexity

Software Project Complexity

Complexity in software projects are quite similar with projects in other domains with regard to the factors that influence it, for example the tools, the processes, the restrictions to name a few [38]. Hughes et al. [39] and Kiountouzis [40] state that software projects differ since are immaterial, complicated, supple and technology dependent. Furthermore, Xia and Lee [29] state that information systems projects "are inherently complex because they deal not only with technological issues but with organizational factors largely beyond the project team's control".

Regarding the top ten factors that lead to project success or to project challenged and failure as they described in various in studies like "Chaos report" [10, 11] and "why software fails" [12], it is obvious that most of them identify project management aspects among them, both in terms of "hard" and "soft" management aspects. Indicatively, are referred as success or failure factors, factors that are related to proper planning, requirements management, scope management, risk management, procurement management, communication management, human resource management, executive management support, user involvement and technology related issues [10, 12]. Most of these failures have been attributed to the complexity of the projects. Project complexity lead to project failure because either complexity is very high [9, 41], either project complexity has been underestimated [42]. Considering the above, it is obvious that many failure factors would have been restrained, if not eliminated, if there was an indication of project complexity level, so properly management techniques would have been applied to handle this complexity. The relationship between project complexity and PM has already started to be investigated by researchers [43].

Consequently, in order to reduce the possibility of project challenge or failure, caused by complexity, we have to control it. The first step to succeed this is to know the level of expected complexity by measuring it. Whitty and Maylor [32] propose the use of complexity as a metric, to measure complexity in a system. So complexity should be deal as a variable that we should to measure, if we want to be helpful in PM. In this case we can use complexity as thermometer by developing—"complexometer" [35] and in the question "How complex is this project?" to reply "Its complexity is. . ..." [32].

Even though studies have proposed methods and techniques to measure project complexity [4, 29, 30, 37], the need for a more practical framework that combines a project complexity measurement model and an automated tool to support it, is evident.

Complexity Metrics

A key element for developing an effective, efficient and trustworthy complexity measurement framework is the determination of the appropriate metrics. Metric is a property of a project that can be measured and recorded as a number, a percentage, a count or a rating scale [44]. The determination of good metrics allow the proactive rather than reactive project management and hence can reduce complexity stemming from projects process faults and dysfunctions. Metrics should satisfy a number of requirements such as to be reliable, repeatable, easy to use, consistent and independent in order to be useful. Metrics and tools in software projects are emphasizing in measuring the software product or software development process mainly

Table 1 Complexity factors and metrics

Complexity factors	Indicative metrics
Project Integration Management	
Software requirements volatility	Number of changes requests
Project novelty	Number of similar software projects within organization Number of similar software projects in the literature
Project Environment	Legislation and regulations Competition in market
Project Scope Management	
Software size	Metrics such as FPA and UCP Number of deliverables
Software structure and architecture	Number of components and modules Number of different technologies used within project
Project Time Management	
Project Schedule	Project duration Number of activities
Project schedule difficulty	Activities Synchronization Number of critical activities Number of constrains
Project Cost Management	
Cost estimation	Availability of cost estimations data Sufficient time for cost estimations
Level of accuracy	Quality and level of detail of tender documentation
Cost planning	Size of project Duration of project
Project Quality Management	
Quality planning	Number of quality management procedures Importance of quality in relation to cost and schedule objectives
Project Human Resource Management	
Team structure	Number of stakeholders Geographical distribution of team members Cultural differences Personnel availability
Project Communication Management	
Communication planning	Reporting frequency Expected average percentage of labor spent in communication procedures
Project Risk Management	
Risk Planning	Number of Risks identified
Project Procurement Management	
Procurement planning	Number of procurement orders Number of new external contractors
Procurement execution	Number of different organizations involved

Table 2 Example of metric's scale

Complexity metric	Very low (1)	Low (2)	Moderate (3)	High (4)	Very high (5)
Expected average percentage of labor spent in communication procedures	Up to 1 %	Up to 3 %	Up to 5 %	Up to 10 %	>10 %

and fragmentally deal project management as an entity. The most widely known measurement models are focusing on software development cost estimation (e.g. COCOMO II [45]), or on analyzing the properties of software (e.g. McCabe's cyclomatic complexity [46]) Our approach in assessing project complexity is holistic and proactive, which means that we want to build a framework for assessing the project complexity affecting all areas of PM at the beginning of the project and provide by that way the project manager with the appropriate information in order to successfully manage it. To succeed that, an extensive literature review is performed and properties of projects and properties of project management process are identified in order to be determined the appropriate complexity factors and metrics. These metrics will be evaluated by experts for their contribution in total project complexity and will be prioritized. Indicative measures for each of the PMBOK subject areas are presented in Table 1 [14].

Further, the definition of the evaluation scale for each metric is an important component in the measurement process. It should clearly define the boundaries and the scale of each metric. An example of an evaluation scale for the metric "Expected average percentage of labor spent in communication procedures" can be seen in Table 2.

Modeling Project Management Complexity and PMCAT Tool

One of the main objectives for the design of Project Management Complexity Assessment Tool (PMCAT) was to design an overall software service that will allow project managers to experiment, develop their own complexity models, if needed, and to apply these models to evaluate projects. The intention is to use this tool as a collaborative tool for the PM community either for complexity model development and validation or for project complexity assessment.

Five basic concepts/entities were used, namely: *Project*, *Model*, *Complexity Factor*, *Metric* and *Evaluation Scale*. The *Project* entity is used to describe each project under evaluation. Each project is evaluated by the use of one or more models. These models are custom developed models, for the needs of the specific project, or are reused from a set of available to the PM community models.

Each *Model* is composed of a number of *Complexity Factors*, factors that are combined with a unique way for the needs of specific project or for categories of projects. In every *Model*, each *Complexity Factor* is correlated with a specific weight that represents the contribution of this factor to the project complexity. It is not unlikely that the same factor has different weights when it is participating in different models. The calculation of the *Complexity Factor*'s

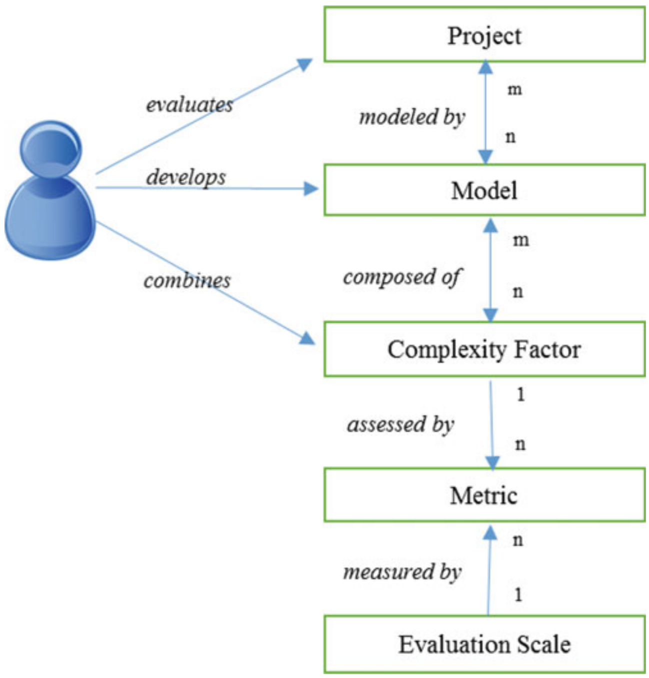


Fig. 1 PMCAT model

weight is outside the scope of this paper, however it is done with the use of statistical methods and group decision techniques.

Similarly, a *Complexity Factor* is correlated with a number of *Metrics*. In the simple case a *Complexity Factor* corresponds to a *Metric*, however it is not uncommon to have composite Complexity Factors that require more than one Metric to be measured. Finally, an evaluation scale is used to indicate how is Metric is measures and can be numerical, ordinal, scale, yes/no, etc. Predetermined evaluation scales satisfies the need for consistency and homogeneity in metrics evaluation.

By using this structure, a project can be associated with different models allowing the execution of different scenarios in order to evaluate different project conditions and the impact to the expected project complexity.

Furthermore, "advanced user" may introduce new complexity factors, metrics and evaluation scales that will allow the PM community to fully parameterize the tool according their project type and the specific project requirements.

According to this model, the Project Complexity (PC) is calculated by the use of the following formula:

$$PC = \sum_{j=1}^{n} CFVj * CFWj$$

where

CFWj: is the Complexity Factor Weight, and

CFVj: is the Complexity Factor Value.

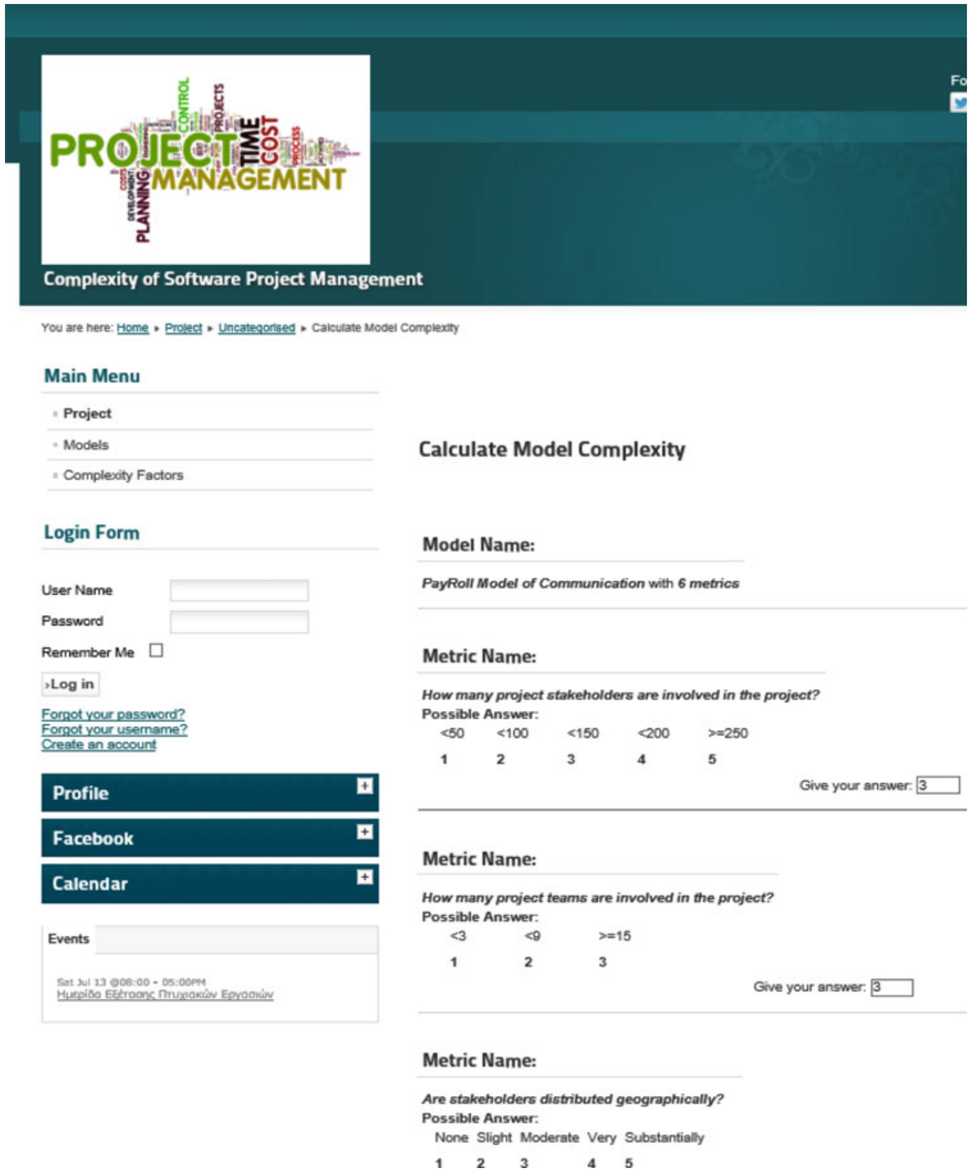
In case of composite Complexity Factors the *CFV* is calculated similarly by the following formula:

$$CFVj = \sum\nolimits_{i=1}^{n} MtrVi * MtrWi$$

In every case the weights of metrics that correspond to a complexity factor are summarized to 1. The same applies and for the weights of complexity factors that belong to a model.

The logical structure of the tool, as described above, is presented in Fig. 1, while a sample screen of PMCAT tool (http://pmc.teilar.gr/), where project complexity of a sample project is calculated is given in Fig. 2.

Fig. 2 PMCAT tool



Conclusion

In this paper we have presented our proposed software project management complexity framework that can be used for systematic assessment of software project's complexity. This framework is based on the assumption that the project complexity is a composite phenomenon and in order to evaluate it, we need to examine all software project management processes. For this reason, the study of PMBOK subject areas, for identifying complexity sources, has been recommended. The advantage of this recommendation is that PMBOK is well-known and widely accepted framework by project managers and organizations and project managers are familiar with its processes and can easily identify and assess complexity factors sourcing from these processes. Thus we believe that we can result with a complexity framework that would be efficient, effective and reliable. Further we propose and describe the architecture of PMCAT tool that utilizes a software project management complexity framework in order to automate the complexity assessment process. PMCAT tool may be used by the PM community in a collaborative fashion or by individual users attempting to evaluate their own projects.

In the future we are going to develop further this complexity framework, by systematically identifying complexity sources, and by systematically evaluating different complexity models. Further, we will extend the functionality of PMCAT tool by providing support for collaborative model development and collaborative project assessments with the use of social networks.

Acknowledgments The research presented in this paper has been co-financed by the European Union (European Social Fund) and Greek National funds through the Operational Program "Education and Life-long Learning" of the National Strategic Reference Framework. Research Funding Program: ARCHIMEDES III. Investing in knowledge society through the European Social Fund. Further, we would like to thank Costas Athanasiou that contributed significantly with the implementation of PMCAT tool.

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Performances of LEON3 IP Core in WiGig Environment on Receiving Side

Natasha Tagasovska, Paulina Grnarova, Aristotel Tentov, and Danijela Efnusheva

Abstract

WiGig is incorporating in the group of wireless standards which tend to release users from being tethered to their devices. It employs 60 GHz frequencies to achieve transfer rate up to 7 Gbps, ten times faster than current fastest Wi-Fi network based on IEEE 802.11n. This new standard, labeled as 802.11ad is an exclusive technology for instant data transfer.

Higher frequencies allow higher data rates, which leads to inter symbol interference problem. This obstacle in WiGig is exceeded by employing OFDM modulation technique elaborated in this paper, with attention focused on receiver—what are the limitations, appropriate transfer synchronization and utilization of FFT required for signal demodulation.

LEON3 processing core is a synthesized VHDL model of 32-bit processor compliant with SPARC V8 architecture. The purpose of this paper is to determine the efficiency of this type of IP core in WiGig environment, by testing LEON3s' performances on algorithms for FFT implementation, since modulation is the essential factor in OFDM deployment. An effort was made to discover an optimal algorithm (with features obedient to the 802.11ad requirements) which escalates LEON3 IP Core performances for receiving side.

Keywords

802.11ad • FFT • LEON3 • OFDM • WiGig

Introduction

We are all aware of the impact of wirelesses in terms of contentment, as well as straightforward and efficient communication. A consequence of this impact is the inevitable trend and effort being made in improving communication technologies, and increased interest in the research area of informational and communication sciences. Latest advancement, defining a significant discovery in the wireless networks, is a result of the publishing the WiGig specification, presenting a hardware capable of transmitting up to 7 Gbps on 60 GHz frequencies. Working in 60 GHz frequency band is very favorable because of the large amount of unlicensed spectrum available in most of the countries. Furthermore, having the convenience of beam forming, results in decreasing inter symbol interference, signal fading and loss of data. Altogether, it seems as a quite promising idea which is expected to be cost effective for large number of clients.

This attractive techniques' realization complexity, originates from of its key feature—high data rate.

A fundamental principle in telecommunications is modulating carrier signal with information to be send. This process has been researched and upgraded through the years ever since modulation techniques have arisen. Today we can choose between different implementations capable of providing us with maximum channel utilization and sending data with high transmission rates. One of the modulation techniques compatible with evaluated 802.11ad standard and

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_31, © Springer International Publishing Switzerland 2015

used for the purpose of this project is Orthogonal Frequency Division multiplication—OFDM.

Since the goal of this paper is testing the performances of LEON3 IP Core on receivers' side, detailed attention was set on OFDM implementation in receivers, signal (de)modulation, channel estimation and synchronization.

Basic function of a receiver is to convert analog to digital signal, transforming received data from time domain to frequency domain by means of Fast Fourier Transformation—FFT, all this preceding the final step of demodulation for retrieving raw data.

FFT is referred many times as "the most important algorithm in our lives" because of its wide utilization in digital signal processing, image processing, business estimations, spectral analyses etc. Hence, a motivation for optimization and successful realization of Fourier transform algorithms. FFTW being the most popular library of optimal algorithms for computing FFT, drew us to applying it in the process of performance testing.

LEON project was started by the European Space Agency (ESA) for studying and development of high performance processors in European space projects. As testing object for this paper LEON3 IP Core was chosen—synthesizable VHDL 32-bit processor compliant with SPARC V8 architecture. Its capabilities were determined in terms of speed executions of the optimal FFT algorithms available.

Current State

LEON3 IP Core has been an object to great number of researches and experiments. Their goals can be divided in three major groups:

1. *Improving system or function specific performances of the core*
2. *Comparing performances with other IP cores*
3. *Finding most suitable hardware implementations*

According to [1] and [2], cores' performances can be enhanced in several ways. A methodology for implementing was described, based on instruction-level characterization. [1] Introduces a way of saving processing power trough joining instructions in pairs—creating composite instructions. This study also takes into account the effect that data switching activity and registers correlation have on energy. Instruction Set Extensions for Multi-Threading in LEON3 is considered in [3].

Furthermore, LEON3 performances have been compared to other microprocessors by executing computations with high processing demands, included in many applications, such as Inverse Discrete Cosine Transform, Matrix multiplication and IIR filters and FFT. According to [3], TMS 320 C 64XX is more efficient then LEON3.

Interesting results are obtained in [2] and [3], where LEON3 performances' were tested in terms of executing Viterbi's function, FFT and Fibonacci, on multi-cored platform.

All these results and methods were taken in consideration while working on this project, although this is the first paper which elaborates on LEON3 IP core's performances in WiGig environment on receivers' side.

WiGig and 802.11ad Standard

What Is WiGig?

The extensive availability and use of digital multimedia content has created a requirement for faster wireless connectivity that current commercial standards can't support. This induced a demand for a single standard that can uphold progressive applications such as wireless display and docking, as well as more entrenched usages such as networks access.

Figure 1 presents the evolution of wireless standards, their data rate and use case, from 1997 until 2012.

Wireless Gigabit (WiGig) Alliance was formed to meet this requirement by constituting a unified specification for wireless communication at multi-gigabit speeds; a specification designed to guide a global ecosystem in interoperable products [1].

WiGig specifies Medium Access Control—MAC and Physical—PHY layers, which allow data rates up to 7 Gbps, ten times faster than today's fastest Wi-Fi networks based on IEEE 802.11n [4].

It operates in the unlicensed 60 GHz frequency band, having much more spectrum available than the 2.4 GHz and 5 GHz bands, [4, 5], which are used by existing Wi-Fi products, thus, enabling having wider channels that support faster communication speeds.

WiGig Alliance based its specification on the existing IEEE 802.11 standard [4], which is at the core of hundreds of millions of Wi-Fi products used worldwide. The specification includes inherent support for Wi-Fi over 60 GHz; new devices with tri-band radios, able to smoothly integrate into current 2.4 GHz and 5 GHz Wi-Fi networks.

An expansive range of advanced uses is supported, including wireless docking and connection to displays, as well as virtually instantaneous wireless backups, synchronization and file transfers between computers and handheld devices. For the first time, consumers are able to enjoy an entire computing and consumer electronics experience without congestion of wires.

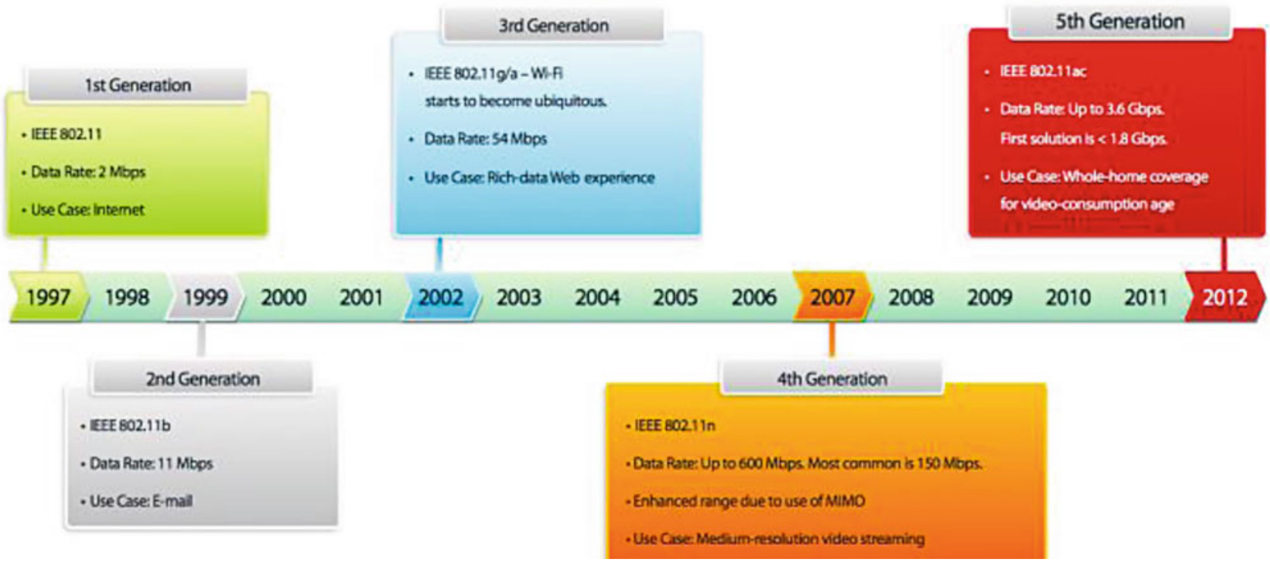

Fig. 1 Evolution of wireless standards

Key Features of WiGig

Key features are included in the WiGig specification [6, 7], tending to maximize efficiency, minimize implementation complexity and cost, at the same time residing compatible with existing Wi-Fi and provide enhanced security:

- Support for data transmission rates up to 7 Gbps; all devices based on the WiGig specification are capable of gigabit data transfer rates;
- Designed from the ground up to support low-power handheld devices such as cell phones, as well as high-performance devices such as computers—advanced power management included;
- Based on IEEE 802.11—provides native Wi-Fi support and enables devices to transparently switch between 802.11 networks operating in any frequency band including 2.4, 5 and 60 GHz;
- Support for beam forming, maximizing signal strength and enabling robust communication at distances beyond 10 m;
- Advanced security using the Galois/Counter Mode of the AES encryption algorithm;
- Support for high-performance wireless implementations of HDMI, DisplayPort, USB and PCIe;
- Low latency, which means that it works with minimal delays, reducing lags in HD video streaming and on-line games;

802.11ad Standard

IEEE 802.11ad is an amendment to the 802.11 WLAN standards which enables up to 7 Gbps data rates in the unlicensed and globally available 60 GHz band [5, 7]. With much more spectrum available than the 2.4 GHz and 5 GHz bands, the 60 GHz band has wider channels, enabling higher data rates over short distances (1–10 m). Primary 802.11ad applications is to remove wires between High-Definition multimedia, computer displays, I/O and peripheral, peer to peer data synchronization and higher speed LAN.

A shared MAC layer with existing 802.11 networks enables session switching between 802.11 networks operating in the 2.4, 5 and 60 GHz bands, resulting in uninterrupted wireless data communications. The 802.11ad MAC layer has been extended to include beamforming support and address the 60 GHz specific aspects of channel access, synchronization, association, and authentication [4].

The design and integration of 802.11ad devices present challenges in tackling ten times higher frequencies and 100 times wider modulation bandwidths than previous WLAN standards [5]. A mix of modulation technologies (both Single Carrier and OFDM) and the use of active directional antennae further complicate matters (Fig. 2).

The ad-hoc network is established and managed through bi-directional protocol exchanges using a low data-rate control channel (MCS 0) while bulk data transfer takes place over an appropriate higher-rate mode (MCS1to MCS31) [2]. Higher rate modes employ beam steering techniques for NLOS operation.

802.11ad uses RF burst transmissions that start with a synchronization preamble (common to all modes) followed by header and payload data. The preamble is always single carrier modulated, the header and data may use single-carrier (SC) or OFDM modulation depending on the target bit rate [6].

Fig. 2 PHY modes (packet overview)

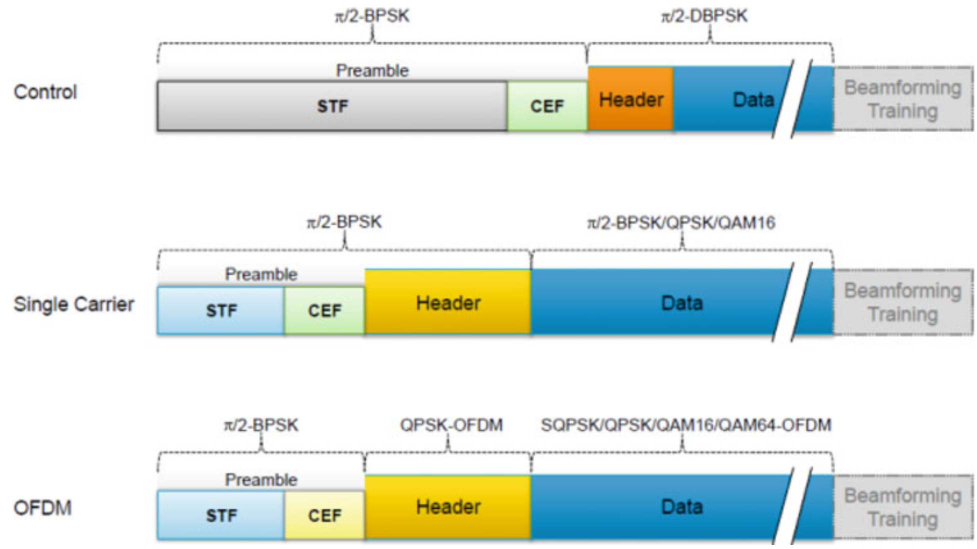

Table 1 OFDM modulation parameters compliant with 802.11ad standard [1]

Parameter	Value
Occupied BW	1.825 GHz
Ref. sampling rate	2.640 Gsamples/s
No. of subcarriers	512
FFT period	~ 194 ns
Subcarrier spacing	5.15625 MHz
Cyclic prefix	128 symbols, ~48.4 ns
Symbol duration	~242 ns
Data subcarriers	336
DC subcarriers	3
Pilots	16
Null subcarriers	157
Modulation	SQPSK, QPSK, 16-QAM,64-QAM
Error Protection	LDPC 1/2, 5/8, 3/4 or 13/16

This project was oriented towards OFDM application in IEEE 802.11ad standard, thus following are presented the fundamental parameters established for this case (Table 1).

Since OFDM has become an essential basis for this project, the next part is dedicated for brief explanation of OFDM implementation in receiver and its key features (Fig. 3).

OFDM Implementation in WiGig Receiver

OFDM can be seen as either a modulation technique or a multiplexing technique. One of the main reasons to use OFDM is to increase the robustness against frequency selective fading or narrowband interference. In a single carrier system, a single fade or interferer can cause the entire link to fail, but in a multicarrier system, only a small percentage of the subcarriers will be affected. Error correction coding can then be used to correct for the few erroneous subcarriers.

In a classical parallel data system, the total signal frequency band is divided into N non-overlapping frequency subchannels. Each subchannel is modulated with a separate symbol and then the N subchannels are frequency-multiplexed. It seems good to avoid spectral overlap of channels to eliminate interchannel interference. However, this leads to inefficient use of the available spectrum. To cope with the inefficiency, the ideas proposed from the mid-1960s were to use parallel data and FDM with overlapping subchannels, in which, each carrying a signaling rate b is spaced b apart in frequency to avoid the use of high-speed equalization and to combat impulsive noise and multipath distortion, as well as to fully use the available bandwidth [8].

To realize the overlapping multicarrier technique, the crosstalk between subcarriers should be reduced, which means that orthogonality between the different modulated carriers is required. The word orthogonal indicates that there is a precise mathematical relationship between the frequencies of the carriers in the system. In other words, when one subcarrier experiences a peak, all of the other subcarriers have zero values. In a normal frequency-division multiplexed system, many carriers are spaced apart in such a way that the signals can be received using conventional filters and demodulators. In such receivers, guard bands are introduced between the different carriers and in the frequency domain, which results in a lowering of spectrum efficiency.

It is possible, however, to arrange the carriers in an OFDM signal so that the sidebands of the individual carriers overlap and the signals are still received without adjacent carrier interference. To do this, the carriers must be mathematically orthogonal [8].

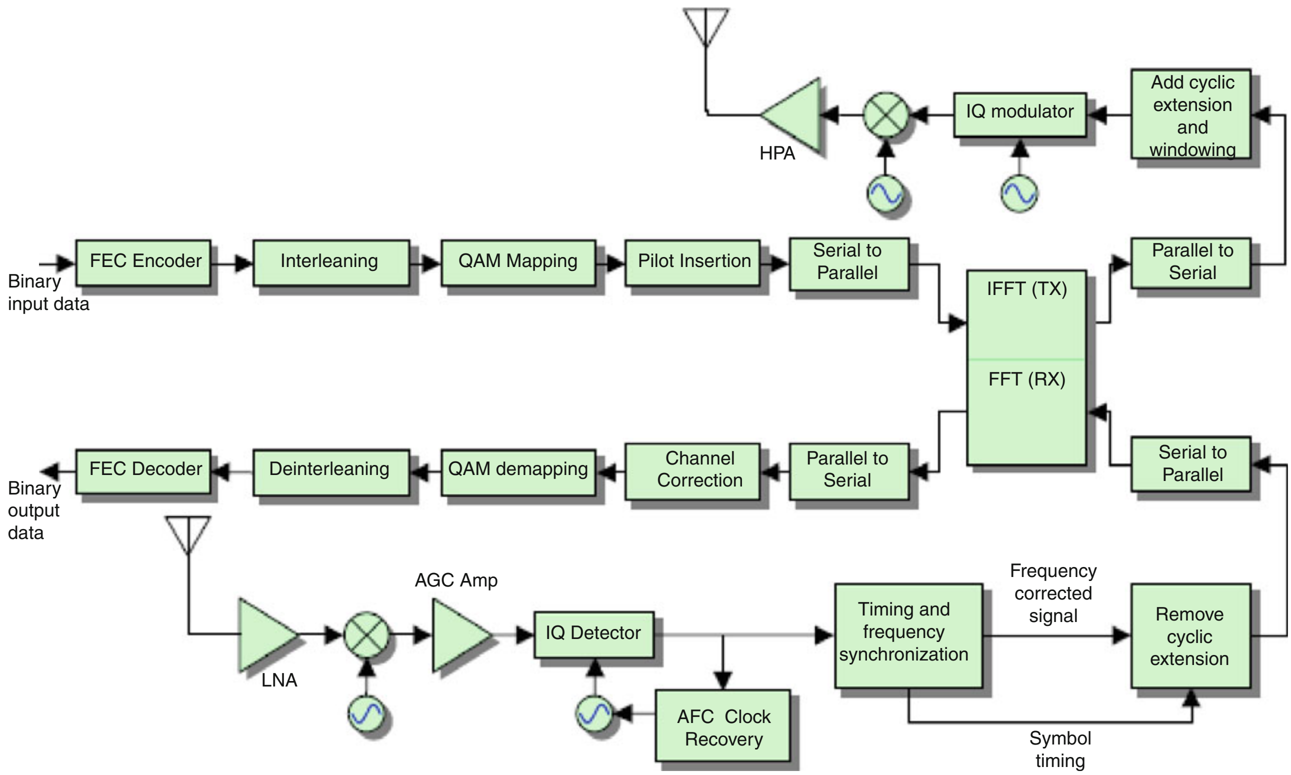

Fig. 3 Block diagram of an OFDM transceiver

The receiver acts as a bank of demodulators, translating each carrier down to DC, with the resulting signal integrated over a symbol period to recover the raw data. If the other carriers all beat down the frequencies that, in the time domain, have a whole number of cycles in the symbol period T, then the integration process results in zero contribution from all these other carriers. Thus, the carriers are linearly independent (i.e., orthogonal) if the carrier spacing is a multiple of 1/T.

It is shown that at the center frequency of each subcarrier, there is no crosstalk from other channels [8]. Therefore, if Discrete Fourier Transformation—DFT at the receiver is used and the correlation values with the center of frequency of each subcarrier are calculated, then the transmitted data will be recovered with no crosstalk. In addition, using the DFT-based multicarrier technique, frequency-division multiplex is achieved not by bandpass filtering but by baseband processing.

Moreover, to eliminate the banks of subcarrier oscillators and coherent demodulators required by frequency-division multiplex, completely digital implementations could be built around special-purpose hardware performing the fast Fourier transform (FFT), which is an efficient implementation of the DFT. Recent advances in very-large-scale integration (VLSI) technology make high-speed, large-size FFT chips commercially affordable. According to this, the receiver is implemented using efficient FFT techniques that reduce the number of operations from N^2 in DFT down to N log N [8].

The most complex and demanding block in OFDM receiver is performing an FFT operation. According to analysis about 70 % of resources dedicated for transceiver deployment, go to receiver implementation because of the complexity consisted in FFT and Viterbi decoder [4]. For this reasons, the evaluation of efficiency of LEON3 IP core was made by executing optimized FFT algorithms and measuring parameters fundamental for accordance with WiGig (802.11ad) standard.

Building Environment and Architecture

There is no such thing as “optimal algorithm” for any computer system/processor. Speed of execution always depends on the hardware. This includes not only processors frequency, but also processors architecture, number of cores, instruction set etc.

LEON3 IP Core

The LEON3 processor core is a synthesizable VHDL model of a 32-bit processor compliant with the SPARC V8 architecture [6]. The core is highly configurable and particularly suitable for system-on-a chip (SOC) designs. The configurability allows designers to optimize the processor for performance, power consumption, I/O throughput, silicon area and cost. The core is interfaced using the AMBA 2.0 AHB bus and supports the IP core plug&play method provided in the Gaisler Research IP library (GRLIB) [9]. The processor can be efficiently implemented on FPGA and ASIC technologies and uses standard synchronous memory cells for caches and register file.

LEON3 Features

- SPARC V8 integer unit with seven-stage pipeline
- Hardware multiply, divide and MAC units
- Separate instruction and data caches
- Support for 2–32 register windows
- Radix-2 divider (non-restoring)
- Single-vector trapping for reduced code size
- Advanced debug support unit
- Optional IEEE-STD-754 compliant FPU
- 20 DMIPS at 25 MHz system clock
- Fault-tolerant version available
- Support for Fusion, IGLOO, ProASIC3E/L, RT ProASIC3, Axcelerator and RTAX [10].

To promote the SPARC architecture and simplify early evaluation and prototyping, the processor and the associated IP library are provided in full source code under the GNU GPL license for evaluation, research and educational purposes. Thus, LEON3 processor IP core source code was simply downloaded from gaisler.org using the GNU GPL license. For completing the resources needed for testing LEON3 performances in WiGig environment, a set of algorithms was also required.

FFTW

Fastest Fourier Transformations on the West—FFTW is a C subroutine library for computing the discrete Fourier transform (DFT) in one or more dimensions, of arbitrary input size, and of both real and complex data (as well as of even/odd data, i.e. the discrete cosine/sine transforms or DCT/DST) [11].

FFTW does not use a fixed algorithm for computing the transform, but instead it adapts the DFT algorithm to characteristics of the underlying hardware in order to maximize performance.

Hence, the computation of the transform is split into two phases. First, FFTW's planner "learns" the fastest way to compute the transform on the machine where transforms are to be computed. The planner produces a data structure called a *plan* that contains this information. Subsequently, the plan is executed to transform the array of input data as dictated by the plan. The plan can be reused as many times as needed. In typical high-performance applications, many transforms of the same size are computed and, consequently, a relatively expensive initialization of this sort is acceptable.

Benchmark Methodologies

In order to obtain and afterwards compare performances of executing FFT variations, a benchmark was deployed on previously mentioned LEON3 architecture.

Raw Data Files

In the raw benchmark data output, the speed for all routines, for both forward (FFT) and backward (IFFT) transforms, is collected in the space-delimited format:

Name-of-code transform-type transform-size mflops time setup-time

Where the times are presented in seconds.

Transform-type is a four-character string consisting of precision (double/single = d/s), type (complex/real = c/r), in-place/out-of-place (=i/o) meaning that entry array is either overwritten while computing FFT-in place, or another, output array exists, and forward/backward (=f/b). For example, transform-type = dcif denotes a double-precision in-place forward transform of complex data [11]. For the purpose of this project we were interested not in the optimal FFTW algorithm from FFTW library, but in the optimal algorithm which fulfills the requirements based on the WiGig standard:

- Transformation on 512 bit array
- FFT time $< 0.194\ \mu s$

Because of the fact that on OFDM receivers' side, data arrives in real format, in the process of evaluation only algorithms with this type of entries were considered.

Timing

Our FFT timing measurement is intended to reflect the common case where many FFTs of the same size, indeed of the same array are required. Thus, measurement is broken into two parts:

Initialization

First any initialization/setup routines provided by the core are called. This setup may include calling the FFTs' code once, if it performs initialization on the first call. Setup is measured separately from the FFT performance below but only as a rough indicator, no attempt is made to perform repeated measurements or to make initialization preparations as efficient as possible.

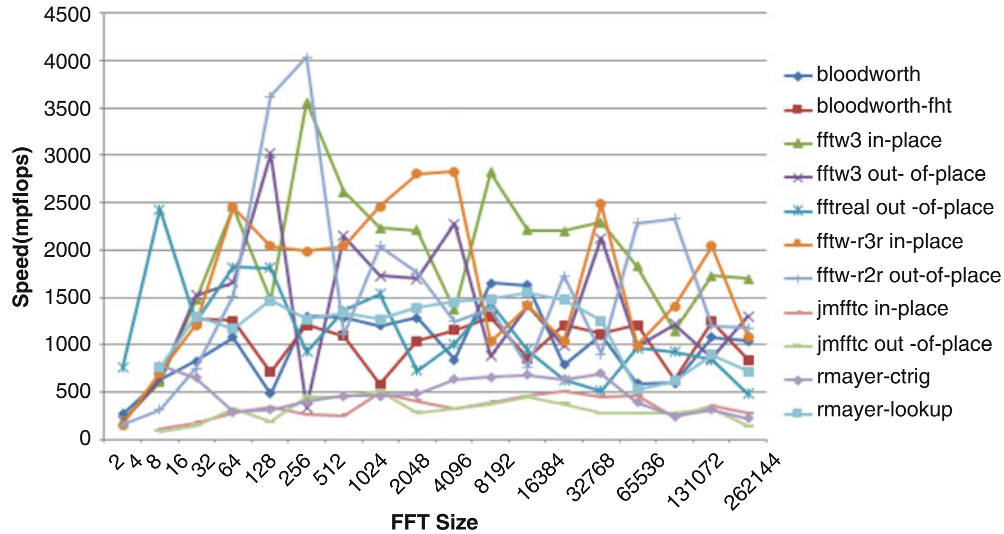

Fig. 4 Double precision real data, one dimensional transforms

Performance Measures

To report FFT performance, *mflops* are plotted for each FFT which is a scaled version of speed according to [11], defined by:

$$\mathbf{mflops} = 5\mathrm{Nlog}_2(\mathrm{N})/\text{time for one FFT in microseconds,}$$
$$\text{for complex transforms} \tag{1}$$

$$\mathbf{mflops} = 2\mathrm{Nlog}_2(\mathrm{N})/\text{time for one FFT in microseconds,}$$
$$\text{for real transforms} \tag{2}$$

Since on receivers' side only real input values are considered for FFT implementation, mflops in this project were calculated by Eq. (2).

Reporting Results and Discusseion

Benchmark Results

Although FFTW represents a vast set of different implementations of FFT, all of them are variations of the most famous algorithms: Cooly-Tukey, Prime Factor and Winogard. From the benchmark results we set aside only the ones with features compliant with characteristics of FFT on receivers' side i.e. algorithms with real input values. As a result detailed attention was set on: FFTW3, Bloodworth, fftw real and fftw3-r2r. Further, only algorithms which fulfill Eq. (3), are acknowledged as appropriate for LEON3 implementation as a WiGig receiver.

$$T_{DFT}:\ IDFT/DFT\ period <= 0.194\,\mu s \tag{3}$$

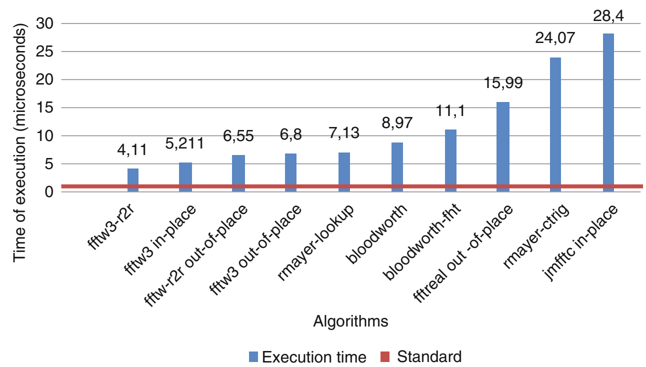

Fig. 5 Execution time compared with 802.11ad requirements

Figure 4 represents most of the algorithms employed in the performance testing process, each one of them plotted with its speed of execution in mflops for appropriate FFT sizes. Since the execution of the benchmark results in terms of execution time for each algorithm with specific FFT size, from Fig. 4 it is noticed that for different number of points, different algorithm appears as optimal. Only valuable data was extracted from accumulated results, meaning computing parameters for FFT size of 512 points.

A logical conclusion is that the algorithm with best execution performances for 512 FFT size, is the one with shortest time of execution, thus it shall be the optimal algorithm, closest to the time set by the standard. This algorithm in our study was proven to be *fftw3 r2r in-place*.

In interest of getting a better perspective on the performances, another chart was plotted (Fig. 5) where algorithms are sorted by their time of execution on LEON3 IP core for FFT with 512 number of points. The red line drawn in the chart represents the boundary set by the 802.11ad standard i.e. 0.194 μs $T_{DFT.}$ Observing Fig. 5, it

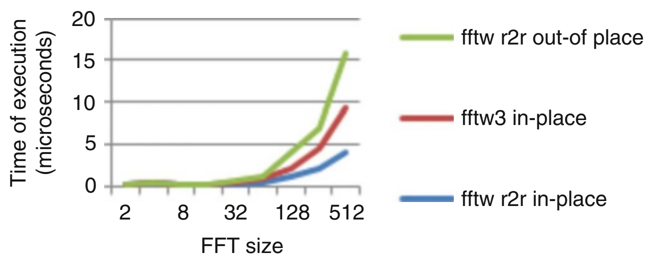

Fig. 6 Best three algorithms FFT size/execution time dependence

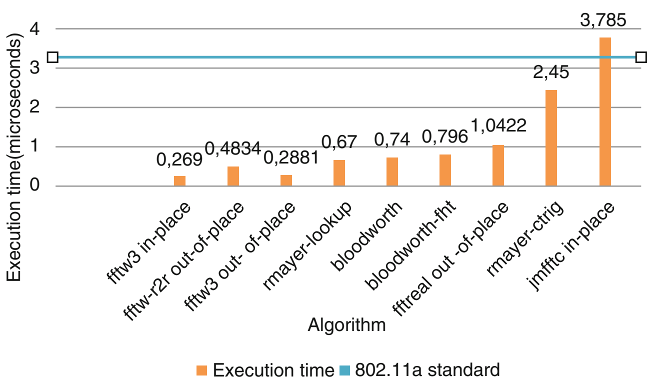

Fig. 7 LEON3 performances in comparison with 802.11a standard

is obvious that neither of the algorithms from FFTW library executed on this type of architecture fulfilled the 802.11ad requirements. The optimal algorithm ftw3 r2r in place, needed 4.11 μs to finish calculations which is far from desired 0.194 μs.

For better overview of the outcome, Fig. 6 is proposed, where the best three algorithms are presented with time of execution for different FFT sizes. An obvious conclusion is that for larger FFT sizes, longer time is required for execution.

Discussion

From the performances tests conducted, only the optimal algorithms for execution on SPARC V8 architecture were considered. The final conclusion is that even the optimal algorithm does not achieve required time of execution—0.194 μs for computing Discrete Fourier Transformation of 512 size. Examining the final results it is perceived that most of the algorithms executed on LEON3 meet the 802.11a requirements, where FFT size is set to 64 and T_{DFT} should be 3.2 μs or less (represented by blue line in Fig. 7). Observing Fig. 7, fftw3 in-place appears to be optimal with time of execution 0.269 μs. Summarizing impressions from gathered results, the final outcome is that single cored LEON3 processor is not convenient for utilization in WiGig environment. This research can be continued towards experimenting with different number of IP cores and parallelized FFT implementation algorithms. Another option is to combine two or more FFT algorithms in order to achieve better performances, due to the fact noticed that some of the algorithms are faster in performing the pre-process sampling, while others are more suitable for computing DFT.

Conclusion

WiGig standard established in 2010, today in year 2013 already has its realization and first products are appearing on the market. This significant progress in area of wireless networks was an inspiration for this research, where performances of LEON3 IP Core were measured and evaluated as if it was used as WiGig receiver. 802.11ad specification has clear boundaries and requirements for devices which are to operate on 60 GHz: signal modulation with OFDM (or SC), accomplished by computing FFT with 512 points, while the period of executing DFT cannot exceed 0.194 μs. After conducting performance testing of LEON3 by employing fastest known FFT algorithms, it is concluded that in neither case, with neither of those algorithms, standards; requirements were met. By conducting this research, LEON3 core performances were acknowledged, when it comes to computing with high performance demands, as in this case, Fast Fourier Transformation. Additionally, we grasp knowledge about what exactly and how much can be expected of LEON3 IP Core implementation in WiGig environment.

Acknowledgment This work was partially supported by the ERC Starting Independent Researcher Grant VISION (Contract n. 240555).

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Clustering-Based Topic Identification of Transcribed Arabic Broadcast News

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Abstract

In this research different clustering techniques are applied for grouping transcribed textual documents obtained out of audio streams. Since audio transcripts are normally highly erroneous, it is essential to reduce the negative impact of errors gained at the speech recognition stage. In attempt to overcome some of these errors, different stemming techniques are applied on the transcribed text. The goal of this research is to achieve automatic topic clustering of transcribed speech documents, and investigate the impact of applying stemming techniques in combination with a Chi-square similarity measure on the accuracy of the selected clustering algorithms. The evaluation—using F-Measure—showed that using root-based stemming in combination of spectral clustering technique achieved the highest accuracy.

Keywords

ASR • Speech Transcripts • Speech Transcription Errors • Topic Clustering • Topic Identification

Introduction

The growing amount of audible news broadcasted on TV channels, radio stations and on the Internet demands reliable and fast techniques to organize and store those vast amounts of news in order to facilitate future search and retrieve.

Although the field of multimedia information retrieval has advanced rapidly in the past decade, it still confronts many challenges. The main problem challenging researchers on this field is the unstructured nature of audio and video. For audio, analyzing audio documents has mainly focused on two main directions. First, developing audio data classification schemes to segment an audio document into coherent chunks of different types of audio classes—music, speech, speech and music etc. [1–4]. Second, generating transcripts of audio documents through Automatic Speech Recognition (ASR)—a technology that allows a computer system to identify the words that a person speaks into a microphone or telephone and convert them to written text—and then performing analysis for automatic indexing, retrieval, and other automatic tasks [5–9]. Those efforts have shown the possibility of applying ASR to audio documents to perform indexing, retrieval and other automatic tasks with a good degree of success. However, many factors affect the degree of success of many of those approaches including the size and quality of transcription.

This work adapts the approach of applying ASR technology to audible Arabic news documents, and then applying preprocessing and clustering techniques on the transcribed textual documents produced by the ASR as in Fig. 1. To overcome the problem of transcription errors produced by the ASR, two stemming techniques are applied: root-based

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_32, © Springer International Publishing Switzerland 2015

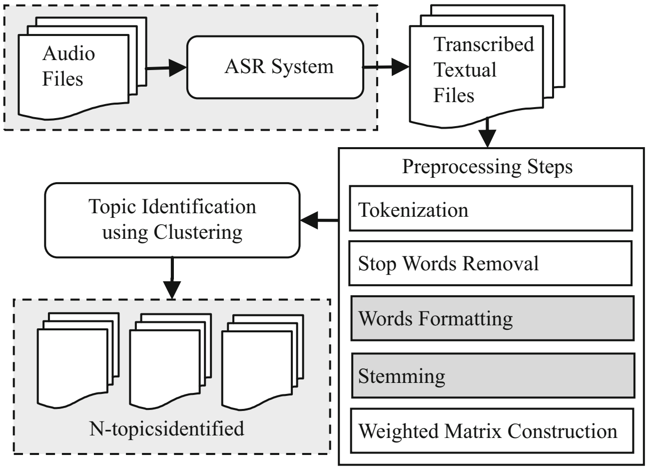

Fig. 1 The procedure of clustering-based topic identification of transcribed textual files obtained out of audio files

and light stemming. To achieve topic identification, two clustering techniques are utilized: k-means [10] and spectral clustering [11]. K-means is selected as a simple and fast traditional clustering algorithm. Spectral clustering is selected as it is one of popular and simple to implement modern clustering techniques. Both clustering algorithms group transcribed documents into topics by applying a similarity measure based on the Chi-square method [9]. This similarity measure is designed to eliminate non informative words (usually erroneous words when applied on transcribed documents).

The topic clustering accuracy is evaluated for the two selected clustering algorithms in three situations: when the transcribed documents are clustered without applying any stemming techniques, when light stemming is applied, and when root-based stemming is used.

This research is organized as follows: in section “Speech transcription challenges”, speech transcription challenges are discussed. In section “Data set preprocessing”, data set preprocessing steps are discussed. In section “Topic identification”, topic identification, using K-Means and spectral clustering techniques, is discussed in details. In section “Experimental results”, experimental results are evaluated. The last section concludes the research.

Speech Transcription Challenges

The process of transcribing audible media to textual form using ASR system confronts many challenges [9] that are typically not present in normal textual documents. The main challenges can be summarized as following:

- Transcription Errors: typically, transcription errors are defined as occurrence of irrelevant or erroneous words in transcripts. The main causes of such errors results from drawbacks of the ASR system.
- Grammatical errors: a common problem in conversational speech.
- Out-of-Vocabulary Problem (OOV): presence of unknown words that appear in the speech but not in the recognition vocabulary of the ASR. The daily growth of natural languages is the main cause of such problem.
- Combination of the previously mentioned problems.

The occurrence of such problems can seriously restrict the transcription process efficiency and hence restricts any further analysis applied on the transcripts.

When dealing with news, problems like usage of grammatically incorrect sentences are restricted by the fact that news is usually written by language skilled news editors and read by selected professional news broadcasters. In addition, topic identification using clustering operates on the word level, not on the sentence level, so such problem doesn’t affect the work in this research.

In the case of Arabic news broadcasts, most news networks use standard Arabic, which limits the problem of OOV to some degree. Solving such problem is an active field of research and it is outside the scope of this research.

Like OOV problem, the transcription errors occur due to limitation in the ASR system. The correction or elimination of such errors is a challenge and requires understanding the nature of these errors. According to authors’ observation, the transcription errors regarding Arabic language are categorized into four sets:

- **Omission errors**: happen when the ASR fails completely to recognize a word or a series of consecutive words. In this case, the words are dropped out from the transcribed text and recognition process is continued. Omission errors are irrecoverable.
- **Word insertion errors**: occur when the ASR confuses word syllables with a separate word or multiple words. In this kind of errors, the original word is irrecoverable.
- **Misidentification errors**: identifying a pronounced word as a different word similar in pronunciation. The transcribed word may or may not belong to the valid Arabic vocabulary set.
- **Minor spelling errors**: a spoken word is identified correctly, but spelled wrong in transcription. These errors usually affect the way a word should be pronounced and it may affect its meaning as well. Common minor spelling errors generated by ASR are replacing the letter ‘ه’ with ‘ة’ at the end of the word and vice versa, replacing one of the following letters with one another ‘ا’, ‘أ’, ‘إ’, and ‘آ’, and diacritics related errors. In the next sections,

transcription errors problem is further descried and proposed solutions to restrict its negative effects are presented.

Data Set Preprocessing

As any written text documents, the transcribed documents produced by the ASR system are of unstructured format. Before working on these documents, it is required to transform these unstructured documents to structured format using preprocessing in order to facilitate any further analysis applied on them. The following are the steps involved in the pre-processing applied in this work.

- **Tokenization**—the process of mapping sentences from character strings into strings of words. For example, the sentence "اللغة العربية تعد من أشهر اللغات" would be tokenized into "اللغة/", "العربية/", "تعد/", "من/", "أشهر/", "اللغات/".
- **Stop words removal**—Stop words are typical frequently occurring words that have little or no discriminating power, or other domain-independent words. Stop words removal can increase the effectiveness of the information retrieval process [12, 13], especially when dealing with large volume of text [14]. Stop words identified in this work are prepositions, pronouns, and conjunctions.
- **Words Formatting**—An extra step applied in this work to unify all different shapes of the same letter to one form and also to remove some unwanted suffixes as in Table 1.
- **Stemming**—Removes the affixes in the words and produces the root word known as the stem. Typically, the stemming process will be performed so that the words are transformed into their root form. Automatic Arabic stemming is effective technique for text processing for small collections as in [15, 16] and large collections of documents as in [17, 18]. It also can enhance clustering as in [16]. Arabic stemmers are categorized as either root-based as in [19, 20] or stem-based (light stemmers) as in [17, 18].
- **Weighted matrix construction**: the process of representing the text document into a machine readable form [21, 22].

Besides transforming the unstructured transcribed text to structured form, the preprocessing is also used as the first phase to reduce the transcription errors by either correcting or help overcoming some of these errors. This happens during the preprocessing steps: words formatting and stemming. The following discuss how applying preprocessing can correct or help overcoming some of the transcription errors.

In Tables 2 and 3, two samples of transcribed text with various transcription errors are presented with the erroneous words being underlined. Words like "بدات", "امس, "اذا", "الفتره", "التجريبيه" and "ملاحقه" are examples of minor spelling errors. According to Arabic syntactic rules [23], the correct spelling for these words should be: "بدأت", "أمس", "إذا", "الفترة", "التجريبية" and "ملاحقة". Such a problem is common in ASR systems and leaving it without handling would cause problems in any further analysis as for any computer system words like "بدات" and "بدأت" aren't the same and actually will process them as two separate words. In this kind of errors, the correct spelling is determined on pronunciation basis, and the correct pronunciations depends on the meaning of the word which is determined according

Table 1 Steps of words formatting process

• Remove diacritics
• Remove punctuations
• Remove non-letters and non Arabic letters
• Replace 'أ', 'إ', 'آ', with 'ا'
• Replace 'ة' with 'ه'
• Replace 'ى' with 'ي'
• Remove the suffixes 'و', 'ا', and "وا" only if the result word is more than three letters

Table 2 Sample (1) of transcribed text with various transcription errors

بدات شركة مايكروسوفت امس السبت بتجربه ميزات ترك رسالة فيديو عن طريق برنامج سكايب اذا لم يكن المستخدم اتصل وتمكن الخدمة المستخدم من تسجيل مقطعه فيديو مدته لا تتجاوز 3 دقائق ومن ثم إرسالها إلى إحدى جهات الاتصال غير المتوافرة حاليا لمستخدمي ماك اندرويد و أي أو أس تجربه المميزه من خلال 20 رساله مجانيه توفرها الشركة دون الكاش عن الرسوم الإضافيه بعد الانتهاء من الفتره التجريبيه

Table 3 Sample (2) of transcribed text with various transcription errors

تتويجا يوفنتوس رسميا بلقب الدوري الإيطالي لكرة القدم للمرة الثانية على التوالي وذلك بعدما
اللعب على ضيفه باليرمو

وأقدم داخلة فريق المدرب انطونيو كونتي المباراة وهو بحاجة الي نقطة 1 فقط لحس اللقب للمرة
ثانية على التوالي والتاسعة والعشرون في تاريخه لكوني يتصدر الترتيب بفارق ١١ نقطة عن
ملاحقة نابولي قبلة أربعة مراحل على انتهي الموسم

Table 4 Sample (1) of transcribed text after applying preprocessing with light stemming

بد شرك مايكروسوفت امس سبت بتجرب ميز ترك رسال فيديو طريق برنامج سكايب مستخدم
اتصل
تمكن خدم مستخدم تسجيل مقطع فيديو مدت تتجاوز دقائق ارسال احدى جه اتصال متوافر حاليا
لمستخدم ماك اندرويد اي اس تجرب مميز خلال رسال مجان توفر شرك الكاش رسوم اضاف
انتهاء فتر تجريب

Table 5 Sample (2) of transcribed text after applying preprocessing with light stemming

تتويج يوفنتوس رسم بلقب دور ايطال لكر قدم مر ثان توال لعب ضيف باليرم
اقدم داخل فريق مدرب انطوني كونت مبارا بحاج نقط فقط لحس لقب مر ثاني توال والتاسع عشر
في تاريخ لكون يتصدر ترتيب بفارق نقط ملاحق نابول قبل أربع مراحل انته موسم

Table 6 Sample (1) of transcribed text after applying preprocessing with root-based stemming

بدات شرك مايكروسوفت امس سبت جرب ميز ترك رسل فيديو طرق برنامج سكايب خدم صلي
مكن خدم خدم سجل قطع فيديو مدي جوز دقق رسل احدى جهي صلي وفر حلي خدم ماك
واندرويد اي اس جرب ميز خلل رسل جني وفر شرك الكاش رسم ضيف نهي فتر جرب

to its context, thus the only way to detect and correct such errors is searching massive dictionary of correctly spelled Arabic words and retrieving the correct syntax if the word is misspelled. Such process is inefficient in terms of the processing time and power required. The most suitable solution is not to correct these errors but to work around them by unifying all different formats of a letter into one form. The unification process is performed at the words formatting step. Some suffixes are also removed at the word formatting step to fine-tune the input text for the stemming.

Two stemming techniques are utilized in this research: light stemming represented in Larkey's light10 stemmer [18] and root-based stemming represented in Khoja's root-based stemmer [19]. The stemming techniques are applied in this work to unify vocabulary and also to overcome some transcription errors.

The words "اتصل", "مقطعه", "الكاش", "تتويجا", "اللعب", "أقدم", "داخلة", "لحس", "لكوني", "قبلة", and "انتهي" in both Tables 2 and 3, are examples of misidentification errors. The original Words are "متصلا", "مقطع", "الكشف", "توج", "تغلب", "قد", "دخل", "لحسم", "لكونه", "قبل", and "انتهاء". If a word is misidentified into one of its relative forms like "اتصل" and "متصلا", they are of the same root "وصل", so there is a good chance that this mistake would be overcome when stemming is applied. The light stemmer would leave the word unfixed as in Table 4, but the root-based stemmer would return it to its root as it is actually a valid Arabic word (Table 6). Unfortunately, according to a limitation in the root-based stemmer the word is transformed to the root "صلي" instead of the correct one "وصل". The promising thing is that such a limitation wouldn't be effective because the stemmer would transform all the forms of the root "وصل" like "اتصال", "وصل" and "اتصل" to a unified root "صلي" and even it is grammatically wrong it is not important. The important thing is that the all forms of the same word are recognized and unified, and thus the error is neutralized. This worked perfectly for the erroneous words "اتصل", "مقطعه", "داخلة", "لكوني", "قبلة", and "انتهي" when root-based stemming is applied.

On the other hand, the light stemmer actually tends to correct the error in case of occurrence of inserted letters in the start and the end of the erroneous word as long as the inserted letters exist on its prefix/suffix removal list. The erroneous word "مقطعه" on Table 2, is transformed to its correct form "مقطع" (Table 4) by eliminating the inserted letter 'ه'. The root based stemmer removed the inserted letter 'ه' and transformed the word to its root "قطع" (Table 6). The root-based stemmer can also correct an erroneous word if by chance the original word is the same as the root of erroneous

Table 7 Sample (2) of transcribed text after applying preprocessing with root-based stemming

توج يوفنتوس رسم لقب دور الايطالي كور قدم مرر ثاني ولي لعب ضيف باليرم قدم دخل فرق درب انطوني كونتي برا حوج نقط فقط لحس لقب مرر ثاني ولي تاسع عشرون ارخ كون صدر رتب فرق نقط لحق نابولي قبل أربعة رحل نهي وسم

word. The word "تتويجا" (Table 3) is a good example, after removing the suffix 'ا' at the word formatting step the result is "تتويج" which would be transformed by the root-based stemmer to "توج" (Table 7) which is also the same as the original word in spelling. Both techniques fail when the erroneous word is substituted by completely another word of a different spelling and meaning like "الكاش", "اللعب", "أقدم", and "لحس" as in Tables 4–7. If such erroneous words are not repeated frequently along the whole set of documents, they would probably have poor information contribution, and hence they would be eliminated by the chi-square based similarity measure if their information contribution assessment doesn't comply with a certain threshold, otherwise they would be retained. The Chi-square similarity measure will be discussed in the next section. After applying stop words removal, words formatting and stemming, and in order to process the transcribed documents for topic identification, they must be represented in a machine readable form. Widely used document representation methods are Vector-Space Model (VSM), Multiword term, Character n-gram representation. VSM is identified as one of the first formal models that were most widely and effectively used for proximity estimation between text documents [21, 22]. VSM model is the model applied in this work because of its conceptual simplicity.

In VSM all documents are represented as points in an n-dimensional space of terms. Thus, every document is treated as a vector, where elements of the vector are weights assigned to the terms by a weighting scheme that evaluates their significance in a document and the overall collection of documents. At the recent time, there are a number of well-known methods that have been developed to evaluate term weight [24], in this work, Okapi method [25] is applied, which is a modification of a classic TFIDF (Term Frequency × Inverse Document Frequency) weighting scheme and proved to be efficient in a number of applications [5, 7].

Topic Identification

Topic Identification is the process of assigning one or more labels to text documents. These labels are chosen from a predefined list of topics. Topic identification is achieved by means of clustering in this work. Clustering is a basic unsupervised machine learning task of finding groups of similar objects in the data set. The similarity between the objects is measured with the use of a similarity measure. A Chi-square based similarity measure is utilized in this work. This measure implements the following steps:

- **Transcript intersection**: Determine the word co-occurrences between matching transcripts.
- **Information contribution**: Evaluate the amount of information contributed by every matching document to the intersection.
- **Informative closeness**: Assess informative closeness of co-occurring words by applying the Chi-square technique.
- **Similarity measure**: Calculate the similarity of matching documents.

To determine the transcript intersection, first, Combined Weight of the word (CW) is calculated using (1) according to the Okapi method [29].

$$CW\left(w_i|D_j\right) = \frac{(k+1) \times CFW(W_i) \times TF\left(w_i, D_j\right)}{k \times \left((1-b) + b \times NDL\left(D_j\right)\right) + TF\left(W_i, D_j\right)} \tag{1}$$

The quantity $CFW(w_i) = log \frac{N}{n(w_i)}$ is the data set collection frequency weight; N is the total number of documents in the collection and $n(w_i)$ is the number of documents containing the word w_i. The quantity $TF(w_i, D_j)$ is the frequency of occurrences of word w_i in the document D_j and $NDL(D_j)$ is the length of the document D_j normalized by the mean document length. The constant b controls the influence of document length and is empirically determined to the value 0.75. The other constant K acts as a discounting parameter on the word frequency: when K is 0, the combined weight reduces to the collection frequency weight; as K increases the combined weight approaches *tf*itf*. K is set to 1.25 in this work.

The next step is to get the Weight of a Document (DW). DW can be calculated for the whole document or any of its parts via applying (2).

$$DW(D_i) = \sum\nolimits_{w_k \in X_i} CW(w_k) \tag{2}$$

To obtain co-occurring words between documents, all words in transcripts are sorted by their weights and retain only those whose weights are greater than some preset threshold which has been determined empirically in this work. Thus non-informative words including low frequently repeated erroneous words should appear at the bottom of the sorted list and hence eliminated according to the empirically determined threshold.

The co-occurring words between documents generally convey different amount of information with respect to the documents that they belong to; hence the amount of information conveyed by every document to the intersection is evaluated using (3):

$$Inter\left(D_i, D_j\right) = \frac{DW\left(D_i \cap D_j\right)}{DW(D_i)} \tag{3}$$

and it can be concluded from (3) the following inequality, which is true when $D_i \neq D_j$:

$$Inter\left(D_i, D_j\right) \neq Inter\left(D_j, D_i\right) \tag{4}$$

Before calculating the similarity between documents, information closeness between intersected words must be evaluated first. To carry out this step, Chi-square technique is applied. An assumption is made that words w_k of the document D_i with the corresponding weights $CW(w_k|w_k \in D_i)$ forms the set of words with the expected weights distribution, and, the same words $w_k^{'}$ but belonging to the document D_j with the weights $CW(w_k^{'}|w_k^{'} \in D_j)$ forms the set of words with the observed distribution of weights. Finally, assume that null hypothesis, stating that two sets fit each other with some value of significance, is true. Thus, the significance value σ that makes the null hypothesis true can be determined by evaluating (5).

$$X^2 = \sum\nolimits_{w_k \in D_i \cap D_j} \frac{\left(CW\left(w_k \middle| w_k \in D_i\right) - CW\left(w_k \middle| w_k \in D_j\right)\right)^2}{CW\left(w_k \middle| w_k \in D_i\right)} \tag{5}$$

Once the value significance is determined the similarity between two documents is calculated by applying (6). The calculated similarity will range between 0 and 1 and it will be equal to 1 if and only if $D_i = D_j$.

$$sim\left(D_i, D_j\right) = \sigma \times Inter\left(D_i, D_j\right) \tag{6}$$

For Topic identification process, two clustering techniques are utilized: K-Means and a spectral clustering.

K-means [10] is based on the idea that a center point (centroid) can represent a cluster. It is one of the most popular traditional data clustering algorithms because of its simplicity and computational efficiency. The main problem with this clustering method is its tendency to converge at a local minimum and the final results highly depends on the initial choices of centroids. The K-means works as follows:

1. Select K points as the initial centroids.
2. Assign all points to the closest centroid.
3. Recompute the centroid of each cluster.
4. Repeat steps 2 and 3 until the centroids don't change.

Spectral clustering [11] is one of the most popular modern clustering algorithms. It is simple to implement, can be solved efficiently by standard linear algebra methods, and very often outperforms traditional clustering algorithms such as the k-means algorithm. The idea of spectral clustering techniques is reformulating the clustering process using a similarity graph G = (V, E) where the goal is to find a partition of the graph such that the edges between different groups have very low weights i.e., points belong to different clusters, and the edges within a group have high weights i.e., points belong to the same cluster. The similarity graph used in this work is the fully connected graph because the Chi-square similarity measure itself models local neighborhoods, so it best suite this kind of graphs. The steps involved in the spectral clustering algorithm utilized in this work are the steps specified by the normalized spectral clustering according to normalized cuts algorithm (Shi–Malik algorithm) introduced by Jianbo Shi and Jitendra Malik [26]:

- Construct a fully connected similarity graph. Let W be its weighted adjacency matrix.
- Compute the unnormalized Laplacian L.
- Compute the first k generalized eigenvectors $u_1, \ldots, u_k$ of the generalized eigenproblem $Lu = \lambda D_u$.
- Let $U \in R^{n \times k}$ be the matrix containing the vectors $u_1, \ldots, u_k$ as columns.
- For $i = 1, \ldots,$ n, let $y_i \in R^k$ be the vector corresponding to the i-th row of U.
- Cluster the points $(y_i)_{i = 1, \ldots n}$ in R^k with the k-means algorithm into clusters $C_1, \ldots, C_k$.

Experimental Results

The dataset used in this research consists of audio news stories collected manually from various Arabic news networks broadcast: Al-Jazeera, Al-Arabiya, and BBC Arabic. The dataset size is about 30 hours of recorded Arabic news stories. The average length of the news story is 2 min. The news stories are transcribed generating 1,000 text files divided into five topics: culture and arts, economics, politics, science, and sports.

The reason behind the manual selection of the news stories is to minimize speaker related problems, like unclear pronunciation and grammatical errors. The collected news are then transcribed into text documents using ASR system "Dragon Dictation" [27], and then preprocessed for topic-clustering.

After applying preprocessing steps on the documents and performing clustering techniques, the accuracy of clustering is evaluated using F-Measure (7), a measure that combines the recall (9) and precision (10) ideas from information retrieval [21]. The recall is defined as the probability that, if a random document D_x should be categorized under cluster C_j, this decision is taken. The precision is defined as the probability that, if a random document D_x is categorized under cluster C_j, this decision is correct.

$$F = \sum_{i} \frac{n_i}{n} Max\{F(i,j)\} \tag{7}$$

The max is taken over all clusters at all levels, and n is the number of documents and $F(i, j)$ is defined as in (8).

$$F(i,j) = \frac{2 \times Recall(i,j) \times Precision(i,j)}{Precision(i,j) + Recall(i,j)} \tag{8}$$

$$Recall(i,j) = n_{ij}/n_i \tag{9}$$

$$Precision(i,j) = n_{ij}/n_j \tag{10}$$

The quantities *Recall* and *Precision* are calculated as in (9) and (10), where n_{ij} is the number of members of class i in cluster j, n_i is the number of members of class i, and n_j is the number of members of cluster j.

The dataset is divided into subsets of sizes ranged from 50 to 200 documents per category. Experiments are carried out on each of the subsets three times for each clustering algorithm: when no stemming is applied, when light-stemming is applied, and when root-based stemming is applied. The accuracy of the clustering is evaluated for each subset, and then the average accuracy is calculated among all the subsets (Table 8). The dataset is divided into subsets to consider the effect of the change in dataset size and hence the change in the amount of information contained in each subset on the average clustering accuracy. The reason why clustering is first applied on non stemmed data is to measure the impact of both stemming techniques on improving the accuracy of the clustering algorithms operating on such erroneous data.

The results have showed that both stemming techniques have improved the accuracy of both the clustering algorithm. Root-based stemming has showed more efficiency improvement than the light stemming when applied on the data used by the two clustering algorithms. The reason behind that is the nature of the Arabic language and its related transcription errors. Light stemming techniques only removes certain prefixes and suffixes which will not truly and effectively transform all similar words to one root, hence limit its ability to overcome misidentification errors. In contrast root-based stemming transforms all similar words to one root, except for some limitations that may exist in the algorithm performing the transformation, which makes it more efficient.

Table 8 Clustering accuracy evaluation

	Average accuracy		
Clustering approach	Non-stemmed	Light-stemmed	Root-stemmed
K-Means	44.3 %	47.6 %	56.5 %
Spectral clustering	46.5 %	53.8 %	68.9 %

The spectral clustering algorithm achieved more accuracy than the k-means algorithm in all cases which may be explained due to the nature of the data set. In contrast to spectral clustering, k-means tends to perform best in linearly separable data. Due to the broad limited category topics chosen, this increases the chance of existence of cross topic documents that in turn make the process of linearly separating data more difficult. Although the similarity method used is based on informative closeness between the textual documents, it is best utilized by the spectral clustering algorithm. The highest accuracy of 68.9 % achieved is promising considering that unrecovered and eliminated transcription errors may have destroyed the amount of the information that govern to which category the document should belong.

Conclusion

In this research a set of transcribed textual documents obtained from a set of spoken documents are clustered into topics, and the impact of applying two stemming techniques along with the Chi-square similarity measure on the accuracy of the topic-clustering process is measured. The results showed that root-based stemming is more effective in overcoming some of the problems that occur during the transcription process: mostly transcription errors. Light-stemming is less effective than root-based stemming in overcoming transcription problems.

The clustering accuracy evaluation showed that the best results are achieved by applying the spectral clustering algorithm in combination with root-based stemming scoring average accuracy of 68.9 % which is promising considering the noisy nature of the dataset.

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Epilepsy Seizure Detection in EEG Signals Using Wavelet Transforms and Neural Networks

E. Juárez-Guerra, V. Alarcon-Aquino, and P. Gómez-Gil

Abstract

An electroencephalogram (EEG) is a record of the electric signal generated by the cooperative action of brain cells, that is, the time course of extracellular field potentials generated by their synchronous action. EEG is widely used in medicine for diagnostic and analysis of several conditions. In this paper, we present a system based on neural networks and wavelet analysis, able to identify epilepsy seizures using EEG as inputs. This work is part of a research looking for novel models able to obtain classification rates better that the state-of-the-art, for the identification of normal and epileptic patients using EEG. Here we present results using a Discrete Wavelet Transform (DWT) and the Maximal Overlap Discrete Wavelet Transform (MODWT) for feature extraction and Feed-Forward Artificial Neural Networks (FF-ANN) for classification. By using the benchmark database provided by the University of Bonn, our approach obtains an average accuracy of 99.26 % tested using threefold cross-validation, which is better than other works using similar strategies.

Keywords

Electroencephalogram (EEG) • Epileptic seizure detection • DWT • MODWT • Self-recurrent Wavelet neural networks (SRWNN)

Introduction

The human brain is a complex system that exhibits rich spatio-temporal dynamics. Epilepsy is a common brain disorder that affects about 1 % of the world population, where 25 % of such patients cannot be treated properly by any available therapy [1]. Epileptic seizures are manifestations of epilepsy; these seizures are seen as a sudden abnormal function of the body, often with loss of consciousness, an increase in muscular activity or an abnormal sensation [2]. Among the noninvasive techniques for probing human brain dynamics, electroencephalography provides a direct measure of cortical activity with a millisecond temporal resolution. An EEG signal can provide valuable insight and improved understanding of the mechanism causing epileptics disorders. Since in the human brain there are millions of neurons interconnected in a very complex manner, the resultant EEG signal is complex, nonlinear and non-stationary in nature. A non-stationary signal is one whose basic statistical properties, such as the mean and variance do not remain constant over time [3].

Analysis, detection and classification are required in many applications where signals are non-stationary and/or multicomponent [4]. Lately, the EEG analysis has been mostly focused on epilepsy seizure detection diagnosis [2, 5, 6], which consists of normal and seizure EEG signals using methods such as Empirical Mode Decomposition (EMD), Wavelet Transforms and Artificial Neural

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_33, © Springer International Publishing Switzerland 2015

Networks. EMD is a spontaneous multi resolution method that represents nonlinear and non-stationary data as a sum of oscillatory modes inherent in the data, called Intrinsic Mode Functions (IMFs) [7]. Wavelet transform is one subclass of time-scale transforms. It has been used for representing various aspects of non-stationary signals [8]. Traditional methods rely on experts to visually inspect the entire length EEG recordings of up to 1 week, which is tedious and time-consuming [9]. Therefore, in recent years several models have been proposed, some of them based on wavelet analysis and artificial neural networks. The combination of both theories seeks to exploit the features of analysis and decomposition of wavelet processing along with the properties of learning, adaptation and generalization of neural networks. Despite of all works recently published, still there is a need to improve the classification accuracy obtained by the available models, as well as the generalization capabilities of such classifiers. As a result we are looking for novel models based on neural networks and wavelet analysis [10]. In this paper, we present the results obtained by a classifier based on Infinite Impulse Response (IIR) and Finite Impulse Response (FIR) filters, Wavelet Transforms (WT) and Feed-Forward Artificial Neural Networks (FF-ANN). The database provided by the University of Bonn [11, 12] was used to assess this model and to compare it with similar works. Our model, tested using threefold cross-validation, was able to obtain an accuracy of 99.26 %, which is better than the results obtained by similar works using the same database [9, 13, 14].

The rest of this paper is organized as follows. Section "Related work" presents an overview of recent works related to epileptic classification, using wavelet analysis and neural networks; the EEG database, pre-processing, feature extraction and classification method proposed in this work are described in Section "Materials and methods"; Section "Results" provides the experimental results and Section "Conclusions" summarizes the conclusions drawn from previous sections.

Related Work

An electroencephalographer, although guided by the general definitions for epileptogenic sharp transient waveforms, uses additional subjective criteria based on contextual information and others heuristics to reach a decision [15]. Visual screening of EEG records requires highly trained professionals. An automated EEG epilepsy diagnostic system would be very useful to improve this medical diagnosis. Several approaches have been proposed for this task. We briefly describe some works that report results using the Bonn database, which is used in our research (see Section "Experimental Data" for further details). Performance metrics such as accuracy, sensitivity and specificity are described in Section "Results".

A method of analysis of EEG signals based on WT and classification of EEG signals using FF-ANN and logistic regression (LR) are presented by Subasi and Ercelebi [9]. They used lifting-based discrete wavelet transform (LBDWT) as a preprocessing method. A LR and a FF-ANN classifiers were compared using EEG data owned by the authors. This database consists in 500 segments of EEG signals. A classification accuracy of 89 % of EEG signals was obtained by logistic regression and a classification accuracy of 92 % by FF-ANN trained using Levenberg-Marquardt algorithm.

Tzallas et al. [6] demonstrated the suitability of the time-frequency (*t-f*) analysis to classify EEG segments for epileptic seizures and they compared several methods for (*t-f*) analysis of EEGs. Short-time Fourier Transform and several (*t-f*) distributions were used to calculate the power spectrum density (PSD) of each segment. A FF-ANN was used for the classification of the EEG segments (that is, to determine the existence of an epileptic seizure). The method is evaluated using a benchmark EEG dataset of the University of Bonn [11, 12] obtaining 89 % of classification accuracy.

Another proposal based on FF-ANN incorporating a sliding window technique for pattern recognition is presented by Anusha et al. [15] for detection of epilepsy based on EEG signals. This work used 50 segments of EEG, 25 cases of healthy patients and 25 of epileptic patients of the database of University of Bonn (it uses two data sets, Z and S) [11, 12]. The classification accuracy obtained was 93.37 % for signals of normal patients and 95.5 % for epileptic patients.

A wavelet-chaos-neural network model for classification of EEGs of healthy (normal), ictal (seizure-active), and interictal patients is presented by Gosh et al. [14]. Interictal refers to the period between seizures. Wavelet analysis is used to decompose the EEG into delta (δ), theta (θ), alpha (α), beta (β), and gamma (γ) sub-bands (see Section "Feature Extraction" for the meaning of each sub-band). Three parameters are employed to represent each segment of the EEG: standard deviation, correlation dimension, and largest Lyapunov exponent. A mixed-band feature space consisting of nine parameters and a Levenberg-Marquardt Backpropagation Neural Network (LMBPNN) obtained a classification accuracy of 96.7 %, using the EEG database of the University of Bonn [11, 12]. Experiments were performed using the data sets named Z, F and S.

A classification system for epilepsy based on FF-ANN and features extraction from EEG, based on wavelet is presented by Shaik and Srinivasa [13]. They used features of Energy, Covariance Inter-quartile range (IQR) and Median Absolute Deviation (MAD) from each sub-band of EEG as input to a classifier based on FF-ANN. The authors

Table 1 Comparison of several published works related to detection of epilepsy

Authors	Type classifier (hidden nodes)	Features extraction	Dataset	Accuracy (%)	Sensitivity (%)	Specificity (%)
Subasi et.al. [9]	FF-ANN (21)	LBDWT Db4	Own data	92	91.6	91.4
Tzallas et.al. [6]	FF-ANN (15)	T-F Analysis	Bonn (O, Z, F, N, S)	89	89.0	89.1
Anusha et.al. [15]	FF-ANN (20)	T-F Analysis	Bonn (Z, S)	93.3	–	–

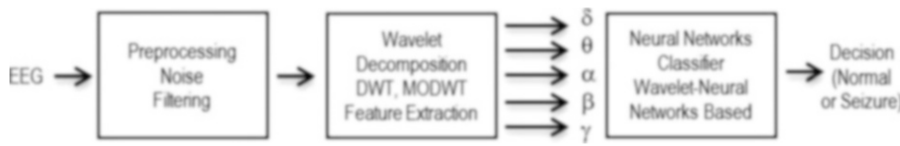

Fig. 1 General block diagram for seizure classification [10]

divided all segment of EEG signal of the database into 23 sub-segments (1 s each) generating 2,300 samples from each set of the database from University of Bonn [11, 12]. This work obtained 98 % of classification accuracy.

In a previous work we reported the use a FF-ANN for classification of ictal/normal states [10]. An accuracy of 90 % was obtained using a Maximal Overlap Discrete Wavelet Transform (MODWT) [19] based on a second order Daubechies (Db2) for characterizing the signal and a FF-ANN with 12 hidden nodes for classification. The MODWT was applied on segments of 23.6 s taken from the subset Z and S of the database provided by the University of Bonn [11, 12]. The EEG signals were filtered using a digital Butterworth low-pass filter of order 10 and cut off frequency of 64 Hz.

Table 1 summarizes these works in terms of classification accuracy, sensitivity and specificity. Notice that the highest accuracy is 98 %.

Materials and Methods

In this work, Wavelet Transforms (WT) are used to extract features of EEG signal and ANN are used to classify Epileptic seizure. The results of two experiments each using different filters, wavelets and size of samples of the database are described. Figure 1 shows the general block diagram of the proposed approach, which is divided into three modules: preprocessing, feature extraction and classification. In what follows, we explain each module.

Experimental Data

The EEG database used in the experiments was provided by the University of Bonn [11, 12]. This collection contains EEG data coming from three different events, namely, healthy subjects, epileptic subjects during seizure-free intervals (known as interictal states) and epileptic subjects during a seizure (ictal states) [11, 12].

The collection contains five datasets identified as: O, Z, F, N and S; each set holds 100 segments of EEG signals of 23.6 s. The sampling frequency of these signals was 173.6 Hz, so each segment contains 4,096 samples. Sets O and Z were obtained from healthy subjects with eyes open and closed respectively; sets F and N were obtained during interictal states in different zones of the brain and set S was gotten from a subject during ictal state [6]. In order to make a fair comparison with some of the works described in Section "Related Work", sets Z and S were used only for the results reported here.

Preprocessing

A filtering of the EEG signals was performed in order to remove noise added during recording. Some physiological researchers consider that EEG frequencies above 60 Hz are noise and can be neglected [17]. Considering this value, the cut-off frequency of the low-pass filters used here was set to 64 Hz. The value 64, which is an exact power of two, was used instead of 60 Hz, in order to obtain more easily the frequency sub-bands of the EEG during the wavelet analysis. Two approaches for filtering were tested: Finite Impulse Response (FIR) and Infinite Impulse Response (IIR). A FIR filter is one whose impulse response (or response to any finite-length input) is of finite duration, because it settles to zero in finite time. An IIR filter may has internal feedback and responds indefinitely, usually decaying [16]. Low-pass filters were designed with 3 dB of ripple in the pass-band from 0 to 64 Hz and at least 60 dB of attenuation in the stop-band [16]. An IIR Chebyshev type II filter of order 24, an Elliptic filter of order 9, a FIR filter Equiripple of order 343 and a Least Squares filter of order 350 [16] were implemented using Matlab 2010a and the Signal Processing Toolbox Version 6.19.

Figure 2 shows a segment of 4,096 samples of a filtered EEG from a healthy subject (Fig. 2a) and an ictal subject (Fig. 2b) with their corresponding frequency spectrum. Notice the differences in the frequency range of each subject.

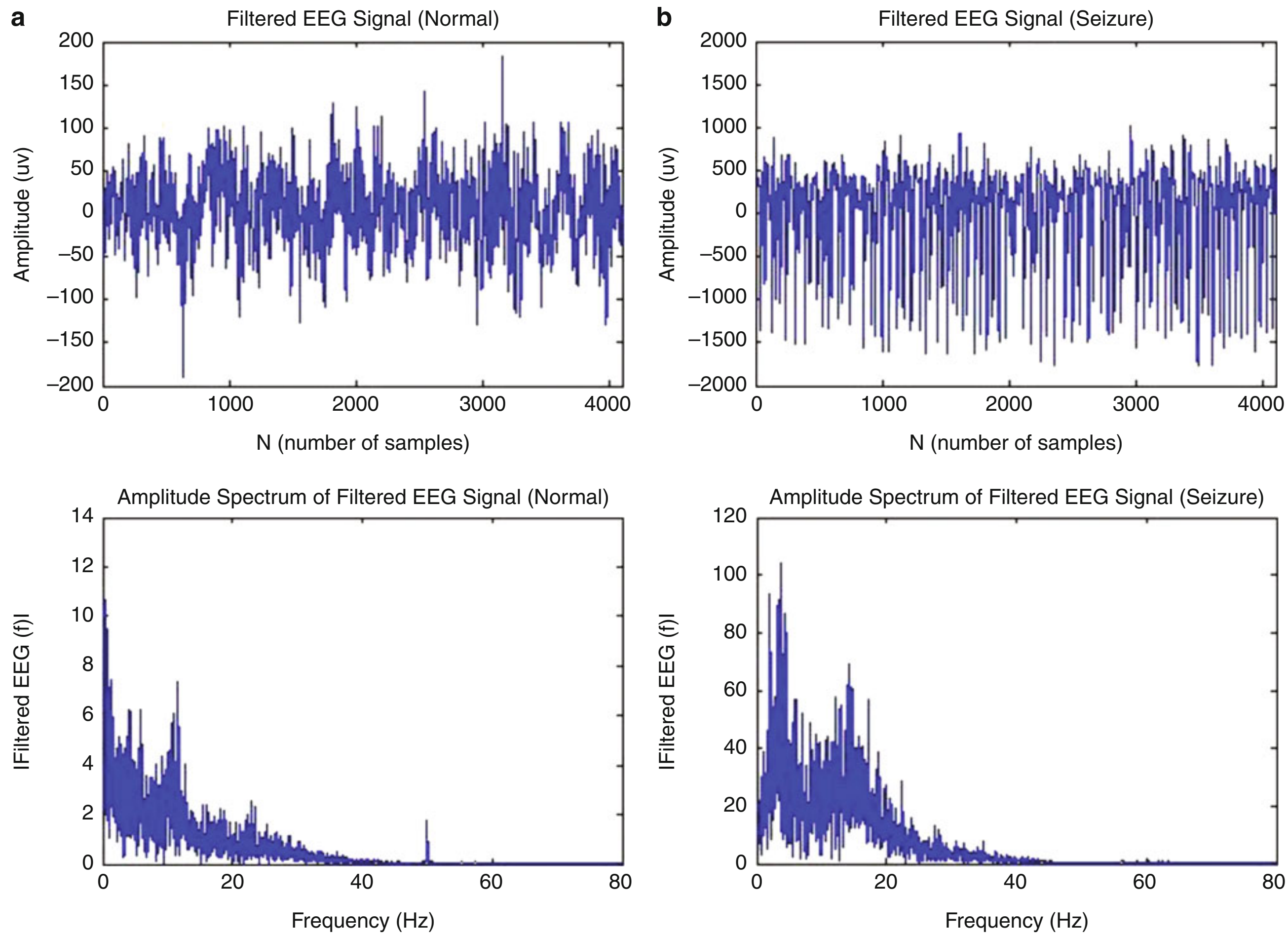

Fig. 2 Filtered signals EEG by a Least Squares FIR filter and its frequency spectrum of: (**a**) Healthy subject, (**b**) Ictal subject. Upper plots are samples from EEG signals and the lower plots show the frequency components of these EEG signals

Upper plots of Fig. 2 correspond to EEG segments and lower plots are the corresponding frequencies. Notice that frequency components above to 64 Hz have been eliminated due to the low-pass filtering.

Feature Extraction

In this work wavelet analysis was used to decompose the EEG signals into delta (δ), theta (θ), alpha (α), beta (β), and gamma (γ) sub-bands. Delta (δ) waves are between 0 and 4 Hz, shown during deep sleep, infancy and serious organic brain disease [18]. Theta (θ) waves have frequencies between 4 and 8 Hz, shown mainly in parietal and temporal regions in children and during emotional stress in some adults [18]. Alpha (α) waves have frequencies between 8 and 12 Hz; they are found in EEGs of almost all normal subjects when they are awake but in quiet, resting and relaxed condition [18]. Beta (β) waves normally occur in frequencies from 12 to 30 Hz. A beta wave is normally associated with active thinking, active attention or problem solving, that is, during intense mental activity [18]. Gamma (γ) waves show frequencies above 30 Hz, related to information processing and the onset of voluntary movements [18]. According to Ravish [18] and Sunhaya [2], the delta and alpha sub-bands provide useful information to localize a seizure. Therefore, only these sub-bands of the EEG signal were used in this work. The wavelet analysis was carried out using both a Discrete Wavelet Transform (DWT) and the Maximal Overlap Discrete Wavelet Transform (MODWT) [19]. In both cases a Haar wavelet, a second order Daubechies (Db2) wavelet and a fourth order Daubechies (Db4) wavelet were used. The selection of the wavelet must be related to the common features of the events present in real signals. That is, the wavelet should be well adapted to the events to be analyzed. Different wavelet families have a trade-off between the degree of symmetry (i.e., linear phase characteristics of wavelet) and the degree to which ideal high-pass filters are approximated (i.e., frequency response functions). The degree of symmetry in a wavelet is important

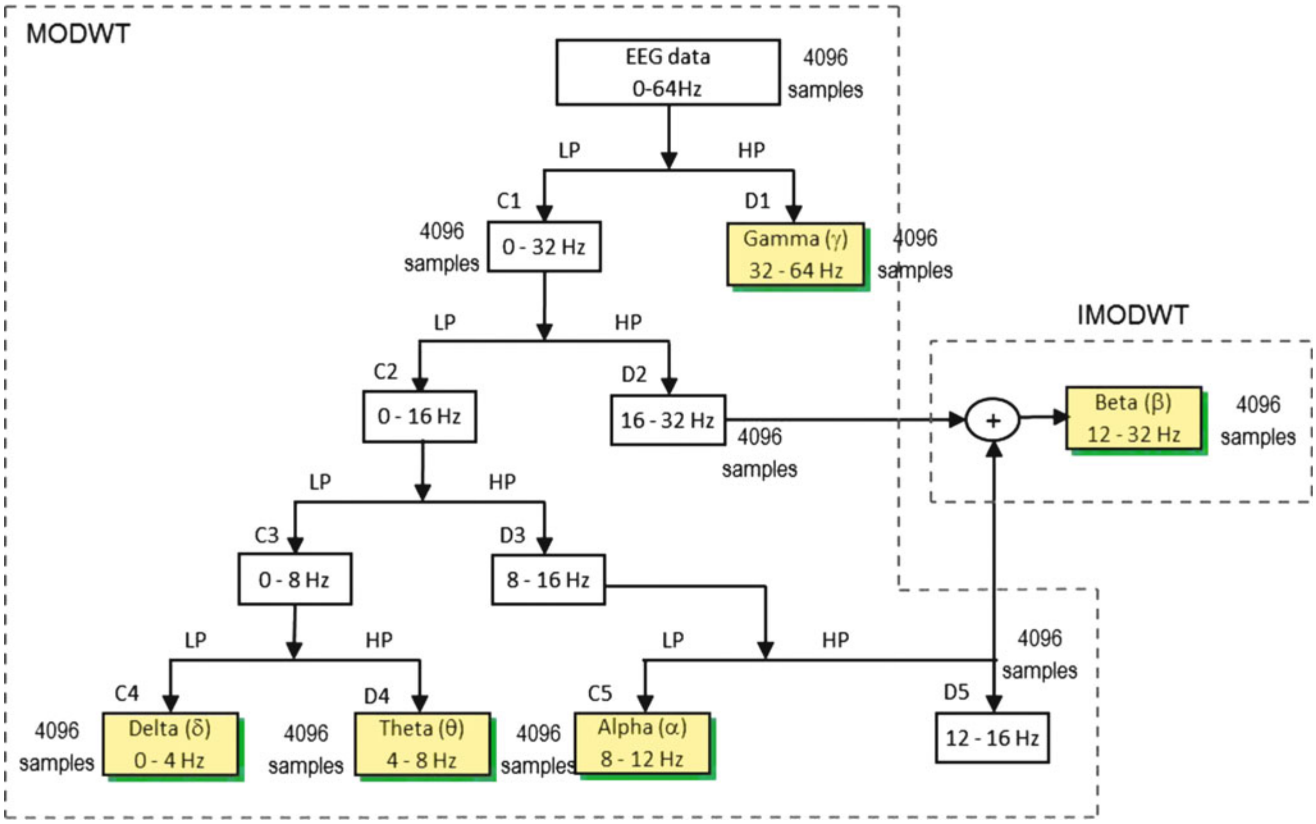

Fig. 3 Decomposition of EEG in physiological sub-bands by MODWT. The figure shows the name of sub-bands and its respective frequency ranges [10]

in reducing the phase shift of features during the wavelet decomposition. If the phase shift is large, it can lead to distortions in the location of features in the transform coefficients [19].

Figure 3 illustrates the decomposition of an EEG time series using a four-level MODWT extracting five physiological sub-bands [10]. Figure 4 shows the delta (0–4 Hz) and alpha (8–12 Hz) sub-bands of an EEG segment of a healthy subject, obtained using MODWT (Db2). Plots on the left side of the figure correspond to sub-bands and plots on the right side correspond their frequency components. Figure 5 shows the same for an epileptic subject. Each segment of EEG was represented by a feature vector of six components, built using the mean, absolute median and variance of both delta and alpha sub-bands. These features were also used in the work reported in [10].

Classification

A FF-ANN with one hidden layer was used to build the classifier. The network has six input nodes (one for each feature) and two output nodes (one for each class). The experiments reported here were executed using the code provided by [20], which is implemented in Matlab 2010a using the Neural Network Toolbox Version 6.0.3. The stopping criterion for learning algorithm was set to a value of 0.01 in the Mean Square Error (MSE); the learning rate was fixed at 0.5. The number of training epochs was fixed at 1,000 and the activation function for all nodes was a sigmoid. These values were experimentally chosen and similar to the ones reported by [20]. In order to find the best number of hidden nodes for the network, several tests were done using 6, 9, 12, 15, 16, 18, 21 and 24 nodes in the hidden layer of the FF-ANN. The network was trained using 200 segments of EEG signals of the database of University of Bonn [11, 12].

Results

Two experiments were carried out. In the first experiment, feature vectors to train the network were obtained using the whole segments coming from sets Z and S in the database (see Section “Experimental Data”). Note that in order to use a training set with a total number of elements divisible by three, the last two segments were eliminated. A total of 198 samples were generated; 132 samples were used for training

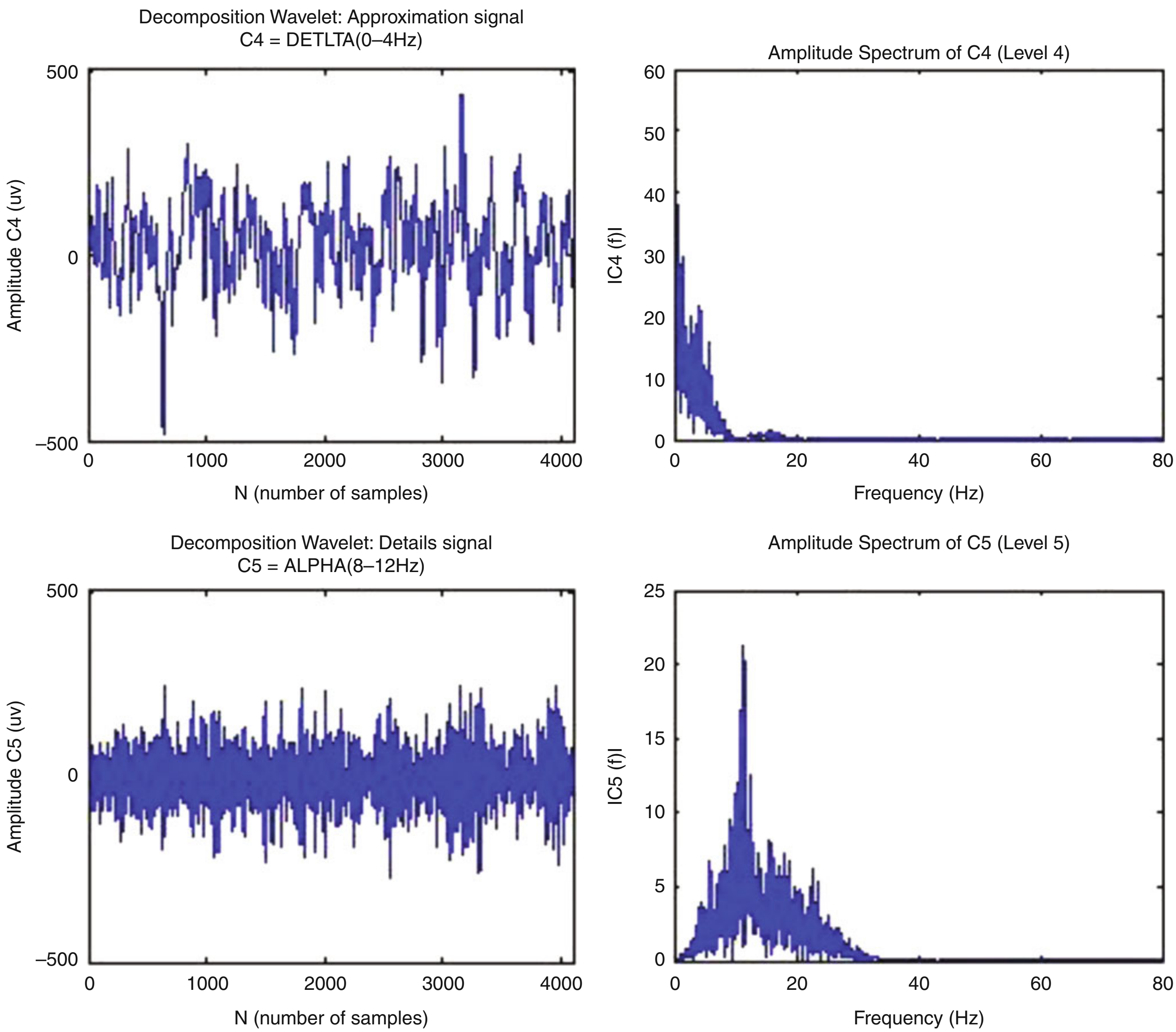

Fig. 4 Delta and Alpha sub-bands of an EEG signal obtained by MODWT (Db2) of a healthy subject (normal). The *graphs* on the *left side* show the obtained sub-bands and the *graphs* on the *right side* show its corresponding frequency spectrum

and 66 patterns were used for testing the FF-ANN. For the second experiment, features are obtained using portions of EEGs available in the database. We divided each EEG segment into 23 sub-segments (1 s) for decomposition by DWT, whereas that for decomposition by MODWT each segment was divided into 32 sub-segments (0.7375 s). This is done in order to provide the classifiers with more data to be learned. A total of 4,599 patterns (3,066 for training and 1,533 for testing) are used for the classification when DWT is used, and 6,399 patterns (4,266 for training and 2,133 for testing) when MODWT is applied.

Each experiment was tested using threefold cross validation. Besides, in order to avoid bias generated by the randomness of initial weights in the networks, each case was executed five times, and an average of the performance is reported. The results are evaluated in terms of classification accuracy, sensitivity and specificity. Sensitivity (also called *the recall rate*) measures the proportion of actual positive results which are correctly identified as such. Specificity measures the proportion of negative results which are correctly identified as such [21]. Sensitivity and specificity are calculated as follows:

$$sensitivity = \frac{TP}{TP + FN} \times 100\% \tag{1}$$

$$specificity = \frac{TN}{TN + FP} \times 100\% \tag{2}$$

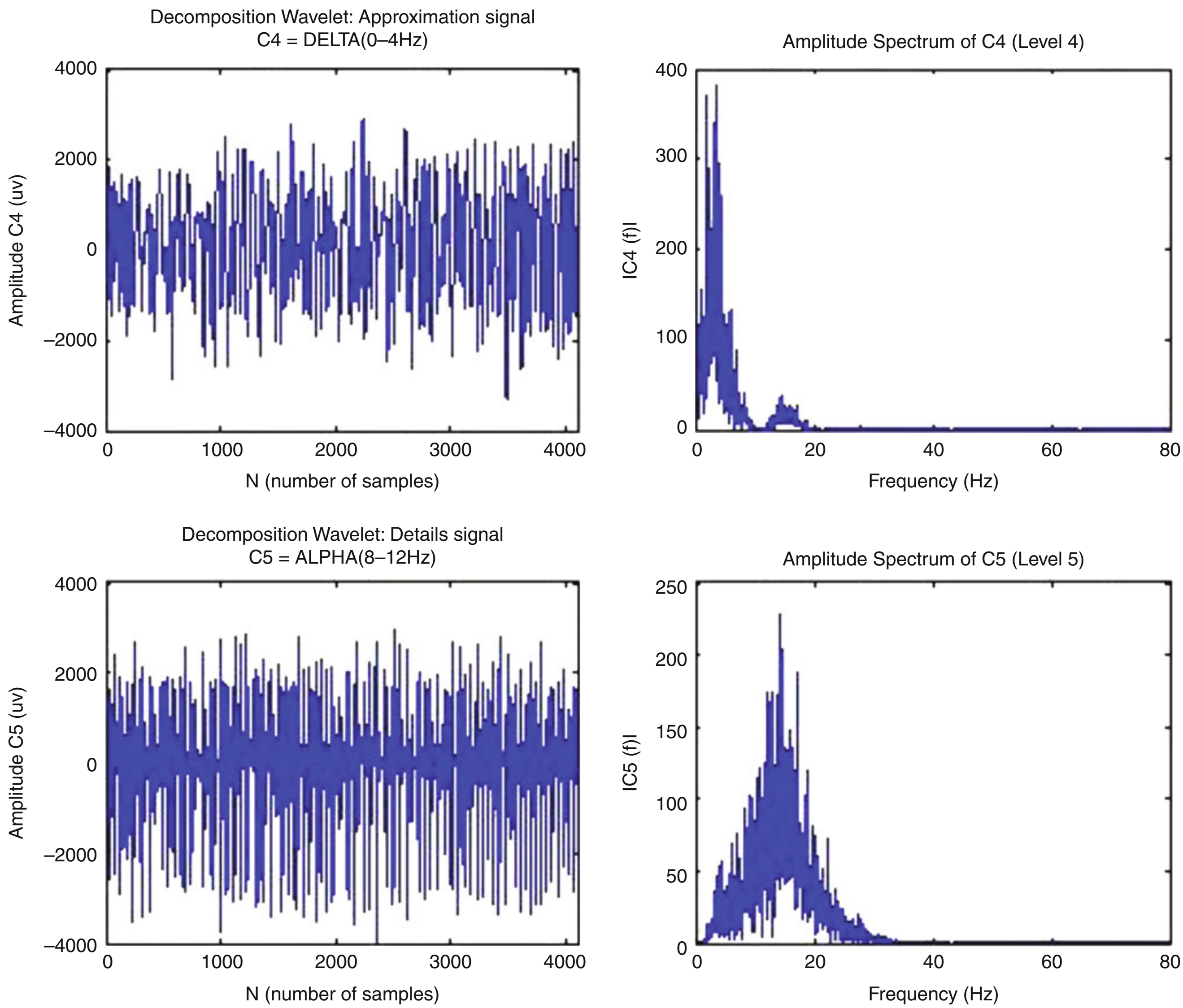

Fig. 5 Delta and Alpha sub-bands of an EEG signal obtained by MODWT (Db2) of a ictal subject (seizure). The *graphs* on the *left side* show the obtained sub-bands and the *graphs* on the *right side* show its corresponding frequency spectrum

where TP (True positive) = correctly identified; FP (False positive) = incorrectly identified; TN (True negative) = correctly rejected and FN (False negative) = incorrectly rejected [21].

Experiment I

As explained previously, during this experiment FF-ANN classifiers were trained using patterns coming from whole segments of EEG. The EEGs were filtered using [16]: low-pass IIR Chebyshev type II and Elliptic; low-pass FIR Equiripple and Least Squares. Table 2 shows the results obtained in each case using the FF-ANN classifier with different number of hidden nodes. The best result obtained in this experiment was 93.23 % of accuracy using features calculated by a Least Squares FIR filter and by DWT (Db2) with six hidden nodes in the FF-ANN.

Experiment II

For this experiment low-pass filters Digital Chebyshev type II and digital Elliptic were used [16]. Table 3 shows the best results obtained in each case using the FF-ANN classifier for different filters and number of hidden nodes. The best result obtained in this experiment was 99.26 % of accuracy using features calculated by a Chebyshev II filter and by DWT (Haar) with 18 nodes in the hidden layer of the FF-ANN. The fact that Haar wavelet has compact support

Table 2 Experiment I: results of the FF-ANN classifier using whole segments

Filter	Wavelet	Hidden nodes	Accuracy (%)	Standard deviation	Sensitivity (%)	Specificity (%)
Chebyshev II	DWT—Haar	9	82.82	21.10	82.79	74.64
Chebyshev II	DWT—Db2	15	83.73	21.52	82.56	82.04
Chebyshev II	DWT—Db4	6	91.11	14.59	91.72	88.37
Chebyshev II	MODWT—Haar	9	83.33	16.58	80.78	86.42
Chebyshev II	MODWT—Db2	6	84.94	16.40	86.06	84.52
Chebyshev II	MODWT—Db4	18	84.44	21.33	86.44	85.74
Elliptic	DWT—Haar	21	88.38	16.76	86.46	91.58
Elliptic	DWT—Db2	6	80.30	21.73	79.87	76.85
Elliptic	DWT—Db4	9	82.82	20.20	80.19	87.61
Elliptic	MODWT—Haar	21	85.45	18.21	84.13	82.21
Elliptic	MODWT—Db2	6	90.00	12.91	88.17	96.32
Elliptic	MODWT—Db4	24	87.17	14.85	87.22	85.61
Equiripple	DWT—Haar	18	87.17	19.09	85.47	90.96
Equiripple	DWT—Db2	12	83.03	20.57	81.48	85.36
Equiripple	DWT—Db4	18	86.56	17.05	85.90	91.14
Equiripple	MODWT—Haar	6	87.07	17.66	84.83	92.02
Equiripple	MODWT—Db2	6	88.88	18.52	89.17	82.80
Equiripple	MODWT—Db4	6	85.52	19.04	83.71	84.74
Least Squares	DWT—Haar	6	84.44	21.32	80.00	82.82
Least Squares	DWT—Db2	6	93.23	14.85	93.87	90.07
Least Squares	DWT—Db4	18	82.72	20.13	91.24	81.42
Least Squares	MODWT—Haar	9	83.73	18.32	82.71	90.97
Least Squares	MODWT—Db2	21	84.14	15.96	82.76	87.16
Least Squares	MODWT—Db4	21	87.37	15.82	85.88	90.89

Table 3 Experiment II: results of the FF-ANN classifier using sub-segments

Filter	Wavelet	Hidden nodes	Accuracy (%)	Standard deviation	Sensitivity (%)	Specificity (%)
Chebyshev II	DWT—Haar	18	99.26	0.26	98.93	99.59
Chebyshev II	DWT—Db2	18	99.03	0.27	98.75	99.32
Chebyshev II	DWT—Db4	18	96.57	5.87	95.38	98.91
Chebyshev II	MODWT—Haar	15	99.24	0.32	98.86	99.64
Chebyshev II	MODWT—Db2	24	95.80	12.84	94.77	96.19
Chebyshev II	MODWT—Db4	24	97.72	4.55	96.76	99.13
Elliptic	DWT—Haar	21	95.49	11.10	97.48	95.30
Elliptic	DWT—Db2	21	95.96	12.33	96.65	96.23
Elliptic	DWT—Db4	9	98.44	1.26	97.91	99.06
Elliptic	MODWT—Haar	18	95.98	12.52	95.49	99.74
Elliptic	MODWT—Db2	6	99.12	0.40	98.73	99.51
Elliptic	MODWT—Db4	24	95.34	12.62	91.27	96.14

and only one vanishing moment compared with others wavelets that have more vanishing moments may be a reason for obtaining the best result using this type of wavelet. That is, the resulting wavelet basis functions are unsuitable as basis functions for classes of smoother functions and the EEG signals are not smooth signals. Notice that the second and third best results were 99.24 % and 99.12 % of accuracy, respectively. These results were obtained with features calculated by MODWT. The values about standard deviation, sensitivity and specificity are similar to the best result of this experiment.

Conclusions

In this work, we present the results of a model based on wavelet analysis and neural networks for identification of seizures events of epilepsy. Inspired from previous results reported in [10], and [13], and in order to find a suitable combination to improve the results reported for this problem we tested several filters, wavelets and wavelet transformations. In addition, we tested the use of segments and sub-segments for training the classifier. Two types of wavelets transforms (DWT and MODWT) with Haar, Db2 and Db4 were used for decomposition of EEG segment and sub-segments. Six features were used to train a FF-ANN: mean, absolute median and variance of Delta and alpha sub-bands. When using whole segments for training, 93.23 % of accuracy was achieved, whereas when using sub-segments for training, 99.26 % of accuracy was achieved. The result of this experiment improved the 98 % obtained in [13]. As future work we will analyze the use of a classifier based on a Self-Recurrent Wavelet Neural Network (SRWNN) [21, 22] for classification and EMD for feature extraction. Furthermore, other training algorithms and features will be explored.

Acknowledgment The first author gratefully acknowledges the financial support from the Universidad Autónoma de Tlaxcala and PROMEP by scholarship No. UATLX-244. This research has been partially supported by CONACYT, project grant No. CB-2010-155250.

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A Bargaining Approach for Disseminating Context Information to Context-Aware Services

Elarbi Badidi

Abstract

The services landscape is changing with the growing acceptation of cloud computing and the proliferation of smartphones and Internet-enabled handheld devices. Users are increasingly demanding services that can adapt to their current context. Hence, context-aware services need up-to-date context information to be able to adapt their behavior to current situations of users. We propose, in this paper, a framework for context information dissemination, which relies on negotiated Context Level Agreements (CLAs) between context consumers and context providers. A Context Broker (CB) helps context consumers select suitable context providers that can offer their required context information and fulfill their quality-of-context (QoC) requirements. Furthermore, the CB is in charge of negotiating the CLA terms—using a multi-attributes negotiation model—with a selected context provider on behalf of the context consumer.

Keywords

Context services • Quality-of-context • Context level agreement • Context negotiation

Introduction

Mobile users are increasingly demanding services tailored to their current context and circumstances. Therefore, business services should be context-aware to adapt their behavior to the changing circumstances/environment of the user. Several definitions of the notion of context have been provided in the literature. According to Dey [1], "*Context is any information that can be used to characterize the situation of an entity. An entity is a person, place, or object that is considered relevant to the interaction between a user and an application, including the user and applications themselves.*" Context information that supports such adaptation is typically obtained from a context management system, which collects, aggregates and supplies context information from various context information sources.

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Existing context-aware services implicitly consider that context information used to adapt their behavior is correct and reliable. This hypothesis is obviously not well matched when considering the effective conditions in real pervasive situations, where raw context data is obtained using various, and possibly unreliable, sensors. Moreover, the aggregation process of raw context data might introduce additional biases. To cope with this reliability issue, context information is characterized by some properties referred in literature as quality-of-context (QoC) indicators. Buchholz et al. [2] defined the QoC as: "*Quality of Context (QoC) is any information that describes the quality of information that is used as context information. Thus, QoC refers to information and not to the process nor the hardware component that possibly provide the information.*"

Buchholz et al. [2] and Sheikh et al. [3] identified the following QoC indicators: *precision*, *freshness*, *temporal resolution*, *spatial resolution*, and *probability of correctness*. *Precision* represents the granularity with which context information describes a real world situation. *Freshness* represents the time that elapses between the

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_34, © Springer International Publishing Switzerland 2015

determination of context information and its delivery to a requester. *Spatial resolution* represents the precision with which the physical area, to which an instance of context information is applicable, is expressed. *Temporal resolution* is the period of time during which a single instance of context information is applicable. *Probability of correctness* represents the probability that a piece of context information is correct. QoC has a great impact on the behavior of context-aware services. Using lower quality context information might increase the risk of making incorrect context-based decisions. In other words, the more reliable context information is, the more accurate and adaptable service that a context-aware service might offer to its customers.

One of the main issues being faced in the area of pervasive computing is managing and distributing context information efficiently to enhance personalized service delivery to mobile users. Context sources as well as context-aware services, which require context information, are very often physically distributed. For instance, context sources providing information about the current temperature may be far from the applications that need to adapt their services to the prevailing weather conditions. Furthermore, it is likely that these context sources provide the same context information but with different QoC. Context-awareness raises new challenges like aggregation of context information in a structured format, discovery and selection of suitable context sources.

To deal with these issues, we set out to develop a framework that will: (1) allow context consumers to express their context and QoC requirements, (2) allow implementing QoC-driven selection of context providers, (3) allow implementing CLA negotiation, and (4) permit assessing of CLAs implementation. The main component of our proposed framework is the Context Broker (CB), which is in charge of mediating between context consumers and context providers in order to reach agreements that explicitly describe expected context information and QoC levels. Besides, The CB carries out other activities such as selection of context providers and CLA negotiation.

The rest of this paper is organized as follows. The next section describes background information on the concepts of context services and Context Level Agreements. Section "Related work" describes related work. Section "CLA-driven context dissemination framework" presents an overview of the proposed framework. Section "CLA Negotiation" describes the CLA negotiation process. Section "The negotiation model" describes the corresponding multi-issues negotiation model. Finally, Section "Conclusion and future work" concludes the paper and describes future work.

Background

Context Services

To adapt their behavior to the changing environment and user circumstances, context-aware services have traditionally used one of the following two methods:

- Each service obtains raw context information directly from sensors or from other context sources, aggregates the raw data to obtain high-level context information, which is then used to make decisions.
- Services use a common context management system that collects raw data from numerous context sources, aggregates that data to provide high-level context information to services.

With the advent of service oriented computing, these context management systems have evolved to what is called *context services*. A context service typically provides infrastructure support for collection, management, and dissemination of context information concerning a number of subjects. Subjects may be users, objects such as handheld devices and equipment, or the environment of users. The context service acquires context information from various context sources. Sources are usually third parties that collect and provide context information. For example, consider the "temperature" at the current location of the mobile user. This information may be obtained directly from the mobile device of the user. It can also be obtained from a local weather station. Alternatively, it may be obtained from weather TV channels providing weather information nation-wide.

Several research works have investigated the design and the implementation of context services. Schmidt et al. [4] designed and implemented a generic context service with a modular architecture that allows for context collection, discovery and monitoring. This context service provides a Web service interface that allows its integration in heterogeneous environments. The implementation uses OWL to describe context information and SPARQL to query and monitor context information.

Lei et al. [5] described the design issues and the implementation of a middleware infrastructure for context collection and dissemination. They implement this middleware infrastructure as a context service. To allow for wide deployment of the context service, this work has addressed the following issues: extensibility of the context service architecture by supporting heterogeneous context sources, integrated support for privacy, and quality of context information support. Coronato et al. [6] proposed a semantic context service that relies on semantic Web technologies to support smart offices. It uses ontologies and rules to infer high-level context information, such as lighting and sound level, from low-level raw information acquired from context sources.

Given the massive amount of context data processed and stored by context services and the wide acceptance of the cloud computing technology, we proposed in our previous work [7] the deployment of context services in the cloud. One of the underlying advantages of the deployment of context services in the cloud is the economy of scale. By making the most of the cloud infrastructure provided by a cloud vendor, a context provider can offer better, cheaper, and more reliable services than is possible within its premises. The context service can utilize the full processing and storage resources of the cloud infrastructure if needed. Another advantage is scalability in terms of computing resources. Context providers can scale up when additional resources are required as a result of a rise in the demand for context information. Conversely, they can scale down when the demand for context information is decreasing. Another benefit of the approach is to enable context-aware application services to acquire their required context information on a pay-as-you-go basis and to select cloud-based context services on the basis of the price they have to pay and other criteria, such as the QoC they can get.

Context Level Agreements

A CLA is an arrangement between a context provider and a context consumer concerning the guarantees of delivered context information. It describes common understandings and expectations between the two parties. The guarantees concern the context information and the QoC levels to be delivered.

The typical components of a CLA are:

- *Parties*: represents the parties involved in the CLA and their respective roles (context consumer and context provider).
- *Activation time*: represents the period of time at which the CLA will be valid.
- *Scope*: defines the types of context information covered in the agreement.
- *Context-level objectives* (CLOs): Represent the levels of QoC that both parties agree on, and habitually include a number of quality indicators such as accuracy and freshness.
- *Penalties*: specifies the penalties for not meeting the stated context level objectives, such as getting discount or having the right to terminate the contract in light of unsatisfactory context levels.
- *Exclusions*: specifies what is not covered in the CLA.
- *Administration*: defines the processes to assess the CLA objectives, and describes the responsibility of the context provider regarding the control of each of these processes.

A CLA life cycle includes five phases, which are: Development, Negotiation and Sales, Implementation, Execution, and Assessment.

Related Work

The importance of considering QoC in designing and managing context-aware systems has been recognized in several works on context-awareness [2, 3, 8, 9]. Few works [10, 11] investigated the issues of modeling and measuring QoC. Filho et al. [10] described an OWL-DL based QoC model and methods for measuring QoC by taking into account the fact that context information might be modified after sensing and described into a high semantic level. Manzoor et al. [11] considered QoC to be composed of two components, QoC sources and QoC parameters. QoC sources represent the information concerning the sources, which collect context information, the subjects about which context information is collected, and the environment where context information is sensed and collected.

Context information brokerage is the subject of several research efforts over the last few years. Kiani et al. [12] present a model of a large-scale system for context dissemination that relies on a federation of context brokers interconnected through a publish-subscribe model. The model addresses scalability and mobility issues faced in a single-broker system that uses synchronous communication between the system's components. They also introduce a mobile broker to facilitate the involvement of mobile devices in context provisioning and consumption efficiently. Reetz et al. [13] evaluate the performance of a context information provisioning system that relies on a context broker using black-box measurements and a simulation model. They identify major parameters and related models for the response delay of the main components of the system. Chen et al. [14] describe a context broker-based architecture, called CoBrA, which uses Semantic Web technologies to provide support to pervasive context-aware systems. It employs the Web Ontology Language OWL for modeling context ontologies and for supporting context reasoning. It also enforces users privacy policies when sharing their context information.

Few works in the area of context-awareness in pervasive environments have investigated the issue of context negotiation. Rather, most research works on context-awareness focus on designing and implementing frameworks and middleware infrastructures for managing context information. Other works investigate the design and implementation

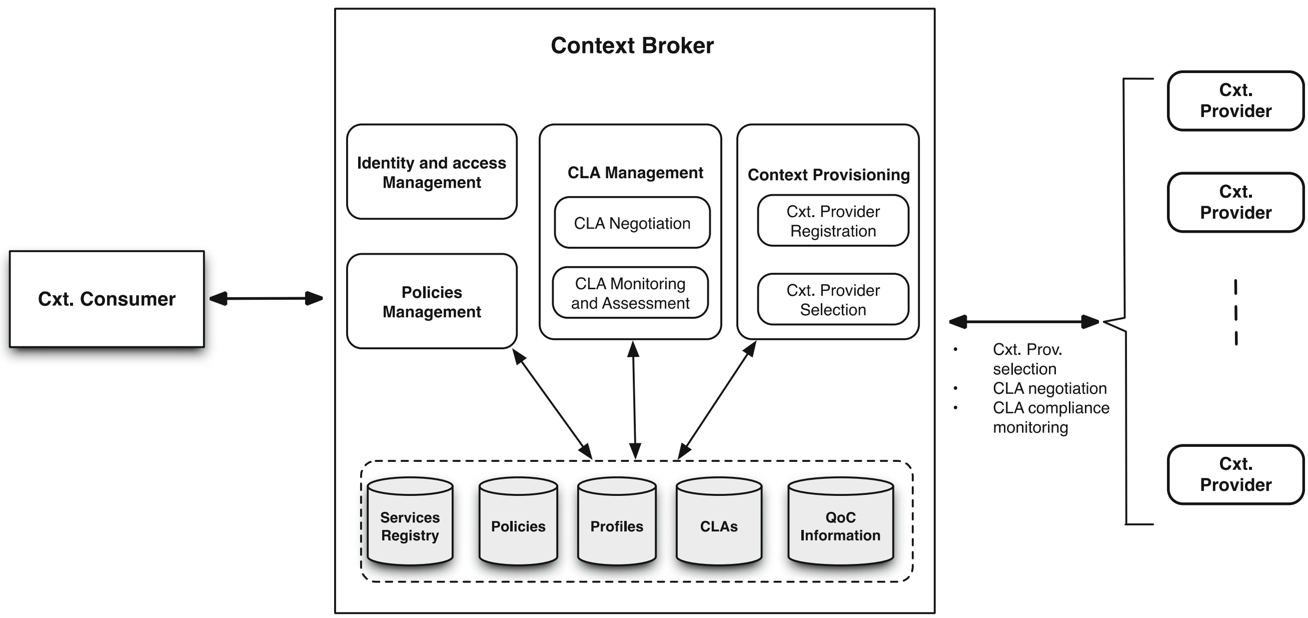

Fig. 1 Broker-based framework for context dissemination

of adaptive context-aware services. Moreover, many surveys have been made in order to understand the features and shortcomings of existing systems [15, 16, 17]. Baldauf et al. [15] provided a survey on many context-aware systems and compare them in terms of sensing support, context model, context processing, resource discovery, historical context data, security, and privacy. To the best of our knowledge, the most significant work that investigated the issue of context negotiation and establishing and negotiating CLAs is the work of Khedr et al. [18]. In this work, the authors described a multi-agent middleware, which uses a negotiation protocol, to facilitate development of adaptive context-aware personalized applications, and an ontology model to represent context information.

Our work aims at facilitating mediation between context consumers and context providers and providing support for automated CLA negotiation and management. Finding the right context offering is not an easy task for context consumers given the variety of context offerings. Moreover, dealing with a context provider requires knowledge of its operating environment, the availability of context management tools, and the service terms and conditions. Collecting this information for multiple context providers is likely to be an arduous task that is expensive and time consuming. The Context Broker with its know-how and value-added services will assist context consumers in: (a) finding appropriate context offerings, (b) negotiating CLA terms, and (c) offering a single interface to interact with multiple context providers.

CLA-Driven Context Dissemination Framework

Figure 1 depicts the proposed framework for CLA-based context dissemination. The main components of the framework are: context consumers (CCs), the Context Broker (CB), and Context Providers (CPs).

Context-Consumers

In our framework, context-aware services (CAS) are the consumers of context information obtained from context-providers. A CAS can be implemented as a Web service that can understand situational context and can adapt its behavior according to the user's activity and location and the changing circumstances in the user environment.

Context Broker

As we mentioned earlier, the CB is a mediator service that decouples context consumers from context providers. Given that context consumers do not normally have the capabilities to negotiate, manage, and monitor QoC, they delegate management tasks, such as selection of appropriate context providers and CLA negotiation, to the CB. The architecture of the CB includes several management operations that cooperate in order to deliver personalized services to clients.

These operations are: *Identity and Access Management (IAM)*, *Policies Management (PM)*, *CLA Management (CLAM)*, and *Context* Provisioning *(CP)*. They are under the control of a *Coordinator* component. The back-end databases maintain information about providers' policies, consumers' profiles and preferences, CLAs, and dynamic QoC information. The *Selection Manager,* which implements CP management operations, is in charge of implementing different policies for the selection of suitable context providers, based on the consumer's QoC requirements and the context providers' QoC offerings. In our previous work [19], we described a QoC-based algorithm for the selection of context services.

The *CLA Manager*, which implements CLAM management operations, is in charge of carrying out the CLA negotiation process between a context consumer and a selected context provider. It approaches this context provider to determine whether it can ensure the required level of QoC. Then, the context consumer and the context provider sign a contract. The contract describes the type of context information, the QoC level to ensure, the cost of service, and actions to take in case or repeated violations of the agreement. If the selected context provider is unable to deliver the required level of QoC, the broker selects another context provider and reiterates the negotiation process. The *Profile Manager*, which implements IAM management operations, is responsible for managing context consumers' profiles, including their preferences in terms of context information types and required QoC. The *Policy Manager*, which implements PM management operations, is responsible for managing different kinds of policies such as authorization policies and QoC-aware selection policies of context providers.

Context Providers

Context providers implement one or several context services, which may offer several types of context information—such as *location*, *temperature*, and *user activity*—that they may obtain from several context sources. In order to determine their current QoC offering, context providers need to use monitoring and measurement techniques to collect sensed data at selected measurement points. By aggregating collected data with its timestamps and data about sensors, a context provider can determine the current value of each QoC indicator for each category of context information. The *CLA Manager* of a context provider is responsible for managing CLA templates, negotiating the CLA terms with the CB, or directly with context consumers, and implementing CLAs.

CLA Negotiation

Figure 2 depicts the CLA negotiation process. The steps of this process are as follows:

Step 1: The context consumer submits a CLA request to the CB to find out an appropriate context provider that can provide required context information and fulfill its QoC requirements.

Step 2: After authenticating the context consumer, the *Coordinator* requests its profile from the *Profile Manager*. Then, it requests from the *Selection Manager* to select a suitable context provider, which can deliver required context information according to the context consumer requirements.

Step 3: The *Coordinator* requests policies of the selected context provider from the *Policy Manager*.

Step 4: If the context consumer profile is available in the profile repository, for example, because the context consumer had previously used some services of the CB, the *Coordinator* may determine whether the context providers, found by the *Selection Manager*, can handle or not the context consumer request. This decision relies on the profile of the context consumer and policies of the selected context providers.

Step 5: If the context consumer's profile is not available in the profile repository, then the *Coordinator* asks the context consumer to provide information, such as preferences and desired levels of QoC, in order to create a new profile for the context consumer.

Step 6: If at least one context provider can meet the context consumer requirements, the *Coordinator* requests from the *CLA Manager* to negotiate with that context provider the terms and conditions of context information delivery.

Step 7: The CB' *CLA Manager* forwards the CLA request, to the context provider's *CLA Manager*, requesting a proposal from the context provider. The context provider's *CLA Manager* parses the CLA request and validates it against its CLA templates.

Step 8: If the CLA request is acceptable to the context provider, then its *CLA Manager* responds to the CLA request by sending back a CLA proposal. The CB analyses the CLA proposal to determine whether it meets all requirements of the context consumer.

Step 9: If the context consumer's expectations can be met, then the CB accepts the offer of the context provider and sends a CLA confirmation to the context provider. Otherwise, it rejects the offer and makes a counter-proposal with different conditions, terms, costs, etc.

The operations of the *CLA Manager* of the context provider, with regards to CLA Negotiation, are as follows:

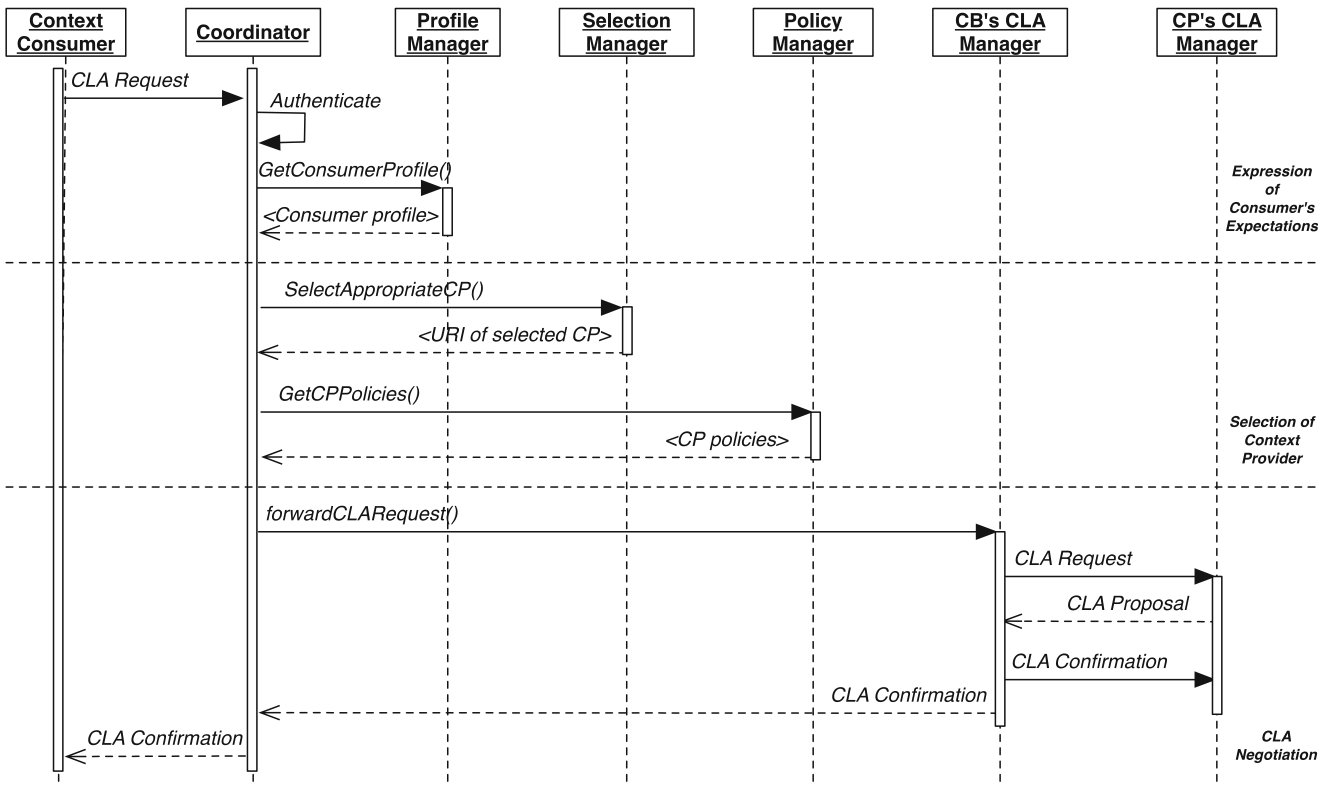

Fig. 2 CLA negotiation process

Step 1: After receiving a CLA request, the *CLA Manager* analyses the CLA request.

Step 2: If there is no issue with the structure of the CLA request, then the *CLA Manager* validates the CLA request against available CLA templates.

Step 3: If the CLA request is acceptable, then the context provider's *CLA Manager* creates a CLA proposal and sends it to the CB. At this point, the context provider will wait for CLA approval or dismissal from the CB.

Step 4: If the CLA request is not acceptable to the context provider, then its *CLA Manager* may suggest an alternative CLA to the CB and waits for acceptance or rejection.

Step 5: Upon reception of the CLA approval from the CB, the *CLA Manager* subscribes the context consumer in its registry and the CLA becomes ready for implementation.

In step 8 and step 9 of the above CLA negotiation process, a multi-attributes negotiation takes place between the *CLA Managers* of both parties according to the negotiation model that we describe in the next section.

The Negotiation Model

Selection algorithms do not guarantee that the selected context offer is the best one. Therefore, to reach an agreement, the context consumer and the selected context provider need to negotiate the various quality levels of context information to be delivered. We assume that QoC indicators are in normalized form with values between 0 and 1. A value of 1 means highest quality and 0 means lowest quality. When submitting a CLA request to the CB, the context consumer specifies her preferences with regards to the normalized QoC indicators that she can tolerate.

Let $X = \{X_1, X_2, \ldots, X_n\}$ be the list of QoC indicators considered in the system. Therefore, the context consumer and the context provider need to bargain on n QoC parameters. Multi-attributes bargaining is a process in which the two parties negotiate multiple issues concurrently. This is a common problem in business. The parties, very often, recognize the existence of differences of interest over several issues; but the need for win-win cooperation is the incentive for seeking to reach a compromise by making concessions.

The *CLA Managers* of both parties have to go through several rounds of negotiation of offers and counter-offers until they reach an agreement or reach a predefined maximum number of rounds. In each round, the CB's CLA Manager evaluates the utility function of each QoC attribute and the global utility function to determine whether the offer of the context provider is acceptable or not. In the following, we consider two quality attributes, *freshness* and *probability of correctness*, to illustrate the approach.

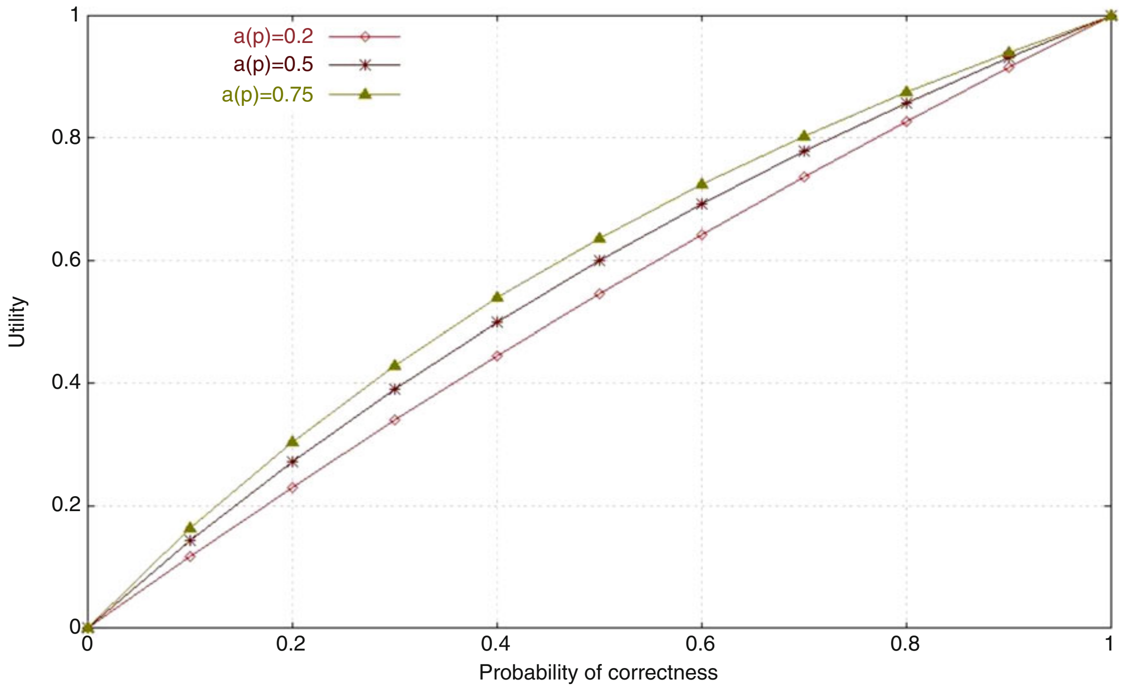

Fig. 3 Utility versus probability of correctness

$U_p(p)$ is the utility function for the *probability of correctness* quality attribute p. It has its maximum value of 1 when p = 1 and decreases to 0 when p is decreasing to 0.

$$U_p(p) = \frac{p(1+\alpha_p)}{1+\alpha_p}p \quad (1)$$

In Eq. (1), p is the probability of correctness and α_p is the CLA value for p.

Figure 3 shows $U_p(p)$ for $\alpha_p = 0.2, 0.5,$ and 0.75 respectively.

We consider also $U_F(f)$ the utility function for the *freshness* quality attribute f. It reaches its maximum, which is 1, when f is 0 and decreases to 0 when f reaches 1 as we are considering the normalized values.

$$U_p(f) = \frac{1-f}{1+\alpha_F f} \quad (2)$$

In Eq. (2), α_F is the CLA for the freshness.

Figure 4 shows $U_F(f)$ for $\alpha_F = 0.2, 0.5,$ and 0.75 respectively.

A global utility U is a function of the individual utility functions $U_i 1 \leq i \leq n$

$$U = f(U_i), 1 \leq i \leq n \quad (3)$$

U_i represents the individual utility function associated with the QoC attribute X_i. If we assume that the QoC attributes are independent, the global utility function U can be expressed by the additive linear utility function as:

$$U = w_1U_1 + w_2U_2 + \ldots + w_nU_n \quad (4)$$

w_i is the importance weight that the context consumer assigns to that attribute. Each weight is a number in the range (0,1) and $\sum{}_1^n w_i = 1$.

Conclusion and Future Work

Given the proliferation of mobile handheld devices, users are increasingly demanding services that can adapt to their current context and to their environment conditions. Context-aware services, which provide adaptive services to the users, are facing the challenge of finding appropriate context providers that can supply them with relevant and high quality context information about numerous subjects.

In this paper, we have presented a framework for context information dissemination that relies on a Context Broker, which is in charge of mediating between context consumers and context providers, selecting appropriate context

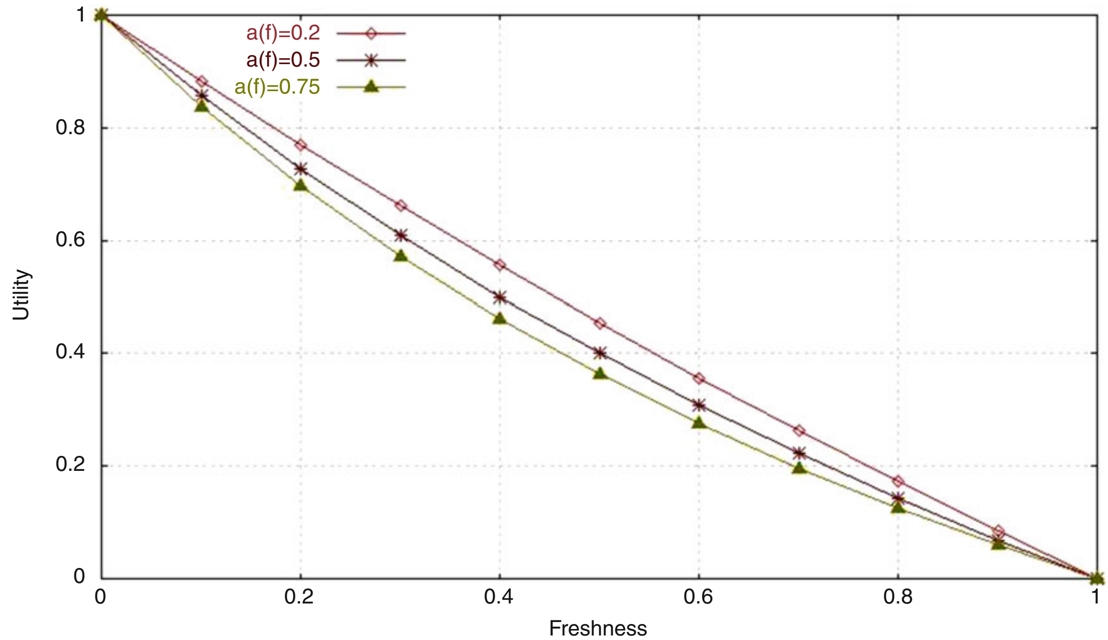

Fig. 4 Utility versus freshness

providers, and negotiating the CLA terms on behalf of context consumers. CLA negotiation involves the negotiation of multiple CLA parameters with both parties trying to maximize their utility function in a series of offers and counter-offers. We have considered two quality attributes—*freshness* and *probability of correctness*—to illustrate examples of individual utility functions and the global additive utility function to allow the context consumer to evaluate the offer of the context provider.

As a future work, we intend to explore the use of non-linear utility functions for the multi-attributes negotiation model and build a prototype of the framework together with some real scenarios for CLA-based context dissemination.

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Automatic Generator of Decoupling Blocks Using Genetic Programming

M. Montes Rivera, M. Paz Ramos, and J.L. Orozco Mora

Abstract

This paper describes a new method for decouple systems with an automatic generator of decoupling blocks using genetic programming and shows that the decoupling of MIMO systems by this method is easier and more powerful than the results obtained using a regular decoupling state feedback technique in a paper production machine process and an irrigation of fields process.

Keywords

Automatic decoupling • Decoupling multivariable systems • Genetic programming • Regular decoupling state feedback

Introduction

The main objective of control theory is to achieve a desired behavior in controlled variables, and it could be the input tracking or system stabilization [1].

When systems have several inputs and outputs are denominated Multiple Input-Multiple Output (MIMO) and systems with only one input and one output are called Single Input-Single Output (SISO) [2, 3].

Multivariable Control (MC) techniques are capable of MIMO system control, even if there are relationships among their variables. MC is important because many processes work the same way as MIMO systems do, and it is common to see that different enterprises use them, but many MIMO systems are controlled as though they were SISO when it is possible because of the complexity of MC, but this generates restrictions manipulating variables at the same time [4–7].

When MIMO systems are decoupled, there is no relationship among their variables. In this scenario, it is possible to separate them easily into several SISO systems and control them, using SISO control techniques. On the other hand, MIMO systems are coupled when there are relationships among their variables. Decoupling coupled MIMO systems imply the use of MC techniques, whose focus is on suppressing the relationship among variables [8–10].

Decoupling blocks or gains, are commonly used to get decoupled MIMO systems, but complex mathematical models are needed to obtain the decoupling blocks and this does not guarantee results, i.e., with the mathematical model it is not always possible to get a decoupled system [3, 11].

Genetic Programming (GP) is an alternative way based on artificial intelligence which could be used to obtain the decoupling blocks by an automated process. The GP is a revolutionary technique proposed by John Koza, this technique is able to develop computer programs by a stochastic form which is related to the genetic evolution [11, 12].

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_35, © Springer International Publishing Switzerland 2015

Getting linear and nonlinear models using GP does not need human work although the algorithm which obtains them, needs to be developed. In addition, this genetic program algorithm is able to test several cases until it finds the solution and suppress the complex analysis used in classical MC techniques to get the mathematical models [11].

In this paper an automatic generator of decoupling blocks for linear and no linear MIMO systems is developed by using GP and is tested and compared to a decoupling state feedback MC technique.

The processes of a paper transport machine and a coupled resistance-capacitor circuit that represents an irrigation process, were selected to test the automatic generator of decoupling blocks against the decoupling static state feedback technique. Paper transport machine was selected to test the generator athwart to a complex MIMO system and the resistance-capacitor circuit was used to test a simpler scenario.

Decoupling Static State Feedback Technique

The decoupling of MIMO systems using a static state feedback technique has the form in Fig. 1, this demands a mathematical model and its state space representation as $\Sigma(A, B, C)$, where systems are called square if they have the same number of inputs and outputs [8, 9].

System inputs can be known from the number of columns in matrix B and outputs number is associated to rows number in matrix C [8].

Decoupling static state feedback technique is regular if the number of inputs are minor or equal to outputs number and in any other case the technique used is non-regular case [13].

The decoupling case required in this paper is regular because system processes are square.

Regular Decoupling State Feedback

Regular static state feedback technique has the form described in Fig. 1, where the matrixes F and G must be obtained.

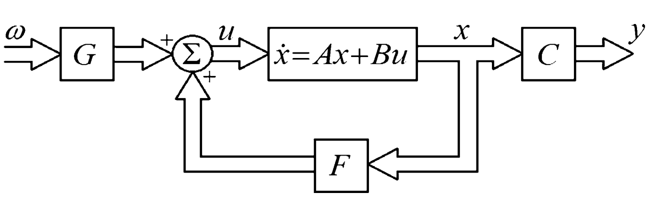

Fig. 1 Decoupling static feedback technique

The development of a regular decoupling static state feedback technique is complex and not always is possible.

Theorem 1. Regular decoupling static state feedback technique have solution if and only if the matrixes A and B are stabilizable and range of system is given as in Eq. (1) [8, 9].

$$rank\left(\begin{bmatrix} A & B \\ C & D \end{bmatrix}\right) = m + n \tag{1}$$

In addition to know if it is possible to suppress the relation among variables it is necessary verify infinite information of system, which is contained in the interactor matrix associated to the system transfer matrix.

Interactor matrix $\Phi(s)$, can be obtained with Hermite form of transfer matrix by columns with elemental operations inside biproper space denoted by the matrix $V(s)$, applying elemental operations the transfer matrix is taken to the form in Eq. (2) [8, 9].

$$T(s)V(s) = \begin{bmatrix} \Phi^{-1}(s) & 0 \end{bmatrix} \tag{2}$$

Theorem 2. If system is square and interactor matrix is diagonal, then system can be decoupled [14].

Once pass conditions in theorems 1 and 2 it is possible to develop a regular static state feedback technique that takes transfer matrix of system to the form in (3)

$$T_{FG}(s) = C(sI - A - BF)^{-1}BG \tag{3}$$

Regular decoupling state feedback technique has F and G matrixes given as shown in Eqs. (4) and (5) [8, 9].

$$F = -(B^*)^{-1}A^{**} \tag{4}$$

$$G = (B^*)^{-1} \tag{5}$$

Matrixes A^{**} and $B*$ are can be obtained using Eqs. (6) and (7) [8, 9]

$$A^{**} = \begin{bmatrix} C_1 & A^{d_1+1} \\ C_2 & A^{d_2+1} \\ C_m & A^{d_m+1} \end{bmatrix} + \begin{bmatrix} \sigma_1 & C_1 \\ \sigma_2 & C_2 \\ \sigma_m & C_m \end{bmatrix} \tag{6}$$

$$B^* = \begin{bmatrix} C_1 & A^{d_1} & B \\ C_2 & A^{d_2} & B \\ C_m & A^{d_m} & B \end{bmatrix} \tag{7}$$

The values $d_1, \ldots, d_m$ can be known with Eq. (8), $C_1, \ldots, C_m$ are the m row in C and $\sigma_1, \ldots, \sigma_m$ are the places where poles will be located after decouple system [9].

$$d_m = \min\{j \in N | C_j A^{d_j} B\} \neq 0 \quad (8)$$

The decoupled transfer matrix using poles movement will be like in Eq. (9)

$$T_{FG}(s) = \begin{bmatrix} \frac{1}{s+\sigma_1} & \cdots & 0 \\ \vdots & \ddots & \vdots \\ 0 & \cdots & \frac{1}{s+\sigma_m} \end{bmatrix} \quad (9)$$

Regular Decoupling State Feedback with Stability

When regular decoupling state feedback cannot be used because of the order and the form functions in transfer matrix, system must be decoupled by using a regular decoupling state feedback with stability.

Decoupling using state feedback with stability, it is necessary to verify if system is asymptotically stable using Smith-McMillan form, because this decoupling technique only preserves system stability [15].

Infinite information must be analyzed with stable interactor matrix $\Phi_s(s)$, wich is obtained using elemental operations in the space of biproper and bistable matrixes denoted by the matrix $V_s(s)$, then Hermite form by columns can take transfer matrix to description in Eq. (10) [15].

$$T(s)V_s(s) = \left[\Phi_s^{-1}(s) \quad 0\right] \quad (10)$$

Theorem 3. If system is asymptotically stable and the stable interactor matrix has diagonal form, then system can be decoupled using regular decoupling state feedback with stability.

If theorem 3 is covered, then the decoupling matrixes F and G of decoupling state feedback with stability will be calculated with Eqs. (11) and (12) [15].

$$G = X^{-1} \quad (11)$$

$$F = -X^{-1}Y \quad (12)$$

The X and Y matrixes are calculated obtaining the solution of diophantine equation in (13) [15].

$$XD_R(s) + YN_R(s) = Q(s) \quad (13)$$

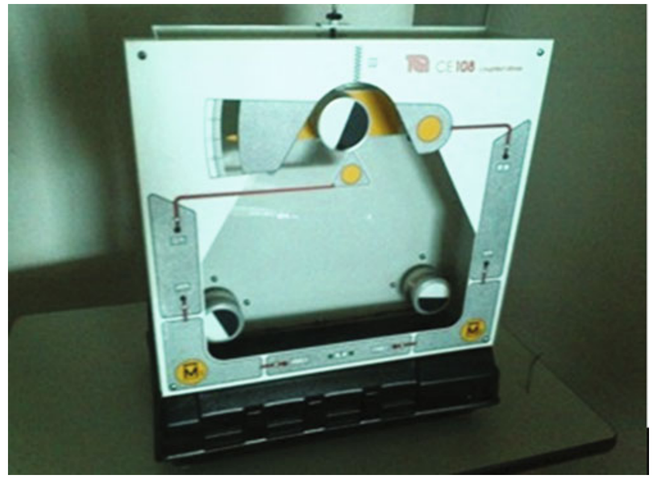

Fig. 2 CE108 device of TecQuipment

Matrixes D_R and N_R are the right coprime matrix fraction description of system $\Sigma(A, B, I_n)$ and $Q(s)$ must be a polynomial matrix that can be known with Eq. (14) [15].

$$Q(s) = V_s^{-1}(s)D_R(s) \quad (14)$$

Decoupling Paper Transport Process

Paper industry involves MC on paper production because paper is transported in a continuous production line, which required the control of speed and tension in paper, while is transported by several cylinders linked to motors that produce movement during the entire paper production operation [7, 16, 17].

The main problem controlling speed and tension during paper production process, is that paper transport is a MIMO coupled system because there is a relation between speed and tension of paper and while the speed of paper transport is changed, tension must be controlled to avoid the paper rupture [7, 16].

A representative plant of a paper machine is the coupled transmission device CE108 develop by TecQuipment, this apparatus has a belt that represents paper and use three pulleys, two are linked to motors that produce the movement in paper transport and a third upper pulley allow measurement of speed and tension in belt (Fig. 2) [17, 18].

The use of classical MC techniques for decoupling paper machine process require a mathematical model, which is obtained simplifying the CE108 device using its free body diagram [18].

State space model of apparatus CE108 with a representation $\Sigma(A, B, C, D)$ can be known with the state equations obtained from the mathematical model, and transfer matrix is obtained substituting in $\Sigma(A, B, C, D)$ the constant values given by the manual of the apparatus, and using equation (15) [18].

$$T(s) = C(SI - A)^{-1}B + D \tag{15}$$

$$T(s) = \begin{bmatrix} T_{11}(s) & T_{12}(s) \\ T_{21}(s) & T_{22}(s) \end{bmatrix} \tag{16}$$

With the elements of $T(s)$ given as:

$$T_{11}(s) = \frac{-3419s - 38460}{100s^5 + 2338s^4 + 87090s^3 + 1638000s^2 + 9374000s + 2758000} \tag{17}$$

$$T_{12}(s) = -T_{11}(s) \tag{18}$$

$$T_{21}(s) = \frac{3125s^4 + 37920s^3 + 2295000s^2 + 25360000s + 7660000}{50s^5 + 1169s^4 + 43545s^3 + 819000s^2 + 4687000s + 1379000} \tag{19}$$

$$T_{22}(s) = T_{21}(s) \tag{20}$$

Using transfer matrix in (16) system interactor can be obtained using Eq. (2) which is given in (21), and its components $\Phi_{11}(s)$ and $\Phi_{22}(s)$ are shown in Eqs. (22) and (23)

$$\Phi(s) = \begin{bmatrix} \Phi_{11}(s) & 0 \\ 0 & \Phi_{22}(s) \end{bmatrix} \tag{21}$$

$$\Phi_{11}(s) = \frac{100s^5 + 2338s^4 + 87090s^3 + 1638000s^2 + 9374000s + 2758000}{3419s + 38460} \tag{22}$$

$$\Phi_{22}(s) = \frac{50s^5 + 1169s^4 + 43545s^3 + 819000s^2 + 4687000s + 1379000}{6250s^4 + 75840s^3 + 4590000s^2 + 50720000s + 15320000} \tag{23}$$

Using Eq. (1) system rank is obtained in (24), which result means that theorem 1 is not fulfilled, although interactor matrix in (21) and can be decoupled according to theorem 2.

$$7 \neq m + n \tag{24}$$

Regular decoupling state feedback with stability, is another way for decoupling speed and tension in paper production process.

Decoupling with asymptotic stability is based on achieve of theorem 3 which is checked by getting Smith-McMillan form and from this verify that finite poles in (25) are in the negative part of real s plane.

$$\begin{aligned} p_1 &= -0.4395 + 26.8514i \\ p_2 &= -0.4395 - 26.8514i \\ p_3 = p_8 &= -11.3522 \\ p_4 = p_9 &= -10.8380 \\ p_5 = p_{10} &= -0.3108 \\ p_6 &= -0.4395 + 26.8514i \\ p_7 &= -0.4395 - 26.8514i \end{aligned} \tag{25}$$

Stable interactor matrix in (26) must be known in order to analyze infinite information.

$$\Phi_s(s) = \begin{bmatrix} \dfrac{1}{(s+5)^4} & 0 \\ 0 & \dfrac{1}{s+5} \end{bmatrix} \tag{26}$$

The theorem 3 is fulfilled since stable interactor has diagonal form and poles are in the left semi-plane s, then system can be decoupled with stability.

Matrix $Q(s)$ is known by using (14) and has diagonal form like is shown in (27).

$$Q(s) = \begin{bmatrix} q_{11}(s) & 0 \\ 0 & q_{22}(s) \end{bmatrix} \tag{27}$$

The calculus of feedback matrixes F and G can be obtained with (13) and should be found by a computational method using polynomial toolbox in Matlab™, but due to the great difference between small values and greater in the polynomials coefficients in $Q(s)$, $N_R(s)$ and $D_R(s)$ the solution cant not be acquired, although theorem 3 is fulfilled and $Q(s)$ is a polynomial matrix.

Decoupling of an Irrigation Plant Process

The irrigation process (Fig. 3) involves the use of water pumps to take water from wells to the irrigated fields and valves are used to control water flow while irrigating. It is possible to use an analog resistor–capacitor circuit that represents the irrigation plant, where power supplies in circuit imitate water pumps, resistors act as water valves and capacitors reproduce irrigated fields.

State space model of irrigation process can be known with the circuit state equations, which are obtained using nodal analysis.

Substituting capacitors value with 0.001 mF, resistors with 100 Ω and assigning inputs V_1 and V_2 in the state space equations, the system transfer matrix in (28) can be obtained with Eq. (15)

$$T(s) = \begin{bmatrix} \frac{10(s+20)}{s^2+40s+300} & \frac{100}{s^2+40s+300} \\ \frac{100}{s^2+40s+300} & \frac{10(s+20)}{s^2+40s+300} \end{bmatrix} \tag{28}$$

Interactor matrix in (29) can be known using transfer matrix system and Eq. (2)

$$\Phi(s) = \begin{bmatrix} \frac{10}{s+20} & 0 \\ 0 & \frac{5(s+10)+5(s+30)}{(s+10)(s+30)} \end{bmatrix} \tag{29}$$

Using Eq. (1) the system rank is given by (30)

$$4 = m + n \tag{30}$$

Since rank is equal to the sum of inputs and outputs and system is stabilizable, then theorem 1 is accomplish

The system poles in (31) can be known with Smith–McMillan form.

$$\begin{aligned} p_1 = p_3 = -10 \\ p_2 = p_4 = -30 \end{aligned} \tag{31}$$

Analyzing interactor matrix and system poles can be conclude that theorem 2 is achieve.

Decoupling matrixes F and G using regular decoupling state feedback technique were obtained like is shown in (32) and (33).

$$F = \begin{bmatrix} \frac{9}{10} & -1 \\ -1 & \frac{7}{20} \end{bmatrix} \tag{32}$$

$$G = \begin{bmatrix} \frac{11}{100} & 0 \\ 0 & \frac{11}{100} \end{bmatrix} \tag{33}$$

Automatic Generator of Decoupling Blocks

GP generates individual programs to find a solution for different problems, in this case blocks for decoupling a coupled MIMO systems.

The population of the automatic generator of decoupling blocks is a set of individual programs developed following a tree base representation like is shown in Fig. 4 [19].

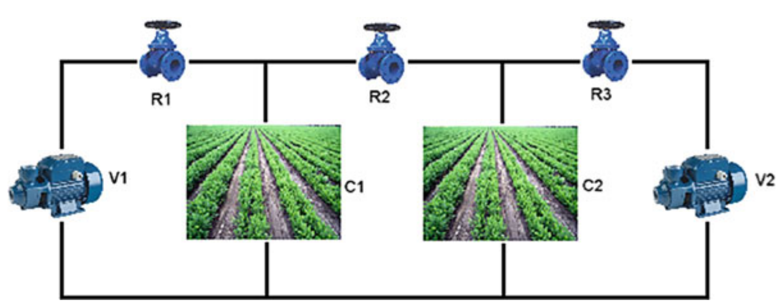

Fig. 3 Representative plat of an irrigation process

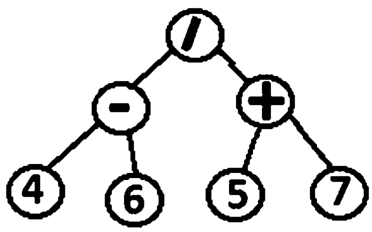

Fig. 4 Scheme of a genetic program element

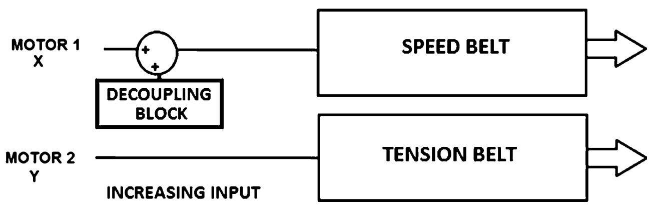

Fig. 5 Evaluation of aptitude function using decoupling blocks with GP

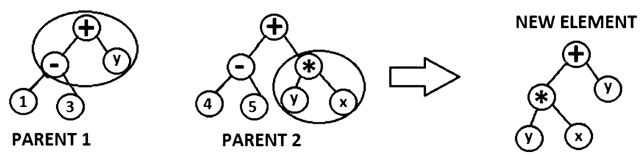

Fig. 6 GP reproduction process

Once population is generated individual aptitude most be evaluated using fitness function, in order to know which programs are better adapted to decouple the system [19].

Fitness function is evaluated by linking the tested block to all inputs, but not to the one increasing like in paper machine process in Fig. 5, then aptitude function sums the difference in system outputs that should not change while the unlinked input is increased, like in (34).

$$\sum_{a=0}^{10} |out_i(a) - out_i(a-1)| \tag{34}$$

After testing fitness in all individuals a random selection process called tournament select population sets where the most adapted individuals to solve the problem, inherit its genes to new individuals by a reproduction process like in Fig. 6 [9].

The first generation is obtained when all new individuals are put into the population group and ordered according to its aptitude value, then mutation and elitism are performed [19].

Mutation is an asexual reproduction that allows to include required genes that was not in the initial population. The process of mutation was programmed making a random selection of an individual in the population and mixing it by crossover operation with a new created individual [19].

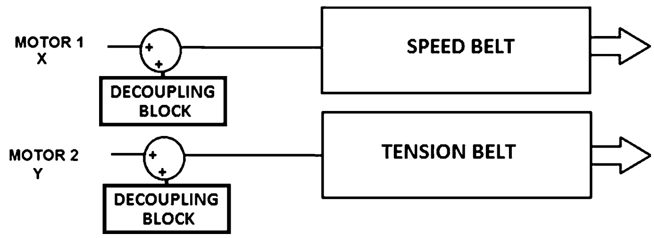

Fig. 7 Decoupled paper machine process

The elitism maintains only the better adapted individuals for its reproduction and is programmed to avoid crossover operation of the worst adapted individuals by deleting them, it is also present in tournaments where only better adapted individuals are selected for crossover operation [19].

After several generations, the decoupling block can be found and then the process must be repeated until find one decoupling block per input so that system can be decoupled, like in paper machine process in Fig. 7.

Results

Simulations and GP structures were made in MATAB™ 2010 and data acquisition device was NI-USB-6211 manufactured by National Instruments.

Paper Machine Production Process

The behavior of uncoupled system shows that the increase of band speed in Fig. 8 requires increase the input voltages in every motor (axes motor 1 and motor 2 in graphs), and the increment of tension in Fig. 9 demands the increase of difference between the voltages applied to motors (axes motor 1 and motor 2 in graphs), which imply a relationship between voltages in the input motors.

In Section "Decoupling Paper Transport Process" the decoupling feedback state could not get decoupled the paper production process, although the complex mathematical model and matrix forms were obtained, but better results were produced with GP.

GP algorithm was initialized with: 4000 generations, 15 individuals of population size, depth of 4, 4 individuals mutated per generation, 5 individuals per tournament and 14 individuals admitted according to its fitness value.

Automatic generator of blocks, produce the schema in Fig. 10 after 101 generations for motor 1 and 40 for motor 2, decoupling time depends of process settling time, in paper machine is an average of 6.4 s per individual tested.

System decoupled shows that speed band in Fig. 11 can be controlled only by using motor 1 in ranges where motor 2 is minor than 3 V i.e. the tension can be modified in ranges from 0 to 3 V so that system remains decoupled.

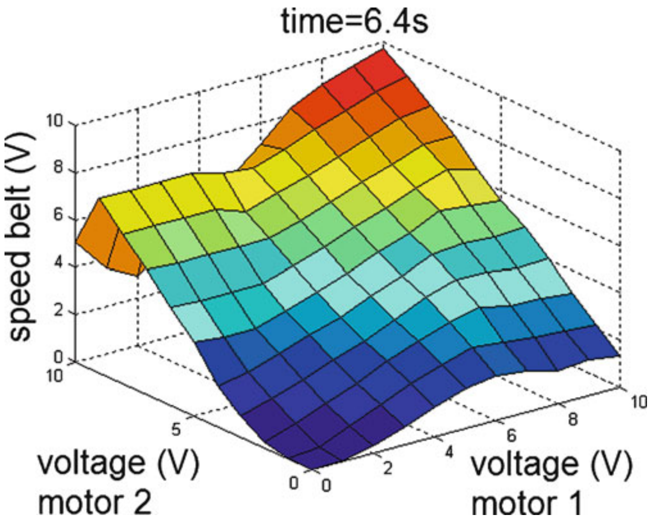

Fig. 8 Speed belt response at 6.4 s in coupled system

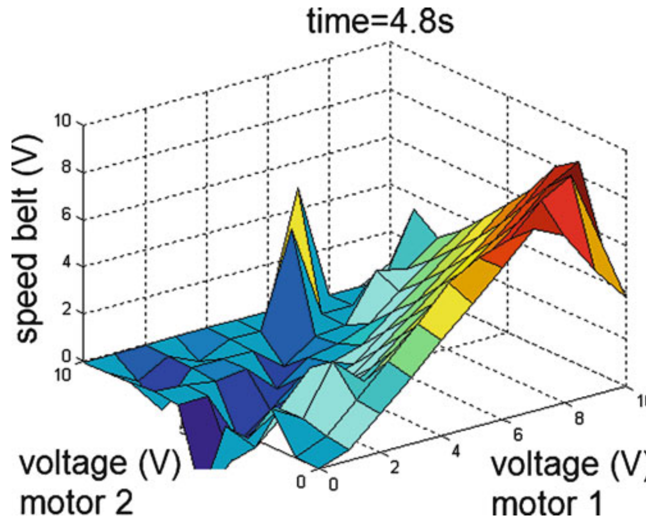

Fig. 11 Speed belt response at 4.8 s in decoupled system

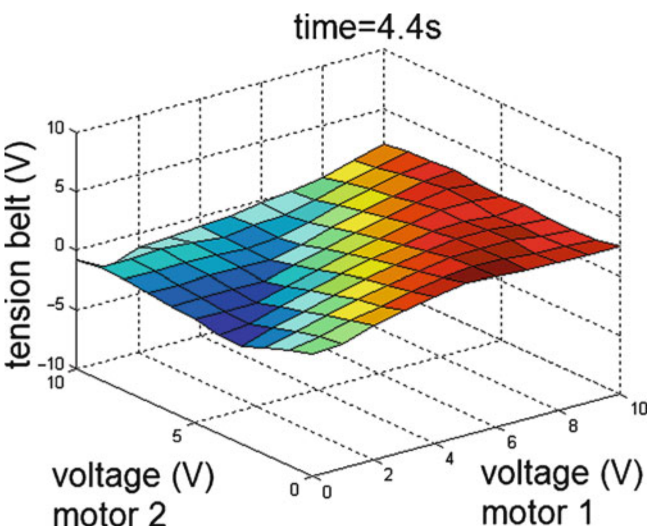

Fig. 9 Tension belt response at 4.4 s in coupled system

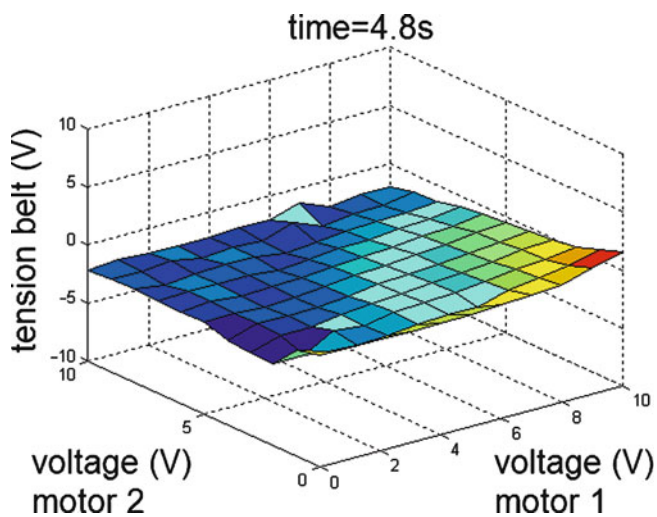

Fig. 12 Tension belt response at 4.8 s in decoupled system

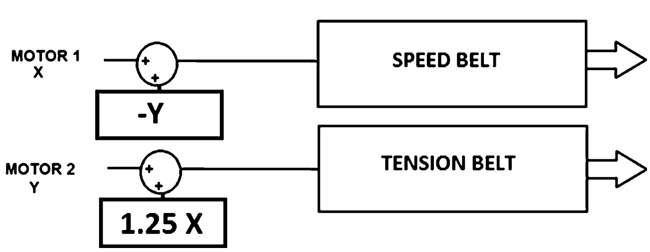

Fig. 10 Decoupling blocks implemented in paper production machine

Analyzing Fig. 12 it is shown that tension belt is decoupled in every range with a low contribution of voltage in motor 1 when voltage in motor 1 is higher than 4 V.

Automatic generator achieve a successful result, despite the relationships remained on the decoupled system for certain ranges, this is because commonly is required the control of paper speed maintaining tension near zero in the paper production process.

Irrigation of Fields Process

The circuit response was originally coupled moving all capacitors voltage when any input voltage is moved as shown in Figs. 13 and 14 (axes of capacitor 1 and capacitor 2 voltages). Decoupled results in Figs. 15 and 16 are obtained using matrixes F and G of decoupling feedback state technique, this shows that capacitor voltage 1 control is made by changing voltage input 1 and capacitor 2 with voltage input 2, but remains an small relation between input voltage 1 and input voltage 2.

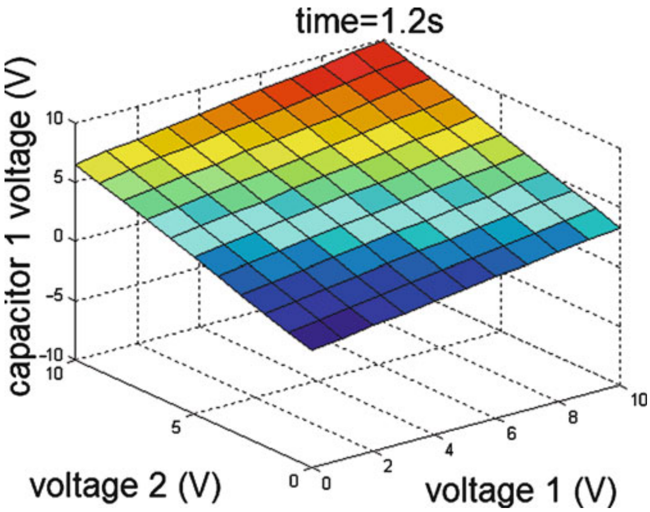

Fig. 13 Voltage capacitor 1 at 1.2 s in coupled process

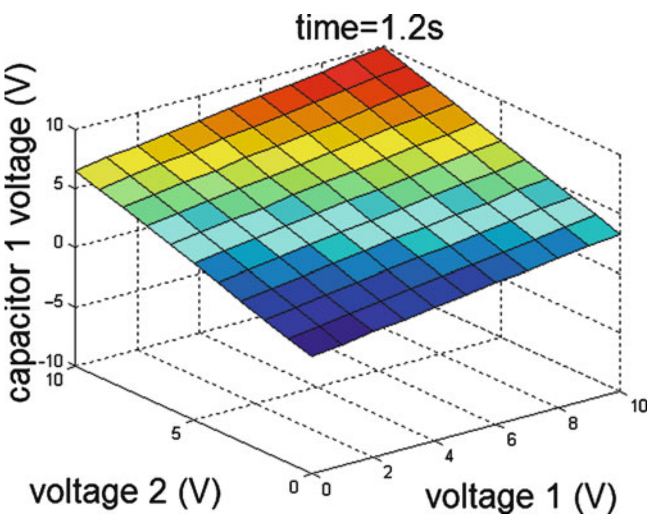

Fig. 14 Voltage capacitor 2 at 1.2 s in coupled process

GP algorithm for irrigation process was initialized with: 4000 generations, 15 individuals of population size, depth of 4, 4 individuals mutated per generation, 5 individuals per tournament and 14 individuals admitted according to its fitness value.

The automatic generator of decoupling blocks use the scheme of blocks in Fig. 17, obtained after 49 generations per block, with an average time of 1.2 s per individual tested, this implies a better performance in the automatic generator of blocks because of the easier way solving the decoupling problem without need a mathematical model. Decoupling blocks obtained with GP also produce better results since there is no relation between voltage in capacitor 1 and capacitor 2 after link the decoupling blocks, like is shown in Figs. 18 and 19, where the control of each capacitor only

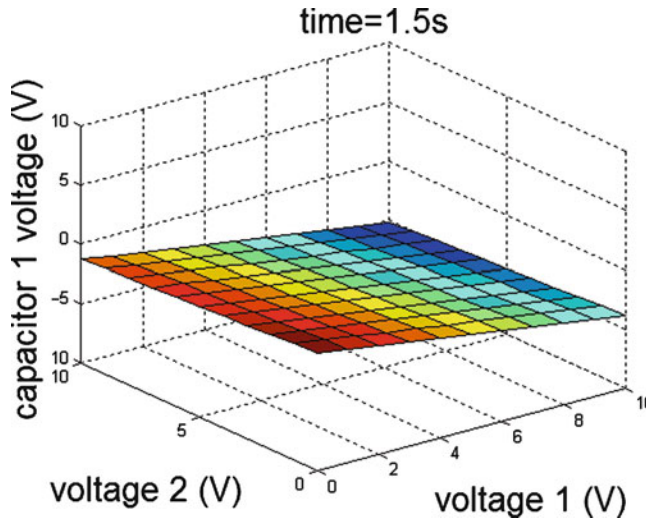

Fig. 15 Voltage capacitor 1 at 1.5 s in decoupled process with feedback state technique

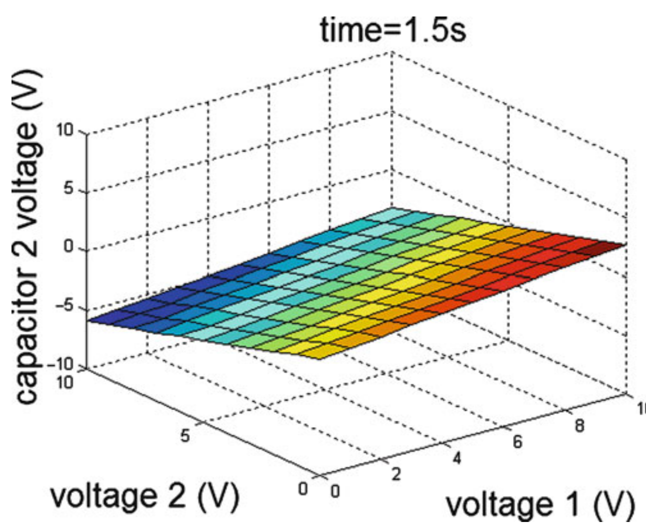

Fig. 16 Voltage capacitor 2 at 1.5 s in decoupled process with feedback state technique

requires one input voltage and the change of the other input voltage does not affect the capacitor.

The calculus of a numerical decoupling error is important to compare the decoupled results and it is calculated by increasing an input and checking the difference produced in the output that should not change in a decoupled system, this is made by using Eq. (35)

$$\sum_{b=0}^{n}\sum_{a=0}^{n}|salida_i(0,b) - salida_i(a,b)| \qquad (35)$$

Decoupling error of paper transport process is obtained with (35) and it is shown in Table 1 for speed band and in

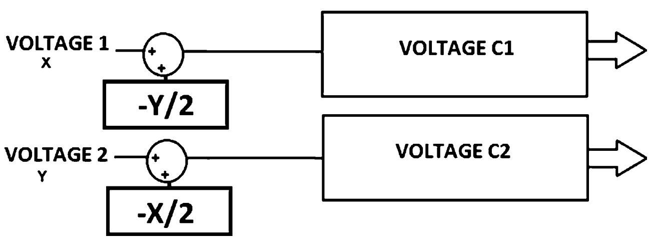

Fig. 17 Decoupled schematic of coupled circuit

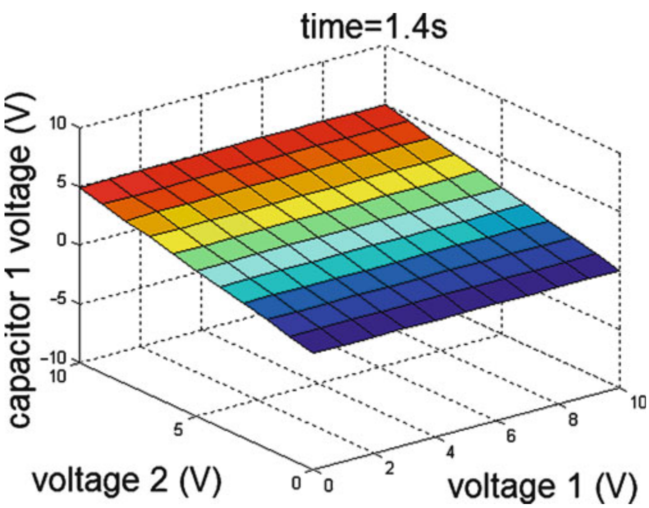

Fig. 18 Voltage capacitor 1 at 1.4 s in decoupled process with GP

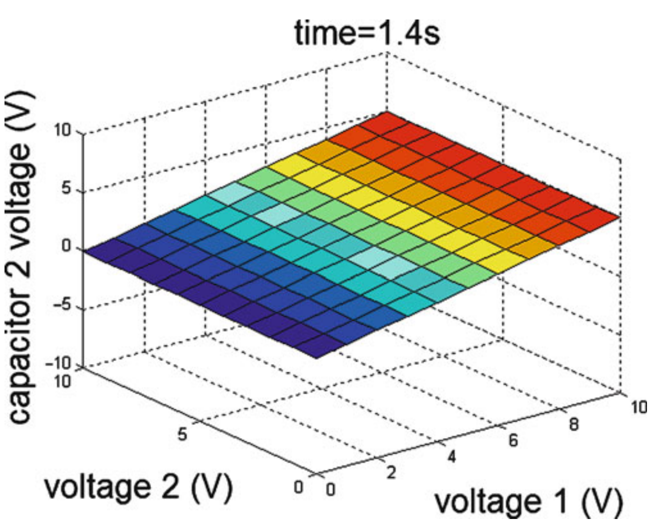

Fig. 19 Voltage capacitor 2 at 1.4 s in decoupled process with GP

Table 2 for tension band. This results show that the speed in band will be changed using two of the inputs (Table 1), but tension is better decoupled since the relation in Motor 1 have been reduced (Table 2), this makes possible to maintain tension band in zero while increasing the speed in paper.

Table 1 Decoupling error in the speed band in paper transport process decoupled with GP

Input	Decoupling error in coupled system (V)	Decoupling error in decoupled system with GP (V)
Motor 1	169.2099	308.8669
Motor 2	401.3224	317.0679

Table 2 Decoupling error in the tension band in paper transport process decoupled with GP

Input	Decoupling error in coupled system (V)	Decoupling error in decoupled system with GP (V)
Motor 1	150.8193	84.1998
Motor 2	198.5391	171.6962

Table 3 Decoupling error in the capacitor 1 in the irrigation of fields process decoupled with decoupling static state feedback technique

Input	Decoupling error in coupled system (V)	Decoupling error in decoupled system with GP (V)
Voltage 1	205.8395	236.8222
Voltage 2	402.4473	77.6248

Table 4 Decoupling error in the capacitor 2 in the irrigation of fields process decoupled with decoupling static state feedback technique.

Input	Decoupling error in coupled system (V)	Decoupling error in decoupled system with GP (V)
Voltage 1	407.0213	160.1663
Voltage 2	200.8564	359.9976

The decoupling error in the irrigation process using decoupling static state feedback technique was calculated with (35) and the results are shown in Table 3 for capacitor 1 and Table 4 for capacitor 2. This results shows that capacitor 1 and capacitor 2 can be separately controlled, but with relationships between their variables because there is a total variation of 77.6248 V for the capacitor 1 and 160.1663 V for the capacitor 2, when an input that should not produce change in the output is increased.

Decoupling error in the irrigation process using GP is obtained with (35), and the results are shown in Table 5 for capacitor 1 and Table 6 for capacitor 2. This results show that capacitor 1 can be controlled using voltage 2 and the change produced for the input voltage 1 is only of 3.6569, which is a very good result because voltage 1 is almost decoupled and the same occurs for capacitor 2 because can be controlled using voltage 1 and the change produced by voltage 2 is only of 1.7719, i.e., system is fully decoupled using GP.

Table 5 Decoupling error in the capacitor 1 in the irrigation of fields process decoupled with GP

Input	Decoupling error in coupled system	Decoupling error in decoupled system with GP
Voltage 1	205.8395	3.6569
Voltage 2	402.4473	301.1334

Table 6 Decoupling error in the capacitor 2 in the irrigation of fields process decoupled with GP

Input	Decoupling error in coupled system	Decoupling error in decoupled system with GP
Voltage 1	407.0213	304.1605
Voltage 2	200.8564	1.7719

Conclusions

This paper proposed an alternative technique for decoupling coupled MIMO systems using GP without need a mathematical model, this technique is tested in a paper transport process and an irrigation of fields process.

The obtained results are very pleasant since we found decoupling blocks that reduce or suppress the relationships in the tested coupled MIMO systems by using GP.

Paper transport process was partially decoupled with our proposed generator of decoupling blocks using GP, but with successful results for paper transport neediness, while decoupling static state feedback technique could not get the system decoupled. The irrigation of fields process were fully decoupled with our proposed technique using GP, while the decoupling static state feedback technique only partially decouples the system.

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MEMS Dual Axis Accelerometer with H-T Shape Structure

Xingguo Xiong, Lidong Qiang, Linfeng Zhang, and Junling Hu

Abstract

MEMS accelerometers have been used in air bag deployment system and other applications. Most MEMS accelerometer devices are sensitive to acceleration input along one direction. In this paper, a dual-axis MEMS accelerometer which can simultaneously measure acceleration inputs along X and Y directions is proposed. The proposed MEMS accelerometer utilizes two sets of folded beams perpendicular to each other to sense acceleration inputs along X and Y directions within the device plane. The working principle of the dual-axis MEMS accelerometer is analyzed. A theoretical model is developed to predict the performance of the accelerometer. A set of example design parameters of the device is obtained based on the analysis. ANSYS simulation is used to verify the function of MEMS accelerometer. The fabrication flow of the MEMS accelerometer is suggested. Combined with a Z-axis accelerometer, the proposed dual-axis MEMS accelerometer can be used for acceleration measurement along complete X, Y and Z directions for 3D inertial navigation system.

Keywords

Microelectromechanical systems (MEMS) • Accelerometer • Dual-axis • ANSYS • Differential capacitance sensing

Introduction

Microelectromechanical Systems (MEMS) technology integrates both electrical circuits and mechanical components in the size range of microns into a single chip using microfabrication technique. A typical MEMS chip includes microsensors, microactuators and microelectronic circuits and together they can implement more powerful function than a single component. Various MEMS products have been developed for different applications [1]. Compared to traditional mechanical counterpart, MEMS devices have the advantages of smaller size, low cost, low weight, low energy consumption, high resolution and multi-functionality. MEMS inertial sensors [2] are important category of MEMS devices. MEMS inertial sensors include MEMS accelerometers (used to measure linear accelerations) and MEMS gyroscopes (used to measure angular velocities), etc. Together they can offer the kinetic information needed for inertial navigation system. They are widely used for automobile, sailing and aerospace applications.

Various MEMS accelerometer designs based on different structure and working principles have been developed. Most MEMS accelerometers are sensitive to acceleration input along one direction. In this way, it can read acceleration along its sensitive axis. However, for a complete 3D inertial navigation system, the acceleration values along all X, Y and Z axes are required. Thus there is a strong demand for dual-axis or even 3-axis MEMS accelerometers. Dual-axis MEMS accelerometer detects acceleration inputs along two

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_36, © Springer International Publishing Switzerland 2015

orthogonal directions. A dual-axis accelerometer can be achieved by simply putting two single-axis accelerometers together, with the sensitive axes of both accelerometers aligned to be perpendicular to each other. However, such duplication of single-axis MEMS accelerometers is not cost effective. To be more compact and cost-effective, a single MEMS accelerometer which can simultaneously measure acceleration inputs along both axes is needed. Several MEMS dual-axis accelerometer devices have been reported [3–7]. In [3], a capacitive bulk-micromachined dual-axis accelerometer is reported. The device has a single inertial mass symmetrically suspended by four pairs of folded elastic beams. Movable fingers extrude from central mass and constitute differential capacitance with left and right fixed fingers. Under acceleration input, the movable mass experiences inertial force which leads to the displacement of movable fingers along X and/or Y directions. By sensing the differential capacitance change, the value of the input acceleration can be derived. The folded beams are flexible along both X and Y directions. Thus the response of the device for accelerations along X and Y directions may be mixed together. In order to separate the mixed signals, frequency-division method is used for signal-sensing circuit design. The proposed design has empty area in four corners, hence it is not very area efficient. In [4], a 1-g dual-axis linear accelerometer using standard 0.5 μm CMOS technology for high-sensitivity applications is proposed. This design also utilized double folded-beam and single mass structure. It designs extra comb fingers to the four corners, hence maximizing the utilization of device area. The overall device features a full scale of 1 g acceleration range with sensitivity of 200 μg. Switched-capacitor low power filter is used in signal sensing circuit. In [5], an octagon-shape dual-axis MEMS accelerometer is proposed. The single central movable mass was supported by flexible springs located at the four corners of the device. Differential capacitance was used for sensing the displacement of movable beams due to inertial force. In [7], a monolithic 3-axis thermal convective accelerometer is reported. Compared to capacitive sensing, thermal sensing consumes more power and the response time is relatively slow.

In this paper, a bulk-micromachined H-T shape dual-axis capacitive MEMS comb accelerometer device was proposed. The device has two separate movable masses to sense acceleration along X and Y direction respectively. The top portion consists of four vertical folded beams, a movable central mass and top movable/fixed comb finger groups. It is sensitive to acceleration input along X direction. The bottom portion consists of two horizontal folded beams, a central movable mass and horizontal movable/fixed comb finger groups in its left and right sides. It is sensitive to acceleration along Y direction. The vertical folded beams and top mass designed to sense acceleration along X direction look like "H" shape, while the horizontal folded beams and bottom mass designed to sense acceleration along Y direction look like "T" shape. The T-shape portion is embedded inside the H-shape portion, resulting in a very compact rectangular design. The overall structure is very simple. Due to separate beam and mass design, the coupling effect of output response between X and Y axis is minimized. Differential capacitive sensing is used to sense the displacement of movable masses in response to acceleration inputs. The proposed dual-axis MEMS accelerometer can be used for inertial navigation systems and other applications.

Device Design and Analysis

Structural Design of H-T Shape Dual-Axis MEMS Accelerometer

The structure diagram of the H-T shape dual-axis MEMS accelerometer is shown in Fig. 1. As seen in Fig. 1, the top portion of the device consists of four vertical folded beams, a top central mass and 16 vertical movable/fixed finger groups. It looks like "H" shape, and is used to sense acceleration input along X direction. The bottom portion include two horizontal folded beams, a vertical central mass, and 16 horizontal movable/fixed comb finger groups (8 groups in left side and 8 groups in right side). It looks like "T" shape and is sensitive to acceleration input along Y direction. The T-shape portion is embedded inside H-shape portion, which makes it a very compact design.

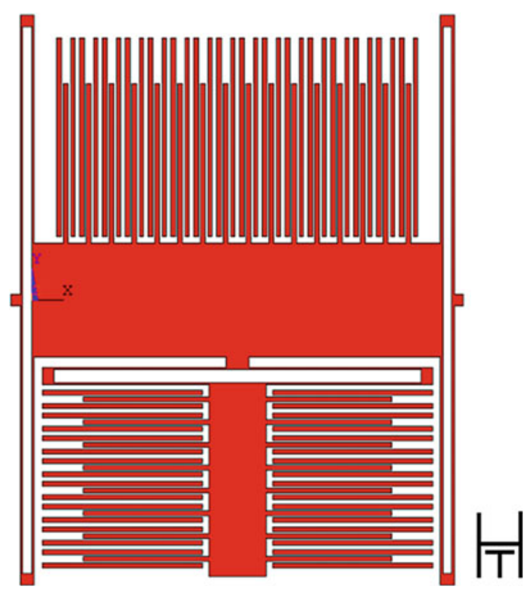

Fig. 1 Structure diagram of H-T shape dual-axis MEMS accelerometer

The working principle of the H-T shape dual-axis MEMS accelerometer is explained as below. The top vertical movable fingers form differential capacitances C_1 and C_2 with their left and right vertical fixed fingers. The bottom horizontal movable fingers also form differential capacitance C_3 and C_4 with their top and bottom fixed fingers. When there is no acceleration input, all the movable fingers stay in the middle between left/right or top/bottom fixed fingers. As a result, $C_1 = C_2$ and $C_3 = C_4$. The differential capacitance output of H-portion is:

$$C_H = C_1 - C_2 = 0 \tag{1}$$

The differential capacitance output of T-portion is:

$$C_T = C_3 - C_4 = 0 \tag{2}$$

However, when there is acceleration input a_X along X direction, the top and bottom central masses experience inertial force along –X direction. As a result, the four vertical folded-beams bend along –X direction. Hence the differential capacitance gaps change, and the differential capacitance changes too. The more acceleration input, the more resulted displacement, hence the more differential capacitance change. By measuring the differential capacitance change $C_H = C_1$-C, the input acceleration along X direction (a_X) can be derived.

Similarly, when there is acceleration input a_Y along Y direction, the bottom central mass experiences inertial force along −Y direction. As a result, the two horizontal folded-beams bend along −Y direction. Hence the differential capacitance gaps change, and the differential capacitance changes too. By measuring the differential capacitance change $C_T = C_3 - C_4$, the value of acceleration input a_Y can be measured.

The vertical folded beams of H portion and the horizontal folded beams of T portion are perpendicular to each other. This ensures that acceleration input along X direction only results in the bending of vertical beams, and acceleration input along Y direction only results in the bending of horizontal beams. This can minimize the cross-coupling between the device response in X and Y directions.

Analysis of H-T Shape Dual-Axis MEMS Accelerometer

The H-T shape dual-axis MEMS accelerometer utilizes differential capacitance sensing to sense the displacement of movable mass due to inertial force. As shown in Fig. 2, each movable finger constitutes differential capacitance c_1 and c_2 with its left/right fixed fingers. Assume there are N finger groups in the device. As shown in Fig. 2a, when there is no acceleration input, movable finger stays in the middle of capacitance gap, hence left/right device capacitance:

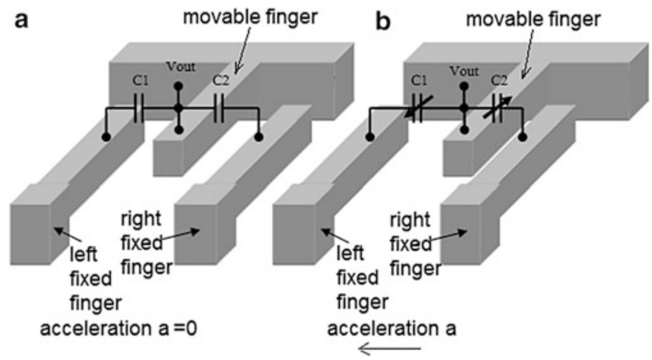

Fig. 2 Differential capacitance sensing of H-T shape dual-axis MEMS accelerometer. (**a**) Without acceleration input, (**b**) with applied acceleration

$$C_1 = nC_1 = C_2 = nC_2 = C_0 = \left(\frac{N\varepsilon S}{d_0}\right) = \left(\frac{N\varepsilon L_{ov}t}{d_0}\right) \tag{3}$$

where ε is the dielectric constant of air, S is the overlap area between movable and fixed fingers ($S = L_{ov} \cdot t$), L_{ov} is the overlap length between movable and fixed fingers, and t is the device thickness. Thus differential capacitance output when there is no acceleration input is

$$\Delta C = C_1 - C_2 = 0 \tag{4}$$

When there is acceleration input along X direction, the capacitive accelerometers convert the acceleration into a differential capacitance change. As shown in Fig. 2b, if there is acceleration along −X direction, the movable mass experiences inertial force and move toward +X direction with displacement of x. Thus left capacitance decreases and right capacitance increases.

$$C_1' = \left(\frac{N\varepsilon L_{ov}t}{d_0 + x}\right) \approx \left(\frac{N\varepsilon L_{ov}t}{d_0}\right) \cdot \left(1 - \frac{x}{d_0}\right) = C_0 + \Delta C \tag{5}$$

$$C_2' = \left(\frac{N\varepsilon L_{ov}t}{d_0 - x}\right) \approx \left(\frac{N\varepsilon L_{ov}t}{d_0}\right) \cdot \left(1 + \frac{x}{d_0}\right) = C_0 - \Delta C \tag{6}$$

Thus the differential capacitance output becomes

$$\Delta C = C_1' - C_2' = 2\Delta C = 2\left(\frac{N\varepsilon L_{ov}t}{d_0}\right) \cdot \left(\frac{x}{d_0}\right) \tag{7}$$

We can see that differential capacitance output ΔC is directly proportional to displacement x of movable fingers, which is directly proportional to acceleration input:

$$x = \left(\frac{F_{inertial}}{k_{tot}}\right) = \left(\frac{M \cdot a}{K_{tot}}\right) \tag{8}$$

where $F_{inertial}$ is the inertial force due to input acceleration:

$$F_{inertial} = \text{-}M \cdot a \quad (9)$$

In it, M is the mass of movable mass and movable fingers, and a is input acceleration, K_{tot} is the effective spring constant of the device.

Hence by measuring the differential capacitance change ΔC by a high-resolution differential capacitance sensing circuit [8], we can know the value of input acceleration. This is the working principle of capacitive accelerometer device.

Device Design and Optimization

Assume the width, length and thickness of the H-folded beams (folded beams of H-portion) are W_{bH}, L_{bH} and t_b separately, and the width, length and thickness of T-folded beams (folded beams of T-portion) are W_{bT}, L_{bT} and t_{bT} separately. The width, length and thickness of H-movable finger are W_{fH}, L_{fH} and t_f. The width, length and thickness of T-movable finger are W_{fT}, L_{fT} and t_f. The width, length and thickness of H-shape mass are W_{mH}, L_{mH} and t_m. The width, length and thickness of T-shape mass are W_{mT}, L_{mT} and t_m. For the device, $t_b = t_f = t_m = t$ (device thickness). Young's modulus of Si material is E.

For the H-shape portion, each folded beam consists of two sections connected in series. Each beam section can be treated as a double-clamped beam model with spring constant of [9]

$$K_{bH} = \frac{EW_{bH}^3 t_b}{L_{bH}^3} \quad (10)$$

Two beam sections are connected in series, thus the spring constant of one folded beam of H-portion is

$$K_{bfH} = \frac{K_{bH}}{2} = \frac{EW_{bH}^3 t_b}{2L_{bH}^3} \quad (11)$$

Four folded beams are connected in parallel, thus the total spring constant of H-portion is

$$K_H = 4K_{bfH} = 4 \times \frac{K_{bH}}{2} = \frac{2EW_{bH}^3 t_b}{L_{bH}^3} \quad (12)$$

The movable mass of H-shape portion is

$$M_H = \rho W_{mH} L_{mH} t_m \quad (13)$$

where ρ is density of silicon, $\rho = 2.33 \times 10^3$ kg/m^3. Thus the displacement sensitivity of H-shape portion to input acceleration along X direction is (note: $t_b = t_m$)

$$S_{dx} = \frac{M_H g}{K_H} = \frac{\rho W_{mH} L_{mH} t_m g \cdot L_{bH}^3}{2EW_{bH}^3 t_b} = \frac{\rho W_{mH} L_{mH} g L_{bH}^3}{2EW_{bH}^3} \quad (14)$$

Similarly, we can find out the displacement sensitivity of T-shape portion to input acceleration along Y direction. For the T-shape portion, two folded beams are connected in parallel and each folded beam consists of two sections connected in series. Each beam section can be treated as a double-clamped beam model with spring constant of [9]

$$K_{bT} = \frac{EW_{bT}^3 t_b}{L_{bT}^3} \quad (15)$$

Two beam sections are connected in series, thus the spring constant of one folded beam of T-portion is

$$K_{bfT} = \frac{K_{bT}}{2} = \frac{EW_{bT}^3 t_b}{2L_{bT}^3} \quad (16)$$

Two folded-beams are connected in parallel, thus the total spring constant of T-portion is

$$K_T = 2K_{bfT} = 2 \times \frac{K_{bT}}{2} = \frac{EW_{bT}^3 t_b}{L_{bT}^3} \quad (17)$$

The movable mass of T-shape portion is

$$M_T = \rho W_{mT} L_{mT} t_m \quad (18)$$

Thus the displacement sensitivity of T-shape portion to input acceleration along Y direction is (note: $t_b = t_m$)

$$S_{dy} = \frac{M_T g}{K_T} = \frac{\rho W_{mT} L_{mT} t_m g \cdot L_{bT}^3}{EW_{bT}^3 t_b} = \frac{\rho W_{mT} L_{mT} g L_{bT}^3}{2EW_{bT}^3} \quad (19)$$

The resonant frequency of H-portion of accelerometer is

$$f_H = \frac{1}{2\pi}\sqrt{\frac{K_H}{M_H}} \quad (20)$$

The resonant frequency of T-portion of accelerometer is

$$f_T = \frac{1}{2\pi}\sqrt{\frac{K_T}{M_T}} \quad (21)$$

Based on the equation, in order to improve the sensitivity of acceleration, the most effective way is to increase beam length or decrease beam width because the sensitivity is proportional to L_b^3 and reversely proportional to W_b^3. Figure 3 shows the relationship between spring constant (reversely proportional to device sensitivity) and corresponding beam length/width. We can see that spring

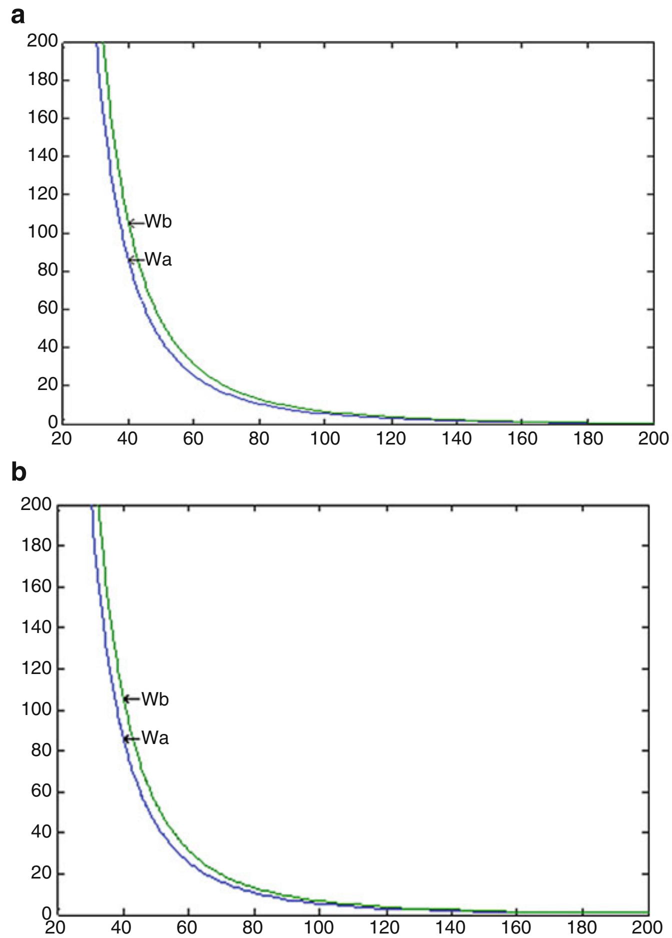

Fig. 3 Relationship between beam length/width and spring constant. (**a**) Spring constant (K_H, unit: N/m) versus H-folded beam length (L_{bH}, unit: μm) for different H-beam width (Wa = 3 μm, W_b = 4 μm). (**b**) Spring constant (K_T, unit: N/m) versus T-folded beam length (L_{bT}, unit: μm) for different T-beam width (Wa = 3 μm, W_b = 4 μm)

constant decreases with beam length, and increases with beam width. That is, we can increase device sensitivity by increasing beam length and decreasing beam width. However, increasing beam length leads to significant increase of device area. Thus the most efficient way to improve device sensitivity is to decrease beam width. On the other hand, if beam width is decreased to be less than 1 μm, the beams may be too fragile. In our device design, we set the beam width of both H- and T-portions to be 2 μm.

In device design, we need to find a set of optimized design parameters to meet the performance requirements. Main performance considerations for the evaluation of the MEMS accelerometers are sensitivity, resolution, nonlinearity, bias instability, and cross-axis sensitivity, etc. In order to increase the resolution of the accelerometer, its natural resonant frequency must be lowered, and the sense capacitance must be increased as much as possible. To achieve maximum resolution, spring constant of the accelerometer should be minimized, and the sensing mass should be maximized. When setting the length of H-portion and T-portion masses, we need to set them properly so that they provide enough space for totally 32 movable fingers (16 movable fingers on the top and other 16 movable fingers on both sides of T-portion mass). Base on above analysis, a set of optimized design parameters of the H-T shape comb accelerometer device are obtained, as listed in Table 1.

Table 1 The optimized design parameters of the MEMS accelerometer

Name	Amount	Length (μm)	Width (μm)
H-mass	1	360	100
T-mass	1	50	170
H-movable fingers	16	140	4
T-movable fingers	16	110	4
H-folded beams	4	250	2
T-folded beams	2	171	2
H-fixed fingers	32	174	4
T-fixed fingers	32	140	4
Anchors	2	10	10
H-T connector	1	20	11
H-folded beam connector	4	12	10
T-folded beam connector	2	10	15
Capacitance gap	2		
No. of finger groups (H-portion)	16		
No. of finger groups (T-portion)	16		

ANSYS Simulation and Discussion

In order to verify the performance of the designed H-T shape comb accelerometer, ANSYS simulation is used. Based on FEM (Finite Element Method), ANSYS gives accurate prediction about the device performance, such as resonant frequencies, stress distribution, sensitivity and capacitance, etc. ANSYS simulation results of the proposed H-T shape dual-axis MEMS comb accelerometer are listed as below.

Frequency Simulation

The first five vibration modes of this accelerometer are extracted during ANSYS resonant frequency simulation. ANSYS animation is used to identify the working modes of the MEMS accelerometer. Only the modes in which the H-portion mass moves left and right along X direction and the T-portion mass moves along Y direction are the working modes. They were found to be located as the first vibration mode and the third vibration mode. Other modes involves the twisting and rotation of the folded beams, hence apparently they are not the working modes of the device. The vibrational mode of H-portion (sensing acceleration along X direction) is shown in Fig. 4, with resonant frequency found to be $f_x = 6.6415$ kHz. The vibrational mode of T-portion (sensing acceleration along Y direction) is shown in Fig. 5, with resonant frequency found to be $f_y = 17.145$ kHz. This verifies the designed dual-axis MEMS accelerometer does sense acceleration inputs along X and Y directions. Furthermore, we can see that the resonant frequencies for both working modes are separately far away from each other. This ensures the vibration of X and Y directions in working mode will not be easily coupled into each other, which is exactly what we want. The folded beams for H and T portions are designed to be perpendicular to each other. This helps avoid unwanted signal coupling between X and Y directions during working mode.

Sensitivity Simulation

In order to simulate the displacement sensitivity of the H-T shape dual-axis MEMS accelerometer, unit gravity acceleration ($1\ \text{g} = 9.8\ \text{m/s}^2$) is applied as constant load to the device along X and Y directions separately. The simulation results are shown in Figs. 6 and 7 respectively. Negative value of displacement indicates the displacement of the

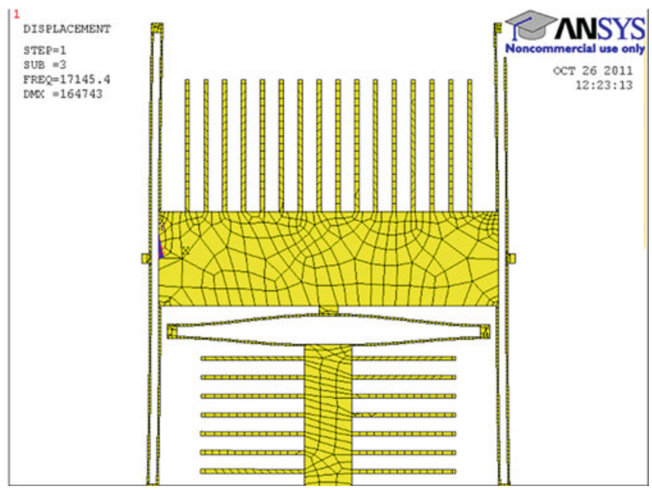

Fig. 5 ANSYS simulation of vibration mode along Y direction

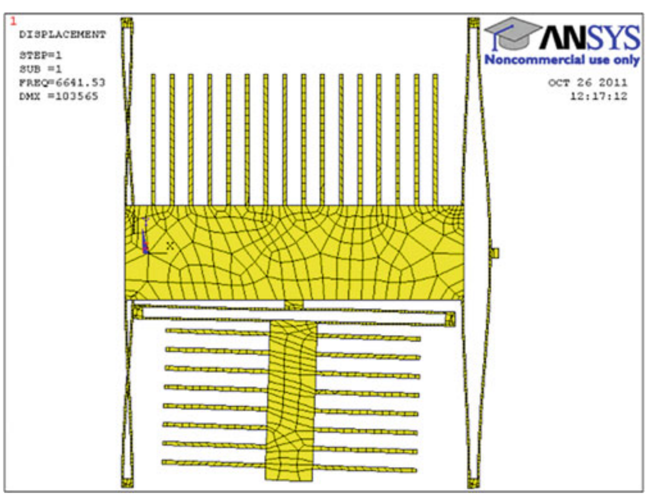

Fig. 4 ANSYS simulation of vibration mode along X direction

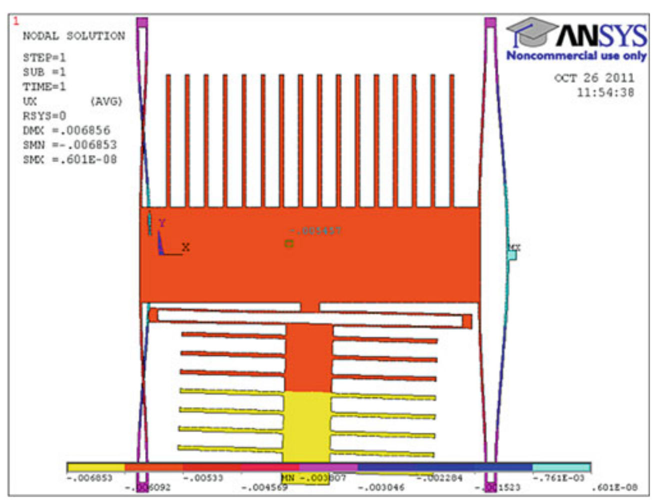

Fig. 6 ANSYS X-axis sensitivity simulation of MEMS dual-axis accelerometer

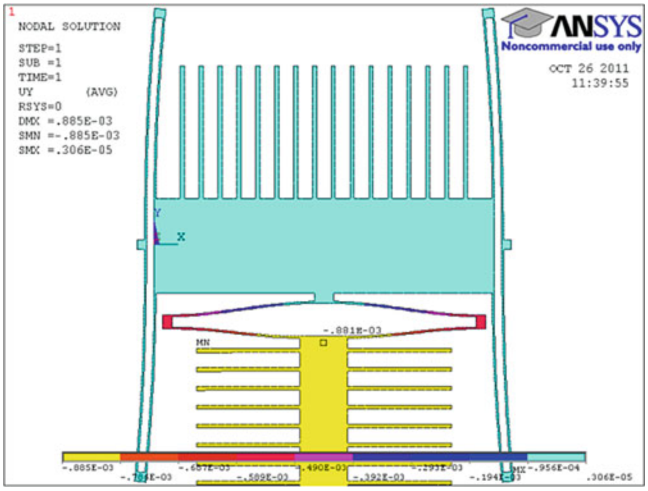

Fig. 7 ANSYS Y-axis sensitivity simulation of MEMS dual-axis accelerometer

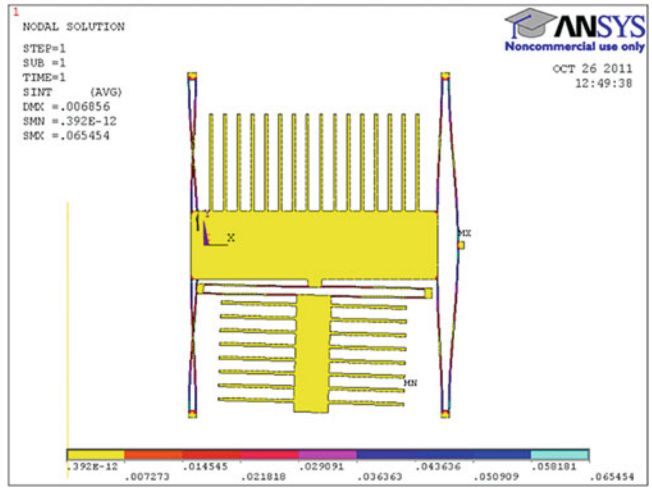

Fig. 8 ANSYS stress simulation of accelerometer for X-axis input acceleration

beam is along -*X* or -*Y* direction, which is the direction of resulted inertial forces. Neglecting the bending of comb fingers, the resulted displacement of device is read as the displacement on the movable mass. From Figs. 6 and 7, it is shown that the displacement sensitivity of dual-axis accelerometer along X-axis is $S_{dx} = 0.005457$ μm/g, and the displacement sensitivity along Y-axis is $S_{dy} = 10^{-6}$ μm/g. As we can see, the displacement sensitivity along Y-axis is smaller than that along X-axis. This is because the H-portion beams are much longer than T-shape beams, and the X-axis sensing mass includes sensing masses of H-portion and T-portion, but the Y-axis sensing mass only includes T-portion mass. The proposed design is suitable for high-g applications. In order to use it for low-g applications, we need to further enlarge the device so that larger sensing mass and longer folded-beams can be obtained.

ANSYS Stress Simulation

During the working mode, the folded beams bend in response to input acceleration. Such bending induces stress on the beam structure. If the maximum stress during working mode exceeds the material strength of silicon, the beams may be broken and the device is out of function. Fortunately, the displacement of beams in working modes is generally very small (generally in the range of nanometers or less). Thus the induced stress is well within the material strength. However, during unusual occasions such as dropping to the ground or working in a vibration environment, the acceleration input may be extremely high during a very short period of time. Thus it is necessary to simulate the stress distribution of the MEMS accelerometer. Take the X-axis input as an example, the simulated stress distribution of accelerometer is shown in Fig. 8. From the simulation result, we can see that the maximum stress generally occurs at the root of the beams connecting to the mass or the connectors. The sensing mass is generally non-deformable, as a result, it has the smallest stress level. To ensure reliability, for accelerometer working in harsh environment or high-g applications, the root portion of the beams may need to be widened in order to extend the lifetime of the device.

Device Fabrication Flow

The proposed H-T shape dual-axis MEMS accelerometer is to be fabricated with silicon bulk-micromachining and silicon-glass anodic bonding technique. Deep Reactive Ion Etching (DRIE) technique is used to release the movable microstructure. By using DRIE technique on single-crystal silicon, device thickness can be increased to 100 μm or more. Compared to surface-micromachining, this can greatly increase the device capacitance, hence make the signal detection easier. The proposed H-T shape dual-axis MEMS accelerometer can be fabricated using similar microfabrication flow as reported in the authors' previous work [10]. Silicon wafer is pre-etched at the bottom side of free-standing parts by 10 μm as releasing channel. The Si wafer is then bonded with glass wafer by silicon-glass anodic bonding technique. The silicon wafer is etched down uniformly to reduce its thickness to about 80 μm. The etched surface should be smooth to ensure the success of following photolithography process. After that, Al thin film (0.2 μm) is deposited on top of Si wafer by PVD (physical vapor deposition) and patterned as etching mask for Si DRIE etching. Finally, DRIE etching is used to etch down the exposed silicon area so that the device is finally released.

Conclusions and Future Work

In this paper, a novel bulk-micromachined MEMS dual-axis accelerometer device is proposed. The proposed MEMS dual-axis utilizes a unique H-T shape structure, which makes it very compact and can sense acceleration inputs along X and Y axes simultaneously. Such dual-axis accelerometer allows acceleration measurements along X and Y axes without utilizing multiple sensors, which leads to improved efficiency and reduced cost. Due to the perpendicular beam design for H and T portions, the signal cross-coupling between X axis and Y axis is minimized. The working principle of the device is analyzed and equations for device sensitivity along X and Y axes are derived. Based on analysis, a set of optimized design parameters of the accelerometer are suggested. ANSYS simulation is used to verify the correct function of the dual-axis MEMS accelerometer. The fabrication flow of the MEMS accelerometer is also suggested. The proposed MEMS dual-axis accelerometer can be used for 3-D inertial navigation system of automobile, airplane, sailing and other applications.

In the future, we will further improve the device design so that its sensitivity can be improved for low-g applications. We may further improve the device into a three-axis MEMS accelerometer, so that it can simultaneously measure acceleration inputs along all X, Y and Z directions.

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Prior Data Quality Management in Data Mining Process

Mamadou S. Camara, Djasrabe Naguingar, and Alassane Bah

Abstract

Data Mining (DM) projects are implemented by following the knowledge discovery process. Several techniques for detecting and handling data quality problems such as missing data, outliers, inconsistent data or time-variant data, can be found in the literature of DM and Data Warehousing (DW). Tasks that are related to the quality of data are mostly in the Data Understanding and in the Data Preparation phases of the DM process. The main limitation in the application of the data quality management techniques is the complexity caused by a lack of anticipation in the detection and resolution of the problems. A DM process model designed for the prior management of data quality is proposed in this work. In this model, the DM process is defined in relation to the Software Engineering (SE) process; the two processes are combined in parallel. The main contribution of this DM process is the anticipation and the automation of all activities necessary to remove data quality problems.

Keywords

CRISP-DM • Data mining • Data quality • Data warehousing • Software engineering

Introduction

Data Mining (DM) projects are implemented by following the knowledge discovery process. Several techniques for detecting and handling data quality problems such as missing data, outliers, inconsistent data or time-variant data, can be found in the literature of DM and Data Warehousing (DW). A literature review allows establishing the complex and timing consuming nature of these techniques. The present research work hypothesizes that this heaviness is the result of a posterior management of data quality problems. Indeed, most current researches assume that the DM process is subsequent to the Software Engineering (SE) process. The software resulting from the SE process produces data which is stored in databases or in a data warehouse in a first time, and later supplied to the mining process. A DM process model designed for the prior management of data quality is proposed in this paper. This model defines the DM process in relation to the SE process; the two processes are combined in parallel.

The paper is structured as follows: a review of the literature related to DM, DW and Data Quality Management is given in section "Literature Review". Section "Literature Review" also includes presentation of the limitations of existing DM process models, and a presentation of the concept of Data Mining Engineering. Section "Design of Work" includes a description of the design of the present research work. An illustrative example to demonstrate the

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DOI 10.1007/978-3-319-06764-3_37, © Springer International Publishing Switzerland 2015

applicability of the proposed DM process model is presented in section "Application Example".

Literature Review

The first sub-section presents the general principals of DM and DW. The following sub-sections present techniques defined in the DM or DW literature for detecting and handling data quality problems. The two last sections present researches on the Data Mining engineering and the limitations of existing DM process models, respectively.

Datamining and Data Warehousing

Knowledge discovery in databases is the nontrivial process of identifying valid, novel, potentially useful, and ultimately understandable patterns in data. Knowledge Discovery and Data Mining (KDDM) process models are meant to provide prescriptive guidance towards the execution of the end-to-end knowledge discovery process [1]. Cross Industry Standard Process for Data Mining (CRISP-DM) is reported to be the most widely used methodology in DM projects [1, 2]. The life cycle of a DM project, defined in the CRISP-DM, consists of six phases (the sequence of the phases is not strict) [2, 3]: Business Understanding, Data Understanding, Data Preparation, Modeling, Evaluation and Deployment or Implementation.

Tasks that are related to the quality of data take place mostly in the Data Understanding and in the Data Preparation phases. The Data Understanding phase contains four tasks: Collect initial data, Describe data, Explore data and Verify data quality. Verify data quality include to check if the data is complete (does it cover all the cases required), and if data contain errors or missing values [3]. The Data Preparation phase consist of five tasks: Select data, Clean data, Construct data, Integrate data and Format data. In the Clean data task, decisions and actions are taken to address the data quality problems reported during the Verify data quality task of the Data Understanding phase [3].

A Data Warehouse is "a collection of subject-oriented, integrated, non-volatile, and time-variant data to support management's decisions" [4]. In the multidimensional (MD) modeling, which is the foundation of DW, information is structured into facts and dimensions tables. The dimension tables contain the textual descriptors of the business [5]. A fact table is the primary table in a MD model where the numerical performance measurements of the business are stored [5]. A measurement (e.g., quantity sold or dollar sales amount) is taken at the intersection of all the dimensions (e.g., day, product, and store).

Missing Data

Many reasons can lead to missing data [6]:

1. procedural factors, such as errors in data entry, disclosure restrictions, or failure to complete the entire questionnaire;
2. the response does not apply (e.g., years of marriage for respondents who have never been married);
3. the respondents' refusal to answer certain sensitive questions (e.g., income level);
4. the respondent has no opinion or insufficient knowledge to answer the question.

Little and Rubin [7] distinguish between data that are missing at random versus data that are not missing at random. There are three ways to treat missing data [6]:

- Deletion procedures: Listwise deletion and Pairwise deletion;
- Replacement procedures: Mean-based, Regression-based and Hot-deck imputation;
- Model-based procedures: Maximum likelihood and Expectation maximization.

Zha, Song et al. [8], revealed some limitations that can be encountered when dealing with missing data using these procedures:

- deletion of missing data reduces the statistical power and affects the accuracy of estimate parameters such as correlations and so on;
- the accuracy of parameter estimation of the replacement-based techniques, especially the estimation of parameter uncertainty (i.e., standard errors), would be biased;
- the complexity of the iterative algorithms hinders the wide application of these model-based methods.

Outlier Detection

An outlier is an observation that deviates so much from other observations as to arouse suspicions that it was generated by a different mechanism [9]. The current methods to outlier detection can be classified into the following five categories [10, 11]:

- Distribution-based method: It is based on some standard distribution model (Normal, Poisson, etc.) and those objects which deviate from the model are recognized as outliers [12]. Its greatest disadvantage is that the distribution of the measurement data is unknown in practice.
- Depth-based method is based on computational geometry and compute different layers of k-d convex hulls and flags objects in the outer layer as outliers [13]. However, the algorithms employed suffer from the dimensionality curse and cannot cope with large k [11].

- Clustering-based method detects outliers as by-products [14]. However, since the main objective is clustering, it is not optimized for outlier detection.
- Distance-based method defined outlier as follows: A point p in a data set is an outlier with respect to parameters k and R if at least k points in the data set lie at a distance greater than R from p [15]. The drawbacks are the amount of computation time required and the impossibility to assign to each object a degree of being an outlier [10].
- Density-based method defined the Local Outlier Factor (LOF) that indicates the degree of outliers of an object using only the object's neighborhood [11]. The disadvantage of this solution is that it is very sensitive to parameters defining the neighborhood [10].

Inconsistent Data

A database instance r is consistent if r satisfies IC in the standard model-theoretic sense; r is inconsistent otherwise [16]. IC is a set of integrity constraints which the database instances are expected to satisfy [16, 17]. Consider a schema with relation P(A, B, C) and the functional dependency (FD) $A \rightarrow B$. The instance $D = \{P(a, b, c), P(a, c, d), P(a, c, e), P(b, f, g)\}$ is inconsistent wrt this FD [17]. In many cases, cleaning the database from inconsistencies may be costly, non-deterministic, and may lead to loss of potentially useful data.

Database integration allows bundling a data warehouse with multiple databases of different reliability and similar contents. Source databases may violate referential integrity and their integration may uncover additional referential integrity problems [18]. Referential integrity is a mandatory condition in a data warehouse where all the keys in the fact tables are legitimate foreign keys relative to the dimension tables [5].

A dimension in a data warehouse (DW) is a set of elements connected by a hierarchical relationship, where each element belongs to a category [5, 19]. The elements are used to view summaries of data at different levels of abstraction. Two main classes of integrity constraints, strictness and covering constraints, are used to check whether such computations, called summarizations, are correct [19]. Strictness constraints are used to require rollup relations to be functions (many-to-one relations) [20]. Covering constraints are used to require a rollup relation from category A to B to connect all the elements in A to at least one element in B [20].

Time-Variant Data

The time oriented databases must record the time varying nature of the information managed by the enterprise [21]. A time oriented database is based on the Temporal Entity Integrity (TEI) which is the temporalized form of the relational constraint of entity integrity.TEI is entity integrity restricted to a valid-time period [22]. A time period is represented by means of a begin point in time and an end point in time. TEI violation is the overlap of valid time periods, for the same object [22].

Temporal data mining deals with the harvesting of useful information from temporal data [23]. Temporal data mining is an important extension of data mining as it can be used to mine the activity rather than just states, and thus, infer relationships of contextual and temporal proximity, some of which may also indicate a cause-effect association [24]. Two types of temporal data are dominant in the development of temporal data mining. They are time-series data and sequence data [24].

The research works on temporal data mining focus on technical aspects, like algorithms, to deal with time-varying-data. They do not propose the necessary transformation in the overall DM process model to support the temporal data mining. Also, the other data quality problems are not taken into account in the proposed methods.

Temporal data warehouses join the research achievements of temporal databases and data warehouses in order to manage time-varying multidimensional data. Temporal data warehouses raise many issues including consistent aggregation in presence of time-varying data, temporal queries, storage methods, temporal view materialization, etc.

In a data warehouse, non-volatility ensures data durability while time-variation indicates the possibility to keep different values of the same information according to its changes in time [4]. Kimball and Ross [5] proposed three basic techniques for dealing with dimension attributes changes (overwrite the value, add a dimension row, add a dimension column), along with a couple of hybrid approaches. According to Malinowski and Zimanyi [4], these slowly changing dimensions techniques no not preserve the entire history of data or are difficult to implement. The changes in facts data are recorded using different types of facts tables depending on the time period represented:

1. Transaction Fact Tables represent an event that occurred at an instantaneous point in time, with a view of the business's operations that is at the individual transaction level [5].

2. Periodic Snapshot Fact Tables take a picture (hence the snapshot terminology) of the activity at the end of regular, predictable time intervals (day, week, or month) [5].
3. Accumulating Snapshot Fact Tables have multiple date stamps, representing the predictable major events or phases that take place during the course of a lifetime of a transaction or discrete product (or customer) [5].

A conceptual model for temporal data warehouses from Malinowski and Zimanyi [4], proposed the inclusion of temporal support in different elements of the model, i.e., in levels, hierarchies, and measures.

Limitations of Previous DM Process Models

Data collected for mining purpose contain noise, and may be subject non-quality problems. The previous sections show that the implementation of the methods for data quality can result in complex and time-consuming activities. This research work hypothesizes that an important amount of effort is needed in the Data Preparation phase because the management of missing data, outliers and inconsistent data is not based on the anticipation of data quality problems.

Time oriented databases and temporal data warehouses are good initiatives to manage time-varying data but the proposed solutions are not incorporated in the DM process models. In other words, the proposed approaches are not guided by requirements related to mining objectives.

Datamining Engineering

This section presents research works in which it is proposed to incorporate ideas from the software engineering in the DM process. Four fundamental activities are identified in software process: software specification, software development, software validation and software evolution [25]. Marban, Segovia et al. [26], conducted a comparison in order to identify what SE model elements (activities, tasks) are applicable to DM projects and are not covered by CRISP-DM. The comparison highlighted that CRISP-DM fails to address many tasks related to project management, organization and quality in enough detail to be able to deal with the complexity of DM projects. Thus, Marban, Segovia et al. [26], proposed a model for DM engineering that covers these aspects. Marban, Segovia et al. [26], aims to assure the completeness and quality of the DM project functions. Data quality problems are not addressed.

Design of Work

Even if researches in DM process models do not always reference to SE process, the two processes are strongly related. The source of the relation is that the data utilized in the DM process came from databases or data warehouses. The data stored there are produces by software created through an SE process. In the general case, the DM process model assumes that DM process must be executed after the SE process. The present work aims at designing a new DM process model based on the assumption that DM and SE processes are executed in parallel. The proposed changes will mostly concern the Data Understanding and the Data Preparation phases of the DM process in order to improve efficiency of the latter. The proposed DM process model is called the Data Preparation and Mining (DPM) process model. The main condition for making an organization or an enterprise use the DPM process model is when this one simultaneously has objectives and requirements related to:

1. specific business purposes for which a software is to be developed,
2. a DM problem definition related these same business purposes.

The interactions between the DPM process and the SE process are depicted in Fig. 1. The structure of SE process interacting with the DPM process is not subject to changes. However, new requirements related to the DM problem definition are defined in addition to traditional requirements like the need to support customer business processes. The DPM process is divided in two sub-processes: the traditional CRISP-DM process and the Prior Data Quality Management (PDQM) process. The software produced by the SE process will include features to automate the collection of the initial dataset to use in the CRISP-DM process. The PDQM sub-process consists of two phases:

- The Prior Data Understanding phase aims at defining requirements related to the way the software will be used for anticipating data quality problems in the collection of the initial dataset. The requirements are based the risk of the data to be missing or to contain incorrect values or outliers.
- The Prior Data Preparation phase aims at defining design and implementation level constraints to be integrated to the development activity of the SE process. These design and implementation elements are based on methods defined in the literature for detecting and handling data quality problems such as, missing data, outliers, inconsistent data or time-variant data. Four steps may be part of the design process: Architectural design, Interface design,

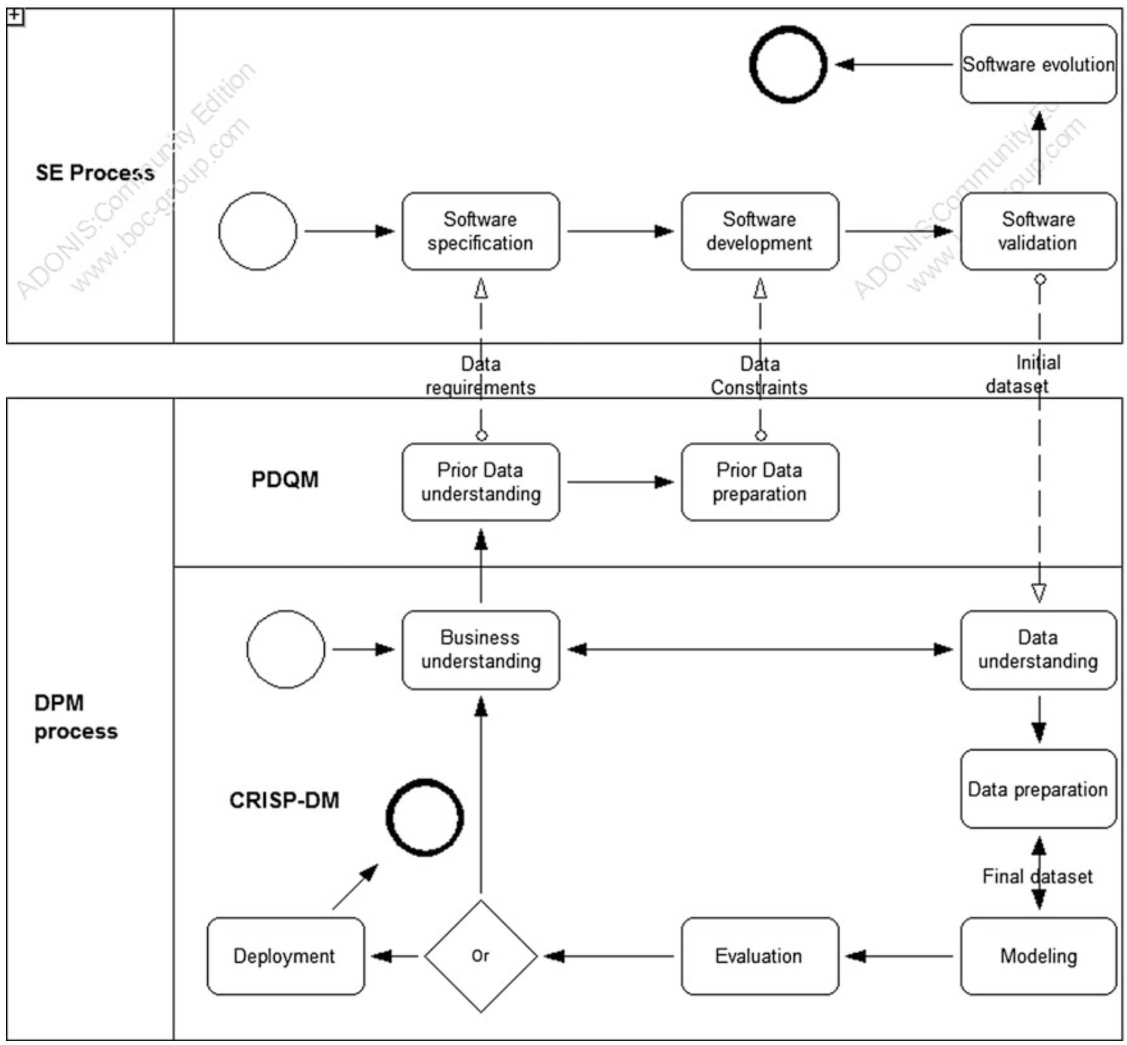

Fig. 1 Data preparation and mining (DPM) process model

Component design and Database design [25]. As the software development activity of the SE process begins with the design process, the constraints defined in the Prior Data Preparation phase must be used in one of its four steps.

After the validation of the software, the data produced are supplied to the Data Understanding phase of the CRISP-DM process via databases or a data warehouse.

The main contribution of PDQM sub-process in the DPM process is the anticipation and the automation of all activities necessary to remove data quality problems. To measure the importance of this automation, one has to consider the potential repeated execution of the data quality management activities and the accumulation of the complexity and the efforts generated by this repetition. Indeed, the repetition of data quality management activities can result from the necessity to update a model. Plan monitoring and maintenance is one step in the Deployment phase of CRISP-DM process. One activity of the plan monitoring and maintenance step is to determine when the data mining result or model should not be used any more [3]. This decision is based on criteria like validity, threshold of accuracy, new data, change in the application domain, etc. The solution if the model or result could no longer be used may be to update the model, set up a new data mining project, etc [3]. Applying the DPM process will be particularly interesting in the case were the model has to be updated using new data. In fact, data quality control mechanisms provided by the software (produced by the combination of SE and DPM processes) will remain functional regardless of the number of times that the model must be updated. Several DM techniques from supervised (decision trees [27], bayesian structure learning [28]) or unsupervised (clustering algorithms [27]) learning naturally support incremental learning (i.e. updating learned models). Incremental learning means to incorporate newly inserted data (i.e., database updates) into an existing model instead of relearning a new model from scratch [29].

Application Example

The illustrative example to demonstrate the applicability of the DPM process is taken from a software project of a higher education institution, the ESP (French acronym for Polytechnic Higher School of Dakar). The following sections present the execution of phases belonging either to the SE process or the DPM process.

Software Specification

The functionalities of the software to build are grouped in two modules: deliberation and scheduling. Figure 2 presents the initial Use cases Diagram of the deliberation module. This initial diagram is the one related to the specific business purposes for which software is developed. It does not include requirements for the DM problem. The ESP has two examination sessions: the first time examination session and the retake examination session. The marks obtained in these evaluations are used in the calculation of the Weighted Average Marks (WAMs) for each student. The marks of the retake examination session are registered by updating values for the first time examination session. This solution was adopted in the software specification because it was sufficient for the calculation of the WAMs.

Business Understanding

In addition to the needs related to the operational management of the programs, the ESP formulated needs for discovering knowledge about the factors of success or failure in the education of the students. The application of the DPM process in parallel to SE process was then proposed. The main objective of the DM process is to create a predictive model to be utilized as a support tool dedicated to finding the best candidates in the student recruitment process. That model is built for the prediction of the academic success of a student based on characteristics related to the student himself, his preceding grades, his school, or his social environment.

For the definition of academic achievement in the mining problem, a differentiation must be made between exclusion for unsatisfactory results, the repetition of the academic year, academic success in the first time examination session and academic success during the retake examination session.

Prior Data Understanding

The Prior Data Understanding phase of the example enables to define solutions for problems related to time-varying data.

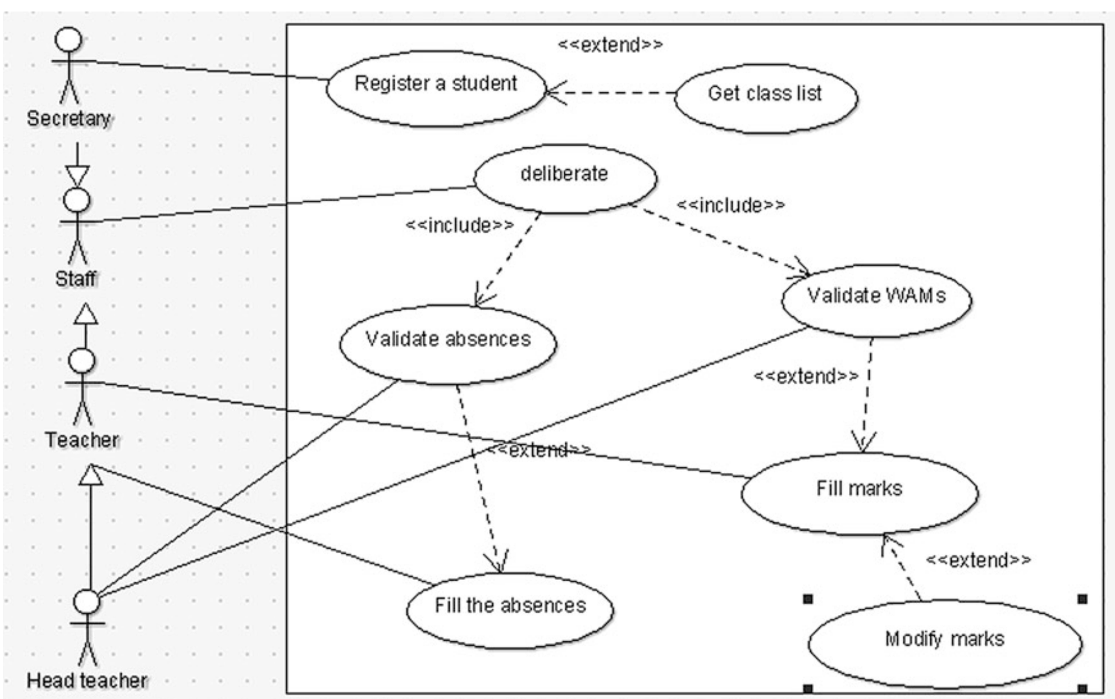

Fig. 2 Initial use cases diagram for deliberation module

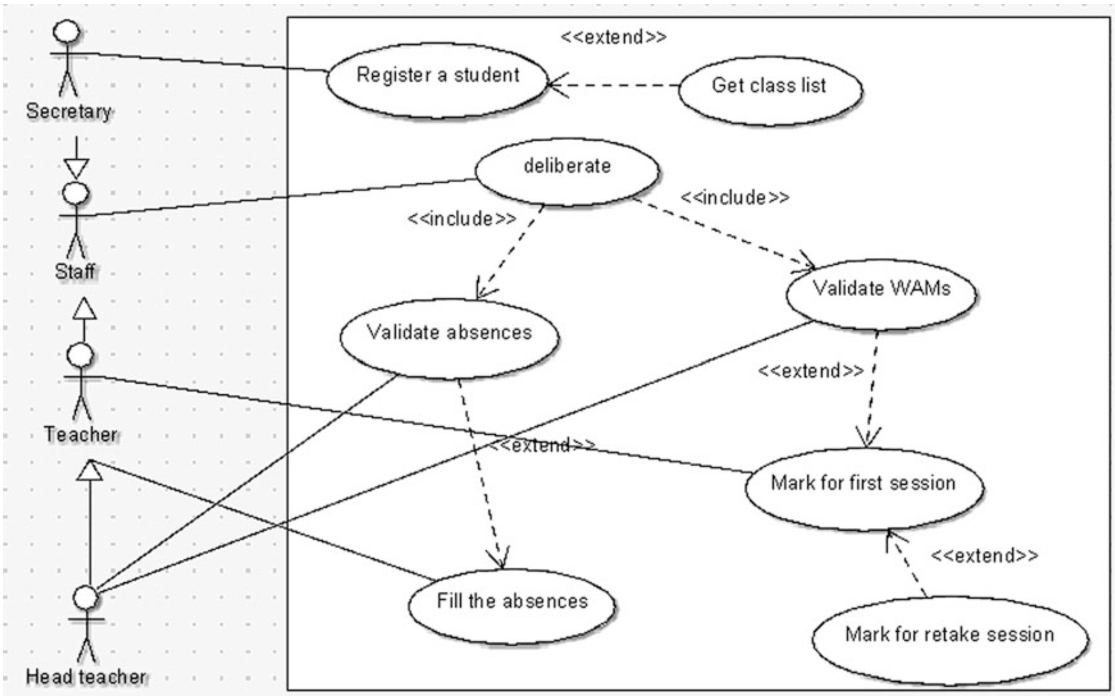

Fig. 3 Modified use cases diagram for deliberation module

Indeed, the replacement of first examination session mark by the one of the retake examination session led to the impossibility to differentiate between academic success in the first time examination session and academic success during the retake examination session. A direct consequence of this is a loss of information that can be important for the knowledge discovery because this differentiation is important in the definition of academic achievement (see the Business Understanding phase). The initial use cases diagram was modified to take in account these requirements (see Fig. 3). The use cases “Mark for first session” and “Mark for retake session” enable the time-variant information to be recorded correctly in two separate values. The time dimension is represented here by the sessions.

A single activity diagram has to be created for each use case in the system [30]. The activity diagram helps to sort out all the alternative flows in the use case [30]. For the example, if the initial use cases diagram (Fig. 2) is considered, the activity diagram representing the “Fill marks” use case is different to the one representing the “Modify marks” use case. If the business process modeling is based on the modified version of the use cases diagram (Fig. 3), a unique activity diagram (Fig. 4) can be used to depict these two use cases. Indeed, in order to register a mark(s) of a student(s) for a module, the teacher has only to specify the concerned session in the “Select session” activity (see Fig. 4).

The elicitation of the database requirements is done in the Conceptual Database Design (CDD). CDD develops a high-level description of the data to be stored in the database, along with the constraints that are known to hold over this data [31]. In the applicative example, this step was carried out using an UML class diagrams.

Prior Data Preparation

The prior Data Preparation phase of the example has brought changes in both the Component Specification and the Database Specification. The Database Specification consists essentially in a Logical Data Model obtained in the design phase of the development activity of the SE process. Logical Database Design converts the CDD into a database schema in the data model of the chosen Database Management Systems (DBMS) [31]. In the applicative example, only the relational DBMSs were considered. Therefore, the task in the logical design step was to convert an UML class diagram into a relational database schema (see Fig. 5). The primary keys are underlined whereas the foreign keys (referential integrity) constraints are displayed as directed arcs (arrow) from the foreign key attributes to the referenced tables (Fig. 5).

Conclusion

The main goal of this research work was to reduce the complexity and the amount of effort needed for the data quality management activities of the DM process. The

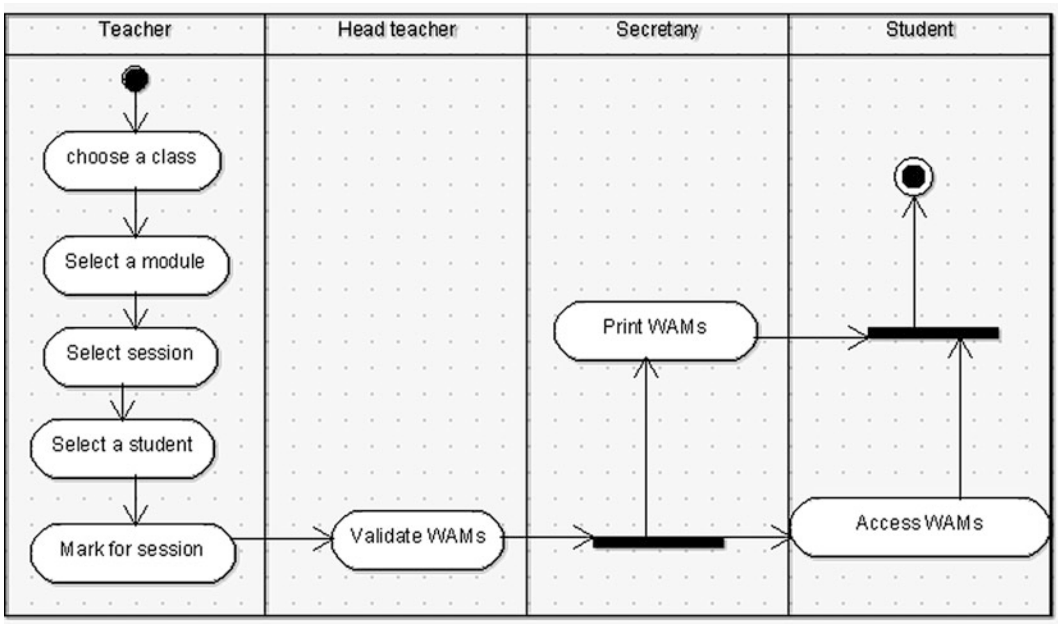

Fig. 4 Register marks activity diagram

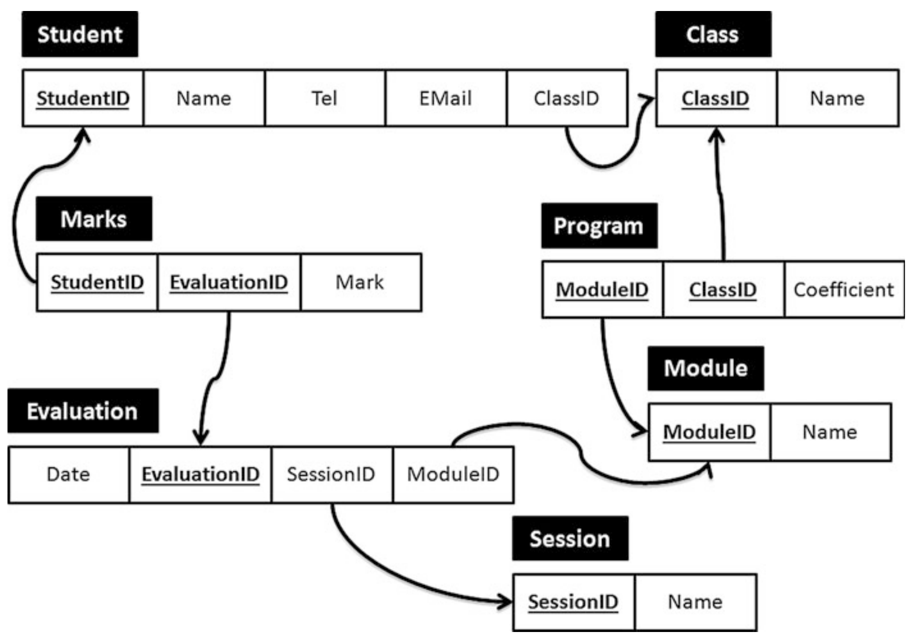

Fig. 5 Relational data model

proposed DM process is designed to be deployed in parallel to the SE process. It is assumed that the software must guarantee, in addition to supporting its principal functional requirements, that the data needed for the DM process will be available in the databases or in a data warehouse with the desired quality. For that reason, a variety of information concerning the DM problem is needed by the SE process in both its Specification and Development activities. All these information are supplied by the Prior Data Understanding and the Prior Data Preparation phases of the proposed DM process model.

The applicative example used to illustrate the application of the proposed DM process model is used to address data quality problems related to time-varying data. On perspective of this work will be to develop a complete case study covering the entire scope of data quality problem categories (inconsistency, missing data, outliers and time-varying data).

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Towards a Better Infrastructure Supporting the e-Education as a State Public Service

D. Kumlander

Abstract

The importance of e-education in the modern world is not debatable as sufficiently increases the auditorium and improves the students per teacher ratio. It also may serve the re-educational needs of a state and therefore should be recognized as an important part of public services that government provides via e-channels. The digital intelligence concept required building e-government is also based on educating citizens to be able to consume, process and formulate a feedback to government via IT channels. Unfortunately the lack of a general approach to the system in majority of states (countries) holds back the process of implementing such e-educational services. In order to bridge this gap the paper defines first areas of interest followed by expanded list of requirements needed to unify and improve such services. The paper also proposes technical solutions in form of hybrid SaaS (cloud) and the id-cards based identification system that satisfy all earlier stated requirement including reducing the cost of implementation.

Keywords

Cloud • Education • e-Government • IT infrastructure • Public services

Introduction

There are two approaches to the educational system in the modern society. One of them concentrates on educating young people only and can be seen as a university centric. It bases on believe that a correct, well-balanced education can be organized only gradually moving students from schools to universities and then to practice and it should give enough skills to enable them to upgrade knowledge later individually without involving mentors as should lay down the analytical and educational base for it. Within this approach the state builds up centers of massive education for young people and funds them both as educational and research centers concentrating the science and education within one, mainly public organization, although private centers also possible and are more or less developed depending on the country and the state approach to organization of such centers [1]. This approach has both positive and negative aspects [2, 3] and in the modern society can be seen as a public service to citizens within the government to citizens' collaboration. At the same time such educational form can be criticized as a kind of education concentrating only on a certain citizens' cluster (young people) and organized around accumulating the well-known knowledge in order to spread it without major updates of courses over decades is required for the quality education [3]. Universities do argue that this is much more stable method of educating as proper understanding of fundamental science (basics, techniques) [5]. Typically there are 3–2 such centers per one million of state population.

Another approach consumes the life-long education concept and offers a set of courses targeted to re-educate the state population either enabling the movement of the workforce between different industrial branches or motivating people to continue education on a constant basis in order to increase their productivity, the professional level or

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_38, © Springer International Publishing Switzerland 2015

educating them to the level from which they will be able to organize own workplace [6]. This approach is heavily used by dedicated programs of increasing the workforce professional level in order to decrease the number of low cost/low productivity work places or by unemployment offices that offers such re-education programs via partners. The main challenge here is maximizing the university ability to transfer the knowledge to practitioners [4]. Summarizing, earlier said we would like to highlight that in either approach we have a set of centers offering the educational programs and majority of them are the public services of the government to citizens just varying by the organizational forms and partners.

The quick development of electronic channels [6, 7] enable to provide access to such centers from anywhere sufficiently improving the availability of the education increasing the number of students that can be involved without increasing the number of teachers (which are known to be often the bottleneck as the teachers productivity per student within the old form was low and so the payments per teacher producing a sufficient inequality in the society decreasing the teacher status from year to year). Unfortunately attempts to consume such electronic channels were mostly local and lacked the strategic, governmental force driving the development of such virtual education environment. Besides such virtual education means also that we are facing completely new challenges in the process of building such systems—the requirements, activities processing and storing information are not the same any longer—differ from activities and requirements in the classroom environment based on a face to face collaboration of teachers and students and neither simple or obvious as contains a number of dangers and pitfalls and therefore requires clearly the governmental strategic approach.

Within this paper we would like to review such potential requirements and dangers and produce a general view on the better approach to organize the IT infrastructure supporting such "public service" educational initiatives highly required in the modern technological world. It is also important to note that this paper bases very much on EU approach and therefore can be less applicable to other regions although the requirements and conclusions do not change that much—probably other regions will add own requirements and the approach will vary in rather small details than in general.

Local Centers

A decision on the planned education service infrastructure for majority new projects is obvious as the small initial size dictates the choice of local servers and locally hosted and administrated software. This even applies to large institutional projects and the only exception occurs if the institution has a sufficient experience executing such projects and therefore may have formulated a strategy and IT infrastructure vision for such e-projects. The last is not very common and therefore the appearance of small local infrastructures mainly containing only one server is wide spread. The more attention the educational project gets the more important become non-context related requirements and at certain stage the project faces a challenge to be compatible with them. The reviewed situation can be named a "bottom up" as the appearance of such projects is dictated by the local needs/interests to offer an e-educational product. The other way, which is also important to understand, is the "top-down" approach where the public governance places the request to produce such projects to its subsidiaries (offices, departments, institutions). The evolutional approach in such projects has shown that first projects are appearing as very local projects also appearing on restricted, local infrastructures as primary goal and evaluation of such is done by the content, usability etc. leaving the IT properties (initially) out of the primary scope. Notice that even the more common approach to organizing such projects by outsourcing such courses to external partners by public institutions has no better position around earlier stated IT questions as those external companies usually are forced to develop own (and mainly from scratch) services making again and again common mistake been restricted in the budget.

As the main goal of executing e-educational programs is both to involve as many people as possible and control the cost of such projects the outsourcing become less and less attractive except open projects or charity driven projects. Even in that case the challenges remain for the security and reliability of such external projects as it will be described next.

Areas of Interest

The paper started reviewing the e-education strategies from the traditional understanding of this term: we would like to provide a professional knowledge to students as universities or schools do. At the same time it is not the only target on the electronic education. The e-government is traditionally divided to the government to government, government to business and government to citizens' clusters. Leaving to a separate discussion the state profit from educated citizens and so programs designed for it, we still would like to discuss other e-education related advantages in the government to citizen collaboration. As the EU e-government declaration says, the goal of the implementing the e-government is to empower citizens by e-government services designed around their needs increasing the access to the information and involving them into the decisions making process.

Notice that here the government is targeting to provide better public services with fewer resources, where among other services the following are mentioned: collaborative production of services, re-use of public sector information, personal mobility, green government and many others. Producing those requires a government centric approach that we do lack at the current stage.

In this situation the digital intelligence concept become extremely important in order to understand the paradigm shift to be done for making the e-government to be successful. In the part that is related to citizens it defines that the digital intelligence is the citizens' ability to receive information, process it and produce a efficient and meaningful responses through the same e-government system consuming dedicated IT technologies and communication channels. A special prerequisite is to educate citizens to use IT channels and expand via education their ability to solve problems and provide feedback individually, locate and consume required services via e-government, communicate their needs to e-government etc. We need to broad citizens vision on how the government works through dedicated e-educational projects expanding their current abilities to the e-era standards.

Requirements

Availability

The availability is important attribute of any software systems and can be described as an ability of the system to correctly serve the consumer needs. For example, It can be defined as a ratio of time the system is available and working vs. time the system is not able to serve requests or provide information due inability to connect to, use it or get the required result. This factor is important for the e-educational project as the low availability system sufficiently reduces the students wish to communicate or consume either this particular project's courses or be a part of any educational projects. Considering the fact that involving a student into education is very problematic by itself (see below) this factor can define the success of the public e-education.

Sometimes this is related to the next item called reliability.

Reliability

An ability of the system to provide the correct result is no less important and goes after the availability. The system should not only be available but also fully functional allowing students to resume earlier started coursed, get correct content of a lesson and correctly submit answers progressing along the course logic.

Integrity

There are less and less systems that are fully independent and self-sufficient. Therefore the integrity and correctness of co-work is recognized as the next important requirement placed to any software system including the e-educational one. For example it is important to create the central log on system as otherwise (again) the usability of the system and the wish to use it will sufficiently degenerate. Another example—the course is rarely taken individually as normally exists within some kind program where output of one course is an input for other courses. Such connectivity (integrity) is very important as allows decreasing the content of the course concentrating exclusively on the directly related topics excluding prerequisites and decreasing the ratio of failed students as guarantees that only prepared students enter the course.

Scalability

Another important requirement is the scalability. This property is rather technical although may depend on the course organization and teacher to student communication method. It is defined by the system ability to serve more students or ability to correctly serve the growing number of entering into the course students. It is the advantage of the electronic course—the ability to serve much more students than in the ordinal classroom approach, but still inadequate choice of the supporting IT infrastructure (servers, telecommunication channels) may lead to a threshold, which are lower than we would expect organizing such electronic courses.

Maintainability

The technological evolution nowadays increases its' speed from day to day. The course managers constantly need to reevaluate platforms and devices students are allowed to use accessing the course. Besides the need to reeducate citizens and teach them to consume electronic channels requires an ability to use a wide range of devices working on course from anywhere and anytime—for example commuting, waiting somewhere in a queue etc. It means we have to maintain the IT infrastructure constantly—updating it, closing security gaps and reconfiguring. The maintenance process should not be complex, time consuming or chaotic.

Confidentiality and Security

The access to the educational software data is a multidimensional question. The access to the system, its data and components require special policies and subsystems to guarantee that nobody can manipulate with them, steal it or modify the system in order to ruin its' work or distribute through it an unwanted software like viruses etc. The right level of the cyber security is required to protect channels, a dedicated log on system to the software data and components. Unfortunately this protection applies only to the virtual part of the system while access to the physical servers, to the building etc. is often forgotten. More importantly it can be the case, when the central e-government or e-public administration systems are following certain directives and rules, but the local sub-systems lack such protection although are working within the same logical unit—just by been outsourced and therefore much less controlled from the environment point of view exposing sufficient risks for the entire system. Notice that the rules described above regulate also very questions like how the servers' building should be constructed and protected from fire, earth quake etc.

The confidentiality question is a question of finding a balance between exposing data only following the direct expression of wish from the customer and consuming his data within the system or externally. Obviously our system should guarantee that no other person than the student or the teacher can access the student data unless it is accepted by the student. At the same time we clearly need to process the data within the institution for statistical and other purposes and quite often it is solved via a contract between the student and the educational organization when s/he enters the course. The data processing can be either local or global and exposing data to external companies it is important to anonymise it—a public data mart the other organization may be interested in (like how many students have certain skills before planning building a factory in one or another place) will be important public information allowing planning better business expansion or co-work with state educating required for-worse. Unfortunately the confidentiality of data is a complex question which is often underestimated and sometimes solved by disabling processing such data or exposing it.

In order to illustrate the importance and reasons the confidentiality we consider the following case: the soft identification of students within e-courses is the extremely important as the face to face collaboration is either impossible or extremely restricted [6, 10]. The soft identification system can be built using the biometrical markers [7–9] employing, for example, an architecture of the system presented on the Fig. 1 where confidential data about the students are biometrical patterns saved in a database and

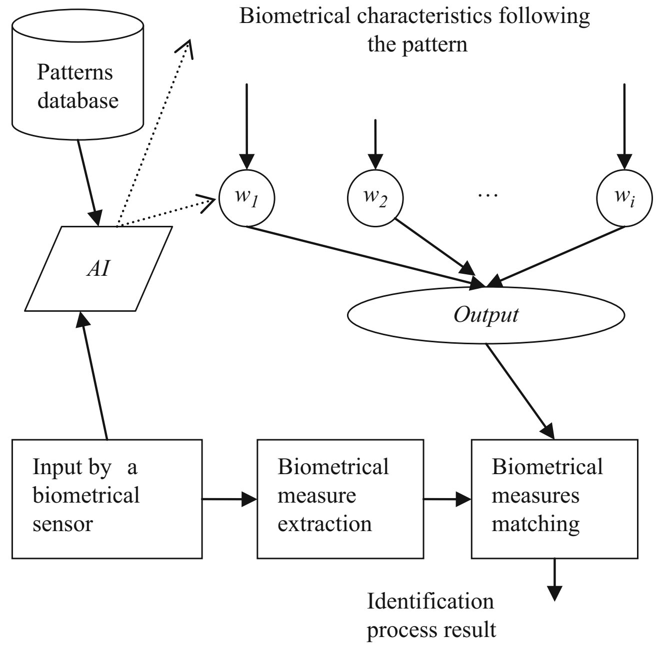

Fig. 1 Architecture of the soft-identification system based on a biometrical characteristics store as patterns in the system

consumed by an artificial system collecting the student input data constantly on the background. If we have several systems then we clearly lack a possibility to reuse collected patterns within other courses and will be forced to rerecord patterns every time sufficiently decreasing the productivity of the biometrical patterns systems at least during the first lessons. If we try to organize the transfer of such information between systems then the confidentiality question arise and the student acceptance is required to transfer such patterns. It will void the entire idea of biometrical markers as a soft-identification system working in the background "against the student" allowing issuing alerts to teachers on a possible misuse of the log on and examination systems—students may do that in order to involve other persons into the examination for obtaining certificates he has no knowledge and consequently rights for.

Cost-Related

Although the importance of education is not debatable, the education as a public service provides a possibility to have cost-saving effect from the size of the project as the large scale of those allows revising and decreasing the cost per student much more than any local project can do. As nearly no government has a wide range of other under-funded projects this property is also highly important. The cost saving opportunity open when the network of local projects is large and it is possible to employ the core property discussed in the paper—be a public state driven project and in the result produce the cost-saving effect instead of requiring more funding on the implementation and development.

Solutions

The previously described shows that we need a standardization of approaches via both—dedicated governmental strategic vision, which should produce directives, rules and standards and a special IT technological solution required to overcome local centers' gaps and produce a common practise for all platforms unifying the access, approach to building and consuming courses and resolving the technical requirements in one place avoiding a need to replicate them again and again and so minimizing the risk of misconfiguration and so not meeting the requirements. The ability to concentrate the servers in one place generates both economical effect (as let as to reduce the overall ownership cost) and the security and maintenance advantage (as sufficiently simplifies maintenance and control over the hosting center) producing sufficient competitive advantages over solutions/projects build individually.

The cloud approach has clear benefits which can be defined as the following:

a. Improves the assets utilization;
b. Improves productivity;
c. Increases the maintainability, scalability and availability (if correctly organized and managed);
d. Adds and ability to involve other parties by allowing them providing own courses and services still ensuring the cloud forced standardization, security etc.;
e. Improves the integrity of services resolving also confidentiality questions especially in specific and so hard cases like the soft-identification of students;
f. Shifts the focus of services providers away from IT hardware questions and allowing them to concentrate on software and content related questions.

The selection of the correct cloud platform is the next challenge designing the dedicated infrastructure. There are several possible solutions recognized as a cloud solution:

1. IaaS: infrastructure as a cloud—here we provide the infrastructure concentrating the servers in one place and allowing service providers build on this whatever software they want. While this fulfills certain our requirements (like access to servers, hardware related scalability, the building construction requirements), it only partly satisfies others (like software related scalability, confidentiality—all that depends on the particular software implementation) and do not satisfy other (like standardization etc.);
2. PaaS, i.e. platform as a service. Here, additionally to IaaS, the solution provider delivers also the virtual environment including web services, an operational system and so forth. Although the requirements coverage is improved here, the utilization increase is not sufficient and consequently is equal to the previous from the type of e-educational, state level projects point of view.
3. SaaS, i.e. software as a service. It can also be called "on-demand" software, i.e. here not only the infrastructure (hardware and software) is provided but also the dedicated software. The partners may fill it with additional modules or content (like courses content) been still restricted by implemented standardization of the particular software the cloud owner implements and provides to the end-user (student, citizen).

The selection among PaaS or SaaS model is debatable, but the earlier proposed requirements will certainly lead us to the conclusion that SaaS model is the best choice especially considering the wish to align the provided services to the e-government public services model.

Clouds can be also divided by the access method and so includes public, private and hybrid clouds.

1. The *public* cloud is designed for the public i.e. open use. Although it still maintained by a single operator, the access to the cloud and ability to build own SaaS solutions is open for every interested company. Notice also here the legal aspect since the access to the cloud data is open to the governmental organizations of the country the cloud operator is located in. This kind of the cloud certainly do not suit for e-educational projects provided within e-government initiative and even less suits for building any service of the e-government;
2. The *private* cloud is vice versa built by one organization to fulfill only their own needs and mostly is exposed to their customers in a form of SaaS. The collaboration to partners is extremely limited or impossible. Although this kind of the cloud technology seems to fit best the e-government services it still restricts too much the ability to involve partners and defines that the state should exclusively hold, maintain and develop services. In some cases it is much beyond abilities of the state personnel and so makes us to consider employing the model described next;
3. The *hybrid* cloud is the type of cloud sharing properties different cloud types described above, i.e. part of it is public, while another is private. Which part is bigger, the public or the private one depends on the targets. Here the e-government can consume the model of building the core part by the private cloud principles while exposing some services via the public cloud services, which are either designed for the public usage and therefore are not confidentiality sensitive (public data marts) or relying on external services to compute something exposing the sensitive data by anonymising it and using internal identifiers to be able to match to sensitive data left inside the system.

Note: There is another type of cloud as well, which is called a community cloud. It can be seen as a set of private clouds sharing common standards, jurisdiction etc., where the cost is spread to several organization. This type of cloud is a typical evolutional pre-stage of the full e-government where services are provided by different subdivisions (departments), which are, at the certain moment, start to concentrate within the most sophisticated departments. Still the lack of central vision makes them hard to cooperate and integrate and therefore the paper authors would still propose employing the hybrid cloud based on one major private cloud node.

It is very important to mention that certain services of e-government and e-education may be built in a form of social network allowing people to collaborate not only to the institutions but also to each other. Within the educational system such system allows to involve students more into the course, let them to discuss topics not only with a teacher but also within the students' community or perform certain tasks together. Within the state scope it promotes social movements and activates citizens to help each other, promote their social believes and fulfills the social target of the state. The ability to implement such social network (instead of consuming them) within e-government central IT infrastructure has sufficient benefits. First of all, it opens for us a possibility to implement police and other rescue informational systems designed to improve the rescue response time (involving also the broad auditorium into the process and activities like, for example, the search operations) or a notification system designed to notify and prevent people from doing something (like for example moving into a place under risk due a snow storm etc). Secondly, it let the state to direct or observe the social movements. Notice an important difference between directing and restricting or manipulating. Here the task is to lead the process of making social decisions through a set of legal stages in order to obtain the law status (the e-democracy is the best example of such system). In the e-education case such social system may help to improve courses, redesign them or introduce new.

Another important subsystem is the identification one. The access problem can be solved differently. For example the EU directives states that the actions plans foresee unification of citizens' access to public services strategies in a form of the ID cards solution. Although this approach is very much criticized in certain countries like UK and USA, EU still finds it to be a future technology approach. The most important here not the implementation technique, but the goal, which is—we should find a common (for a country) way to access public services like e-educational courses in order to unify and simplify access for the entire nation.

Conclusions

In the first part of the paper we have presented the e-education or virtual environments educational courses in the broader scope proposing to see it as a public service provided within the e-government initiative. Beyond teaching citizens professional subjects allowing them moving along the individual career the state development clearly requires to develop citizens' abilities to consume the government electronic services, improve their understanding and skills to communicate to the government via electronic channels within the digital intelligence initiative. In the second part the requirements for such e-educational environment were formulated in order to show potential problems associated with an improper realization. It was found that not following such requirements can sufficiently increase the cost, produce vulnerabilities and even sufficiently decrease the citizens wish to enter the educational program. It is very hard to ensure full coverage of requirements in the local implementation since it doesn't have the state level general strategy and the enforcing power to standardize the approach. In the last part of the paper we have shown that all this can be done by concentrating different systems (implemented by state offices, departments, institutions) within one standard IT infrastructure consuming the cloud approach to building the soft- and hardware environment. The access to such environment should be unified and one approach is using the id-card system, which is very much supported by recent EU decisions in the e-government field of technology. Building the environment by such approaches should both show the strategic wish of the state to educate citizens and help the state to be an attractive, light weighted environment with minimal cost or even reduced cost in compare to the current that are spread among different institutions and so consumed inefficiently in many countries.

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Novel Time-Stamped Digital Card (TSDC) to Determine User's Identity

Munif Alotaibi, Aziz Alotaibi, Adel Qahmash, and Zhengping Wu

Abstract

Time-stamped digital signature scheme can be employed to derive a new solution that improves the authentication, non-repudiation and data integrity of current payment gateway. This paper suggests designing a new method to process online payment transactions by adding an additional step to the current payment gateway which will verify the bank cardholder's identity through a trusted third party, a certificate authority (CA). Therefore, a new Time-Stamped Digital Card (TSDC), which relies on the concept of asymmetrical key algorithm to accomplish its task, will be proposed. Time stamp is examined to determine how old the TSDC is and whether it has been previously used or not. It will also be used to prevent replay attack. The protocol of TSDC consists of five steps and has been analyzed. TSDC needs to be generated, signed and submitted to the payment gateway along with the credit card information by the customer prior to processing any online transaction for identification purposes. Therefore, TSDC aims to reduce the high volume of fraud transactions and suspicious activities currently experienced during payment gateway due to lack of identity verification of bank cardholders.

Keywords

Security • Time-stamped digital card • Online payment • Payment gateway • TSDC

Introduction

Information technology has rapidly grown to become an essential part of many businesses. Network technology has played an extensive role in serving businesses in not only accessing information, but also in exploring new ways of communication and rendering services to consumers. For instance, selling online products, and processing online payment [1].

However, using network technology in business has also introduced several issues along with these benefits. For example, a long term concern for businesses is the lack of a secure network system which affects the processing online payment [2]. Security of online payment system has stood out as one of the services for business owners where security assurance becomes problematic. There have been several attempts to protect online payment processing from being taken advantage of through ports in the payment system [3]. Yet, an efficient and stable solution to the problem has not been found. Consequently, this paper introduces a new approach which employs time-stamped digital signature scheme for coming up with a new solution with the intent of improving the security of existing online payment method. Time stamped digital signature is an authentication method that has been used in many applications such as Microsoft Word and Adobe PDF to name a few. It enables the sender to attach a code to the message as a signature, in order to provide data integrity, authentication, and non-repudiation. Digital signature alone is a combination of the sender's private key and the encrypted message. The receiver would decrypt and validate the message using the

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_39, © Springer International Publishing Switzerland 2015

sender's public key. The purpose behind using the digital signature is to assure that the message has not been altered during transmission and to verify the identity of sender. The digital timestamp is a specific time that indicates when the digital signature is created. Timestamp is used to validate the lifetime of the digital signature and prevent a replay attack.

Related Work

A payment gateway is known as an e-commerce application service that allows many e-businesses and online retailers to authorize online payments. The payment gateway basically works as a middleman between customers and online businesses by facilitating the online transaction. The most critical fact in payment gateway security system is determining user identification. Some researchers have proposed solutions to solve the issues of payment gateway; however the current security standards have not reached the customers' needs. Moreover, in the current payment gateway's system, there are some mechanisms that aim to enhance the security of online transactions, such as fraud detection tools.

A recent research on payment card fraud recommended some techniques for detecting payment card fraud such as the use of Address Verification Service to match the billing address of the cardholder on file with the card Issuer and Card Verification Value/Code (CVV2/CVC2) [4].

An informative research conducted by Mahdi, M.D.H., Rezaul, K.M., and Rahman, M.A, provided statistics on how customers suffered from online fraud payment transactions in UK. The survey presented data for various categories such as those who have been victims of online identity fraud once in their lives (20 %), twice (14 %), three times (8 %), four times (1 %), and five times (1 %). 56 % of the respondents were not victimized because they do not use online banking [5].

Based on these statistics, the author suggested some ways to reduce online fraud and recommended few steps for the customers such as ensuring the presence of the lock padlock or unbroken key mark in the web browser, which indicates that that site is secure. The web site URL should include the protocol of "https" which show that the connection is secure. The research also recommended additional tips for online customers [5].

Another research that provided valuable insight was, "Key Technologies for Security Enhancing of Payment Gateway." This research focused on payment gateway security and enhancing its security phase, the research proposed a technique, which is based on fusion-based payment protocol blinding SSL and SET. In this approach, the author proposed a solution where the hash algorithm "MD5" is used, and integrated with two protocols SSL and SET for security needs. The author mentioned ten steps that users will go through during the transaction. This approach aimed to achieve one major element of payment gateway security, which is authentication [6].

As many researches on payment gateway security attempted to come up with a comprehensive solution, a new system for processing online payment with identity verification method using Handwritten Signature Verification system has been proposed [7]. The system will do algorithm analyses for both the static features of a signature such as shape, size, and its dynamic features such as velocity, timing, and pen-tip pressure. The algorithm analyses aims to make a judgment about the signer's identity by comparing both the Handwritten Signature, which was provided during the registration time with the one that submitted at sale point.

Furthermore, another method that has been proposed in an attempt to solve this issue is designing a special credit card that has an inbuilt chip. The function of the inbuilt chip is to generate a new credit card number after every new transaction by using hash function method, and then delete the previous credit card number. The same function will also be used by the card issuer to compute and update the user's new card number. In order to make an online payment, the user will require a card reader to swipe his/her card through [8].

Problem Statement

For online consumer, SSL technology can easily help to identify if the website is safe or not. However, payment gateway cannot recognize with whom it is interacting. For instance, any user is capable of accessing the payment gateway and submitting any credit card information even if does not belong to them. Figure 1 explains the issue in a visualized way.

Therefore, due to the lack of identity verification of bank card holders, current payment gateway system experiences a high volume of fraud transactions and suspicious activities.

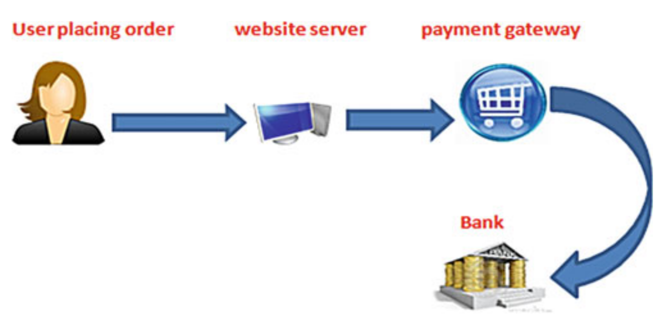

Fig. 1 Traditional process of online payment

The goal of this paper is to come up with identity verification solutions that can help to prevent online fraud transactions and suspicious activities. Time-Stamped Digital Card (TSDC) is suggested as a potential solution that would be used to secure the payment gateway and reduce the high volume of fraud transaction.

Proposal Design

TSDC Approach

As mentioned earlier, Time-Stamped Digital Card (TSDC) aims to reduce the high volume of fraud transactions and suspicious activities which payment gateway experiences due to the lack of identity verification of bank cardholders. Therefore, it will be used as digital identification card that will verify the identity of the customer. TSDC needs to be generated, signed and submitted to the payment gateway along with credit card information by the customer before processing any online transaction in order to identify his/her identity.

Figure 2 illustrates the proposed scheme that will be used in this paper in a visualized way.

TSDC will be valid and used only for one-time, per one transaction. If more than one online transaction is required; a user has to generate and submit more than one TSDC. Every TSDC will be unique due to unique identifiers—timestamp and nonce (random number). Time stamp is also used to deal with the expiration of TSDC, it will determine how old the TSDC is and make sure it has not been previously used. Thus, it will also prevent any replay attack. Once generated, TSDC will be expired within a certain period of time. Algorithm such as the following will be used to do the function of determining if TSDC has expired or not.

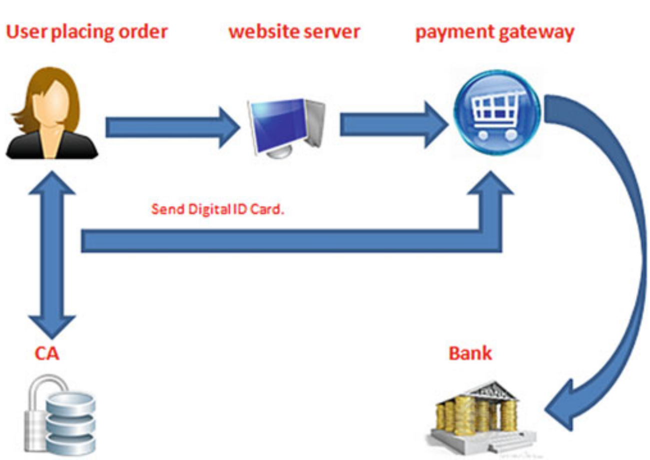

Fig. 2 The process of online payment after the proposed TSDC

Table 1 Elements of TSDC

Elements of TSDC

- Name of Customer
- Expiration Date of TSDC (life Time)
- URL of Payment Gateway
- Random Number Generated by CA

```
If (time – current Time) >= 1 hour
{
        TSDC == veiled;
    }
Else
        TSDC == unveiled;
```

Time is not the only element that determines if TSDC is valid or not, there are other elements, such as making sure the content of TSDC has not been modified and the TSDC is not fake. Hash function will be used in this case; it mathematically identifies the content of the TSDC so that even the smallest change to the code and content of TSDC will invalidate it. The same hash function will be used at the time of creating the TSDC and at payment gateway to automatically validate the TSDC.

Furthermore, for the privacy of consumer, name of consumer and expiration date are the only two elements that the TSDC contains and the certificate authority collect about costumers each time they do an online payment. The elements of TSDC are shown in Table 1.

There should be one federated format for the file of Time-Stamped Digital Card (TSDC).

The CA in this approach can be either a banking self-built CA, a government CA, or a business CA. however, in all cases the CA should be assured of the identities of its users upon the time of registration.

TSDC Protocol

Since our concern is to use timestamp digital signature technique to verify the user identity, a Timestamp Digital Card (TSDC) protocol is proposed. The TSDC protocol aims to generate the TSDC and sign the transaction securely. The TSDC protocol defines the format and the order of all messages exchanged between the consumer, CA and payment gateway, as well as any actions taken during the exchanged messages. The protocol has the potential to generate a unique Digital Card that its information will be stored on the CA for security concern in the future use.

The TSDC protocol is summarized into five steps:

a) The user will send a request to CA.
b) CA will generate a random number, store it, and send it back to the user.
c) The user will submit the TSDC along with credit card information.
d) The payment gateway will communicate with CA to check if the TSDC information such as random number and timestamp are valid or invalid.
e) The CA will respond to the payment gateway after checking if the TSDC has been issued to that user and still valid.
f) Finally, the payment gateway verifies the user's identity, and responds to the user by accepting or rejecting the transaction.

First Step

$$A \rightarrow CA: \quad ID_a || URL || N_1$$

In case, the customer wants to issue a new (TSDC), customer will send a request with his/her ID, URL, and T. After the CA received the request, CA will check the user public key certificate. The CA will use the user ID as unique key to store the TSDC information. Based on the payment gateway's URL or Domain name, CA will send the website's public key (which is the payment gateway) to the sender. Furthermore, in the future, CA will not provide any TSDC information such as user name to the gateway till the payment gateway's URL's verified. Time will be attached to assure that the request is fresh and to prevent any replay attack as well as to make the TSDC unique along with URL, ID and RN.

Second Step

$$CA \rightarrow A: \quad E\big(PR_{auth,}\,[ID_a \,||PU_{website}||\, N_1]\,\big|\big|\,E(PU_{website,}\,[Name\_Of\_Customer \,||RN||\, LifeTim])\,\big]\big)$$

After the CA receives the request from the sender, it will provide three main elements that will be the main content of the (TSDC) which will be encrypted by the website's public key to be secure over the transmission: First, user name will be fetched and can be known in the CA by using the user's ID. Later, the payment gateway can decrypt and authenticate the TSDC. Payment gateway will be able to read the customer's name and match it with the name of customer on the credit card. However, gateway will not process the transaction till it checks with CA that the digital card with this customer's name is valid. The other element is the timestamp (Lifetime); it will be used to sign the TSDC with a time limit to validate the card. Random number also will be generated by the CA and will be included in the (TSDC), therefore, the gateway can check with CA to determine if this random number is belong to the user and has not been used before.

The Linear Number Generators technique is used in the TSDC protocol to produce a random number and attach that to the TSDC. The Linear Number Generators algorithm is:

$$Xn + 1 = (aXn + c) \bmod m$$

Third Step

$$\begin{aligned} A \rightarrow B: E\big(&PU_{webstie}, [(Name\_Of\_Customer \,||RN \,||\, LifeTim)]\big|\big| \\ &E\big(PR\ user,\ [H(TSDC)]\big|\big|ID_a\big|\big|E\big(PR\ user,\ [T\ \big|\big|\ N_b]\big) \end{aligned}$$

As mentioned above the main elements of the TSDC are (username ||RN || LifeTime).

The Signed TSDC will look like

$$\begin{aligned} = E\big(&PU_{webstie}, \big[(Name\_Of\_Customer \,||RN \,||\, LifeTim)\big|\big| \\ &E\big(PR\ user, [H(TSDC)]\big|\big|ID_a\big|\big|E\big(PR\ user,\ [T\ \big|\big|\ N_b]\big) \end{aligned}$$

In Fig. 3, the RSA approach is used to generate a digital signature. The TSDC will be used as an input to the hash function which produces a secure hash value. The hash value will be encrypted using the user's private key. Both TSDC and the digital signature are transmitted to payment gateway. The content of the TSDC will be encrypted using the website's public key to provide confidentiality over the transmission. Timestamp is sent to inform the payment

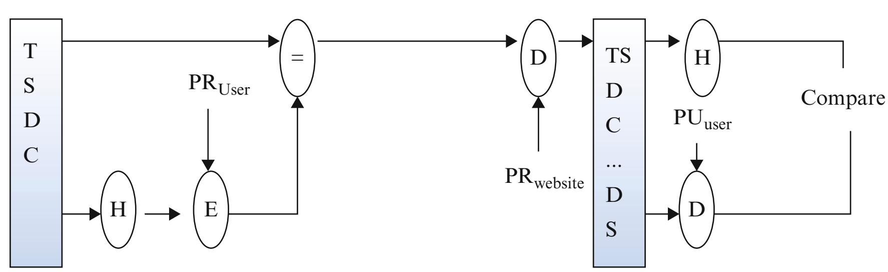

Fig. 3 Generating TSDC using digital signature

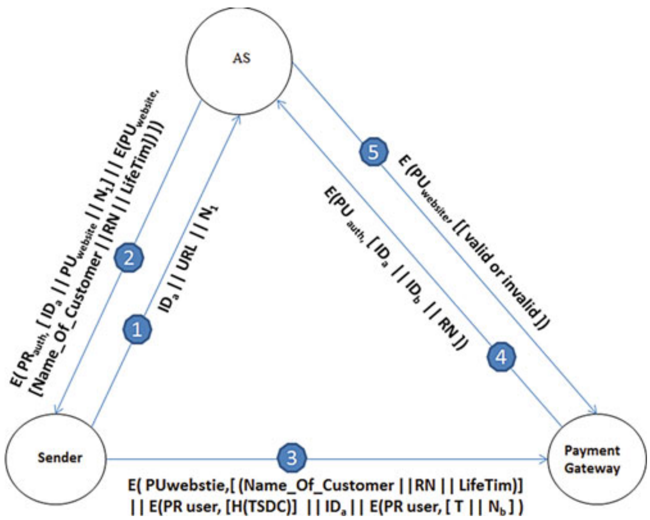

Fig. 4 The five steps of TSDC

gateway that the TSDC is timely and not a replay. Nonce, is a random number, has been used as a unique identifiers for each transaction.

Fourth Step

$$B \rightarrow CA: \quad E(PU_{auth,} [\, ID_a \| \, URL \| \, RN \,])$$

After B used the digital signature to authenticate A, B will send IDa , URL, and the RN to the CA encrypted with the CA's public key to verify that the user with ID and RN has a valid a TSDC. CA who generated the random number (RN) can verify that the random number which associated with IDa and URL has not been used before.

Fifth Step

$$CA \rightarrow B: \quad E\left(PU_{website}, \; [[\, \text{valid or invalid} \,]\right)$$

The CA will replay back to the payment gateway with response indicating if the TSDC is valid or not. The above five steps can be described in Fig. 4.

Working Prototype for TSDC

In the widely used current payment system, consumer who wants to do online payment needs only to submit his/her credit card information to process the payment transaction such as the name of consumer, credit card expiration and the address of the credit card holder. However, applying TSDC approach to the current procedure of the payment gateway system requires consumer to undergo two extra steps which are—generating and uploading the TSDC before submitting the transaction. User has to have a public key certificate through a third party in order to generate the TSDC.

Both payment website that is compatible with the TSDC and an application to generate TSDC have been created to test and analyze TSDC approach. The TSDC application has a "login" interface to the CA to allow the user to login to the TSDC generator application and raise a TSDC request to the CA.

The CA would reply with information which includes a unique random number and the consumer's full name.

As stated earlier in the second protocol, the user will receive certain information from the CA which will form the content of the TSDC.

In that application, the user will enter the website's URL and submit the request of generating a new TSDC. The URL is payment gateway domain name. It will be stored on the CA side among the user information. The TSDC will be generated and stored on the user's computer as file with execution *.TSDC. The file is to be submitted along with credit card information to the payment gateway. Likewise, more professional friendly web interface that uses the mechanism and protocol of TSDC can be designed in future to achieve the same goal. This experiment showed great results. Once the user submitted his/her TSDC with the credit card information, the payment gateway can then authenticate the sender by using the digital signature technique. It will also compare the name of the user on the credit card, which will be submitted in Credit Card Holder Name field as shown in Fig. 4, with the name of the user in the TSDC.

The following code I used:

```
If (NameOfCustomerOnTSDC = NameOfCustomerOn credit card)
{
        Transaction == valid;
}
Else
        Transaction == invalid;
```

Furthermore, per the submission of the TSDC protocol, there will be communication between the payment gateway and the CA to authenticate TSDC. If the payment gateway accepts the transaction, the user will receive a confirmation; otherwise, error message will be showed.

In the above experiment, simulation system has been created to test and measure the performance and the function of TSDC system. TSDC protocol has been tested and analyzed in this experiment, as well as the new payments system which is designed to include the extra steps that are required by the TSDC approach.

A Comprehensive Comparison

In the related work section, many methods to protect online payment transactions have been provided, however, comparing with our model, in [4, 5] and [6], the steps which those authors mentioned do not substitute the need for implementing a new secure mechanism for payment gateway, some of these steps are only to secure the communication and can't verify the identity of the user. In [7], the authors came up with a nice idea. Yet, the system is not very accurate since its performance achieves an overall error rate of 2:1 % in the best case. Comparing with the TSDC, the average memory consumption of TSDC would be better than the Handwritten Signature (HRS) method because the average size of the signature is larger than the average size of the TSDC as indicated in Fig. 5.

Hence, one of the advantages of TSDC approach is to provide security to the existing payment gateway by using Time-stamped digital ID as identification. Another advantage of the TSDC is that it uses the URL of a specific website which narrows its usage. This means each TSDC is designed to be used one time for one website and for one online payment transaction.

In the meantime, implementing the TSDC approach will increase the cost since every consumer needs to gain the public key-certificate through a third party. However, The proposed protocol will help payment gateway to verify the identity of the cardholder by matching the information on the credit card with the information of the generated TSDC. The proposed protocol is used to make the TSDC trusted by both parties: users and payment gateway. Also, the protocol has been designed to provide more secure method to protect the TSDC information. However, all these processes will require the CA to be able to perform huge transactions, which puts a huge burden on the CA to register and provide too many TSDCs at the same time.

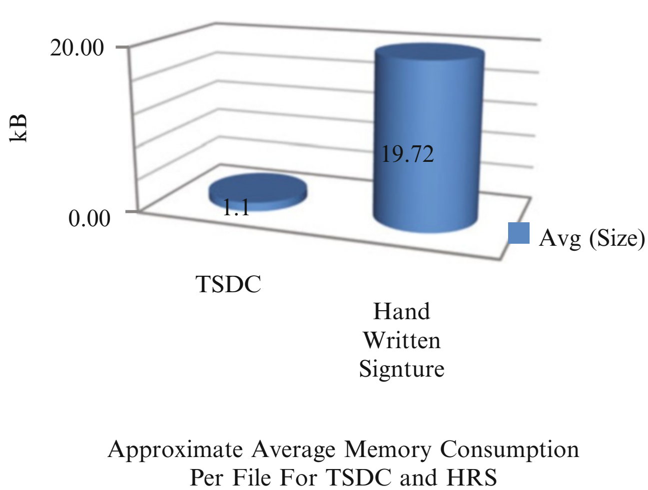

Fig. 5 Comparison between TSDC and HRS

Moreover, to generate and upload the TSDC may require more time to finish the online payment transaction. However, both sides, user and the payment gateway, would be more secure. Authentication, integrity, and none-repudiation are the security services that have been achieved through the TSDC protocol. Digital time stamped signature is used to provide authentication and prevent replay attack. Since the payment gateway has the ability to verify the TSDC content with the CA, the integrity has been achieved through the entire protocol.

Conclusion

The main goal of this paper is to propose a new mechanism that can reduce the high volume of online credit card fraud. To the best of the authors' knowledge, the proposed solution in this paper is one of the only few proposed solutions that attempted to solve the lack of identity verification of bank cardholders in the payment gateway. The TSDC is a new approach that utilizes time-stamped digital signature scheme to improve the security of current online payment method.

It requires redesigning the current payment gateway system by adding a new step which is intended to verify credit card holder by involving third trusted party (CA). Moreover, TSDC integrates the concept of time-stamp digital signature which depends on both public key and private key. The TSDC protocol consists of five steps and has been tested and analyzed in this paper. First step is that the consumer sends a request using the TSDC application to the CA. The second step of the protocol is that the CA generates a random number and sends it back to the consumer along with Public key of the website. The TSDC is stored on the consumer machine and has four main elements: Name of Customer, URL of Payment Gateway, the Expiration Date of TSDC (life Time), and Random Number Generated by CA. The Third step is that the timestamp digital signature will be formed on the TSDC content. The TSDC will be submitting through the payment gateway page. Fourth step is that the payment gateway reads the TSDC file and sends a user name and random number to CA to check validates TSDC. Finally, CA checks the TSDC content if it has been registered before and has not been used, and replays to payment gateway with a valid or not valid message. The protocol has been used to provide three security services: authentication, integrity, and non-repudiation.

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PSPICE Implementation of Block-Wise Shut-Down Technique for 8 × 8 Bit Low Power Pipelined Booth Multiplier

Umatri Pradhananga, Xingguo Xiong, and Linfeng Zhang

Abstract

VLSI continues to shrink the feature size of transistors. Nowadays the technology is advancing to deep submicron level in which leakage power becomes dominant in VLSI power consumption. Traditional logic shut-down technique only eliminates dynamic power when circuit is idling, but leakage power cannot be eliminated. To eliminate leakage power, the voltage source (V_{dd}) should be completely shut down when circuit is idle. However, shutting down and enabling V_{dd} in a large circuit results in a large transient current, which may lead to error of circuit. Block-wise shut-down technique has been reported by researchers to avoid this issue. It shuts down and recovers back V_{dd} of blocks in a pipelined circuit sequentially when circuit is idle. This totally eliminates leakage power, while avoiding the large transient current in circuit, hence resulting in less glitches. To verify its effectiveness in power saving, we implemented block-wise logic shut-down in an 8 × 8 pipelined Booth multiplier. Whenever the multiplier is idle, supply voltage is turned off block-by-block, eliminating both dynamic and static power. The schematic of the circuit is designed using PSPICE. Simulation results verify the correct function and the expected shut-down of the designed Booth multiplier. PSPICE power simulation demonstrates effective power saving of the Booth multiplier for the given input pattern sequence.

Keywords

Low power VLSI • Block-wise shutdown • PSPICE power simulation • Booth multiplier • Pipelined multiplier

Introduction

Multiplication is the general arithmetic operation which is of great use like other arithmetic operations. Multipliers are one of the important building blocks of microprocessors and play vital role in DSP (Digital Signal Processing) application as well. As the multiplier plays an essential part in the microprocessors and DSP applications, several low power multiplier designs have been introduced. During a DSP computation, it has been found that multiplier consumes lot of power. Hence, it is essential to design low power multipliers for a DSP system.

Traditionally, in Complementary metal-oxide-semiconductor (CMOS) technology, dynamic power dissipation is comparatively very large than the static power. Dynamic power dissipation can be reduced in the multiplier by minimizing switching activities. In order to optimize the switching activity, hardware bypassing technique is applied. However, the extra circuitry added for bypassing actually consumes more power. The multiplier designs shown in [1–3] are the modifications done in the array or parallel multiplier which reduces the switching activity to minimize the dynamic power dissipation. However, such low power technique cannot eliminate leakage power.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_40, © Springer International Publishing Switzerland 2015

All the design techniques mentioned in [1–3] are especially for minimizing dynamic power dissipation. But as the technology is advancing towards a deep sub-micron level, leakage power is becoming dominant in the multiplier as well. Several leakage power minimization techniques such as logic shut down technique, multi-threshold voltage CMOS (MTCMOS) and block-wise shutdown techniques are developed. Many conventional shutdown techniques in electronic devices are introduced to save leakage power dissipation. However, there is unnecessary wastage of energy when such systems have to decide when to shut down or start the system. The newer approach of the predictive shutdown technique shuts down the system by analyzing the computation history which makes the system more power efficient [4]. Multi-threshold CMOS was introduced in which high threshold voltage transistor is connected in series with the logic of low threshold voltage transistors [5]. Whenever the circuit is idle, high threshold voltage transistor is turned off creating virtual ground to minimize the leakage current [5]. Generally, N-type metal-oxide-semiconductor (NMOS) are mostly preferred as a sleep transistor since it has high ON-resistance and also can be implemented in smaller size than that of the P-type metal-oxide-semiconductor (PMOS) transistors. Several approaches in the MTCMOS circuits were introduced such as sleep transistor sizing methodology to improve the performance level by putting delay constraints in each functional block [6]. Booth multiplier is also one of the most power efficient multiplier [7]. In [7], the pipeline registers are inserted in three different stages to decrease the path delay. However this technique can be more properly utilized if the block-wise shutdown technique is also applied. In [8], a 32 bit × 32 bit low power multiplier based on modified Booth algorithm is reported. It was implemented with three-stage pipelined block-wise shutdown technique. The multiplier was designed using top-down strategy. It was designed with Verilog HDL (hardware description language) and synthesized with 0.13 and 0.09 μm Taiwan Semiconductor Manufacturing Company Limited (TSMC) standard cell library. In reference to [8], we designed an 8 × 8 pipelined Booth multiplier with block-wise shutdown technique using bottom-up strategy. The Booth multiplier was designed in PSPICE schematic in transistor level. PSPICE power simulation is used to simulate the power consumption of the designed Booth multiplier for given input pattern sequence. The block-wise shut-down is verified with PSPICE simulation and it demonstrates effective power saving due to block-wise shut-down technique.

Block-Wise Shut-Down Pipelined Booth Multiplier Design

In reference to [8], we have designed the 8 × 8 pipelined Booth multiplier with block-wise shutdown technique in transistor level in PSPICE. The architecture of the pipelined Booth multiplier and the detail structure are shown in Fig. 1. The multiplier is divided into multiple pipeline stages. A CMOS transmission gate is used as a switch to control the voltage supply (V_{dd}) to each pipeline stage. Whenever the multiplier is idle, the switches turn off V_{dd} to each pipeline stage block by block. When active input is detected, the V_{dd} to each pipeline stage will be turned back on block by block. In this way, the leakage power of the multiplier can be totally eliminated without introducing large power line noise.

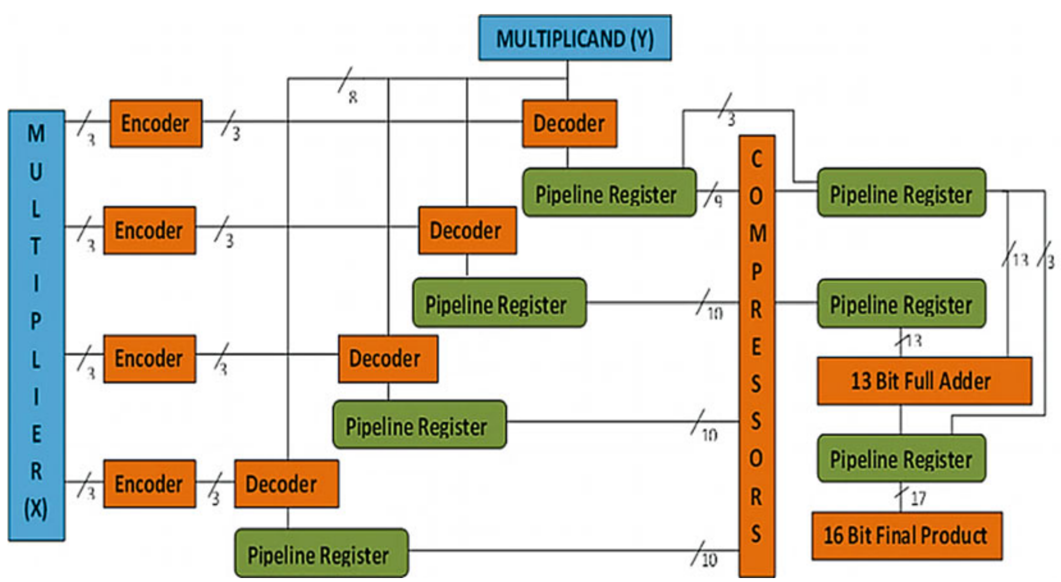

Fig. 1 Detail structure of the proposed schematic design

Booth Multiplier Algorithm

Booth Algorithm is a popular algorithm developed by Andrew Donald Booth in 1950. This algorithm is used to multiply two signed binary numbers in two's compliment notation [9]. It is known to be one of the fastest multiplication techniques with smaller circuit size. Every second column of the multiplier term is taken and is multiplied by ±1, ±2 or 0 such that a number of the partial product generated is half compared to conventional multiplication technique.

The multiplier term is booth recoded by dividing the terms into block of three bits by overlapping the blocks with one another as shown in Fig. 2. The division of block starts from the least significant bit of the multiplier term. For the first block, least significant bit is assumed to be always zero. According to the block bits, the partial product generating strategy is shown in Table 1 [9].

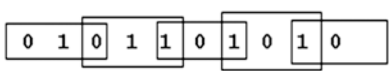

Fig. 2 Division of block starting from least significant bit of the multiplier term [9]

Table 1 The partial product generating strategy using booth algorithm technique [9]

Block	Partial product
000	0
001	1 * Multiplicand
010	1 * Multiplicand
011	2 * Multiplicand
100	−2 * Multiplicand
101	−1 * Multiplicand
110	−1 * Multiplicand
111	0 * Multiplicand

Sign Extension

When a partial product is generated in the signed multiplication, it is required to be sign extended to the most significant bit (MSB) position. In [10], sign extension prevention scheme is shown.The MSB of the first partial product row is repeated once. Then the MSB of each partial product row is inverted to increase one more bit and "1" is added in front of each partial product row except the first. The "Neg" output has to be added to the corresponding partial product row as shown in Fig. 3.

Architecture of the Block-Wise Shutdown Pipelined Booth Multiplier

The block-wise shut down pipelined booth multiplier consists of transmission gate switches, 3-bit encoders/decoders, the compressors, the final full adder and the pipeline registers. There are three transmission gate switches which turn on and off in the sequential manner according to the control signal. The three pipeline registers divide the whole multiplier block into three stages. First stage consists of the encoder and the decoder block. Altogether four 3-bit encoder blocks and four 9-bit decoder blocks are implemented. In this stage, four rows of the 10 bit partial product is generated. In the second stage, three 3:2 compressors, three half adders and six 4:2 compressors are used to add the partial product generated in the first stage such that there are only "sum" and "carry" bits. These sum and carry bits are then added separately by using 13-bit full adder in the third pipeline stage.

PSPICE Design of Booth Multiplier

In this paper, we implemented the block-wise shut-down technique [8] using an 8 × 8 pipelined Booth multiplier as an example. The pipelined Booth multiplier is designed and simulated in PSPICE using hierarchy design. The block-wise

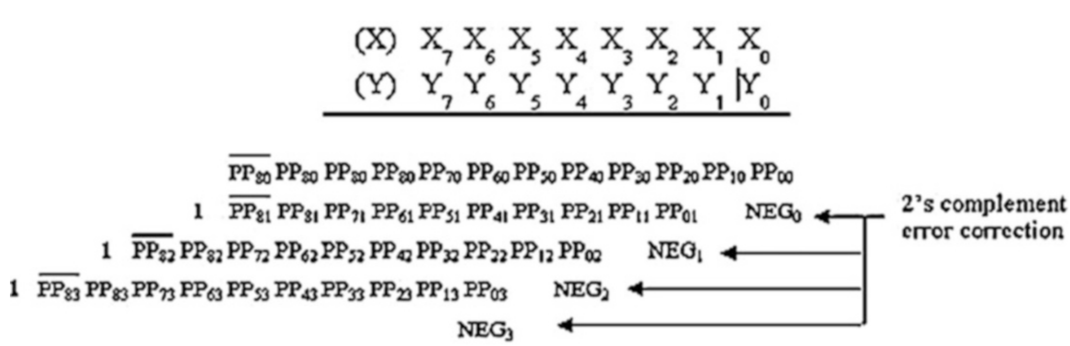

Fig. 3 Simplified sign extension method [10]

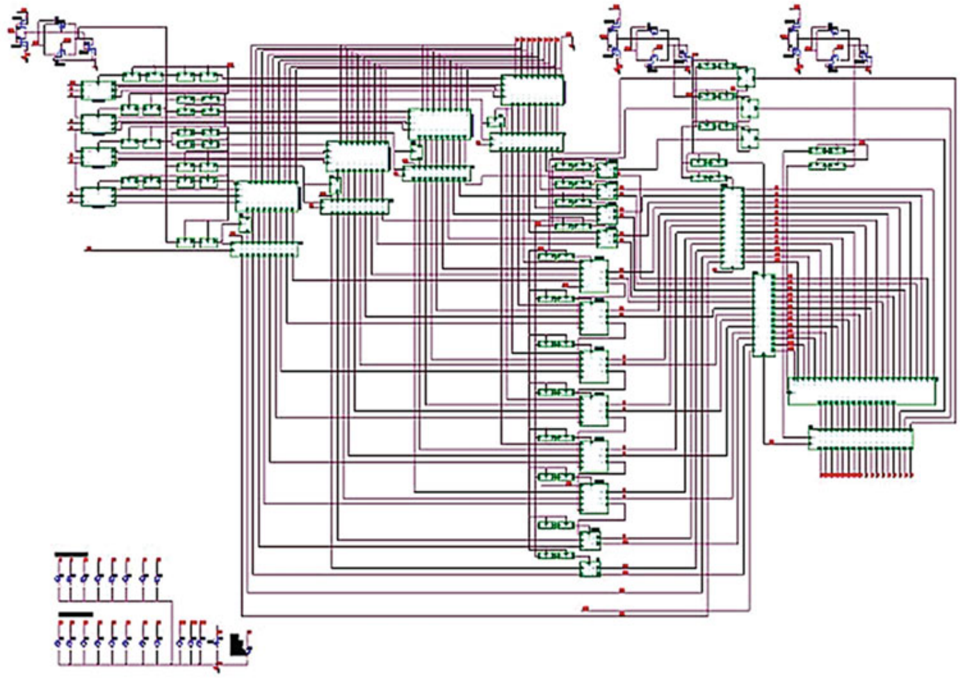

Fig. 4 Schematic design of the main circuit of the block-wise shutdown pipelined booth multiplier

shut-down technique is implemented in the designed Booth multiplier. Simulation results verify the correct function of the multiplier, as well as its block-wise shut-down to eliminate the leakage power when the circuit is idle. As an effective low-power technique, the block-wise shut-down technique can be extended to any other CMOS VLSI as well.

The top-level schematic design of the main circuit of the block-wise shutdown pipelined booth multiplier is shown in Fig. 4. The multiplier, multiplicand and the control signals are applied using "Piecewise Linear Voltage" source as shown in the figure below. The DC voltage of 5 V is supplied. Clock cycle is applied using pulse signal. The maximum voltage of 5 V and the minimum of 0 V are applied. The clock period is set to 1,000 ns and pulse width is set as 500 ns. Rise time and fall time is set to 0.1 ns. The PMOS transistor size is set as $(W/L)_p = 9\lambda/2\lambda$ and NMOS transistor size is set as$(W/L)_n = 3\lambda/2\lambda$.

The schematic design of the single booth encoder block is shown in Fig. 5. Two inverters are then inserted between the X_{2i+1} and "Neg" output to avoid any processing confliction.

Totally four encoder blocks are used in our design. The schematic of the complete encoder block is shown in Fig. 6. The multiplier bits divided in group of three are serially fed to the four encoder blocks. The three output codes generated are passed to the decoder blocks. The hierarchy design is used to form the complete four blocks of encoder.

Schematic design of the single bit decoder circuit is shown in Fig. 7. The outputs from the encoder and the multiplicand are fed as the inputs to this block, generating the partial product. In the design, hierarchy block for the XOR, OR and NAND gates have been used.

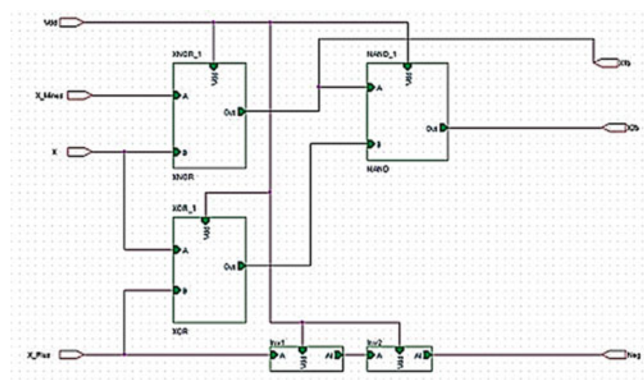

Fig. 5 Schematic design of single bit encoder

The complete 9-bit decoder block with cascaded half adder is shown in Fig. 8. Hierarchy block of the decoder and the half adder unit is used in the complete decoder design.

In our design, there are four 9-bit decoder blocks which generate four rows of partial products. These partial products are then sign-extended. Figure 9 shows the encoder and the decoder blocks with sign extension.

The partial product adding operations are carried out using different compressors, as shown in Fig. 10. Instead of using only the conventional 3:2 compressor, 4:2 compressors are also used to increase the speed. The 3:2 compressors are simply a full adder. 4:2 compressors are constructed by cascading two 3:2 compressors. To implement the 3:2 compressors, the hierarchy block of the XOR gate and the Multiplexer are used [11]. The structure is like a three input adders generating the "sum" and "carry" output. The 4:2 compressors are designed by cascading two hierarchy blocks of 3:2 compressors [11]. The propagation delay for 4:2 compressors are more than that of 3:2 compressors.

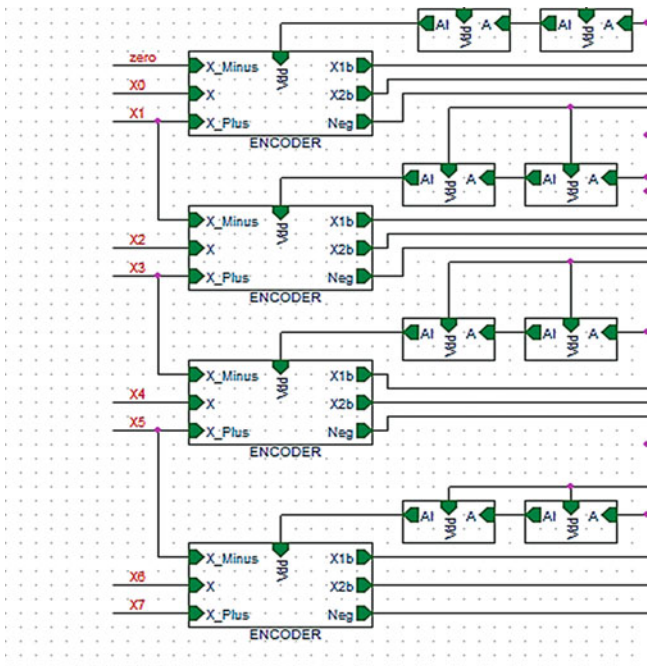

Fig. 6 Schematic design of complete encoder

Thus, we have increased the clock cycle in our simulation to accommodate this propagation delay.

We have applied the block-wise shutdown technique in the regular booth multiplier. In order to shut down the multiplier block sequentially, transmission gate switch is designed. The schematic design of the switch is shown in Fig. 11. In the switch, we used PMOS and NMOS transistors with very large sizes (W_p = 1,800 μm and W_n = 600 μm). To prevent the slow cutoff process, large transistors are being used. In the actual layout design, interdigitated layout should be used for such large transistors for compact design. As a level restorer, NMOS transistor is connected between switch output and the ground as shown in the figure, so that logic low can actually go down to 0 V. To reduce delay, the output of transmission gate is passed through a buffer before going to the V_{dd} of the hierarchy block.

In order to implement block-wise shut-down for three pipeline stages of the Booth multiplier, three transmission gate switches (with control signals Ctrl1 ~ Ctrl3) are used to turn ON and OFF the V_{dd} power source for each pipeline stage. These switches are turned ON and OFF sequentially to avoid large power line noise due to transient current generated in turning these switches on/off. Whenever a certain block is ideal, the switch for that particular block can be turned off in order to prevent the leakage power dissipation. However, before these switches are turned off, certain extra time should be given for the signal of this

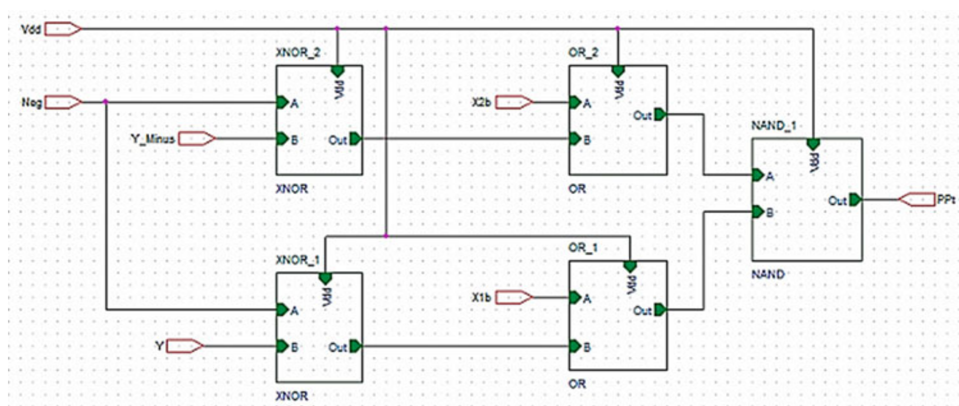

Fig. 7 Schematic design of single bit decoder block

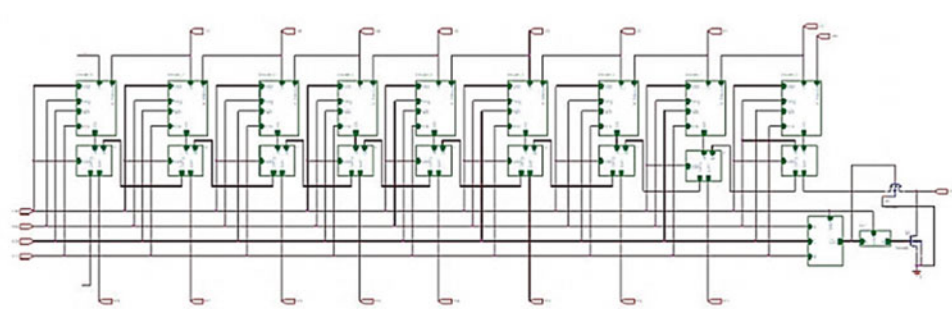

Fig. 8 Schematic design of 9 bit decoder block

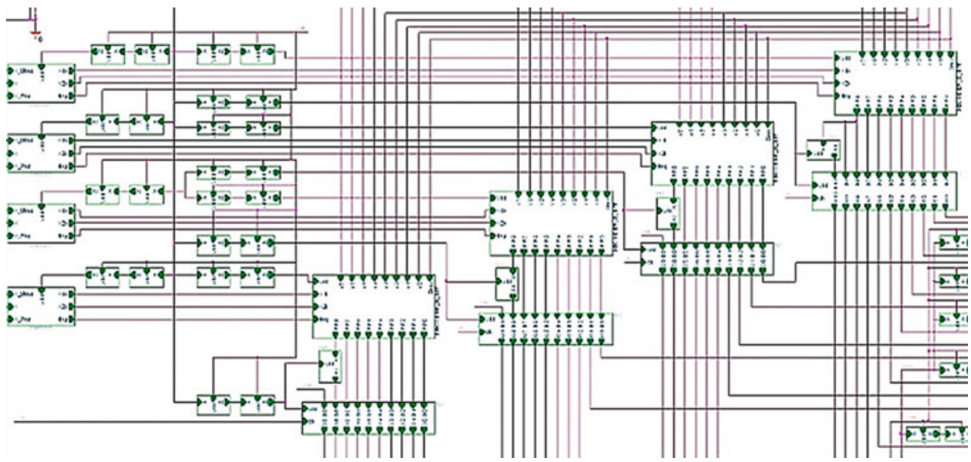

Fig. 9 Schematic design of complete encoder and decoder blocks generating the partial product

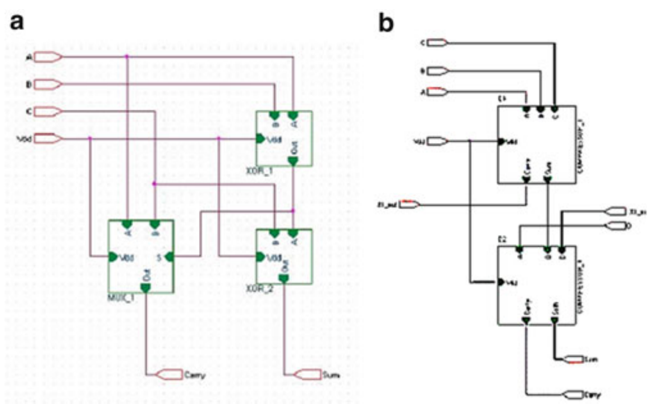

Fig. 10 Schematic design of compressors (**a**) 3:2 compressor (**b**) 4:2 compressor

pipeline stage to be passed to the next pipeline stage to ensure the correct outputs will be generated by the current input patterns.

Results and Discussion

Test Patterns and the Final Simulated Product of the Main Pipelined Booth Multiplier Circuit

The circuit is simulated using certain random patterns. Six test patterns are applied to simulate the circuit, as shown in Tables 1, 2, and 3. Each pattern is set to last for 1,000 ns. Among these six patterns, we assume that the second and third pattern are not useful patterns. Thus during this period the multiplier is ideal and we can shut down the multiplier block-wise using CMOS transmission gate switches so that power can be saved (Table 4).

The clock period is set as 1,000 ns. Since there are three pipeline stages, the expected output for the first input pattern does not show up immediately at the same clock cycle. Instead, it becomes available at the third rising edge of clock ($t = 2{,}500$ ns) considering the delay due to pipeline stages. The expected output for the remaining patterns can be observed in the subsequent cycles. In our case, we need to observe outputs corresponding to the first, fourth, fifth and sixth input patterns, but discard the second and third patterns for which the CMOS transmission gate switches are turned "off" accordingly. Three different control signals (Ctrl1 ~ Ctrl3) are used to control these switches individually. For example, let's consider the time interval of 0–3,000 ns. Second and third patterns are assumed to be invalid thus is not required to be processed. So "Ctrl1" is assigned a value "1" from 0 to 2,000 ns only then it is turned off. Also since the expected output for the first pattern can be observed only at third rising edge of the clock, i.e. at 2,500 ns, the second and third switch are turned on only at second and third clock cycles. Thus, each block is turned "on" sequentially decreasing the power line noise.

Figure 12 shows the final product simulated for the test patterns. The result is observed in the rising edge of clock starting from 2,500 ns. From Table 5, we observed that for each test pattern, the expected correct product outputs are obtained. Though the output obtained is 17 bit, we are discarding the MSB. The output is actually 16 bit. The simulated final product obtained for each pattern is as expected. Here, the final product is observed for first, fourth, fifth and sixth patterns. The test pattern that we have used is only the positive binary numbers.

PSPICE Power Simulation

Once the simulated final product has been verified, the net list of the design is extracted and is used to determine the power consumption using PSPICE power simulation. The current-controlled-current source coefficient in the power measurement auxiliary circuit can be calculated as:

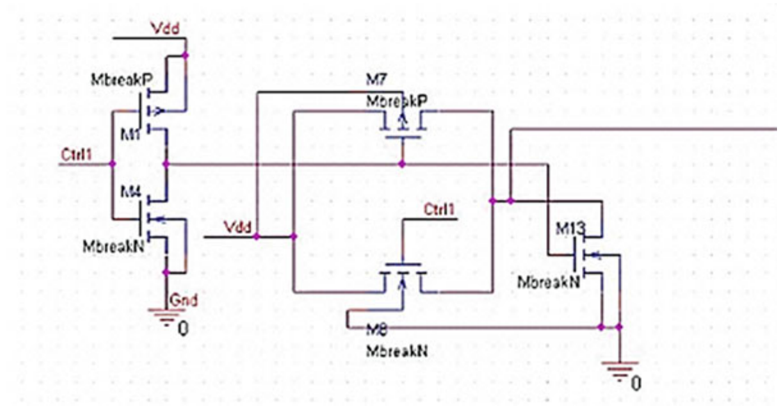

Fig. 11 Schematic design of switch

Table 2 Test patterns of multiplicand Y

Inputs	0–1 μs	1.0001–2 μs	2.0001–3 μs	3.0001–4 μs	4.0001–5 μs	5.0001–6 μs
Y7	0	0	0	0	0	0
Y6	1	0	1	0	1	1
Y5	1	0	1	0	1	1
Y4	1	0	1	1	0	1
Y3	0	0	1	1	0	0
Y2	1	0	1	0	0	1
Y1	1	0	1	1	1	0
Y0	1	0	1	0	0	1

Table 3 Test patterns of multiplier X

Inputs	0–1 μs	1.0001–2 μs	2.0001–3 μs	3.0001–4 μs	4.0001–5 μs	5.0001–6 μs
X7	0	0	0	0	0	0
X6	1	0	0	1	0	1
X5	1	0	0	0	1	1
X4	1	0	0	0	1	0
X3	0	0	0	1	1	0
X2	0	0	0	1	0	1
X1	1	0	0	0	1	1
X0	1	0	0	1	1	0

Table 4 Control signal test patterns

Inputs	0–1 μs	1.0001–2 μs	2.0001–3 μs	3.0001–4 μs	4.0001–5 μs	5.0001–6 μs	6.0001–7 μs	7.0001–8 μs	8.0001–9 μs
Ctrl1	1	1	0	1	1	1	1	0	0
Ctrl2	0	1	1	0	1	1	1	1	0
Ctrl3	0	0	1	1	0	1	1	1	1
Output			First			Fourth	Fifth	Sixth	

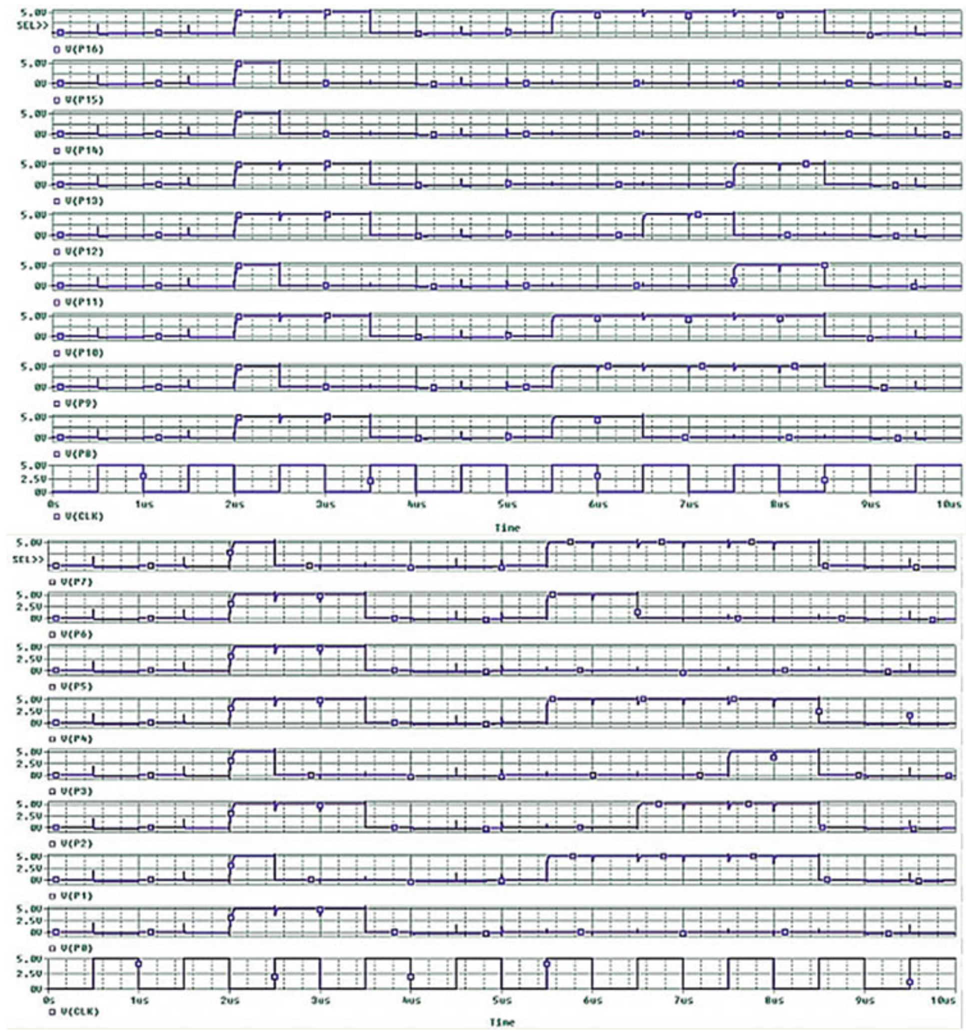

Fig. 12 Final simulated waveform in PSPICE

$$K = (V_{dd} * C)/T$$
$$= \left(5 \times 100 \times 10^{-12}\right)/\left(10{,}000 \times 10^{-9}\right) = 0.00005 \quad (1)$$

From the plot shown in Fig. 13 at t = 10 μs, V(P_{av}) = 1.0627 mV, thus the average power of the multiplier during time period of t = 0 ~ 10 μs is:

$$P_{avg} = 1.0627 \ \text{mW} \quad (2)$$

From Figs. 13 and 14, we observe that in the interval 6,000 ~ 10,000 ns the power consumption rate is not increasing in larger extent, as we can see each switch is turning OFF one at a time thus completely shutting down the unused block of the circuit. For example, let's consider the time interval of 7–10 μs. At t = 8 μs power consumed is slightly less than that at t = 7 μs because at t = 8 μs two unused block of circuit is completely shut down with the switches. While in the interval of 9,000 ~ 10,000 ns when all the switches are turned off, the power consumption of the circuit is almost zero. The small power curve increase is due to the power consumption of switch circuits. This verifies that the block-wise logic shut down does save power for the Booth multiplier especially when the circuit remains unused for the longer interval of time.

Conclusions and Future Work

Block-wise logic shut down reported by other researchers [8] is an effective low power technique to eliminate both dynamic power and static power when circuit is idle.

Table 5 Simulated and expected output patterns

(a) First pattern	
Multiplicand (Y)	01110111
Multiplier (X)	01110011
Expected Product	0011010101110101
Simulated Product	0011010101110101
O/P as expected?	Yes
(b) Fourth pattern	
Multiplicand (Y)	00011010
Multiplier (X)	01001101
Expected Product	0000011111010010
Simulated Product	0000011111010010
O/P as expected?	Yes
(c) Fifth pattern	
Multiplicand (Y)	01100010
Multiplier (X)	00111011
Expected Product	0001011010010110
Simulated Product	0001011010010110
O/P as expected?	Yes
(d) Sixth pattern	
Multiplicand (Y)	01110101
Multiplier (X)	01100110
Expected Product	0010111010011110
Simulated Product	0010111010011110
O/P as expected?	Yes

Compared to whole circuit shut-down, block-wise shutdown can avoid large transient current hence reduce circuit glitches. In this paper, we have designed an 8 × 8 pipelined Booth multiplier with block-wise shutdown technique in PSPICE using bottom-up strategy. Compared to the top-down design implementation in [8], bottom-up design in transistor level allows us to customize the design for optimized performance. Through transistor/gate sizing, combining different logic family and various other techniques, we can improve circuit speed, reduce power consumption and minimize transistor count of the circuit. The transistor design also allows us to evaluate the power consumption of the circuit with PSPICE power simulation for given input pattern sequence. The Booth multiplier has been designed in PSPICE schematic and its correct function was verified by PSPICE simulation. Power simulation demonstrates effective power saving of the designed Booth multiplier for given input pattern sequence. In the future, we will further extend this technique to other CMOS VLSI (e.g. ALU) design as well.

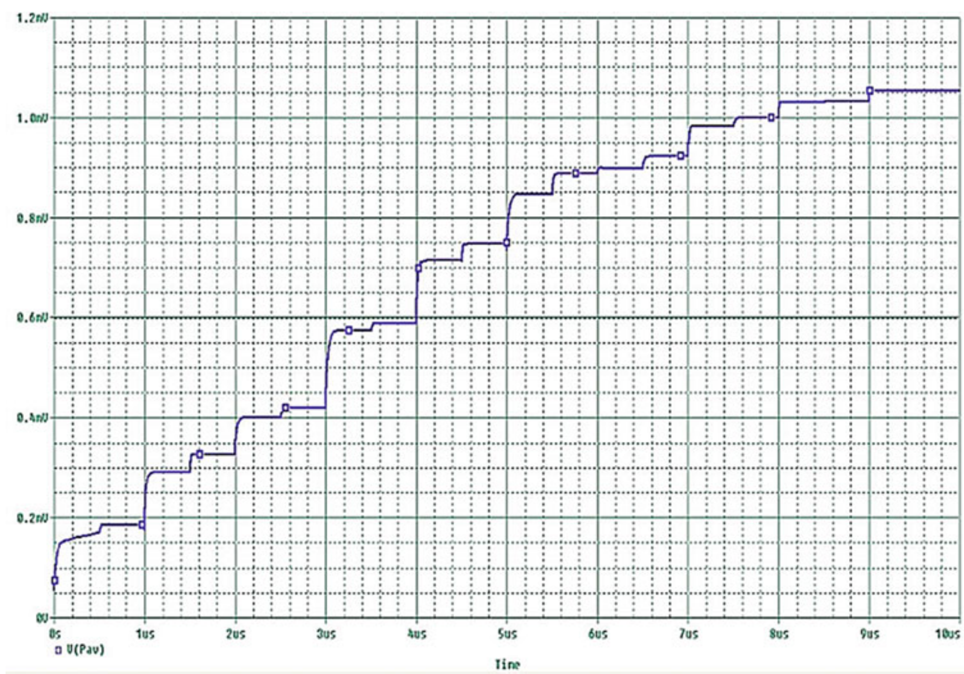

Fig. 13 Power curve of block-wise shutdown pipelined booth multiplier

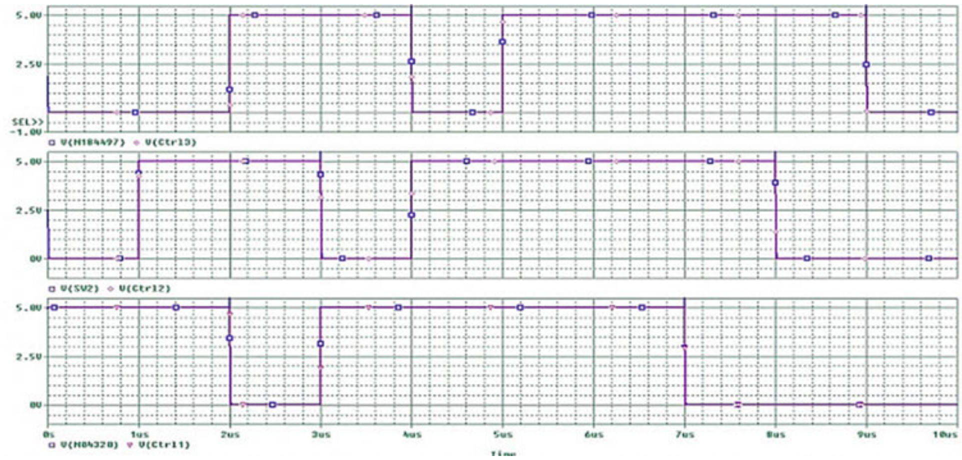

Fig. 14 Transmission gate output and the control signal

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Trust Models in Cloud: A Survey on Pros and Cons

Usha Divakarla and K. Chandra Sekaran

Abstract

Cloud is the recent emerging technology in all aspects. The basic concern with the usage of this cloud technology is security. Security poses a major drawback with data storage, resource utilization, virtualization, etc. In the highly competitive environment the assurances are insufficient for the customers to identify the trust worthy cloud service providers. In a nut shell all the entities in cloud and cloud computing environment should be trusted by each other and the entities that have communication should be trusted by each other. This paper throws light on different Trust Models developed and their drawback with respect to resource security. A strong Trust Model is recommended to enhance the security of the resources in Cloud.

Keywords

Trust aspects • Trust models • Drawbacks of trust models

Introduction

The major factor of any human interaction is Trust. Though this trust has no specific meaning researchers coin their own meaning according to the scenario. In psychology trust is defined as belief. The more the degree of belief the more the trust on that entity. In technology trust has no definite meaning as-well. In is defined as the degree of trustworthiness. If an entity is to be trusted then the degree of trustworthiness must be more. The less the trustworthiness the more is the risk to the system. Trust is often measured/related with terms like co-operation, confidence and predictability. According to Gambetta [1] trust is the probability that an entity will perform an action that is beneficial or at least not detrimental to us is high enough for us to consider engaging in some form of co-operation with it.

In the current world competition increases the risk of untrust worthiness where one entity does not trust the other entity in the system. To overcome this risk of untrustedness a strong trust evaluation scheme is need though many models do exist.

Trust in Cloud Environment

Cloud is the emerging technology for the users to easily work with minimum effort and minimized cost. In cloud every service is tendered with as pay-as-use term. So users can use the cloud technology to maximize their profit with minimum cost and effort.

Given the quick adoption of cloud computing in the industry, there is a significant challenge in managing trust among cloud service providers and cloud service consumers. Recently, the significance of trust management is highly recognized and several solutions are proposed to assess and manage trust feedbacks collected from participants [26, 27]. However, one particular problem has been mostly neglected: to what extent can these trust feedbacks be credible. Trust management systems usually experience malicious behaviors from its users. On the other hand, the quality of trust feedbacks differs from

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_41, © Springer International Publishing Switzerland 2015

one person to another, depending on how experienced he/she is.

The major components that are affected by the barriers in cloud trust are [28]:

(1) **Security**—Mechanisms (e.g. encryption) which make it difficult or uneconomical for an unauthorized person to access some information.
(2) **Privacy**—Protection against the exposure or leakage of personal or confidential data (e.g. personally identifiable information (PII)).
(3) **Accountability**—Defined in [29] as the obligation and/or willingness to demonstrate and take responsibility for performance in light of agreed-upon expectations. Accountability goes beyond responsibility by obligating an organization to be answerable for its actions.
(4) **Auditability**—The relative ease of auditing a system or an environment. Poor auditability means that the system has poorly-maintained (or non-existent) records and systems that enable efficient auditing of processes within the cloud.

The various aspects on which trust depends for evaluation are discussed in next section.

Various Aspects of Trust

Trust being the old notion there exists many definitions according to the researchers and their scenarios. Though trust is belief in psychology or Trust value in technology, to reach at the specified threshold value various aspects are to be considered. Some of the aspects of the Trust are listed below.

Characteristics of Trust

The common characteristics used in trust definition are:

(1) **Benevolence**: means care and motivation to act for somebody else's goodness [22].
(2) **Integrity**: concluding an agreement with benevolence [23].
(3) **Competence**: skills or capabilities necessary to do what we need to do.
(4) **Predictability**: actions of partners that are consistent enough to predict future situations [24].

Trust Principles

To arrive at the maximum trust of any entity some basic principles are followed.

Some of the trust principles traced are:

(1) **Trust Transitivity**: if A trusts B and B trusts C then A can derive trust on C using the B trust referral.
(2) **Trust is a function of risk perception**: as trust is a belief on one person for his correct actions, so trust also evaluates the uncertainty that the other party will work upon.
(3) **Trust is determined by Time**: as trust is built on past experience, i.e. the data collected over time.
(4) **Trust can be quantified**: trust can be quantified in numeric world as between [0,1]
(5) **Culture influence trust**: trust depends on organizational and national culture.
(6) **Formal and Social tools for control are significant for Trust development**: based on the organizational and national culture trust can be formulated using different formal methods.

Factors Affecting Trust

Trust cannot be specified in a crisp value. To arrive at calculative value some basic factors always have an exclusive influence. Some of the few factors are listed below.

Some of the key factors that are common to trust irrespective of the platform are:

(1) Trust plays an important role in uncertainty and risky environment.
(2) Trust is the base platform on which certain decisions are made.
(3) Trust is built using prior knowledge and experience.
(4) Trust is a subjective notion based on opinion and values of an individual.
(5) Trust is dynamic and new knowledge and experience will be overriding over the old ones.
(6) Trust is context-dependent.
(7) Trust is multi-faceted.

Types of Trust

Trust can be defined or calculated in different ways based on the scenario it is used. There exists basically three types of Trust [25] used viz:

(1) **Interpersonal trust**: trust in a specific actor based on reliability.
(2) **Institutional trust**: based on capability or reliability or both.
(3) **System level trust**: Multiple, overlapping mechanisms like Institutional assurance mechanisms, Institutional organizations, roles, Economic incentives and reputation, Social norms and ethical values, Experience.

Trust can also be classified as online trust and offline trust. Offline trust involves offline activities of organization like direct sales, channel sales, etc. online trust involves the business activities of the organization in electronic medium. In offline trust object of trust is typically a human where as in online trust the technology itself is object of trust.

Various Trust Models in Cloud Environment

Multiple trust computing models are designed to evaluate trust in cloud. Some of them are listed below.

Khan and Malluhi [5] have looked at the trust in the cloud system from users' perspective. They analyze the issues of trust from what a cloud user would expect with respect to their data in terms of security and privacy. They further discuss that what kind of strategy the service providers may undertake to enhance the trust of the user in cloud services and providers. They have identified control, ownership, prevention and security as the key aspects that decide users' level of trust on services. Diminishing control and lack of transparency have identified as the issues that diminishes the user trust on cloud systems. The authors have predicted that remote access control facilities for resources of the users, transparency with respect to cloud providers actions in the form of automatic traceability facilities, certification of cloud security properties and capabilities through an independent certification authority and providing security enclave for users could be used to enhance the trust of users in the services and service providers.

Sato et al. [6] have proposed a trust model of cloud security in terms of social security. The authors have identified and named the specific security issue as social insecurity problem and try to handle it using a three pronged approach. They have subdivided the social insecurity problem in to three sub areas, namely; multiple stakeholder problem, open space security problem, and mission critical data handling problem. The multiple stakeholder problem addresses the security issues created due to the multiple parties interacting in the cloud system. As per the authors, three parties can be clearly identified. They are namely, the client, the cloud service providers, and third parties that include rivals and stakeholders in business. The client delegates some of the administration/operations to cloud providers under a Service Level Agreement (SLA). Even if the client would like to have the same type of policies that it would apply if the resources were hosted on site on the delegated resources, but the provider's policy may differ from that of the client. The providers are bound only by the SLA signed between the parties. The SLA plays the role of glue between the policies. Also the authors opine that once the data is put in the cloud it is open for access by third parties once authenticated by the cloud provider. The open space security problem addresses the issue of loss of control on where the data is stored and how they are physically managed once control of data is delegated to the cloud provider. They advise to encrypt the data before transferring data converting the data security problem to a key management problem as now the keys used for encryption/decryption must be handled properly. The mission critical data handling problem looks at the issue of delegating the control of mission critical data to a service provider. They advise not to delegate control of this data but to keep them in a private cloud in a hybrid setup, where the organizations have unhindered control. However setting up of a private cloud may not be an option to small and medium sized organizations due to the high costs involved. Hence enhancement of security of the public cloud is the only option to serve everybody. Authors have developed a trust model named 'cloud trust model' to address the problems raised above. Two more trust layers have been added to the conventional trust architecture. These layers have been named internal trust layer and contracted trust layer. Internal trust layer acts as the platform to build the entire trust architecture. Internal trust layer is installed in the in house facilities and hence under the control of the local administration. Identity and key management are handled under the internal trust. Also any data that is considered critical or needs extra security must be stored under this layer. Contracted trust has been defined as the trust enforced by an agreement. A cloud provider gives the trust to a client based on the contract that is made up of three documents known as Service Policy/Service Practice Statement (SP/SPS), Id Policy/Id Practice Statement (IdP/IdPS) and the contract. Level of trust required can be negotiated by parties depending on the level of security needed for the data. A cloud system thus installed is called a secure cloud by the authors.

Manuel et al. [10] have proposed trust model that is integrated with CARE resource broker. This trust model can support both grid and cloud systems. The model computes trust using three main components namely, Security Level Evaluator, Feedback Evaluator and Reputation Trust Evaluator. Security level evaluation has been carried out based on authentication type, authorization type and self-security competence mechanism. Multiple authentication, authorization mechanism and self-security competence mechanisms are supported. Depending on the strength of individual mechanism, different grades are provided for trust value. Feedback evaluation also goes through three different stages namely feedback collection, feedback verification and feedback updating. The reputation trust evaluator computes the trust values of the grid/cloud resources based on their capabilities based on computational parameters and network parameters. Finally the overall trust value has been computed taking the arithmetic sum of all the individual trust values computed.

Both Shen et al. and Shen and Tong [11, 12] have analyzed the security of cloud computing environment and described the function of trusted computing platform in cloud computing. They have also proposed a method to improve the security and dependability of cloud computing integrating the Trusted Computing Platform (TCP) into the cloud computing system. The TCP has been used in authentication, confidentiality and integrity in cloud computing environment. Finally the model has been developed as software middleware known as the Trusted Platform Software Stack (TSS).

Alhamad et al. [13] have proposed a SLA based trust model for cloud computing. The model consists of the SLA agents, cloud consumer module, and cloud services directory. The SLA agent is the core module of the architecture as it groups the consumers to classes based on their needs, designs SLA metrics, negotiates with cloud providers, selects the providers based on non-functional requirements such as QoS, and monitors the activities for the consumers and the SLA parameters. Cloud consumer module requests the external execution of one or more services. Cloud services directory is the one where the service providers can advertise their services and consumers seek to find the providers who meet their functional requirements such as database providers, hardware providers, application providers, etc.

A model called a multi-tenancy trusted computing environment model (MTCEM) for cloud computing has been proposed by Li et al. [14]. MTCEM has been proposed to deliver trusted IaaS to customers with a dual level transitive trust mechanism that supports a security duty separation function simultaneously. Since cloud facilities belong to multiple stakeholders such as Cloud Service Providers (CSP) and customers, they belong to multiple security domain and serve different security subjects simultaneously. The different stakeholders may be driven by different motives such as best service, maximization of the return on investment and hence may work detrimental to the other party involved. Hence cloud computing should have the capability to compartmentalize each customer and CSP and support security duty separation defining clear and seamless security responsibility boundaries for CSP and customers. MTCEM has been designed as two-level hierarchy transitive trust chain model which supports the security duty separation and supports three types of distinct stakeholders namely, CSP, customers and auditors. In this model, CSP assume the responsibilities to keep infrastructures trusted while the customer assumes responsibility starting from the guest OS which installed by the customer on the Virtual Machines provided by the CSP. The auditor monitors the services provided by the CSP on behalf of the customers. The authors have implemented a prototype system to prove that MTCEM is capable of being implemented on commercial hardware and software. But no evaluation of the prototype on performance has been presented.

Yang et al. [15] have studied the existing trust models and firewall technology. The authors have found that all the existing trust models ignore the existence of firewall in a network. Since firewall is an integral and important component of any corporate security architecture, this non-inclusion of firewall is a huge shortcoming. The authors have proposed a collaborative trust model of firewall-through based on Cloud theory. This paper also presents the detailed design calculations of the proposed trust model and practical algorithms of measuring and updating the value of dynamic trust.

Fu et al. [16] have studied the security issues associated with software running in the cloud and proposed a watermark-aware trusted running environment to protect the software running in the cloud. The proposed model is made up of two components namely the administrative center and the cloud server environment. The administrative center embeds watermark and customizes the Java Virtual Machines (JVM) and the specific trusted server platform includes a series of cloud servers deployed with the customized JVMs. Only specific and complete Java programs are allowed to run on the JVMs while rejecting all the unauthorized programs like invasion programs. The main advantage of this approach is that it introduces watermark aware running environment to cloud computing.

Ranchal et al. [17] have studied the identity management in cloud computing and proposed a system without the involvement of a trusted third party. The proposed system that is based on the use of predicates over encrypted data and multi-party computing is not only capable of using trusted hosts but also untrusted hosts in the cloud.

Takabi et al. [18] have proposed a security framework for cloud computing consisting of different modules to handle security, and trust issues of key components. The main issues discussed in the paper are identity management, access control, policy integration among multiple clouds, trust management between different clouds and between cloud providers and users. The framework identifies three main players in the cloud. They are cloud customers, service integrators and service providers. The service integrator plays the role of the mediator who brings the customers and service providers together. Service integrator facilitates collaboration among different service providers by composing services to meet the customer requirements. It is the responsibility of the service integrator to establish and maintain trust between provider domains and providers and customers. The service integrator discovers the services from service providers or other service integrators negotiate and integrate services to form collaborating services that will be sold to customers. The service integrator module is composed of security management module, trust management module, service management module and heterogeneity management module. The heterogeneity management module manages the heterogeneity among the service providers.

Li et al. [7] propose a domain-based trust model to ensure the security and interoperability of cloud and cross-cloud environment and a security framework with an independent trust management module on top of traditional security modules. They also put forward some trust based security strategies for the safety of both cloud customers and providers based on this security model.

The family gene based cloud trust model [8, 9] that is fundamentally different from the Public key Infrastructure based trust models have been proposed by several researchers. These researchers have studied the basic operations such as user authentication, authorization management and access control, and proposed a Family-gene Based model for Cloud Trust (FBCT) integrating these operations.

Thus the literature survey throws light on the different trust models proposed to evaluate trust in distributed/cloud environment for various platforms.

Drawbacks of Existing Trust Models

Though various Trust Models are developed to solve the trust issue still trust is a major concern. An extensive literature survey reveals some of the drawbacks found in the various trust models explained in the above section. The issues are listed as below:

Sato et al. [6] have proposed a cloud trust model which adopts two layers of trust called internal trust layer and contracted trust layer. In internal trust layer the trust is based on the internal trust of the organization and the trust of resources is based on this internal trust. In contracted trust layer the trust is based upon the contract of the CSP and user identity. Both the layers though provide trust in layered way but the trust calculated is internal to the organization. The CSP has nothing to do with the security of the resources. So the organization has to have a private cloud to secure its data which is not possible with small/medium organizations.

Alhamad et al. [13] have proposed SLA based trust model only and no implementation or evaluation has been developed or described. Hence each and every module will have to be evaluated for their functionality and the effectiveness and finally the overall model will have to be evaluated for its effectiveness. This model is reputation based trust that has a disadvantage that user with high scores for reputation can cheat user in few transactions even though they receive negative feedback. This model has a centralized architecture, so all the services and reputation information has single point of failure.

Shen et al. and Shen and Tong [11, 12] have proposed trusted computing technology for trust evaluation. The basic disadvantage of this model is that the underlying architecture is based on Trusted Computing Platform [TCP] which is difficult to integrate with cloud computing with respect to hardware.

In Role Based Trust model the trust is based on the roles, ID used for TCP, standard certificate for assurance. The hardware maintains a master key for each machine and it uses master key to generate unique sub key for every configuration of the machine. The data encrypted for one configuration cannot be decrypted in another configuration of same machine. If the configuration of the machine changes the session key of the local machine will not be useful.

In CARE [10] resource model conventional scheduling is done through FIFO. So computation/process starve for the necessary resources. The priority of resources for the critical jobs is not taken care.

In Family Gene based trust models [8, 9] the trust model is just proposed for authentication and is tested by simulation. The model does not deal with security aspects either of data or of resources. A real time implementation is not done.

The Active Bundle Scheme [17] proposed based on Identity Management model approach is independent of a third party, it is less prone to attack as it reduces the risk of correlation attacks and side channel attacks, but it is prone to denial of service as active bundle may also be not executed at all in the remote host

The consolidated summary of the various trust models is summarized in Table 1. The table throws light on the various trust models and their performance with respect to different parameters.

Though a lot of work is done in trust area still no researcher has proposed any trust model for trusting the resources in the cloud. As resources are the entities in cloud the security of these entities is very essential. Though the security as whole is taken care by the service provider still security of the resources in cloud is at stake. It is noted from the literature review that trust plays an important part of the security in cloud, but trust as whole in terms of services provided is taken into account. When an entity enters the cloud the trust is calculated by the service provider in accordance with the other resources. The above table though explains the various trust models proposed by various researchers with specific to some of the parameters, does not deal/specify about trust with respect to resources.

Table 1 Comparative summary of the previous work done in trust in cloud

Authors	Type	Authentication	Data security	Heterogeneous systems support	Remarks
[5]	Establishing trust in cloud	Discussed	Discussed	Yes	Discussed the issues
[6]	Social security based	Discussed	Discussed	No	Discussed the issues
[8, 9]	Family gene based	Discussed	No	No	Proposed Model is tested using simulation
[10]	Integrated with CARE Resource Broker	Yes	Yes	Yes	Proposed Model is tested using simulation
[11, 12]	Built on trusted platform service	Yes	Yes	Yes	Only a model has been proposed
[14]	Built on trusted computing platform	No	No	No	Concept has been proved with a prototype
[15]	A collaborative trust module of Firewall-through	No	No	Yes	Model has been tested using simulation
[16]	Watermark based security	No	No	No	Concept has been proved with a prototype
[17]	Based on active bundles scheme	Yes	No	Yes	Concept has been proved with a prototype

Conclusion

Though trust is not defined in true terms, being an integral part of any system it is more of concern. Currently the trust is based on Service Level Agreement (SLA) that is negotiated between the service provider and the customer. The SLA does not guarantee the security of the resources/applications/data deployed in cloud but it only assures of backup in case of loss/theft of the data. Many Trust Models are developed which try to solve the trust problem in the system and their drawbacks are discussed. Due to the draw back a strong Trust model is to be developed which would enhance the security of the entities in the system thus making the Cloud more secure with respect to storage, computation and efficient usage.

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Face Recognition Security System

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Abstract

Today's institutions are facing major security issues; consequently, they need several specially trained personnel to attain the desired security. These personnel, as human beings, make mistakes that might affect the level of security.

A proposed solution to the aforementioned matter is a Face Recognition Security System, which can detect intruders to restricted or high-security areas, and help in minimizing human error. This system is composed of two parts: hardware part and software part. The hardware part consists of a camera, while the software part consists of face-detection and face-recognition algorithms software. When a person enters to the zone in question, a series of snapshots are taken by the camera and sent to the software to be analyzed and compared with an existing database of trusted people. An alarm goes off if the user is not recognized.

Keywords

Digital image processing • Face detection • Face recognition • Biometrics

Introduction

Real time face recognition is part of the field of biometrics. Biometrics is the ability for a computer to recognize a human through a unique physical trait. Face recognition provides the capability for the computer to recognize a human by facial characteristics. Today, biometrics is one of the fastest growing fields in advanced technology. Predictions indicate a biometrics explosion in the next century, to authenticate identities and avoid and unauthorized access to networks, database and facilities.

A facial recognition device is a device that takes an image or a video of a human face and compares it to other image faces in a database. The structure, shape and proportions of the faces are compared during the face recognition steps. In addition, distance between the eyes, nose, mouth and jaw, upper outlines of the eye sockets, the sides of the mouth, location of the nose and eyes, and the area surrounding the check bones are also compared [1].

When using a facial recognition program, several pictures of the person must be taken at different angles and with different facial expressions [1]. At time of verification and identification the subject stands in front of the camera for a few seconds, and then the image is compared to those that have been previously recorded.

Facial recognition is widely used because of its benefits. The advantages of facial recognition are that it is not intrusive, can be done from a faraway distance even without the person being aware that he/she is being scanned [1]. Such thing is needed in banks or government offices for example, and this is what makes facial recognition systems better than other biometric techniques in that they can be used for surveillance purposes like searching for wanted criminals, suspected terrorists, or missing children.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_42, © Springer International Publishing Switzerland 2015

Face recognition devices are most beneficial to use for facial authentication than for identification purposes, because it is easy to alter someone's face, and because the person can disguise using a mask. Environment is also a consideration as well as subject motion and focus on the camera [1].

Facial recognition, when used in combination with another biometric method, can improve verification and identification results dramatically.

Existing Solutions

Many face recognition software have been implemented during the past decade. Each software uses different methods and different algorithm than other software. Some facial recognition software extracts the face features from the input image to identify the face [1]. Other algorithms normalize a set of face images and then compress the face data, the saves the data in one image that can be used for facial recognition. The input image is compared with the face data [2].

New method for face recognition is being used which is the three-dimensional facial recognition. In this method, a 3-D sensor is used to capture information about the shape of the face so that only distinctive features of the face, such as the contour of eye sockets, nose and chin, are used for face recognition [1]. This new method offers some advantages over other algorithms in that recognition it is not affected by the change of light, and the face can be identified from a variety of angles, including profile view.

Another new technique in facial recognition is called skin texture analysis. This technique uses the visual details of the skin, as captured in standard digital or scanned images and then turns the unique lines, patterns, and spots apparent in a person's skin into a mathematical space.

Below is an introduction for some of the existing facial recognition programs that were used for security reasons.

A. *FaceFirst*
FaceFirst is a software that provides a fully automated, user friendly, turnkey mobile and live-video surveillance facial recognition system. This software generates an alert whenever a face is recognized; and this occurs when the match of the input face with a face in the database is above a user defined probability. The advantage of FaceFirst system is the availability to work in low resolution environments enabling real-world performance [1].

B. *MorphoTrak*
MorphoTrak provides biometric and identity management solutions to a broad array of markets including law enforcement, border control, driver licenses, civil identification, and facility/IT security. MorphoTrak is part of the world's largest biometric company and leading innovator in large fingerprint identification systems, facial and iris recognition, as well as secure credentials [1].

C. *Cross Match Technologies*
Cross Match Technologies is a leading global provider of biometric identity management systems, applications and enabling technologies to governments, law enforcement agencies and businesses around the world. Offerings include biometric technologies capable of wireless, mobile or stationary use that encompass facial recognition systems and other systems [1].

Approaches for Face Recognition

There are many difficulties related to human facial recognition. The fact that human faces are all relatively similar, yet produce varying facial expressions makes it more difficult to generalize an algorithm. Except in the case of identical twins, the face is arguably a person's most unique physical characteristic. Each face has certain distinguishable facial features. These are the peaks and the valleys that make up the different facial features. Lighting conditions and the angle from which the facial image is taken are other factors to consider. Taking all this into account, it is important to note that humans themselves can distinguish a multitude of different faces quickly and with high accuracy [3]. The facial recognition software is based on the ability to first recognize faces, which are a technological feat in itself, and then measure the various features of each face [2].

The planned testing approach is to have a database of numerous faces that is used to test the recognition algorithm against certain particular faces. The variables that need to be tested for a face against the database include the size and condition variations, illumination changes, different facial expressions, and the angle from which the image is taken. The approach is similar to what may be done in real world applications, where a facial image is acquired, but not necessarily in ideal conditions and needs to be matched against a database of somewhat ideal facial images. There are many applications that this algorithm could be used for, such as surveillance and security systems. Face recognition is a widespread technology used for Access Control. The task is stated as follows. There is a group of authorized people, which a recognition system must accept. All the other people are unauthorized or 'alien' and should be rejected. Security identity, whether in the physical or virtual world, has always been a business critical issue for the world's leading organizations. Whether access to property, to valuable IP on corporate networks or simply proving your identity-adequate and robust security is essential [2].

Three main tasks of face recognition may be named: “document control”, “access control”, and “database retrieval”. The term “document control” means the verification of a human by comparison his/her actual camera image with a document photo. Access control is the most investigated task in the field. Such systems compare the portrait of a tested person with photos of people who have access permissions to joint used object. The last arises when it is necessary to determine name and other information about a person just based on his/her one casual photo. Because of great difference between the tasks there is not a universal approach or algorithm for face recognition.

Proposed Solution

The proposed solution is a real-time face recognition system that reads a video from a camera connected to the computer running the software, detects any face present in front of the camera, and then checks if this face is present in a set of face images in a database using face recognition technique. The software is divided into two parts: face detection and face recognition.

Algorithm Suggested for Face Detection

First, the image is taken as an input into the software. The program then converts it from its color mode (RGB) into gray-scale then resizes it. An edge detection operation is then applied by calculating the gradient of the image. To calculate the gradient of an image, the Sobel operator is used, which creates a binary mask using a user-specified threshold value. The threshold value is determined by getting the mean of all gray values in the image. The next step is a dilation operation in order to make the borderlines thicker, followed by a filling technique in order to fill the hole in the face. After holes filling, erosions followed by dilations are applied to get rid of other smaller objects in the image and maintain the region of interest intact. Finally, the image is resized back to its original size.

1. *Filtering*
   After taking the initial picture, the median filter technique is applied on the image. The purpose of this technique is to eliminate the noise that will appear during the capturing, and to enhance the edging procedure. So when using other kind of filter it will affect the edging and an error will appear.
2. *Resising*
   After filtering the image, it is resized from $2048 \times 1536 \times 3$ (resolution of the camera) to $512 \times 384 \times 3$, because it easier to MATLAB to deal with small image and the processor will be faster.

Fig. 1 Image converted from RGB to HSV

3. *Color Mode Conversion and Skin Detection*
   The initial image taken from the camera is in RGB color mode which is not good to analyze, because the luminous and the cruminous are mixed together and then the removing of the important parameter will be difficult, so the picture will be converted to (HSV) that facilitate the process of taking the important parameters.
   HSV: stand for Hue, Saturation and value and also it is called HSB for brightness [4].
   H: It is an angle between (0 and 360) degree in general, in each cylinder the angle around the central vertical axis corresponding for ‘Hue’, but when there is a skin the range of angle must be satisfied between (0 and 50).
   S: The distance from the axis corresponds for ‘Saturation’, for skin must satisfied the interval $0.18 < S < 0.68$.
   V: The distance along the axis corresponds to lightness ‘Value’, and also for the skin must satisfied the following interval $0.35 < V < 1$.
   Figure 1 shows an image converted from RGB to HSV and Fig. 2 shows an image with the skin pixels detected and converted to white pixels while the non-skin pixels are transformed to black pixels.
4. *Morphological Operations*
   After doing all previous procedure, the image will be processed using morphological operations.
   (a) *Dilation*
   Dilation is the process of converting the black color which has the (value 0) near to the white one (value 1) into white color. Dilation is one of the two basic operators in the area of mathematical morphology (the other being erosion). It is typically applied to binary images, but there are versions that work on grayscale images. The basic effect of the operator on a binary image is to gradually enlarge the boundaries

Fig. 2 Skin detection

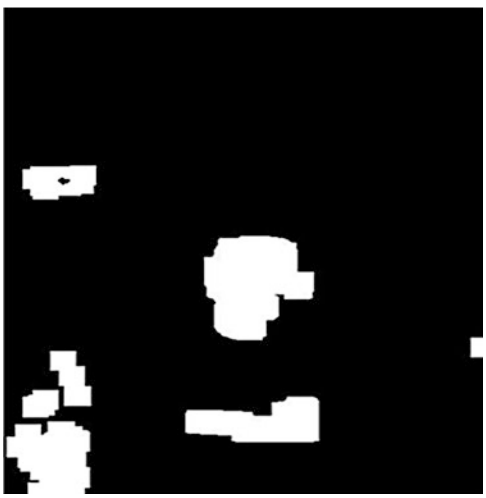

Fig. 3 Image dilation

of regions of foreground pixels (*i.e.* white pixels, typically). Thus areas of foreground pixels grow in size while holes within those regions become smaller. The dilation operator takes two pieces of data as inputs. The first is the image which is to be dilated. The second is a (usually small) set of coordinate points known as a structuring element. It is this structuring element that determines the precise effect of the dilation on the input image [5]. In MATLAB, you can use the built-in function imdilate which takes two arguments, the image and a structuring element and return the dilated image. To get the structuring element object, you can use the built-in function strel [6].

Figure 3 shows the result of the image after dilation.

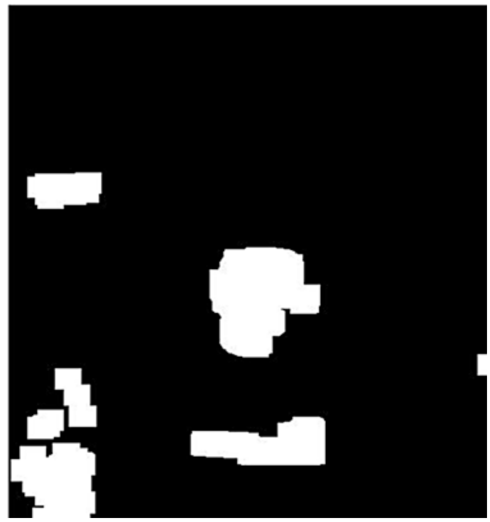

Fig. 4 Image filling

(b) *Filling*

Filling is the process when a hole surrounded by white color (value 1) will be filled to a white color [7]. In other words, if in a white region there exist some black pixels, those pixels will be transformed to white pixels. This case occurs after the skin detection algorithm because the eyes are not considered skin pixels, and so they are converted to black pixels. In MATLAB, the function imfill takes a binary image as input argument and returns an image with the holes filled.

Figure 4 shows the result of the image after the hole filling operation.

(c) *Erosion*

Erosion is second basic operator in the area of mathematical morphology (the other being dilation as stated before). Again, it is used as the dilation operator, so it is typically applied to binary images, but there are versions that work on grayscale images. The basic effect of the operator on a binary image is to erode away the boundaries of regions of foreground pixels (*i.e.* white pixels, typically). Thus areas of foreground pixels shrink in size, and holes within those areas become larger. The erosion operator, as the dilation operator, takes two pieces of data as inputs. The first is the image which is to be eroded. The second is a (usually small) set of coordinate points known as a structuring element. Again, it is this structuring element that determines the precise effect of the erosion on the input image [8]. In MATLAB, you can use the built-in function imerode which takes two arguments, the image and a structuring element and return the eroded image. To get the structuring element object, you can use the built-in function strel [9].

Fig. 5 Image erosion

Fig. 6 Image after the pixel technique operation

Figure 5 shows the result of the image after dilation.

5. *Elimination of the Non Face Regions*
   After doing the previous functions, the elimination of the non-face area will be performed using two techniques:
   (a) *Pixels Technique*
   The correct face has at least 500 pixels, so when we have a skin region with less than 500 pixels, it will be automatically eliminated.
   Figure 6 shows the image after the pixels technique operation.
   (b) *Ratio Technique*
   The correct face has a ratio of height and width between 0.4 and 2.5 respectively, so any skin region that does not fulfill this condition will be eliminated. Figure 7 shows the image after the ratio technique operation.

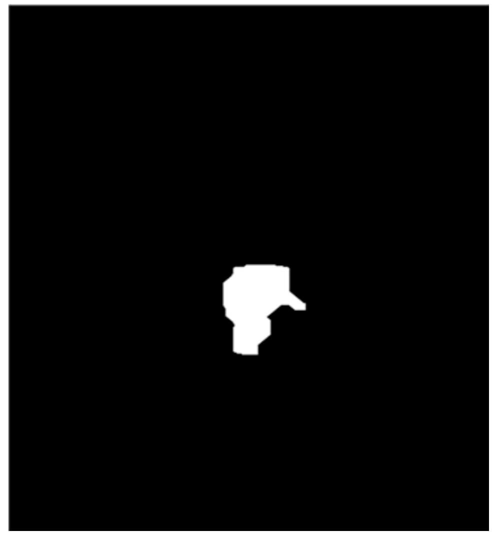

Fig. 7 Image after the ration technique operation

After both techniques, only face region remains and the location of the face will be detected from the image.

Algorithm Suggested for Face Recognition

In the Recognition procedure, the initial image will be gray scaled to accelerate the processing. After applying the face detection algorithm, only the face part is used now to be compared with the database face images.

First, the face is detected using the face detection algorithm discussed previously. The face is split into three sub-images emphasizing on the special features of the face. Then, a correlation is performed on each sub-image with the database images followed by results averaging to check if the face is recognized or not.

(a) *Lightining Effect*
In this step, the effect of lighting will be reduced to be similar to that of the images in the database. Figure 8 shows the difference between the original image and the image after the reduction of the lightning.

(b) *Scaling*
All the images in the database should have the same scale. When the face is detected, the image is cropped to the region of the face, and then it is scaled to the same size of the images in the database.

(c) *Correlation*
After scaling the image, the captured face is correlated with each image in the database. If the maximum value of the correlation is greater than 0.9 or less than 0.5, no need for segmentation, otherwise it moves further to the segmentation step.

Fig. 8 Reducing the light effect

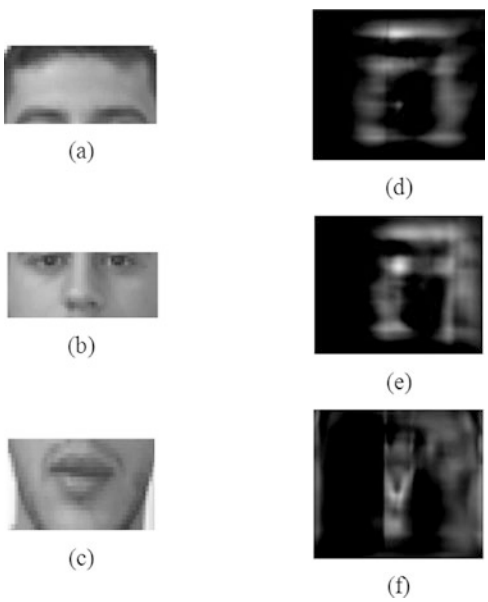

Fig. 9 (**a**) Segment 1 (**b**) Segment 1 correlation with correct face (**c**) Segment 2 (**d**) Segment 2 correlation with correct face (**e**) Segment 3 (**f**) Segment 3 correlation with correct face

(d) *Segmentation*
Segmentation is the process of dividing the face into three regions: upper level, middle level and lower level [10]. Then each segment is correlated with each face image in the database. Figure 9 shows the results of the segmentation process.

Results and Testings

In order to test and validate the proposed face recognition security system, a GUI implementation of the algorithm was applied using MATLAB. The database was created and accessed by the software during the processing of the input image.

This system is a real-time face recognition system that reads a real-time video from the camera connected to the computer running the software, takes an image from this video, processes it to detect any human face presented in front of the camera, and then recognizes the face using a set of face images in a database.

This system was tested on several cases, and it achieved a face detection accuracy of 98 % and a face recognition accuracy of 90 %.

Conclusions and Future Work

Face recognition systems are going to be used more and more in the future for security reasons because they provide better performance over other security systems.

An experimental study face recognition system is presented, which may be applied in identification systems and access control. The proposed face similarity meter was found to perform satisfactorily. The software for the system was coded in MATLAB and was based on face detection and recognition. Although its accuracy is above 90 %, this system may be improved by utilization of additional features.

Light normalization and accurate segmentation of face may allow the threshold value to improve. Cruising the warping space more efficiently, e.g. using a corresponded face rotation and gesture geometric model, may speed up the execution time.

Future work may include improvement of the Face recognition using specific characters in the face (distance between eyes) and also analyze the face in 3-D by using the combination of two cameras and by using these two methods, the probability of error will decrease and the system will be more accurate and with a very low cost.

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Evaluation of Sound Perception to Identify Candidate Frequency for Wireless Networking

Kuruvilla Mathew, Chong Eng Tan, and Biju Issac

Abstract
Wireless technology has been introduced and growing since early twentieth century, but there are still environments the current technologies find it difficult to penetrate. The dense jungle terrain, for example, pose a huge challenge for the 0.12 m wavelength of the Wi-Fi signals, but the FM radio frequency signals at a wavelength of 3 m function a lot better. This paper studies the possibility of using a very low frequency, down to the range of audible frequencies to try and identify the frequency band that can be used, ubiquitously and unobtrusively. Sound can be considered as a ubiquitous signal due to obvious reasons and the search is to find the unobtrusive frequency band that can be a candidate frequency for data carrier signals. The paper is presented in two sections, the first section does a geographically and age neutral survey to identify the unobtrusive signal and second section analyses the noise profiles in these frequency bands.

Keywords
Ubiquitous computing • Rural networking • Low frequency network signal • Wireless networking

Introduction

As the world is moving closer to bringing network technology closer to the people, the need for ubiquitous mode of operation is now highly pronounced. The Ubiquitous wireless networks need more than the current Wi-Fi signal architectures in order to be more power efficient, better performance in obstructions and for application varied types of environments. One area in which the Wi-Fi in the 2.4 GHz band and other high frequency signals used for communication under the FCC regulations fall short in delivering efficient connectivity is in dense jungle type of environments [10]. Where high frequency signals fail to perform in environments with obstructions, research for low frequency signals to deliver the required connectivity begins. Low frequency signals are not only expected to perform better in the presence of obstacles due to their longer wavelength, they also need much lower power to generate, and hence more sustainable in domains with limited power availability. However, it is also to be noted that as the frequency becomes lower, the maximum bit-rate that can be encoded is also lower and hence usually suitable for low bandwidth connections. It is also advantageous to be able to use some ubiquitous signal, which allows use of existing devices with little or no change for it implementation. Such a system is expected to have minimum cost impact, minimal or no training requirements (for the new system) and a very high acceptance factor. In this light, we are studying the

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_43, © Springer International Publishing Switzerland 2015

possibility of using a very low frequency, ubiquitous signal, which is “sound”.

Sound can be generated and sensed with very cheap hardware and is known to travel quite well around obstacles. However, sound being an audible signal, and hence perceivable to the human ear, can cause unwanted distractions or disturbances its environment of use. Hence the acceptance factor will be higher if sound frequencies outside the hearing range for humans are used. The accepted norm for hearing range in humans is from 20 to 20,000 Hz, but the hearing threshold in humans is normally not tested above 8 kHz [9]. We would therefore do random survey to try and measure the practical audible frequency range humans can “perceive”. We have conducted this as a culturally, demographically and age neutral study and noticed that the audible frequency band is actually lower than the accepted norm. The participants of the study was from India, Indonesia, Malaysia, Australia, Netherlands, Hungary, Korea, Brazil, Japan, Switzerland and England with an age group ranging from 15 to 55 years. We will use this output to identify the candidate frequency band that we can use for the network architecture using sound as the carrier signal. We will then conduct an analysis of the ambient noise profiles in these frequency bands and compare them as an approach to looking at the practical implications of making use of these frequency bands.

Related Work

K. Mathew and B. Issac presented sound as candidate carrier for low bandwidth, low power communication, ubiquitously [1]. The paper demonstrated as proof of concept, the ubiquitous data communication using consumer hardware in smart devices and sound as the carrier signal.

K. Mathew, C.E. Tan and B. Issac studied the ambient noise in various natural environments [2] in order to identify the candidate frequency band for communication. The study compares the noise profiles in common environments.

M. Weiser, presented pervasive computing technologies that disappear into everyday life as they ubiquitously blend into our daily activities so that they are indistinguishable [3]. The paper introduced concepts of tabs, pads and boards, and opened up some challenge in networking which the nature of the devices will present.

Madhavapeddy, Scott and Tseworked worked on audio networking as a forgotten technology [4]. They successfully sent and received data using sound as the carrier signal, with common computing platforms to use high frequency audio (ultrasonic) for the communication.

Chen and Lee looked at the inter-relationships among major research themes in ubiquitous paradigm as a bibliographic study on Ubiquitous Computing [5].

Jurdak, Lopes and Baldi proposed using acoustic signals to uniquely identify and locate a user in an acoustic identification scheme for location systems [6].

Madhavapeddy, Scott and Sharp presented context aware computing with sound [7], analysing some location aware applications including pickup and drop interface, digital attachments in voice, etc.

Mandalet et al. proposed application of indoor positioning with 3D multilateration algorithms using audible sound [8] and was able to give accuracy to about 2 feet almost 97 % times using cheap consumer hardware.

A.R. Moller provides detailed information about the physiology and anatomy of the entire auditory system in humans as it describes disorders in hearing in his book Hearing: Anatomy, Physiology and Disorders of the Auditory System [9].

Popleteev, A., Osmani, V. and Mayora, O. presented an investigation of indoor localization with ambient FM radio stations [10], exploring the performance of Wi-Fi and other signals in the indoor vs. outdoor scenarios.

Theory and Background

As computer networks grew, the users also became more and more mobile and wireless networks came into being. Wireless technologies makes use of various frequencies in the radio frequency spectrum, ranging from low frequency signals as low as a few kHz, the TV broadcast signals in the, the more recent mobile phone networks, the very noisy, unlicensed 2.4 GHz band used by a number of short distance wireless protocols including Wi-Fi, Bluetooth, Zigbee and many more in the IEEE 802.11(x) specifications, and some ultra (UHF), extended (EHF) and Super (SHF) high frequencies. The details of these specifications are found in the Federal Communications Commission (FCC) online table of frequency allocations [11]. A quick view of the popular wireless bands in use in the electromagnetic spectrum is shown in Fig. 1. Each of the bands works in their specific domains for providing specific services.

Popular Wireless Signals

The Wi-Fi is a short range signal operating within a room or a small building. Mobile operators establish large towers called base stations to cover a much larger areas and mobile devices connect with these to establish communication Bluetooth was introduced as a cable replacement standard and works in very short ranges, most often to connect peripheral devices like headphones, microphones, keyboards etc. to computing devices. The Zigbee related

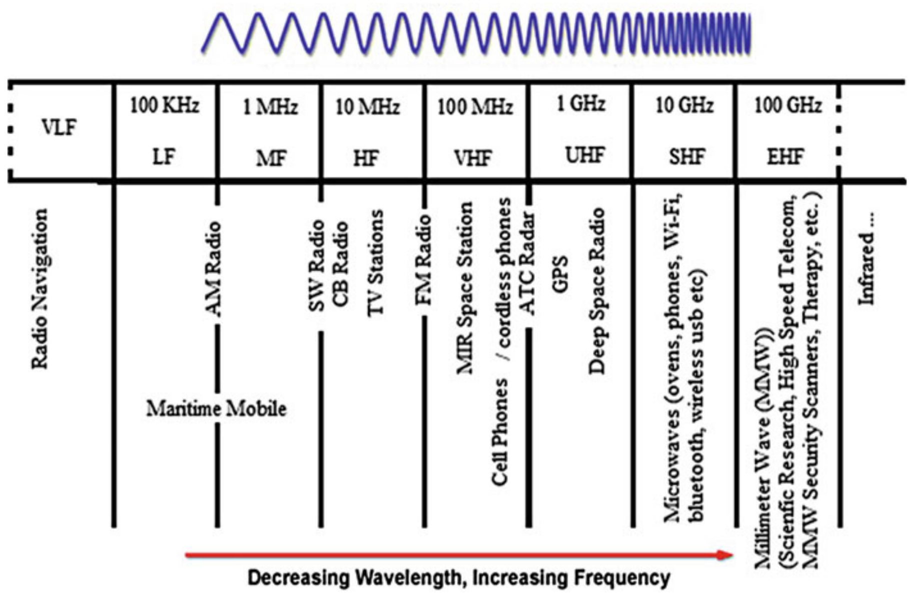

Fig. 1 The electromagnetic spectrum

protocols operate in under the IEEE 802.115 standards to bring wireless network connectivity to low powered devices. Performance of all of these assumes normal urban environment and have seen to fall short in rural environments with thick foliage. This could mainly be due to the fact that the hilly and jungle terrain pose many obstacles in the form of hills and valleys, trees, bushes, rocks and many unpredictable entities. This calls for the need for alternate architecture for effective networking in the rural environments.

The Nature of the Signals

The popular signals we see so far in use for wireless data communications, namely Wi-Fi, mobile, WiMax, etc. use very high frequency, above the 800 MHz range. The high frequency signals require more power to generate, but can support higher speeds of data over greater distances. However, high frequency signals have shorter wavelength and hence do not travel very well around obstacles causing the range of coverage to drop drastically in such environments.

The Terrain

The popular wireless signal performs well in the urban scenarios for which they are designed for, and we have evidenced over the past that the speeds and Quality of Services have kept on increasing with time. These are quite inefficient in rural setting with lots of obstacles. The jungle with thick foliage, with bushes in particular, is a proven signal killer and is known to reduce the usable range by about 25 %.

The high frequency signal of Wi-Fi, operating at a frequency of 2.4 GHz has a wavelength of 0.12 m, and is affected even by moving leaves [10]. This type of terrain calls for the need of a different kind of signal and an architecture to make communication possible in remote terrains which may be limited not just by the obstacles, but also by the limited availability of power.

The New Signal Idea

An approach to tackle the above problem is to make use of a low power, longer wavelength signal (which directly translates to lower frequency) and low cost signal that can function over the terrains in consideration. Lower frequency implies a longer wavelength, which translates to better traversal around obstacles. The FM frequency, operating at about 100 MHz has an effective wavelength of about 3 m and therefore travels well around most naturally found obstructions like leaves and bushes. Hence the idea to study the use of much lower acoustic frequencies arises. Acoustic frequencies are the range of frequencies audible to the human ear, and this is commonly seen as from 20 to 20,000 Hz. To make the communication effective, we will need to use an acoustic signal above the ambient noise levels. This however creates additional human perceivable noise in the environment and hence may not be well accepted. We will therefore study the range of frequencies normally perceivable by people using a random sample study as it is possible that the actual perceivable frequency range is much shorter than the theoretical maximum range. If this is possible, then this translates to the fact that there are some frequencies in the acoustic spectrum that we can utilize for data transmission.

The Survey

A study is to be carried out among the general population to try and identify the perceivable range of audible frequencies. This study can be used to get an idea of the frequencies that is outside the perceivable range and can be used as communication signals. We have the option of utilizing subsonic and supersonic frequencies. The subsonic frequencies are expected to be closer to the 20 Hz mark and hence should be able to traverse well across obstructions. The supersonic frequencies, closer to the 20 kHz mark, should theoretically be able to carry more data as it has more signals per second. The survey will therefore ask the participants to listen to pre-generated signals of various frequencies and identify which of these they can hear and which they cannot. This will help us get to a range of frequencies that practically falls into the audible space.

Fig. 2 The hardware used for the experiment and survey. (1) Apple iPad used to play back the signals, (2) AKG K99, Zoom H4n with remote, consumer headphones (Samsung and HTC)

The Survey, Analysis and Results

The experiment has two phases, the first phase carried out a survey with some generated signals and specific hardware to evaluate the perception of the signals among the crowd. The second phase evaluates the spectrum outside the audible frequency range and performs a noise analysis on this. This range is our candidate frequency band for our communication.

The Sound Signals for the Survey

Phase 1 of the experiment involves a survey. In the survey, we have generated sound samples of specific frequencies, namely, 15, 20, 25, 30, 35, 40, 50, 100, 150 and 250 Hz in the low frequency spectrum and 10, 12, 14, 16, 17, 18, 19, 20, 21 and 22 kHz. The signals were generated using the free open source application "Audacity" and the frequency of the sound signals generated was verified by plotting the frequency of each signal on Matlab to notice that the generated signal pattern is a sinusoidal wave of the correct peak frequency.

The Survey Equipment and Hardware

The Survey involves playing back frequencies ranging from 20 Hz to 20 kHz. This implies that the audio hardware should be capable of handling this wide range of frequencies, which cannot be expected from usual consumer hardware. We used the following hardware to maintain the frequency range is reproduced as close as possible to natural, uncoloured sound. Figure 2 shows the hardware used in the survey.

1. *AKG K99 Studio Monitor Headphones*
   The AKG K99 Studio Monitor headphones offer a frequency response range from 18 to 22,000 Hz. They provide natural, uncoloured sound which is a requirement for the validity of this survey.
2. *Audio Player (Apple iPad)*
   We needed any audio player that will reproduce the digital audio signals we have generated without any colouring. We used the Apple iPad to play out the sound through the AKG headphones for the experiment.
3. *Other Devices for Comparison*
   We also tried a comparison using other smart devices and with popular consumer earphones to notice the variation in the participant response and noticed the colouring the consumer devices adds to the original sound signals. The result of this comparison is not included as it does not directly contribute to this study. Some of the devices were Samsung and HTC Smartphones along with their bundled earphones, an Asus laptop with standard headphones etc. We also noticed some playback software can also cause or add colouring to the signal, which therefore was carefully avoided before the survey.

The Survey Process

The survey was done in normal user environment, which can be considered to be moderate to low noise environments. The participants were first asked to listen to a white noise in to set a comfortable listening volume. Once this is set, the signals are played back one by one, from the lowest to higher frequency for the low frequency band and from the highest to lower frequency for the high frequency band. The very low frequencies are not audible (15 Hz) and the participant will respond to the first audible signal, which is recorded as the lowest frequency the person can "hear" or "perceive as present". Once the lowest perceivable frequency is identified, we start from the very high frequency, at 22 kHz, which also is not normally audible, and play back the lower frequency signals, one at a time. The participant will respond when they can hear the audio signal, which is recorded as the highest frequency that they can hear. In the case of high frequency signal, we expect that they can actually hear and not "feel" the sound as in the case of low frequency.

The sample selected for the survey included participants from all over the world, from the Americas, Africa, Europe, Asia and the Oceania/Australia and hence is culturally and geographically neutral. The population is also age neutral and includes participants from various age groups within 15 years and 55 years, which helps the survey results not to be biased towards a particular age group. The identities of the participants are kept confidential in order to protect their privacy.

Survey Results

Table 1 shows the results of this study, where we notice that the lowest frequency of perceivable audio signal observed in the study was 25 Hz and the highest was 18 kHz. This allows us to deduce that majority of the population may not perceive the presence of a sound signal below 30 Hz, at normal amplitudes, though at a really loud amplitude we may "feel" the very low frequency vibrations [10]. The distribution of participants' response to the perceivable low frequency signal is shown on Fig. 3 and the perceivable high frequency signal is shown on Fig. 4. We can observe that from the population, no one could hear the tones of 15 and 20 Hz. Though some could "perceive" the presence of the 25 Hz signal and majority could perceive only from 35 Hz and above. On the high frequency side, though we can notice that a few can "perceive" or actually "hear" the audio signal up to 18 kHz, majority cannot perceive above 14 kHz, and some can perceive up to 17 kHz. 18 kHz, can therefore, be considered a safe upper limit of "perceivable audible sound frequency" according to the results of this survey.

Table 1 Results of audio perception survey

DocN	Geo	AG	Low (Hz)	High (kHz)
JL1325-01	India, Middle/ S. Asia	<15	25	17
JL1325-02	Indonesia, E. Asia	<30	30	16
JL1325-03	Malaysia, E. Asia	<30	50	17
JL1325-04	Australia, Oceania	<30	35	17
JL1325-05	Netherlands, Europe	<30	35	16
JL1325-06	Hungary, Europe	<30	35	14
JL1325-07	Korea, E. Asia	<55	50	14
JL1326-01	Brazil, S. America	<30	30	14
JL1326-02	Malaysia, E. Asia	<30	30	14
JL1329-01	Japan, E. Asia	<55	25	14
JL1329-02	Switzerland, Europe	<30	30	14
JL1330-01	Malaysia, E. Asia	<30	40	18
JL1330-02	England, Europe	<30	30	17

DocN, participant response document number (name not included to protect respondent privacy); Geo, geography/country; AG, age group; Low, lowest perceived frequency; High, highest perceived frequency

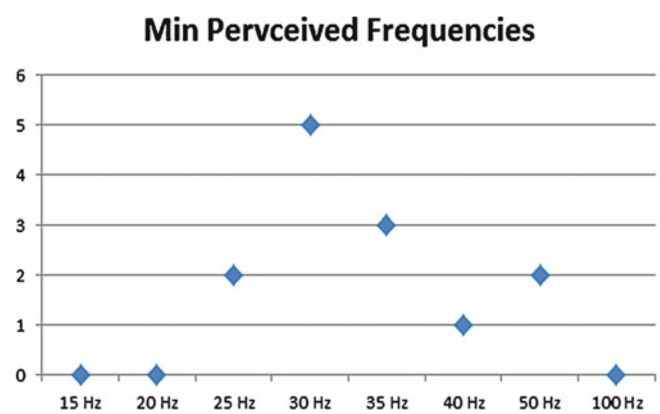

Fig. 3 Survey result plotted for frequency of sound against number of participants who's lowest perceivable frequency

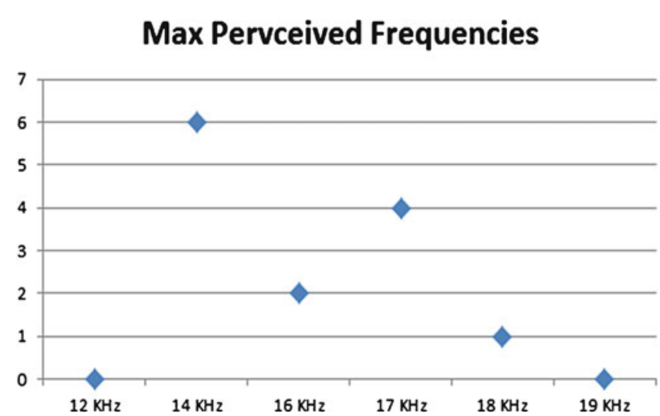

Fig. 4 Survey result plotted for frequency of sound against number of participants whose highest perceivable frequency

Noise Analysis of Candidate Frequency Band

The survey has helped us narrow down on candidate frequency band that can be considered available for "audio networking". However, these being sound signals, is still open to a lot of ambient noise that may be present in the natural environments of intended use. Hence we also carried out a noise profiling for the candidate frequency bands.

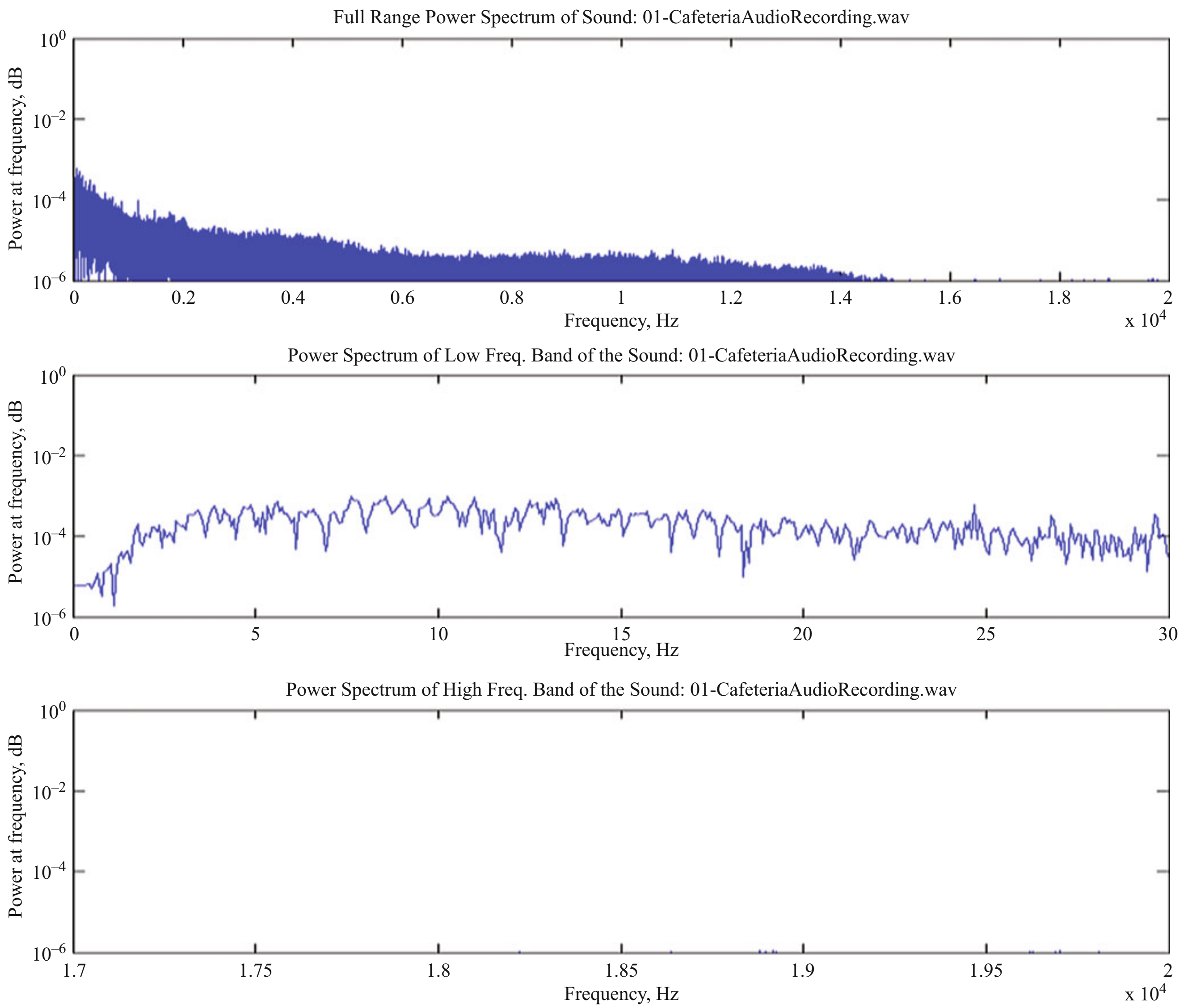

Fig. 5 Cafeteria noise analysis

The Candidate Frequency Bands

As a result of the study, we have identified that the frequency bands below 25 Hz and above 18 kHz can be considered for the intended audio networking technology. The noise profiling was done in the four environments we studied, namely, the quiet office, the busy cafeteria, the beach and the urban roadside. For the purpose of comparison, we have divided the zones into the low frequency zone and the high frequency zone. The further sections describe the analysis and the results of the studies.

The Experiment Process

This analysis was conducted by recording the ambient noise from various environments using the Zoom H4n recorder, shown in Fig. 2. All the sound samples were recorded with the same device, using its high response built in microphones at a consistent gain of 40 % in order to get comparable signal. The recorded signals were analysed using Matlab to plot the actual spectrum of the noise and then specific bands we are interested in using a band (low pass and high pass) filter and plotted within 0 to 10^{-5} db. The results of the analysis are discussed in the next sections.

The Low Frequency Zone

The very low audio frequencies outside the perceivable range noticed are below the 25 Hz mark. The very high audio frequencies outside the perceivable range noticed are above the 18 kHz mark. The study of the comparison of noise profiles in "The Cafeteria", "Car Park", "City Side Park", "Beach at Night with Footsteps" and "The Crowded Lobby" is shown in Figs. 5, 6, 7, 8 and 9 respectively.

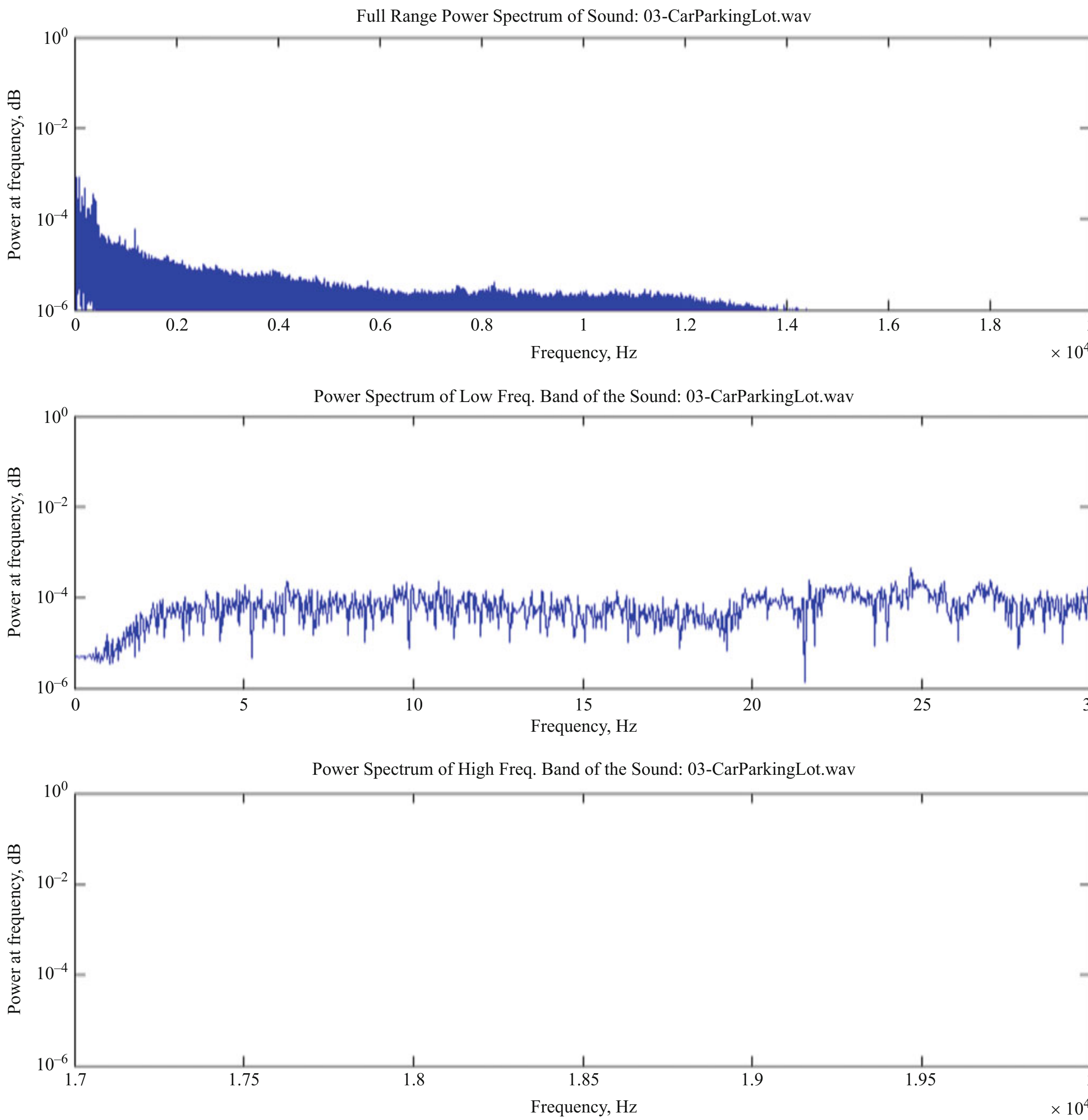

Fig. 6 The car park noise analysis

Few more environments were analysed, but not included as it did not add more light to the results.

We can notice a very consistent pattern in all the signals recorded and analysed. We can see that in almost all cases, we have strong low frequency signal in the range of 10^{-5} db to 10^{-2} db, but the high frequencies above 18 kHz is almost absent. The high frequency signal, on the other hand, has a much larger bandwidth above 18 kHz and is theoretically expected to be able to support better data rates. Hence, considering the noise to signal ratio and possible future performance requirements, the high frequency band above 18 kHz is a better recommended candidate for the intended audio networking architecture.

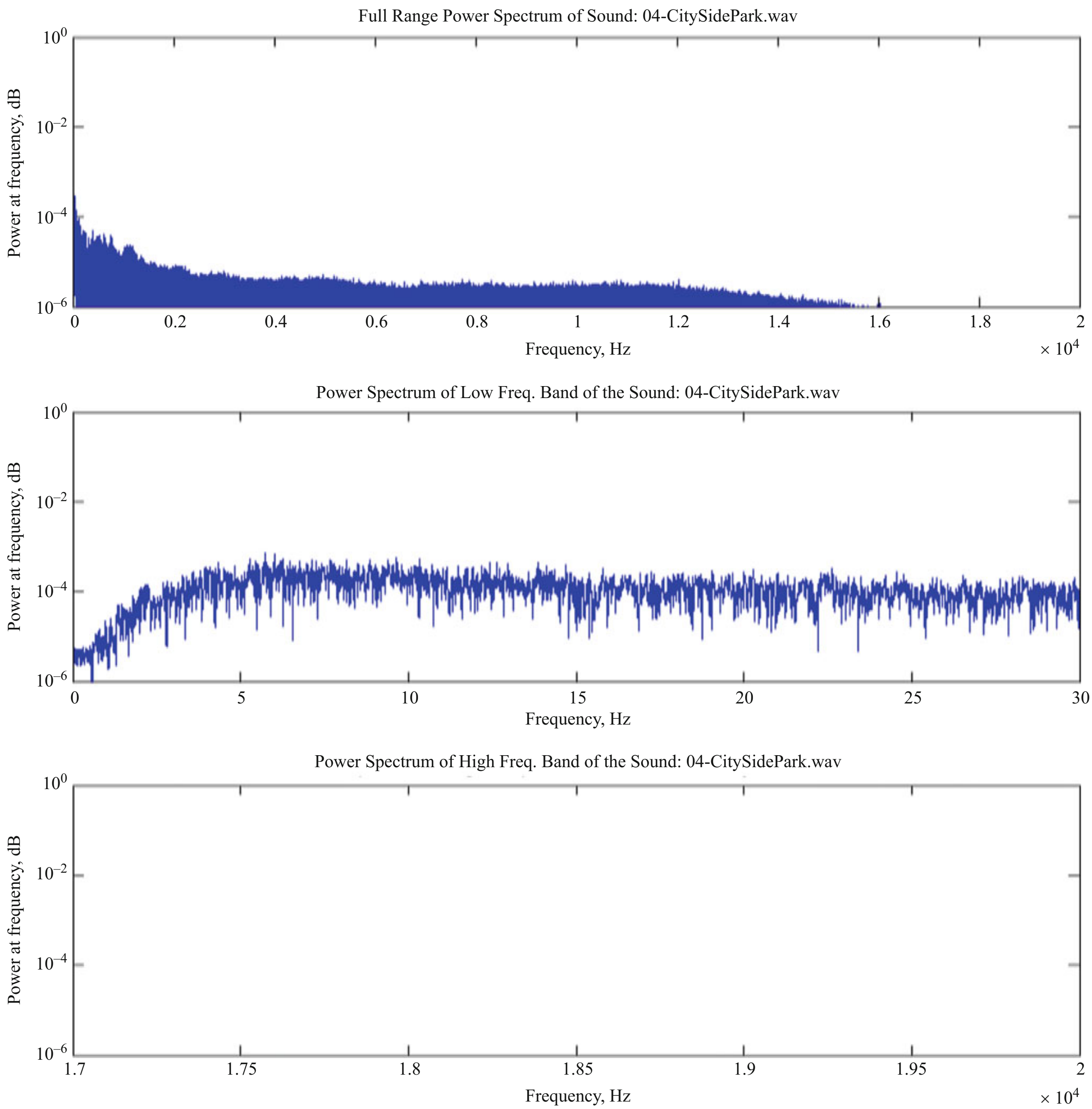

Fig. 7 City side park noise analysis

Conclusion

The study was aimed and has contributed towards identifying a candidate sound frequency for signal transmission, for use in network technology, unobtrusively. Using of the frequencies outside the perceivable range will allow network communication to carry on using sound signals that blend or fade away into the background.

In an effort to further the study, we have also done a noise sampling for a number of environments and did a profiling on the identified candidate frequency bands to notice that ambient noise in the high frequency band is a lot "quieter" that the low frequency band, suggesting that as a better possible channel for network communication. The available frequency range in this band is a lot wider as well as having a much higher signal rate, suggesting possibility of a theoretical higher data rate. It has also been noticed that

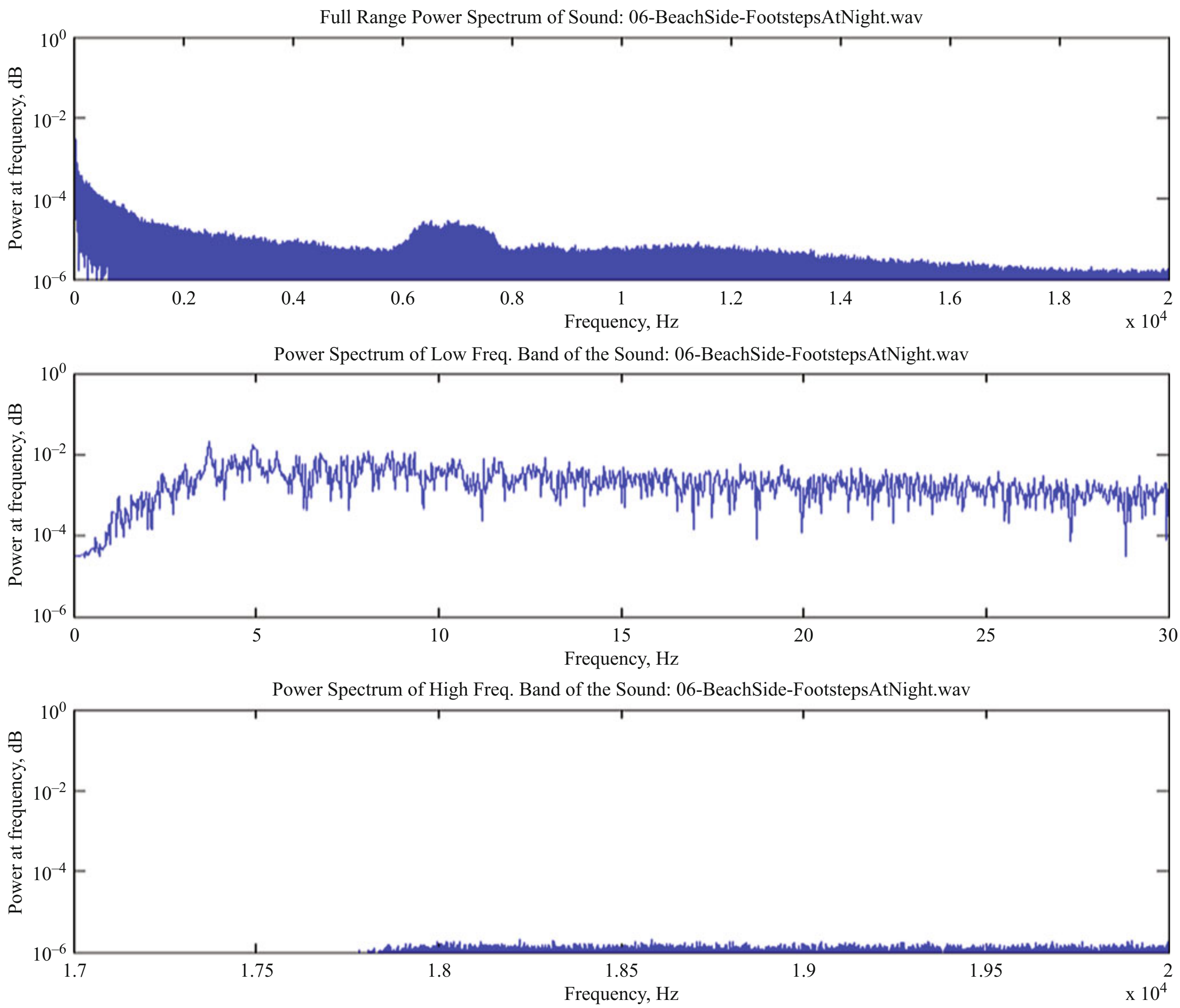

Fig. 8 The beach at night with footsteps noise analysis

very low frequency sounds played at very high amplitude, even though may not heard, can be “felt” or “perceived” by the humans [11]. Further studies need to be carried out to identify ubiquitous devices that can handle the frequencies well. The actual attenuation of the various signal frequencies in different environments also need to be studied further to evaluate sufficiency in terms of signal propagation in our target environments.

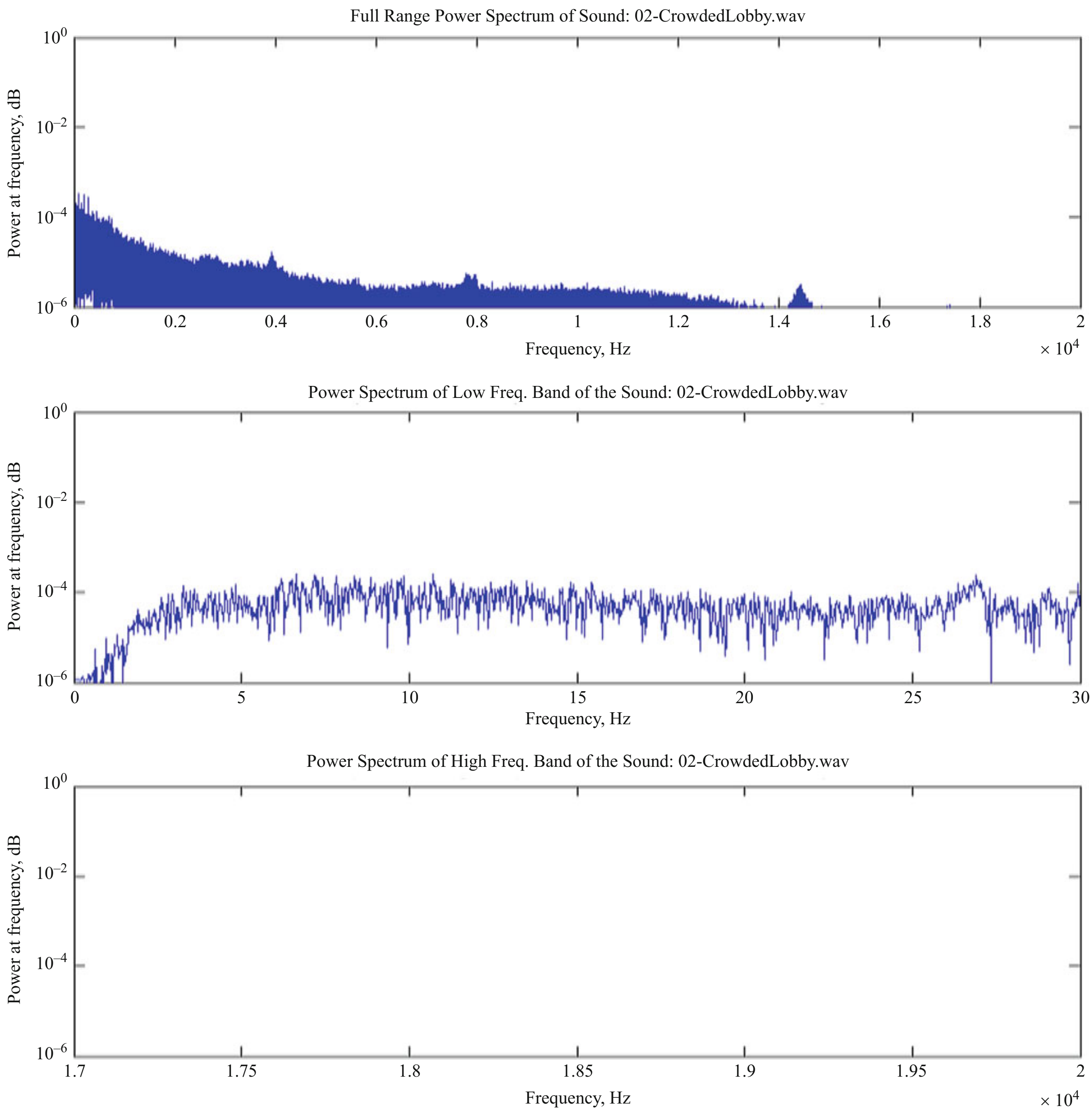

Fig. 9 The crowded lobby noise analysis

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The Efficiency of the Code Parallelization in Multi Core Environment on the Basis of Image Processing in 3D Space

Krzysztof Oleszko

Abstract

Grains parameters like volume, surface area or shape factor are important in geological issues connected with rocks and coal mining. For quick and efficient way to calculate that parameters are use some complicated algorithms which are based on source data written in flat 2D images. Almost always raw images need to be filtered and always need to be transformed into 3D images and process in 3D space to obtain reliable results. Even if algorithms which can do that operations are run on efficient computers, operations performed in 3D space consumes a lot of processing time. Authors made attempt of parallelization procedures performed in 3D space to improve efficiency on computers equipped in widely used, multi core processors and presents results.

Keywords

3d image analysis • Image processing • Grains reconstruction • Parallel processing

Introduction

Image processing performed in 2D space can be very time consuming, especially when object of interest is written in high resolution image. Almost always processing is composed of many separated algorithms which are applied one after another in exact way to the image, to obtain desired result. When processing is moved into 3D space [17], then whole process becomes more and more time consuming, especially when procedures are compound of algorithms based on mathematical morphology. In that kind algorithms, to process single pixel it is necessary to read and process neighborhood of it, few times more than in 2D algorithms. In geology, 3D image processing is present, but not commonly used [14]. Especially in topics connected with grains and its structure [1, 3]. That kind of measurement are important in rocks and coal mining. There are some algorithms, applied with success to compute volume, surface area or shape factor of grains [11]. Yet, if there are a lot of data to process, algorithms processed on single processor can last many long minutes or even hours. To speed up this process, algorithms with some small modifications can be run simultaneously to shorten time of image processing and analysis.

Many authors describes different parallel image processing algorithms [6] in very narrow application context, or describes general parallel concepts and problems [4, 5, 8, 12]. Also many of scientists performs popular parallel computing on GPUs [2, 13]. GPU processing is time saving process, but creating such algorithms is not as easy as parallel algorithms performed on CPUs. Most of researches are more theoretical, and it is difficult to find exact results of some parallelization proved by real tests and comparisons [16]. That is why author decided to prepared algorithms and compare them on different CPUs to present obtained results.

For the project, there was created new code, written in JAVA based on existing code used in previous research [11]. To 2D and 3D images, were applied algorithms based on mathematical morphology [15]. On the images used for the project were presented group of grains and each grain

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_44, © Springer International Publishing Switzerland 2015

needed to be processed and analyzed. As a result of research, authors proposed simple way of code parallelization, and presents promising results.

Code Parallelization

In general, computer programs are written to be processed on single processor or single core, where problem is divided into some small pieces, where each piece is executed, one after another. Whole program can be rewrite in order to execute with success in parallel environment where can be achieved grate speedups. But not always there is available complex parallel code or parallel environment. However, nowadays, single processor or single core computers seem to be history. For some years, majority of personal computers are equipped with multi core processors. When user, in the same time is executing few programs, operating system can manage that situation, and provide best efficiency utilizing all of available cores. But in situation, when programmer creates some algorithms which are executed one after another on exact part of data, operating system can do nothing with that situation. Algorithms are being processed on single core, even if there are available 4 or even 8 cores. That situation is common in image processing and analyzing issues, where there is available single image, and there need to be applied some filters on that image or some objects presented on the image. When programmer writes his code in some high-level programming language like JAVA, there are provided some mechanisms which can speed up image processing and analysis executing same code on different cores processing different individual parts of data. What is more, those parallel mechanisms can be easily applied in standard serial code. A programmer only need to keep in mind to write parallel loops instead of standard ones. After all, code is executed in more efficient way and saves time.

The Algorithm

The algorithm in serial version was used in previous research, where is described with details [10, 11]. In this project code was rewrite in JAVA and some crucial parts of it were parallelized. Basically, algorithm applies transformations and filters like watershed, border cleaning or different morphological filtrations on 2D source image, next there is created 3D image based on 2D one using advanced mathematical transformations based on image symmetrical reflection [9]. After that, separately for each grain, there is applied spatial morphological filtration algorithm and there are computed grains parameters like volume, surface area or grain dimensions [7]. Operations performed in 3D space were parallelized to improve efficiency.

Test Environments

The program was tested in two different kind of environments. First one was PC personal computer with 16 GB RAM and Intel Core i7 2.93 GHz 64 bit processor. Second environment was PC laptop computer with 2 GB RAM and Intel Core 2 Duo 2.0 GHz 32 bit processor. On both machines, was installed Windows 7 Professional operating system. In first environment the tests were run on JRE (Java Runtime Environment) 1.6 32 and 64 bit, and JRE 1.7 32 and 64 bit. In second environment the tests were run on JRE 1.6 and 1.7 32 bit.

Experiment

For the tests there were used two images containing real data. Each image contained about 50 grains. After grains separation, each one was processed separately in different CPU process to improve efficiency. The program were run on different number of cores, depending on the environment. In the environment with two cores, program were run on single and on two cores for both images. In the environment with eight cores, firstly, program was run on single core, and then number of cores was successively increased by one to number of eight. Each test containing single image was repeated five times on given number of cores. From obtained five results was taken average value which was used for further deliberations. All the tests were run in both environments. In the Intel Core 2 Duo tests were run on two kinds of JRE's, in the Intel Core i7 environment tests were run on four kinds of JRE's. That situation was caused by particular CPU architecture. On 32bit CPU can be used only 32bit JRE, but on 64bit CPU can be used both, 32bit and 64bit version of JRE.

Selected Results and Discussion

In performed tests, due to similar grain sizes presented on processed images, there was not applied any algorithm dealing with data irregularity. Also, processed tests proved, that the data taken under consideration were quite regular and all the processes were executing equally.

In the following discussion there are presented results for one image for all test cases. Results for both images were more or less similar and that is why one of them will be presented.

Table 1 Results for Intel Core 2 Duo environment

JRE	Number of processes	Read time (s)	Preparation time (s)	Processing time (s)	Program time (s)
1.6	1	5.56	5.66	432.64	443.80
	2			229.37	231.65
1.7	1	5.37	5.75	196.73	207.81
	2			105.61	116.77

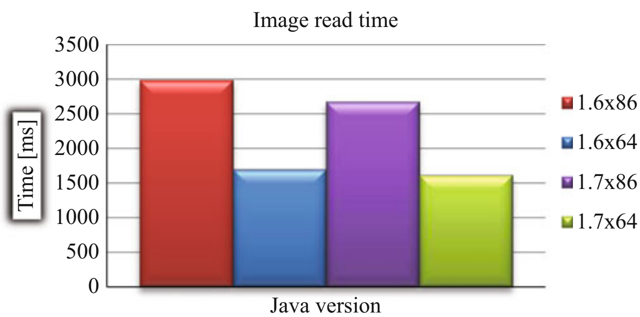

Fig. 1 Image read time

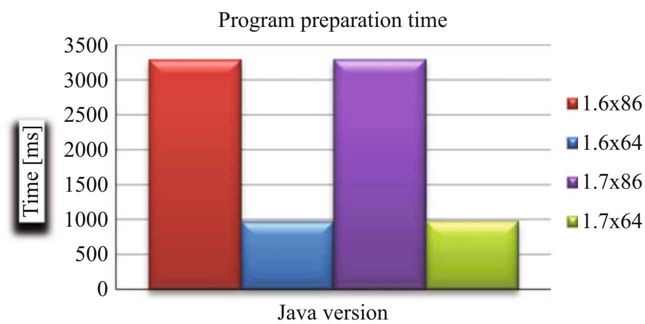

Fig. 2 Image preparation time

Intel Core 2 Duo Results

In Table 1 are presented results for Intel Core 2 Duo environment. In first column is presented version of JRE, in second number of processes. In third column is presented time of image reading, in fourth time after image reading and before proper processing which mainly includes initial filtration in 2D space. Fifth column presents 3D processing time and sixth presents full program execution time. It is clearly noticeable that JRE 1.7 is more efficient and JRE 1.6. The majority of program time was generated by operations in 3D space. Speedup for whole program for JRE 1.6 is equal 1.92 and for JRE 1.7 is equal 1.78. But speedup JRE 1.7 in relation to JRE 1.6 for single process is equal 2.14. Therefore, in this scenario should be used only JRE 1.7 and parallel code which provides best efficiency.

Intel Core i7 Results

An Fig. 1 shows image read time for all the JRE's configurations. For JRE x64 and x86 there is a big difference in those times and for JRE x64 efficiency is better. Also between JRE 1.6 and 1.7 there is a small difference in image read time, it can be noticed that JRE 1.7 achieved better execution times.

On Fig. 2 image preparation time is presented, which mainly contains initial filtration (preparation) time in 2D space. Same, like for image read time, there is quite big difference between JRE x64 and x86—the JRE x64 is more efficient in both versions, 1.6 and 1.7. However there is almost no difference between JRE 1.6x64 and 1.7x64.

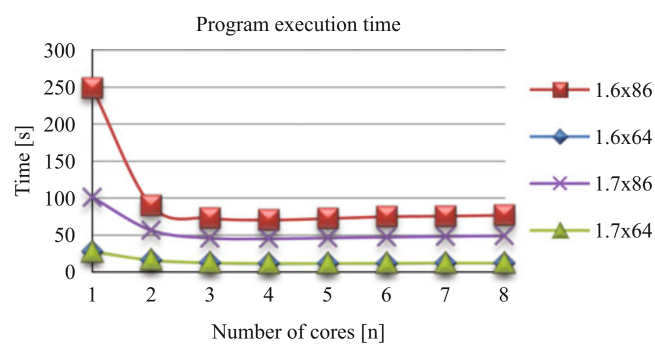

Fig. 3 Program execution time for all types of JRE

Similar to Intel Core 2 Duo, in the Intel Core i7 environment most of the program execution time is generated by parallel processing of data in 3D space. Thus, there will be presented time of full program execution and its speedup, and there will not be separately considered 3D processing time.

Figure 3 shows full program execution time, on the x axis the number of cores is presented. On the y axis the execution time in seconds is presented. Each line represents different JRE version. It is noticeable, that JRE's x86 is the slowest one of all of them. For example, on single core on JRE 1.6x86 program was executing 248.41 s but on JRE 1.7 only 27.37 s. It is caused by processor architecture, which is 64bit architecture. There is simple conclusion, that on 64bit processor should be only used JRE x64.

On Fig. 4 there are presented speedups for x64 JRE's. For both, JRE 1.6 and JRE 1.7 speedups are similar and there is no significant difference between them. The interesting thing is, that speedups are increasing up to 4 cores and they are quite good, after that, for 5–8 cores speedups are worse than

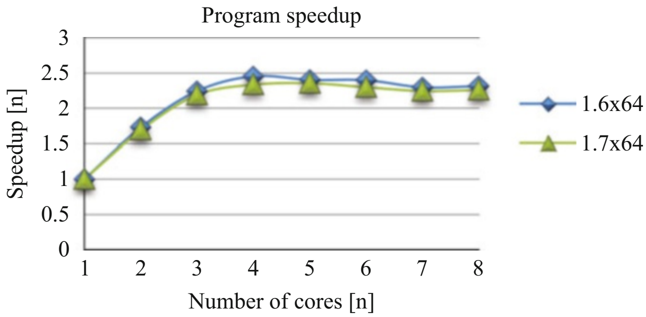

Fig. 4 Program speedup for two the fastest JRE versions

for 4 cores. In spite of Intel Core i7 provides eight logical cores, the real speedups can be achieved only for four physical cores. Therefore, to achieve best results for that kind of environment, parallel code should be used for maximum four cores which gives speedups about 2.45 comparing to single core.

Summary

The problem of program execution efficiency in multi-core environment was presented in this paper. The main task of the experiment was to improve efficiency of image processing algorithms applied in 3D space using commonly user processors. Obtained results proofs that speedup of program run in multi core environment can be easily achieved. However, not always good hardware specifications brings satisfactory results. Also, the programmer must be aware of traps which are connected with proper hardware—software configurations.

Acknowledgment This work was financed by the AGH University of Science and Technology, Faculty of Geology, Geophysics and Environmental Protection as a part of Dean's grant number 15.11.140.213

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Evaluation of the Contract-Aware Software Development Process in a Controlled Experiment

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Abstract
Contract-Aware Software Development (CASD) process combines the Model Driven Engineering (MDE) approach with the Design by Contract ideas performed at the modeling level. Software engineering solutions need empirical investigation on the impact of methodology on the developed products. Therefore we have designed and performed a controlled experiment analyzing the crucial parts of the process. We focused on the CASD process specialized for UML models with contracts specified in the Object Constraint Language (OCL). Models with contracts are automatically transformed into C# code. In the experiment different development phases and their products were evaluated. As a result a high consistency between contract specification at a model and a code level was confirmed. The evidences stressed very high requirements on the tool support, and some inconveniences that still limit widespread application of the MDE paradigm.

Keywords
Computer aided software engineering • Contracts • Object oriented modeling • Software development management

Introduction

The complexity of information systems puts pressure on the constant verification during the whole system development and evolution process. The correctness rules can be specified at all process stages as contracts and transformed together with other software artifacts. This idea states behind a generic process called Contract-Aware Software Development [1] that combines the Model Driven Engineering (MDE) approach [2] with the Design by Contract ideas [3] performed at the modeling level.

The general goal of the process is quality improvement of the software artifacts, its testability and maintainability. The main potential advantages should be recognition of incorrectness at early stages of software development and preserving of consistency between different levels of software artifacts, supported by benefits of contract utilization at the source level.

However, an open question remains: how the best practices promoted in the process can be introduced in practice. We have therefore performed a controlled experiment focused on evaluation of a Contract-Aware Software Development (CASD) process specialized for UML as a modeling notation [4], Object Constraint Language (OCL) [5, 6] for contract specification at the model level, and C# with Code Contracts library [7] at the implementation level.

Adequate assessment of the software quality is difficult if the maintenance phase is not taken into account, though we tried to evaluate the artifacts created at each development stage as well as to question the participants of the experiment. The experiment confirmed that a high consistency of contract specification at model and code level can be achieved, but also pointed at the obstacles of the approach.

In the next Section, the background of the experiment is presented. In section “Experimental Design” we describe

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_45, © Springer International Publishing Switzerland 2015

the basic features of the controlled experiment. Results of the experiment are discussed in section "Analysis of the Experiment". Finally, section "Conclusion" concludes the paper.

Background

Contract-Aware Software Development Process

CASD is an approach to a generic process that combines features of the contract-based and model-driven development [1]. A distinguishing feature of the process is encountering of dual artifacts in various phases, from the analysis to the implementation one. Artifacts realizing certain functionality have their corresponding artifacts specifying constraint rules—contracts at a given level of abstraction. Transformations between consecutive process levels should preserve a constrain dependency between the corresponding artifacts.

The CASD process should be adapted in order to implement this general idea. One of possibilities is utilization of models in the UML notation [4]. Furthermore, the models can be specialized with dedicated profiles creating a kind of Domain Specific Language. One of many languages that can be applied for contract definition at models is the Object Constraint Language (OCL) [5, 6]. It is a declarative specification language used in the UML definition.

Transformation of models with contracts at the specification level, to the corresponding source code and appropriate contracts is a crucial part of CASD. In the specialized process, discussed here, this transformation together with the profiling of UML models can be supported by T.O.F.I.C. [8]. It is an extension of the IBM Rational Software Architect tool (RSA in short) [9]. A new version of T.O.F.I.C. [1, 10] is enhanced with visual facilities for efficient creating of UML models with C#-dedicated profiles. It also supports comprehensive interpretation of OCL constraints (invariants of classes, pre- and post-conditions of operations) and transform them into contracts from the MS Code Contracts library [7]. Code Contracts bring the advantages of design-by-contract programming to .NET programming languages.

OCL Constraint Transformation

With the following, simple example we illustrate how an OCL constraint is transformed into its corresponding code. In an airport model a *FlightList* class manages a set of flights. This class is associated with a class *Flight*. An end of the association is specified with "to many" multiplicity. However, it is required that the list includes at least one flight. It could be expressed with an invariant of the *FlightList* class stating that an appropriate collection is not empty. The appropriate code of C# is automatically generated using a contract method from the Code Contract library and C# mechanism of delegates (Listing 1). The original OCL constraint is incorporated as a comment into the source code.

```
/*OCL invariant: flight->notEmpty()*/
public partial class FlightList   {
[ContractInvariantMethod]
  private void FlightListInvariant ()   {
  Contract.Invariant(Ocl.invoke(delegate   {
    return this.flight.set().notEmpty ( ) ;
            }));  }
…}
```

Listing 1. OCL class invariant and its implementation

Related Work

Design-by-contract paradigm was proposed and established in the context of programming languages [3]. Here, we would like to benefit from the approach, but the specification effort is shifted to the former phases of the software development.

OCL constraints are commonly used in different MDE approaches. Automatic generation of component skeletons implementing modeled components with OCL constraints is presented in [11]. The approach is focused on Java CORBA as a target platform. In [12] business rules associated with models and written in OCL are statically validated using the Octopus/OCL tool and transformed into Java code by a prototype generation tool—AutoPA.

UML models with OCL specification can also be transformed to another formalism in which they are verified using available solvers, for example with Constraint Satisfaction Problem (CSP) tools [13]. However, there are some limitations and very high requirements on the model and constraints completeness that are usually not met in the industrial practice. Moreover such verification of the modeled contracts could be performed before model to code transfer and would supplement the process discussed in this paper.

Constraint-driven Modelling (CDM), which also uses OCL constraints, focus on generation of constraints from the source model [14], whereas in our approach constraints are used for the software verification and are the object of transformation together with software artifacts.

Implementation of MDE is still an open issue, and convenient tool support is one of its main obstacles [15]. In many CASE tools we can use OCL together with modeling facilities, but only a few of them support code generation, so important for MDE approaches. Comparison of OCL tools can be found in [16] and discussion of code generation capabilities in [10]. The most comprehensive OCL tool is Dresden OCL [17], but it does not work with C#.

Experimental Design

The purpose of the experiment was investigation of basic parts of the CASD process. It was mainly focused on the impact of automatic generation of code with contracts into the software development process. We also studied the maturity of the T.O.F.I.C. tool support, whether it would be helpful in completing contract modeling and its transformation.

Subjects

The experiment was carried out during the laboratories in Advanced Software Engineering (ASE) course in Institute of Computer Science, Warsaw University of Technology. Experiment subjects were sixth-semester students attending the Computer Science bachelor degree studies. During the previous semesters they had done different courses on object-oriented programming (C++, Java) and the basic course of Software Engineering. During laboratory tutorials of this course they prepared a preliminary project covering all basic types of UML diagrams with support of a CASE tool, and experienced major development activities from requirements to code generation and implementation.

Direct before the experiment, during the ASE course, the participants extended their knowledge about requirement engineering, UML modeling, and application of design patterns at the model level. They also learned about contracts and fundamentals of OCL.

Experimental Tasks

During the experiment a system simulating an airport was designed. The system was divided into modules corresponding to different system tasks, such as: general management, flight control, flight schedule, catering, airplane technical service management, luggage service, customs service, airport security, and crew management.

Experiment Procedure

The experiment scenario consists of the following main steps:

1. Functional and non-functional requirements were analyzed and specified within a CASE tool.
2. Based on the requirements, use case models were prepared. Specification of use cases was written in the form of predefined structured, textual descriptions. It included, among others, pre- and post-conditions of use cases and their invariants, if applicable. The prepared requirements of a module and use case diagrams with their specification were handed over to other participants. This procedure corresponded to a typical situation, were requirements are prepared by analysts and passed to a design team. Each participant wrote a review of the obtained specification.
3. UML class models were designed to meet the requirements of a module. The subjects were encouraged to use design patterns in the class model. The models were specialized using C# profiles.
4. Structural models were enhanced with contracts covering class invariants, pre- and post-conditions of non-query operations, and constraints of attributes. The contracts were expressed in OCL.
5. A mapping of structural model to C# code was created. Stereotyped class model with OCL constraints was transformed into the corresponding C# code.
6. The subjects implemented selected functionality of the application with the contracts.
7. The implemented modules were tested using developed unit tests. The tests covered implemented functionality as well as the transformed contracts. Selected tests carried out activities that contradicted rules specified in the contracts. Such tests were verified whether they falsify the contract conditions, i.e. error occurrences were expected.

Experiment Outcome

There were three kinds of experiment outcomes. The first type of outcomes consist of intermediate results of the laboratory tasks presented during the consecutive meetings of the experiment and the final results delivered at the end of the semester. The artifacts submitted by each participant included: a requirement description, a use case model, a review of the obtained requirements and of use case specification, a UML project with classes, design patterns and OCL constraints, a C# application with contracts and unit tests.

The second type of outcomes were different software metrics calculated on the direct artifacts mentioned above.

The third type of results stands for the data collected in the survey performed at the end of the experiment.

Apparatus and Tools

In this point we listed tools used for experiment realization and its evolution. It should be stressed that the CASD process was not fully supported, especially we did not make automatic transformation from requirements to use cases, or from use case contracts to contracts in a design model. Moreover many backward traceability facilities, foreseen in the process, were not automated.

Software requirements of the system modules were stored and analyzed using the IBM RequistePro tool with the MS Access data base. The requirement data base was provided as an input to the IBM Rational Software Architect (RSA) [9]. This CASE tool was also used for UML modeling and assisted in application of design patterns. Introduction of OCL constraints was supported by RSA, including constraint parsing facilities.

Model specialization towards C#, as well as transformation of classes with OCL contracts to C# code were realized using T.O.F.I.C.—an extension of RSA [1, 8, 10]. Final C# applications were development in MS Visual Studio with support of the Code Contract library [7].

The analysis of UML general models and models defining C# code structure were performed with Eclipse lightweight plug-ins (so called pluglets) implemented for the experiment. The measurement of OCL contracts was also based on an implemented pluglet that checks the OCL constraints and analyses utilization of AST (Abstract Syntax Tree) constructs according to the OCL specification.

In the final C# applications, various software metrics were measured with a static analyzer tool NDepend. The unit tests were run and evaluated using MSTest tool included in the MS Visual Studio. The code coverage of tests was verified with the NCover tool.

Survey Procedure

The subjects expressed their opinion and evaluated the experiment in a survey carried out at the end of the experiment. The questions of the survey considered the following issues:

- Self-assessment of experiences and knowledge of the object-oriented technology,
- Labor intensity of the project,
- Estimation of usefulness of the various tools used in the project, especially T.O.F.I.C. options supporting C# and contract modeling and transformation,
- Opinion about the tool maturity,
- Impact of the contract-aware software development on the code quality, consistency with the specification and task realization time,
- Applicability of tools and the methodology to future projects.

The survey ended with an open question about the subject's attitude to the experiment and observed advantages and disadvantages.

Analysis of the Experiment

Requirements and Use Case Models

The use case models were used for description of actor-system interactions and available services. Completeness of requirements as well as correctness and sufficiency of use case models were checked both by the supervisor and the subjects after exchanging of tasks.

A challenging activity in this step of the experiment was introduction of constraints into the textual specification of use cases. As recommended, almost all specifications of use cases included some contracts, written in a structured natural language, associated with logical equations or arithmetical relations if necessary. At this process stage, contracts were only qualitatively assessed. Presence of contracts was checked and whether they were logically correct formulated. However, we could not use a qualitative measure that evaluate the future usability of these use case contracts during the design of the contracts specified at the modeling level.

Evaluation of UML Design Models

Evaluation of the UML structural models concentrated on the profiled models specialized to the desired language. The calculated metrics were devoted to complexity of models, number of C# stereotypes, variety of C# structures used for the model refinement, coverage of UML model elements with the corresponding C# stereotyped model elements.

For the verification reasons, UML models of C# implementation and code mapping were also analyzed with the Software Analyzer module of the RSA.

The class models designed for the implementation were fully stereotyped as C# code models, with the coverage from 98 to 100 % of all modeling items. The correctness of association between C# profile items and modeling concepts was partially verified by the T.O.F.I.C. internal methods.

The specialized models were not complex as far as the variety of types is concerned. The mostly used C# concepts specified as stereotyped modeling items were: classes, methods, constructors, destructors, fields, properties, and their accessors. As less commonly used concepts we have observed class inheritance, interfaces and their realizations, enumerations, namespaces, and delegates.

Evaluation of OCL Contracts

OCL contracts specified at the design level consisted of invariants associated with 60 % of classes profiled as C# classes. The pre- and post-conditions of operations were used only for selected operations that modify system states. About 85 % of operations stereotyped as C# methods were specified with such contracts.

It should be noted that the structural models of any module were designed by other subject than the one preparing requirements and use case models. In result, the design contracts often do not reflect the contracts from the corresponding use cases. Some conditions were overlooked, but some new were added. This observation confirms the necessity for a tool to support tracing of contracts for all stages of the software development, as suggested in the CASD process.

The application of a specification language, OCL in this context, was new for the subjects, which were familiar with different imperative programing languages but not with declarative ones. We examined the usage of various concepts of the constraint specification language. The contracts were commonly specified with variable expressions, calls to attribute values and operations, and literals especially with *null* value. The subjects did not used more complicated OCL structures, like a variable definition, conditionals, or general iteration over a collection; probably due to lack of experience with the new language.

However, using only a basic subset of the specification language, the experiment participants were able to formulate the logically correct and meaningful contracts. One advantage of simple contracts was that they were easily to be interpreted and checked. If the contracts, which are a source of the verification code, were erroneous the whole verification activity would be in vain.

Code Transformation

In the final C# applications the following software metrics were measured: IL—number of virtual lines of the intermediate language of .NET corresponding to the implementation, LOC—number of logical source code lines, NM—number of methods, NF—number of fields, NT—number of types.

The measurement covered only the parts of the code that could be modified by the subjects, excluding modules of MS Code Contracts and of OCL standard library delivered by T.O.F.I.C. We summarized average metric values for the blank applications, i.e. obtained direct after transformation of model and constraints (Table 1). The third column includes average metric values for the applications with completed method bodies for selected functionality.

Table 1 Average metric (per module) of blank and completed application

Metric	Application		Implementation factor
	Blank	Completed	
IL	2,649	3,668	0.4
LOC	194	347	0.5
NM	175	198	0.22
NF	70	81	0.21
NT	27	31	0.2

Evaluating the amount of code that was manually supplemented to an application, we developed an Implementation Factor (*IF*) for each metric. It was calculated as a ratio of the difference between a metric of a blank application and its completed equivalent, over the metric in the completed application.

$$IF(Metric) = \frac{Metric(ComplApp) - Metric(BlankApp)}{Metric(ComplApp)} \quad (1)$$

The last column of Table 1 gives an average implementation factor for given metrics. The factor is averaged over the modules implemented by different subjects.

The implementation factor was decreased for some subjects who deleted a part of automatic generated code. There were two reasons of such situations. First, the generated code was not necessary in the implementation of the selected functionality. Second, the idea was not properly modeled, and the valid implementation was given in the code without updating the corresponding model. The second case points out the necessity of backwards traceability (from the code to models), which is still not sufficiently tool supported.

Taking into account the implemented functionality, at least one-third of the code of the final applications was automatically generated. To this code belonged implementation of all contracts.

Except of the above mentioned basic metrics, the NDepend tool examined some code quality factors. None of such factor was invalidated. The warnings of the code quality concerned a more advanced programming features, such as suggesting of refactoring of the complex operations, introducing a static type, introducing a sealed type for classes without ancestors, etc. Summing up, the transformed code has satisfactory quality and some application shortcomings were caused by the limited experience of participants in C# programming.

Application Testing

Unit tests were implemented for each module. Participants tested and compared the execution of the application without and with contracts. Two kinds of tests were designed:

– *Positive tests* that should end with success when the Code Contracts functionality was incorporated,
– *Negative tests* that should fail, invalidate a contract, when the Code Contract facility was switched on in the application. These tests could end successfully when the Code Contracts were switched off.

The code of contracts was covered by tests from 64 to 90 %. The rest of the application was only covered in 7–68 %. But it should be noted, that the tests were designed mainly for those parts that were responsible for realization of selected, implemented functionality.

The running of tests gave results that confirm the expectations and showed a correct execution of contracts.

Survey Results

All participants of the experiment answered all questions of the survey. The subjects estimated the labor intensity of the tasks comprised in the experiment. This intensity was expressed as a number of hours spent to complete each step. Average values per one participant are shown in Table 2.

Table 2 Distribution of estimated labor intensity of the project

Project phase		Average labor intensity (h)
1	Requirement analysis	5.6
2	Use case modeling and specification	7.5
3	Class modeling with design patterns and C# stereotypes	16.0
4	Specification of contracts in OCL	4.4
5	Model to code mapping	1.3
6	Implementation of application	8.6
7	Testing	3.3

The most of subjects spent a few hours on code modeling or implementation. The high effort (more than 30 h) was devoted to these activities only in exceptional cases, such as one student who preferred low level programming and in the contrary to his declaration did not know the C# language, or another student who rewrite the whole code in order to apply a desired, special C# library.

The usefulness of different functionalities supported by T.O.F.I.C. and RSA was estimated as a number of a range 0–10. The average results are summarized in Table 3. The best scores were assigned to the functionalities supporting UML modeling specialized with profiles (first three positions). Stereotypies of the profile can be easily allocated to one or many model elements, or alternatively, model elements with stereotypies are selected from a specialized palette and ready placed in a diagram. Without such tool support the specialized modeling would be a tedious work, hardly acceptable by model developers.

Some support concerning code generation was on average counted as less useful. Although, the opinions were diversified, e.g. separation of compilation units from the non-modified code (7th position) were found very useful by some subjects (mark 8–10), whereas as useless (mark 0–2) by others.

Impact of the technology on the project development was evaluated using the following scale: negative impact (−1), neutral (0) and positive (+1). The following three aspects of the development process were taken into account: code quality, its consistency with the specification and a task realization time.

The most of experiment participants positively charged compliance of the application with the specification. No one assessed this feature in a negative way, and the average grade was above 0.6. However, different scores were given to the impact on the application quality: negative, positive and neutral, whereas the negative predominated (−0.125 on average). The most of problems concerned a general MDE practice of code modeling and transformation and not the contract specification. The third factor, impact on the development time, was also differently evaluated, but the overall

Table 3 Usefulness of various kinds of tool support

Functionality	Average usefulness [0–10]
1. Visual stereotyping—graphical representation of stereotypies in UML diagrams and in project explorer	8.4
2. Palette extension—extension of the standard RSA menu with UML elements with stereotypes	7.9
3. C# Action Tool—additional view for rapid stereotyping of model elements	8.3
4. Map/Code separation—C# code model is separated from a model of the project structure (directories, compilation units etc.)	7.4
5. C# naming support—mapping of qualified names, C# types, *using* policies, etc.	5.6
6. OCL support—on-line verification of OCL expressions and code generation	4.5
7. Support for multiple code transformation—generated code reserved for the tool is separated from compilation units that are modified by a programmer	5.1

averaged score was negative (−0.25). The necessity of creating of models extends the whole time in the contrast to direct code development. However the participants admitted that some delays originated from the fact that they used the crucial technologies and their environment (OCL, T.O.F.I.C, Code Contracts) for the first time. During development of a next project some activities could be realized faster.

Threats to Validity

While interpreting experiment results several threats to validity should be taken into account [18].

Mono-operation bias is a threat as the experiment was conducted on a single development project.

Threats to external validity are conditions that limit the ability to generalize the results of experiments. The main source of such threat concerns students that were subjects of the experiment. However, students are often accepted as a substitute of more experienced professionals [19]. Moreover, when new approaches are concerned, the educational level in the reference to a contract specification language or modeling facilities can be comparable to subjects from the industry, which have former or less comprehensive educational background. The threat was also lowered by cooperating with students from the advanced course, attending the sixth semester of their studies and participated as volunteers in this kind of laboratory. They declared from few to about 80 months (38 of average) of active, usually industrial, experience in program development using OO languages.

In general, all subjects had a similar level of experience working with OO modeling method in UML, specifying contracts in OCL, using the CASE tool and development environment. It was based on the courses and laboratories comprised in the common curriculum. Whereas, the most discrepancy referred to the comprehension level of the C# language and skills of using the MS Visual Studio.

The statistical conclusion is limited by a small number of objects participated in the experiment (eight person). Therefore we can give some averaged results, but cannot evaluate statistically significant measures.

The survey results were provided to the supervising teacher after the end of the semester, i.e. after the project results had been evaluated. Therefore the collected opinion has no influence on the scores of the course given to the subjects and could be more outspoken.

Conclusion

Summing up our experiences and opinion of the experiment participants, the main advantages of the approach realized by T.O.F.I.C. can be summarized as follows:

– A structure of a project and C# code features can be precisely represented using appropriate stereotypes.
– Separation of modeling of a project and UML code increases flexibility of the solution and allows specifying various mappings for a single code model.
– Extended GUI covers graphical representation of model elements refined with the stereotypes.
– The refined UML model is validated in the accordance to the C# code generated from the model.
– OCL contracts are handled in models and transformed into calls of the Microsoft Code Contracts library.

The automatic generation of code contracts improves the consistency between specification and the source code. On the other hand, if developers are not involved in the maintenance phase or at least do not consider this activity are less prone to take care on this consistency. The most criticism is directed to the modeling activities. Many subjects believe that the same result can be directly implemented with contracts in a programming language, but without models.

The CASD process and other similar model-driven methodologies need strong tool support. Application of the T.O.F.I.C. tool with the RSA led to the high productivity, but the solution has still some limitations. Especially helpful was the full integration of the tool with the used modeling environment. Without a user-friendly support at each development step, effective traceability, and full reverse engineering, an MDE approach can hardly be successfully introduced into the industrial projects. Learning of a new technology requires some time and this is also an obstacle.

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Benchmarking GPenSIM

Reggie Davidrajuh

Abstract
Petri Nets is a family of modeling formalisms, consisting of various types of Petri nets with different interpretations and abstraction levels. General Purpose Petri Net Simulator (GPenSIM) is a new Petri Net simulator that implements many of the Petri Net types. This paper presents first a short introduction to the various types of Petri Nets; second, GPenSIM is tested for its implementation of various Petri Net types, using the classical benchmark known as the problem of "Buffered Producers-Consumers with shared channel (BPC)"; the modeling and simulations given in this paper show that the classical BPC problem can be solved by a variety of Petri Net extensions implemented in GPenSIM. In addition to the Petri Net extensions, some facilities are also provided in GPenSIM (e.g. resources), with which some specific problems can be conveniently solved.

Keywords
Benchmarking • GPenSIM • Petri nets • Discrete event systems

Introduction

Petri Nets is a family of modeling formalisms, consisting of a variety of Petri net types with different interpretations and abstraction levels. General Purpose Petri Net simulator (GPenSIM) is a new Petri Net simulator that implements many of the Petri Net types.

This paper presents first a short introduction to the various types of Petri Nets; second, GPenSIM is tested for its strengths and weaknesses using the classical benchmark known as the problem of "Buffered Producers-Consumers with shared channel (BPC)"; the BPC problem is solved using some of the Petri Net types such as the Ordinary Petri Net, and Petri Net extensions such as Petri Net with Inhibitor arcs, Color Petri Net, Petri Net with Priority, and GPenSIM resources.

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Petri Nets

The Petri Net Types

The Petri Nets is a family of modeling formalisms that includes the following five types of Petri Nets:

1. Ordinary Petri Net and Generalized Petri Net [1],
2. The extensions of Petri Net, e.g. Colored Petri Net, Petri Net with priority, Petri Net with inhibitor arc, etc., [2]
3. The extensions to incorporate time on Petri Nets; there are three main ways to incorporate time to ordinary Petri nets: Time Petri nets, Timed Petri nets, and Petri nets with time-windows [3, 4]
4. The restrictions of Petri Net (also known as Petri Net subclasses) such as Marked Graphs, Finite State Machines, simple Petri Nets, Free-choice Petri Nets, etc. [1, 5], and,
5. The fifth type of Petri nets are that created for solving specific problems *conveniently*. These types of Petri Nets neither have increased modeling power nor increased analytical power; perhaps, modeling convenience may be increased by using these types of Petri

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_46, © Springer International Publishing Switzerland 2015

Net while solving specific problems. GPenSIM provides "resources" for modeling and simulation of discrete systems conveniently, when large number of resources is involved in the system [1, 6].

Capabilities of Petri Nets

There are three *capabilities* involved when Petri Nets are used for modeling and simulations [6]:

1. The modeling capability: capability of the Petri Net for modeling a specific problem.
2. Analytical capability: whether the mathematical tools associated with Petri Net provide enough analytical power, and
3. Decision-making capability: does the Petri Net environment enables advanced decision-making from the modeling, simulations and analysis?

The following subsections explore the capabilities further.

Modeling Capability of Petri Nets

Ordinary Petri Nets do not possess the modeling power to model Turing machines, meaning it cannot be used for modeling certain discrete event systems [1]; this is true also for Generalized Petri Nets. Thus, extensions are needed to make the Petri Net capable of modeling Turing machine and any discrete event system.

The three following extensions are equivalent and are capable of modeling Turing machine: Petri Net with priority, Petri Net with inhibiting arcs, and enabling functions [7, 8]. However, the extensions increase modeling capability of Petri Nets at the cost of the analytical capability.

Analytical Capability of Petri Nets

Analytical power of Petri Nets generally comes from the manipulation of the incidence matrices such as D- and D. Theses incidence matrices are linear algebraic equivalent of the Petri Net graph. Matrix D- can be used to check whether an event can occur at a specific state, and together with matrix D, the reachability tree can be generated to represent all the possible states due to occurring of different events at different states [1, 9].

However, the extensions of Petri Net, while increasing the modeling power, they usually lower the analytical power. This is because, the extensions create a gap between the Petri net graphical versions and their incidence matrices; in other words, some information is lost while transforming the graphs into their linear algebraic equivalents (matrices). Thus, the incidence matrices no longer represent the Petri Net graphical version precisely, thus cannot be used for analysis as much as in the case of Ordinary Petri Nets.

In some cases, there is a need to increase the analytical power while not sacrificing the modeling power too much. This can be achieved by restricting Petri nets (as opposed to extensions). Marked Graph, Finite State Machines, Free-Choice Nets, and Simple Nets are some of the restricted Petri Nets (also known as Petri Net subclasses) that offer more analytical power, at the expense of modeling power. For example, marked graph is a restricted Petri Net in which every place is restricted to have exactly one input and one output transition; marked graph offers additional analysis techniques like minimum cycle time for performance analysis and optimization.

Decision-Making Capability

Is it possible to make advanced decisions, by simulation and analysis of the Petri Net models? Can the simulation and analysis of Petri Net models provide enough information for optimization and performance evaluation compared with the other alternative methodologies? In some cases, the Petri Net environment may not provide the necessary tools and functionality to make advanced decisions from the modeling and simulation of Petri Net models.

GPenSIM provides a set of additional functions that are useful for analysis of Petri net models. But the most important support is from the MATLAB environment: since GPenSIM is realized as a toolbox on MATLAB platform, various toolboxes that are available on the MATLAB platform (e.g. toolboxes like control systems, fuzzy-logic, optimizations, statistical, data collection, etc.) can be freely used with the Petri Net models developed with GPenSIM.

GPenSIM

GPenSIM is designed for increasing the decision making potentials from the Petri Net models, by implementing a variety of Petri Net extensions, supplemented by utility functions, and allowing easy interface to numerous functions and resources available on the MATLAB platform. GPenSIM is freely available from the website given in the reference section as [10]. Reference [11] gives an introduction to GPenSIM.

As GPenSIM is realized on MATLAB platform, a Petri Net model developed with GPenSIM consists of M-files. A typical Petri Net model consists of the following M-files:

- Petri Net Definition File (PDF): PDF defines the static Petri Net structure: this files declares the set of places, the set of transitions, and the set of arcs,
- A main simulation file (MSF): MSF declares the initial dynamics of the model. E.g. the initial marking (tokens in

different places), firing times of the non-primitive transitions, and system resources (explained later),

- COMMON_PRE: this file defines the enabling conditions (also known as guard conditions) of the transitions. Whenever a transition is enabled, the conditions defined in this file are checked to determine whether the enabled transition can start firing, and
- COMMON_POST: this file defines the post-firing actions (if any) for transitions. For example, after firing, a transition may release some resources that were used during the firing; releasing the resource can be coded in the COMMON_POST file.

The Benchmark: Producers-Consumers with Shared Channel

The classical problem of "Buffered Producers-Consumers with Shared Channel (BPC)" is used as a benchmark to test the strengths and weakness of different extensions of Petri Nets implemented in GPenSIM. The BPC problem was originally put forward in 1970s [12, 13].

Details of the BPC problem (Fig. 1):

- Four agents: there are four agents (or processes) involved; two of them are producers and the other two are consumers.
- Producer-consumer pairs: The first producer (P1) produces items only for the first consumer (C1). Similarly, the second producer (P2) produces items only for the second consumer (C2).

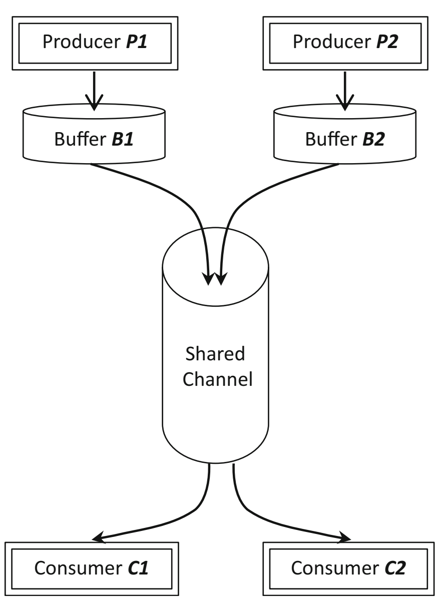

Fig. 1 The Benchmark: "Buffered Producers-Consumers with Shared Channel" (adapted from [1])

- The buffers: Items which are produced by the producers are kept in the corresponding buffers (B1 and B2) until transported away to the consumers by the shared channel.
- The shared channel: Bandwidth of the shared channel is one, meaning only one item can be transported through the channel at a time.
- The producers are passive: Producers merely produce items and place them into the respective buffers.
- The consumers are active: Consumers actively control the use of the channel so as to take appropriate items that are transported through the channel.
- Preferential treatment: the producer-consumer pair P1-C1 has priority over the P2-C2 pair; this means, transportation of items from B1 to C1 has priority over items from B2 to C2. This also means, as long as the B1 is nonempty, the transportation channel only transports items from B1 to C1.

In summary, the solutions for the BPC problem must satisfy the following two requirements:

- Requirement-1: Only one item at a time through the shared channel, and
- Requirement-2: P1-C1 pair has priority over P2-C2 pair

Literature study shows that the BPC problem is already solved many times, using a variety of Petri Net extensions [1, 12–14]. Hence, this work providing the yet another solution to the classical BPC problem, serves only as a benchmark for testing GPenSIM.

Benchmarking Gpensim

In this section the problem of buffered producers-consumers with shared Channel (BPC) is solved in using five different Petri Net types: first with ordinary Petri Net, the next three solutions using different Petri Net extensions, and the final one with GPenSIM convenience:

1. Ordinary Petri Net,
2. Petri Net with Inhibitor arc extension,
3. Colored Petri Net,
4. Petri Net with Priority extension, and
5. Using GPenSIM resources.

The following subsections present the solutions.

Ordinary Petri Net Model

Figure 2 shows an Ordinary Petri Net model for the BPC problem. The model shown in Fig. 2 satisfies the first requirement ("only one item at a time through the shared channel") by utilizing a semaphore mechanism consisting of the place pCh and the transitions tCh1 and tCh2; the transitions tCh1 and tCh2 that transport through the channel

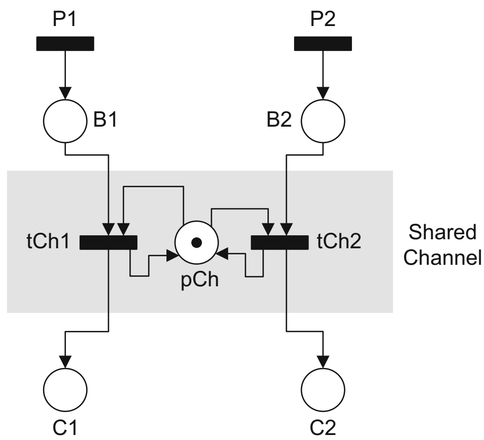

Fig. 2 Ordinary Petri Net model of the BPC problem

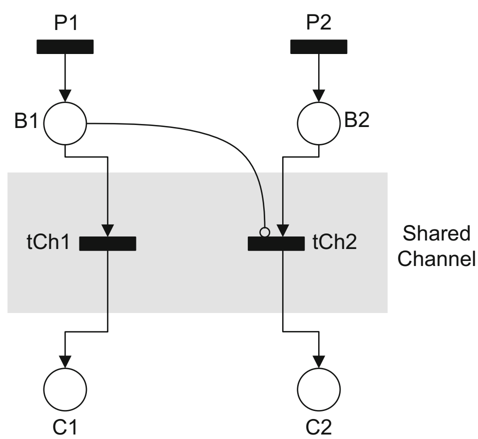

Fig. 3 Petri Net model using inhibitor arc extension

can be enabled at the same time, but can never fire simultaneously, as there is only one token ('semaphore') in common the input place pCh.

Though the first requirement is satisfied, the model will not satisfy the second requirement ("P1-C1 pair has priority over P2-C2 pair"). This is because, if both tCh1 and tCh2 are enabled at the same time, the model does not prioritize tCh1 over tCh2; when tCh1 and tCh2 are enabled at the same time, it is indeterminate whether tCh1 will fire blocking tCh2 from firing. As there is no mechanism in Ordinary Petri Net to give priority to tCh1, Ordinary Petri Net cannot be used solve the BPC problem.

Petri Net Model with Inhibitor Arc Extension

Figure 3 shows the model the BPC problem using Petri Net with the inhibitor arc extension. In the model shown in Fig. 3, the inhibitor arc B1-tCh2 makes sure that tCh2 is disabled as long as there are items in buffer B1. This means, if there are items in B1, only tCh1 is enabled and allowed to

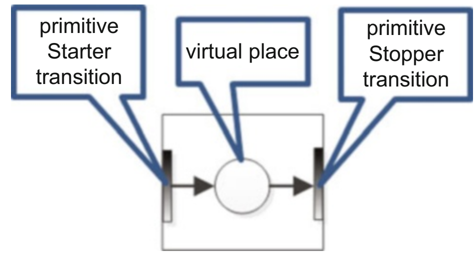

Fig. 4 Composition of a non-primitive transition in GPenSIM [15]

fire. Only if there are no items on B1, then tCh2 is enabled and allowed to fire as long as there are items in B2. Thus, the two requirements are seemingly satisfied by the model. However, there is one big difference between the model shown in Fig. 3 and its GPenSIM realization:

In the model shown in Fig. 3, all the transitions are assumed to be primitive transitions satisfying the atomicity property. However, in GPenSIM, the transitions are generally non-primitive; each non-primitive transition in GPenSIM takes time ('firing time') to fire, and consists of a primitive starter transition, a virtual place, and a primitive stopper transition; see Fig. 4.

Non-primitive property of transitions in GPenSIM makes the model shown in Fig. 3 not fulfill the first requirement of the BPC problem, in some situations: imagine a situation where the buffers B1 and B2 have one item (token) each. Since tCh2 is disabled because of the item in B1, tCh1 is enabled and starts firing by removing the item in B1. However, tCh1 will takes some time to complete firing (tCh1 takes some time to transport the item through the channel). In the meantime, absence of items in B1 and presence of an item in B2 makes tCh2 enabled and it starts firing as well. This means, both tCh1 and tCh2 are firing at the same time, breaching the first requirement of the BPC problem. Thus, an additional enabling condition must be imposed on the transition tCh2: "tCh2 is only enabled if tCh1 is not firing".

It is possible to generate a reachability tree for the Petri Net model with inhibitor arcs, by making the unbounded system shown in Fig. 3 into a bounded system; this conversion from an unbounded into a bounded system can be done by limiting the firings of the 'cold' transitions P1 and P2 by adding an input place each, and inserting a limited number of initial tokens into these input places. The resulting bounded Petri Net gives a useful reachability tree for analysis. Thus, using the inhibitor arc extension, analytical part of the model seems somewhat improved over its ordinary Petri Net model, even though, theoretically, the analytical power should remain the same for both Petri Net types. Theoretically, for bounded systems, the modeling capability of Ordinary Petri Net and the formalism with inhibitor arcs should have the same effect.

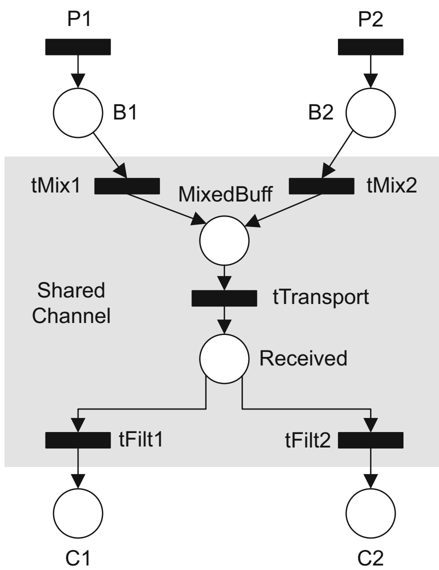

Fig. 5 Colored Petri Net model

Colored Petri Net Model

The Colored Petri Net model of the BPC problem is shown in Fig. 5. In order to perform color manipulation, the model possesses some more transitions and places compared with the model using the inhibitor arc extension.

The color extension is mainly to impose the following actions and enabling conditions on transitions:

- When the transition tMix1 fires, it adds a tag (color) "P1" on the items from B1 and deposits them into the common buffer 'MixedBuff'. Similarly, transition tMix2 adds tag "P2" on the items from B2 that are also deposited into the common buffer MixedBuff.
- Transition tTransport is the only transition that transports items through the channel. tTransport picks an item from MixedBuff, transport the item, and deposits them into the buffer 'Received'. However, when picking items from MixedBuff, tTransport prefers item with tag "P1".
- The buffer at the end of the channel is 'Received'. 'Received' may hold items with tag "P1" and "P2". Thus, transition tFilt1 (short for Filter-1) takes only the items with tag "P1" and deposits them into the consumer C1; similarly, tFilt2 takes only the items with tag "P2" and deposits into the C2.

Simulations show that the Colored Petri Net model satisfies both requirements. However, it was only possible with additional structures such as the pre-channel common buffer "MixedBuff", and the post-channel common buffer "Received", both of them are not defined in the BPC problem.

Figure 6 shows the COMMON_PRE file coding the color manipulations for enabling the different transitions.

```
function [fire,transition] = COMMON_PRE
(transition)

tname = transition.name;
tokID = 0;  % initially

if strcmpi(tname, 'tMix1'),
  transition.new_color='P1'; % color of tokens
  fire = 1;
elseif strcmpi(tname, 'tMix2'),
  transition.new_color='P2'; % color of tokens
  fire = 1;
elseif strcmpi(tname, 'tTransport'),
  % tTransp "prefer" tokens with color 'P1'
  tokID = tokenColors('MixedBuff',1,{'P1'});
  fire = 1; % "prefer"
elseif strcmpi(tname, 'tFilt1'),
  % tFilt1 must select tokens with color 'P1'
  tokID = tokenColors('Received',1,{'P1'});
  fire = tokID;
elseif strcmpi(tname, 'tFilt2'),
  % tFilt2 must select tokens with color 'P2'
  tokID = tokenColors('Received',1,{'P2'});
  fire = tokID;
else
  % allow P1 and P2 just fire
  fire = 1;
end;
transition.selected_tokens = tokID;
```

Fig. 6 COMMON_PRE file for coding enabling conditions

The reachability tree obtained for the Petri Net model with color extension is not useful; this is because, the color manipulations that enable the transitions are lost in translation from the graph to the incidence matrices; thus colored Petri Net model can only be used for simulations and analysis of the results; structural analysis using the incidence matrices are not possible.

Petri Net Model with Priority Extension

The Petri Net structure (static graph) of the model with priorities is exactly the same as the one for Ordinary Petri Net (shown in Fig. 1). However, in the model with priorities, the transition tCh1 is given higher priority than its competitor tCh2, thus tCh1 is always allowed fire if it is in conflict with tCh2. Thus, the model with priority extensions satisfies both requirements.

However, the reachability tree obtained for the Petri Net model with priority extension is again the same as the one that is obtained from the ordinary Petri Net model. This is because, as in the case of colored Petri Net model, priority manipulation is lost in the translation of the graph model into the incidence matrix. Thus, the analytical power is decreased in this model.

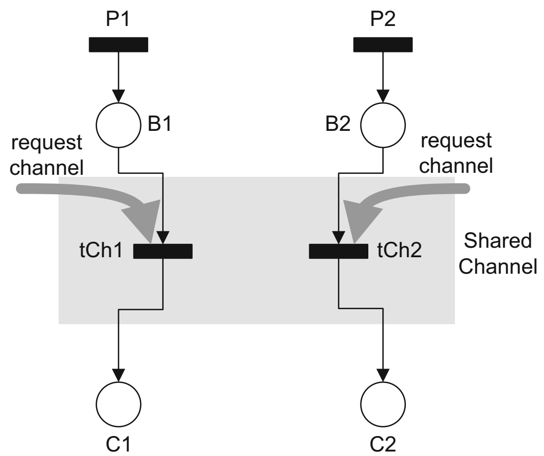

Fig. 7 Petri Net model using GPenSIM Resources

```
function [fire, trans] = COMMON_PRE(trans)

tname = trans.name; %name of the enabled trans

if strcmpi(tname, 'tCh1'),
    fire = request(tname, {'Channel',1});
elseif strcmpi(tname, 'tCh2'),
    fire = request(tname, {'Channel',1});
else
    fire = 1; %allow other transitions to fire
end;
```

Fig. 8 COMMON_PRE file for coding enabled transitions requesting resource

```
function [] = COMMON_POST(transition)

% release all the resources used by
% this transition when it completes firing
release(transition.name);
```

Fig. 9 COMMON_POST file for coding transitions releasing resource at the completion of firing

Using GPenSIM Resources

As stated in the beginning of this section, the first model is based on Ordinary Petri Net; the next three models are based on Petri Net extensions (inhibitor arcs, color extension, and priority extension) that are necessary to model a system that is otherwise not possible to model with Ordinary Petri Net; thus, these three extensions are out of necessity. This subsection presents an enhancement (convenience) known as "resources" that is available in GPenSIM for creating compact models.

Literature report the huge size of Petri Net models of discrete event systems as one of the main problems associated with Petri Nets [16]. This problem is especially true when modeling resource scheduling problems; even for a resource scheduling problem with few activities competing for a few resources, the resulting Petri Net model can be very large [16].

GPenSIM offers "resources" with which compact models of resource sharing and resource scheduling can be obtained. By using resources in GPenSIM, the size of Petri Net models are reduced as mainly the activities are shown in the Petri Net models; the resources are grouped into two groups as focal resources and utility resources, and only the focal resources are shown in the Petri Net model and the utility resources are pushed into the background [16].

Figure 7 shows the Petri Net model that uses GPenSIM resources. In this model, the channel is assumed as a utility resource thus not represented in the model as a place; total absence of the channel makes this model the most compact among all the models.

When B1 and/or B2 are filled with items, tCh1 and/or tCh2 is enabled; in order to start firing (start transportation), these two transitions request the channel. Only if the channel is available, then the underlying system will grant the channel to the requesting transition, and the transition can start firing. When the transition completes firing, the channel is released to the underlying system. Transitions tCh1 and tCh2 requesting the resource channel to start firing is coded in the COMMON_PRE file; see Fig. 8.

The COMMON_POST file is used for coding the transitions releasing the channel after firing; see Fig. 9.

The Petri Net model that uses GPenSIM resources (Fig. 7) satisfies the first requirement of the BPC problem (on channel bandwidth). However, the second requirement on the higher priority for P1-C1 pair is not satisfied, as the resource usage has nothing to do with priorities. Hence, unless the resource usage is combined with the extension for priority, this model will not solve the BPC problem.

The Petri Net model that uses GPenSIM resources is intentionally compact and less detailed; the reachability tree for the Petri Net model will be trivial and not useful for any analysis. Hence, it is obvious that analytical power is sacrificed in order to make simple and compact models; the resource details are taken away from the Petri Net model, and moved into the underlying system (GPenSIM) that manages the resources.

In summary, the advantages of using resources in GPenSIM are twofolded:

- Petri Net models of manageable size can be obtained even for a system with many resources
- GPenSIM provides a detailed report of resource usage.

Discussion

In this paper, we investigate a new Petri Net simulator called GPenSIM, by testing some of the extensions and facilities that have been implemented on this tool. The classical Buffered Producers-Consumers with Shared Channel (BPC) was used as a benchmark to measure the ability of GPenSIM in using the different Petri Net extensions.

Theoretically, the modeling capability of Ordinary Petri Net is exactly the same as of Colored Petri Net; Petri Net with inhibitor arcs should have the same modeling capability of Petri Net with priority, and these two Petri Net types should have much higher modeling power than the Ordinary Petri Net; this is because, the two Petri Net types—Petri Net with inhibitor arcs and Petri Net with priority—can model Turing machines, whereas ordinary Petri Net cannot (this is not true for bounded systems); when modeling bounded systems, all the Petri Net types should possess the same modeling power. All the Petri Net models shown in section "Benchmarking Gpensim" are unbounded systems (as initial tokens are continuously generated by 'cold' transitions P1 and P2, which are always enabled); simulations of these models correctly prove the theory.

The modeling and simulations given in this paper show that the classical BPC problem can be modeled and analyzed by a variety of Petri Net extensions (such as inhibitor arc, color, and priority). In addition to the Petri Net extensions, some facilities are also provided in GPenSIM, e.g. resources. GPenSIM's resources are a convenient way for minimizing the size of the Petri Net models, especially when there are many resources involved in the discrete event system. When using resources, GPenSIM provides a set of functions to make decisions from the compact Petri Net model, thus increasing the decision power of the environment.

GPenSIM program codes for the Petri Net models discussed in this paper are simple and easy to study. The implementation code for simulation is not shown here due to space limitation, but fully available at the webpage indicated in the reference as [17].

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A Field Experiment of System to Provide Tourism Information Using Image Recognition Type AR Technology

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Abstract

In recent years, the growth of the Internet and communications networks for mobile phones have led to the development of services to provide tourism information via mobile information devices at tourist sites. In Japan, various tourism-related organizations are conducting field tests to improve tourist satisfaction through the provision of appropriate information.

In this research, we propose a tourism information system using augmented reality. The proposed system uses image recognition type augmented reality to superimpose tourism-related video content on photo images taken with a smartphone. Field experimentation of this system was conducted with tourists walking through the Otaru Canal area. As a result, a basically good evaluation was obtained regarding the ease of operation and appeal.

Keywords

AR • Image recognition • Tourism information system

Introduction

In recent years, the mode of tourism has shifted from group travel, which was previously the mainstream, to small group travel by families, groups of friends and individuals. In addition, the scope of tourism information linking tourist destinations with tourists has grown "much broader and deeper" due to the development of high-speed information and communications infrastructure, and the dissemination of mobile phones and other mobile devices.

Against the backdrop of trends such as changes in international economic conditions, and development of a high-speed transportation system in Japan, tourism has been positioned as a key national policy, and the Tourism-based Country Promotion Act was passed in 2006. As a result, the Tourism Nation Promotion Basic Plan was formulated in 2007, and various measures are being promoted, to achieve specific targets regarding issues such as "developing attractive tourist destinations, and an environment to promote tourism".

In the Machi Meguri Navigation Project (referred to below as "Machi Navi") organized by the Ministry of Land, Infrastructure, Transport and Tourism (MLIT) in FY2006 and FY2007, services to provide tourism information using mobile phones and other media were rolled out in 56 regions throughout Japan, and efforts were made to improve satisfaction by providing appropriate information to tourists. In demonstration experiments using this Machi Navi system, it was shown to be "possible to provide previously unavailable tourism information by utilizing technologies such as IT." However, some issues were also pointed out at the same time, i.e., "there are cases where mobile phones are not an efficient means of providing information due to factors such as the difficulty of mastering operation" and "the respective advantages of paper media and IT equipment are not being fully exploited."

Therefore, in this research, the authors focus on augmented reality (AR) as a technology for resolving such issues involved in the provision of tourism information,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_47, © Springer International Publishing Switzerland 2015

and propose a system for providing tourism information using smartphones for use by tourists at their destinations. AR is a technology which attempts to augment/extend reality by adding digital information to the actual world [1]. It relates the virtual world to the actual world, and is attracting attention as a technique for providing information to support activities of people in the actual world.

In the proposed system, tourism information is smoothly presented by using the built-in camera of a smartphone to take a picture of a photographic image of a tourist spot or other point of interest printed on a paper map, and then automatically overlaying a display of video content providing a detailed explanation on the photo image on the screen. In this way, the system strives to achieve smoother and more appealing tourism by improving the ease of operation of the system for providing tourism information using mobile devices, enabling easier access to tourism information, and better supporting tourists from the standpoint of providing information.

Issues of Mobile Tourism Information Systems

Process of Tourism Behavior

The travel behavior of tourists is regarded as having five stages: (a) Anticipation, (b) Outward trip, (c) Behavior at the destination, (d) Return trip, and (e) Reminiscence [2]. At each of these stages, the tourist requires information on tourism and movement. In this research, the discussion focuses on the "Behavior at the destination stage" between the "Outward trip" and "Return trip," i.e., tourism information at the destination.

Issues

With "Machi Navi," an effort was made to provide tourist information at each destination using mobile devices such as mobile phones and PDAs. Based on the results of evaluating this effort, and on prior research, the following summarizes issues when providing tourism information at a destination.

Systems to provide tourism information using mobile phones were developed as the main means of providing tourism information at 41 of the 56 areas throughout Japan (73 % of the total). With these systems, techniques were devised so that users could obtain a wealth of tourism information, even while on the move, by comparing with conventional fixed guidance signs and paper media such as town maps, and using features such as the search function.

However, the results of the "Machi Navi" project also revealed a number of issues. First were issues relating to ease of operation of the device for providing tourism information. In providing information using mobile devices, if a large amount of information is provided, then a greater amount of operation tends to be necessary to reach the information needed by the tourist.

Second are issues relating to the content of tourism information. Tourism information provided by people on the destination side is often selected based on thinking in the local area, and in some cases there is inadequate information on the dining and shopping tourists are looking for. Furthermore, even if such information is provided, the content and amount of information may have been insufficiently vetted, and may be hard to understand, due, for example, to too much text which is hard to read.

Third are issues relating to appeal of tourism. At tourist destinations where people want to relax and enjoy the historical or natural scenery, the "tourism atmosphere" may be ruined if techniques are not devised regarding the content and display method for information displayed on the small screen of a mobile phone.

Based on the above, the issues when providing tourism information at a destination using mobile devices can be summarized in the following three points:

Issue 1: If a large amount of information is supplied to the tourist, the steps of operating the mobile device will increase proportionally, and it may be impossible to reach the needed information with simple operation.

Issue 2: The content and amount of tourism information provided on the destination side has been insufficiently vetted, and easy-to-understand tourism information is not provided.

Issue 3: There is inadequate innovation and consideration to increasing the enjoyment and interest of tourism by exploiting the characteristics of mobile devices.

Related Research

Previously, a variety of studies have been conducted relating to tourism information systems for supporting tourists at their trip destinations. Developed systems include, for example, a tourism guidance system which displays image and text information such as tourist spots on a tablet PC [3] and push-type systems for providing tourism information using active RFID and mobile phones which can respond to diverse types of tourists [4].

As mobile AR systems for tourists, one AR application uses the camera function of smartphones [5]. This prototype system can present content consisting primarily of text and photographs, and it has not reached the point of displaying video content. Although it is not for tourists, a mobile AR system using paper maps and mobile phones [6] has also been proposed. However, this AR application displays icons on the screen of the mobile phone which reflect the paper map, and this enables distribution of video, but it is based on location information obtained from GPS, and the system proposed here uses a different approach.

Proposed System

System Design Concept

As an approach aiming to resolve Issues 1, 2 and 3 described in section "Issues", three system design concepts were established, one corresponding to each issue [7]:

Concept 1: To ensure that the atmosphere of tourism is not ruined, it must be possible to easily obtain the desired information, without the need for the tourist to carry out a complicated procedure to operate the mobile phone.

Concept 2: In introducing tourist spots, restaurants and so on, the information must be communicated to the tourist in an easy-to-understand form, by using more effective media such as video and audio.

Concept 3: When presenting information to the tourist, the enjoyment and interest of tourism must be improved by providing the tourist with fresh excitement and inspiration. This will be achieved by devising expressive techniques for video content.

System Architecture

To achieve the system design concepts of "devising expressive techniques" and "handling video," it was felt that an image recognition type AR would be appropriate, and it was decided to develop the system as an AR application for mobile terminals.

Smartphones

Smartphones equipped with touch panels were selected to provide tourists with operation that is simple and easy to understand. Photography of images to be recognized and shaping/playback of tourism video content was achieved as a smartphone AR application.

Image Recognition Server

The image recognition server holds comparison source images for analyzing and matching images photographed with the AR application. In addition, the server performs matching of comparison source images against the photographed image sent to the image recognition server by the AR application, and has a search function for that purpose.

Content Server

The content server holds the video content, audio information and other data corresponding to the comparison source images stored in the image recognition server. It has a function for sending information such as the matching tourism video content to smartphones, and a function for keeping URL information for related web pages.

Implementation of Prototype

A prototype was implemented in order to verify the basic functions of the proposed system [7]. Figure 1 shows the composition of the prototype. The following describes the system composition and function of the implemented prototype.

Composition of Prototype

The HT-03A made by HTC was used as the smartphone. Workstations made by HP were used as the image recognition server and content server. Smartphones and workstations were connected via the mobile phone communications network and the Internet. The tourism map was printed in A3 size, and the map of the tourist destination was placed near the center. Photographs of tourist spots, restaurants and so on (images to be photographed with the mobile phone) were placed around the map, with text content explaining each photo.

Functions of Prototype

The functions of the implemented prototype are described below, in the sequence photography, analysis and playback.

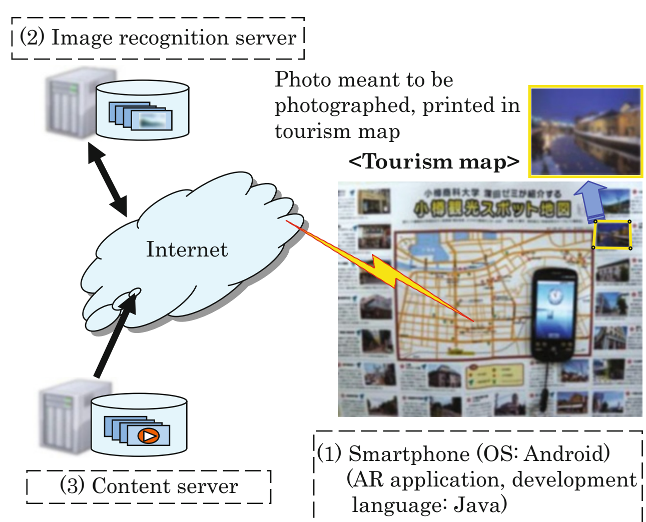

Fig. 1 System architecture of the prototype

Fig. 2 Photographing a picture with a smartphone (*left*) and playing back video content (*right*)

Photographing the Pertinent Image

The tourist photographs one of the photos on the map using the smartphone AR application. The photo is taken so that, among the photo images on the tourism map, the image of the tourist spot the tourist is interested in is included in the shot (Fig. 2, left). In terms of smartphone operation, a special-purpose icon is placed on the screen, and the user can easily take the photograph by simply gently touching the icon.

Analysis of Photographed Image

The AR application sends the photographed image to the image recognition server for analysis, and then video content is played back based on the analysis results. At this time, the video is shaped to fit in the outline of the photo appearing in the photographed image, and the video content is overlaid (Fig. 3). In addition, when playing back the overlaid video content, the first frame is made the same as the comparison source image.

By using these expressive techniques, a visual effect is created where the photo on the smartphone screen seems to start moving, and when the tourist first looks at the video content, they get a feeling of freshness and surprise.

Local feature values robust under rotation, enlargement and reduction were used for visual recognition. Local feature values are extracted beforehand from comparison source images, and stored in the image recognition server. Then feature values are extracted for the photographed image sent in from the AR application, and compared with the set of feature values for the comparison source images. If the number of matching feature points exceeds the threshold value for a comparison source image, that image is regarded as appearing in the photographed picture. The transformation matrix is obtained from the coordinate information for the feature points, and the coordinate information of the comparison source image is calculated.

Playing Back Video Content

The AR application overlays the pertinent video content downloaded from the content server onto the photographed image, and then plays it back (Fig. 2, right). In addition, tapping the screen during playback is taken to be a trigger for switching to video content with audio, or to a related web page. This makes is possible for the tourist to arrive at detailed information on a tourism facility he or she is interested in via simple operation.

Prototype Use Procedure

The system user operates it in the following procedures.

- Step 1: Starting the AR application
  The user starts the application by touching the AR application button on the smartphone screen.
- Step 2: Photographing the pertinent image
  When the AR application is started, the system switches to the camera mode. The user takes a picture of a photograph printed on the tourism map using the smartphone camera.
- Step 3: Sending the photographed image
  The photographed image is automatically sent to the image recognition sensor via the mobile phone communication network.
- Step 4: Analyzing the photographed image
  The image recognition server analyzes whether or not a comparison source image appears in the photographed image, and if so, where it appears.
- Step 5: Acquiring the analysis results
  From the image recognition server, the AR application acquires information on which comparison source image appears in the photographed image, and coordinate information indicating where it appears in the photographed image.
- Step 6: Acquiring content
  Based on the information acquired in Step 5, the AR application downloads video content relating to the photographed comparison source image from the content server.
- Step 7: Shaping and playing back video
  The AR application first shapes the downloaded video content, using the coordinate information acquired in Step 5, to match the outline of the image meant to be photographed within the overall photographed picture. Then the video content is overlaid on the

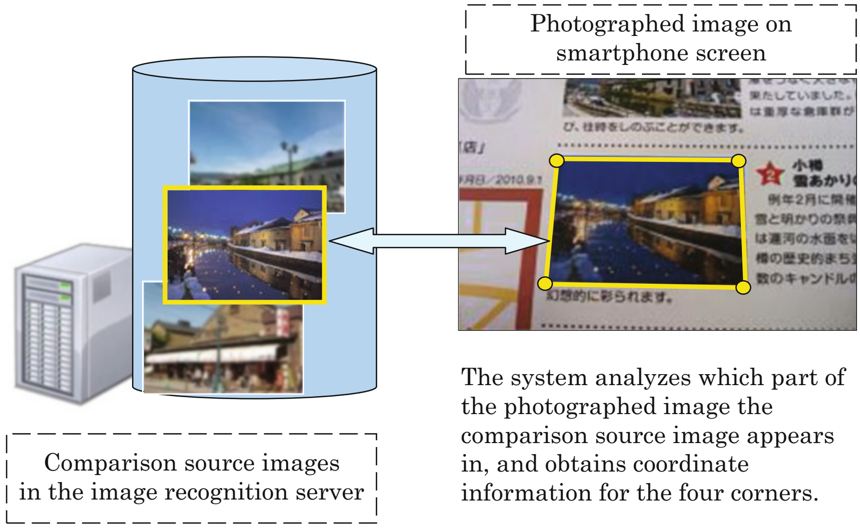

Fig. 3 Analysis image of photographed picture and overlaid display on paper map

Table 1 Categories and number of items of tourism video content used in the experiment

Tourism resource/facility	Number of video content items	Existence of audio information etc.
Urban scenery (Seasonal scenes, street scenes etc.)	5	Have audio information. However, street scenes are video only
Historical scenery (streets scenes photographed in the early Showa period)	3	Added date information photographed with subtitles
Historical buildings	3	Have both audio information and subtitles
Famous restaurants and souvenir shops	3	Have both audio information and subtitles
Local shops (featured infrequently in travel magazines etc.)	5	Video only, no audio information or subtitles

image. Furthermore, if the user taps this video, the system will download video content and provide a full-screen display.

Field Experiment Using Prototype

Overview of Evaluation Experiment

A field experiment was conducted using the implemented prototype, with the aim of evaluating the basic functions of the proposed system from the standpoint of the tourists who are prospective users of the system. The subjects in this case were pedestrian tourists whose tourism start and end point was taken to be JR Otaru Station. The field was an area centered on the Otaru Canal in Otaru City, Hokkaido. This is an area with many typical tourist spots of Otaru, located in a region about 2 km long east to west, and 1.5 km long north to south.

The field experiment was conducted from September 16–25, 2011 for 7 days, excluding the 20th, 21st and 22nd, which were working days between holidays. The time period each day was, as a rule, 8 h from 10 a.m. to 6 p.m.

Tourism Video Content for Experiment

Table 1 shows the categories and nature of the tourism video content prepared for this demonstration experiment. Photographs relating to the 19 content items shown in the table were printed on the tourism map prepared for the experiment. As the tourism facilities, restaurants and other locations listed on the tourism map, tourist spots frequently visited by tourists who come to Otaru were selected based on the results of a tourist movement survey conducted by Otaru City. In addition, information unique to the local area, of the sort tourists seek out, was gathered by asking the opinions of workers in Otaru City and local advertising related companies etc.

Experimental Method

At JR Otaru Station, which was chosen as the field experiment, a booth for explaining the experiment (referred to below as the "booth") was set up outside of the entrance hall. A system was adopted wherein smartphones installed with the AR application were loaned to tourists visiting the booth for them to try out. Figure 4 shows the scene at the booth.

In this field experiment, the functions implemented in the prototype were explained to tourists who wished to use the AR application for actual tourism in Otaru (referred to below as “actual use tourists”), and then the smartphone was loaned out together with the tourism map. During their touring, they were asked to use the system freely, without any accompaniment by experiment staff. When they returned to the booth after finishing their touring, they returned the smartphone, and were handed a questionnaire form which they were asked to fill out.

Result and Discussion

This section summarizes and discusses the results of the questionnaire evaluation conducted after the end of the experiment. The subjects of the demonstration experiment were comprised of both groups and individuals. Therefore, there was one questionnaire for each individual tourist, but for groups there were a number of cases: the case where only one representative of the group responded, the case where a few members of the group responded, and the case where all the members of the group responded. A total of 21 questionnaires were collected.

To evaluate the proposed system, it was decided to evaluate the degree of satisfaction of tourists who tried out the prototype, based on the perspective of utility. The evaluation items were determined to be: ease of operation, ease of understanding, and appeal. The questions were designed based on these three perspectives. Each question could be evaluated with one of five levels, with the highest evaluation being 5 and the lowest 1. These evaluations were converted to points, with the most positive evaluation scoring 5 points, and the most negative 1 point. Figure 5 shows the questionnaire results for each evaluation item.

Fig. 4 Scenes of the booth for the experiment

Ease of Operation

To assess linkage with solving Issue 1 in section “Issues”, subjects were asked about the ease of operation of the prototype corresponding to Design Concept 1. Item ① in Fig. 5 shows the results relating to ease of operation. In the evaluation results, 76 % of actual use tourists gave a positive evaluation, and the average was 4.0, so the obtained results were basically good. It is thought that these results indicate appreciation of the simplicity of the system, whereby tourism video content is automatically played back through the single operation of photographing an applicable photograph with a smart phone.

Ease of Understanding

For Item ②, the evaluation item was set corresponding to Design Concept 2 to examine solution of Issue 2, and subjects were asked about the ease of understanding

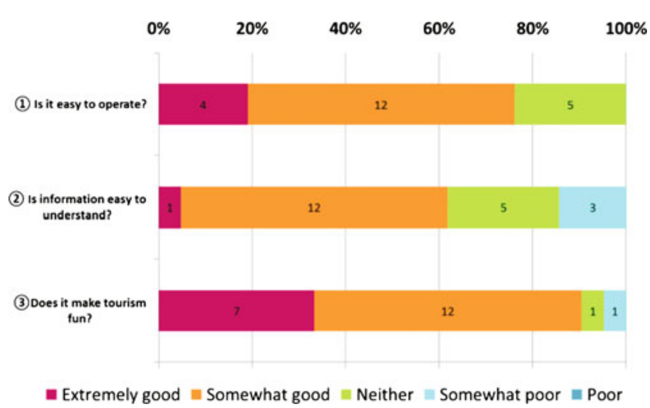

Fig. 5 Result of effectiveness evaluation

information presented as tourism content. In the results, 62 % gave a positive evaluation, and the average was 3.5.

It is believed that, the inability to sometimes achieve video playback in a single operation had an affect on the evaluation in this case. In addition, one of the comments in the space for free comments on the questionnaire was that "It is hard to hear the audio, so it would be best to wear earphones, or redesign the system so the audio can be turned up." In the actual outdoor environment, the noise was louder than expected beforehand, due to factors such as the running noise of automobiles, and as a result there were likely cases where the audio explanation could not be heard.

Appeal

Two items were set for evaluation corresponding to Design Concept 3 for resolving Issue 3. For Item ③, subjects were asked whether, by using the proposed system, "they felt it made touring Otaru more fun and interesting." In this evaluation of appeal, 91 % gave a positive evaluation, with an average of 4.2, and thus the evaluation was good. This is thought to be connected to the fact that, with the AR application, photographs on the screen appeared to start moving, and the realization of this expressive technique gave the tourists a sense of freshness and surprise, and boosted the appeal of the system.

Conclusion

In this research, a system for providing tourism information using image recognition type AR technology was proposed, for use by tourists at their destinations. The system is based on design concepts which aim to resolve the issues relating to providing tourism information using mobile devices. A prototype was developed based on the proposed system architecture, and a demonstration experiment was carried out using as subjects tourists who visited Otaru City, Hokkaido. A basically good evaluation was obtained regarding the ease of operation and appeal of the basic functions implemented based on the design concepts.

In terms of issues with the proposed system, the first is improving the image recognition accuracy. The authors would like to continue improving the technology so that stable image recognition can be achieved even in the outdoor environment. The next task will be to consider the use of devices such as Bluetooth monaural earphones to address the issue of difficulty hearing the video content audio while outside.

As an issue for achieving practical use, an important requirement is how to maintain freshness of the tourism video content, and thereby ensure the appeal of tourism. A program of system management must be established to ensure there is no stagnation in updating of the provided tourism information.

Acknowledgment

This research received support for Scientific Research in Priority Areas at the Otaru University of Commerce. In preparing the tourism map used in the demonstration experiment, and procuring tourism video content and smartphones, tremendous support was provided by staff of the Otaru Tourism Promotion Office (Industry and Ports Division, Otaru City), Otaru City General Museum, Hokkaido Branch of NTT Docomo, Inc., and O-Plan Co., Ltd.

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Metrics for Data Warehouse Quality

Bharti Suri and Prerna Singh

Abstract

Information in organizations is managed efficiently by adopting data warehouses. Organizations are using data warehouses for integrating data from various heterogeneous sources in order to do analysis and make decision. Data warehouse quality is crucial because lack of quality in data warehouse may lead to rejection of the decision support system or may result in non-productive decision. A set of metrics have been defined and validated to measure the quality of the conceptual data model for data warehouse. In this paper, we first summarize the set of metrics for measuring the understand ability of conceptual data model for data warehouses. We focus on providing empirical validation by the family of experiments performed by us. The whole empirical work showed us that the subset of proposed metrics can be used as an indicator of conceptual model of data warehouses.

Keywords

Conceptual model • Data warehouse • Quality

Introduction

In order to solve the problem for data to provide correct information, organizations are adopting a data warehouse which is defined as a subject oriented, integrated, non volatile data that support decision process [1].

Information quality of the data warehouse comprise of the data warehouse quality and the data presentation quality. Database management system quality, data quality and data model quality directly influence Data warehouse quality. Data model quality can be categorized into conceptual, logical and physical [2] (Fig. 1).

According to Manuel Serrano et al., data warehouses are used for making strategy decisions, Data warehouse quality is crucial for organizations [3]. They have used global methods for defining and obtaining correct metrics. They performed theoretical validation of the proposed metrics and had carried out family of experiments for empirical validation of metrics. The above study has been replicated in this paper by performing empirical validation on the metrics defined by global methods [3].

In this paper, the set of metrics that are already defined for conceptual model for data warehouse [3] are summarized. Then empirical validation is performed by conducting family of experiments by us. The family of experiments showed that several of the proposed metrics are the practical indicator of the understanding ability of the conceptual model of data warehouse.

The paper is structured as follows: Section "Related Work" summarizes the most relevant related work, Section "Metrics" defines the method used for defining metrics, and Section "Empirical Validation" defines the family of experiments carried out to prove empirical validation of metrics. Section "Conclusion and Future Work" defines the conclusion drawn from the family of experiments.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_48, © Springer International Publishing Switzerland 2015

Related Work

In this section, the related work is defined regarding (a) multidimensional modeling, (b) metrics for software system data warehouses.

Multidimensional Modelling

A variety of multidimensional data models have recently been proposed by both academic and industry communities. The basic notions are the dimension and the data cube. A dimension represents a business perspective under which data analysis is to be performed and is organized in a hierarchy of levels which correspond to different ways to group its elements [4].

A data cube represents factual data on which the analysis is focused and associates measures and coordinates defined over a set of dimension levels [4].

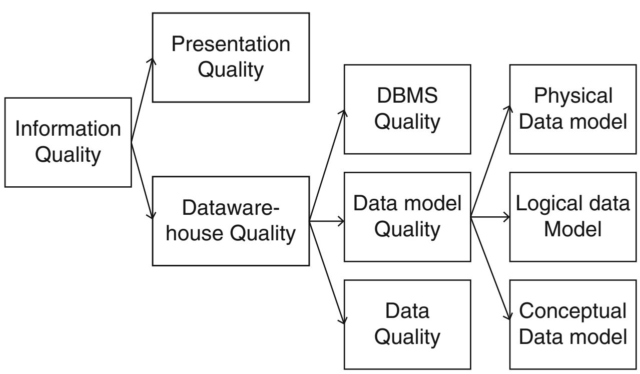

Fig. 1 Data warehouse quality

Data modelling is an art [1], even if the product of this activity has the prosaic name of the database scheme. The data model that allows the designer to devise schemes that are easy to understand and can be used to build a physical database with any actual software system, is called conceptual data model [5]. Conceptual data model represent concepts of the real world. It is widely recognized that there are at least two specific notations that any conceptual data model for data warehousing should include in some form: the fact (data cube) and dimension. A widespread notation used in implementation in this context is the "star schema" [6] in which fact and dimension are simply relational tables connected in specific way. One of the uses of multidimensional model is that they can be used for documentation purposes, as they are easily understood by non-specialists. They can also be used to describe in abstract terms the content of data warehousing application already in existence.

The dimensions are organized into hierarchy of levels, obtained by grouping elements of the dimension according to the analysis needs. A dimension has three main components: a set of levels, a set of level description and a hierarchy over the levels.

A Multidimensional data model dimension scheme D can be defined as [4].

- A finite set L of names called levels.
- A finite set Δ of names called level descriptions for each level in L; each description is associated with a base type t.
- A partial order $\leq$ called roll up relation on the levels in L, if L1 $\leq$ L2 we say L1 rolls up to L2.

In this representation, levels are depicted by means of round cornered boxes and there is a direct arc between the two levels. Small diamonds depict the descriptions of a level. Figure 2 describe some examples of dimension scheme.

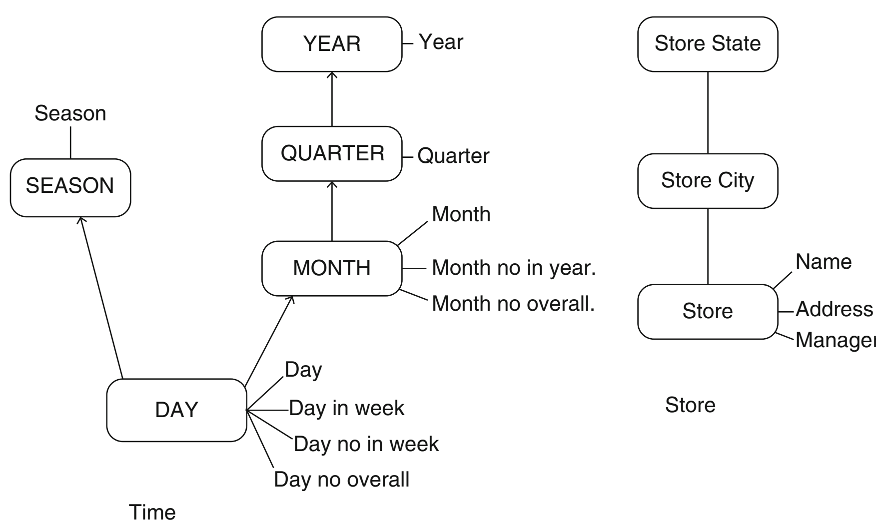

Fig. 2 Dimension scheme in the MD model [4]

A multidimensional data model is a direct reflection of the manner in which a business process is viewed [2]. The Dimensional Fact (DF) Model by Golfarelli [7], the multi-dimensional/ER model by Sapia et al. [8], the StarER Model by Tryfona [9], the Model proposed by Husemann [10] and the Yet Another Multidimensional Model (YAM) by Abello et al. [11] are examples of multidimensional models. According to Pedersen [12] and Blaschka et al. [13] multidimensional modelling is summarized as Explicit separation of structure and content, explicit notion of dimension and data cube, explicit hierarchies in dimension, multiple hierarchies in each dimension, level attributes, measure sets and symmetrical treatment of dimension and measures. In recent years, different authors have proposed some useful guidelines for designing multidimensional data models [2]. However more objective indicators are needed to help designers and managers to develop quality multidimensional data models [14].

Quality is a crucial issue during multidimensional design and there are several approaches focusing on the quality of the multidimensional models [15]. The approaches on the quality of the multidimensional models lack concrete and quantitative metrics for the measurement of multidimensional models quality [2].

Several data models have been proposed with the main goal of studying specific data warehousing application problems such as incomplete information, efficiency issues, heterogeneous dimension, dimension updates and temporal OLAP queries and so are well suited for them [2].

Metrics for Software System and Data Warehouses

The role of metrics in software quality is well recognized. However, they are yet to be standardized and integrated into development practices across software industry. While process, project and product metrics share common goal of contributing to software quality and reliability, utilization of metrics has been at minimum. The more they are utilized the more the effective and productive the organizations becomes. Software metrics are utilized during the entire software life cycle. Data is gathered, analyzed and evaluated by the project managers and software developers. According to Tom DeMarco [16] statement, “You can’t control what you can’t measure”. These disciplines apply measurements to gain better control of their projects and quality of products. Estimation models [17] help better plan and execute software projects. Mathematical models, such as Boehm’s COCOMO [18], Putnam’s SLIM [19], and Albrecht’s Function Points [20], can be used. Process, Project and Product are the three common categories for software metrics.

Process metrics focus on software development and maintenance. They are used to assess people productivity, also called private metric and the productivity of the organizations, also called as public metrics. Project metrics are related to project characteristic and execution. They also contribute to the development of process metrics. Common software project metrics [21] include order of growth, Lines of code, Cyclomatic complexity, Code coverage. Product metrics measure the key characteristic of the software product.

Commonly used product metrics [21] include Specification quality metrics, System size metrics, Architectural metrics, Length metrics, Complexity metrics, testing effectiveness metrics.

Software measurement is a very important part of organizations that wants to reach high level of maturity. This is proved with current progress maturity and improvement such as CMMI [22], IEC 9003 [23].

Metrics

A metric is a way to measure the quality factor in a constant and objective manner. They are used to understand software development and maintain projects. They are used to maintain quality of the system and determine the best way which helps the user in research work. It should be defined according to organization needs. The main goal is to assess and control the quality of the conceptual data warehouse schema.

This section describe the metrics proposed by Serrano et al. [2] for data warehouse multidimensional model. The proposed metrics are defined in Table 1.

Table 1 Metrics description

Metrics	Description
NDC(S)	Number of dimensional classes of the stars
NBC(S)	Number of the base classes of stars
NC(S)	Total number of classes of the stars NC(S) = NDC(S) + NBC(S) + 1
RBC(S)	Ratio of base classes. Number of base classes per dimensional class of stars
NAFC(S)	Number of FA attribute of the fact class of the stars
NADC(S)	Number of D and DA attribute of the dimensional classes of the stars
NABC(S)	Number of D and DA attribute of the base classes of stars
NA(S)	Total number of FA, D and DA attribute NA(S) = NAFC(S) + NADC(S) + NADC(S) + NABC(S)
NH(S)	Number of hierarchy relationship of the stars
DHP(S)	Maximum depth of the hierarchy relationship of the data stars
RSA(S)	Ratio of attributes of the stars. Number of attributes FA decided by the number of D and DA attributes

Empirical Validation

The suggestion provided by [24] were used on how to perform controlled experiments. The formats proposed in [24] have been used. In this section, we will present the current empirical validation for the metrics. We have used the GQM (Goal, Question, Metric) approach [25] in defining the goal of the experiment.

Definition

GQM is the acronym for goal, question, metric and is an approach to software metric [25] that has been promoted by Victor Basili of the University of Maryland, College Park and the Software engineering laboratory at the NASA Goddard Space Flight Carenter. GQM defines a measurement model on three levels: (i) Conceptual level(goal) which defines the goal for the object for a variety of the reason with respect to various model of quality from various point of view and relative to particular environment, (ii) Operational level(question) in which the set of question that are use to define models of study and then focuses on that object to characterize the assessment or achievement of a specific goal, (iii) Quantitative level(metric) defines a set of metric based on the models which is associated with every question in order to answer it in a measurable way. Six steps of Goal-oriented measurement process in are (i) characterize the environment, (ii) identify measurement goals and develop maintenance plans, (iii) define data collection procedure, (iv) collect, (v) analyze and interpret data, (vi) perform post-mortem analysis and interpret data and (vii) package experiences.

The experimental goal can be defined by using the GQM (Goal Question Metric) template [26]. The goal of the experiment is defined as follows:

Analyse the metrics for data warehouse conceptual models for the purpose of evaluating if they are useful with respect to the data warehouse understanding ability from the view of researcher in the context of students [2].

Planning

Context Selection

The context of the experiment is a group of twenty students of Master of Technology; Computer Science. The students are addressed as subjects. The experiment addresses a real problem by investigating the correlation between data warehouse structural complexity metrics and the understanding ability of data warehouses.

Table 2 Metrics value for all schema

	NDC	NBC	NC	RBC	NAFC
S01	3	5	9	1.67	2
S02	4	9	14	2.25	3
S03	3	4	8	1.33	3
S04	4	8	13	2	2
S05	3	8	12	2.67	3
S06	4	6	11	1.5	1
S07	3	5	9	1.67	5
S08	3	5	9	1.67	3
S09	3	6	10	2	3
S10	3	4	8	1.33	2
S11	3	6	10	2	3
S12	3	5	9	1.67	3

	NADC	NABC	NA	NH	DHP	RSA
S01	12	5	19	2	3	.11
S02	11	11	25	4	3	.13
S03	10	6	19	2	2	.18
S04	10	11	23	3	3	.09
S05	6	8	17	3	3	.21
S06	8	6	15	3	3	.07
S07	8	14	27	3	2	.22
S08	8	5	16	3	3	.23
S09	6	7	16	3	3	.23
S10	5	5	12	2	3	.2
S11	8	6	17	3	3	.21
S12	6	5	14	2	3	.27

Selection of Subjects

The subjects were students of Master of Technology, Computer Science. The subjects had no prior knowledge of data warehouse design.

Variables Selection

The independent variables are the variables for which the effects should be evaluated. These variables correspond to the structural complexity. Table 2 present the value for each metric in each data warehouse conceptual schema.

The dependent variable is the understand ability of the data warehouse schema. Table 3 present the understanding time of each subject. The understanding of the tests was measured as the time each subject used to perform the tasks of each experiment test.

Instrumentation

The objects used in the experiment were 12 data warehouse schemas. The independent variables were measured by metrics already defined in Section “Metrics”. The dependent variable were measured by asking each subject to answer some question related to schema and note then understanding time.

Hypothesis Formulation

We wish to test the following hypothesis.

Table 3 Data collected of understanding time (s)

	S01	S02	S03	S04	S05	S06
Sub1	170	300	343	234	107	62
Sub2	160	200	300	200	100	70
Sub3	200	218	260	160	230	180
Sub4	230	360	262	120	180	100
Sub5	300	120	180	240	120	180
Sub6	177	135	262	175	180	120
Sub7	180	180	120	210	172	111
Sub8	157	201	171	86	134	107
Sub9	300	240	120	135	172	98
Sub10	178	200	210	118	135	90
Sub11	179	160	140	130	115	108
Sub12	180	180	120	140	95	84
Sub13	229	162	114	109	110	114
Sub14	120	180	120	110	100	120
Sub15	131	119	129	64	102	68
Sub16	188	165	200	200	100	120
Sub17	120	180	300	300	180	180
Sub18	86	86	120	240	184	122
Sub19	120	185	125	185	242	185
Sub20	140	120	240	180	300	180
	S07	S08	S09	S10	S11	S12
Sub1	64	125	183	100	120	130
Sub2	89	100	108	107	124	120
Sub3	142	197	112	130	105	110
Sub4	191	100	120	120	100	100
Sub5	110	157	178	120	149	100
Sub6	180	240	180	178	120	166
Sub7	130	231	170	129	130	122
Sub8	–	112	78	114	89	120
Sub9	120	117	99	110	92	83
Sub10	91	125	93	100	108	83
Sub11	83	110	111	100	77	83
Sub12	119	147	120	100	105	111
Sub13	150	120	111	106	108	128
Sub14	100	90	120	60	120	90
Sub15	115	78	71	96	75	105
Sub16	110	108	128	124	100	105
Sub17	240	60	124	111	108	90
Sub18	64	65	100	120	130	120
Sub19	120	185	100	104	108	111
Sub20	300	180	107	111	105	104

Null Hypothesis, Ho

There is a no a statistically significant correlation between the structural complexity measures and the understanding time of the schemas.

Alternative Hypothesis, H1

There is statistically significant correlation between structural complexity measures and the understand ability of the schemas.

Experiment Design

We selected experiment i.e. all the task had to be solved by each of the subjects.

Operation

Preparation

Subjects were given an intensive explanation session before the experiment taken place. However, subjects were not aware of the aspects we intended to study. Neither were they informed of the actual hypothesis stated. We prepared the material we handed to the subjects, consisting of twelve data warehouse schemas. The diagrams were related to different universes of discourse and were general enough to be easily understood by each of the subjects. The structural complexity of each diagram was different. Table 2 describe the values of the metrics for each data warehouse schema. In Fig. 3 we can see the example of an Object oriented data warehouse conceptual model. In this example we analyse the procurement schema. The Fact contains the specific measures to be analysed, i.e. Quantity and dollar amount. The main dimension along with which we would like to analyse measures are:-Time, Vendor, Contract and Product. Finally Base classes are:-Week, Quarter, Year, City, State, Country, Category and Department.

Execution

Twelve conceptual data warehouse schemas were used for performing the experiment. The domains of all the schemas were different. Each design includes a data warehouse schema and a question/answer form. The question/answer included the task that had to be performed and space for the answers. For each design, subjects have to analyse schemas and answer some questions about the design. All the design were based on real world task. Before starting the experiment, the subjects were explained all the exercises they had to perform, the answers they had to provide and how they had to record time spent performing the tasks i.e. before studying each schemas they had to annotate the starting time (hour, minutes, seconds) and once they have answered they had to annotate final time (hour, minutes, seconds). Each subject had to work alone. In case of doubt, they could only consult the supervisor who organized the experiment.

Data Validation

We collected the entire test, controlling whether they were complete and correct in order to know if some of them must be discarded. All the tests were completed in all aspects and we were able to work with the twenty results for each schema.

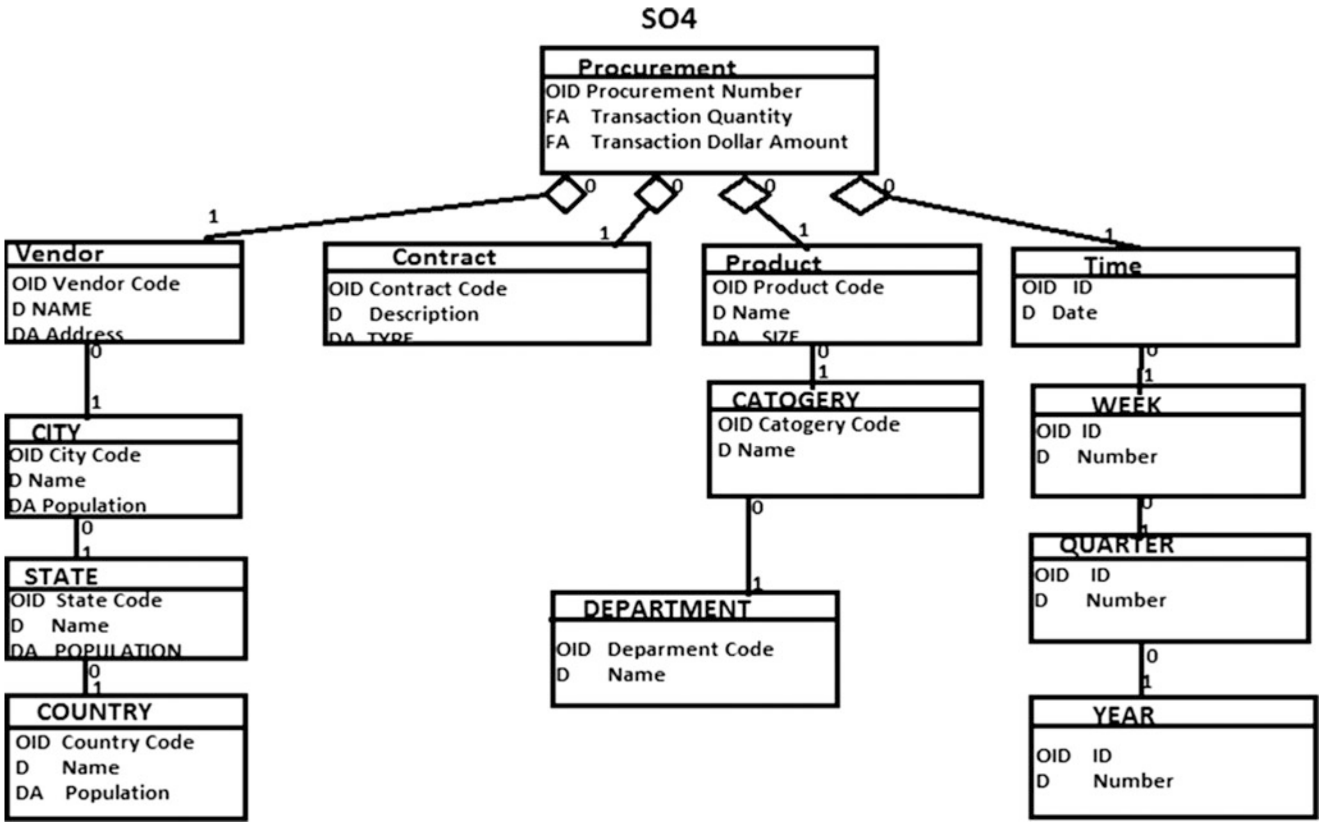

Fig. 3 Procurement schema

Box Plot of Understanding Time

Box plot of understanding time is used to measure the schemas and their corresponding understanding time. As with the help of the box plot we can easily find out the median of all the schema and then divide the schema in to two quarter Q1 and Q3. Q1 represent the median of the first halves above the median. Q3 represent the next half of the quarter below the median.

The minimum value is represented by the bottom whiskers and the maximum value is represented by the top whiskers. With the help of box plot we can analyze the understanding time for each schema individually. With the help of analysis it can be easily found out which schema took lot of time for getting answers and hence which schema was answered in less time. Figure 4 shows the box plot of the understanding time. Analyzing Fig. 4 we can conclude that S07 took very little time for understanding by one of the subjects and the maximum time was 300 s by Subject 20. For S01 and S02 minimum time are 86 s and the maximum time is 300 s and 360 s respectively. Hence box plot helps us to analyze the understanding time of the schemas.

Correlation between Metrics and Understanding Time

Table 4 shows the results obtained for the correlation between each of the metrics and the time used by each subject (on each schema) in performing the tasks. Analysing Table 4, we can conclude that there exists a correlation between understanding time used and the metrics NADC, NABC and DHP (the p-value is lower than or equal to the spearman coefficient 0.05).

Conclusion and Future Work

The quality of data warehouse is very important because when data warehouse is constructed properly, it provides organizations with a foundation that is reusable.

In this paper we have used already defined metrics for data warehouse in order to control their quality. We have concluded that there exists a strong correlation between the understanding time and some of the metrics. We can derive that metric NADC (Number of D and DA attribute of

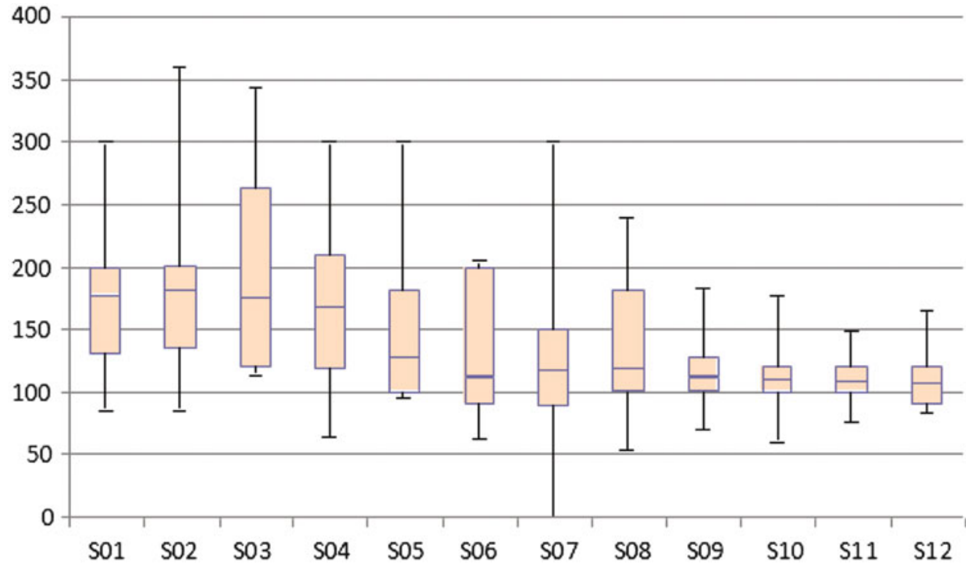

Fig. 4 Box plot of understanding time

Table 4 Correlation between metrics and understanding time

Metrics	NDC	NBC	NC	RBC	NAFC
Correlation	.305	.318	.336	.150	.060
p value	.336	.314	.286	.641	.854

Metrics	NADC	NABC	NA	NH	DHP	RSA
Correlation	.800	.243	.537	.090	.261	.506
p value	.002	.007	.072	.780	.003	.094

dimensional classes) is correlated with understand ability of data warehouse schemas.

These metrics will help in measuring the understand ability and the efficiency of the designers and users working in the schemas. Using these metrics, designer can choose among several data warehouse models. A set of experiments have been done in order to prove the validity of the proposed metrics. After performing the set of experiments we can conclude that several metrics are correlated with understand ability of the models and within the efficiency of the subjects.

The empirical work done is not enough and it is necessary to continue the experimental work in at least one of the following ways:

- To design more experiments with more cases and different value of metric.
- Use professional subjects.
- To make replication of the experiment already done.

The immediate future work will be applying these metrics in the real world projects.

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A Framework Based on Model Driven Engineering to Support Schema Merging in Database Systems

Marcus Vinícius Carvalho, Denivaldo Lopes, and Zair Abdelouahab

Abstract

Model Driven Engineering (MDE) aims to make face to the development, maintenance and evolution of complex software systems, focusing in models and model transformations. This approach can be employed in other domains such as database schema integration. In this paper, we propose a framework based on MDE to integrate database schema. In MDE context, database schema are viewed as database model. A metamodel for creating database models, an algorithm for database model matching and an algorithm for database model merging are presented. We provide a prototype that extends the MT4MDE and SAMT4MDE tools in order to demonstrate the implementation of our proposed framework. An illustrative example helps to understand our proposed framework.

Keywords

Model driven architecture • Model matching • Model merging • Database integration

Introduction

An organization has diverse sectors that generate a large information volume. However, sometimes this information volume is structured in some and different databases. In order to use this information in decision making, the managers need an unified view of the databases, i.e. an integrated database. Database integration is not an easy task, because systems are developed by different teams, having different views of the same domain.

To integrate different databases is needed find similarities between their database schemas, i.e. mappings between schemas [1]. Schema matching consists in finding mappings between elements of two or more *schemas*. *Schema matching* is a problem that is present in the database integration, ontology and XML schemas, and so on. *Schema matching* is being studied for a long time ago [1–3].

The manual task find schema mappings is tedious work consumes so much time and is prone to errors. So, domain specialist work can become easier with mechanisms that support the task to find mappings between schemas. In order to automatize or semi-automatize the schema matching process, researchers are looking for algorithms and tools that increase the quality of matched elements [3, 4]. In the literature, we find some research works that aim to support the evolution (i.e. merging, mapping or integration of database schemas) of database schema [5, 6]. However, the integration of database schema remains a problem without a complete solution.

Bringing to MDE context, schema matching can be called model matching, and schema mapping can be called model mapping, and schema merging can be called model merging. So, *Model Driven Engineering* (MDE) is a paradigm that can help to minimize the domain specialist work in the task of finding model matching (i.e. schema matching). MDE is an approach that drives the software development process in the model creation and transformation.

A *database schema* is a formal structure that represents an engineering artifact, such as SQL schema and XML schema [4]. So, a *schema* is a model, and being a model,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_49, © Springer International Publishing Switzerland 2015

MDE can contribute to minimize the task to find mappings among schema, in order to assure productivity, portability and interoperability [7].

This paper proposes a methodology to support the database integration process through an approach based on MDE and describes a framework that aims support the database model matching in a semi-automatic manner.

This paper is organized as follows. Section "Background" presents concepts around MDE, and schema matching and merging, and related work. Section "Framework to Support Database Schema Integration" presents a proposal to take care of schema matching based on MDE providing a framework and methodology. Section "An Algorithm for Database Schema Matching" presents an algorithm for database model matching. Section "An Algorithm for Schema Merging" presents an algorithm for database model merging. Section "An Illustrative Example" presents an illustrative example to demonstrate our framework and tool. Section "Conclusions and Future Directions" presents some considerations, conclusion and future directions about this research.

Background

This section present some technologies related to our research such as MDE, database schema matching and merging.

Model Driven Engineering

Models are the basis to the software development based on MDE. So, different models can represent a specific viewpoint of a system, and transformation definitions can pass information from a model to another model until to obtain the source code or script, i.e. the final artifact. *Model Driven Architecture* (MDA) [8] and *Eclipse Modeling Framework* (EMF) [9] are examples of MDE.

MDA is a framework to develop software systems based on standards from *Object Management Group* (OMG) [8]. In MDA context, a *Platform Independent Model*—PIM is an abstraction of a system without details about platform, and *Platform Specific Model*—PSM contains the business logic and platform details. A transformation definition takes as input a PIM and generates a PSM, another transformation definition takes as input this PSM and generates a source code or configuration file.

EMF is a framework that supports the software system modeling, assists the generation of source code from models and helps in the creation of tools focused on models [9].

Operators: Schema Matching and Merging

Schema matching consists in finding correspondences among elements of two or more schemas [1] and results in a mapping model between two or more schemas. According to [1], a *schema matcher* can be classified in *individual matcher* that use only one criteria for matching and *combining matcher* that combine multiples criteria to find *matchings*.

A mapping relates elements from S_1 schema and elements from S_2 schema. A mapping relates elements from two or more schemas that are equals or similar following a definition of equality or similarity [2].

In order to combine two schemas in a new schema, a merging process requires a mapping. *Schema merging* aims to create a unified schema from two other schemas [10].

Related Work

Tools as Rondo, Clio and COMA++ have been presented to assist the *match* process of database schemas.

Rondo [11] has operators that can be applied to XML schema, relational schema and SQL view. Operator *Match* proposed in Rondo uses an algorithm called Similarity Flooding (SF).

Clio [12] creates mappings between XML schema and -Object-Relational database generating queries in SQL. Clio provides a GUI that helps the domain specialist to define mappings to complete those found by the operator match.

COMA++ [13] creates mappings of SQL, *database schema*, W3C XSD and OWL. COMA++ uses a strategy based on *combining matcher*, implementing *matching* based on fragments and *matching* based on reuse. It provides a GUI where a specialist can view and configure strategies of *match*, and can correct mappings.

Our extension proposed to MT4MDE and SAMT4MDE and our solution to merge database models are a proposal to handle database schemas in the MDE context. The scripts obtained by the tools Rondo and Clio are queries applied to schemas, while the framework MT4MDE/SAMT4MDE create a script containing the structure of the integrated database, allowing queries to it. As the same way as COMA++ and Clio, our framework MT4MDE/SAMT4MDE allows the domain specialist to create mappings that were not found by the matching algorithm. It allows to define the elements that must be merged.

Framework to Support Database Schema Integration

The proposed framework aims to support the database schema integration following a MDE approach. A methodology is provided in order to help in the database schema integration. For this purpose, models, transformation definitions, and database schema mapping are present in our methodology. In the MDE technological space, a schema is viewed as a model, a schema matching is viewed as a model matching, a schema merging is viewed as a model merging, and database schema integration is viewed as database model integration.

Figure 1 presents the proposed framework to integrate database schemas, i.e. database models. This *framework* is an extension from the research work presented in [14] that aimed to use metamodel matching to generate mapping models and transformation definitions in ATL [15]. In this paper, we provided extensions in the MT4MDE and SAMT4MDE in order to support database model matching, database model mapping and database model merging. The extensions are the following:

- A new metamodel to support *model matching* is proposed. In [14], an approach was proposed to provide metamodel matching in order to generate transformation definition, and in this paper we provide an approach to database model matching. The module *SchemaMatchDB* implements the interface *ITFMatch* (see Fig. 1), and reuse the metamodel *schema matching* in order to create a model that registers the corresponding elements among two database models.
- We propose a metamodel for *database model merging* included in the *Merge* module. Figure 1 shows that *Merge* module implements the interface *ITFMerge*. *Merge* creates a mapping model whose elements are superposed to generate an integrated database model.
- *Integrated Model Generator* generates a database model that contains all information needed to build a SQL script.
- The concrete class that implements *ITFGenLanguage* is conceived for generating a SQL script instead of generating transformation definition in ATL.

So, thanks to these extensions, the proposed *framerwork* MT4MDE/SAMT4MDE is enabled to execute the *merging operator* having as input two or more database model.

A Methodology to Integrate Database Models

Our methodology aims to guide the steps to obtain database model integration in the MDE context. This methodology is described as follows.

1. ***Import database schema***: consists in import the database schema and generate an equivalent model conforms to a database metamodel. If the database schemas exist, then the domain specialist can import them. This can be done reading a database schema or reading a SQL script with the structure of the database or reading XML schema. The domain specialist also can to complete the model

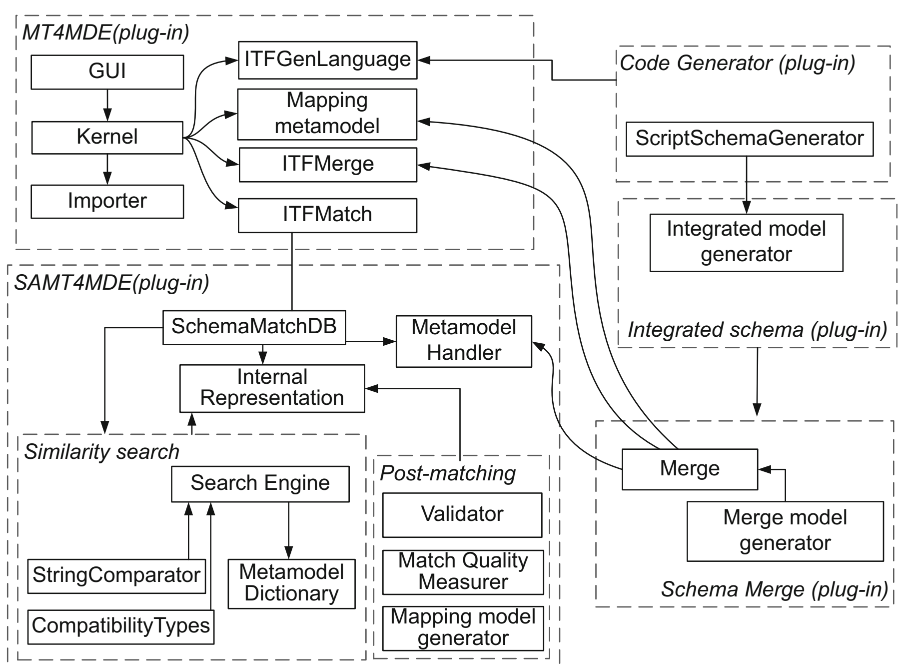

Fig. 1 A framework to integrate database schemas

with information that was not obtained in the imported database schema. All imported models are saved in the database model repository.

2. ***Create or recover database schema*** (**model**): consists in creating a model of a database or recover a model of a database from a repository. In the recovery database schema model the domain specialist can choose database models to be integrated. While in the creation database schema model the domain specialist create a model of database conform to metamodel database defining all entities and their attributes, primary key and foreign key. A database model created by the domain specialist are saved in a repository for future recovery.
3. ***Generating a mapping model from database models***: in this step, once we have database models, and an algorithm of model matching is applied to find corresponding elements between database models. This algorithm results in a database mapping model that relates the equal or similar elements of these two database models. This database mapping model is required in the step of database model merging.
4. ***Generating a merging model from database models***: in this step, the database models (those imported in the step of mapping model generation) and taken as input by an algorithm of model merging. So, this algorithm takes two database models (M_1 and M_2) and a mapping model (C), and determines the differences between M_1 and C and the differences between M_2 and C. Afterwards, this algorithm gives as output a merging model *Me* relating M_1 and M_2.
5. ***Generating an integrated database model***: in this step, a merging model is used to generate an integrated database model. For this purpose, transformation definitions written in *ATLAS Transformation Language* (ATL) take as input a merging model and provides a database model containing the merged information from the database models M_1 and M_2.
6. ***Generating a SQL script from an integrated database model***: in this step, an integrated database model is transformed in a SQL script thanks to transformation definitions written in ATL.

Metamodels for Defining Database Model and Mapping Model

In order to create an integrated database model and a database mapping model, we propose the following metamodels. In MDE context, a database schema is imported as a database model in order to be manipulated in the MDE technological space.

A Metamodel for Database. Figure 2 presents a metamodel for database that is used to instantiate database models. So, a domain specialist or MT4MDE can create or import a database model from a database schema. So, the domain specialist or MT4MDE can instantiate *Entity*, *Attribute* and *Relationship* elements. As seen in Section "A Methodology to Integrate Database Models", models conform to database metamodel can be created or imported.

A Metamodel to Create Mapping Model. Figure 3 presents a metamodel to create mapping model in order to store correspondences between elements of two database

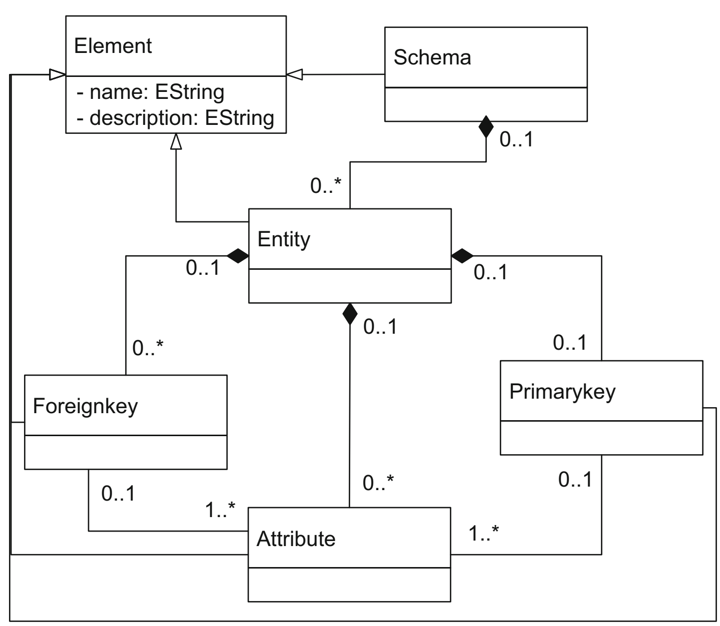

Fig. 2 A metamodel for database (fragment)

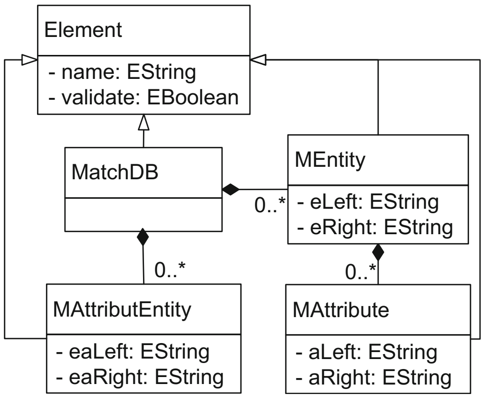

Fig. 3 A metamodel to create mapping

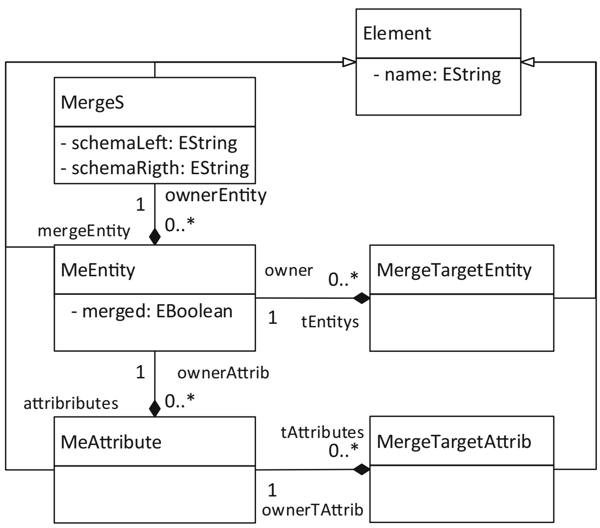

Fig. 4 A metamodel to create merging model (fragment)

models. This metamodel is based on [16]. Given as input the database model S_1 and the database model S_2, this mapping metamodel allows to instantiate a model that registers corresponding elements from S_1 and S_2, determining if they are equals or similar. As presented before, the first step to integrate database models is find the corresponding elements between database models.

A Metamodel to Create Merging Model. Figure 4 presents a metamodel to create merging model and it aims to register the elements from database models that are superposed in an integrated database model. Moreover, it registers the database elements that are not contained in the mapping model. Therefore, this metamodel defines what database elements will constitute the integrated database model.

An Algorithm for Database Schema Matching

In MDE context, schemas are viewed as models. So, a database schema is imported in MDE technological space as a database model. An algorithm for database model matching takes as input database models and gives as output a mapping model that links the equal or similar elements from database models.

Our framework has a hybrid matching algorithm for database models. Our proposed hybrid matching algorithm implements the operator *Match* that has as input two database models and gives as output a mapping model.

The operator *Match* is defined as follows: $Match(M_1(S_1)/M_s, M_2(S_2)/M_s) = C_{M_1 \to M_2}/M_c$, where given a database model M_1 that represents a system S_1 and is conform to database metamodel M_2, and a database model M_2 that represents a system S_2 and is conform to a database metamodel M_s. The operator *Match* generates a mapping model $C_{M_1 \to M_2}/M_c$, that is conform to mapping metamodel M_c. $C_{M_1 \to M_2}/M_c$ corresponds to *MatchDB* presented in Fig. 3.

We can define $C_{M_1 \to M_2}/M_c$ as $C_{M_1 \to M_2}/M_c \supseteq \{M_1 \cap M_2\}$, where $\cap$ returns elements of M_1 and M_2 that are equal or similar. Similar elements are elements where there is a relationship between them, but this relationship cannot be well defined [2].

We can define the sets M_1, M_2 and $C_{M_1 \to M_2}/M_c$ as:

- $M_1 = \left\{e_{1_i}, f_{1_i} \middle| 0 < i \leq \Lambda 0 < j \leq m\right\}$, where e_{1_i} are entities, f_{1_j} are attributes and n and m are the amount of entities and attributes from M_1 respectively.
- $M_2 = \left\{e_{2_x}, f_{2_y} \middle| 0 < x \leq n' \Lambda 0 < y \leq m'\right\}$, where e_{2_x} are entities, f_{2_y} are attributes and n' and m' are the amount of entities and attributes from M_2 respectively.
- $C_{M_1 \to M_2} = \{c_1, c_2, \ldots, c_p\}$ constitutes a set of correspondent elements of M_1 and M_2, where $c_i = \{e_{1_i}, e_{2_x}\} \vee \left\{f_{1_j}, f_{2_y}\right\} \vee \left\{f_{1_j}, e_{2_x}\right\}$. Thus, c_i may be a *match* of entities to entities, or attributes to attributes, or attributes to entities.

Once we have the defined sets M_1, M_2 and $C_{M_1 \to M_2}/M_c$, we present the proposed algorithm to match database models as follows:

1. Create *MatchDB*: $MatchDB = \emptyset$.
2. Select all entities e_{1_i} from M_1 and e_{2_x} from M_2.
3. For each pair $\{e_{1_i}, e_{2_x}\}$ apply the function *simString* (e_{1_i}, e_{2_x}, *<method>*) that returns the information about equal or similar entities:
   (a) If $e_{1_i} = e_{2_x}$, then,
      (i) For each pair $\{f_{1_i}, f_{2_x}\}$

A. Verify if attributes are similar (apply the function $simString(e_{1_i}, e_{2_x}, <method>)$ calculating percentage of equal or similar attributes.

(ii) Case the percentage be big than a limit value, then include this entity in *MatchDB*.

(b) Else, if $e_{1_i} \neq e_{2_x}$, verify if entities are equal or similar according to attributes, calculating percentage of equal or similar attributes as in item (a.i.A) and include them in *MatchDB* when be big than a limit value.

(c) Entities that are not equal or similar do not must be included in *MatchDB*.

4. **Compare attributes and entities**. For each pair $\{f_{1_i}, e_{2_x}\}$ apply the function *simAttributeToEntity* $\{f_{1_i}, e_{2_x}\}$ that compares if an attribute from e_{2_i} is represented by an entity e_{2_x}. Returning true, include f_{1_i} and e_{2_x} in *MatchDB*.

The function $simString(e_{1_i}, e_{2_x}, <method>)$ returns the similarity degree between two strings. The returned value can vary in the range [0..1], where the returned value 1 means that the two strings are equal. A limit value must be defined in order to consider strings as similar. After test limit value, the value 0.7 was the most positive correspondence returned, being used in *simString*. The parameter $<method>$ determines the metric of similarity between strings. We use the Jaro-Winkle, Levenshtein and N-Gram solutions as distance metric in our matching algorithm. Thus, a domain specialist can test which is best metric database entry.

The function *simString* use also a synonymous dictionary to determine similarity between *strings*. Our framework provides a *metamodel for dictionary* that allows to create models containing lists of domain synonyms. Models conform to a *metamodel for dictionary* are used by the function *simString* to determine the corresponding elements from database models. The function *simAttributeToEntity* $\{f_{1_i}, e_{2_x}\}$ implements a search of attributes f_{1_i} from M_1 that are similar to entities e_{2_x} from M_2.

An Algorithm for Schema Merging

In MDE context, a schema is viewed as a model. So, a database schema is imported into MDE technological space as a database model, and a merging algorithm can merge two or more database models.

An operator *merge* combines two database models in one database model (in database domain, combines two database schemas in one database schema [11]). Therefore, we propose the operator *Merge* that has as input two database models and a mapping model and provides as output a *model merging* (see Fig. 4). Our operator *Merge* was created observing the requirements of preserving elements proposed in [16]: equality preservation, relationship preservation and similarity preservation.

The operator *Merge* takes as input a triple $(M_1, M_2, C_{M_1 \to M_2}/M_c)$. Thus, *Merge* $(M_1, M_2, C_{M_1 \to M_2}/M_c) = Me[(M_1 - C_{M_1 \to M_2}) \cup (M_2 - C_{M_1 \to M_2}) \cup C_{M_1 \to M_2}]$, where:

- *Me* is s*chema merge model* between M_1, M_2 and $C_{M_1 \to M_2}$ conform to *schema merging metamodel*.
- $M_1 - C_{M_1 \to M_2} = \{m_{1_i} | m_{1_i} M_1 \Lambda m_{1_i} \notin C_{M_1 \to M_2}\}$.
- $M_2 - C_{M_1 \to M_2} = \{m_{2_j} | m_{2_j} M_2 \Lambda m_{2_j} \notin C_{M_1 \to M_2}\}$.

The degree of correspondence between entities $\{e_{1_i}, e_{2_x}\}$ models M_1 and M_2 is considered in the task of merging. Therefore, case entities present a ***no match***, i.e. entities from model M_1 do not have any entity correspondent in model M_2, an entity will be create in the *merging model*. The same is applied in the inverse, in the case where entities $\{e_{1_i}, e_{2_x}\}$ present a ***total match***, a unique entity must be created in the *merging model*. In the case entities $\{e_{1_i}, e_{2_x}\}$ present a ***partial match***, we can have:

- **Scenario 1 has *high match***: in this case, there is a high degree of correspondence where few attributes of entities are not corresponding. Entities are merged and an attribute identifier is used to remember the origins of the instances (tuples).
- **Scenario 2 has low *match***: in this case the number of corresponding attributes is too low and a merging element is not created. So, distinct entities must be created in the *merging* model.
- ***Scenario 3 has average match***: the number of corresponding elements between the entities is not classified in the previous scenarios. Therefore, the proposed solution is *generalization of correspondence*} where matched attributes from $\{e_{1_i}, e_{2_x}\}$ are generalized in a new entity called *generalized entity*, and for the attributes no correspondents are created new entities containing the same primary key of the *generalized entity*.

The conflicts of attributes types are handled as follows: between a character and a string, prevails the type string, between integer and float, prevails the type float. All relationship of entity source and target must be preserved when is applied a *generalized correspondence*.

In this way, an algorithm for *database model merging* follows these steps:

1. **Create *MergeS***: $MergeS = \emptyset$.
2. **Select entities *no match***: create a set E: $E = \{e_{1_i}, e_{2_x} / e_{1_i} \notin C_{M_1 \to M_2} \Lambda e_{2_x} \notin C_{M_1 \to M_2}\}$;
3. **For each** $\{e_{1_i}, e_{2_x}\}$ ε E:

   (a) Include new entities *MeEntity* for $\{e_{1_i}, e_{2_x}\}$ in *MergeS* containing their attributes.
4. **Select entities classified as *total match***: create a set T containing entities e_{1_i} that are present in the mapping $C_{M_1 \to M_2}$. The *merging model* is based on the elements contained in the database model M_1;
5. **For each** $e_{1_i} \varepsilon T$:
   (a) Include a new entity e_{1_i} *MeEntity* in the *MergeS* containing their attributes.
   (b) Create one or more classes *MergeTargetEntity*. The class *MergeTargetEntity* maps entities e_{2_x} that correspond to e_{1_i}.
   (c) Create one or more classes *MergeTargetAttrib*. The class *MergeTargetAttrib* maps *MeAttribute* f_{2_y}/e_{2_x} that corresponds to f_{1_j}/e_{1_i}.
6. **Select entities classified as *partial match***: create a set Y as entities e_{1_i} that are present in the mapping $C_{M_1 \to M_2}$, but do not have all their attributes corresponding**.**
7. **For each** $e_{1_i} \varepsilon Y$**:**
   (a) If the degree of similarity for *low match*, then
      (i) Include a new entity *MeEntity* in *MergeS* to e_{1_i} and to e_{2_x} containing their attributes identical to original attributes.
   (b) Case the similarity degree is *high match*, then
      (i) Include a new entity *MeEntity* inside *MergeS* containing their attributes identical to original attributes. Include also a *MeAttribute* containing *name* ="*typeEntity*" to identify the origin of tuples.
      (ii) Execute the same actions of steps 5. (a), 5.(b), 5.(c).
   (c) Case the similarity degree is *average match*, then (apply *generalization of correspondence*).
      (i) Create a new entity *MeEntity* containing the attributes *MeAttribute* f_{1_j}/e_{1_i} that correspond to attributes f_{2_y}/e_{2_x}. Include also a *MeAttribute* containing *name*="*typeEntity*" to identify the original tuples;
      (ii) Include new entities *MeEntity* inside *MergeS* to e_{1_i} and to e_{2_x} containing only the attributes f_{1_j}/e_{1_i} and f_{2_y}/e_{2_x} that are not correspondent. *MeEntity* that will represent e_{1_i} and e_{2_x} must have the same primary key of entity *MeEntity* created in the previous step (i);
      (iii) Execute the same actions of steps 5. (a), 5.(b), 5.(c).

The *merging model* maintains only references to model elements of M_1 and M_2. In the moment of the creation of integrated model of M_1 and M_2, all entities are created containing their attributes, primary keys and foreign keys.

An Illustrative Example

We choose the software systems developed to Unique Health System (*Sistema Único de Saúde*—SUS) as a business domain to illustrate our proposal to integrate database schemas. SUS provides to Brazilian population a free health service [17]. SUS provides diverse software systems to manage all information around the health system of Brazil. These systems store a large volume of information, but they are not integrated. As consequence, the decision making is difficult because the information is stored in different systems and different database schemas.

The chosen systems to test our *framework* were the *SISMAMA* and the *SISCOLO*, and they were created to store data about breast cancer and uterus cancer, respectively. The system *SISMAMA* has 17 tables containing 741 columns. The system *SISCOLO* has 17 tables containing 481 columns.

First, we use the import module of our framework to import the database schemas as the source and target *database model*. The *database model* was constructed for recovering the information about metadata of database systems. Once the *database models* were imported and the steps 3 and 4 of our methodology presented in Section "A Methodology to Integrate Database Models" were executed, we obtain as a result the mapping model of the source and target database model. Figure 5 presents our *framework* and the *mapping model* in the center, on the left *database model* of *SISMAMA* and on the right the *database model* of *SISCOLO*.

Once we have the *mapping model* relating the *database models*, we can apply the merging algorithm to create the *merging model* (see Fig. 4) containing the elements that will be *merged*. Figure 6 present an entity *MeEntity municip* from *SISMAMA* that was superimposed to *MergeTargetEntity municip* from *SISCOLO*. Once the *merging model* is defined, the steps 5 and 6 of the methodology presented in Section "A Methodology to Integrate Database Models" are applied to generate the *integrated database model* conform to our *database metamodel* and generate the SQL script to create the integrated database.

Conclusions and Future Directions

MDE contributes to minimize the efforts to match database models, and the domain specialist can work and visualize the solution of a problem in a high level of abstraction. A database model is submitted to transformations until to generate the schemas in a language like as SQL.

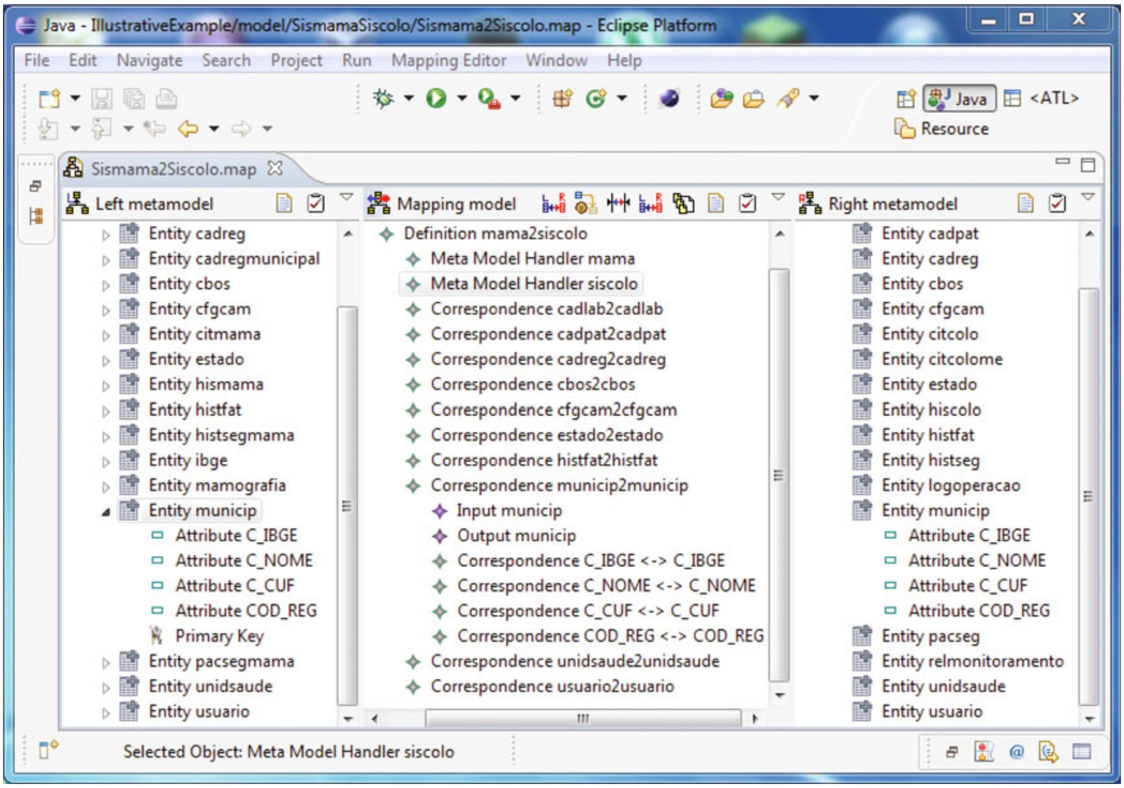

Fig. 5 Framework MT4MDE/SAM4MDE presenting the matching model between two database

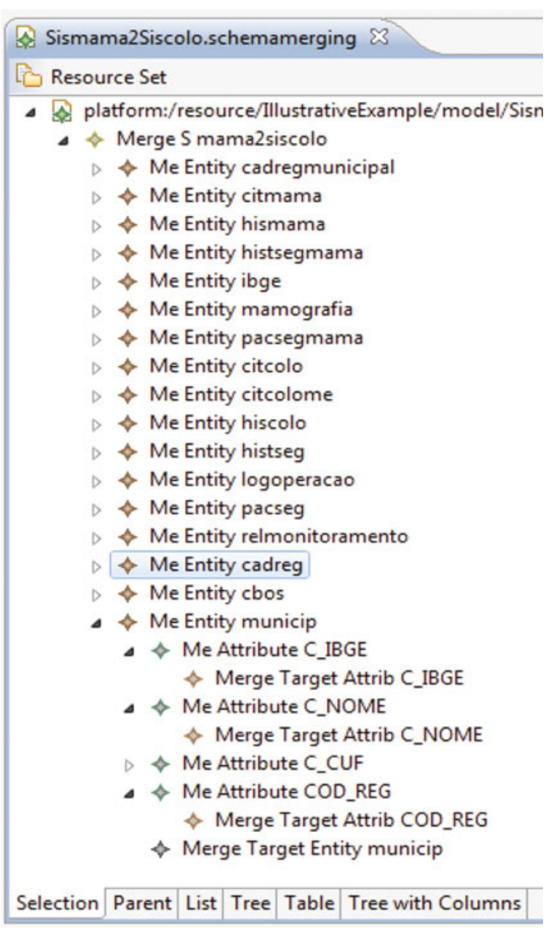

Fig. 6 Merging model of SISMAMA and SISCOLO (fragment)

The main contributions of this research work were: our proposed algorithm to match and merge database models; metamodel to define *database models*; metamodel to define match models; extension of tools MT4MDE and SAMT4MDE in order to work in the model level to be applied in database domain. Furthermore, the framework helps the domain expert from creating model database to generate a script integrated database, thus reducing the effort to integrate the database.

In next research work, we aim to implements other match algorithms to our *framework* like composite match in order to obtain a better result and consequently an integrated database model.

Acknowledgment The work described in this paper is supported by CAPES, FAPEMA (through the grant APP-00944/09) and CNPq (through the grant 560231/2010-5).

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Work in Progress: Model Design to Measure the Efficacy of Students Learning Preferences—Does Media Matter?

Ashfaque Ahmed Chowdhurya, Aruna Jayasuriya, and Elizabeth A. Eschenbach

Abstract

With the advance of online Learning Management Systems (LMS) it is easier than ever to provide students content providing multiple types of media. Many contend that this generation of students prefers video to text. This paper proposes a study that will examine if first year engineering students learn more about given case studies using the recommended textbook or the video prepared by the text publisher. Students are required to learn about the wonders of the industrial world in their first year engineering course. Students are tasked to learn about the wonders from the recommended text. Sometimes students choose to use to learn about the wonders from the video that is being available from the publisher of the book. In this study, it is planned to assess how much the students learn depending on their choice of media as well as study style and time on task. The students' performance in the reflective assessment item will be combined with the survey results to determine if there are any significant effects on student learning. The results of this study are not yet known. It could be that students learn better with the video vs. the text or vice-versa. It may be that the most influential aspect is how long the student spent studying or if the student reviewed notes before taking the assessment. Regardless of the specific outcome of the results, the overall conclusions will be useful in helping future first year engineering course students be successful in this particular assignment and in the course over all. This Work In Progress paper is presented with the intention of receiving feedback on the study and survey design.

Keywords

Using video in engineering education • Video aided learning • Text aided learning

Introduction

Many contend that the current generation of University students has a new set of preferred ways of learning, that the generation of Millennials prefers to learn with technology [1, 2]. However, we do not yet have a good understanding if introductory engineering students share the same preferences and beliefs about learning with technology. In addition, we do not yet have a good understanding if students are able to effectively use technology to meet stated learning outcomes. In the specific exercise, students will be given cases to study from engineering achievements that ensued during the Industrial Revolution. The cases are available

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_50, © Springer International Publishing Switzerland 2015

both in text and video format. Students have the freedom to select the preferred format to study the cases.

It is understood that if the students learn the case studies well, they should be able to exhibit their learning about engineering and the profession in general and be able to:

- Demonstrate evidence of developing professional capacity to communicate, work, and learn.
- Articulate an appreciation of the complex nature of engineering activities including ill-defined situations and problems involving uncertainty, imprecise information, and conflicting technical and non-technical factors.
- Describe, apply and justify risk assessment and workplace health and safety in engineering activities.
- Discuss the socio-technical role of the professional engineer.
- Articulate an appreciation of the uncertain nature of engineering design [3].

This paper presents the design of an engineering education experiment that will explore the beliefs, preferences, actions and efficacy of those actions, when introduction to engineering students have a choice of technology to use to reach stated learning outcomes.

Background

Learning Preferences

Researchers have attempted to characterize engineering students' learning preferences. Carver [4] attempted to enhancing student learning for different learning styles through hypermedia courseware. The author tried two approaches; students were provided a number of tools to prepare for lessons in one approach. The author found that the value of a particular multimedia tool to a student varied widely based on an assessment of the multimedia and hypertext documents. An adaptive hypermedia interface was suggested which, in authors view, is the most effective for students learning.

Mohler [5] reported that educators are examining different ways to implement publisher generated materials or custom, self-developed digital utilities into their curriculum and suggested educators to continue to integrate digital tools into their classrooms which provide active learning opportunities to the students. The author also suggested educators to undertake research to determine the effective level of interactive multimedia for students to understand the engineering concepts and identify the technologies that may have most successful impact on student learning.

Ambikairaja et al. [6] conducted a survey over a three year period to better understand the learning approaches and behaviours of year 3, 4 and 5 engineering students. They found that students over the 3 years were less interested in attending lecture 100 % of the time (48–30 %) while they became more interested in attending tutorial sessions 75 % or greater (75–85 %). They also observed students preferences were moving toward more learning via self-study (30–48 %). A small proportion used the textbook as a study aid (10 %). Most students relied on past exams (35 %), lecture notes (30 %) and worked examples (25 %).

Beliefs

An interesting finding from Ambikairaja et al. [6] was that, in the first year of the survey, 70 % students believed they learn the more from lectures than working by themselves, but this value fell to 53 % by the third year of the survey. Student critiques of lectures included: hard to understand, difficult teaching techniques, or boring presentations. The students tended to spend less than the recommended 20 h a week on study outside of class, with 30 % of the students stating they study less than 5 h a week. The authors noted a correlation between non-attendance and students with part-time work.

Learning Preference and Efficacy

Belski [7] provided third year electronic engineering students a choice between static (text and pictures) and dynamic recordings (video and audio) of problem solutions. Students statistically significantly preferred the dynamic worked examples over the static examples and they also performed statistically significantly better on the final exam. Only three of the 30 students preferred the static questions. Wandel [8, 9] also found that students preferred dynamic worked examples. These examples of using videos to present solved problems demonstrated the concept of reducing cognitive load by having students review solved problems first before trying to solve problems on their own [10].

Calm et al. [11] reported the effectiveness of video in on-line maths courses where they provided video tutorials along with other educational resources. The results were presented in terms of student satisfaction and their academic outcome. The authors reported that students satisfaction in the course increased by 20 % in the first semester of 2011–2012 when they introduced videos resources compared to 2010–2011 when the same resources were not available to the students. Moreover, the authors reported sustained upward trend of the percentage of the students finally passed the course.

How Students Use Technology

Jonassen and Reeves [12] and Burns and Ungerleider [13] reported that the effects of technologies on learning are variable and inconclusive.

Taylor [14] performed an experimental study on the use of captioned video with the beginning students of Spanish and reported that the first year students found the captioned distracting and had difficulties to attend the information with captioned video compared to non-captioned video. However, 3rd and 4th year students scored better than the first year students in the captioned video.

Lowerison et al. [15] suggested that there is need to perform controlled studies of technology integration and the methods of technology integration to realise its impact on students perceptions of learning.

In Belski's [7] work, with third year electronic engineering students Belski found students used the dynamic solutions right before summative assessment opportunities.

Purpose of the Study

With the advance of online Learning Management Systems (LMS), it is easier than ever to provide students content using multiple types of media. Currently, CQUniversity engineering programs use a mix of video and text based learning material. Types of learning material used in a typical ABC University engineering course include:

- Study Notes, lecture slides (text).
- Full 1–2 h lectures (video).
- Short videos addressing a specific course content (video).
- Step by step problem solving (videos and text).
- Textbook (hard copy).
- Additional online material (text/video).

With the availability of wide variety of material, students may feel overwhelmed by the amount of information they have to go through during a course. Also the educators need to spend a considerable time preparing different types of material using different technologies for the course. Furthermore, some content may be made redundant by other content. For example it may not be necessary to view the tutorial answers after watching problem solving videos. Hence it is important to understand what learning material best serves the student's learning requirements.

We intend to address four major research questions in this study.

1. Preference: In what format do first year engineering students prefer their learning content (text or video or a combination)?
   (a) If a student prefers video, what type of video does the student prefer (short, long, problem solving)?
2. Beliefs: Do first year engineering students believe they learn better with video or text?
3. Learning Preference and Efficacy: Do first year students actually learn better with video or text? Is there a certain type of video they prefer (e.g. long, short, problem solving)? Is their preference statistically correlated with their performance? Do students learn better when they spend more time on task—regardless of media choice? Do students learn better when they take notes or when they review their notes?
4. How Do Students Use Technology Do students use more video than text (as measured in number of downloads)? How do students use videos—do they review them before an assessment or do they rewind and watch particular parts?

Answers to these questions will allow the academics to more effectively and efficiently prepare course material. Furthermore the teaching staff would be able to provide student guidelines as to how to best use different course material (e.g. "Students in the past have learned the most when they watch the video first, take notes, and then attempt a set of questions).

Study Method

The data collection for the study will occur in two subsequent offerings of the Engineering Skills course, a common first year course for all CQUniversity engineering students. In the course, students are required to study case studies on the seven wonders of the industrial world. Both video and text material are available that provide the necessary learning content.

Engineering Skills is part of CQUniversity [3] two unique degree pathways in engineering – one with a Co-op experience (the dual award program Bachelor of Engineering (Co-operative Education)/Diploma of Professional Practice (Engineering) [16] and one with Distance Education option, but no Co-op option (Bachelor of Engineering). Both of the degree options integrate Project Based Learning (PBL) in all years of the degree program [17]. Approximately 32 % of the CQUniversity enrolled engineering students take their courses as part of the distance education program.

The engineering skills course, which is offered in the first year of the CQUniversity engineering degree program, has been developed to satisfy the pedagogical requirements of the Engineers Australia and the University's graduate attributes. The course requires students to develop and/or enhance technical and professional knowledge, skills and attributes required for team work, self and team management along with other core learning outcome of the courses. The course offers a number of learning approaches including text and video based courseware, face to face lecture and/or via multimedia streaming technologies, tele-tutorials, online video conferencing, residential school etc. Students are to participate in a number of learning activities and reflections including reflective journals, team based and individual activities, case studies, laboratory experiments, instructional design, and hands on activities [18]. The courses are offered across four regional campuses and by distance mode with academic support in each of the campuses to ensure equitable learning opportunities for students.

In the proposed experiment, students will be able to choose their preferred mode to learn about the six case studies. Students will either use the video prepared by the publisher, or the recommended textbook or both. About 210 students take this course each year across the 4 campuses and in the distance mode. Hence over 400 students are expected to participate in the study over the 2 years.

Before students are given the case study assignment, students will be given a survey that will query them about their learning preferences. Table 1 provides a sample of the types of questions that will be part of the preference instrument.

Students will then be given one of the case study assignments where they can choose to use the course text, the video provided with the course text or both to learn the required content. Students also have the freedom to source additional resources if they prefer to do so.

The students will then be assessed on how much they learned. In the specific exercise, students will be given a case to study from engineering achievements that ensued during the Industrial Revolution. Students are required to prepare a reflective paper highlighting their capacity and propensity for teamwork, individual work, communication, use of information well (research, investigate etc.), socio-technical role of engineers, account for the uncertain nature of design, use of engineering language effectively etc. Students will also conduct a self-assessment on their learning from the learning resource. Students will also be given a post-survey where they will report on how they used the video or text and increase/decrease self-reported performance and overall performance. Table 2 and 3 below provides an outline of the proposed survey instrument.

Lastly, students usage data will be collected from the learning management system. Student uses of video and text content will be measured against their preferences and intentions.

Table 1 Anticipated number of students in each cohort

Campus	Number of expected participants
On-campus Rockhampton	80
On-campus Gladstone	25
On-campus Mackay	30
On-campus Bundaberg	25
Distance	50

Data Analysis

1. Preferences
   The data from the pre-assignment survey will be summarized to determine student preferences. These data will be separated into categories of face to face students, regional campus students and distance students.
2. Beliefs
   The data from the pre-assignment survey will be summarized to determine student beliefs about their ability to learn with different types of media. In addition, statistical analyses will be completed to determine if there is any correlation between the student beliefs about their ability to learn with different media types and the student's score on the assessment item.
3. Learning Preference and Efficacy
   Statistical analyses will be completed to determine if there is any correlation between the student preference for video or text and the students score on the assessment item.
4. How Students Use Technology
   The data from the post-assignment survey will be summarized to determine student reported use of technology. The data from the learning management system will be summarized and analyzed to determine if there are any reportable trends. Lastly, statistical analyses will be completed to determine if there is any correlation between the student self-reported use of the media and the measured student use of the media. Also, statistical analyses will be performed to determine if there is a correlation between student performance and measured use of the media on the learning management system.

Anticipated Outcomes

The results of this study are not yet known. It could be that students learn better with the video vs. the text or vice-versa. However, it may be that the most influential aspect is how long the student spent studying or if the student reviewed notes before taking the assessment. Regardless of the specific outcome of the results, the overall conclusions will be useful in helping future first year engineering course students be successful in this particular assignment and in the course over all. It will also assist to understand how the type of media selected by students assist them to shape perceived their ideas of engineering and arouse their desire to learn the basic skills required in athe professional life.

Table 2 Draft set of questions for pre-assignment preferences survey

• From the available resources (textbook or textbook provided video), which type of study resource do you prefer to use for this case study?
• Do you intend to use other type of resource later on? (for example extra web based resources, textbook if you decided to use video initially)
• What factors influenced your choice of resources? Check all that apply (estimated time to view/read the resource, personal preference, ease of sourcing the material, ease of use, content covered by the resource, efficacy of the resource)
• How confident are you (from 1 to 5 in Likert scale, where 1 is not confident at all and 5 for extremely confident) that your choice of resource is the best type of resource for your study?
• How confident are you (from 1 to 5 in Likert scale, where 1 is not confident at all and 5 for extremely confident) that you will not need any other type of resources to complete the study?

Table 3 Draft set of questions for study survey

• Which media did you choose for the reflective assessment? (video, text, both)
• What factors influenced your choice of resources? Check all that apply (estimated time to view/read the resource, personal preference, ease of sourcing the material, ease of use, content covered by the resource, efficacy of the resource)
• Did you read the text or watch the video more than once?
• Did you take notes while reading the text or watching the video?
• How long did you study for the reflective assessment?
• If you took notes, did you review your notes before the reflective assessment?
• How satisfied are you (from 1 to 5 in Likert scale, 1 for not satisfied at all and 5 for extremely satisfied) about your choice of resource?
• Would you select the same type of material for a similar assessment next time? Why?
• How confident are you that you (from 1 to 5 in Likert scale, 1 for not confident at all and 5 for extremely confident) have learned well from the selected type of learning resource?

Study Limitations

In this study, we will not focus heavily on broad learning patterns of students. Hence, it is not impossible that we may not see any particular correlation between the material used and the students' grades. In our study, we propose to let students choose their preferred media. If the results from this study are inconclusive, the authors plan to design a study where students will be randomly allocated a particular type of material and later surveyed and assessed in a similar manner.

Concluding Remarks

The results of this proposed study will provide information to help course designers provide the best type of media to support students to learn as effectively.

In addition, the results of this proposed study should provide information to help future first year engineering students make more informed choices on how to study more effectively. The study results will be useful in helping future first year engineering course students be successful in this particular assignment and in the course over all. The results of the study can be fed forward to students so they can learn the practices that made most students successful in the assignment.

The authors anticipate receiving helpful feedback from reviewers and colleagues at the CISSE conference.

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Computerizing Exams: The Michigan Tech Testing Center

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Abstract

Universities such as Brigham Young offer a central facility for computerized testing. Michigan Technological University is following this model with the establishment of the Michigan Tech Testing Center (MTTC) in 2012. The center creates a space that supports flexible, high integrity computerized exams. This paper focuses on the pilot testing of a spreadsheet lab practical using file submission through Canvas and Jotform.

Keywords

Computerized testing • Students • Online course • Instructors • Centralized testing center

Introduction

Many universities have added computerized testing centers primarily for standardized testing (FE, GMAT, GRE, TOEFL, etc.) [1–4]. At Michigan Tech, Brigham Young and the University of Nebraska-Lincoln, in addition to the standardized tests, the testing center optimizes the use of the center through offering testing services for students needing accommodations and faculty wanting to use the center for testing and standardized computer work [5–7]. This allows the center staff to be utilized when the center is not being used for the standardized tests, along with the center and associated testing rooms. Universities such as Weber State University and the University of Nebraska-Lincoln rely heavily on the use of testing centers in the administration of exams. In 2011 alone, approximately 120,000 exams were administered in the three testing centers at UNL and 300,000 at Weber State [7, 8].

For many exams at Michigan Technological University (Michigan Tech), the combination of ADA accommodations and university approved absences often results in up to 15 % of registered students needing alternative exam times and/or space [9]. These alternative exams have historically been left largely to faculty, resulting in inconsistent testing environments and supervision. In addition, at Michigan Tech, almost two-thirds of students enrolled in online courses are in residence [10]. Online course instructors are therefore expected to proctor exams which necessarily occur on multiple days/times to accommodate the flexibility expected by online students. Both of these factors mean significant faculty time and effort is spent scheduling and administering alternative exams for students. This can be dramatically reduced by utilizing a centralized testing center. This paper discusses the implementation of the Michigan Technological University Testing Center and the usage of the testing center for a standardized first-year engineering lab practical.

First-Year Engineering at Michigan Tech

The enrollment in the first-year engineering courses at Michigan Technological University (Michigan Tech) is approximately 900 students [11]. One of our assessment components in first engineering course is a spreadsheet lab practical. Students are provided with an Excel spreadsheet

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_51, © Springer International Publishing Switzerland 2015

template where they must (a) enter data, (b) complete calculations, (c) use spreadsheet functions, and (d) create graphs within a specific timeframe to test their basic spreadsheet skills. Planning a lab practical to be completed on computers, in real time, during class for this number of students is a complex, logistical challenge.

In order to assess these skills, the students cannot have access to the lab practical information ahead of their assigned lab practical time or (for test security) access the template after completion. We have traditionally run the lab practical with the use of a special user accounts created by our IT department prior to the lab practical. The computer accounts are set up such that students can only have access to Microsoft Excel and the lab practical template; they cannot access the internet or any of their personal files. The creation of these accounts and associated passwords requires about 2–3weeks to complete.

From 1999–2012, the lab practical for the first-year engineering program was scheduled using the following process:

1. Instructors logged into the lab practical user accounts and opened the lab practical template. This process takes about 15–20 min.
2. Groups of 3–4 students were scheduled per lab practical time.
3. At their designated time, these groups were brought into one of the first-year engineering classrooms. A timer was set for 20 min and the students began to work on their lab practical. The directions were placed in sheet protectors and taped to the desk.
4. Every 5 min, another group of students entered the classroom.
5. When one group finished, the instructor saved the files, deleted the previous students work, and opened a fresh version of the lab practical template.
6. Once all students had cycled through, the instructor logged off the computers. This took approximately 5–10 min.
7. The instructor then graded the lab practical files.

Three classrooms are traditionally used to administer the lab practical that have: 24 computers, 32 computers, and 34 computers, respectively, for a total of 90 computers. A single instructor, however, had access to only one classroom at a time and only for their specific class period of 80 min. With the time required to set up the computers, administer the lab practical, and log off, instructors were rushed and would occasionally run over time into another instructor's class. This time and labor intensive process drastically needed an update. An alternative came with the creation of the Michigan Tech Testing Center (MTTC) in 2012.

Michigan Technological University Testing Center

The creation of the Michigan Tech Testing Center was motivated by faculty frustration with test integrity, increasing ADA accommodations, and student exam conflicts in conjunction with the maturation of learning management and homework system testing technologies. After studying Brigham Young University's testing center model, where virtually ALL testing on campus is done centrally, the goal became to create a space that supports faculty work toward more flexible, higher integrity computerized exams and provides centralized, consistent accommodations for make-up and online exams [5].

Static multiple choice examinations given in "Scantron" format present increasing integrity problems as virtually every student now carries a camera and communication device (in the form of a smartphone or tablet) on their person. Computerized exams allow a virtually infinite number of exam permutations, both confounding student communication about answers and allowing the exam to be given at multiple times without loss of integrity. Essay answers can be typed, which students generally prefer and faculty find more legible.

Within a testing center, computerized exams dramatically decrease the workload and overhead associated with transferring the exam to the center and returning the exam to a faculty member. By securing exams within a variety of learning management and "locked down browser" systems, student access to other applications can be controlled. Exams can easily be allowed from only specific IP addresses, limited to certain times, and/or secured by a proctor password.

With all of these benefits, however, come the costs of staffing and equipment maintenance for a testing center. Michigan Tech's testing center has been seeded by a generous gift from alumnus William G. Jackson, but will be made sustainable by offering sponsored tests such as the NCEES Fundamentals of Engineering exam and the GMAT. These increasingly computerized, revenue-producing tests have specific infrastructure requirements similar to what will be used for computerized course tests. By using the center evenings and weekends for sponsored tests, we should be able to defray some of the cost of providing accommodated and online course testing for faculty.

ENG Lab Practical Logistics in MTTC

In the Fall of 2012, the testing center was open for faculty use. The pilot group, which included instructors testing the spreadsheet lab practical described above, proved somewhat challenging. Our lab practical was switched to an online format using CANVAS, our learning management system. To administer the test, students had to access the lab practical template only during their testing time for test security, and could not access it after their submission. The use of the "lock down browser" feature would not allow students to access the spreadsheet template. Nor was there a ready mechanism in Canvas for file submission without future student access. Students could submit a file through an assignment, but could access the file later.

To address these challenges, we used a Canvas quiz to provide the background information for the lab practical. This information included instructions and a link to the template file. The template was uploaded into Canvas and locked so that students could not access it unless the instructor linked to it. The quiz itself was IP address locked to the testing center location and specific lab practical times. Therefore students could not access the quiz unless they were in the testing center during their scheduled time.

Submission of the completed Excel file was provided through the Jotform Website (http://www.jotform.com/) [12]. Jotform is a tool that allows the creation of a web form through a word-processing-like interface. We created a simple form that just asks users to identify and upload a single file. This form was then published in HTML code (essentially a single iframe tag) within the Canvas quiz question, giving us the ability to upload a student's completed Excel template.

There are three ways that the files have been accessed for grading. In the traditional method that was used from 1999–2012, grading either occurred during the lab practical or the files were stored in an alternate drive. In 2012–2013 academic year, the files were saved to JotForm and the instructor accessed the files via web links. Then the files were downloaded to the instructor's drive to be graded. In 2013–2014 academic year, Canvas was updated such that the student could upload the file to Canvas without access to the file later. The instructor could access the file for grading through "Speed Grader" or could use "Speed Grader" as an avenue to download the files and then grade them. The latter method was used this fall. The teaching assistants downloaded the student files, followed by the instructor grading each individually.

Two instructors used this new method in one of the first-year engineering courses (ENG1001) with approximately 200 students. Our updated procedure using the testing center was as follows:

1. The students were scheduled in groups of 38 students (or about ½ of the class of 60 students). Students who needed ADA accommodations were scheduled in the first group such that if they needed extra time, it was during the scheduled testing period.
2. Students were given brief instructions regarding the process of starting EXCEL and accessing the Canvas quiz.
3. Students came at their scheduled time, completed their exercise and uploaded their file to Canvas.
4. All instructions were on Canvas or within the EXCEL template. For the MATLAB program, all the instructions were on Canvas.
5. The files were graded outside the Lab Practical.

As shown, the logistics of using the testing center were much simpler than using the classroom machines. Additionally, managing the process was easier because it just required having half of a class complete the lab practical and Canvas kept track of the time. The classroom environment was quieter because once the students began working, there were few distractions. In contrast, the prior method had students entering and leaving the classroom every five minutes, along with the instructor grading and saving the files.

Student Feedback

Since this method was drastically different from the previous testing procedure, students were asked to complete a short survey on Canvas after the Lab Practical to assess their perception of the lab practical process. The results indicated that 83 % of the students felt the instructions were clear and easy to follow, while 1.5 % of the students had indicated that they had several questions on the procedure. In this same survey, we asked students about their stress level before and during the lab practical. They ranked this on a scale from 1 to 5 (1 being very stressed, and a 5 being not stressed at all). As shown in Fig. 1 below, generally students had the same level of stress before and during the lab practical. There is a slight difference in the left side of the graph (stress levels of 1 and 2). This change, however, is not statistically significant ($p = .23$). From this, we concluded that the use of this lab practical procedure and testing center did not adversely affect the students due to stress.

As part of the lab practical evaluation survey, we asked for student comments. The comments focused on five main problem areas as shown in Fig. 2: (a) computer issues, (b) timing, (c) instructions, (d) submission issues, and (e) exam proctors. Out of the comments, about half were positive. Approximately 15 % of student comments dealt with computer issues they experienced during the lab practical. Ten students stated issues with the computers (broken keyboards, long login time, slow computers) and seven cited issues with the file submission in Canvas. Fourteen students would have liked more time on the lab practical itself or for file submission and 19 % thought the instructions needed some improvement.

These comments were used to improve the lab practical for future offerings. Based on the success of this first pilot run with the spreadsheet lab practical, first-year engineering instructors also used the testing center for a MATLAB lab practical in the Spring of 2013. Combined with the testing from other pilot faculty, the MTTC was used to administer

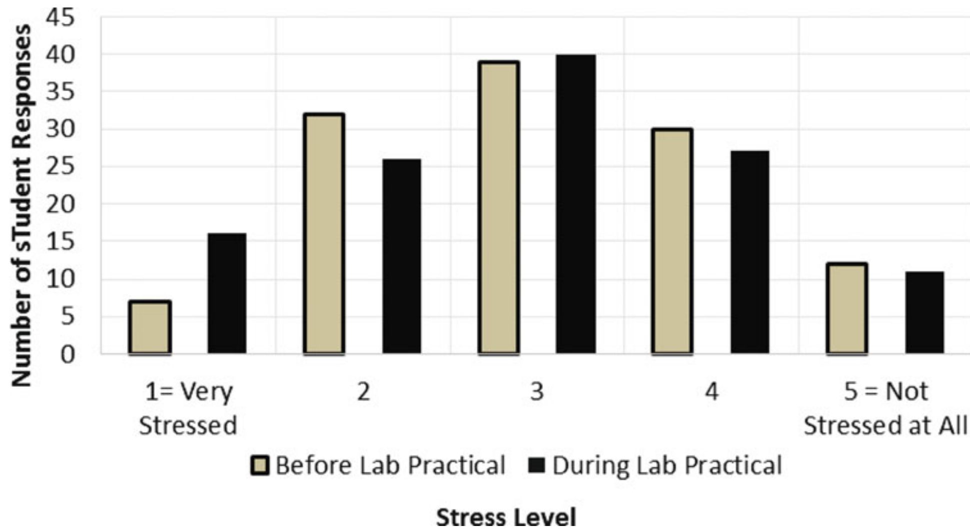

Fig. 1 Self-reported stress level of lab practical on students (n = 120)

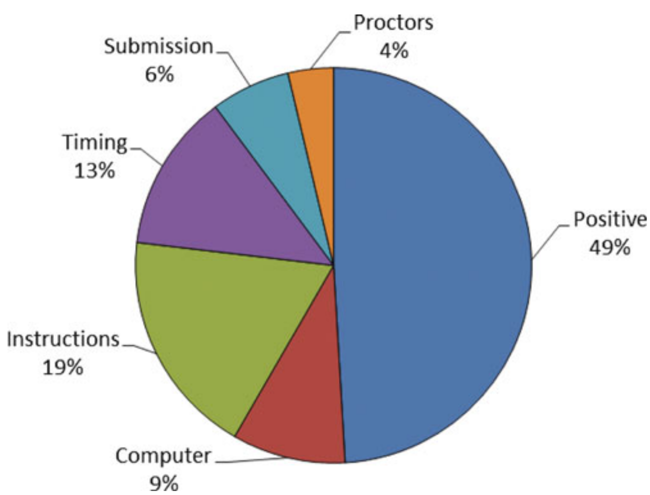

Fig. 2 Student comment breakdown by topic (n = 108)

approximately 1,500 exams during the Spring of 2013. This method of testing is quickly becoming popular on campus. Looking at data from the current semester, approximately 2,200 exams have been administered in September and October alone [6].

Conclusions and Final Thoughts

The establishment of the Michigan Tech Testing Center addresses faculty concerns regarding test integrity, ADA accommodations, and flexibility with online learning. The pilot tests administered by pilot faculty have demonstrated the feasibility of the testing center for use with the first-year engineering lab practicals and other exams. Survey results from first-year engineering students have indicated no significant change in stress level while taking the lab practical in the testing center and the logistics of the lab practical are simpler for the instructor.

While this has been demonstrated at a pilot scale, challenges still exist to the full implementation for our 900 student population primarily due to the number of computer seats available in the current testing center. Lessons learned indicate that the lab practical can be successfully administered through the Canvas LMS using Jotform submission or the file submission question through a Canvas quiz. Issues brought up through student comments on instructions have largely been addressed. Using the IP lock can limit access to the lab practical to the classroom computers, which would increase our computer pool, but this has not been completely tested.

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Gravitation Search Training Algorithm for Asynchronous Distributed Multilayer Perceptron Model

Natalya P. Plotnikova, Sergey A. Fedosin, and Valery V. Teslya

Abstract

More popular principles of neural networks distribution are trainset parallel processing, matrix operations parallel processing or parallel processing of neural networks collections. This paper describes architecture of asynchronous distributed system for training and simulating multilayer perceptron based on the actor model. Usual backpropagation training algorithms for multilayer perceptrons, such as gradient training, Levenberg-Marquardt training, RPROP training have a common disadvantage: all of them are partially synchronous. Global optimization algorithms (genetic training, annealing algorithm etc.) are more appropriate for distributed processing but most of them are slow and inefficient. Gravitation search algorithm is new global optimization procedure combining good convergence, efficiency and algorithm step dispensability. The paper develops asynchronous distributed modification of this algorithm and presents the results of experiments. The proposed architecture shows the performance increase for distributed systems with different environment parameters (high-performance cluster and local network with a slow interconnection bus).

Keywords

Actor Model • Gravitation Search • Multylayer Perceptron • Neural Network • Training Algorithm

Introduction

Multilayer perceptron is a type of artificial neural networks. It is capable of solving quite a wide area of problems: function approximation, classification, image recognition and others [1–4]. Nowadays, at the time of great data sets current scientific and practical tasks are multidimensional and large-scaled, for example, climate prediction, medical expert systems, scientific experiment modeling. For this reason, creating efficient reliable scalable distributed neural network has become an important research area. The most popular neural network distribution techniques are: parallel processing of the neuron groups, trainset parallel processing, matrix operation parallel processing, neural networks collection parallel processing.

The most obvious approach to gain the efficient of multilayer perceptron distribution is to unite neurons inside the layer into a few groups and perform parallel calculations for each group.

Trainset parallel processing does not touch training or simulation algorithms. The main idea of this approach is to distribute parts of trainset between computational nodes. And each part of trainset is processed by the whole network simultaneously. There are no parallel steps in training or simulating algorithms.

Matrix operations are the base of training and simulation. And for large neural networks (about 1000-2000 neurons) their parallel or distributed implementation allows to get a significant performance increase. But in this case neural network model becomes fully mathematical and its original nature (from neural networks biological prototypes) gets lost.

Neural networks collections solve a problem by using several different neural networks. All of them solve the same problem parallel. And at the end of simulation or

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_52, © Springer International Publishing Switzerland 2015

training a handler chooses the neural network with the best result.

The main purpose of this research is to develop a new efficient neural network distribution approach and training and simulation procedures for it. For this we have used methods of system analysis, artificial neural networks theory, optimization theory, functional and distributed programming, and experiment planning theory.

The main idea of the proposed approach is to create a fully distributed neural network model based on the principle "one neuron—one process" and to distribute training and simulation algorithm steps between simple units—neurons. The best concept for these purposes is the actor model [5]. The actor model describes calculation process as a result of interaction between active objects—actors. Actors interact with each other by asynchronous message passing. At a time each actor may process the received message according to implemented behavior or wait for a new message [6].

To implement this idea we need to choose a proper training algorithm.

Training Algorithm

Algorithm Review

Multilayer perceptron training algorithms may be classified into two main groups: local optimization algorithms and global optimization algorithms.

Local optimization algorithms, such as different variation of gradient training, use backpropagation procedure to determine local minimum of error function and find optimal set of network weights and biases [7–9]. Multilayer neural network architecture affects backpropagation training process in such a way, that the operations going during this process are partly synchronous. Only neurons of one layer may perform calculations simultaneously. Distributed version of gradient algorithms does not allow using all the benefits of parallel processing.

Global optimization algorithms try to find global minimum of error function and use unusual ways of processing this function. For example, the genetic algorithm models natural evolution process and introduces new mathematical terms in accordance with the real evolution and reproduction principles [10]. Most of the known versions of neural network global optimization algorithms include calculations independent for all neurons. The only bottleneck is a forward signal propagation process during error function calculation.

Gravitation Search Algorithm

Gravitation search is a new algorithm of global optimization proposed by Rashedi in 2009 [11, 12]. This algorithm uses principles and laws of mass movements and gravitation interactions. Original algorithm includes seven steps:

1. To generate corpuscle system

$$S = \left\{p_i = \left(p_i^1, p_i^2, \ldots, p_i^n\right) \epsilon X\right\}_{i=1}^{N} \tag{1}$$

   where N—maximal number of corpuscles, p_i—i-th corpuscle coordinates in n-dimensional space
2. To calculate fitness function $f(X)$ for each corpuscle
3. To calculate current value of gravitation constant

$$G(t) = \frac{G_0}{e^{\alpha t}} \tag{2}$$

   where G_0—gravitation constant initial value, α—control constant, and mass

$$M_i(t) = \frac{m_i(t)}{\sum_{j=1}^{N} m_j(t)} \tag{3}$$

   where

$$m_i(t) = \frac{f(p_i) - \max_{j\epsilon\{1\ldots N\}} f\left(p_j\right)}{\min_{j\epsilon\{1\ldots N\}} f\left(p_j\right) = \max_{j\epsilon\{1\ldots N\}} f\left(p_j\right)} \tag{4}$$

4. To calculate the resulting force for each corpuscle

$$F_i(t) = \sum_{j=1, j\neq i}^{N} \xi_j F_{ij}(t) \tag{5}$$

   where ξ_j—random value, uniformly distributed on the interval (0; 1),

$$F_{ij}(t) = G(t)\frac{M_i(t) M_j(t)}{\left\|p_i, p_j\right\| + \varepsilon}\left(p_j(t) - p_i(t)\right) \tag{6}$$

   where ε—a small constant.

   Eq. (6) represents interaction force between corpuscles i and j.
5. To calculate corpuscle velocities

$$v_i(t+1) = (\zeta_1, \ldots, \zeta_n)^T * v_i(t) + \frac{F_i(t)}{M_i(t)} \tag{7}$$

   where $(\zeta_1, \ldots, \zeta_n)^T$—vector of random values, uniformly distributed on the interval (0; 1); *—element-wise vector multiplication.

6. To update corpuscle positions

$$p_i(t+1) = p_i(t) + v_i(t+1) \tag{8}$$

7. To repeat steps 2-6 until one of the following criteria takes place: exceeding the fixed epoch count or reaching the desired fitness function value.

Based on these steps asynchronous distributed modification of gravitation search optimization has been developed and implemented for training multilayer perceptron. Weights of multilayer perceptron represent multidimensional coordinates of one corpuscle. So, the system of corpuscles is a set of several potential weight values. Fitness function in this case is represented by the value of neural network SSE (Sum Squared Error). Dimension of optimization problem is equal to the number of interconnections between multilayer perceptron neurons.

The existing training algorithms based on gravitation search use only some ideas of original algorithm and include mostly other principles and steps of other optimization techniques [13, 14].

The next essential step of our research was to choose a development platform.

Development Platform

As a perfect tool for developing asynchronous distributed neural network model Erlang/OTP platform was chosen—a set of libraries written in functional programming language Erlang, designed for creating reliable distributed applications [15–17].

Erlang/OTP includes predefined modules implementing generalized behaviors of actors for the actor model: gen_server (generalized server for message passing), gen_fsm (generalized finite state machine), gen_event (generalized events handler). Since neural networks are presented as a set of interacted units—neurons, the most appropriate behavior for the system entities is generalized server. Behavior gen_server maintains such an important ability for the actor model as handlers for asynchronous message processing.

The functional programming language allows writing mathematically correct programs almost without debugging. Erlang virtual machine implements lightweight internal processes inside each Erlang-node and an efficient message queue. This ability provides fast and reliable interconnection not only between processes of one node, but also between processes of different nodes on different physical devices. Moreover, Erlang virtual machine maintains several standard interconnection protocols, in particular ssh, so there is no need to implement additional protocols and construct new structured packets for messages passing between neurons and other logical elements of the model [18, 19].

These advantages make application development much easier, so we may mostly focus on architecture features.

Asynchronous Distributed Model

Asynchronous distributed multilayer neural network model consists of three abstraction levels: NEURON, MLP and NNET. The first level—NEURON—encapsulates functions of elementary neural network units—neurons. The second level—MLP—generally implements a concrete neural network type. In this research we have considered multilayer perceptron. That is why this level includes layer and neuron initialization, training and simulation process handling. The third level—NNET—is a generalized neural network. This level starts and controls other levels and also contains handlers for service purposes, such as data input and output, neural network main settings.

All levels are represented as Erlang/OTP behavior gen_server. For NNET and MLP levels one process per level is started. For NEURON level the number of processes is the same as the size of modeled neural network. Each neuron process has a state for saving some important information. There is no data redundancy, so each neuron "knows" only what it should, in particular, the state contains weights and biases only for input links connected with the current neuron.

Detailed interconnection scheme for levels MLP and NEURON is presented in Fig. 1.

According to the scheme training process begins with block "Start epoch". This block initializes the trainset and prepares data for simulation. After that the only synchronous part of the model,—simulation process,—begins. Since we have used asynchronous message passing, we need an additional synchronization procedure. This procedure has been implemented in three steps. "Start simulation" block of MLP level sends messages to all input neurons on NEURON level and quits. Neurons propagate the received signal from layer to layer for each corpuscle saved in neuron's state, until output layer is reached. "Stop simulation" block of MLP level gathers messages from all output neurons, and when the count of received messages becomes equal to the number of network outputs this block sends the gathered data to the next block—"Simulation result". The only purpose of "Simulation result" block is to calculate error function value and make decision about the next step of training algorithm. If the best error (among errors for all corpuscles) is more than the desired value, the algorithm goes to the next block of MLP level—"Start gravitation search". This block calculates masses and gravitation constant according to gravitation search algorithm and then sends a control message to "Calculate distance" blocks of all NEURON level neurons. Neurons use their parts of information about corpuscle coordinates to calculate their parts of Euclid distances (the sum of corresponding square difference of coordinates) between all the possible

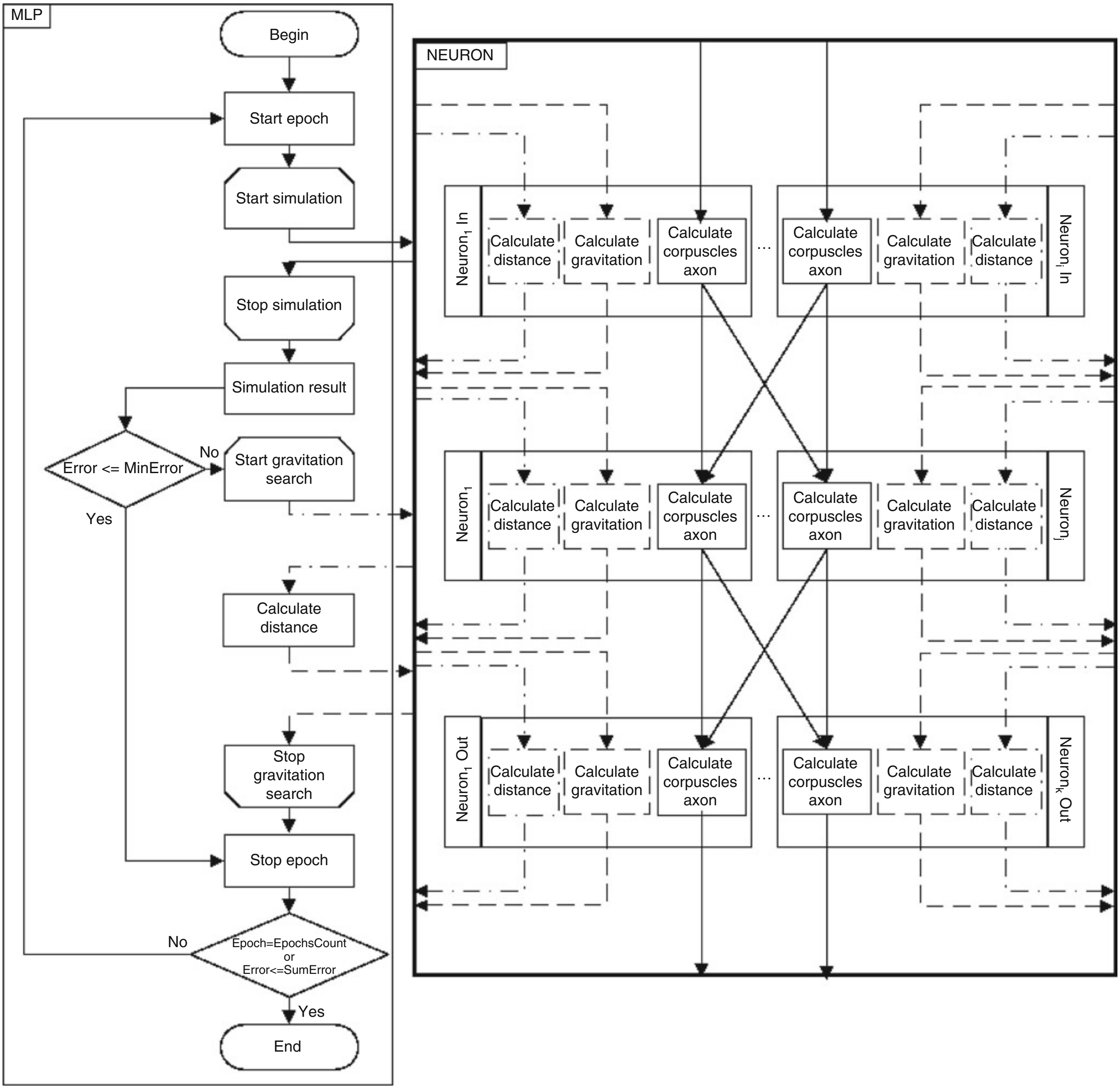

Fig. 1 Interconnections between MLP and NEURON levels of asynchronous distributed neural network model during training process. Gravitation search training algorithm

corpuscles. "Calculate distance" block of MLP level gathers all partial sums and finally calculates all the distances and all the values of F_{ij}. After that, values of F_{ij} are sent to neurons to calculate gravitation forces, velocities and new corpuscle coordinates. This step is performed by "Calculate gravitation" block of NEURON level. "Stop gravitation search" block of MLP level gathers messages about coordinate correction from all neurons and when all the parts of corpuscle coordinates become updated this block sends a message to "Stop epoch" block to complete the current epoch. If there are no appropriate conditions to stop training process, this block starts a new epoch by sending a message to the first logical block of MLP level—"Start epoch".

Experiments and Performance Tests

Multilayer Perceptron Architecture

All experiments and tests have been carried out on multilayer perceptron with the following architecture. The number of layers is equal to three (input, hidden and output). Such networks are capable of solving problems of any dimension, if quite sufficient number of input and hidden neurons is used [1–3]. The number of input and hidden neurons is varied from 100 to 1000 with step 100. The number of outputs is equal to one. The sum number of neurons is equal to 2*N + 1. Hidden neurons use sigmoid activation function. Output neuron uses linear activation function.

Trainset size varies from 10 to 50 pairs with step 10. Input data are randomly generated and presented in binary form.

Corpuscle count varies from 10 to 50 sets with step 10.

Experiment Conditions

All the experiments have been conducted on the systems of three types:

1. Distributed system deployed to the high-performance IBM cluster is represented as 15 virtual machines interconnected by internal cluster high-speed bus.
2. Distributed system deployed to the local network is represented as 15 physical machines (1 core for each machine) interconnected by slow bus (usual LAN cable with maximal transmission speed 100 Mbit/s).
3. Hybrid system is based on the system 2 but each physical machine has four processor cores.

Table 1 presents the main characteristics of high-performance IBM cluster.

Table 2 describes experimental platforms for all types of systems.

We have examined an asynchronous model with different sets of options: fixed trainset size and corpuscle count, varied neural network size and node count; fixed trainset size and neural network size, varied corpuscle count and node count; fixed neural network size and corpuscle count, varied trainset size and node count.

Table 1 High-performance IBM cluster characteristics

Characteristic	Value
Blade count	3
Processors	2 per blade 4 cores per processor
RAM	16 GB per blade
Interconnection bus speed	16 Gbit/s
High-level software	VMware vCenter Server Version 5.0.0 Build 455964

Table 2 Experimental platform characteristics

Characteristic	Value
Processors	System 1: 1 processor 1 core System 2: 1 processor 1 core System 3: 1 processor 4 cores
RAM	2GB
Operation system	CentOS 6.4 Linux 2.6.32–279.el6.x86_64
Development software	Erlang R15B01 (erts – 5.9.1)

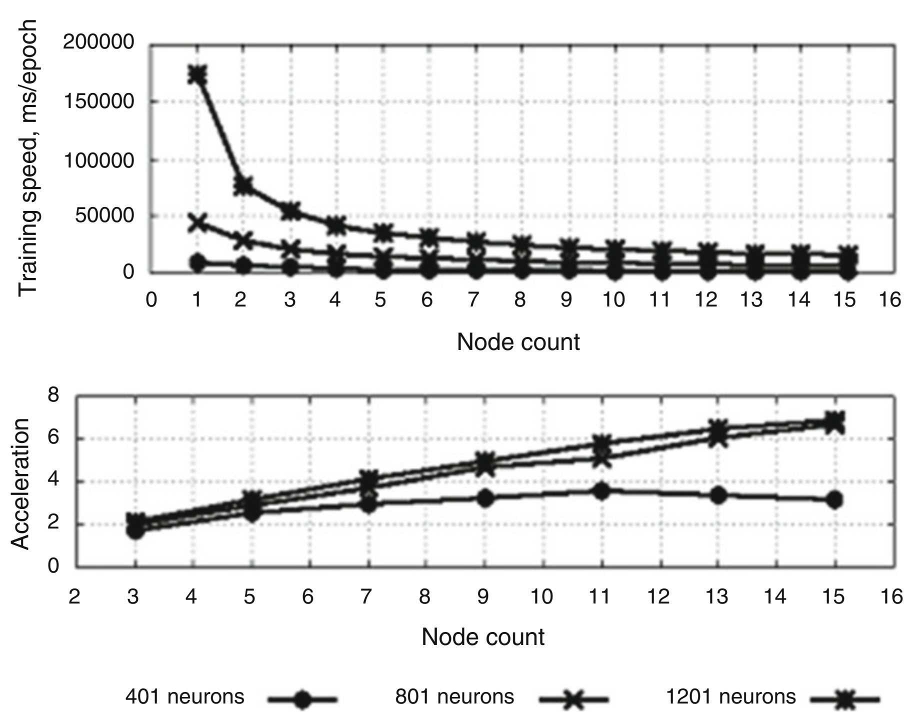

Fig. 2 Experiment results for system of type 1

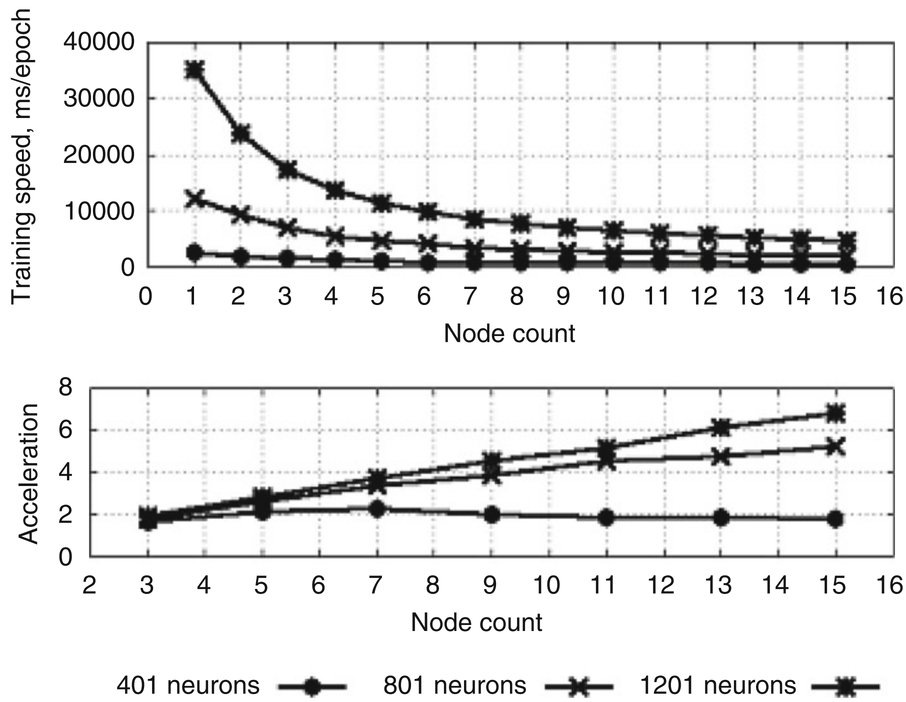

Fig. 3 Experiment results for system of type 2

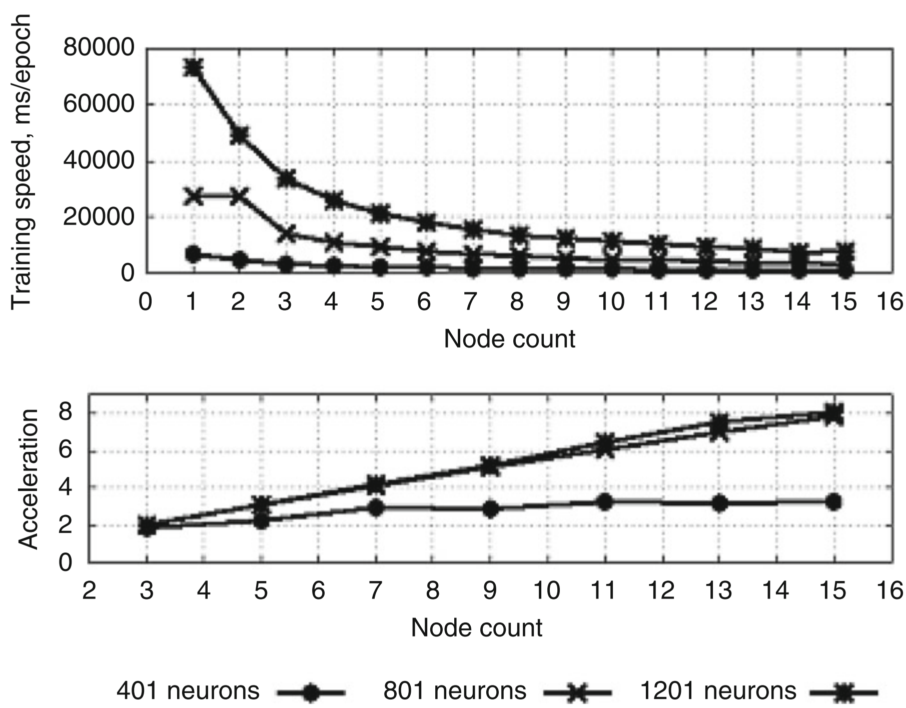

Fig. 4 Experiment results for system of type 3

The obtained results of our experiments are presented in a set of charts. The most interesting results reflect dependency of training speed and acceleration on the node count with fixed trainset size (10 pairs) and corpuscle count (10 corpuscles). These dependencies for the described systems and neural networks with 401, 801 and 1201 neurons are presented in Figs. 2, 3, and 4.

The presented results evidence a good distribution ability of gravitation search, especially for large networks (801 and more neurons). Acceleration depends on the node count almost linearly. If we use 15 nodes we can get about eight times acceleration of the training process.

Conclusion

In this paper, we have proposed asynchronous distributed actor multilayer perceptron model with global optimization training algorithm—gravitation search.

A new model has been developed using functional programming language Erlang and platform Erlang/OTP with predefined base implementation of actor model functions.

The model includes three abstraction levels: NNET, MLP and NEURON. Such architecture allows encapsulating some general features on general levels and some specific for the considered neural networks features on special levels. Asynchronous message passing between levels allow to differentiate synchronous and asynchronous parts of training and simulation algorithms and, as a result, to improve the use of resources.

The distributed implementation of the gravitation search inside the actor model have showed good convergence tendency and distribution ability. Our experiments with the developed system and different types of distribution platforms have evidenced efficiency of the proposed model.

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Application of Image Processing Techniques to the Identification of Phases in Steel Metallographic Specimens

Adarsh Kesireddy and Sara McCaslin

Abstract

Metallographic image processing focuses primarily on image segmentation, edge detection, and approximating grain size. This paper presents the results of applying a radial basis function neural network to the image texture data obtained from steel metallographic specimens to determine the feasibility of the automated recognition of steel phases.

Keywords

Image processing • Metallography • Artificial neural networks • Texture

Introduction

Metallographic image processing technology is one of the easiest, widespread and effective methods of research and testing in materials science. Metallographic testing is very important in various countries, as shown by the ISO international material test standard and ASTM material test standards. For example, ASTM E1382-97 covers automatic and semiautomatic determination of average grain size using image processing [1]. New methods of testing are continually being developed.

This paper combines the concepts of image processing with neural networks to recognize the predominant phases in steel specimens when viewed with a metallurgical microscope.

Background

Digital Imagery

An image is defined as a two-dimensional light intensity function $f(x,y)$, where x and y are spatial coordinates. The values of f at any given point are proportional to brightness or gray level of the image at that particular point. An image can be considered as matrix in which each row and column identifies a point in the image and the corresponding matrix element value represents gray level at that point. These elements are called pixels.

Computer vision is the recognition of objects or structures from images with the help of their properties, which can be either geometric or material. The purpose of computer vision is to differentiate the state of the physical world from noisy or ambiguous images. It is basically concerned with analyzing the surface and properties of three dimensional objects from their two-dimensional representations.

A dot is the minimum unit of visual communication. When dots are very close to each other, patterns, like a line or circle, are formed. This effect is known as grouping. It is also possible that what is perceived from an image may not be ambiguous or illusory, but can be globally unrealized, meaning that one cannot physically construct the perceived three-dimensional object in this entirety from what is observed.

Computerized methods of analyzing metallographic images are preferred because of their repeatability and consistency in interpreting data, and in some instances have been found to be up to 100× faster than human interpretation [2]. Digital image processing is a subdivision of such computerized methods that is growing in popularity, as evidenced by a recent tutorial by Grande [3].

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_53, © Springer International Publishing Switzerland 2015

Various metals have been studied in this context. Codaro et al. used image processing to characterize the shape and parameters of corrosion pits in aluminum and titanium [4]. Lee et al. used digital image processing for automatic detection and characterization of cracked constituent particles/inclusions in wrought aluminum alloys [5]. Petersen et al. also used texture of ore samples, but their focus was on characterization of their properties rather than classification [6]. Singh and Rao used the visual texture of ore samples combined with a radial basis neural network to sort and classify ore, and achieved an overall accuracy of 88.71 % [7].

The proposed research focuses on iron and steel specimens. Latala and Wojnar demonstrated that computer aided methods of assessing grain size in a single phase material, such as austenitic stainless steel, had a definite advantage over manual methods [2]. Dengiz et al. used a combination of fuzzy logic and neural networks to detect grain boundaries in superalloy steel [8]. Zeljkovic used automatic pattern classification on steel [9], and Peregrina-Barreto et al. were able to determine grain sizes in steel using automated methods (without human intervention) [10]. Jiang studied image edge detection use mathematical morphology in the context of metallography for edge detection in steel [11]. Kotas et al. combined segmentation and spectral methods to retrieve the image of interest from a steel specimen, then used statistical methods for analysis and classification [12]. Cavallini et al. studied graphite nodules using image processing [13], as did Prakash et al. [14]. Prakash et al. developed an analysis system to testing and quantifying ductile cast iron [15] and classify cast iron [16]. Cracked particles with width of 0.5 μm or larger can be detected and segmented using image processing [17].

In addition, these methods have applications far beyond examining microscopic images of metal specimens. Yan et al. was able to extract geometric parameters of metal beads caused by fire [18]. Barron et al. made of use neural networks for recognizing the size distributions of blast fragments [19]. Elunai used image texture analysis to analyze the surface of roads [20].

This works focuses in phase identification, rather than geometry identification, similar to the work of Singh and Rao with the classification or ore samples [7].

Methodology

Sample Preparation

The first step of image processing is acquiring the image. This involved preparing steel specimens for viewing under a metallurgical microscope.

Metal samples were cut to desired dimensions using a Buehler Samplmet-2 Abrasive Cutter. With the help of Buehler Sampler-Kwick liquid and Buehler Sampler-Kwick powder, an epoxy disc containing the specimen was made to securely hold the sample for the grinding/polishing/etching process and later for clear viewing under microscope.

The specimen was then processed on a Buehler Ecomet 6 variable speed grinder/polishers for flattening the surface of the specimen and polishing it to make it clear to view under microscope.

After processing through the grinding and polishing steps, two drops of 3 % Nital etchant were applied to the surface of the specimen, allowed to remain for approximately 2 s, rinsed, and air dried. Etchant was used to distort the smooth surface, to help view clearly grain structure under microscope.

A Nikon Type 104 microscope with 50*/0.80 CF plan lens and 20*/0.46 CF plan lens and 5*/0.13 Nikon Japan plan lens and 0*/0.30 CF plan lens images were acquisition. DMP 1000 software was used with a digital camera mounted to the microscope to capture the images. The desktop computer which processed this operation had 2 GB of RAM. The entire image acquisition was processed in The Material Science Lab in The University of Texas at Tyler.

Specimens were prepared for the following types of steel alloys:

1. ASTM 1038 steel for pearlite
2. Carbon steel for ferrite
3. Damascus steel for martensite and for cementite

Pre-processing

Pre-processing involves preparing an image for analysis. In this research, the original images were captured as RBG color images, and were then converted to grayscale. Sixteen distinct grayscale levels were used.

Subimages were generated from each main image based on regions of interest representing steel phases were extracted.

For consistency between the images, histogram equalization was used to distribute the gray-scale levels across all sixteen levels, distributing the grayscale contrast across the range of possible values.

Processing: Statistical and Textural Analysis

Texture methods used can be categorized as follows: statistical, structure, model-based and signal processing features. Many methods are available to characterize the texture of materials directly from gray level images; the principal approaches are statistical, structural, morphological, fractal and spectral. This research focused on statistical measures of texture.

The image processing began by finding the mean, median, standard deviation, and histogram for multiple subimages representing a known steel phase.

Next, statistical measures related directly to the image texture were obtained for the subimages. The properties used in this research are listed below, where $P_d(i,j) = P(i,j)/R$ where R is the total number of pixel pairs (i,j).

The complexity of image is studied by entropy. When the value is zero, pixels have same intensity [21].

$$Entropy = -\Sigma_i \Sigma_j P_d(i,j) log P_d(i,j) \tag{1}$$

Local variations in intensity are measured with contrast [21].

$$Contrast = \Sigma_i \Sigma_j (i-j)^2 P_d(i,j) \tag{2}$$

Energy is a measure of the homogeneity of an image. Hence it is a suitable measure for detection of disorders in textures. For homogeneous textures value of energy turns out to be small compared to non-homogeneous ones [7].

$$Energy = \Sigma_i \Sigma_j \{P_d(i,j)\}^2 \tag{3}$$

$$Homogeneity = \Sigma_i \Sigma_j \left(\frac{1}{1+(i-j)^2} \right) P_d(i,j) \tag{4}$$

Generating the Neural Network

There are various types of neural networks that have been developed, and all of them have applications in classification (e.g., classifying, or registering, and image). Radial basis neural networks have been found very useful for image classification. In general, research has shown that neural networks show more superiority than fuzzy logic algorithms in image detection [21].

For this work, the radial basis function network was chosen. The radial basis neural network has a strong resemblance to the distance, weighted regression or memory based and inspired by a curve fitting problem in multidimensional space.

Figure 1 shows a radial basis neural network with a hidden layer. The input layer is composed of input nodes that are equal to the dimension of the input vector x, which means there are the same number of input nodes as there are features used to identify the specimen. The input layer made of source nodes and works as an information recipient. Nonlinear transformations from input space to high dimensional hidden space are in the hidden layer. The last layer is the output layer.

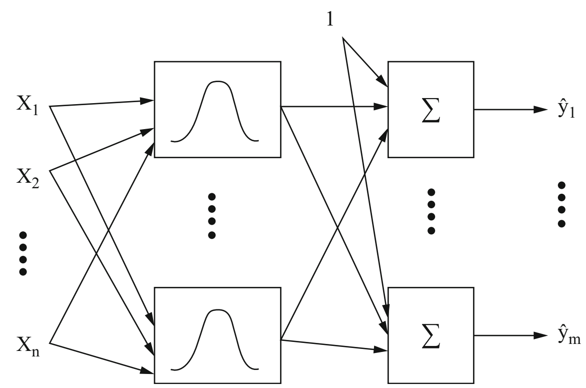

Fig. 1 Basic radial basis function network with multiple outputs and a hidden layer [22]

The output of the *j*th hidden neuron with Gaussian transfer function can be calculated as shown in Eq. 5 [23].

$$h_j = e^{-||x-c_j||^2/\sigma^2} \tag{5}$$

Note that h_j is the output of j^{th} neuron, x| is an input vector, c_j is the j^{th} radial basis function center, **σ** is the center spread parameter which controls the width of the radial basis function and $||.||^2$ represents the Euclidean norm. The output of any neuron at the output layer of radial basics function network is calculated as shown in Eq. 6 [23].

$$y_i = \Sigma_{j=1}^{k} w_{ij} h_j \tag{6}$$

The nonlinear activation function in the neuron is usually chosen to be the smooth step function, such as the sigmoid function [24]. This is given by Eq. 7 below.

$$sigmoid[x] = \left(\frac{1}{1+e^{-x}} \right) \tag{7}$$

The training algorithm implemented was the widely used Levenberg-Marquardt (LM), which is the default training algorithm in the Neural Network add-in package for *Mathematica*. The goal of the algorithm is to minimize the root mean square error by adjusting the parameters of the neural network. Note that it works well even for ill-conditioned problems [24], and outperforms the simple gradient descent and other conjugate gradient methods in a wide variety of problems [25].

A vector of data including the entropy, contrast, energy, homogeneity, number of histogram peaks (for the grayscale version of the image), and the percentage of black pixels present in a binarized format of the image were provided as input to the network. The output of the network would indicate which steel phase was predominant in the image.

A subset of the subimages was selected and used train the neural network by providing their data vector (as described in the previous paragraph) along with an indication of what type of phase was predominant, based on human visual identification. Fifty samples of each phase were used for training. Once the network was trained, another subset of the subimages with known phases was provided to the network to verify its accuracy. Fifty different samples were used for validation.

Results

Image Acquisition

Figures 2, 3 and 4 show the main images used for analysis. Each main image was broken into multiple subimages for manual identification, followed by analysis, network training, and network validation.

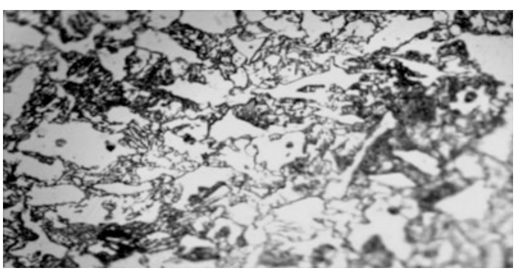

Fig. 2 Image of 1038 steel specimen, where the light colored areas are ferrite and the darker, striped areas are pearlite

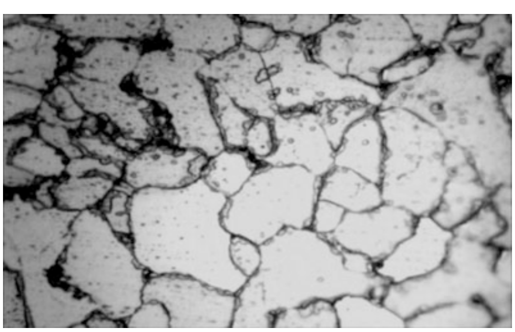

Fig. 3 Image of low carbon steel specimen, where the light colored areas are ferrite

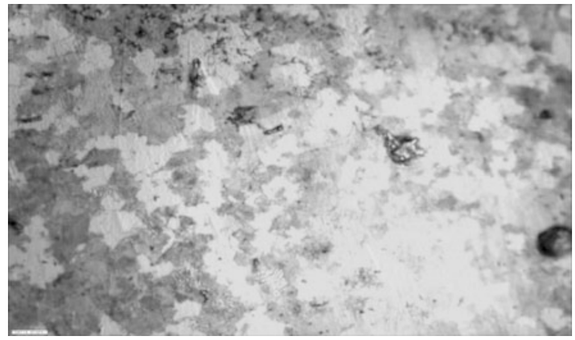

Fig. 4 Image of Damascus steel, with a mixture of cementite and martensite

Table 1 Textural results for 566 pearlite subimages

	Entropy	Contrast	Energy	Homogeneity	Black pixels
Mean	1.39	0.015	0.00086	0.055	41.4
STD	3.11	0.00368	0.00030	0.0018	6.50
Max	1.39	0.022	0.0019	0.060	66.0
Min	1.39	0.0058	0.00052	0.052	26.3

Table 2 Textural results for 245 ferrite subimages

	Entropy	Contrast	Energy	Homogeneity	Black pixels
Mean	1.39	0.012	0.0015	0.057	73.7
STD	2.67	0.0054	0.00058	0.0024	7.49
MAX	1.39	0.036	0.0032	0.061	93.87
MIN	1.39	0.0026	0.00057	0.051	52.14

Table 3 Textural results for 97 matensite subimages

	Entropy	Contrast	Energy	Homogeneity	Black pixels
Mean	1.29	0.0089	0.0020	0.058	62.1
STD	0.15	0.0044	0.00084	0.0022	17.3
MAX	1.39	0.016	0.0038	0.062	91.9
MIN	1.04	0.00044	0.00084	0.054	24.9

Textural Analysis

Tables 1, 2, 3, and 4 show entropy, energy, contrast, homogeneity, % black pixels for extracted subimages, respectively.

Neural Network Training

The results for the radial basis function neural networks with six inputs and one output for each phase, trained with the Levenberg-Marquardt algorithm, are presented for two

Table 4 Textural results for 150 cementite subimages

	Entropy	Contrast	Energy	Homogeneity	Black pixels
Mean	1.34	0.010	0.0016	0.057	44.7
STD	0.12	0.0028	0.00050	0.0014	15.8
Max	1.39	0.018	0.0036	0.062	87.1
Min	1.04	0.0017	0.00078	0.053	20.1

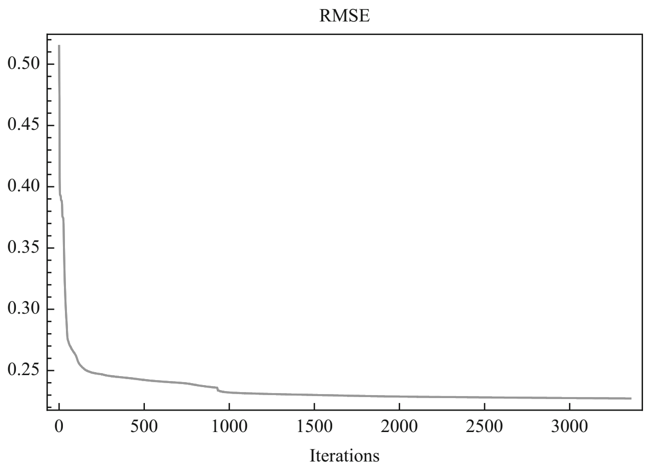

Fig. 5 Root mean square error for training the radial basis function using 50 training samples for each type of phase

cases. Fifty samples of each phase were used for training, and 50 samples were also used for validation.

For the first case, all features were used and four outputs were implemented. Each output had a value between 0 and 1, where 1 is true and 0 is false. The values were rounded to a crisp value of 0 or 1. For example, if the sample was predominantly cementite then the cementite output would have a value of 1, while all other outputs would be 0. Using the *Mathematica* Neural Network add-in package, this network converged after 3,357 training iterations to Root Mean Square Error of 22.7 %. This is shown in Fig. 5.

Neural Network Validation

The classification of validation samples shown in Fig. 6 is summarized in Table 5. Values along the diagonal were correctly classified, while off-diagonal values were noted. This table shows that the greatest confusion occurred with reference to martensite and cementite.

The level of confusion between martensite and cementite were a source of concern, therefore cementite was eliminated from the network to see if the accuracy of the neural network would be increased.

On eliminating cementite the root mean square error was reduced to 17.2 % in 3,336 iterations, as shown in Fig. 6. The classification of validation is summarized in Table 6.

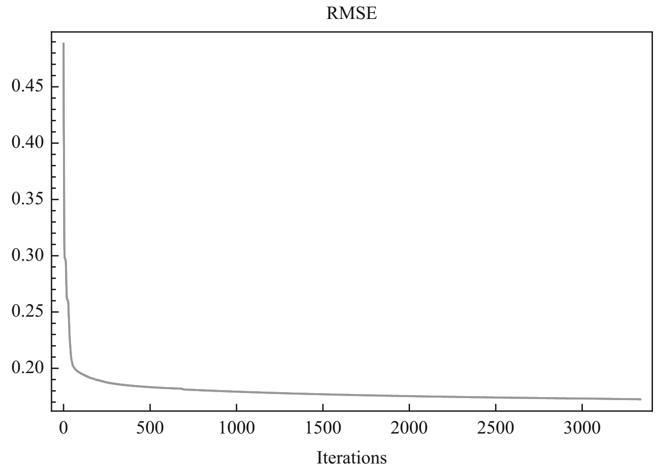

Fig. 6 Plot of root mean square error versus iterations with cementite eliminated

Table 5 Classification of validation samples

	Pearlite	Ferrite	Martensite	Cementite
Pearlite	49	0	1	0
Ferrite	0	48	0	0
Martensite	4	0	39	4
Cementite	4	1	5	41

Table 6 Classification of validation samples, excluding cementite

	Pearlite	Ferrite	Martensite
Pearlite	50	0	0
Ferrite	0	48	1
Martensite	3	1	45

Conclusions

The main aim of this research was investigate the effectiveness of training a radial basis neural network to recognize these steel phases: pearlite, ferrite, martensite, and cementite. The implementation of digital image processing techniques in metallography has been in development for two decades. This work represents a new area for research which will help in automatic recognition of phases present in metals.

Using six features (entropy, energy, contrast homogeneity, number of peaks in the histogram, and percentage of black pixels in the binary version of the image) and one output for each of the phases listed above, the root mean square error for the network was 22.4 %. When cementite was not included in the output, thus reducing the output to pearlite, ferrite, and martensite, the error reduced to 16.2 %.

This is comparable to the work of Sing concerning ore sorting and ore classification, which demonstrated an error of 11.29 %, which is a very different application but based on very similar principles of texture and radial basis function networks [7].

In terms of image based texture, it was discovered that cementite and martensite had nearly the same texture. It would be difficult for any neural network to differentiate between them based on texture alone without additional information such as the percentage of binarized black pixels and the number of significant peaks in the image histogram. The possibility of error would be more if number of features were less than six.

This work demonstrated that pearlite and ferrite can be easily differentiated to a high level of accuracy using only the six features already discussed.

Additional features, such as the power spectrum or additional histogram data, would further improve the accuracy of this methodology.

The image identification methodology presented in this research can be expanded for use in automated quality control in steel manufacturing plants. It can be also adapted for analyzing composite materials, including fiber-reinforced composites and nano-reinforced composites, by recognizing different constituents present. It can also be used for identifying materials present in concrete. In terms of scale, the concepts of texture analysis combined with neural networks can also be applied to images captured using a scanning electron microscope. The authors feel that there is definite potential with continued research.

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Requirement and Interaction Analysis Using Aspect-Oriented Modeling

Sagar Mohite, Rashmi Phalnikar, and S.D. Joshi

Abstract
Aspect-oriented modeling (AOM) has been developed to modularize crosscutting concerns appropriately in UML models. In software engineering, aspects are concerns that cut across multiple modules. In requirements modeling, we analyze interactions and potential inconsistencies. We use UML to model requirements in a use case driven approach. During requirements specification a structural model of the problem domain is captured with a class diagram. Use cases refined by activities are the join points to compose crosscutting concerns. Graph transformation systems provide analysis support for detecting potential conflicts and dependencies between rule-based transformations.

Keywords
Aspect-oriented modeling • Rule-based graph transformations • Aspect • Pointcuts • Crosscutting concerns

Introduction

Aspect-oriented represents aspects during requirements engineering. One of the advantages of aspect-oriented approaches is that they allow software developers to react easily to unanticipated changes in existing software systems, while promoting reusability of already tested and designed software components. Nevertheless, people are reluctant to apply AOP in serious and large projects, not because of a lack of good aspect-oriented programming languages and tools, but because they do not have aspect-oriented modeling and design techniques at their disposal. Aspect-orientation should take into account in all the stages of the software lifecycle, in particular, the design level. Aspect -orientation clearly separate crosscutting concerns from non-crosscutting ones which provide modularization. Separation of concerns will reduce complexity of software design [1]. Aspect-orientation originally has applied at programming level; it now applied over other development phases. Aspect is quite important in software development. Aspects are identifying by analyzing a complex system from multiple viewpoints [2]. Aspect-oriented modeling covers many activities at early stages of the software development.

Related work

When designing software, it is desirable to explore several different designs, i.e. consider several feasible solutions to implement a specific requirement or functionality and compare the advantages and disadvantages of each solution. Unfortunately, software modeling tools are often tedious to use when making significant changes within the design of a large software model.

Aspect-oriented modeling (AOM) is a new modeling technique that allows developers to describe the design of their software using many aspect models. In AOM, each individual aspect model is small in size. To build larger systems, structure and behavior defined in one aspect

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_54, © Springer International Publishing Switzerland 2015

model can be reused within other aspect models. This reuse is achieved by establishing a mapping between the model elements in the two aspect models. OOP already allows for modularizing concerns into distinct methods, classes and packages. However, some concerns are difficult to place as they cross the boundaries of classes and even packages. Security and logging, response time are examples for cross-cutting concern.

Disadvantage on Object Oriented Programming:-

UML, in its current state, allows us to capture the structure and interactions of our aspect-oriented program. However, the resulting model presents some major drawbacks:

- There is no difference between modularization by class and by aspect. The basic concepts of AspectJ, such as point-cuts, introduction and advice, are not explicitly modeled.
- The model does not show that the aspect is a “pluggable” entity. The diagrams give the impression that aspects are static entities, although, in reality, aspects are configured at weave-time, and triggering them can be based on various kinds of execution flows or conditions.
- The model does not provide way to find and remove conflict and dependency in system.
- The requirement, it can be functional or non functional are not properly analyze and manage.

It can naturally capture the functional and structural description of each aspect. An aspect may share the base domain model or add its own concepts. Each aspect can be analyzed for consistency, and the consistency of the entire system consisting of the base and aspects can be analyzed as well. Analysis is even more crucial for aspect-oriented models:

As an aspect is specified once but can be used in many different places of the system.

As an example, consider a simple Heath care air ambulance system. Patient can cancel booking of air ambulance. Patient can registered in the system and can be unregistered as well. Both are conducted by admin on behalf of patient. Also, air ambulance and flight segments are administered by staff members. When patient books air ambulance and if he wants to cancel the booking, then the booking are cancel. But after some time, he wanted to again book for the same air ambulance, then it is not possible because entire data has been deleted during cancellation process. Here the conflict will be created.

The domain classes are given in Fig. 1. A patient can book a book air ambulance.

When doctor and specialist generate the report of patient using machine or other test, in earlier Stage, they find the causes of some particular dieses and generate report for that, but later stage by using test report and medicine report, they find other dieses than earlier one. This could be one type of conflict

Approach

Requirement Specification

In an object-oriented requirements specification the main usage scenarios are captured with use cases. Each such use case has to be described in more detail, either textually or by means of activity diagrams. Also, during requirements specification a structural model of the problem domain is captured with a class diagram. A use case diagram provides a system overview. Each use case is described by trigger, its actors, pre- and post-conditions and its key scenarios. Scenarios are specified using activity diagrams and use cases are the starting point for the aspect-oriented modeling. We model the so-called base of the system with use cases and an integrated behavior model. An aspect is modeled as a use case. The join point for an aspect is an activity of the base. The point cut of an aspect is specified in terms of the activities of the base. While up to now proposed for modeling techniques like UML, an integrated behavior model is also suitable and beneficial for aspect- oriented modeling:

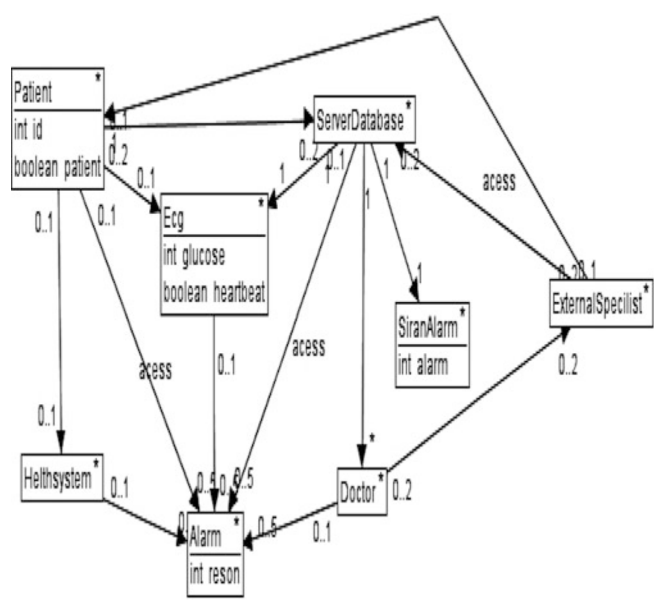

Fig. 1 Type graph for health care air ambulance system

Interaction Analysis Using Aspect Oriented Model

An overview of typical use cases is presented in Fig. 2.

It comprises use cases for administration of passenger data of patient, air ambulance and segment data. For the main usage scenarios it has use cases for booking and canceling air ambulance. In the following, we will focus on the use cases book air ambulance.

The steps, pre- and post- conditions of use case book air ambulance are described in Table 1.

The steps are also presented in the activity diagram in Fig. 3. Our approach uses integrated behavior models and extends them by aspect-oriented features. An integrated behavior model consists of a domain model and a set of activity models. The domain model provides the types of the domain objects.

Each activity is refined by pre- and post-conditions describing the effect of the activity in terms of domain objects. Typically, an initial configuration of the system is provided in terms of domain objects and their relations. The benefit of an integrated behavior model is an early and better integration of the structural domain model with the functional activity model. Pre- and post-conditions are formalized by the theory of graph transformation systems (Fig. 4).

It is conceivable, that the use case Finding Report is expressed as rules such as in Table 2, and Test crisis Management is express as rule in Table 3 (Figs. 5, 6, 7, 8, and 9).

Implementation

We are using AGG tool for implementing Heath care air ambulance system. With this tool we can perform conflict analysis and graph transformation. For graph transformation AGG tool is used which is a rule based visual language. A graph grammar contain in AGG program attributed by Java objects. Graph grammars contain a start graph and a set of rules which may have negative application conditions. A graph consists of two disjoint sets containing the nodes and the arcs of the graph. As a whole, the nodes and arcs are called the objects of the graph. Every arc represents a

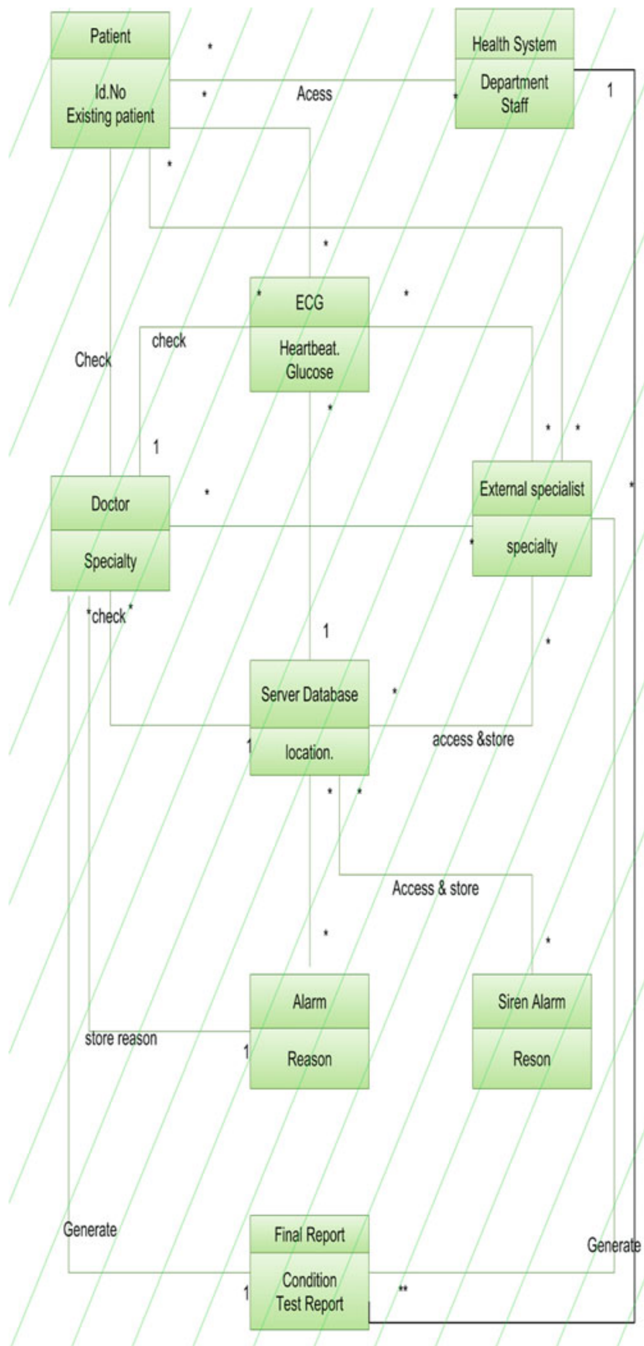

Fig. 2 Class diagram for health care air ambulance system

directed connection between two nodes, which are called the source and target nodes of the arc.

In AGG attribute declaration is same as other programming language. An action can be viewed as a state transition, and obviously, a transition of states can be specified by giving descriptions of the states before an after the action in question. Since states are modeled as graphs in AGG, it follows that basically an action can be described as a pair of two graphs modeling the "before" and "after" states. In the "before" state of an operation, we collect all the preconditions that have to be met for the operation to take place. The left-hand side of a graph rule states the necessary conditions for the specified operation to take place: A rule can only be applied if its conditions are fulfilled by the current concrete state graph. The effect of a rule application at a given match is a state graph transformation, also called derivation or graph transformation step.

Table 1 Description of use case book air ambulance

Use case	Book air ambulance
Actor role	Admin
Trigger [condition]	Patient orders booking
Rule of precondition	Air ambulance exits
Rule of post condition	Each segment of the flight is booked
Use-case scenario	1. Select air ambulance 2. Select patient 3. Reserve segments 4. Book segments 5. Pay

Conclusion

In aspect oriented modeling, the program with the main business logic and the cross cutting concerns are represented by models. There are many cross cutting concerns that are not part of the problem, such as security, logging, persistency, etc. Differently from traditional aspect oriented programming, in aspect oriented modeling there is no preferential entity. This means the weaving can be done in any model (e.g., weavings in the business models or in the cross cutting concerns).

Our approach is towards detecting conflicts and dependencies. The tool was used for specifying the behavior of aspects and objects in terms of pre- and post-conditions and for analyzing conflicts and dependencies between them. The tool computed the necessary input in form of conflicts and dependencies which were then compared with the specified composition.

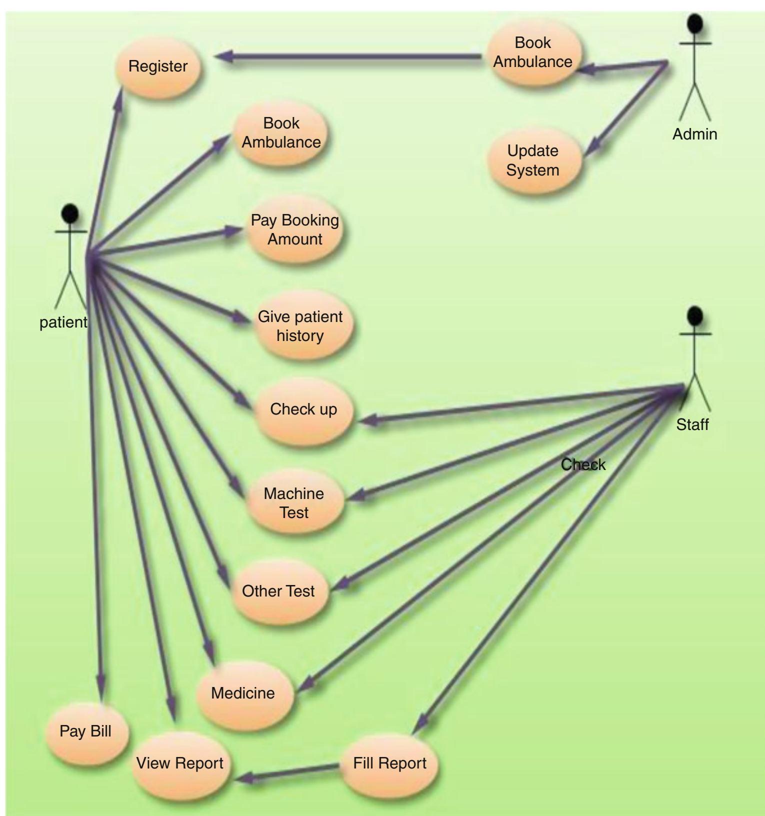

Fig. 3 Use case diagram for health care air ambulance system

Fig. 4 Activity diagram for health care air ambulance system

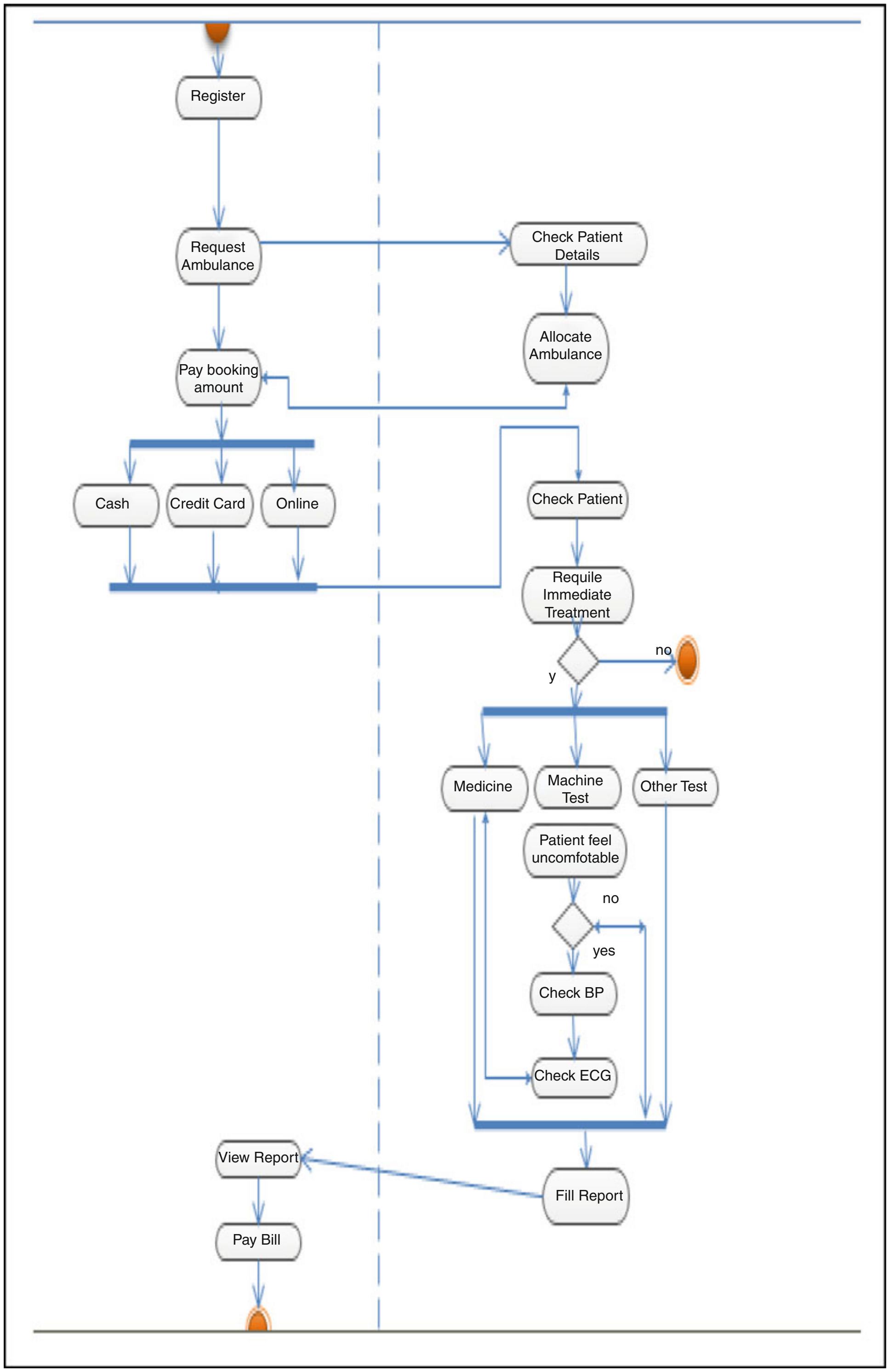

Table 2 Finding report store given as rules

Finding report store scheme ( )
1) Only authorized staff i.e. doctor, nurse, staff etc can access the report of patient
2) Doctor checks the patient, if patient condition normal or abnormal
3) If patient condition is abnormal then store a patient report with time and report—name with a different server database on internet
4) If patient condition is normal then store a patient report to a local server on a system

Table 3 Test crisis management given as rules

Apply test crisis management ( )
Only doctor suggest medical test or other test to a patient
Doctor analyzes the heath condition of patient during test conduction
Patient feel normal or abnormal find in primary analysis
If patient feel abnormal, then conduct medical test with crises management system to a test
If patient feel normal, then conduct medical test without crises management system to a test

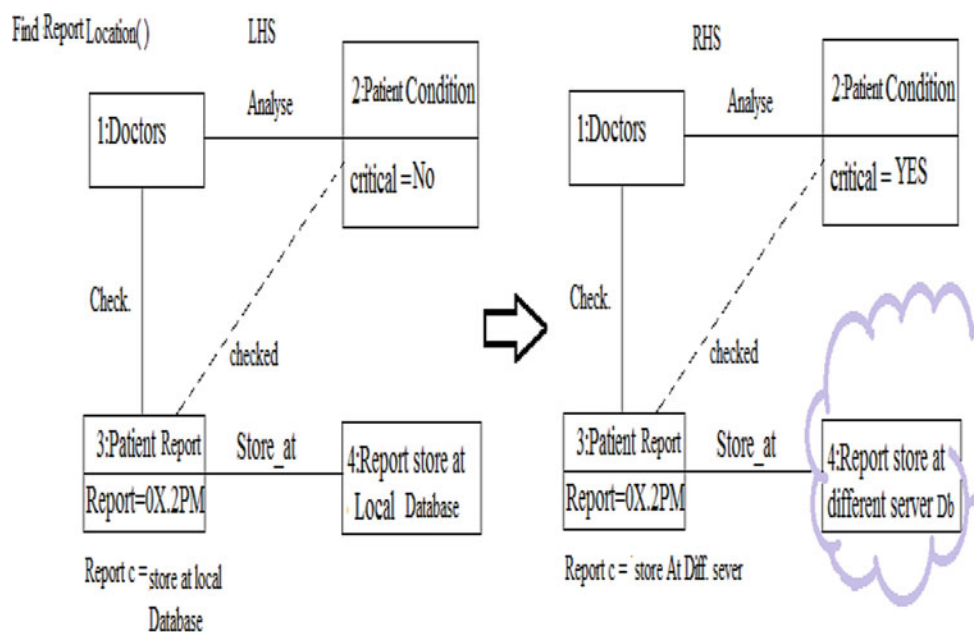

Fig. 5 Rule for report store

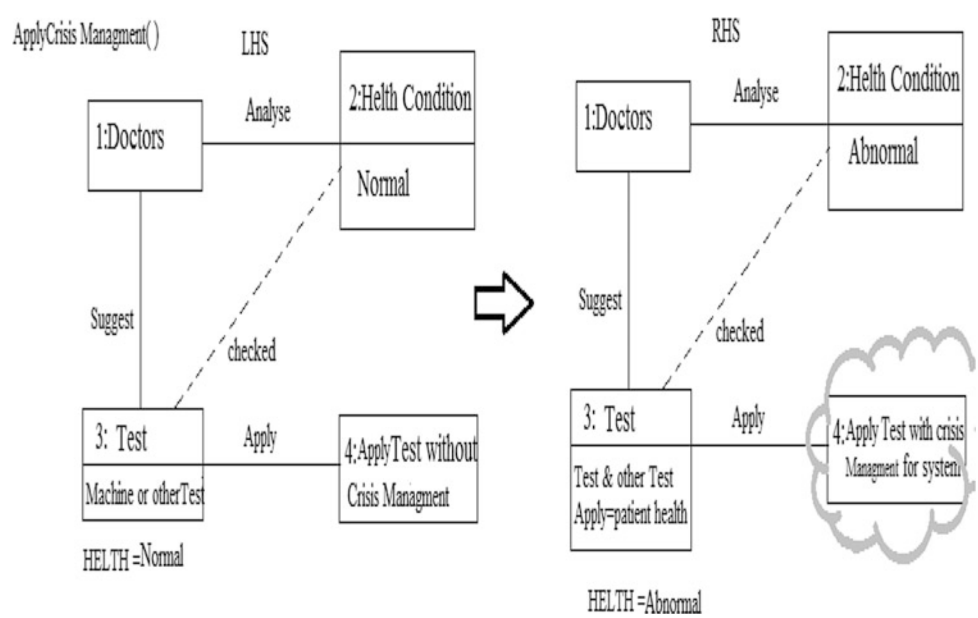

Fig. 6 Rule for crises management for test

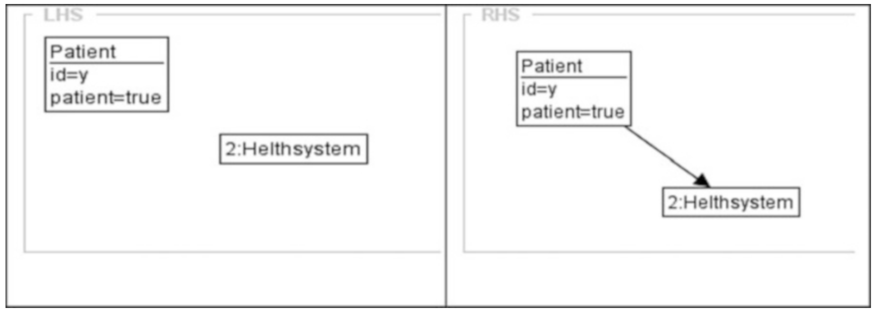

Fig. 7 Rule for patient login

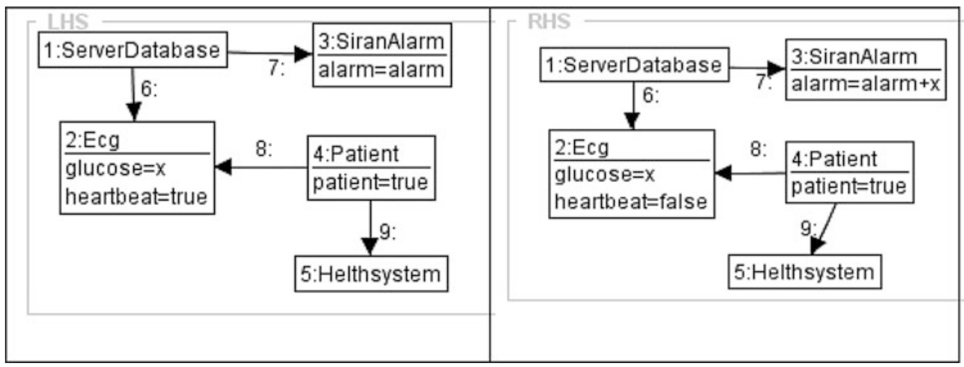

Fig. 8 Rule for siren alarm

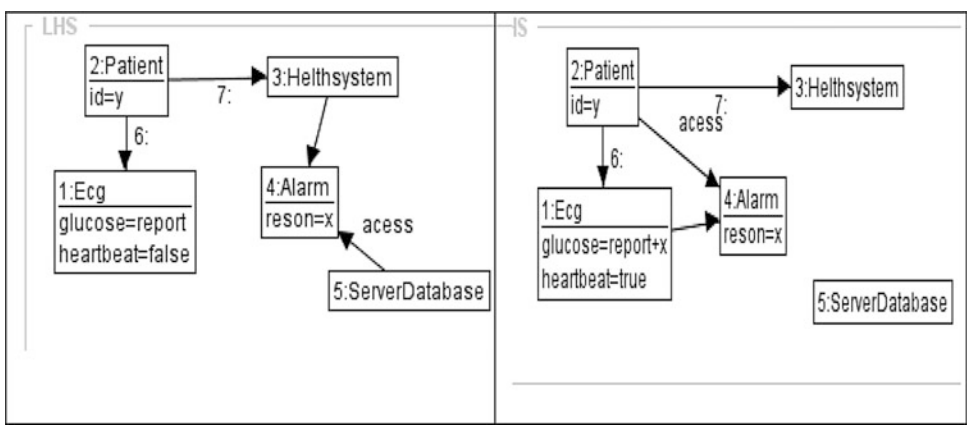

Fig. 9 Rule for alarm reason store in severs

Found conflicts and dependencies are potential conflicts (since they can be interactions) and not every possible conflict since it depends on the accurateness of the pre- and post-conditions. Graph transformation also allow to reason uniformly about object and aspect models [3]. Besides an editor for specifying the rules, the tool also provides all analysis functions as an API. Rules can be read from an XML file. Therefore, AGG is ideal to be used with existing UML CASE Tools [3].

We have proposed to integrate rule specifications with object oriented models in an aspect-oriented way. Graph transformation systems provide means to specify rule-based aspects directly as rules. We have tried to detect conflicts and dependencies. The approach uses formal aid to analyze systematically semi-formal specifications. We use graph transformation to detect conflicts.

In the future, we want to investigate the relationship between graph transformation and aspect-oriented languages further. We feel that pre- and post-conditions are an essential counterpart for an informal language like the UML, making modeling more rigorous. Pre- and post-conditions are analyze for activity diagrams. The approach is not restricted to functional aspects, as presented here. We consider both functional and nonfunctional aspect. Thereby, also interactions between functional and non-functional aspects are automatically covered.

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AGG tool avialble at: http://user.cs.tu-berlin.de/~gragra/agg/index.html

Fingerprint Orientation Image Estimation in the Frequency Domain

Victor-Valeriu Patriciu, Stelian Spînu, and Cezar Pleşca

Abstract

The orientation image estimation is a critical step in fingerprint image processing and feature extraction. The computation of local ridge orientation in low quality fingerprints is still a difficult task. A simple and reliable method for estimating the fingerprint orientation image in the frequency domain is proposed. The results of the experiments conducted on a collection of fingerprints, in order to evaluate the performance of the proposed method, are also presented.

Keywords

Biometrics • Feature extraction • Fingerprint recognition • Spectral analysis

Introduction

The need for reliable person identification is ever growing in today's globally interconnected information society [1].

The use of fingerprints as a biometric identifier is both the oldest mode of automated person identification and the most widely used today [2]. The deployment of fingerprint recognition technology in a wide range of government and commercial applications has been favored by a series of factors. These factors include: the well-known individuality and permanence properties of fingerprints, the enormous success of fingerprint identification in law enforcement applications, the existence of a large market of small and inexpensive fingerprint sensing devices, and the increasing availability of affordable computing power [3].

A fingerprint consists of a pattern of interleaved ridges and valleys. The local ridge orientation at pixel (i, j) in a fingerprint image is the angle $\theta_{ij} \in [0, 180\,°)$ that the fingerprint ridges, crossing through an arbitrary small neighborhood centered at (i, j), form with the horizontal axis [3]. By estimating the local ridge orientation at discrete positions, a fingerprint orientation image is obtained.

The ridge orientation estimation in fingerprint image processing represents a very important step. The computation of local ridge orientation is required in: fingerprint image enhancement [4–9], fingerprint classification [10, 11], feature extraction from a dermatoglyphic point of view [12], and fingerprint image segmentation [13–15].

The rest of the paper is organized as follows. In Section "Previous Work", a review of previous approaches for estimating the fingerprint orientation image is presented. Section "Proposed Approach" describes the method proposed for extracting the local ridge orientation in fingerprint images in the frequency domain. The results of the experiments conducted on a set of fingerprints, in order to evaluate the proposed method, are presented in Section "Experimental Results". Section "Conclusion" concludes the paper.

Previous Work

The approaches proposed in the literature for the computation of fingerprint orientation use image processing in both the spatial and frequency domains.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_55, © Springer International Publishing Switzerland 2015

Ridge Orientation Estimation in the Spatial Domain

Sherlock et al. [4] determine local ridge orientation at a square grid spaced 16 pixels apart. A window of size 32 × 32 pixels is centered at the pixel of interest and rotated to 16 different orientations. At each orientation, a projection along the y-axis of the window is formed and the noise from the projections is removed with a second-order Butterworth band-pass filter. When the window is aligned with its x-axis orthogonal to the ridges, we expect maximum variation of the projection. Conversely, alignment of the x-axis along the ridges should give minimum variation.

The computation of gradients in the fingerprint image is the most widely used approach for extracting local ridge orientation. Ratha et al. [13] divide the input fingerprint image into blocks of size 17 × 17 pixels and compute the gradients at each pixel in the block. Let $G_x(i, j)$ and $G_y(i, j)$ be the gradient magnitudes in the x and y directions, respectively, at pixel (i, j), obtained using the classical Sobel masks. The dominant ridge orientation in a block, centered at pixel (i, j), can be computed using the following equation:

$$\theta_{ij} = \frac{\pi}{2} + \frac{1}{2}\tan^{-1}\frac{\sum_{h=-8}^{8}\sum_{k=-8}^{8} 2G_x(i+h,\ j+k)\,G_y(i+h,\ j+k)}{\sum_{h=-8}^{8}\sum_{k=-8}^{8}\left[G_x(i+h,\ j+k)^2 - G_y(i+h,\ j+k)^2\right]}. \tag{1}$$

Bazen and Gerez [16] demonstrate that the gradient-based method is mathematically equivalent to the principal component analysis of the autocovariance matrix of the gradient vectors.

The orientation image of a good quality fingerprint image can be reasonably obtained with the gradient-based method. Nevertheless, the presence of high-curvature ridges and noise leads to a poor estimate of the local ridge orientation. To overcome this limitation, Jain et al. [17] employ a multi-resolution approach resulting in a fairly smooth orientation image.

Karu and Jain [10] compute the gray-value sum of pixels along a number of eight reference orientations. The local ridge orientation corresponds to the minimum-sum slit for pixels lying on ridges and to the maximum-sum slit for pixels lying on valleys.

Based on the observation that the total variation of the gray levels is minimum along the ridge orientation and maximum in the orthogonal direction, analogous approaches are proposed by He et al. [18] and Oliveira and Leite [19].

Ridge Orientation Estimation in the Frequency Domain

Based on the intrinsic characteristics of the ridge pattern in fingerprint images, Kamei [5, 8] proposes a filter designed in the Fourier domain, having two distinct components, a frequency filter corresponding to ridge frequency and a directional filter corresponding to ridge orientation. The local ridge orientation is selected not only according to the maximum filter response, but also taking local smoothness in the fingerprint into consideration.

Park and Park [11] compute the fingerprint orientation image at a square grid spaced 16 pixels apart. The Fast Fourier Transform (FFT) is applied on the blocks of size 32 × 32 pixels centered at each inner grid point. In order to detect the dominant ridge orientation in each block, a directional filter D_θ is used in the FFT image:

$$D_\theta(i+p,\ j+p) = \begin{cases} \exp\left(-\frac{u^2}{a^2} - \frac{v^2}{b^2}\right), & \text{if } \frac{u^2}{a^2} + \frac{v^2}{b^2} \le 1, \\ 0, & \text{otherwise,} \end{cases} \tag{2}$$

for $-p \le i, j \le p$, where $v = i\cos\theta + j\sin\theta$, $u = \sqrt{i^2 + j^2 - v^2}$, $p = 16$, $a = 1.5$, $b = 32/3$, and $\theta = 0, 10, \ldots, 170\,°$.

The local ridge orientation in each block is perpendicular to orientation θ that maximizes the sum of FFT coefficients weighted by directional filter D_θ:

$$f(\theta) = \sum_{i=0}^{2p}\sum_{j=0}^{2p} D_\theta(i,\ j) \times |F(i,\ j)|, \qquad \theta_{\max} = \arg\max_\theta f(\theta) \tag{3}$$

The algorithm proposed by Chikkerur et al. [9] is based on short time Fourier transform analysis. The fingerprint image is divided into partially overlapping, raised cosine windows. The Fourier spectrum of each window is analyzed and a probabilistic estimate of the local ridge orientation is obtained.

Proposed Approach

The pattern of the ridge-valley structure in fingerprint images can be locally modeled as a sinusoidal-shaped surface. The Fourier spectrum of this surface consists of two peaks, symmetrical about the origin. The direction perpendicular to the line connecting these peaks indicates the local ridge orientation [8].

In our approach, fingerprint orientation image is estimated at a square grid spaced 16 pixels apart. For each window of size 32 × 32 pixels, centered at each grid point, the largest spectral component in the range of valid ridge frequencies is determined. The location of this maximum frequency component is used to estimate local ridge orientation and its magnitude to segment the fingerprint area from the image background.

The fast Fourier transform is applied on each block of size 32 × 32 pixels centered at each grid point. The maximum value of the centered Fourier spectrum is searched in the upper half of the frequency rectangle, in the region of valid ridge frequencies, i.e., the points (u, v) satisfying the double inequality: $f_{\min} \leq \frac{\sqrt{(u-N/2)^2+(v-N/2)^2}}{N} \leq f_{\max}$, for $0 \leq u \leq N/2$ and $0 \leq v \leq N-1$, where $N = 32$, $f_{\min} = 1/15$, and $f_{\max} = 1/5$, for 500 dpi images.

Let u_m and v_m be the coordinates of the maximum value of the Fourier spectrum, $|F(u_m, v_m)|$. These are separately adjusted for each coordinate axis by taking into account the neighbor in the Fourier spectrum with the largest value.

Therefore:

$$u_m' = \begin{cases} u_m - \dfrac{|F(u_m, v_m-1)|}{|F(u_m, v_m-1)| + |F(u_m, v_m)|}, & \text{if } |F(u_m, v_m-1)| > |F(u_m, v_m+1)|, \\ u_m + \dfrac{|F(u_m, v_m+1)|}{|F(u_m, v_m+1)| + |F(u_m, v_m)|}, & \text{otherwise,} \end{cases} \tag{4}$$

and

$$v_m' = \begin{cases} v_m - \dfrac{|F(u_m-1, v_m)|}{|F(u_m-1, v_m)| + |F(u_m, v_m)|}, & \text{if } |F(u_m-1, v_m)| > |F(u_m+1, v_m)|, \\ v_m + \dfrac{|F(u_m+1, v_m)|}{|F(u_m+1, v_m)| + |F(u_m, v_m)|}, & \text{otherwise.} \end{cases} \tag{5}$$

The local ridge orientation at point (i, j) is computed as:

$$\theta_{ij} = \frac{\pi}{2} + \tan^{-1}\frac{u_m' - N/2}{v_m' - N/2} \tag{6}$$

The values of the local ridge orientation obtained with (6) are considered only for the fingerprint ridge area. In order to segment the fingerprint area from the image background, we applied the Otsu's method [20] to threshold the magnitudes of the previously determined spectral components. Morphological closing and opening are further performed on the resulted binary image, in order to eliminate holes and smooth the region of interest for the local ridge orientation estimation.

Experimental Results

Two previous approaches for estimating local ridge orientation (the ones presented in [11, 13]) and the proposed method were implemented in MATLAB®. Experiments were conducted on set B of FVC2004 databases [21] (a total of 320 fingerprint images) in order to compare the three methods. The set of fingerprints includes four different databases (DB1, DB2, DB3, and DB4), collected by using different sensors/technologies (see Table 1).

Figure 1 shows sample images from each of the four databases DB1, DB2, DB3, and DB4.

As indicated in [16], there exists no ground truth for the orientation image of fingerprints. Consequently, it is difficult to quantitatively evaluate the quality of an orientation image estimate. We solved this problem using the same approach as described in [22]. A reliable orientation image can be computed for each fingerprint by using a Gabor filter-bank [23], providing that the number of filters is large enough. We employed a bank of 100 Gabor filters (25 discrete orientations and 4 discrete frequencies).

The orientation images obtained by each of the three methods, the previous approaches presented in [11, 13] and the proposed method, were compared with the orientation image extracted by using the Gabor filter-bank.

Table 1 FVC2004 databases [21]

	Sensor type	Image size	Resolution
DB1	Optical sensor "V300" by CrossMatch	640 × 480 (307 kpixels)	500 dpi
DB2	Optical sensor "U.are.U 4000" by Digital Persona	328 × 364 (119 kpixels)	500 dpi
DB3	Thermal sweeping sensor "FingerChip FCD4B14CB" by Atmel	300 × 480 (144 kpixels)	512 dpi
DB4	Synthetic fingerprint generation SFinGe v3.0	288 × 384 (108 kpixels)	About 500 dpi

Fig. 1 Sample images from each of the four databases DB1, DB2, DB3, and DB4 [21]

Table 2 The average orientation estimation errors

	DB1	DB2	DB3	DB4
Method in [13]	5.24	5.85	5.59	4.78
Method in [11]	5.27	6.41	5.12	4.98
Proposed method	4.91	5.54	4.90	4.34

Nevertheless, there should be noted that the computation of fingerprint orientation image by using a Gabor filter-bank is too computationally expensive and, therefore, not suitable for real-time applications [22].

For each fingerprint, the mean absolute error between the orientation estimate of each of the three approaches and the orientation estimate obtained by using the Gabor filter-bank was computed. The averages of the mean absolute errors of the orientation estimates obtained by using the three methods, for each of the four databases, are presented in Table 2. The results show a decrease of the average error of the fingerprint orientation estimates obtained by using the proposed approach, for all of the four databases.

Figure 2 shows samples of orientation images obtained with the proposed method for each of the four fingerprint databases DB1, DB2, DB3, and DB4. The region of interest is also displayed.

The average processing times for estimating the local ridge orientation are presented in Table 3. These values were obtained using an Intel® Core™2 Quad Processor at 2.4 GHz.

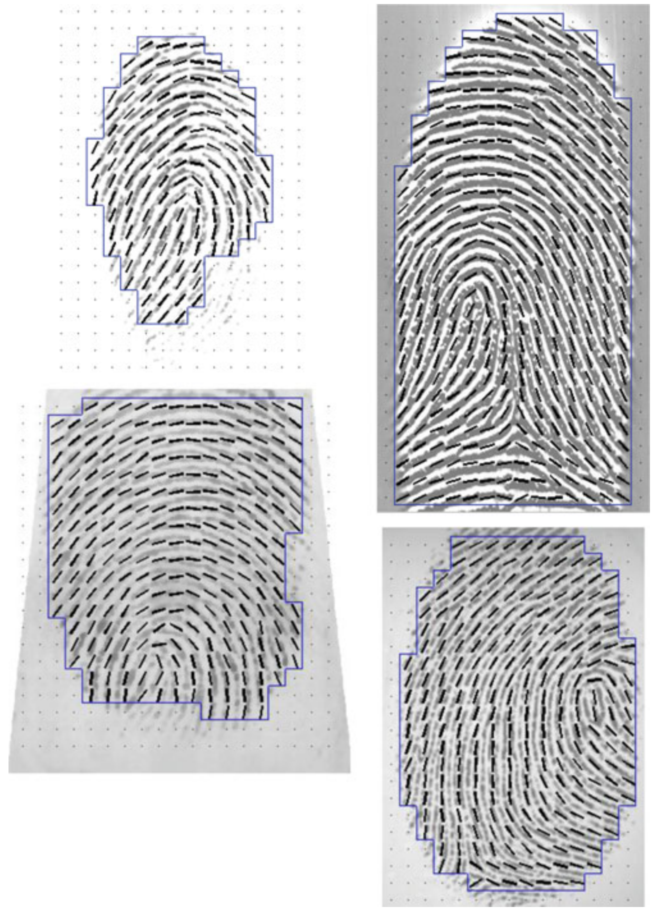

Fig. 2 Samples of orientation images obtained with the proposed method for each of the four fingerprint databases DB1, DB2, DB3, and DB4. The region of interest is also displayed

Table 3 The average processing times

Method in [13]	Method in [11]	Proposed method
54 ms	330 ms	162 ms

Conclusion

The computation of local ridge orientation is an important step in fingerprint image processing and feature extraction. A reliable orientation estimate is required in fingerprint enhancement, feature extraction, classification, and segmentation. The orientation image extraction in fingerprint areas of low quality is still a difficult task and most approaches to the computation of local ridge orientation result in poor estimates.

A simple and accurate method for fingerprint orientation image estimation in the frequency domain was proposed. The results of the experiments conducted on a collection of fingerprints show a better performance of the proposed method compared to previous approaches. The proposed method is robust with respect to the noise specific to low quality images and reliable fingerprint orientation images are obtained. Our future work will apply the proposed method for fingerprint image enhancement.

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Relationship Between Affordance and Cultural Conventions in the Design of IVR Systems for Oral Users

Tembalethu J. Ndwe and Nomusa Dlodlo

Abstract

This paper addresses the significance of affordance in association with cultural conventions in the design of voice user interfaces (VUI) for orally grounded users in the global South (i.e., developing countries of the world). The paper demonstrates that sensitivity towards cultural conventions which subsequently bring about affordance has more credence than the objective usability measures of effectiveness and efficiency as defined by the International Standards Organization (ISO). This demonstration is done with the aid of two case studies of Interactive Voice Response (IVR) systems that were developed for users in developing countries of Southern Africa. The paper specifically presents the concept of taking into consideration the shift between the standard cultural norm and the situational norm experienced during the usage of the IVR system. We have established that orally grounded technology users prefer a VUI that allows them to transfer easily from their standard cultural norms to the situational norm that is determined by the context of use of the presented technology.

Keywords

Cultural conventions • IVR systems • Oral users • Affordance • Usability

Introduction

It is anticipated and expected that speech technology, through the use of the normal telephone, will have a substantial impact in the realization information access and delivery of information services in the developing nations of the world [2]. This is due to the fact that cellular technology penetration is growing more rapidly in the developing countries even more than the developed world [10, 3]. Another important reason to take into account is the fact that the majority of the people living in the developing world have a strong oral tradition with high levels of illiteracy. This phenomenon is even more important to consider within the realm of the region of Southern Africa where a strong oral tradition exists amongst a large low literacy population [1].

Speech technology is a feasible channel for information access in low-literacy individuals since listening and talking do not require literacy. In this paper we support the use of speech technology as a means of information access and present an essential attribute that needs to be incorporated in the design of speech technology for orally grounded users in the developing regions of Southern Africa.

The present research study utilizes two case studies of the development of two IVR systems namely the OpenPhone and the Beautiful Game Results (BGR) systems. The OpenPhone system is an IVR system that allows its users to access information on caregiving for HIV/AIDS infected children in Botswana, Southern Africa. The BGR system is a fun system that allows its users to access the results of recently played soccer matches in the Professional Soccer

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DOI 10.1007/978-3-319-06764-3_56, © Springer International Publishing Switzerland 2015

League (PSL) of the Republic of South Africa. The OpenPhone users are characterized as ranging from illiterate to semi-literate users who are mostly female and mothers of the infected children. The BGR users are soccer fans who are mostly male and range from illiterate to literate users. The intended users for both IVR systems are nevertheless numerically literate and are accustomed to the use of a telephone. The two IVR systems are contrasting in their manner of use as the OpenPhone system has a serious context about a disease that is pandemic in the region of Southern Africa and either infects or affects everybody in the region. On the contrary, the BGR system is a cheerful application that allows its users to interact with a telephonic access to results of soccer games in the Republic of South Africa, where soccer is the favorite sport [16]. The contrast between the two IVR systems has allowed the researchers to discover oral user attitudes and characteristics that would not have been possible with applications that have a similar context of use. We, the researchers, have achieved the contrast and comparison between the two IVR systems by running usability experiments of comparing two modalities of interaction that were presented to recruited intended users for each of the systems.

The two modalities are the speech-enabled modality and the Dual Tone Multi Frequency (DTMF) which are different modes of interaction with an IVR system. In the speech-enabled interaction mode the users interact by giving verbal commands to the system and the system responds by giving back verbal information. The speech-enabled mode is similar to a normal conversation to another person. On the other hand, the DTMF system allows the users to interact with the system by using the telephone keypad as the sole input device. The DTMF system output presents a menu that instructs the user to press a particular number on the phone keypad that corresponds to a particular effect.

For example: *System: To learn about common sicknesses press 4, to learn about ARV medication press 5.*

The user then reacts by pressing whichever number that corresponds to the task that they want to carry out.

The researchers are not aware of any other research or publication that investigates the relationships between affordances and cultural conventions within the usage of IVR systems in developing countries.

Affordance and Cultural Conventions

The term of affordance was first coined by Gibson who defined the concept of affordance as:

> *An actionable attribute of an interaction design between the world and an actor* (actor refers to a person or an animal) [7].

Gibson's view on affordance implies a relationship between the actor and the attribute of an interaction design and affordance exists naturally and does not have to be seen, known or desirable. Cultural conventions are constraints or regulations shared by a cultural group [13]. The concept of cultural conventions is demonstrated vividly through the contrast in the two case study IVR systems that are presented in this research as follows:

In operation, the DTMF modality necessitates the user to interact by pressing buttons on the phone. The interaction is muted to a nearby person and therefore, except for the user, no other person would be able to comprehend the content and the context of the interaction between the user and the system. Conversely, in the speech-enabled modality, the interaction necessitates the user to respond by using their voice as in a normal conversation with another human being. This means that a close by person would be able to overhear the content of the interaction, whether unintentionally or otherwise. Other people who have used the same system would certainly know both the content and the context of the interaction as well.

Within the culture of the indigenous people of Botswana, speaking out loudly on matters that are affiliated with sex is an inappropriate practice. It is well known that HIV/AIDS is an illness that is mostly spread through sex, and therefore speaking out loud about HIV/AIDS is not common unless it is through educational media. The stigma about HIV/AIDS is a complex situation that is interlinked with other stigmas including ethnicity, religion, gender and others [15, 4]. Stigma is considered as one of the main impediments in controlling the disease. It is important to note that almost all of the OpenPhone users are women who are mothers and caregivers to the HIV/AIDS infected children, and subsequently they are HIV/AIDS infected themselves. Being an HIV infected woman, within the socio-cultural ideals of the targeted users bears even more stigmatization, suffering in silence and shame [6]. The targeted users' socio-cultural ideals in combination with the lack of seclusion in the modus operandi of the speech-enabled modality support the postulation that the caregivers' choice of the DTMF modality is based on the privacy of the modality when compared to its counterpart. The researchers view the privacy issue as the most compelling reason behind the users' choice.

An inherent problem with the stigma of HIV/AIDS is that the users would largely not say that the stigma is the reason for their choice of interaction modality. This is partly because from the lectures that they attend and from the media of radio and television the effects and negative consequences of stigma have been addressed over and over, and caregivers know very well that they are not supposed to yield to stigma. Nevertheless, the HIV/AIDS stigma continues to be a problem in their communities and by nature of stigma, it is something no one wants to talk about or be associated with. This discourages the caregivers in revealing stigma as an influence to their choice of interaction modality. In all the 22 participants who performed both modalities and were asked to compare the 2 modalities, only one participant mentioned that they prefer the DTMF because of its confidentiality, stating that in the speech-enabled system other people can hear what the user says. Interestingly, this particular user got her DTMF tasks incorrect and got her speech-enabled tasks correct during the usability tests and still chose DTMF as her modality of choice.

In contrast to the OpenPhone system's access to a stigmatized issue, the BGR provides access to an issue that is not only free of stigma but something that has a culture of being publicly debated and spoken about loudly in public places, private homes, work, and generally everywhere. It is a normal practice to find soccer fans gathered at work, in public places or over a phone call and debating issues about recent games and such conversations are typically accompanied with exuberance and loud speaking and laughter of fans teasing the supporters of teams that recently lost a game. Soccer fans love to speak out loudly about their favorite teams and this loud and exuberant expression of love for soccer is part of South African soccer culture and it is compatible with the oral tradition of the indigenous people of the country as it promotes expression of opinions and chatting freely on a subject that is well-known and loved by the soccer fans. The exuberance of speaking loudly on soccer issues was evidenced on the BGR whereby participants were speaking loud and fast as they gave commands to the system, something they were not requested to do, as they were only requested to speak clearly. This was in direct contrast to the reserved, monotonous and slow style of interaction that was experienced with the OpenPhone participants. According to studies that were conducted in human emotional sensitivity, loud and fast voice was found to correlate to happiness, joy and confidence whilst a low and slow voice was linked to boredom, grief and sadness [17, 5]. The researchers believe that the reason for the overwhelming choice of the speech-enabled modality in BGR is because it allows the users to interact with the system in the same manner that they normally address soccer issues which is, simply, loud. Everything about the game in the region is loud; from the vociferous vuvuzela—a plastic horn played in and outside soccer matches that produces a loud monotone note, to the colorful apparel worn by the spectators, inside and outside the stadia. Likewise, the choice of the DTMF system in OpenPhone was due to fact that DTMF allows the users to exercise confidentiality in an interaction with a stigmatized subject matter within a society that imposes even more stigmatization towards infected women as they are looked upon more negatively than infected males.

In summary it can be stated that during the interaction with OpenPhone system, the users of the technology have a situational norm or a cultural convention of quietness and being mute due to the given situation of interacting with a stigmatized subject. Contrastingly, the BGR users have a cultural convention or situational norm of loudness during the interaction with the technology due to the situation of interacting with a cheerful application that is associated with loud situations.

Development Methodology and Experiments

In this section the methodology that was used in the development of the two IVR systems and their associated prototypes (i.e., DTMF and speech-enabled modalities) is discussed together with the measurements that were taken during the usability tests and the experimental results that were obtained. The types of measurements that are relevant to this study are stated and the researchers also explain the motivation behind the measurements used.

Methodology

The methodology of User Centered Design (UCD) was deployed in the development of the two IVR systems. The choice for UCD was motivated by the researchers' desire to engage the intended users of the technology in every phase of the development lifecycle of the IVR prototype systems. UCD is a methodology for designing technology products/services such that they meet usability objectives. In other words, UCD is a set of activities that need to be taken so as to ensure usability in an ICT product/service design/development. The UCD methodology is a participatory design methodology as it involves the anticipated users throughout the development of the technology. One of the essential activities in UCD is usability testing which allows the researchers, in alliance with the intended users, to evaluate the prototype designs against the technology design requirements.

According to the ISO the definition of usability is:

> *Usability is the extent to which a product can be used by specified users to achieve specified goals with effectiveness, efficiency, and satisfaction in a specified context of use* [9].

Additionally the ISO standard describes the implied usability characteristics of effectiveness, efficiency, and satisfaction in Table 1.

The ISO 9241-11 standard definition of usability is becoming the main reference of usability [12]; hence it has been used extensively in this research.

Table 1 Definitions of usability characteristics [9]

Usability characteristic	Definition
Effectiveness	Accuracy and completeness with which users achieve specified goals
Efficiency	Resources expended in relation to the accuracy and completeness with which users achieve goals
Satisfaction	Freedom from discomfort, and positive attitudes towards the use of the product

Usability Experiments Measures

There are typically two types of results that usability experimenters are interested in, which are the qualitative and quantitative results. The objective measurements are quantitative factors that are obtained from empirical measurements and the subjective results are qualitative representations of the intended users' sentiments and satisfaction levels towards the prototypes. According to the ISO definition of usability, as indicated in Table 1, the usability characteristics of effectiveness and efficiency are the quantitative objective measures and the usability characteristic of satisfaction is the qualitative subjective measure of usability. In both IVR systems the subjective results are used and viewed as more dependable than objective results given the exploratory nature of the experiments. Human experience is generally subjective [11], and user subjective reactions to speech interfaces may well be a more important predictor of real world success [8]. Shackel also regards user satisfaction or users' attitude towards the system as the most important aspect of usability [18], and objective measures of system behavior may not suffice in predicting system acceptability [19]. As the study is centered on the first iteration of the IVR systems development, the researchers are more interested in evaluating the technology's feasibility to add value in the lives of the intended users through the measurement of the users' acceptance of the technology since the technology is new to the intended users. The success or failure of the technology depends on the human experience and the study mainly relies on the intended users' view on whether the technology is enough to serve its purpose and meet the user needs and expectations or not. Nevertheless, the study utilizes both the subjective and objective results in unison and regards both as complementary. In the analyses of the results, a good conclusion is considered as one that has support of both types of results.

Experimental Results

As mentioned in the earlier paragraph, this research views subjective results as more prominent that objective results even though reference is made to the qualitative objective results. The following subsections discuss the results that were obtained from the usability tests for both IVR systems.

OpenPhone Subjective Results

After completing both sessions (i.e., one in DTMF and another in speech-enabled mode), participants were asked a few demographic questions which requested their age, level of education, home language, and others. They were also asked for their subjective evaluation of the systems and their choice of interaction modality between the DTMF and the speech-enabled modalities.

Out of the 22 participants who interacted with both OpenPhone prototypes (i.e., the DTMF and the speech-enabled prototypes), 13 preferred DTMF, four preferred the speech-enabled system, and five were equally happy with both systems which produced a 59.1 % preference of DTMF over 18.2 % for speech-enabled system, with 22.7 % of users who were undecided. Two people out of the four who preferred the speech-enabled system said that they actually like both but they were choosing the speech-enabled system because they think that it would be a better system for elderly users who might have difficulty using the DTMF system. Another participant mentioned that she preferred the speech-enabled system because she envisions doing something else with her hands whilst interacting with the system. The participants who did not have any particular preference over any of the systems claimed that the two interfaces were the same to them.

The substantial majority of 13 participants who chose the DTMF system had well-defined reasons including perceived faster speed. The most common reason was the ease of use and the fact that the DTMF was easy to follow because, "it is impossible to get lost as the system [DTMF] just tells you what to do." Another participant remarked that in the speech-enabled system she had to pay more attention because if a command is missed or misunderstood or misinterpreted, then that can be problematic but the DTMF is more "straightforward." Throughout the interactions, the participants were noted to speak in a low, slow paced and monotonous voice and their facial expressions did not show any particular emotion.

BGR Subjective Results

The BGR subjective questions were conducted in the same manner as in the OpenPhone system by asking similar questions and then asking the participants to compare the two modalities after having performed tasks in both modalities.

In total, out of the entire set of 27 participants who participated in the BGR tests, 23 preferred speech-enabled system and four preferred DTMF, which produced an 85.2 % preference of speech-enabled system over 14.8 % for DTMF. The BGR users were all decisive in their preferences and none of them preferred both modalities as in the case of OpenPhone.

The 23 users who preferred speech-enabled system had different reasons which included the fact that it presented a more natural way of interaction through speaking rather than the extra effort of pressing numbers. These users considered the speech-enabled system to be easier to use and they felt that they had more control when giving verbal commands than pressing numbers in the DTMF modality.

The users also perceived the speech-enabled system to be faster even though objective results illustrated that, on average, DTMF outperformed speech-enabled system in terms of the time taken to complete tasks although task completion times using the two systems were not significantly different. Another reason for speech-enabled system preference was novelty. Users mentioned that speech-enabled system is trendy and it is an advanced technology that is beyond their familiarity with DTMF systems that they have used before.

Overall, it was evident how the BGR users were excited in using the BGR system by the way that they conducted the tests. An obvious observation was that users were speaking out confidently, loud and swiftly in positively articulated utterances during the interaction with the BGR speech-enabled prototype. The caller loudness and confident speaking contributed to the very high recognition levels that were experienced in the speech-enabled prototype, "as confident speakers enunciate more clearly, making the recognition task much easier" (E. Barnard, January 10, 2011, e-mail message to author). This was particularly helpful for users who were calling from cellphones in environments that were not ideal for speech recognition purposes. The researchers did not request the callers to raise their voice at any time but the callers selected this style of interaction at their own discretion.

Observations and Analyses

In our experiments for both the OpenPhone and BGR systems we have observed that the choice between DTMF and speech-enabled system is not influenced by the performance levels of the users, which implies that the effectiveness, of the system, as defined by ISO, does not influence users' choice. This was evidenced by the fact that in both case studies the users preferred systems where they performed equally or less successfully. Also, in the BGR study, the users chose the system that was marginally less efficient to them in terms of time taken to complete the required tasks, which implies that their choice is not influenced by efficiency of the system. From this it can be inferred that the objective usability measures of effectiveness and efficiency, as defined by ISO, do not have a conclusive bearing on oral users' choice of technology.

Through the usability experiments we have established that oral users' choice is affected by how the technology affords the users' current situational norms at the time of interaction with the technology. Different cultures around the world have different cultural norms concerning loudness in speaking. Poyatos (1993) makes several examples of these various norms, for example, in Kenya and Ghana, shouting in the street and talking loudly indoors is unacceptable and Ghanaians find Nigerians too loud whilst loudness is a cultural characteristic in Spain and Italy. In addition to these cultural norms there are situational norms that fluctuate above or below the standard cultural norm as when people enter a large office where people work quietly at their desks, in an exclusive lounge (particularly with an intimate low light level) or even during an abrupt silence whereby people immediately adjust their voice loudness accordingly. On the other hand, other environmental situations can force people to raise their voices, such as noisy factory or in a noisy party [14].

In BGR system the users chose the speech-enabled system because it affords the users the opportunity to shift easily from their standard cultural norm to the situational norm of loud soccer culture. Conversely, DTMF was chosen by its users because it enables them to shift easily to the situational norm of being mute about a stigmatized subject.

The loud culture of soccer evokes a situational norm that is above the cultural norm and this is attested the exuberant loud voice during participant interaction with the BGR system. On the other hand, the reality of stigma in HIV/AIDS illness within the socio-cultural norms of the OpenPhone's targeted users evokes a situational norm that is below the cultural norm as attested by the low, slow and monotonous voice during their interactions.

Conclusions

In both IVR systems the intended users perceived the systems as useful tools that would improve their livelihoods if the systems were to be fully deployed in the relevant regions. OpenPhone users viewed the technology as something that would improve their ability to deal with the illness at any time of the day and from their homesteads instead of having to travel to the clinic, which can be as far as 70 km away for some users, in order to get help. The BGR users regarded their system as a helpful means that gives them results of recently played games whilst travelling or at work without having to wait for sports news on the television or the radio.

The DTMF modality's mute interaction affords the OpenPhone's intended users' requirement to interact with the technology in privacy and the modality also affords the users' shift from their standard cultural norm to the quiet situational norm that is demanded by the circumstances of their socio-cultural predicament of stigma. Similarly, the speech-enabled interaction modality affords the BGR users' need to interact with the technology in the loud manner that they are used to when interacting with other soccer fans. The speech-enabled modality also affords the BGR users to shift from the standard cultural norm to the loud situational norm that is congruent to the loud soccer culture in South Africa.

The researchers propose that the cultural conventions of the intended users need to be incorporated into the design of IVR systems for oral users as essential design attributes because these conventions are carried over into the interactions with speech technologies such as IVR systems, as attested by the reactions of the users when confronted with the different types of applications and modes of interaction. This finding implies that voice-based systems should provide different kinds of user interfaces in different user contexts. IVR systems should provide an interface that corresponds to their typical usage context as the default interface.

Acknowledgment The authors would like to acknowledge the assistance from the Rhodes University Centre of Excellence in Distributed Multimedia, Human Language Technologies Group at the Meraka Institute-CSIR, and the National Research Foundation.

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Analysis and Implementation of Frequency Domain Equalizer for Single Carrier System in the 60 GHz Band

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Abstract

The goal of this paper is to design equalizer for a Single Carrier (SC) system operating in the 60 GHz frequency band. The main focus is to emphasize the important considerations that have to be investigated when implementing an equalizer in the millimeter wave technology, as well as to demonstrate its simplified design when it is being implemented in the frequency domain. In order to accomplish this task, the calculation of the needed coefficients has to be investigated, and therefore the channel characteristics. These coefficients would enable us to implement a simple form of an equalizer, using VHDL, which for the 60 GHz technology is an important module on the receiver side of the communication. At the end of the paper the results of the performed synthesis are being presented.

Keywords

60 GHz • SC-FDE • VHDL equalizer implementation • 802.11ad • WiGig

Introduction

As wireless networks become more and more widely used, the need for faster and more reliable network increases, along with the need for developing newer and better network approaches. Having huge amount of data that has to be transferred at real time requires developing better network solutions that work in real time, with very high data rates of multiple giga-bits per second.

Sending data such as HD video streaming requires about 1 to 1.5 Gb/s bandwidth. Also, in cases when sending for example H.264 encoded signal, previous encoding of the signal is required. Encoding the signal takes time, and consequently, it deteriorates the overall performance and implementation of the system. Furthermore, using the existing 5 GHz frequency domain may conflict with current standards and might not be accepted world-wide. As a solution to the afore-emphasized problems, we propose using uncompressed HD video signal, which requires from 2.5 to 3 GHz bandwidth (which is 60−90 % higher than compressed H.264 HD video). Because of the limited bandwidth available, the 2.4 and 5 GHz frequency bands are below the requirements. Therefore we propose using the 60 GHz frequency band, which has plentiful of spectrum available (up to 7 GHz unlicensed spectrum) [9], in accordance with the IEEE 802.11ad standard, [13]. Moreover, to send the uncompressed HD video, we propose to use from 4 to 5 GHz bandwidth of the available spectrum.

In the millimeter wave wireless technology for high-throughput systems, due to their high bandwidth efficiency, two modulations are being widely used: orthogonal frequency division multiplexing (OFDM) and single carrier (SC) block transmissions. Although OFDM modulation has become very popular and implemented in broadband communications standards, it suffers from several important drawbacks [10] that made the choice between SC and

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_57, © Springer International Publishing Switzerland 2015

OFDM a strongly debated issue and thus each is worth a deeper analysis. In this paper we focus on using the Single Carrier modulation.

Beside its advantages, such as increased throughput and smaller antennas, the 60 GHz technology also contains a number of physical challenges that need to be addressed, that includes high attenuation and severe inter-symbol interference [2]. Thus, the equalization being done on the receiver side of the communication is of huge importance. This becomes even bigger challenge when the power consumption of the system is also being considered, indicating that important design aspects and trade-offs have to be done. To achieve better performance and faster transmission rates, as well as to simplify the implementation and to lower the power consumption of the whole system, in this paper we propose implementing the Single Carrier transmission technique combined with equalization that is done in the frequency domain [11].

This paper is organized as follows: Section "Single Carrier: Frequency Domain Equalization" presents an overview of the SC modulation while emphasizing the main modules of such transceiver where the equalization is done in frequency domain; Section "Single Carrier Receiver at 60 GHz" focuses on the receiver, and gives an overview of the structure of frequency domain equalizer with decision feedback; Section "VHDL Implementation of SC-FDE Receiver" focuses on implementing such equalizer using VHDL, as well as presenting the results of the performed synthesis; and finally, the future plans on this subject, that are aimed in our project and the conclusion of this paper are being presented in Sections "Future Work" and "Conclusion", respectively.

Single Carrier: Frequency Domain Equalization

There are two preferred choices for PHY modulation scheme for 60 GHz wireless high-throughput systems: Single Carrier and OFDM. OFDM transmits multiple carriers in parallel, with each occupying a narrow band, while SC modulation transmits a SC modulated at a high symbol rate. SC modulation benefits from several interesting properties, from which most significant is the low Power to Average Peak Ratio (PAPR). The PAPR of SC transmissions is lower and does depend on the excess bandwidth of the pulse shaping filter and on the constellation, and can be traded off against spectral efficiency in a different manner for each constellation [1].

We propose using the QAM-64 modulation scheme [1], where 6 bits are being used to form a single symbol. QAM can be defined as a digital modulation format where information is conveyed in the amplitude and phase of a carrier signal. QAM combines two carriers whose amplitudes are modulated independently with the same optical frequency and whose phases are shifted by 90° with respect to each other. These carriers are called in-phase carriers (I) and quadrature-phase carriers (Q) [1].

Single Carrier Transmitter

The encoded and spread bits are fed to the constellation mapper. Exact locations of the constellation points are presented in [5], together with the mapping rules. The modulated symbols from the constellation mapper are split into blocks and the blocks can further be split into sub-blocks.

Each block (which length is N symbols) consists of the data symbols that are about to be sent (total M data symbols per block), and a cyclic prefix (CP) with length of N_{CP} symbols, as shown in Eq. (1). The cyclic prefix is a copy of the last part of the data in the block, and its main purpose is to guarantee reception without inter-symbol interference (ISI) [6]. The length of the CP depends of the channel delay spread and is equal to the maximum expected length of the channel impulse response. The IEEE 802.11ad standard recommended CP length is at least M/5.

$$N = M + N_{CP} \tag{1}$$

It should be noted that the CP insertion can be done before the bits have been modulated (for example before the QAM modulation), in which case the lengths M, N and N_{cp} are expressed in bits. Anyway we recommend that the CP is added after the QAM modulation (to have simplified coding) and therefore we use symbols when expressing the lengths. Since we use block size of 64 symbols (N = 64), we are using 12 symbols for the CP (N_{CP} = 12), and therefore M = 52 data symbols.

Single Carrier Receiver

One of the most important modules of the SC receiver is the equalizer which module is designed according to the characteristics of the channel. Although SC modulation benefits with interesting properties over its counterpart-OFDM modulation [1], its main drawback is the need of implementing very complex equalizers in time domain. The purpose of the time-domain equalizer is to minimize time-domain inter-symbol interference [2], and therefore, it is being implemented on the receiver side of the communication (Fig. 1). When implementing such equalizer, beside its high complexity, a negative fact is that the length of the channel has to be known in advance in order to determine the equalizer length [1]. Therefore, it is worth considering going into the frequency domain at the receiver, and to perform the low-complexity single-tap equalization in the frequency domain (as it is done for the OFDM modulation) and then to go back to the time domain.

SC modulation with frequency-domain equalization (SC-FDE) implies that an FFT and an IFFT are being used at the receiver and the equalization is being performed on a block of data, with total length of N (Fig. 1).

Notable difference between OFDM and SC-FDE is that in an OFDM receiver, the received CP is discarded before the FFT processing is done on each block. On the contrary, CP in SC-FDE cannot be discarded before the FFT operation, because it is essential to the proper functioning of the FFT since it makes the received block to appear periodic with period N (which makes the circular convolution possible). Therefore, the CP adds additional overhead while the equalization is being performed. The model of the receiver in a Single carrier—Frequency Domain Equalization is shown in Fig. 2.

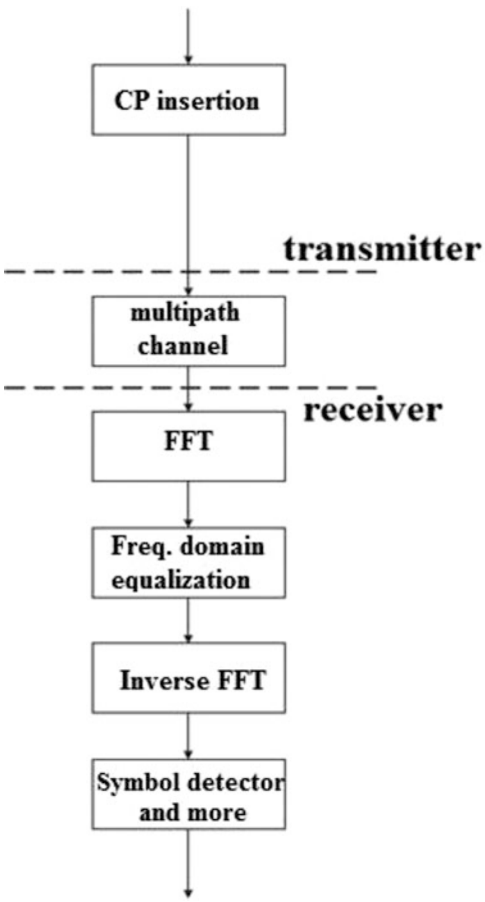

Fig. 1 System diagram for SC-FDE transceiver

Single Carrier Receiver at 60 GHz

Single Carrier Receiver Modules

An improvement of the aforementioned equalizer is the implementation of decision feedback equalization (DFE) which gives better performance for frequency-selective radio channels than linear equalization [4]. Since in a SC-FDE system, the symbols are being detected in time-domain, and the equalization is being done in frequency-domain, a hybrid time-frequency domain DFE approach [3] is often used, illustrated in Fig. 3. This type of equalizer has two main parts: (1) one that deals with the precursor ISI (the multiplication in frequency domain); (2) and a second part that deals with the postcursor ISI [8] that assumes that the detected symbol is correctly received, and therefore trying to remove its interference on the symbols that are received afterwards (Fig. 4).

The received symbols can be sampled once or more than once per single symbol period, where I is the number of samples per symbol period. Higher value for I ensures that the equalizer would be insensitive to sampling phase, as shown in [3], which tradeoffs with the need of high sampling rate (thus higher power consumption). For simplicity and for lower power consumption, we consider a system with $I = 1$. Due to the presence of the cyclic prefix, the received complex samples $\{r_m\}$, sampled at rate $1/T_S$, can be assumed to be periodic ($r_m = r_m \pm LN$, for any integer L). The same assumption can be made for the channel impulse response samples ($\{h(nT_s/I) = h((n/I \pm LM)T_s)$).

If we consider that $n = NI$, and since $I = 1$, $n = N$ then the optimum frequency domain forward filter coefficients can be expressed as given in (2), (for derivation of (2), see [3]):

$$W_l = \frac{H_l^*\left[1 + \sum_{kF_B} f_k^* \exp\left(-j2\pi\frac{kl}{N}\right)\right]}{\sigma^2 + \left|\hat{H}_l\right|^2}, \text{for } l = 0, 1, 2, \ldots (NI-1) \tag{2}$$

where $\left|\hat{H}_l\right|^2$ is expressed as:

$$\left|\hat{H}_l\right|^2 = \frac{1}{I}\sum_{k=0}^{I-1}\left|H_{(l+kN)_{\text{mod}NI}}\right|^2 \tag{3}$$

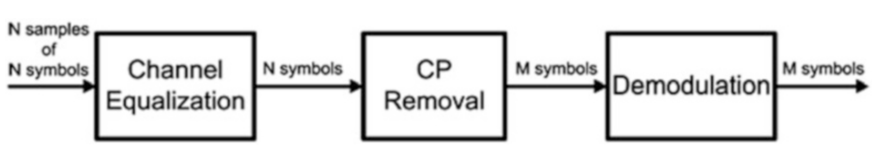

Fig. 2 Model of the receiver side of the SC-FDE

FDE Equalizer at 60 GHz

While calculating the FFT, the values for the H_l can be stored in separate registers for later use when calculating the W_l values (given in (2)). The multiplied values are forwarded to the IFFT module to return the symbols in time domain where each output of the IFFT is sampled once per symbol period thus each output is a symbol in time domain, [6]. The output from the IFFT is subtracted with feedback coefficients to reduce the error in detection (Fig. 3). To simplify the implementation, the feedback coefficients are set to 0, i.e. the F_b is an empty set. Table 1 shows the parameters for the 60 GHz implementation of the SC-FDE without DFE.

The symbol period in a single carrier depends on the frequency bandwidth that is being used, since one symbol is transmitted over the entire available bandwidth. Because we are using 4 GHz of frequency bandwidth, the symbol period is 0.25 ns. Considering the values given in the above specified table, for H_l [3] we get:

$$H_l = \sum_{n=0}^{63} h(nT_S)\exp\left(-j\pi\frac{nl}{32}\right); \quad (4)$$

where h(t) is the channel's impulse response, for which we consider the measured values shown in Fig. 5 [7].

Consequently (considering Eq. (4) and measurements from Fig. 5), we can calculate the values H_l, for $l = 0,1,\ldots,63$ (as for the undefined $h(iT_s)$ values, ranging from $i = 16$ until $i = 63$, we use value of 0).

Since the sampling is done once per symbol ($I = 1$), $|\hat{H}_l|^2 = 0$ (from Eq. (3)) and therefore, the W_l coefficients can be calculated using the following equation:

$$W_l = \frac{H_l^*}{\sigma^2}, \text{ for } l = 0, 1, 2, \ldots, 63 \quad (5)$$

where H_l^* is the complex conjugate of H_l (4).

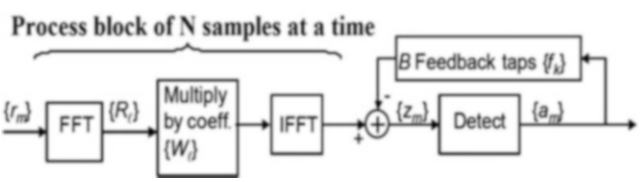

Fig. 3 SC-FDE with decision feedback equalization

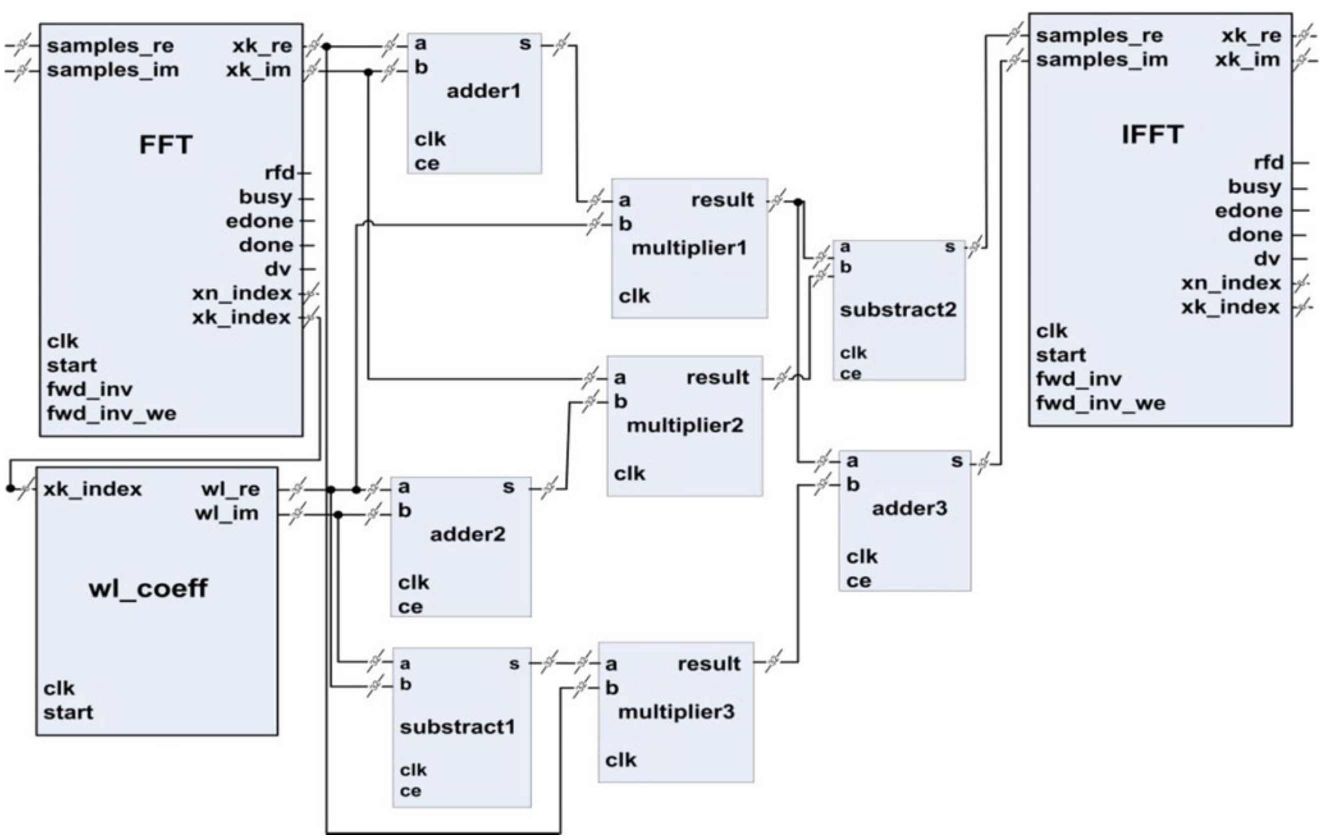

Fig. 4 Modular structure of the SC-FDE equalizer, implemented using VHDL

VHDL Implementation of SC-FDE Receiver

As it was demonstrated, the output (64 complex numbers) of the FFT has to be multiplied with the calculated W_l (64 complex numbers) values. Since we need to multiply complex numbers, where one multiplication of floating point numbers is about 30 times more time-consuming than the addition, we consider the Eq. (6), on which is based the implemented equalizing module illustrated in Fig. 4. Instead of having four multipliers (two multiplications to calculate the real part of the complex result, and another two to calculate the imaginary), based on Eq. (6) we can implement three multipliers (since two multiplications are identical), for a price of increasing the number of used adders.

Table 1 Parameters for SC-FDE for 60 GHz Band and 3 GHz bandwidth

Value	Parameter
$N = 64$	Size of FFT block
$I = 1$	Number of samples per symbol
$T_s = 0.25$ ns	Symbol period
$\sigma^2 = 2$ dB	Variance of additive noise [7]
$f_i = 0, i\varepsilon F_B$	Feedback coefficients

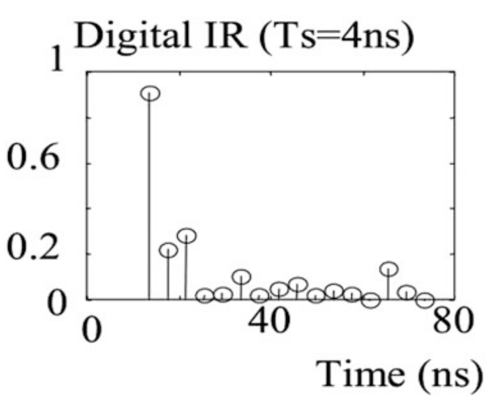

Fig. 5 Measurements of tapped delay on 60 GHz

$$(a + bj) \times (c + dj) = [(a + b)c - b(d + c)] + j[(a + b)c + a(d - c)] \quad (6)$$

As illustrated in Fig. 4, we use two Xilinx LogiCORE™ IP Fast Fourier Transform (FFT) (one FFT and one IFFT), as well as multipliers and adders, and a module where the W_l coefficients are being stored.

The Xilinx FFT module is configurable and is based on the Cooley-Tukey FFT algorithm for calculating the Discrete Fourier Transform [12]. The used ports are given in Table 1, along with their description.

In order to implement the "wl_coeff" component (illustrated in Fig. 4), we used 128 registers where the coefficients are being stored (64 for the real coefficients and 64 for the imaginary ones). For addressing the corresponding register, we use the 6-bit out-port of the FFT component xk_index, which is input for the decoder that selects one pair of three state buffers. The three state buffers (TSB) are needed in order to avoid signal conflict, since there are 64 buffer outputs connected to the same port (wl_re and wl_im). Only one of the buffer outputs should be present at the output wl_re/wl_im and therefore the rest of the TSB buffers should be in high impedance state.

To implement the above demonstrated design (Fig. 4) of the FDE equalizer, we used Xilinx ISE Design Suite along with the integrated simulator ISim where the presented design was tested.

The results of the synthesis are given in Table 3, where we use the device XC5VLX110T, family Virtex5 (with selected speed of -1).

Future Work

The presented results (given in Table 3) are below the aims of our project, which main goal is uncompressed HD video transmission with as low as possible power consumption, and therefore the presented design would be further

Table 2 Used ports of the FFT module and their description

Port name	Description
XN_RE	Input data bus: Real component in two's complement or single precision floating-point format
XN_IM	Input data bus: Imaginary component in two's complement or single precision floating-point format
START	FFT start signal (Active High).
FWD_INV	Control signal that indicates if a forward FFT or an inverse FFT is performed. (1-forward transform, 0-inverse transform)
FWD_INV_WE	Write enable for FWD_INV (Active High)
CLK	Rising-edge clock
XK_RE	Output data bus: Real component in two's complement or floating-point format
XK_IM	Output data bus: Imaginary component in two's complement or single precision floating-point format
XN_INDEX	Index of input data
XK_INDEX	Index of output data
DONE	FFT complete strobe (Active High): DONE transitions High for one clock cycle when the transform calculation has completed

Table 3 Results of the performed synthesis

Parameter	Value
Maximum frequency	229.609 MHz
Maximum combinational path delay	0.55 ns

improved. The leading idea for future improvements of this design is to implement as much parallelization as possible. The so far implemented equalizer presents a good base which would later be improved, and which would also be useful for comparison purpose while separately investigating the different types of equalizers.

Our future plans include improving the FFT modules by designing them as simplified as possible, as well as consuming energy as less as possible [14]. Since we deal only with the precursor ISI so far, our future plans also include implementing the second part of the DFE equalizer.

As mentioned, one of the main goals of our project is to lower the power consumption of the transceiver, so the advantages of the frequency domain equalizer become debatable. This is consequence of the fact that the FFT module requires a lot of processing and therefore consumes a significant amount of energy. Although the simplified design of the FDE cannot be neglected (and thus the hardware implementation is simpler), both approaches should be investigated and measured.

Conclusion

Two possible PHY modulations are used for the 60 GHz band for high throughput system. The SC modulation is being reviewed in this paper. Also, a brief analysis of the possible choices when implementing an equalizer in a SC transceiver system is presented. One of the main decisions that has to be made is whether the equalization should be implemented in frequency or in time domain. Furthermore, to improve the performances of the equalizer it is better to make it adaptable, and therefore FDE equalizers are being widely used.

We implemented equalizer in frequency domain using VHDL, whose module structure was presented in the paper. The 60 GHz channel characteristics had to be explored, that directly affect the calculated values of the W_1 coefficients, needed for implementing the equalizer. The synthesis results are also presented in the paper.

Acknowledgment This work was partially supported by the ERC Starting Independent Researcher Grant VISION (Contract n. 240555).

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Recent Trends and Developments for Standardization of Video Quality Assessment Metrics

Ranjit Singh and Naveen Aggarwal

Abstract

In this paper we have discussed the progress of research in the area of video quality model standardization and the future challenges in that area. First we have discussed the various projects completed by VQEG to evaluate the performances of various video quality models. The video databases used by VQEG projects are discussed briefly. Based upon the proposals of VQEG, ITU has proposed the standardized recommendations for various video communication services. The video quality assessment related various ITU recommendations are presented. After discussing the application areas of video quality assessment, the recent research trends in that area are discussed. In this paper, the role of audio is not discussed and only the issues related to video are discussed for video quality assessment.

Keywords

Video quality assessment • VQEG projects • ITU recommendations • VQEG video databases • Video quality assessment applications • Video quality assessment research issues

Introduction

Recent developments in popular video services like video communication, video streaming over internet and video broadcasting etc., lead to huge volume of video data production and exchange among different parties all over the communication networks. For efficient utilization of network resources, storage space and for many other applications we need video quality assessment.

Video quality is the term which is related with human subject. Video quality is the perceived quality of video by human subject. So efforts have been made to simulate the human perception. Video quality assessment (VQA) by human subject is time-consuming and costly [1, 2]. Therefore objective metrics are needed to automate this process. The ultimate objective of these metrics is to match with human visual perception.

The objective of this paper is to provide some useful information regarding the developments, VQEG model validation efforts, ITU (International Telecommunication Union) standards and research issues in video quality assessment. The paper is organized as follows. In Section "VQEG Video Quality Evaluation Projects" we have discussed various VQEG projects which have been completed. In Section "ITU Recommendations for Video Quality Assessment", the standardized ITU recommendations are discussed. In Section "Application Areas of Video Quality Assessment", the applications areas of video quality are discussed. In Section "Future Trends in Video Quality Assessment", the research challenges in video quality assessment are discussed. In Section "Discussion and Conclusion", some final concluding remarks are presented.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_58, © Springer International Publishing Switzerland 2015

VQEG Video Quality Evaluation Projects

A group of Members from ITU-T SG-9, ITU-T SG-12 and ITU-R SG-11formed the VQEG (Video Quality Experts Group) in October 1997 at CSELT, Turin, Italy. The purpose of VQEG was to assist the ITU standardization body to formulate the recommendations in video quality assessment. VQEG has provided the informal platform for independent evaluation of video quality metrics. The reports of these evaluation tests have to submit to appropriate ITU Study Groups. These study groups then evaluate the final test reports to formulate the recommendations in video quality for various applications. To conduct the test, VQEG has formed the groups and guidelines. VQEG has formed 4 committees as Independent Labs and Selection Committee, Classes and Definitions, Objective Test Plan, and Subjective Test Plan. These committees have designed the guidelines to perform their tasks. VQEG has completed various projects for television, broadcasting services and multimedia applications.

On the basis of availability of source reference information, VQEG has categorized video quality models into Full Reference (FR), Reduced Reference (RR) and No Reference (NR) [1].

- **Full Reference** (**FR**): In this category, models use both the original source video and delivered distorted video for quality assessment purpose.
- **Reduced Reference** (**RR**): In this category instead of using original source video, models use extracted features from the original source video along with delivered distorted video for quality assessment.
- **No Reference** (**NR**): In this category, models use only the delivered distorted video for quality assessment purpose.

The projects which have been completed by VQEG and their related information is presented in Table 1. In these projects a FR metric PSNR (Peak Signal to Noise Ratio) is included as the minimum performance metric. The PSNR is calculated according the ITU recommendation [3].The performance of submitted objective models is compared with PSNR. Only the models whose performance is at least comparable (for RR and NR models) or better than PSNR (for FR models) are considered for proposal in final project report by VQEG.

For subjective assessment, standardized methods described by ITU are used in these projects [4, 5]. There are many subjective methods defined in ITU standards and among these only DSCQS (Double-Stimulus Continuous Quality-Scale) and ACR-HR (Absolute Category Rating with Hidden Reference) methods are used in VQEG projects for subjective assessment. The obtained subjective scores are then compared with the quality scores predicted by objective models for performance testing of submitted objective models. The objective scores cannot be compared directly with the subjective score because there may be a non-linearity in quality scores. Thus a nonlinear regression function is used for mapping the objective scores with subjective scores. FR phase-I and FR phase-II projects have used the logistic mapping and other projects have used the polynomial mapping before testing the performance of objective models.

For performance testing, FR phase-I has used the 4 metrics Pearson Linear Correlation Coefficient (PLCC) between MOS (Mean Opinion Score) and MOSp (predicted Mean Opinion Score), PLCC between DMOS (Differential Mean Opinion Score) and DMOSp (predicted Differential Mean Opinion Score), Spearman Rank Order Correlation Coefficient (SROC) between DMOSp and DMOS and Outlier Ratio (OR). FR phase-II has used the 7 metrics, PLCC between DMOSp and DMOS, SROC between DMOSp and DMOS, OR, RMSE (Root Mean Squared Error), Resolving Power, Classification Errors and F-test to test the model performances. Other 3 projects have used the 3 metrics PLCC, RMSE and OR for testing model performances. The objective models which have not shown the statistical performance comparable with PSNR are not considered for proposal in final project reports. Table 1 contains the range of average PLCC results only for those models which have shown the statistical performance equivalent or better than PSNR. For each experiment in project test, PSNR PLCC is mentioned in Table 1 which acts as minimum performance standard.

The video database which have been used by VQEG and their characteristics are discussed in Table 2. Here, SRC column represents the number of source videos, HRC stands for Hypothetical Reference Circuit which represents the various test conditions such as bit-rate range, encoding and transmission errors etc. PVC stands for Processed Video Sequence and is representing various SRC and HRC combinations. Only the database of FRTV Phase-I is publically available. Also for HDTV (High Definition TV) Phase-I, five video datasets out of total six video datasets are publically available. Other VQEG databases are not publically available and are kept secret.

Now, VQEG is working towards audio-visual quality assessment, hybrid and 3D video models. In 2012, VQEG has completed two more projects HDTV Phase-II and Multimedia Phase-II. But the results and reports of these tests are not completed yet. Currently running VQEG projects are hybrid perceptual/Bitstream, 3DTV,

Table 1 VQEG completed projects details

Project (final report date) Ref.	Target applications	No. of objective models completed the test/submitted	Tested model types	Subjective method used for subjective scores	No. of viewers for subjective quality assessment	Average Pearson linear correlation coefficient (only proposed models) with PSNR score	Proposed model to ITU
FRTV Phase-I (Jun, 2000) [6]	SD TV video (525-line and 625-line)	9/10	FR-09	DSCQS, 5-scale DMOS	287	All FR models-0.80 PSNR-0.80	No proposal
FRTV Phase-II (Aug, 2003) [1]	SD TV video (525-line and 625-line)	6/10	FR-06	DSCQS, 5-scale DMOS	93	**For 525-line**: 0.84 & 0.94 PSNR-0.80 **For 625-line**: 0.78–0.90 PSNR-0.73	FR-2 for 525-line FR-4 for 625-line formats
MM Phase-I (Sept, 2008) [7]	Mobile/PDA and broadband internet communication services	25/31	FR-12 RR-07 NR-06	ACR-HR, 5-scale DMOS	984	**FR models**: ***VGA***: 0.79–0.83 PSNR-0.71 ***CIF***: 0.78–0.84 PSNR-0.66 ***QCIF***: 0.76–0.84 PSNR-0.66 **RR models**: ***VGA***: 0.80 PSNR-0.71 ***CIF***: 0.78 PSNR-0.66 ***QCIF***: 0.77–0.79 PSNR-0.66	FR-12 for 3 resolutions RR-7 for 3 resolutions; No NR model is proposed
RRNR-TV (Jun, 2009) [8]	SD TV video (525-line and 625-line)	7/12	RR-07	ACR-HR, 5-scale DMOS	32	**RR models**: **For 525-line**: 0.80–0.91 PSNR-0.83 **For 625-line**: 0.65–0.90 PSNR-0.86	RR-6 for 525-line RR-4 for 625-line formats; No NR model is proposed
HDTV Phase-I (Jun, 2010) [9]	HD TV video (1080i & 1080p)	8/14	FR-05 RR-03	ACR-HR, 5-scale DMOS	24	**FR models**: 0.87 PSNR-0.78 **RR models**: 0.77 (all 3) PSNR-0.78	FR-1 RR-3 for HDTV format; No NR model is proposed

Table 2 VQEG video databases and their characteristics

Database	SRC	HRC	PVC	Errors type	Video bitrates	Codecs
FRTV Phase-I	20	16(18 for 50 Hz and 60 Hz each; (2 HRCs are common)	360	Coding distortions, A/D conversion distortions (no IP impairments)	768 kb/s–4.5 Mb/s for low quality; 3–50 Mb/s for high quality	H.263, MPEG-2
FRTV Phase-II	13	14 for 525-line NTSC & 10 for 625-line PAL	128	Coding distortions, A/D conversion distortions (no IP impairments)	768 kb/s–4 Mb/s for 625-line PAL ; 768 kb/s–5 Mb/s for 525-line NTSC	H.263, MPEG-2
MM Phase-I	346	13 VGA, 14 CIF and 14 QCIF	5320	Coding and transmission both, pre-and post- processing effects, live network conditions, interlacing problems	128 kbits/s-4 Mbits/s for VGA; 64–704 kb/s for CIF; 16–320 kb/s for QCIF	H.263, H.264/AVC (MPEG-4 Part 10), MPEG-2, MPEG-4 Part 2
RRNR-TV	12	34	156	Coding and transmission both, pre-and post- processing effects	1.0–5.5 Mbit/s For MPEG-2. 1.0–3.98 Mbit/s for H.264	MPEG-2, H.264/AVC (MPEG-4 Part 10)
HDTV Phase-I	13	19	168	Coding only and coding plus transmission errors, both digital and analog impairments	1–30 Mbps for 1080p; 1–30 Mbps for 1080i	MPEG-2, H.264/AVC (MPEG-4 Part 10)

Table 3 ITU video quality recommendations

VQEG report	ITU recommendations (year) Ref.	Target applications	Recommended models (number of models)
FRTV Phase-II	ITU-T Rec. J.144 (2004) [10]	Objective FR methods for digital cable TV for NTSC and PAL video formats	BT, Yonsei, CPqD, and NTIA for NTSC and PAL (FR-4)
	ITU-R Rec. BT.1683 (2004) [11]	Objective FR methods for standard definition digital broadcast TV	BT, Yonsei, CPqD and NTIA (FR-4)
MM Phase-I	ITU-T Rec. J.247 (2008) [12]	Objective FR methods for multimedia applications	OPTICOM, Psytechnics, Yonsei and NTT for 3 resolutions (FR-12)
	ITU-T Rec. J.246 (2008) [13]	Objective RR methods for multimedia services over digital cable TV networks	Yonsei 10 k, 64 k, 128 k for VGA, Yonsei 10 k, 64 k for CIF and Yonsei 1 k, 10 k models for QCIF (RR-7)
	ITU-R Rec. BT.1866 (2010) [14]	Objective FR methods for broadcasting applications using low definition TV	OPTICOM, Psytechnics, Yonsei and NTT for 3 resolutions (FR-12)
	ITU-R Rec. BT.1867 (2010) [15]	Objective RR methods for broadcasting applications using low definition TV	Yonsei 10 k, 64 k, 128 k for VGA, Yonsei 10 k, 64 k for CIF and Yonsei 1 k, 10 k models for QCIF (RR-7)
RRNR-TV	ITU-T Rec. J.249 (2010) [16]	Objective RR methods for digital cable TV for NTSC and for PAL video formats	Yonsei 15 k, 80 k and 256 k, NEC 80 k and 256 k, NTIA 80 k bandwidth models for NTSC (RR-6) Yonsei 15 k, 80 k and 256 k, NTIA 80 k bandwidth models for PAL (RR-4)
HDTV Phase-I	ITU-T Rec. J.341 (2011) [17]	HDTV objective FR methods for digital cable TV	VQuad-HD (FR-1)
	ITU-T Rec. J.342 (2011) [18]	HDTV objective RR methods for digital cable TV	Yonsei 56 k, 28 k, 256 k side-channel bandwidth models (HD FR-3)

AudioVisual HD (AVHD), High Dynamic Range Video (HDR), Ultra HD etc.

ITU Recommendations for Video Quality Assessment

In Table 3, the various ITU standards have been discussed which are resulted from VQEG project reports. No ITU recommendation is proposed based on the VQEG FRTV Phase-I final report because no model performs better than the PSNR metric and all models have shown the statistical equivalent performance with PSNR. Other VQEG project reports have proposed some models for recommendation purpose and ITU recommends some video quality models as standardization. These recommendations and related information have been discussed in Table 3.

Application Areas of Video Quality Assessment

The video quality models can be used for many quality control applications but not limited to:

- Specification, evaluation and acceptance testing of video codecs.
- Pre-service video quality assessment at source based upon the codec specifications and network conditions before starting transmission.
- In-service video quality monitoring and controlling at remote destination as per user requirements.
- Video quality monitoring and controlling at various nodes between source and destination for optimal utilization of transmission system resources.

Future Trends in Video Quality Assessment

Previously, we have discussed the progress in video quality assessment is but still a lot of research work have to done in this field. In this section we are discussing the various research issues in video quality assessment area.

From the above mentioned standardized recommended methods, it has been observed that there is no standardized NR video quality model for video sequences. In real-time environment, no source reference information is available. All the efforts to validate the NR model by VQEG are not successful because the performance of all submitted NR models is below the reference PSNR performance.

Secondly, it has been observed the fact that VQEG is only concerned with the measurement of accuracy of video quality models. The important aspects of computational and resource complexity for video quality models are not considered. For the real-time video quality model, computational complexity and resource requirements must be as low as possible. Also, there is a tradeoff between prediction accuracy and computational complexity. Thus, there is need to consider these issues in model designing and implementation. For complexity issue, the PSNR metric can be used as the minimum complex reference model.

Third, it has been observed that the accuracy of recommended models is not sufficient so that they can replace the subjective assessment with objective assessment. No model has achieved the accuracy comparable to subjective assessment. Thus, there is still a room for improving the accuracy of video quality models so that the accuracy of video quality models can match with the human perception of video quality assessment.

Fourth, the recommended models have their limited use and scope. To use the objective model effectively, there are many constraints upon it to use. Beyond their scope and constraints, they have been failed to predict video quality reliably and the results are degraded. Thus, other than the application oriented video quality models, there is a need of the general model which requires minimum constrains and general scope to use.

Fifth, existing models are only producing the video quality score and there is a need of some more information other than single predicted quality score. These models must provide some mechanism for the optimization of video processing system such as bit-rate control to achieve desired video quality level. But these models are not providing the effective framework for optimizing the video processing system.

Sixth, there is a need of considering the Human Vision System (HVS) in an effective way to improve the model performance. The end user of video quality system is the human being and it judges the video quality based upon the capabilities of HVS. So the complete understanding of HVS process for quality assessment is needed. Research in this field is carried out but the full understanding of HVS is not known yet. For video quality assessment task, there is need of research in HVS areas and need to incorporate the latest research results to improve the video quality models. The areas which are of most importance in this field are the role of cognitive mechanism, the pooling of information in HVS and the role of visual attention (VA) mechanism in quality assessment mechanism.

Seventh, there is a need for comprehensive generic nature video quality database for testing the video quality models. Today the video databases which are available based on particular artifacts and distortions. These databases have shown only limited behaviour of real time environment for video quality assessment. No single database is of generic nature. There is a need of videos representing real time scenarios.

In this section, we have discussed various research issues in the area of video quality assessment and research efforts are needed for problem solving.

Discussion and Conclusion

For various communication services, there is a need of video quality assessment metrics for optimal utilization of limited network and storage resources. Also, many other application areas are described in this paper. In the past, efforts have been made to develop the best perception objective model for video quality assessment. To contribute in this research area, VQEG is providing the platform for independent testing and validation of objective models. This leads to the standardization of video quality models by ITU.

In this paper, we have discussed the various projects which are conducted by VQEG for testing and validation of video quality models. Based on the results of these projects the standards recommended by ITU are also discussed. After discussing these, the current research issues in video quality assessment are discussed. These research issues are of great importance and need researcher's attention to design efficient video quality assessment model.

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An Efficient 64-Point IFFT Hardware Module Design

Danijela Efnusheva, Aristotel Tentov, and Natasha Tagasovska

Abstract

This paper presents the process of designing a 64-point IFFT hardware module, as a 2D structure of 8-point IFFT pipeline modules. The proposed 64-point IFFT module utilizes only two 8-point IFFT modules, which include minimal number of multiplications and additions, and as well provides parallel processing of eight symbols in each pipeline phase. This allows high throughput performances of the proposed 64-point IFFT module and chip area savings of its hardware implementation on Virtex 5 FPGA. The realized hardware design can be easily applied in a high-speed real-time system, such as OFDM-based communication system.

Keywords

64-point IFFT hardware module • 8-point IFFT pipeline modules • Virtex 5 FPGA • Real-time system • OFDM-based communication system

Introduction

Inverse Fast Fourier Transformation represents an efficient algorithm for computing Inverse Discrete Fourier Transformation (IDFT), considering the reduced number of calculations needed to perform the same objectives. In algorithm context, IFFT execution complexity corresponds to $O(N\log_2 N)$ [1, 2]. As a result of this feature, IFFT has extensive application in analyzing and implementing communication systems with real-time data transmission requirements. Furthermore, in wireless network systems it is recommended for the chip to occupy less surface area in order to achieve reduction in the energy consumption [3–5].

DSPs [6] are programmable and flexible, but cannot completely satisfy fast processing requirements. In such cases, specialized hardware circuits are developed, dedicated for IFFT calculation. FPGA components are utterly suitable for implementation on this type of circuits, providing a tradeoff between speed, cost, flexibility and programmability [7–9].

IFFT module executes the main functionality in OFDM systems on sending side, which recently became attractive for their extensive use in modern communication systems. IFFT's purpose is to convert signals from frequency domain to time domain, at the same time enabling orthogonality between the subcarriers in the OFDM system [2–4]. Actually, the process of modulation of the subcarriers in the channel with symbol information and making them orthogonal to each other is performed by means of IFFT on transmitting side. By including this "IFFT module" in OFDM systems, the signal processing complexity is reduced (at both, transmitting and receiving side) and higher transmission rates are achieved.

Because of the processing complexity while executing IFFT calculations, several implementations are introduced [10, 11] in order to attain real time processing and reduction of hardware complicatedness. In general, there are two directions in this area of research. The first one deals with developing algorithms for IFFT and their optimization. Best known algorithms in this field, worth mentioning in this paper are radix-2, radix-4, radix-8 and split-radix variations of the

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_59, © Springer International Publishing Switzerland 2015

Cooley-Tukey (C-T) algorithm [12], as well as Winograd algorithm [6]. By increasing the base in C-T algorithm, the number of operations is decreased, resulting in 64-point IFFT split-radix implementation becoming superior compared to Winograd algorithm [13]. The second approach is focused on hardware improvement and optimization of the IFFT module. This includes adequate techniques for parallel processing and pipelining, memory structures for preservation of previously calculated results, as well as dedicated components (usually multipliers) for more rapid calculations [14–16].

Standard radix-2 Cooley-Tukey 64-point IFFT module contains 192 butterfly processes, which perform exactly 192 ($N/2*\log_2 N$) multiplications and 384 ($N*\log_2 N$) additions of complex numbers [2]. Due to the high complexity of this implementation, this paper evaluates the process of designing IFFT module with size of 64 points, by employing two IFFT modules with 8 point sizes. These modules implement an improved C-T radix-2 algorithm with minimal number of operations, and can operate in pipeline manner. It is expected for the realized 64-point IFFT module to contain less number of operations, which shall reflect in chip complexity reduction, along with shortened time of execution.

This paper is organized in six sections. Section "Current State"discusses variety of algorithms for IFFT implementation and several current modules of IFFT realization. Section "8-Point IFFT Module Design" displays the actual process of designing 8 point IFFT module, later used for implementing 64 point IFFT module, whose architecture is shown in section "64-Point IFFT Module Design". Section "Performance Analyses" elaborates on the performances of the realized 64-point IFFT module, in terms of complexity and speed. The paper ends with conclusion, stated in section "Conclusion".

Current State

Since IFFT module has a fundamental role in the OFDM systems functionality on sending side, recently it has drawn the attention of a huge number of researchers towards developing hardware components solely for IFFT, capable of real-time processing. Most of the researches are performed on FPGA platforms, basically because of their availability and low price [2, 7–9].

For realization of IFFT module diverse algorithms can be used, each of them having its own advantages and disadvantages. However, every one of this algorithms [17], is more efficient then the algorithm for IDFT computation, where algorithms' complexity reaches $O(N^2)$ [2]. Among the most frequently used IFFT algorithms are the Cooly-Tukey based algorithms, with variation of radix values (2, 4, 8, split etc.), whose complexity amounts $O(N\log_2 N)$. The value of the radix (2, 4, 8) indicates that the total number of points used for the transformation can be expressed as 2×, 4× or 8×, correspondingly [17]. These C-T algorithms execute parallel butterfly processes which perform independent operations with 2, 4 or 8 input/output values, accordingly, and therefore allow faster execution of the IFFT computation. On the other hand, split-radix [14], combines C-T algorithms with different radixes, for computing distinctive partitions of Fourier's transformation in order to reduce the number of required operations. In C-T based algorithms, Twiddle factor multiplications are required, so if the decomposition of N is relative prime, the prime factor or Winograd FFT algorithms can be used to reduce the Twiddle factor multiplications [17, 18]. The last algorithms are implemented in circuits where the cost of performing multiplication is much larger than the one for performing addition calculations.

Table 1 presents an analysis of required multiplication and addition operations (with complex numbers) for different IFFT algorithms. From the results, a conclusion can be made that in general, split-radix algorithm achieves best results. Nevertheless, due to its irregular structure, it introduces difficulties while developing hardware implementation [13]. This is not the case with the C-T radix 2, 4 and 8 algorithms. For an instance, the radix 2 C-T algorithm is a common choice for IFFT implementation [2, 9], in spite of the fact that it includes less number of additions, on the account of larger number of multiplications. In comparison with the C-T based algorithms, the Winogard algorithm

Table 1 Multiplication and addition complexity for various IFFT algorithms

	DFT	C-T Radix 2			C-T Radix 4			C-T Radix 8			C-T Split Radix			Winograd		
Points	Total	Adds	Muls	Total	Adds	Muls	Total	Adds	Muls	Total	Adds	Muls	Total	Adds	Muls	Total
16	**496**	64	32	**96**	96	24	**120**				64	8	**72**			
60	**7,140**													486	72	**558**
64	**8,128**	384	192	**576**	576	144	**720**	896	112	**1,008**	384	72	**456**			
240	**114,960**													3,042	324	**3,366**
256	**130,816**	2,048	1,024	**3,072**	3,072	768	**3,840**				2,048	456	**2,504**			
504	**507,528**													8,946	936	**9,882**
512	**523,776**	4,608	2,304	**6,912**		1,728		10,752	1,344	**12,096**	4,608	1,082	**5,690**			
1,008	**2,031,120**													21,546	2,106	**23,652**
1,024	**2,096,128**	10,240	5,120	**15,360**	15,360	3,840	**19,200**				10,240	2,504	**12,744**			

performs less number of multiplications than the split-radix algorithm, but the number of additions is significantly increased by roughly 40 %.

Results presented in Table 1 consider operations with complex numbers, whose real and imaginary components are real numbers. Therefore, the addition operation can be carried out by two operations of adding real numbers: (Xre + Yre) + (Xim + Yim)j, whereas multiplication operation can be executed with two additions and four real number multiplications:(Xre*Yre-Xim*Yim) + (Xre*Yim + Xim*Yre)j. Currently, there are a large number of IFFT realizations which implement algorithms' advancements, by reducing the number of multiplication operations, or accelerating processing by means of pipelined execution of operations. This approach is resolved in [11], where architecture of FFT/IFFT pipelined processor with 8 points is presented, which reduces the number of multiplications in the C-T algorithms to zero. That is achieved through utilization of supplementary operations for addition and shifting, instead of multiplying. Very similar optimization approach, which includes specialized multiplier, is presented in [16].

Usually for IFFT modules' realization, FPGA components are being used. The same platform is also applied for implementing FFT modules that operate on the receiving side of an OFDM system. Basically the FFT modules execute the same set of operations as IFFT, while converting the signal from time to frequency domain. For example, the authors of [8] suggest a 64 point FFT processor realization, as 2D structure made of FFT modules with 8 points, on Spartan 3 FPGA component. Because of its primary hardware improvements, this module has 76 tact cycles latency, while occupying 29 % less resources of the standard/default FFT IP core from Xilinx. Another realization of 64 point FFT module on Spartan 3 FPGA component is presented in [15]. This chip holds 22 % of FPGAs' slice resources and is characterized by 0.081 W total energy consumption. Another FPGA realization of merged FFT/IFFT 64-point module on Virtex4 and Virtex5 components is proposed in [9]. This module [9], has a 163 tact cycles latency and occupies 22 % of the FPGAs' slice resources.

Apart from FPGA realizations, ASIC implementations of IFFT hardware modules also exist [4, 5]. For example, in [4] an FFT/IFFT ASIC chip, implemented in 0.25 μm BiCMOS technology with surface of 6.8 mm^2, is presented. Another two chips, discussed in [5] and [19] are implemented in TSMC 0.13-μm 1P8M CMOS and 0.18 μm CMOS, correspondingly. These ASICs have low energy consumption of 22.36 mW and 15 mW, respectively.

An interesting concept is considering the possibility of designing IFFT processors with configurable number of points. Similar approach, referring to FFT processor design is elaborated in [20].

8-Point IFFT Module Design

Inverse Fast Fourier Transform represents an efficient and fast algorithm for converting signals from frequency domain to time domain [2]. Calculations required for obtaining the result of Inverse Fourier Transformation are given by the following equation:

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)W_N^{-nk}, \quad n = 0, 1, \ldots, N-1 \qquad (1)$$

In equation (1), x(n) represents time output, for the n-th spectral point, N represents number of samples(points), and X(k) is the k-th sample obtained in the process of sampling. The inputs X(k) and outputs x(n) of the Inverse Fourier transformation are complex or real numbers. The variable W_N^{-nk}, used in (1) is called conjugated Twiddle factor and is defined as follows:

$$W_N^{-nk} = e^{\frac{j2\pi nk}{N}} \qquad (2)$$

The realized 8-point IFFT module implements radix 2 Cooley–Tukey algorithm. This algorithm is based on the "divide and conquer" paradigm, and thus performs IFFT computation of the even-indexed and odd-indexed inputs separately and then combines those two results to produce the IFFT of the whole sequence. This approach allows recursive execution of the algorithm in $\log_2 N$ phases, whereas all phases perform parallel computations, organized in butterfly processes. Each butterfly process executes one operation of multiplication and two operations of addition with complex numbers [2]. The flow graph of C-T radix 2 algorithm, executing 8-point IFFT computation is given in Fig. 1.

The module for IFFT calculation with 8 points is implemented in VHDL, by means of Xilinx ISE Design Suite tool. This module is synchronized by a clock signal, and as an input it receives an array of 16-bit real and imaginary components for eight symbols of a signal given in a frequency domain. As an output it produces an array of 16-bit real and imaginary components for eight symbols given in time domain. Instead of that, the module includes control signals for input and output enabling, which indicate if the module is ready to start operating (enable_in = 1) and if an output from the calculations is generated (enable_out = 1). Resetting of the IFFT module is done by an external reset signal (Fig. 2).

For the given 8-point IFFT module, a simplified hardware implementation of the radix 2 C–T algorithm, with reduced number of multiplications by Twiddle factors, is proposed. Assuming that Twiddle(−0) = 1 and Twiddle(−2) = j, the

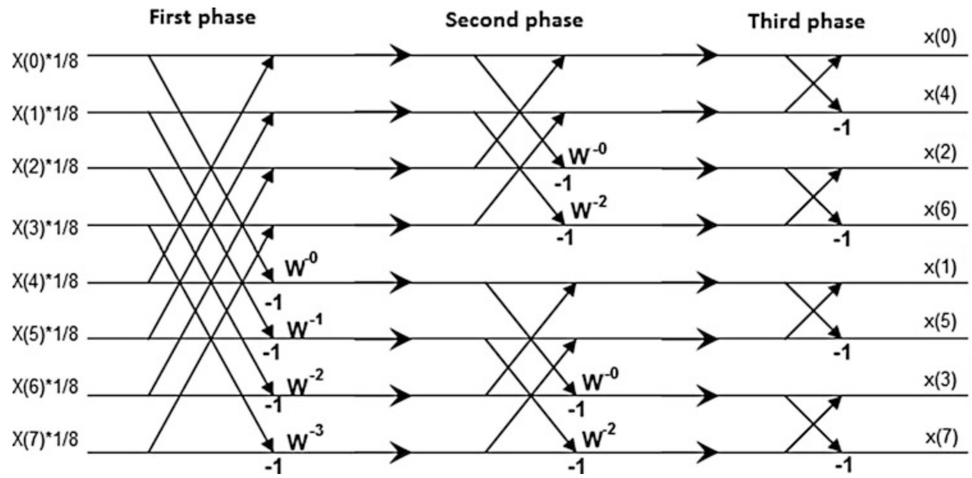

Fig. 1 Flow graph of 8-point IFFT radix 2 Cooley–Tukey algorithm

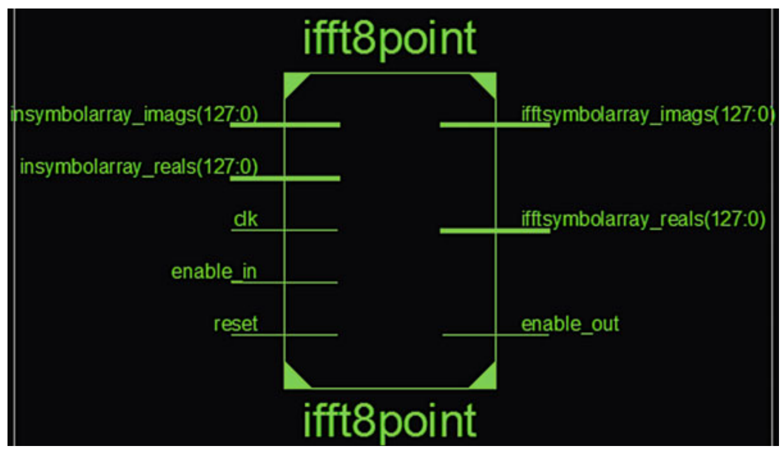

Fig. 2 8-point IFFT hardware module interfaces

multiplications by one are not taken into account and the multiplications with j are performed in such a way that the product (a + jb)*(j) is directly computed as −b + ja in hardware. On the other hand, we considered that Twiddle $(-1) = \frac{\sqrt{2}}{2} + j\frac{\sqrt{2}}{2}$ and Twiddle $(-3) = -\frac{\sqrt{2}}{2} + j\frac{\sqrt{2}}{2}$, so the multiplications of this factors by a complex number a + jb were performed by only two multiplications with real numbers:

$$\left(\frac{\sqrt{2}}{2} + j\frac{\sqrt{2}}{2}\right) \bullet (a + jb) = \frac{\sqrt{2}}{2}(a - b) + \frac{\sqrt{2}}{2}(a + b)j,$$

$$\left(-\frac{\sqrt{2}}{2} + j\frac{\sqrt{2}}{2}\right) \bullet (a + jb) = -\frac{\sqrt{2}}{2}(a + b) - \frac{\sqrt{2}}{2}(b - a)j$$

This way, the IFFT module executes only four multiplications with real numbers and a constant, in the first phase of the C–T algorithm. Furthermore, the initial multiplication by the constant 1/8 in the first phase of the C-T algorithm is realized as a simple right shift operation. As a result of these improvements, the number of adders in the IFFT module realized in Virtex 5 XUPV505-LX110T FPGA, is decreased by 33.33 %.

The proposed IFFT module performs operations over real and imaginary components of symbols that represent such 16-bit real numbers with fixed-point [21]. Comparing to the 16-bit IEEE half-precision floating-point representation, the fixed-point approach is more convenient for use in the proposed IFFT module. This is confirmed by several experiments, which show that the IFFT Virtex 5 LX110T FPGA implementation that operates with floating-point numbers, increase the complexity by 15,2 times, accordingly. Actually, the floating point IFFT module utilizes 76 % of the slice resources of Virtex 5 LX110T FPGA component, while the other IFFT module which works with fixed-point numbers utilizes only 5 % of the slice resources of the same FPGA component.

In order to achieve high throughput and continuous data flow, the IFFT module implements pipeline architecture, divided into control and data path. The control path is used to generate signals that control the operation of the functional units and the register transfers in the data path. On the other hand, the data path is purposed to enable pipeline

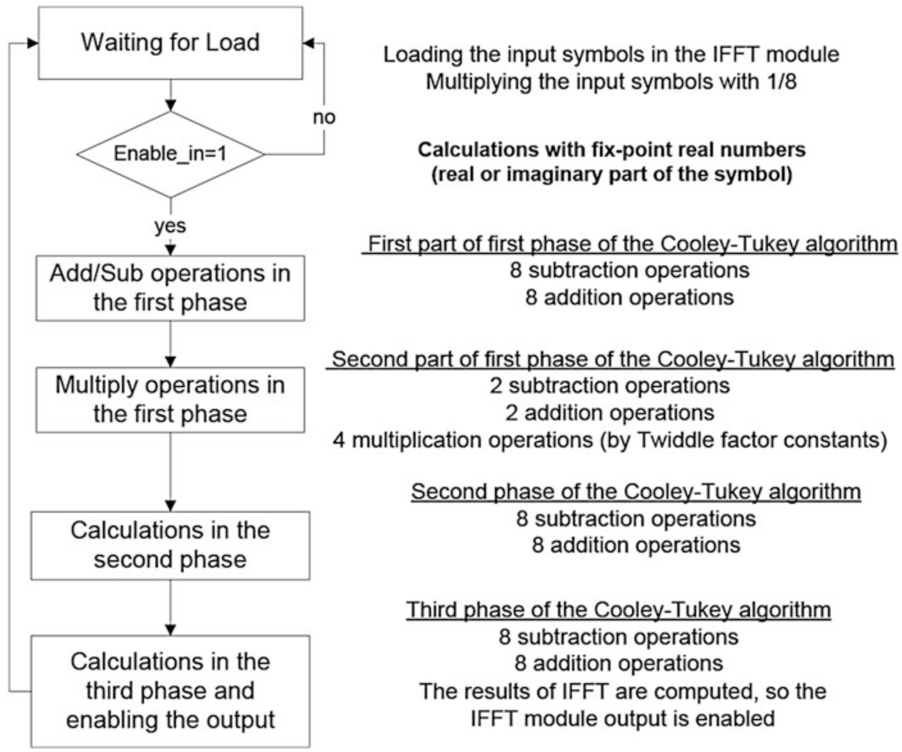

Fig. 3 Pipeline stages in the proposed 8-point IFFT module

execution of the IFFT computations in five stages (Fig. 3): operands load and multiplication by 1/8 in the first phase of the C–T radix 2 algorithm; executing the add/sub operations in the first phase of the same algorithm; executing the multiplications in the first phase; executing the operations in the second phase; and executing the operations in the third phase and enabling the calculated results at the output. During the pipeline execution, the results of each stage are placed in intermediate registers, thus allowing continuous data flow between the sequential pipeline stages. The final result of the IFFT computations is obtained after five clock cycles. If it is considered that the symbols are loaded and outputted serially, via serial to parallel and parallel to serial components, additional delay of 16 clock cycles will be caused. This gives 21 cycles delay in complete.

Intermediate registers store and transfer the intermediate results (imaginary and real parts of a symbol) from one to another pipeline stage. The IFFT module implements four sets of 32 16-bit registers. The input symbols loaded by the input module interface are multiplied by the 1/8 constant and then placed in the first set of intermediate registers: i0re-i15re and i0im-i15im, while the results of the next three pipeline stages are stored in: p0re—p15re and p0im-p15im, x0re—x15re and x0im-x15im, y0re—y15re and y0im-y15im set of registers, accordingly. The output symbols are placed directly on the output module interface. Without using the intermediate registers, pipelined execution will not be enabled, but the number of registers would have decreased four times. However, the complexity of the IFFT module, in terms of occupied Virtex5 LX110T FPGA slice resources, will change to a maximum of 1 %. Thus we can justify the use of intermediate registers in the IFFT module.

The VHDL model of IFFT module with 8 points is simulated by using ISim simulator, in ISE Design Suite tool. For this purpose, a test bench was designed, which input vectors were additionally simulated in Matlab. Through comparing the results obtained by ISim and Matlab it was shown that the IFFT module achieves the required functionality (Fig. 4). This result is of great meaning, since the designed IFFT module with 8 points will later serve as foundation for developing IFFT module with 64 points. However, an efficient 8 point IFFT module that requires minimal logic and execution time was successfully realized.

64-Point IFFT Module Design

The implementation of IFFT module with 64 points, by using C-T radix algorithm includes 192 butterfly processes, which separately execute 4 multiplication operations and 6 addition operations of real numbers. Given the fact that this approach leads to higher circuit complexity in terms of chip area and energy consumption [4, 5], commonly it is practice to use 2D structure of IFFT modules with 8 points while designing 64 point IFFT module. This approach follows the rule [5], that each IFFT module with number of points—N can be presented as a decomposition of other two IFFT modules with number of points p and q, in such a

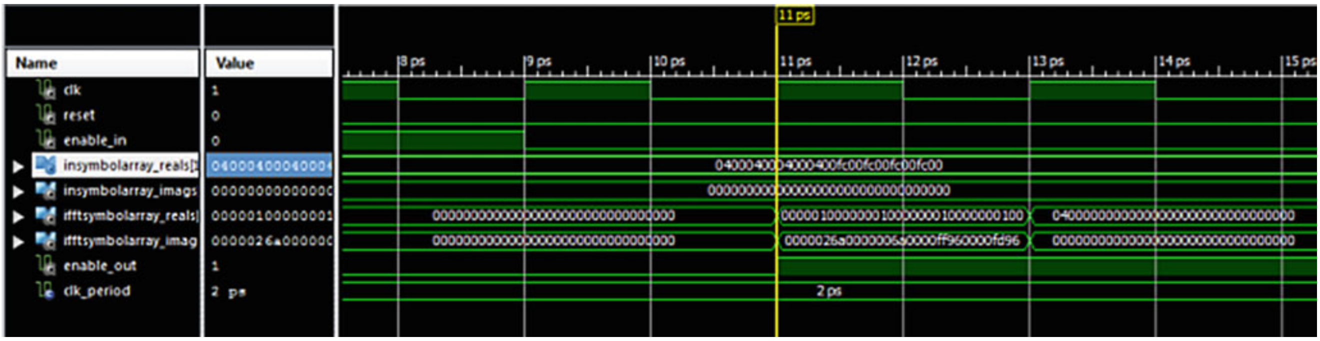

Fig. 4 8-point IFFT module simulation in Isim Xilinx simulator

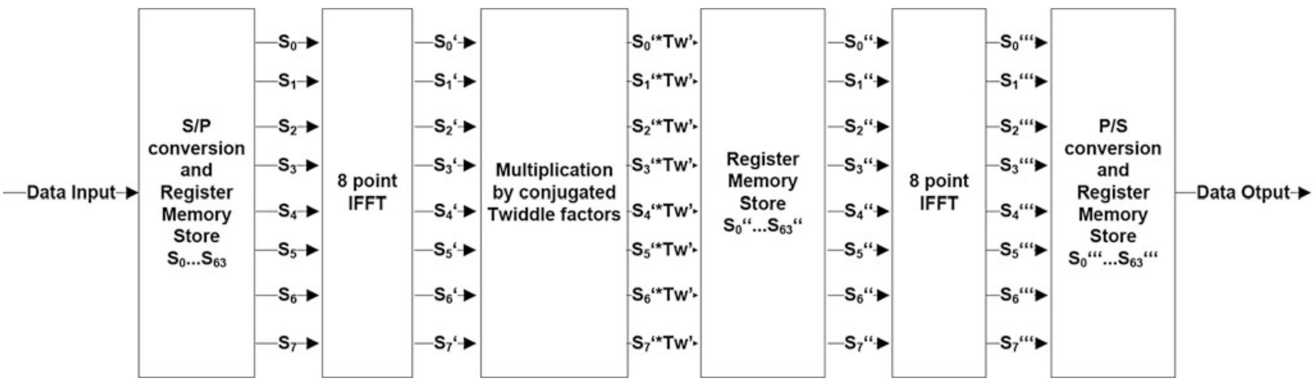

Fig. 5 Execution flow in the proposed 64-point IFFT module

way that N = p*q, assuming that p, q > 1 (in this specific case 64 = 8*8).

For the purpose of computing 64 point IFFT, as a 2D structure made of two 8 point IFFTs, the following five-step algorithm is executed:

1. Arrange 64 point input vector in 8 × 8 matrix, in rows/columns major order;
2. Calculation of 8-point IFFT for each column/row of the matrix and storing the obtained results in the appropriate column/row;
3. Multiplying every (i, j) element of the matrix by conjugated twiddle factor, whereas i = [0–7], j = [0–7]. Assuming that some of the factors' values are 1, the number of multiplications is 49 (with 9 various factors);
4. Calculation of IFFT with 8 points for every row/column of the matrix and storing the results in corresponding row/column;
5. The result is acquired by outputting the derived 8 × 8 matrix, in columns/rows major order;

In order to implement 64-point IFFT module the previous algorithm will be followed, thus several components to perform S/P conversion, 8 point IFFT computation, twiddle factor multiplication, storing in-between results and P/S conversion, are required. Computing IFFT with 8 points is performed in the second and fourth step of the algorithm. In general, each of these phases (2 and 4) can be realized by use of eight or one IFFT module with 8 points, depending on whether all of the eight modules will be working in parallel, or multiplexing in time of one module will be used [13]. In spite of the fact that by employing only one IFFT module with 8 points in the second and fourth step of the algorithm, a large latency exists, yet chip complexity is decreased by 80 %.

The architecture of the realized 64-point IFFT module (Fig. 5) is consisted of two IFFT modules with 8 points, and additional components which offer support for the rest of the steps in the previously described algorithm. The first module (shift register) assures serial to parallel conversion for the input array of 64 symbols, by loading 8 symbols in each tact cycle. Due to the fact that the orientation of the input symbols from the 2D matrix (rows) is inverse to the one required by IFFT module with 8 points (columns), the output (i.e. IFFT input) of the module for serial to parallel conversion is available even in the eighth tact cycle. In the next seven tact cycles, the rest of the seven columns with eight symbols are sent towards IFFT module with 8 points. IFFT calculation is pipeline performed in real-time of five tact cycles. After finishing the calculation, the output symbols of IFFTs' module are multiplied by the appropriate Twiddle factor constants, in one tact cycle. This functionality is executed by the multiplication module, which stores the values of nine different conjugated Twiddle factors as constants. In order to enable the next calculation of 8 point IFFT, once again altering of the inputs of the 2D structure is required, from columns to rows. This orientation is made possible by utilizing separate register memory, in which by the time of eight tact cycles the results from the multiplications will be stored, and in the last tact cycle, as in the following seven cycles, each of the rows with eight symbols is passed to the second 8 point IFFT module. Here, again, in five tact cycles a pipelined IFFT computation is performed with eight symbols from one row. The individual results are stored in output register memory, implemented as shift register. At the moment of storing IFFT calculation results for the eight symbols from the last row, parallel to serial conversion can be initiated, i.e. generating the output symbol array.

The whole architecture of the IFFT module with 64 points enables pipelined operation execution. Still, it can be perceived that there are 3 points of slowdown: symbol input, symbol memorization after multiplying and symbol output. These critical points are actually the price that should be paid for time multiplexing on both 8-point IFFT modules. Considering all of the limitations, the resulting latency in the proposed 64 point IFFT processing is 33 tact cycles. This is computed as: *7 cycles (Mem. Store) + 5 cycles (IFFT) + 1 cycle (Twiddle Mult.) + 7 cycles (Mem. Store) + 5 cycles (IFFT) + 7 cycles (Mem. Store) + 1 cycle (Generate Output) = 33 cycles*. After the initial delay of 33 tact cycles, the 64-point IFFT module generates a stream of eight output symbols in each of the next clock cycles. In comparison with the other IFFT modules presented in section "Current State" [9], one can argue that the realized 64-point IFFT module achieves better results.

Performance Analyses

The realization of IFFT module with 64 points as 2D structure made of two IFFT modules with 8 points is a result of the reduced number of operations with real numbers, comparing to the other algorithms for IFFT computation, shown in Table 2.

The results given in Table 2 show the number of non-trivial real numbers multiplications and additions of different algorithms for calculating IFFT with 64 points [17]. The provided results of the examined C-T based algorithms exclude the trivial conjugated Twiddle factor multiplications by 1 and j, and as well implement each complex numbers multiplication as three multiplications and five additions of real numbers. However, the 64-point IFFT implementation made as 2D structure of C-T Radix 2 based 8-point IFFT modules is far more efficient than the other algorithms, at least by 35 %. Although this 64-point IFFT implementation doesn't provide minimal number of multiplications, it is still close to the minimum, provided by the split radix algorithm (difference of eight operations). The 2D decomposition algorithm can provide even better results, if the previously described method of complex numbers multiplication is utilized (with three muls and five adds). This way, the number of real-numbers additions will be increased to 349, on the account of the reduced number of multiplications to minimum of 155.

Although the realized 64-point IFFT module, made as 2D structure of 8-point IFFT modules, includes less number of operations, yet, it introduces need of additional memory registers for storing results, while changing orientation of the working data (columns or rows). This operations cause additional latency in the pipeline architecture of the 64-point IFFT module, at the same time impacting its complexity.

From the results acquired with VHDL model implementation of the 64-point IFFT module in XUPV505-LX110T Virtex5 FPGA, it is shown that, the realized IFFT chip occupies 19 %, 18 % and 28 % of the slice registers, slices LUTs resources and occupied slice resources, respectively. These results are similar with the ones given in [9], and though additional optimization of the given 64-point IFFT module is expected to enhance (reduce) them. The realized IFFT module achieves maximal frequency of 70,711 MHz.

Table 2 Multiplication and addition complexity (real numbers) of various algorithms for 64-point IFFT design

	Number of additions with real numbers	Number of multiplications with real numbers	Total number of operations
C-T Radix 2	1,032	264	**1,296**
C-T Radix 4	976	208	**1,184**
C-T Radix 8	972	204	**1,176**
C-T Split radix	964	196	**1,160**
Winograd (for N = 63)	1,394	198	**1,592**
2D structure of C-T Radix 2 based 8-point IFFT modules	**202**	**204**	**406**

Conclusion

In this paper, the process of designing IFFT module with 64 points was presented, as a 2D structure of two IFFT modules with 8 points, which results in at least 35 % improvement compared to the other algorithms for 64-point IFFT computation. The realized IFFT module with 64 points executes 406 operations with real numbers in processing time of 33 tact cycles. As a result of the input loading of eight symbols simultaneously, the output from the calculations of IFFT with 64 points is generated as a pipeline, on every 8 tact cycles with initial latency of 33 tact cycles. Thus, IFFT module achieves 565,688 Msymbols/s throughput, which for QAM-64 modulated symbols can reach upper limit of 3.39 Gb/s. Considering this and the fact that the realized IFFT module consumes 28 % of the available XUPV505-LX110T Virtex5 FPGA resources, it can be concluded that it is suitable for application in OFDM based communication system, or else, as a foundation unit for designing IFFT modules with higher number of points (256, 512 or 1,024).

Acknowledgment This work was partially supported by the ERC Starting Independent Researcher Grant VISION (Contract n. 240555).

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The Relationship Between Psychological Distress and Human Computer Interaction Parameters: Linear or Non-linear?

Indika Karunaratne, Ajantha S. Atukorale, and Hemamali Perera

Abstract

During a survey based study, it has been revealed that psychological distress is reflected in the way computer users interact with computers. Based on the feedback of the subjects, the types of stress-reflector interactions were also identified. However, the specific interactions related to the stress, the nature of the relationship between the identified 'stress-reflectors' and the actual level of psychological distress was yet to be investigated. Accordingly, the initial survey was then followed by a series of activities with the objective of uncovering the nature of the relationship exists between the two. Initially, the most frequented interaction behaviors of computer users under stress were identified. Secondly these behaviors were recorded using a background analyzer together with a simultaneous measure of the distress level of the relevant computer user. Finally, behavior measures of individuals were mapped with their personal distress scores, producing the dataset using which the nature of correlations were explored. This paper presents the work carried out during this exploration, keeping the initial focus on linear methods, and provides justifications for the claim: The relationship between the distress score and the human computer interaction parameters is non-linear.

Keywords

Human computer interaction • Behavior parameters • Psychological distress • Correlation analysis

Introduction

While attempting to investigate how human computer interaction patterns can be utilized in understanding people better, potential of a correlation between the psychological distress and computer-user interaction patterns was also ascertained. The initial investigation was conducted as a questionnaire survey which was designed and administered with the objective of identifying the most frequented interactions of computer users when they are under stress. The questionnaire consisted of three sections; namely, General Information, Experience of Stress, and Reflection of Stress. Based on the responses given by the 756 computer literate undergraduates who participated at the survey [1], the mostly relevant stress-reflectors out of the ten (10) possible behaviors listed in the 'Reflection of Stress' section were selected. The mostly voted five activities were; *logging in to social networking sites, making typing errors, checking emails, scroll window up and down, and switching between tasks.* It is equally important to determine the magnitudes of contributions of these behavior parameters in defining the level of psychological distress. Therefore, three research actions were planned; (1) to record behavior parameters of individual users, (2) to identify the distress level of the user at the time; (3) to correlate the behavior parameter measures with the distress level of the user. Accordingly, the Stress Detector Framework (as shown in Fig. 1) was designed and developed in order to record behavior parameters and to correlate them with the distress level of the computer user [2].

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_60, © Springer International Publishing Switzerland 2015

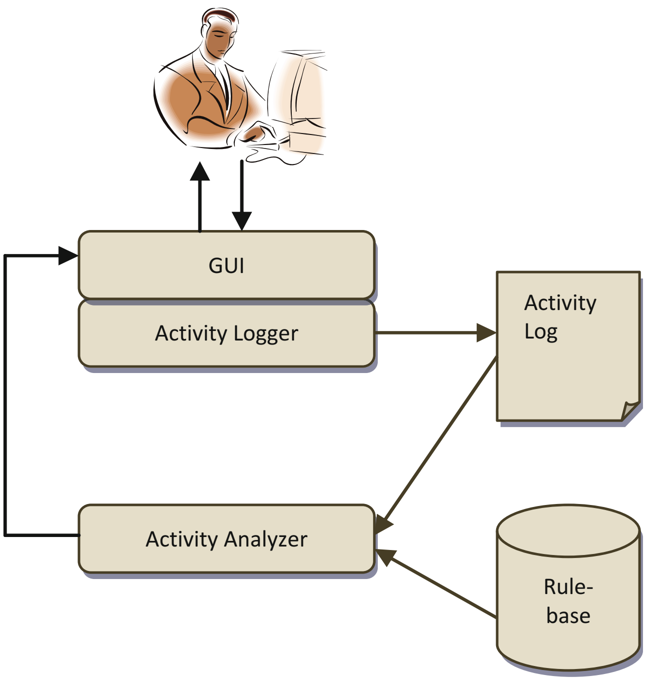

Fig. 1 Stress detector framework

The Activity Logger has been designed to gather four measures except the *Scrolling* measure, which was separately measured through the electronic version of K10 Distress Assessment Questionnaire. K10 distress assessment questionnaire [3, 4] is a widely accepted instrument for psychological distress assessment, and has been used to measure the distress level of the computer user at the time of behavior observation and recording.

Considering the possibility of deliberate alteration of natural behavior patterns [5, 6], the behavior measures were recorded unobtrusively, without informing the user upfront. In order to obtain the most regulated pattern of interaction [7] it was also planned to monitor the user at least for 30 min. Deliberate measures were taken in order to ensure the freedom of action of users, and not to disturb their ordinary interaction patterns due to the experiment. Towards the end of the session, the subjects were instructed to complete the electronic version of K10 distress assessment questionnaire, and recorded the *DistressScore* of individuals. The method of mapping between the *DistressScore* and the individual activity records is illustrated in the section "Data Preparation". Once the data was collected, the subjects were informed about the entire process, given the opportunity to withdraw from the study if they wish, informed about the withdrawal procedure even at a later stage, and written consent was obtained. The behavior parameter values and the respective distress scores were collected from a sample of 256 computer literate individuals. This sample was a subset of the larger sample considered at the initial data collection cycle.

The correlation between the distress score and the behavior parameters are expected to be instrumental in devising a mechanism to predict computer users' distress level based on their interaction patterns with the computers. Covering a single milestone in this long term objective, this paper presents the work performed in exploring the existence and henceforth the nature of a relationship between the distress score and the behavior parameters. Further, more attention is given in finding out nature of such relationship: linear, or non-linear. This exercise was initiated by assuming the relationship to be linear, and testing the dataset to qualify the assumption, failing which the testing for non-linear relationships to be set-out as described in the next two sections respectively.

Exploring Correlations

Methods

It has been noted that there exist several methods in identifying correlations between parameters, and to perform pattern matching and recognition. Among them two broad areas were considered; statistical methods, and machine learning techniques. Following sections will describe different techniques under considerations, outlining their applicability in the context of the work presented.

Statistical Evaluation

Correlation Analysis and Multiple linear regression (MLR) have been run on the dataset in order to reveal the existing correlations among the parameters, and to develop a regression model for *DistressScore* calculation.

Correlation coefficient is considered a measure of strength of the linear association between two quantitative variables. In other words, correlation quantifies the extent to which two quantitative variables, X and Y, associate with each other. Depending on the way the association is formed it will be identified as either positive or negative.

In multiple linear regression, a single response measurement Y is related to more than one predictor variables ($X_1 \ldots X_n$) for each of their observations. In other words, multiple linear regression attempts to model the relationship between two or more explanatory variables and a response variable by fitting a linear equation to observed data [8]. Every value of the independent variable X is associated with a value of the dependent variable Y. The function can be defined as:

$$Y = \alpha + \beta_1 X_1 + \ldots.. + \beta_n X_n$$

where α is the *intercept* and β_i are *slopes* or *coefficients*.

In the problem under consideration, there is a single response measurement '*DistressScore*' to be related with more than one interaction variables; Switching count (*Switching*), Count of Facebook visits (*FB*), Count of Email visits (*Email*), Error Percentage (*ErrPercent*), and Amount of scrolling (*Scroll*). Therefore, it is required to reveal the regression model to predict the Stress Score, given the interaction variables.

$$\begin{aligned} Distress\ \ Score = \alpha + \beta_1(Switching) + \beta_2(FB) \\ + \beta_3(Email)\ + \beta_4(ErrPercent) \\ + \beta_5(Scroll) \end{aligned}$$

Machine Learning

Machine learning techniques will identify the patterns in the data based on the observations presented. The learned model will then be applied to the real world scenarios/observations to predict the outcome. A system is considered 'learned' when it can take an input and provide an output based on the developed model.

Data Preparation

Mapping of Activity Record and K10 Questionnaire Results

The *Activity Logger* writes the values to a text file, with the time stamps for each session start and session end. It also recorded the computer name and the user name using which the login was done. In a single log file several session logs were recorded, and there were several such log files available containing activity records of different computers, may be by different users. These logs needed to be extracted into tabular format in order to prepare for evaluation. The extraction was performed using *ActivityExtractor* developed using Visual Studio.NET. In addition, the subjects of the study were to answer the K10 Distress Assessment Questionnaire made available on the computer, of which the answers are also recorded in a text file together with the computer name, user name, the time of questionnaire submission, the answers to each question and the final score. The Scroll measure captured via the K10 Distress Assessment Questionnaire interface was also recorded together with related information. The questionnaire results and the scroll measure were mapped with the corresponding activity log report.

Extraction and mapping of the records in the K10 results file to the activity log record were performed using the *ResultsMatcher*. If during a given session of computer use, the subject has not submitted the K10 distress assessment questionnaire, the activity record will not have a corresponding distress score, and a scroll measure. Such records were eliminated from the dataset, by keeping only the activity records with a corresponding K10 questionnaire results. The mapping process is illustrated in Fig. 2.

Dataset with Distress Score

For statistical approaches: correlation analysis and multiple linear regression, quantitative values are required, so that the *DistressScore* was kept in the dataset as the response variable. The structure of the dataset is shown in Table 1.

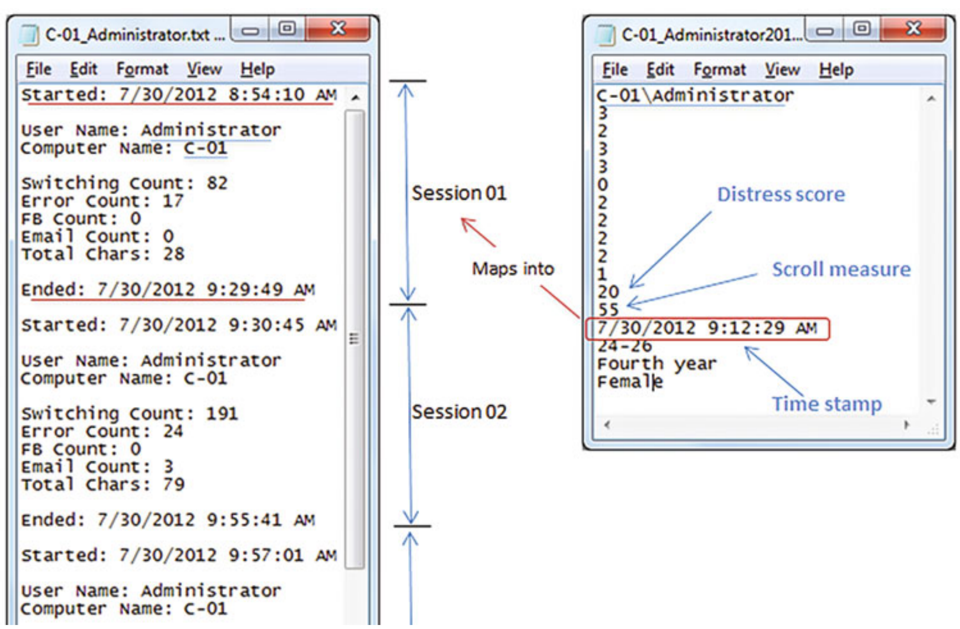

Fig. 2 Activity logger and K10 results mapping

Dataset with Class Labels

In order to support the machine learning approaches under clustering and classification, the dataset was then refined by replacing the *DistressScore* with respective class labels. Four ranges of *DistressScore* were identified: 10–19, 20–24, 25–29, and 30–50. Class labels 1, 2, 3 and 4 were assigned to the identified *DistressScore* ranges respectively. Accordingly, the dataset was created, and Table 2 gives the structure of the dataset.

Table 1 Dataset with distress score

Variable name	Data type	Description
Switching	Numeric	Predictor variable
FB	Numeric	Predictor variable
Email	Numeric	Predictor variable
ErrPercent	Numeric	Predictor variable
Scroll	Numeric	Predictor variable
DistressScore	Numeric	Response variable

Table 2 Dataset with distress class

Variable name	Data type	Description
Switching	Numeric	Predictor variable
FB	Numeric	Predictor variable
Email	Numeric	Predictor variable
ErrPercent	Numeric	Predictor variable
Scroll	Numeric	Predictor variable
Class	Enumerated {1, 2, 3, 4}	Response variable

Data Pre-processing

The nature of the problem is coupled with psychological distress, and there lies a great possibility for the dataset to contain some records of highly distressed individuals. Also, it is expected to observe unusual interaction patterns by those who are highly distressed. Therefore, outliers or unusual observation removal was deliberately suspended.

Testing for Normality

The normality of the dataset has been tested using three graphical methods: (1) Histograms; (2) Probability plots with a confidence interval of 95 %; (3) Empirical Cumulative Distribution Function (ECDF).

All these tests verified the data fits into the Normal distribution, and depicted in Figs. 3, 4, and 5 respectively.

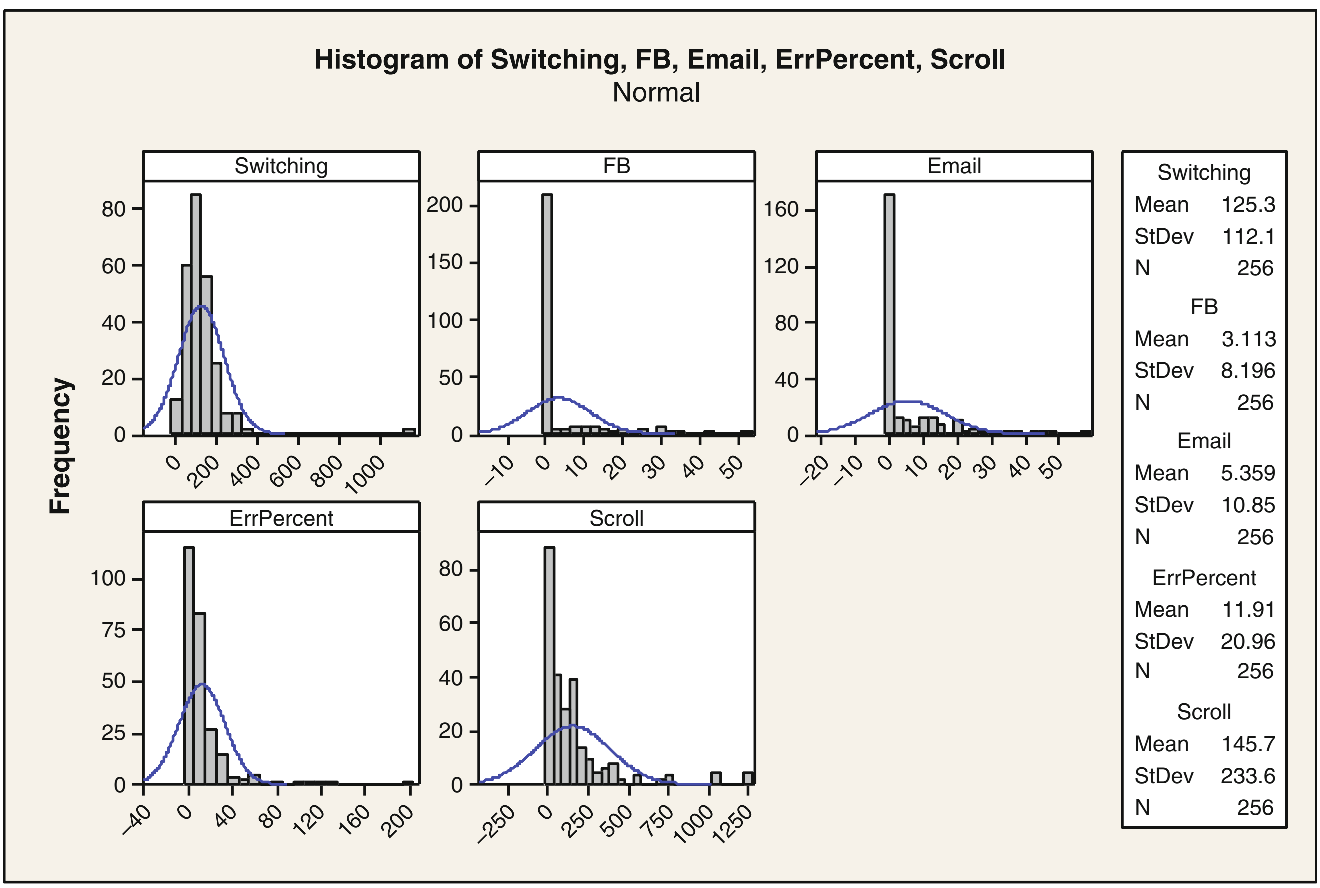

Fig. 3 Normality test using histograms

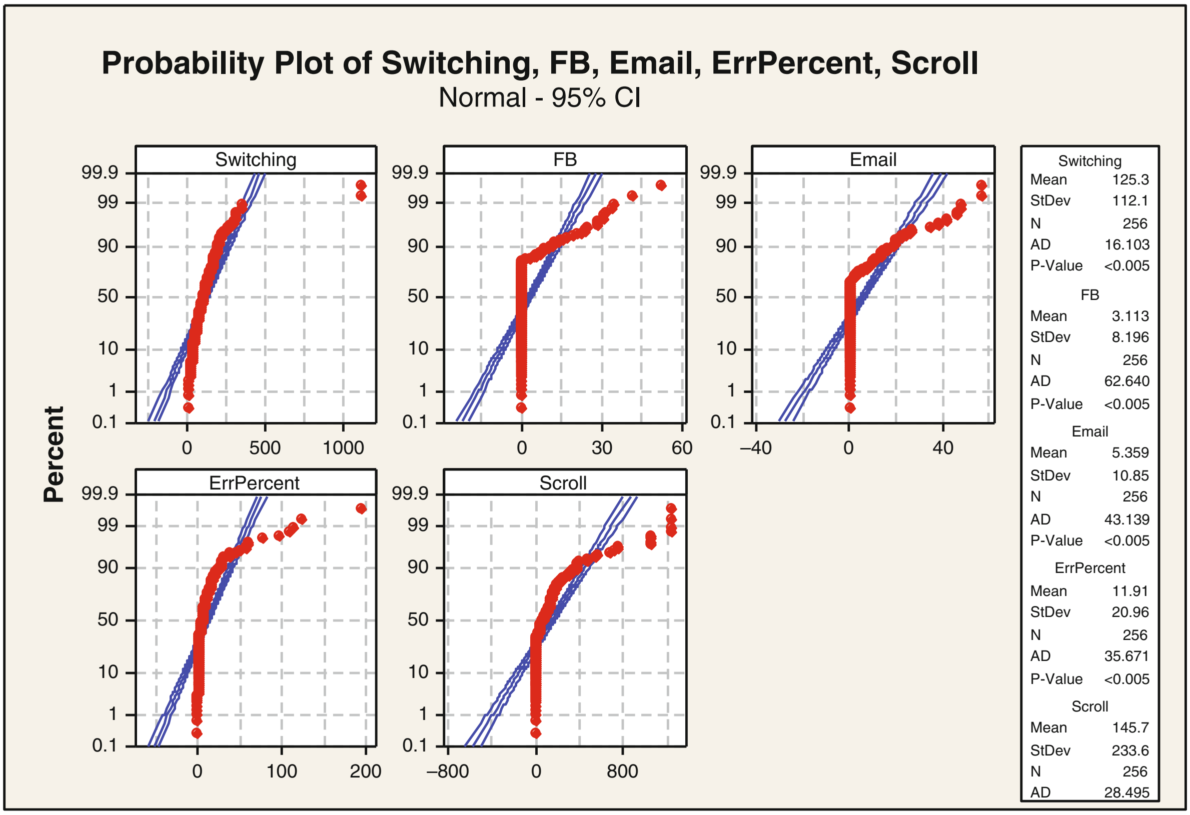

Fig. 4 Normality test using probability plots

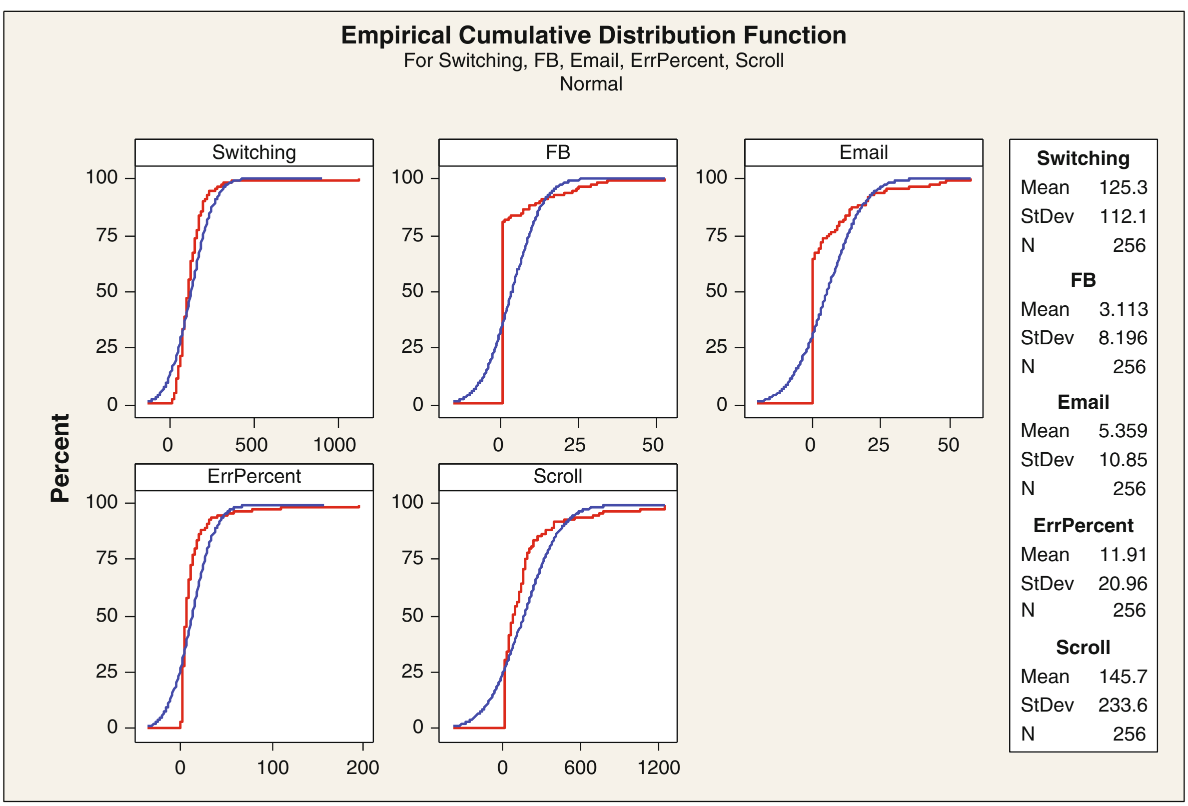

Fig. 5 Normality test using empirical cumulative distribution function (ECDF)

Evaluation

Correlation Analysis

The correlation between each predictor variable and the *DistressScore* was first tested by producing the scatter plots. As visible in Fig. 6, no correlation between the predictor variables and the distress score could be observed using these plots.

The dataset was further analyzed using correlation matrix, in order to reveal correlations which are not visible through scatter plots. Pearson correlation method was used, with the corresponding P-values. The correlation matrix of the dataset under evaluation is given in Fig. 7. Considering 95 % confidence level, the correlations of which the p-value is less than 0.05 were identified as significant, and marked using numbers 1 through 4 in darker circles. The correlation between *Email* and *ErrPercent* reports a p-value slightly higher than 0.05, but may provide a weak correlation. It is marked as 5, in a lighter color circle. Moreover, the only parameter which exhibits a direct correlation to the *DistressScore* is *FB*. The relationships of the other parameters to the *DistressScore* are built through *FB*.

Multiple Linear Regression

Having completed the correlation analysis, the next step was to construct the multiple linear regression model. It is expected to make inferences based on the observed associations between two or more variables related to the study. In other words, the dependent variable (*DistressScore*) can be predicted based on the values of the other independent variables (Switching, FB, Email, ErrPercent, and Scroll). The regression model provides further insight in to how the predictor variables can be used in predicting the output variable. In regression analysis, following assumptions were made.

1. Distress scores are statistically independent of one another
2. Distress score is proportional to independent variables
3. For a fixed value of independent variable, Distress score has a normal distribution
4. The variance of Distress score is constant for any X.

Analysis of Variance (ANOVA) table lists a p-value of 0.013, which is less than the normal 0.05 α level. Though the p-value gives a 'green light' to the model, the R^2 value is very small (5.6 %). The R^2 evaluates how well the model explains the dataset, and larger R^2 values are known to be better. Since the R^2 for the model is smaller (as shown in

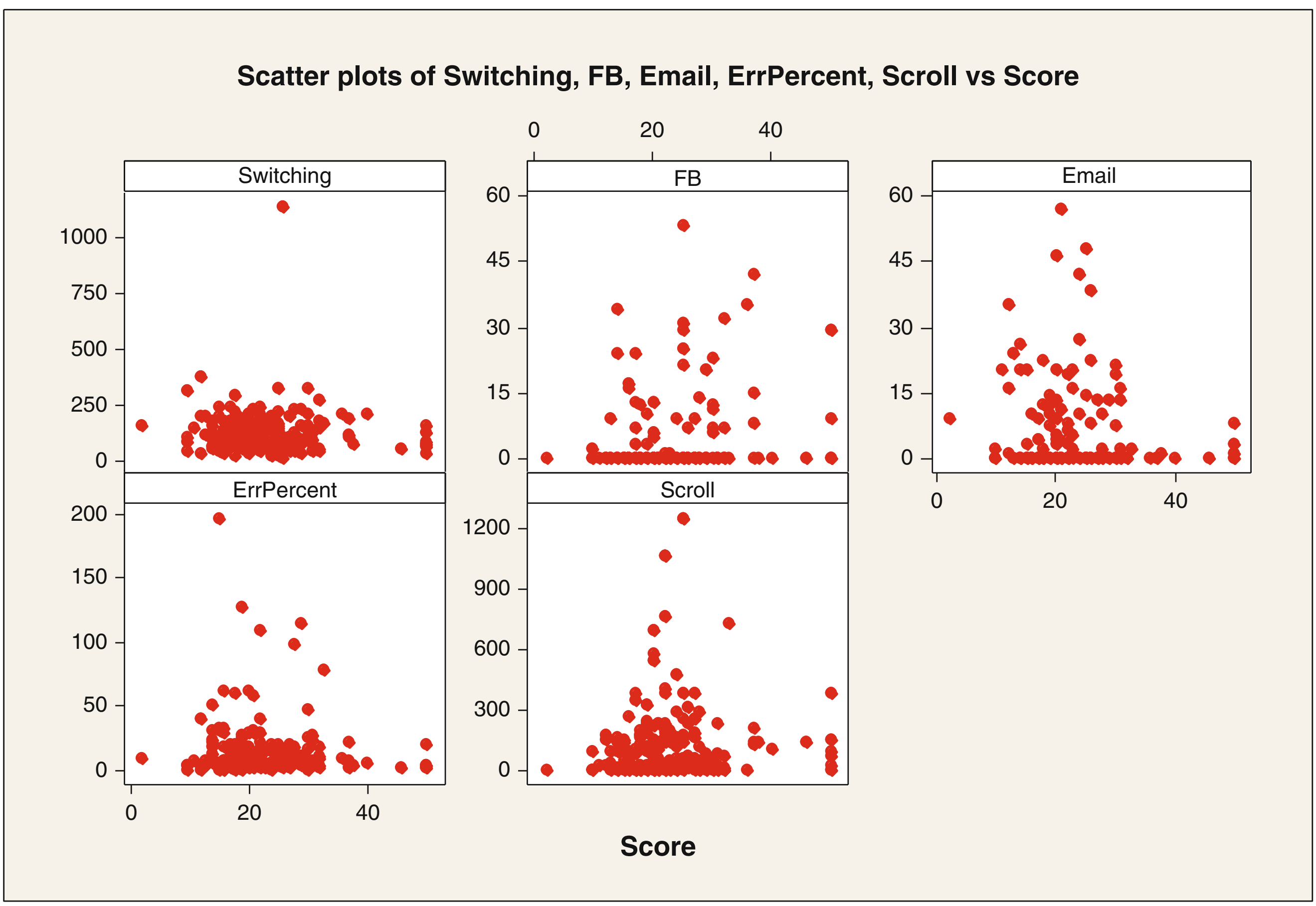

Fig. 6 Scatter plots between predictor variables and distress score

Fig. 7 Correlation matrix of distressscore and other interaction variables

Correlations: Switching, FB, Email, ErrPercent, Scroll, Score

	Switching	FB	Email	ErrPercent	Scroll
FB	0.087				
	0.164				
Email	0.490 ④	0.059			
	0.000	0.347			
ErrPercent	-0.014	-0.023	-0.117 ⑤		
	0.820	0.719	0.061		
Scroll	-0.028	0.205 ②	0.175 ③	-0.019	
	0.660	0.001	0.005	0.758	
Score	-0.061	0.184 ①	-0.098	-0.087	0.047
	0.330	0.003	0.119	0.165	0.453

Cell Contents: Pearson correlation
P-Value

Details of the Correlations

① Score and FB has a positive relationship (Correlation Coefficient = 0.184)
② FB and Scroll has a positive relationship (Correlation Coefficient = 0.205)
③ Scroll and Email has a positive relationship (Correlation Coefficient = 0.175)
④ Email and Switching has a positive relationship (Correlation Coefficient = 0.495)
⑤ Email and ErrPercent has a negative relationship (Correlation Coefficient = - 0.117)

Fig. 8 Outcome of regression analysis (At 95 % confidence level, $\alpha = 0.05$)

Regression Analysis: Score versus Switching, FB, Email, ErrPercent and Scroll

The regression equation is
Score = 23.5 - 0.00160 Switching + 0.183 FB - 0.0852 Email - 0.0372 ErrPercent
+ 0.00093 Scroll

Predictor	Coef	SE Coef	T	P
Constant	23.5087	0.8741	26.89	0.000
Switching	-0.001603	0.005178	-0.31	0.757
FB	0.18288	0.06249	2.93	0.004
Email	-0.08517	0.05448	-1.56	0.119
ErrPercent	-0.03717	0.02399	-1.55	0.123
Scroll	0.000928	0.002237	0.41	0.679

S = 7.96410 R-Sq = 5.6% R-Sq(adj) = 3.7%

PRESS = 16565.0 R-Sq(pred) = 1.39%

Analysis of Variance (ANOVA)

Source	DF	SS	MS	F	P
Regression	5	940.99	188.20	2.97	0.013
Residual Error	250	15856.72	63.43		
Total	255	16797.71			

Source	DF	Seq SS
Switching	1	62.80
FB	1	604.72
Email	1	111.31
ErrPercent	1	151.25
Scroll	1	10.92

Fig. 8), the robustness of the model is not acceptable. In addition, the regression analysis has identified 35 unusual observations accounting for about 13.6 % of the entire dataset.

So that another round of regression analysis was carried out by removing the unusual observations from the dataset. Though R^2 value has only slightly increased (by 0.8) after taking refining measures, it is still not significant enough to recognize the regression model as robust.

Prelude Evaluation of Machine Learning Techniques

It was initially assumed that the ideal case would be to perform unsupervised learning with the dataset. However, as the number of clusters existing is pre-defined (four classes of distress), it was decided to use supervised learning techniques. Hence, the K-Means clustering was applied, by setting the number of clusters to four (04). Setting the number of clusters will in other way support to reduce the amount of vulnerabilities associated with the formation of clusters. The clustering algorithm was trained with a training set of 200 records, and tested using a test set of 56 records. At the training phase, four (04) clusters have been identified. During the evaluation with the test set, only three (03) clusters were recognized.

Conclusion

During this evaluation it has been revealed that Multiple Linear Regression (MLR) is not suitable to predict the *DistressScore*. Even though MLR has been very widely used in many different contexts, the dataset under evaluation does not fit in to a linear model in an effective manner. Therefore, it is required to identify techniques which are capable of handling complexities and uncertainties of real world scenarios, and vulnerabilities attributed to affective dispositions. As machine learning techniques are reputed as suitable in real world scenario, steps were taken to evaluate some existing machine learning techniques, and to find out their suitability to match the context better.

Preferably, unsupervised learning algorithms would give the full freedom to the learning algorithm to identifying the invisible patterns if available, and to cluster the instances accordingly. This method makes it possible to eliminate self-reports of the subjects about their level of stress, which is not considered highly reliable. Reliability of self reports is challenged due to three main reasons: (1) self-reports may be biased to the individuals' characteristics; (2) subjects can deliberately report their-selves incorrectly; (3) subjects may have forgotten how they felt at the time they were asked.

However, with the presence of the class labels, and due to the diversified nature of the dataset, it was also noticed that supervised learning methods provide better results than the others.

Collectively, the study has revealed that the relationship between the human computer interaction patterns and psychological distress score does not fit in to linear model, and henceforth the evaluation should focus more on machine learning techniques. Even among them, supervised learning algorithms seem to be performing better. But this needs further evaluations before making a confident judgement.

Acknowledgment We would like to acknowledge the Ethical Review Committee of Faculty of Medicine, University of Colombo for reviewing and approving the ethical clearance for the questionnaire developed; The director of University of Colombo School of Computing, and the deans of the two faculties, Faculty of Medicine, University of Colombo, and the Faculty of Information Technology, University of Moratuwa for granting permission to administer the questionnaire in their institutes; The academic members for the support extended in administering the questionnaire among the students, and finally the students who have participated in this survey voluntarily.

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Leading Innovation in Universities: From Practice Ahead of Practice

Kristina Zgodavova and Matus Horvath

Abstract

The purpose of this paper is to provide information to support a change in the curriculum and promote the inclusion of new advances in study programs. The resources of this paper come from the shared experiences of the authors working in strategic management at the university, and also from audits from the Slovak Higher Education Institution (HEI) and several projects in the field of education and innovation. The Paper describes a project to develop an advanced general method of improving and innovating study programs according to the expected development of advances in the field. The overall findings have shown study programs required change, it is appropriate to use the Study Program Leading Innovation (SPLI) model and apply it to the creative existing curriculum. The diversity of ideas was and still is limited about the quality benchmark and how to achieve change and improvement.

Keywords

Creativity • Laboratory education • Quality engineering and management • Cloud learning

Introduction

Research and innovation are two of the most important priorities of Europe's 2020 strategy, which puts research and innovation as the pillars of economic growth and development [1]. Organizations evolve and remain competitive only through innovation.

To formulate a sufficiently and specific mobilization concept of innovative education, that will be accepted by the scientific and pedagogical staff of Higher Education Institution (HEI), requires the knowledge of modern trends in the area.

Universities are expected to be a source of new ideas and information. But in reality universities often have trouble keeping pace with evolution their own fields of study. There could be many reasons why, however let's focus on four main points:

– The slow introduction to new concepts;
– Communication gaps and shifting priorities;
– A lack of funding;
– The verification or validation of new or modified study programs is too slow.

According to Merriam-Webster dictionary: Being an advanced means to be ahead of others in progression or idea and leading means providing direction or guidance [2].

The products of universities are study programs; research projects outcomes and publications as a result of their dual research-educational process [3]. These, however, often lag behind the needs and expectations of the future employers of graduates. Offered study programs are presented in curricula. To create such study programs that prepares students in advance to be innovative requires prognostic and creative thinking, modern behavior and flawless performance.

Therefore the HEI should be ahead of the practice and the practice should constantly innovate (of course within the limits of corporate social responsibility).

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_61, © Springer International Publishing Switzerland 2015

The purpose of this paper is to present factual and generalized information for advancement and curriculum change according to needs, requirements and expectations associated with current and modern practices in the field.

The research methodology is based on key words text mining and clustering. The results are presented in the form of:

– Clouds of most frequently used keywords in four areas: (a) provided study programs, (b) student results—the final theses, (c) the results of academics—publishing's outputs and projects, (d) the professional requirements of practice and trends of innovations;
– Simplified methodology, which we named 'Study Program Leading Innovation' (SPLI).

Social implications of results lie in the increasing of the competitiveness of universities and improving the loyalty of employers of graduates, and other interested parties.

The originality of the solution is in the approach to research and related research methodologies. Critical success factors of the SPLI model are:

– Creating innovative and creative research team, involving all stakeholders (especially future employers)
– Reliable assessment of the current state of knowledge and the development trends in the practice (prospective employers of graduates),
– The creation of the initial idea of the optimum (benchmark) desirable outcome of the current state of the solution, its comparison with the recognized state and factual, resource and chronological planning of the SPLI project.

This paper is intended for HEI executives, academics and professional practitioners who are involved in changing and innovating study programs.

Methodology

The development of a new curriculum and its accreditation often takes several months and with staff and instrumentation preparation, testing, start-up and completion of the first graduates, it could take several years.

HEIs have the opportunity to partly change the curriculum and update content after accreditation, but there is always the question of which way the practice will move and what skills for innovation will graduates need.

Europe's future economic growth and jobs increasingly will have to come from innovation in products, services and business models. This is why innovation has been placed at the heart of the Europe 2020 strategy for growth and jobs [4].

The EU 2020 strategy sets up a basic research orientation with a detailed creation of curriculum, close monitoring of development trends (technology) and collaboration with the practice is necessary, especially with top experts—innovators in the field.

In the framework of Education, Research and Innovation knowledge triangle universities have a crucial role to play in creating knowledge and translating it into innovative products and services, in cooperation with research centres and businesses [5]. There is a range of mechanisms by which universities can contribute to regional innovation systems [5]. One of the possibilities is that universities and businesses directly cooperate in curricula design and curricula delivery to ensure that graduates has the right skills and transversal competences [5, 6].

Within the research project ESF Technical University of Kosice (TUKE) ITMS 26110230070 'Elements package for improvement and innovation in education at Technical University of Kosice' we have created a general 'Study Program Leading Innovation' methodology, which meets the criteria in terms of achieving innovation ahead of the industry:

"*– from praxis ahead for praxis –*"

The starting point is a collaborative network of teams from Slovak universities and representatives of employers to generate ideas for advance innovation of study programs. The methodology is being tested on the study programs in the field of 'quality' on six technical faculties of Slovak universities. The methodology can be suitably generalized and used to advance innovation of courses also in other disciplines.

The research methodology is based on searching keywords and subsequent visualization by tag clouds. The research was conducted in the following steps:

Selection of Electronic Databases for Research on the Subject

– PortalVS of Slovak HEIs [7]: collects information about curricula; teachers and researchers outputs: publications and projects.
– Portal Profesia SK [8] and Profesia CZ [9]: collects information on labour supply and demand home and abroad.

Note: We have also investigated Czech portals because of minimum language barrier and the fact that approx. 10 % of Slovak HEI graduates in the field of 'quality' profession are looking for work in the Czech Republic.

Search Criteria Based on Portals Offer

– The title [corresponding to area of research 'Engineering and Technology' (SK classification No 17), study branch 'quality' (SK classification No 5.2.57)]; keywords and content according to the study program.
– The title and keywords of final theses (BSc., MSc. (Ing.), and Ph.D. of university study).

– Authors of publications (name according affiliation to workplace, which guarantees and provide the content of the curriculum), the titles of their publications and keywords (monographs, textbooks, Current Content, Web of Science, Scopus databases and other scientific publications in English language).
– The name of the profession according to the International Labour Organization Classification of Professions [10] and actual databases of job seekers; most offered professions; higher education institutions; study programs and courses; scientific publication in the area of 'quality'.

Fig. 1 Tag cloud for curriculum from TUKE, Department of Integrated

Fig. 2 Tag cloud for theses of students from TUKE, DIM in the field of 'quality' study program

Fig. 3 Tag cloud for educators, researchers and students publications from TUKE, DIM in the field of study 'quality' study program

Creating Clouds

According to [11] "*A tag cloud (word cloud or weighted list in visual design) is a visual representation for text data, typically used to depict keyword metadata (tags) on websites, or to visualize free form text. Tags are usually single words, and the importance of each tag is shown with font size or colour*". In use, tag clouds can evolve as the associated data source changes over time. Discussions around tag clouds often include a series of tag clouds and consider how they evolve over time. However, since tag clouds do not explicitly represent trends or support comparisons, the cognitive demands placed on the person for perceiving trends in multiple tag clouds are high [12]. Therefore, the results of our research will be presented in future, 1 March, 2014 by Parallel Clouds, which integrate Lines [13] into a tag cloud to convey trends between multiple tag clouds [12].

Comparison Within Clouds

The comparison is realised within clouds in categories: curriculum, theses of students, outputs of academicians and SK and CZ profession portal, Slovak and Swedish expert prediction, and selected Scopus journal.

Visualization and Discussion

Visualization

We used tag clouds for visual representations of social tags, displayed in paragraph-style layout (in this case not in alphabetical order) where the relative size and weight of the font for each tag corresponds to the relative frequency of its usage. The data used as input for our tag clouds are social tags (the unstructured keywords and annotation of information created by authors using short textual labels known as 'tags') [14]. Research results in the form of tag clouds are presented in Figs. 1, 2, 3, and 4.

Note: Requirements of employers on graduates' qualification for positions in 'quality' by International Labour Organization SK ISCO-08 [15] from the databases 'Profesia SK' and 'Profesia CZ' (see Table 1):

Discussion

A comparison of the ten most frequent words as a TUKE DIM outputs, and a summary of three SK HEIs are presented in the Table 2. Comparison of praxis requirements, experts' predictions for a period of 20 years of three Slovak and three Swedish universities [16] and Quality Innovation Prosperity Scopus journal are presented in the Table 3.

Fig. 4 Tag cloud requirements of employers on graduates' qualification for positions in the field of 'quality'

Table 1 Requirements of employers for position in the field of 'quality'

	Number of demands	
	Date: November 10, 2013	
Title of profession	Profesia SK	Profesia CZ
Quality Engineer	111	48
Quality Controller	102	59
Quality Manager	67	15
Quality Planner	49	17
Director of Quality	8	3

Model and Procedure of Study Program Leading Innovation

SPLI Model

Based on the knowledge gained from the research we have created a graphical model (see Fig. 5) and the methodology for leading innovation of study programs.

Procedure of SPLI

1. Evaluation knowledge of the status and trends of practice development
   1.1 Creation of initial ideas about study program curriculum based on the practice.
   1.2 Checking the actual state and development trends of the existing curriculum and its comparison with initial vision.
   1.3 Creation of the change concept (innovation) for program of study or revolutionary change (i.e., creating a whole new program of study) based on leading innovation in the area and time.
2. Planning and designing changes to existing or a new program.
   2.1 Creating a plan in accordance with the resources and constraints of HEI stakeholders (internally and externally—the Accreditation Committee and the practice)
   2.2 Designing a model and starting the SPLI project.
   2.3 Vindication or if needed a correction of the project of the change to the existing curriculum or a new program before stakeholders (internally and externally—the Accreditation Committee and the practice).

Table 2 Parallel Cloud displays a Gradient Line that Links the Same Word Occurring in Multiple Tag clouds

Curriculum		Theses of students		Outputs of academicians	
(Retrieved: September 15, 2013)					
TUKE DIM (Fig. 1)	Summary SK HEI	TUKE DIM (Fig. 2)	Summary SK HEI	TUKE DIM (Fig. 3)	Summary SK HEI
Systems	Management	Organisation	Management	Management	Properties
Methods	Technical	Management	System	Production	Measurement
Processes	System	Process	Process	Environment	Production
Statistics	Material	System	Production	Process	Management
Environmental	Technology	Risk	System	Control	Process
Information	Production	Security	Analysis	System	Application
Safety	Engineering	Information	Design	Safety	System
Tools	Informatics	Assessment	Methods	Model	Automotive
Integration	Statistic	Analysis	Safety	Parameter	Evaluation
Metrology	Integration	Application	Statistics	Integrated	Material

Note: Ranked according to the number of occurrences in the searched batch

Table 3 Presentation of the ten most frequent words in the 'Profesia SK and CZ' databases, Slovak and Swedish expert analysis and QIP Journal

Profesia (SK and CZ). Date: November 10, 2013	Expert prediction (SK) 3 Slovak Universities. Date: February 15, 2013	Quality Innovation Prosperity (Scopus) Journal www.qip-journal.eu	Expert prediction Swedish Universities. Date: August, 2012
Process	Leading	Improvement	Development
Product	Creative	Innovation	Customer
Production	Sustainability	6 sigma	Focus
Management	Service	Management	Improvement
Control	Intelligent	Efficiency	Sustainability
Implementation	Measurement	Success	Business
Customer	integrated	Social value	Integrated
Documentation	System	Critical factor	Continuous
Compliance	Self-learning	Process map	Service
Audit	Mobile	Strategy	Network

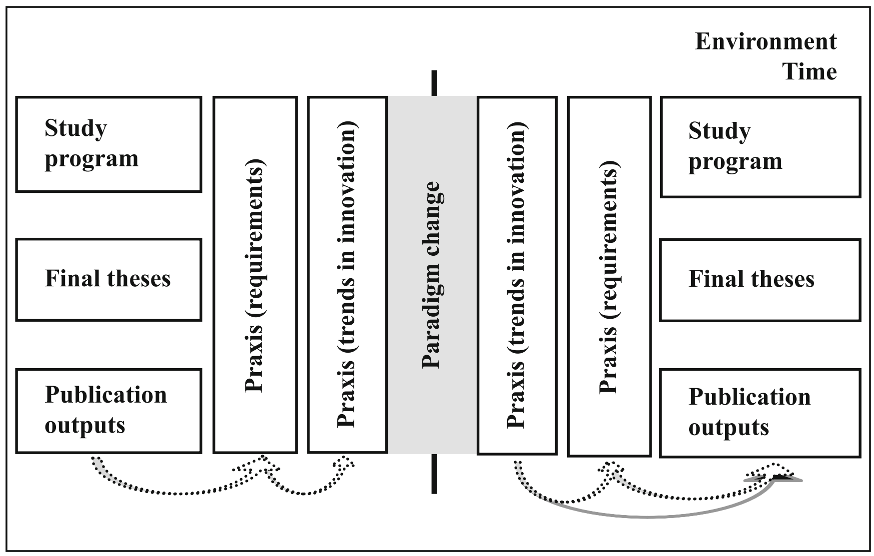

Fig. 5 Graphic model of paradigm shift of curricula preparation and their content

3. Realization of changes in the existing curriculum or creation of new curriculum.
   3.1 The realization of organizational and personnel changes.
   3.2 The realization of technical changes and changes in management and implementation of key (research and training) and secondary (administrative) processes.
   3.3 Design, start-up and normal process of study program, including the creation of documentation and study aids, laboratory training and interaction with the practice (especially) for students theses.
4. Evaluation knowledge of a new state and monitoring of development trends of the improved or new study program.
   4.1 Determination of actual status against the plan in documentation (records).
   4.2 Designing and implementing corrections in all monitored areas.
   4.3 Summarization of knowledge for SPLI knowledge management.

Conclusion

Whoever innovates in advance wins. Universities that prepare graduates for this trend can win in the competition from other universities and their offers of study programs.

Creativity has an essential role to play in education, whether for the purposes of enhancing technical innovation or for creating well-rounded graduates who can truly contribute to society [17].

From a more detailed comparison of the study program, final theses and publications of academics of TUKE DIM with other study programs in Slovakia in the field of 'quality engineering and management' follows that the curriculum content is about the same and differs only in details of the subjects depending on the general technical basis of Faculty to which the program is implemented. For example Slovak University of Technology Bratislava—automotive industry and materials; Slovak Agriculture University—agricultural machinery and information's systems; Technical University

Table 4 Future research keywords in the field of quality

Leadership
Creativity
Improvement
Innovation
Development
Customer
Quality renaissance
Experience economy
Experience quality

of Kosice—metallurgy, foundry, engineering, automotive industry, and mining.

From comparison of the requirements of practice the knowledge and skills of graduates follows, that HEI provide sufficient background for the positions listed in the Table 1, but according to the prediction of experts, it would be useful to include in a curriculum subject/s on creative thinking and innovative behaviour, to extend the selection of subjects relating to the quality of services and develop the students' ability to think in terms of leadership.

From the Table 3 and thematic topics of scientific conferences in 2013, and planned topics of conferences for the year 2014 we can conclude that the future research can be presented with keywords from the Table 4.

To achieve leading innovation in study programs it is required:

– To design and implement the SPLI project and manage information for leading innovation of study programs in the field of study 'quality engineering and management'.
– To continuously benchmark products i.e. study programs and curricula as well as research-learning processes versus the best domestic and foreign study programs in the field of 'quality engineering and management'.
– Theses should be focused on research projects and innovations in practice.
– To monitor trends of innovation in the field of application of knowledge and skills of students.

Acknowledgements This work was supported by the ESF Technical University of Kosice (TUKE) ITMS 26110230070 'Balík prvkov pre skvalitnenie a inováciu vzdelávanie na TUKE/Elements package for improvement and innovation in education at Technical University of Kosice'.

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Implementation of Variants in Component Based Software Production

Yusuf Altunel and Abdül Halim Zaim

Abstract

We created and implemented an algebraic framework to automatically generate component variants out of specifications. In this paper we provide the definition of this approach with its operators and operands, and show how expressions can be parsed, translated and processed for production of component variants. We created a Java application which is widely based on sets, to show the approach can be implemented properly and efficiently.

Keywords

Component • Set • Variant • Java • Component based development (CBD) • Algebra • Formal approach

Introduction

Component based development can solve certain problems of software production and assist to make enhancements in the industry. Components are defined as the software units encapsulating data and methods, having simple and well-defined interface, exhibiting functional specialization, and designed for expectation of reuse [1].

CBSE offers high level of software reuse and has potential of reducing the development cost and time-to-market of software system. Component-based software development methodology would result in productivity improvement between 40 and 400 % and ten times higher of quality in means of defect density measurement [2]. CBSE increases the overall quality of the software system as a result of strict application of quality issues and increasing the standardization. Additionally, it helps to improve the maintainability of the system by replacing the old components with better ones [3].

In this paper we basically concentrate on description of components considering the variation and produce the variants when it is necessary without the need to redefine the commonalities. In our approach we keep the system out of inheritance to prevent the long inheritance hierarchies which produce unpredictable side effects and damage the black-box and individual deployment of components. Additionally, inheritance mechanisms generally reduce the ability to control the containment of features in components, which in return results in unnecessarily chubby components just to make coding easier.

Formal approaches and languages yield the theoretical background needed to create a formalism to describe components out of their features. We used algebraic approach to define the complex relationships between common and different features of a component. In this approach, a component defines a set of features in some cases will be included but not in other cases. So, we can define component features and possible options based on reuse patterns and specific requirements of domains and applications. Additionally, in this way we can guarantee the containment of just the features required in certain reuse of the component,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_62, © Springer International Publishing Switzerland 2015

which in return helps to increase the efficiency of overall system and enhance the maintainability. It is possible to automate variant production out of formal descriptions of such feature relationships.

In this paper we will describe our algebraic approach used to describe the components as set of features, the relationships between these features and how to translate these descriptions into set of variants that can be produced from that component. We will explain the language provided to express the algebraic expressions, the semantics of those expressions and the techniques used to iteratively translate those expressions into proper software units that can be implemented in typical programming languages such as Java.

Component Based Development and Variants

Component Based Development (CBD) is based on constructing software using ready to use software units called components, instead of inventing the wheel again and again. Building software systems out of reusable units is complex since we must consider both the production of appropriate components and generation of the whole system best utilizing such units at the same time [1]. Therefore the methodology must include formal processes to guide the life cycle of the project in defining the team roles, deliverables, and the techniques in solving the problems [4].

A component is accepted as a unit of reuse and integration [5]. A more detailed definition would be "software units encapsulating data and methods, having simple and well-defined interface, exhibiting functional specialization, and designed for reuse expectations in a generic context" [1]. The methodology for Component Based Software Engineering (CBSE) should be '*flexible*' that can be adaptable according to actual needs of the project [6]. Providing the functional variation and the variability is a key element in success of reuse [7].

Certain researchers recommend solving component adaptation problem by using object-oriented inheritance [8]. Inheritance is a very smooth and straightforward way of sharing the common features [9, 10]. Unfortunately, strong dependencies make the derived classes hard to maintain, as well as the component compilation painful. Additionally, low-end products require a just restricted sub-set of the functionality and they pay for the unused functionality in terms of code size and complexity that traditionally implemented components provide [12].

Adaptability might be increased if more features to be used in various applications can be added to the component, but this the scarification of performance and maintainability for the sake of reusability and adaptability [13]. Object oriented approach to component production results is in having unneeded extra features come from the class hierarchies, in return to the reduction in the efficiency caused by extra resource usage, and damaging the code modifiability as a result of complexities in locating and correcting the problems. Additionally, long class hierarchies are hard to understand and package as single units.

Alternatively, producers can construct applications by using component variants to increase the functional variations, as well as assist the system producers to select the best-fitted units with a minimum cost. We can see that the application boundary coincide with the boundary of the component variants, since we know that only those variants that exactly match.

Variants, Variation and Variability

Variants are specific components containing just one specific set of features in order to implement the specific versions of components Variants can become very effective in component customization process when the variant production is automated by means of variant generators [11]. Variants seem to be a straightforward way of satisfying the need for variation in component based development.

Variation is defined as the ability to configure, customize, extend, or change a product when to use in a specific context [14]. In one of our last studies, we investigated certain influences, or abilities in obtaining the software variability, including adaptation, configuration, customization, evolution, extension, modification, portability, scalability, suitability, tailoring, and upgradability [15].

In reverse to classical and concrete software items, variability is especially needed in new approaches which, share more flexible structures with delayed design decisions for functionality and quality [16]. Product-Line approaches especially put emphasis on Variability Management (VM), to identify and manage the commonalities, variations and dependencies across certain levels [17].

Variation Capacity

Variation Capacity, VC, is a metrics defined to calculate the number of possible variation that can be obtained out of features when there are certain relationships defined between them by some means of operators. There are certain relationships in the scope of component that influence the number of possible combinations and in return the variation capacity positively such as optionality, exclusion, OR, whereas there are others influences negatively such as dependencies. Actually, optionality in containment of features provides an exponential order of growth in variation, whereas dependent features should always exist to happen in the same variant, which in return eliminates certain options.

Here the relationships including the optionality might hardly be effected from the domains and application requirements and as more domains are in the scope of the components.

Generative Programming provides a diagrammatic representation of the feature relationships, which is called as "feature diagram". In this means, mandatory, optional, alternative (exclusive), and or-features can be represented using these diagrams. Simple edges with filled ending with filled circles represent the "mandatory features", simple edges ending with empty circles are for "optional features", edges connected by an arc are used for "alternative features", and finally edges with filled arcs are used for "or-features" [18]. The variability in product-line is especially critical to build and maintain products over a specified period of time to maximize the return on investment (ROI) and it is supported with the "variation mechanism" which helps to control the adaptation and support of product-line development for benefiting from the similarities that exist between similar applications [15].

When the options are analyzed, even a simple component with restricted amount of features will have many combination alternatives based on complex relationships among the features. So, manually, it is really a hard job to analyze the relationships and explore those alternatives correctly and in efficient manner. Using software based automation helps to make the whole process manageable. Additionally, code automation techniques can be used to translate the variant descriptions into actual source code written in the programming language preferred. Based on this strategy, it would be very easy to obtain a specific combination of features to satisfy very specific requirements in production of software systems without making vital mistakes such as omitting depending features in generation of variants automatically.

Formal Approaches

Basically formal approaches are used in locating the software defects, description of structural properties, model checking, automated test production, verification, consistency checking, investigation and dealing with complexities [19].

In this study a special care is given to the formal techniques, since they are critical in establishment of engineering studies. Actually, there are certain claims about formal approaches abilities to increase the quality of software, simplify the development processes [20, 21]. With formal approaches, it is possible to automate code production out of descriptions, on the fly generation of systems at run time, description of configurations for reusable architectural structures and verification objectives [22]. In fact, the feasibility of using formal approaches in the field is still in discussion and without proper tools and methods, mathematical approaches are hard to create, use and implement in software engineering.

Formal Languages

Formal languages theory is accepted a joint area of study for computer sciences and mathematics. Languages are main concerns in definition and understanding of things, otherwise expression of facts and description of complex relationships would be impossible. Our currently used programming languages seem to provide no more than classes to define the entities and the relationships between those entities [23].

Chomsky defines four different languages and grammars [18, 24]. Two of them, context-free and regular grammars are used in definition and processing of programming languages. Regular grammars are used to define the basic elements of the programming languages, whereas the context-free grammars are used to define the syntax of those languages. Backus-Naur Form (BNF) which is firstly described by John Backaus, but later modified by Peter Naur, is widely used in description of programming languages [25]. Today, it is standardized as ISO/IEC 14977 and extended with the name of EBNF (Extended BNF).

Formal Method

Formal methods have mathematically defined syntax and normally cannot be used in software engineering without properly translating into software applications [26]. Functional and behavioral properties, scheduling, performance, real-time restrictions, security principles, internal characteristics can be formally defined [27]. Complexities, low level of formalism, strict structures can prevent the acceptance of a language in the software industry [28].

The problem in modular designed systems seems to be the strategy to convert the software starting with conversion of previously defined abstract definition of structures. Later, the derived patterns can be converted into algebraic module definitions, but before that it is necessary to construct reusable libraries [29].

Developers of real-time critical systems feel the thread of ill described specification seriously and therefore they are in search of filling the gap between the system engineering and standard programming languages [30].

It is necessary to include team roles, production methods and techniques to move from the classical development to component based model [4]. At the beginning of development, component groups are uncertain, ambiguous, complex and inadequate. To solve these problems, formal approach can help to remove the missing points, and correct the wrong definitions [27].

Definition of Formalism

Assume $\boldsymbol{\Psi}$ is the set of all elements, $E_i \in \boldsymbol{\Psi}$ is a set of some of elements or a category, $e_{ij} \in \mathbf{E}_i$ is an element in this category, $\boldsymbol{\Phi}$ is feature set that can be assigned to these elements, $f_k \in \boldsymbol{\Phi}$ is any feature, $\boldsymbol{\Delta}$ is the operations sets defined by the algebra, $\boldsymbol{\delta}_i \in \Delta$ is any operation, and $\boldsymbol{\Gamma}$ is the operation-element element combination. We define an algebra $\boldsymbol{\Lambda}$ as follows.

$$\boldsymbol{\Lambda} = \langle \boldsymbol{\Psi}, \boldsymbol{\Phi}, \boldsymbol{\Delta}, \boldsymbol{\Gamma} \rangle$$

$$\boldsymbol{\Psi} = \{E1 \cup E2 \cup \ldots\} = \{e1, e2, \ldots\}$$

$$\boldsymbol{\Delta} = \{\delta 1, \delta 2, ..\}$$

$$\boldsymbol{\Gamma} = \Sigma ei\delta jek \cdots .$$

$$\boldsymbol{\Phi} \subseteq \boldsymbol{\Psi}$$

We define our algebra based on this definition as follows:

$$\boldsymbol{\Lambda} = \langle \boldsymbol{\Psi}, \boldsymbol{\Phi}, \boldsymbol{\Delta}, \boldsymbol{\Gamma} \rangle$$

$\boldsymbol{\Psi}$ = {Entity, Application, Annotation, Package, Class, Component, Variant, Interface, Feature, Repetition, Attribute, Method, Singature, Parameter, Process, Body, Aspect, Comment, Value, Message, ...}

$$\boldsymbol{\Delta} = \{!, +, -, |, \$, *, /, \#, <, >, \{\}, []\}$$

$$\boldsymbol{\Gamma} = \Sigma e_i \delta_j e_k \cdots \ldots$$

$\varepsilon\delta e = e\delta\varepsilon = e$, $\varepsilon \in \boldsymbol{\Gamma}$ is identity element

$$\Phi \subseteq \boldsymbol{\Psi}$$

Entities

An entity is a structural element of the algebra and we include the common entities of software engineering to cover as much approaches as possible, such as component, aspect, generative, feature oriented approaches.

Component

A component can be defined as a vector of features and the relations defined between these features:

$$\mathbf{C} = \langle \mathbf{F_c}, \mathbf{R_c} \rangle, \mathbf{F_c} = [\boldsymbol{f_{c1} f_{c2} \ldots f_{ci} \ldots f_{cn}}], \quad \mathbf{R_c} = [\boldsymbol{r_{c1} r_{c2} \ldots r_{cj} \ldots r_{cm}}]$$

$$\mathbf{i, j, m, n \geq 0} \tag{1}$$

r_{cj} is a j^{th} relationship defined between features of C and if $i = j = m = n = 0$ is e, C = is empty component and can be represented as C.

Feature

A feature is a property of an entity that is included by that entity, such as method, attribute, etc. in object oriented approach.

$$A = \langle \text{Visibility, Type, Name, Initial Value} \rangle \tag{2}$$

These elements are straightforward, so their definitions are not included here for the space reasons.

Similarly, method can be defined as the visibility, return type, name, parameter list, and body.

$$M = \langle \text{Visibility, Return Type, Name, Parameter List, Body} \rangle \tag{3}$$

Return type, name, and parameter list can be further categorized as the signature which is used by various programming languages to differentiate methods from each other. There are various approaches between the programming languages in definition of the signatures of the methods.

$$\text{Signature} = \langle \text{Return Type, Name, Parameter List} \rangle \tag{4}$$

So, we stop further to formally define these elements here, since they are straightforward and space reasons.

Variant

We can define a variant of C_m as a vector of features included by component C_m.

$$F_i = (f_1, f_2, \ldots, f_i), V(F_k) = V_k = (f_1, f_2, \ldots, f_i), \quad f_1 \neq f_2 \neq \ldots f_i \text{ and } f_1, f_2, \ldots, f_i \in C, \, i, k \geq 0, \; i \leq |C_m| \tag{5}$$

To show V_k is a variant of C_m we use the expression $V_k \sim C_m$.

Accordingly, the component C_m can be described as the unification of all variants created from C_m.

$$C_m = V_1 \cup V_2 \cup \ldots V_i \ldots \cup V_n \text{ so that } V_i \neq V_j,$$

where

$$i, j, m, n \geq 0 \text{ and } C_m \sim V_1, \, C_m \sim V_2, \ldots \, C_m \sim V_i, \ldots, C_m \sim V_n \tag{6}$$

From here on we use $V_x(f_1, f_2,.., f_i)$ to represent a variant $V_x = (f_1, f_2,.., f_i)$.

Table 1 Operation Precedence

Precedence	Operations
1	Or: ǀ, Exclusion:$
2	Addition: +, Subtraction: −
3	Cartesian Product: *, Division:/
4	Left dependency: <, Right dependency: >
5	Invert: !
6	Optionality: []
7	Repetition: { }
8	Change of operator precedence: ()

Operations

The relationships between the software elements have critical importance [1].

In our algebra we provide certain operations with their precedence to define the relationships between the software elements including Addition (+), Subtraction (−), Or (ǀ), Exclusion ($), Repetitions ({and}), Dependency (<, or >), Optionality ([and]), Cartesian Product (*), Division (/) and Inverse (!). Additionally, we provide parenthesis to change the precedence of the operators (Table 1).

Feature optionality is one of the main causes of variant productions, since such relations helps to obtain alternatives of a component. Optionality is the result of situations when some features are needed whereas they are totally unnecessary in others. So, as the component reuse options increase the possibility to meet of such optional features increases. To realize the situation, we provide a partial specification for one component Order to represent options with or without Tax.

$$\mathbf{Order} = \mathbf{Product} + [\mathbf{Tax}] + \mathbf{Payment} + \ldots$$

Exclusion can happen to include one of the alternatives at a time but never both of them at the same variant. As an example, we provide the Payment options which are all exclusive.

$$\mathbf{Payment} = \mathbf{Credit\ \ Card}\ \ \$\ \ \mathbf{Check}\ \ \$\ \ \mathbf{Cash}\ldots$$

We provide repetitive and dependent features as well. Order Cost can be calculated using the values of all products included in the order, the tax and shipment price. So, a variant with Order Cost should contain Product Costs, Tax, and Shipment features as well. Product and Product Cost features are examples to represent the repetitions.

$$\begin{aligned}\mathbf{Order} = \{\mathbf{Product}\} + \mathbf{Order\ \ Cost}\\ <\ (\{\mathbf{Product\ \ Cost}\} + \mathbf{Tax} + \mathbf{Shipment\ \ Price})\end{aligned}$$

Optionality and exclusions increase the possible number of variants, whereas dependencies restrict such options, since some of the options will be disabled if the optional features are dependent to each other.

Translating Formal Definitions

Variant Generation Algorithm

Entity Sets and Operation Sets are the main structures used in translation of syntactical rules into the variant definitions. Basically an entity set is a simple set containing certain features whereas an operation set contains many entity sets each of which is a unique set among possible alternatives. These sets are useful in variant generation because the operations can be applied on them one by one and at the end produce an operation set containing all possible alternatives. This is the result that is obtained from the application of the operations defined by the expressions.

Sets are efficient structures in implementation of the system, since they keep only unique elements. Hashing helps to locate a feature within the feature set; however, it is harder to understand that entity set is contained by an operation set, as the contained entity sets should be compared for each of their contained element.

A nice feature of set is the ability to use well defined operations such as intersection, unification, difference, adding and removing elements, all of which are used in the implementation of the system. Fortunately, programming languages such as Java provides libraries to implement sets with ready to use operations and functionality.

Java provides three implementations of Set<E> Interface including HashSet<E>, TreeSet <E>, and LinkedHashSet<E>. Java derives set Interface from super interfaces Collection<E>, Iterable<E>. HashSet<E> is fast since Hash Indexing method is used for fast location of contained elements. TreeSet<E>, is slower but uses tree implementation with all advantages (and of course disadvantages) of tree implementations. LinkedHashSet<E> provides an insertion ordered iteration and similar performance comparing with HashSet<E> [31].

Parsing Formal Rules

A typical algebraic expression in infix form, can be read, scanned, and parsed to identify contained operations and operands. Java Scanner class provides basic lexical analysis operations. So, it is used to scan the expressions and identify the tokens, while Stack class is widely used in parsing and translation to postfix notation. It is easy to create entity sets out of operands and later apply these operations on those sets. The main objective is to convert the complex expressions to operators and operation sets that will be used in application of the operations defined by those operators. The output of the application of operators is an operation set containing unique entity sets.

Entity Set

An entity set is actually a set with any entity which can be any software unit in our system and an entity set typically defines the elements that will be contained in a variant when it contains component features. An entity set can contain other software elements and therefore can be also used to define the other structural units for example. If the entity set contains features of a component, it can be directly converted to a single variant and the specifications of the elements can be used for example to translate them into a programming language such as Java. When we place such importance to the entity sets, they become one of the most critical objectives in the implementation of the variations out of expressions.

Definition **Λ** is an algebra, δ is an operation and f_i is any symbol and E(δ) is an entity set defined by this algebra:

$$E(\delta) = \{f1, f2, f3, f1, \ldots fn\}, \quad i \le n,\ i, n \ge 0, \\ \& \, fi \ne fj \ fi, fj \in E(\delta) \tag{7}$$

An entity set can directly be converted to a variant:

$$V(E(\delta)) = V(f_1, f_2, f_3, f_i, \ldots f_n), \\ f_i \in E(\delta) \Rightarrow f_i \in V(E(\delta)) \tag{8}$$

So, the translation is straightforward, since each element is a member of the variant generated from that entity set. This definition is valid for any software unit that can be defined as a set of other elements, hence if the software unit is specified using expressions, and then proper variants can be produced out of these expressions.

Entity Pairs

Definition Any entity set containing just two elements. So, an entity pair can be defined as:

$$EP_i(a, b) = \left\{a, b \middle| a \in C \Lambda b \in C\right\}, \text{ where C is a component}$$

Entity pairs are used in definition of operations, since it is easier to show how the operations will be applied since the results will be short enough to understand, however, it is possible to define and use other finite sets as well.

Operation Set

Definition **Λ** is an algebra, δ is an operation and f_i is any symbol that can used in the operation and $E_j(\delta)$ is any entity set defined by this algebra:

$$O(\delta) = \{E_1(\delta), E_2(\delta), \ldots, Ei(\delta), \ldots, Ej(\delta), \ldots En(\delta)\}, \\ Ei(\delta) \ne Ej(\delta) i,\ j \le n,\ n \ge 0$$

So, an operation set contains certain entity sets, all of which are different. A similar definition can be done using the elements of each entity set:

$$O(\delta) = \left\{\{f11, f12, f13, \ldots f1n\}\{f21, f22, f23, \ldots f2n\} \ldots \\ \{fm1, fm2, fm3, \ldots fmn\}\right\} i, j \le m, n,\ m, n \ge 0$$

Conversion Rule Each operation set can be converted to unique variants and variants can be converted to operations set one to one.
According to conversion rule, we can find missing definitions out of each other.

Operations on Entity Sets

Operations can be applied on entity sets and the result will always be an operations set. The operations are applied for each possible combination of entity sets, and the result can produce some entity sets which are included by the resulting operation set:

$$\begin{aligned}
&O_1 = \{E_{11}, E_{12}, \ldots, E_{1m}\} \\
&O_2 = \{E_{21}, E_{22}, \ldots, E_{2n}\} \\
&O_1 \delta_k O_2 = \{E_{11}, E_{12}, \ldots, E_{1m}\} \delta_k \{E_{21}, E_{22}, \ldots, E_{2n}\} \\
&\ = \left\{\{E_{11}\delta_k E_{21}\}, \{E_{12}\delta_k E_{21}\}, \ldots, \{E_{1m}\delta_k E_{21}\}, \ldots \right. \\
&\quad \left. \{E_{11}\delta_k E_{21}\}, \{E_{11}\delta_k E_{22}\}, \ldots, \{E_{11}\delta_k E_{2n}\}, \ldots \{E_{1m}\delta_k E_{2n}\}\right\}
\end{aligned}$$

Operation sets are especially useful when the operations are applied iteratively. The operands are added to the operation set and each operation is applied one by one until the whole operators and operands are processed. The result operation set is used to obtain the variant set, since each operation set is convertible to variant set.

Variant Set

Definition A variant set is a set of unique variants produced by an operation δ in algebra **Λ**.

$$\mathbf{V} = \{\mathbf{V}_1, \mathbf{V}_2, \ldots, \mathbf{V_i}, \ldots, \mathbf{V_n}\}, \mathbf{V_i} \ne \mathbf{V_j}\ \mathbf{i}, \mathbf{j} \le \mathbf{n},\ \mathbf{n} \ge \mathbf{0} \tag{9}$$

So, each variant in a variant set should be unique such that contained features by each variant should be different.

Generation of Variants Out of Operation Sets

Typically, an operation set is used to define possible different variations of components and they can later be translated into individual variant definitions. Individual entity sets can be translated to a single variant. Since, set elements have to be unique as a result of the set, each operation set will contain variants all different from each other.

$$
\begin{aligned}
&O(\delta) \leftrightarrow V(\delta) : fi{\in}Ej(\delta){\wedge}Ej(\delta){\in}O(\delta){\wedge}fi{\in}V(\delta)\\
&O(\delta) = \{\{f11, f12, f13, \ldots f1n\}\{f21, f22, f23, \ldots f2n\} \ldots\\
&\qquad \{fm1, fm2, fm3, \ldots fmn\}\}\\
&V(\delta) = \{V1(f11, f12, f13, \ldots f1n),\\
&\qquad V2\{f21, f22, f23, \ldots f2n\} \ldots\\
&\qquad Vm(fm1, fm2, fm3, \ldots fmn)\}
\end{aligned}
\tag{10}
$$

Conclusions

In this paper we presented our approach to implement variants for component based software development based on formal descriptions. The formalism provided in this paper is easy to learn and use, and can be implemented using typical programming techniques. We used Java as the main development language, and implemented the “Stack” and “Set” Interfaces in our applications. With our system, it is possible to represent component feature relationships even if they are complex. With our system, the translation and variant production can be done automatically.

In next step, we plan to automatically generate the code of the variants using code automation techniques in popular languages. Another work would be applying on the fly compilation techniques to generate components and software systems built from those components based on user preferences. We hope to simplify and enhance the application development, if the components were already been defined with their features and relations between them.

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A Methodology for Designing Agent-Based Models: Agent from « UP » for Complex Systems

n Example on the Rift Valley Fever at Barkédji (Senegal)

Ahmed Tidjane Cisse, Aboubacar Cisse, Alassane Bah, Jacques A. Ndjone, and Cheikh M.F. Kebe

Abstract

The designing and implementation of an agent-based model for a system called complex are often guided by clear objectives and often meet a need. Such kinds of models are often used to help for understanding real systems mechanisms and to be at last a decision support. The evolution of a theory that is supported by validation stages through a simulation needs to be guided by a methodology. This one will make possible to organize the work and communicate with the end users in order to achieve the objectives with a reasonable time. We propose here a methodology that extends the Unified process. It provides a set of tools to carry out the modeling work successfully. This methodology is iterative and incremental and is guided by a collaborative approach based on communicating with the end users in one side and specialist in another side. The communication is also supported by tools and a common ontology. The present method formalism is based on Agent UML combined with GAIA. A practical case is also presented with an agent based model on the Rift valley fever.

Keywords

Agent • Methodology • Unified processes • Iterative and incremental approach • Rift valley fever • Complex system

Introduction

Complex system modeling is a field with objectives often including the achievement of realistic simulations. Simulations are the « in sillico » materialization of a theory established by one or several specialists [1]. The process of designing a model to be implemented is often based upon an approach supported by a theory expressed in the form of a mathematical model.

Actually, modeling a complex system to implement a simulation tool often becomes necessary. The modeling activity support research and simulation help to perform a theory by validating the theory model. The modeling work is also guided by clear objectives, following precise needs, it become imperative to achieve within a reasonable time limit. However, achieving these objectives require an organization making it possible to guide the whole of the process to gain time while moving forward coherently.

The idea suggested here is to guide the modeling, implementation and validating work set by a process made up of steps and an organization. These process will promote communication among interested people and is iterative and incremental thus making it possible to move forward as discoveries and new theories questioning some concepts and validating others arise.

The choice is thus made of the Unified Process (UP) [2] which is a process borrowed from software engineering that has to be adapted here and whose appropriateness or relevance has to be checked. The modeling language chosen is Agent UML, which is a UML extension for the agent.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_63, © Springer International Publishing Switzerland 2015

Ontology is also designed to support communication and to give a common language among specialists.

Context

This work is part of the QWeCI project, a project funded by the European Union in the aftermath of the AMMA project (http://www.amma-international.org/) in the environment-Health part (WP 3.4: Health Impact). Scientists have been interested in the relationship between climate variability (inter annual and intra season) and disease outbursts such as meningitis, malaria and Rift Valley Fever (RVF). A great number of African and European institutes that work hard studying the impact of climate changes on the population health in developing countries intervene in this project.

The implementation of this methodology was made within the framework of an agent-based model for the rift valley fever. This work is carried out within a multidisciplinary context involving several research institutes. It has required the setting up of an adequate working and communication framework.

The System Studied and the Objectives

The system studied is the RVF transmission in the Barkedji area (Senegal). The RVF is a tropical region zoonosis due to a Bunyaviridae family arbovirus, a phlebovirus kind [3]. This disease is transmitted to hosts essentially by two vectors which are mosquitoes:

- *Aedes vexans*
- *Culex poicillipes*

They reproduce in water holes and have dynamics varying according to their states and environment.

The main objective is to carry out a simulation taking into account the entity dynamics and their interactions in order to check some experimental results [4] and to finally carry out an Early Warning System (EWS) by detecting epidemics risks in the studied area.

Issue

A complex system is by essence a system considered impossible to reduce to a finished model whatever its size, component number, interaction intensity. Modeling such a system calls upon advanced knowledge but also upon techniques that can help the modeler reach their goal. The system studied here is all the more complex as it depends on several environment, climate and also entomology elements. An exhaustive model has to call upon expert knowledge of these different fields. The integration work is here made easier by resorting to the agent-based model which is interaction-based. So, to overcome the constraints linked to the number of the specialists working in the building of the model, an adequate approach has to be used. These approach concern the whole iterative scientific process that permit to perform a theory about a real system by simulating and validating models (Fig. 1).

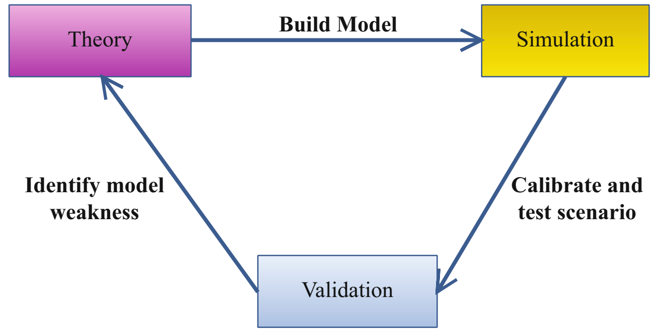


Fig. 1 The scientific process overview

There are several approaches for agent-based modeling. Some of these approaches propose methodology [GAIA, TROPOS, Adelfe], language [AUML, AML] and toolkits (MadKit, MASON, Repast or GAMA). Among the methodologies, some are based on object-oriented processes. In the case of MASUP it is RUP. Regarding complex systems related to ecology, elements of standardization are constructed with the ODD protocol to support model description and validation process. However these approaches, while offering advantages in the description process through implementation of an agent based model suffer from deficiencies in a work environment where communication and collaboration are imperative. Indeed, the requirements provided by the complexity of a system under study can make the construction of a model agent be not longer an end in the research but a tool accompanying the work of thematicians allowing them to experiment or to verify some of their theories. That approach has thus to suggest a project management strategy, a collaborative working tool, description means of the model and an implementation tool.

AUP (*Agent from « UP »*), The Extension

In the present case, the problem is to carry out a simulation which in fact is a reality in "sillico" reproduction. This reproduction attempt is of course not supposed to represent exactly the real object but simply to extract from it some bricks to describe.

The methodology resorted to here calls upon AUML modeling languages [5]; and some GAIA formalisms. To be provided with a complete methodology, the UP process was chosen as a means. However, the latter was built for projects concerning software engineering environment where the problem is to build an information system tending to meet a need often expressed in specifications.

Being guided by the UP process is not new in itself (MASUP). However, the use of this method in modeling complex system, as we will see after, is not obvious. The concept of use cases must thus be revised or adapted. In addition, the choice of Gaia complete description of roles that, from our point of view, is essential. This approach also is particular because it offers ways to integrate through the use of UP and collaborative tools, a multidisciplinary team qualifying research area's specialists of interlocutors and end-users. It also takes care to separate them from designing and implementation of the agent model.

With the UP process and as part of a management application, the starting point is the client's need which is going to be expressed in terms of final features of the application. The Use Cases Diagrams (UCD) are first a means to focus on these features and mark off their outlines. Thus, it is on the basis of these use cases but also of the chosen architecture that the set of the analysis process design, but also implementation and tests is carried out. So actually, the overall objective through up is that the final product conforms and observes all the use cases. Specifically, the users must be able to do anything they are allowed to do and be provided with all the outputs they are entitled to. Thus, everything that comes after in the analysis, design and implementation tends to:

1. Detail these use cases by describing the business entities participating in it and necessarily playing a role in it, the activities, the sequences, the states and their roles,
2. Think about a way allowing to reach a system which fully observes the use cases. This is done by choosing the tools to be used, fully describing means and tricks allowing implementation (design pattern ...) classes, architectures, packages ...,
3. Implement and then test it all, that is to say check if everything that has been developed meet's the use cases pre and post conditions.[1]

This is absolutely logical and perfectly tallies with the systemic approaches whose first and foremost philosophy is shared by the object and the agent that try to answer to the « what ».

This question can be summarized in a classical management application to the expressed need. The agent based simulation emphasizes the entity dynamics identification individually and collectively [6]. These dynamics are the results of entity interactions conformity in pre and post conditions is a guarantee of a reliable simulation given reality.

The parallel between classical UP and this extension of UP is thus made. As a matter of fact, if the classical use cases are successfully adapted to simulation requirements, a guiding line allowing the solution achieved to be analyzed, described and then implemented will be available.

Thus, as in classical UP, it will be possible to

1. Detail these use cases, therefore the interactions by describing the reality entities which interact, which play a role, their state, their roles...
2. Think about a way allowing a solution conforming to the use cases to be reached. Thus choose the implementation tools and fully describe the implementation items while observing possible constraints.
3. Implement and thoroughly test everything. The use cases are good test medium. As a matter of fact, the present system is valid if all use cases are validated. This validation boils down to checking the conformity of the simulated results according to reference scenarios.

Thus, the foundations for the approach used are laid. As a matter of fact, the system entity intrinsic dynamics are described in scientist theories. The agent model force is to put all these dynamics into action and then to show the results of these interactions. "*to imagine a tree*, Paul Valerie makes Leonard de Vinci say, *you are forced to imagine the backgrounds it stands out against*" to perceive that tree is to perceive the interaction between the tree and its context, the action produced by the perception of the interaction (Charles [7]). Thus, these nodes constituted by the interactions are a perfect starting point just by allowing the entities at stake and the role they play in them to be identified. So, they provide a first abstraction which allows these entities to be excluded or included according to their positions in the issue. And the rest of the work is based on this.

Four Process Phases

Like the UP our Process consists of cycles that may repeat over the long-term life of a system. A cycle consists of four phases: Inception, Elaboration, Construction and Transition. Each cycle is concluded with a release, there are also releases within a cycle. The four phases in a cycle are:

- Inception Phase—During the inception phase the core idea is to materialize our vision of the system. In this phase, we discuss with specialists to review and confirm our understanding of system's dynamic.
- Elaboration Phase—During the elaboration phase the majority of the Use Cases are specified in detail and the simulation system architecture is designed.

[1] Pre conditions in use cases refers to necessaries conditions to enter on a use case, Post conditions refers to the conditions we should reach at the end of the use case.

- Construction Phase—During the construction phase the model is being implemented. The architectural baseline grows to become the completed simulation system.
- Transition Phase—In the transition phase the goal is to ensure that the model implemented achieved the expected goals. This phase is often initiated with a beta release of the simulation. The transition phase ends with a postmortem devoted to learning and recording lessons for future cycles.

Implementation

The first question to ask is: « Are the methodological choices made compatible ? ». This question is reflected as being important for the rest. It is often necessary to combine different elements, to create some on condition that it makes sense and that it is useful.

The agent UML first philosophy gives the temptation to answer yes:

When it makes sense to reuse/extend portions of UML, then do it. When it doesn't make sense to use UML, use something else or create something new.

Thus, the present UP extension can well used agent UML as a description language. However, it is believed that the central dimension at the interaction level is the role of interacting entities. As a matter of fact, physically agents exchange. E.g.: « An *aedes vexans* bites an animal. a *culex Poicilipes* bites a human ». But the pertinence of the interaction can be found more at an abstraction level where role is talked about. For in this example, the interactions are the same but it is just the interacting entities that differ. Thus, following this interaction, the entities have quite a different dynamic and take other roles. What will be said is just « a vector bites a host », putting into the context and at the same generalizing the protagonist of the interaction. Given this aspect importance, a set of formalisms with diagrams taken from GAIA are added to the agent role description.

Use Cases

Here the changes Bernard Bauer and James Odell [8] suggested are tentatively used. The main concepts associated with the use case diagrams are:

- Actor
- Use case
- Subject

The subject is the considered system the use case is applied, the users or any other external system in interaction represents the actors. The system expected behavior is specialized by one or several use cases defined in accordance with actors needs.

UCD are applied to define an external point of view on the system. In particular, they define the "what" instead of the how. An UCD is achieved in the prospect of an external communication with partners. So they describe the system, the actor, the use cases and the relationship between actors and use cases.

Agent based systems can also use the same notions of use cases as defined in UML 2.0. The event use of the actor who makes a request the subject has to answer and the events affecting an actor are very appropriate. These are indicated as an association name between an actor and a use case. The associations also indicate the direction of events (the way events are moving to).

The use case major extension is the change in actor definition. UML defines an actor as being « played by an entity interacting with the subject . . . but who is external to the subject » [9]. Actors, in an agent context, can make requests, and the subject internal and external services. Thus, the definition needs extension to indicate that an actor is « played by an entity which interacts with a subject use case. . . but which is external to the use case » in other words the actor can interact with the subject. So it can be internal or external [8].

These changes provide a starting point for the analysis and design. Thus, all go from the use case each agent has of the subject actually constituting its environment in the "ferberian" [6] sense of the world. So all the process is carried out in a collaborative mind. As a matter of fact, the approach chosen integrates this aspect and sets it first and foremost in all that concerns the implementation and design. This attitude is a response to an integration concern of all the participants in the project leading them to appropriate it, which is essential for the success of the work carried out. To facilitate this integration, a common ontology is made to help scientists understand to each other. This ontology is a way to classify terms according the field of its usage (Fig. 2). We thus identify three fields of research to integrate in the model which are:

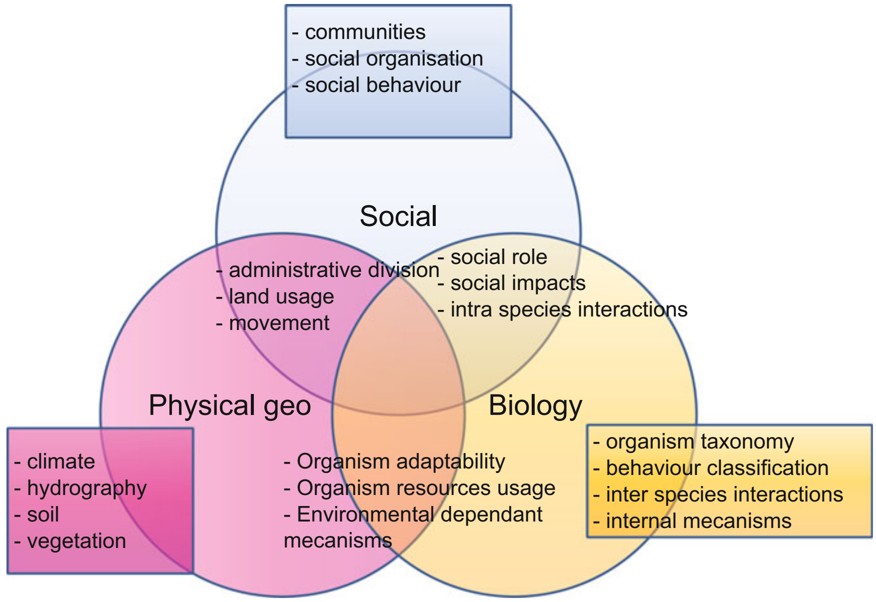
– The social field
– The physical geography field
– The biology field

This classification helps to determine the concern of every specialist and is also a way of collecting terms for constituting the ontology. The ontology is here considered as a common language between specialists.

Tools

Implementing the collaborative environment is an important dimension of the method chosen and whoever talks about "method" necessarily talks about "tools", it is important to make use of it to be able to manage all the aspects of the present context which are taken into account in the method. This work like all the rest is conducted iteratively with some results which have been produced.

Fig. 2 Ontology classification



Application: The RVF in Barkedji (Senegal)

Description of the System

The studied area populating and dynamic are based on water. As a matter of fact, it is the appearance of temporary ponds after the first efficient rain (quantity > 20 mm) which is the activity source or starting with the hatching of the first generation of *aedes vexans* mosquito. Thus, the population of *aedes vexans* evolves as long as the pond dynamic makes it possible. This link can be explained by the need for these eggs to get dry and then be put into water before being able to develop and by their resistance to the dry season. The pond also has a dynamic made up with waterfalls and increases due

– To natural water losses, which are infiltration and evaporation,
– To irregular rains in some rainy season period.

The first rains increase the water level in the pond which extends on the surface gradually putting into water the *aedes vexans* eggs of the previous year that have had the time to get dry. Thus, the latter develop and then hatch. The following generation eggs hatch if the pond increase gives them the time to get dry and then to be put into water to develop. These conditions are observed from the first efficient rain until July. But exceptionally, raining pauses can create these conditions favoring the vector sudden proliferation in periods supposed unfavorable, increasing the risk of epidemic.

Tools

Working on a collaborative environment has required the setting up of a wiki which remains for the researchers a working and communication framework from the members of the team.

A wiki was set up to manage the entire project and to communicate with the team. The wiki constitutes for the researchers:

A communication and a project management tool (Fig. 3);

Analysis and Agent Model

In this part, the model state is summarized after the last iteration achieved.

The analysis, as supposed by the methodology begins by the use case adopting the agent modeling basics and describing the main interactions which define the system.

Thus, four main actors appear which do services to each other, interact though four use cases described in the model bellow (Fig. 4).

These use cases summarize the interactions of the four main actors that are the mosquito, the climate factor, the water hole and the hosts.

The present agent model represents some of the elements which are made up of agent classifiers gathered in packages (Fig. 5). The most important of these are *Aedes vexans* and *Culex poicilipes* mosquitoes, the hosts, the water hole, the vegetation, the rain. . . As the methodology is incremental, the description for the moment will end here on the few agents

The Vector Dynamic

The identified main vectors evolve during their lives and move through two essential states: the egg state and the adult state.


Accueil
A faire
Dates importantes
Définition du sujet
Documentation
L'équipe
Mises à jour
Présentation

6
jours avant
Date fin du projet

Rejoindre notre discussion

Rejoindre notre discussion

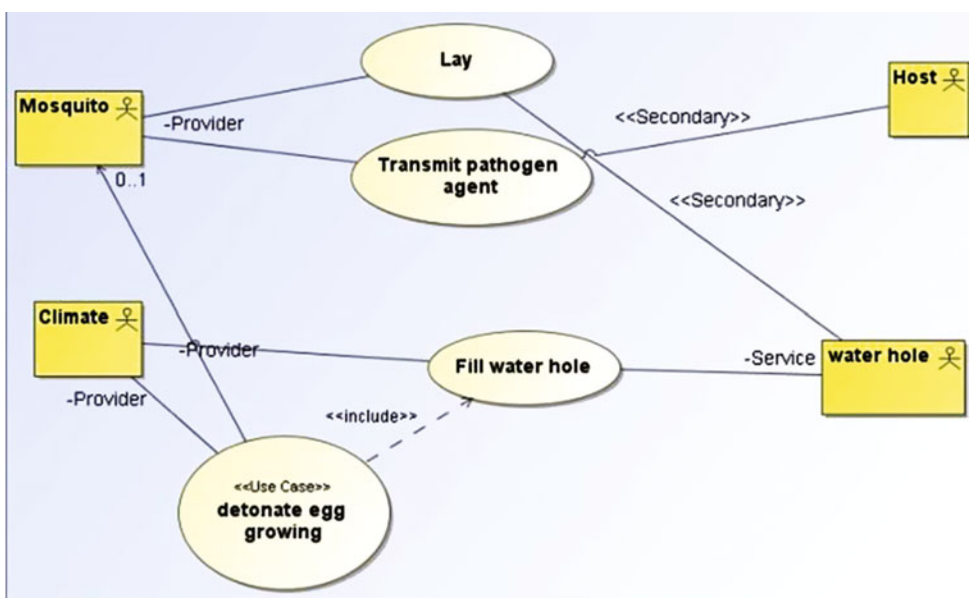
A faire

planning

ID	Activite	Date Début	Date Fin	Pourcentage achevé	dépendence [1]	WBS [2]
1	Début	1-Feb-2010	1-Feb-2010	100.00%		
2	**Inception**	1-Feb-2010	26-Mar-2010	100.00%	1	1.00
3	**Itération 1**	14-Feb-2010	12-Apr-2010	95.00%	1	
4	Etat de l'art	14-Feb-2010	12-Mar-2010	100.00%	1	1.11
5	Etude plateformes et Méthodologies	2-Feb-2010	26-Mar-2010	100.00%	1	1.12
6	Définition des dynamiques	17-Feb-2010	19-Mar-2010	100.00%	4	1.13
7	Analyse	22-Feb-2010	30-Mar-2010		5	
8	Use Case agent		25-Feb-2010	100.00%	5	1.14
9	Diagrammes d'intéractions et d'états	22-Feb-2010	30-Mar-2010	100.00%	7	1.14
10	Conception	8-Mar-2010	5-Apr-2010	100.00%	6	
11	Modèle de Role	8-Mar-2010	5-Apr-2010	100.00%	6	1.15
12	Implémentation	1-Mar-2010	12-Apr-2010	100.00%		
13	Modèles sous Gama	1-Mar-2010	12-Apr-2010	100.00%	10	1.15
14	**Réunion Fin Itération**	**13-Apr-2010**	13-Apr-2010		3	
15						
16	**Elaboration**	26-Mar-2010	30-Apr-2010	100.00%	2	
17	**Itération 2**	13-Apr-2010	28-Apr-2010	100.00%	3	
18	Définition des dynamiques	13-Apr-2010	19-Apr-2010	100.00%	14	2.13
19	Analyse	15-Apr-2010	22-Apr-2010	100.00%	18	
20	Use Case agent	15-Apr-2010	20-Apr-2010	100.00%	18	2.14
21	Diagrammes d'intéractions et d'états	19-Apr-2010	22-Apr-2010	100.00%	20	2.14
22	Conception	19-Apr-2010	26-Apr-2010	100.00%	19	
23	Modèle de Role	19-Apr-2010	26-Apr-2010	100.00%	21	2.15

Fig. 3 Planning of the first two iterations

Fig. 4 The system UCD



For the *Aedes vexans*, the egg state includes a "Drying" sub state which to a large extent determines the evolution in number of agents of this kind in the studied zone in function of the rain temporal distribution (Fig. 6).

The Roles

The system agents evolve according to their states, their environments, according to their well defined roles. The relationships between these roles are of two types:

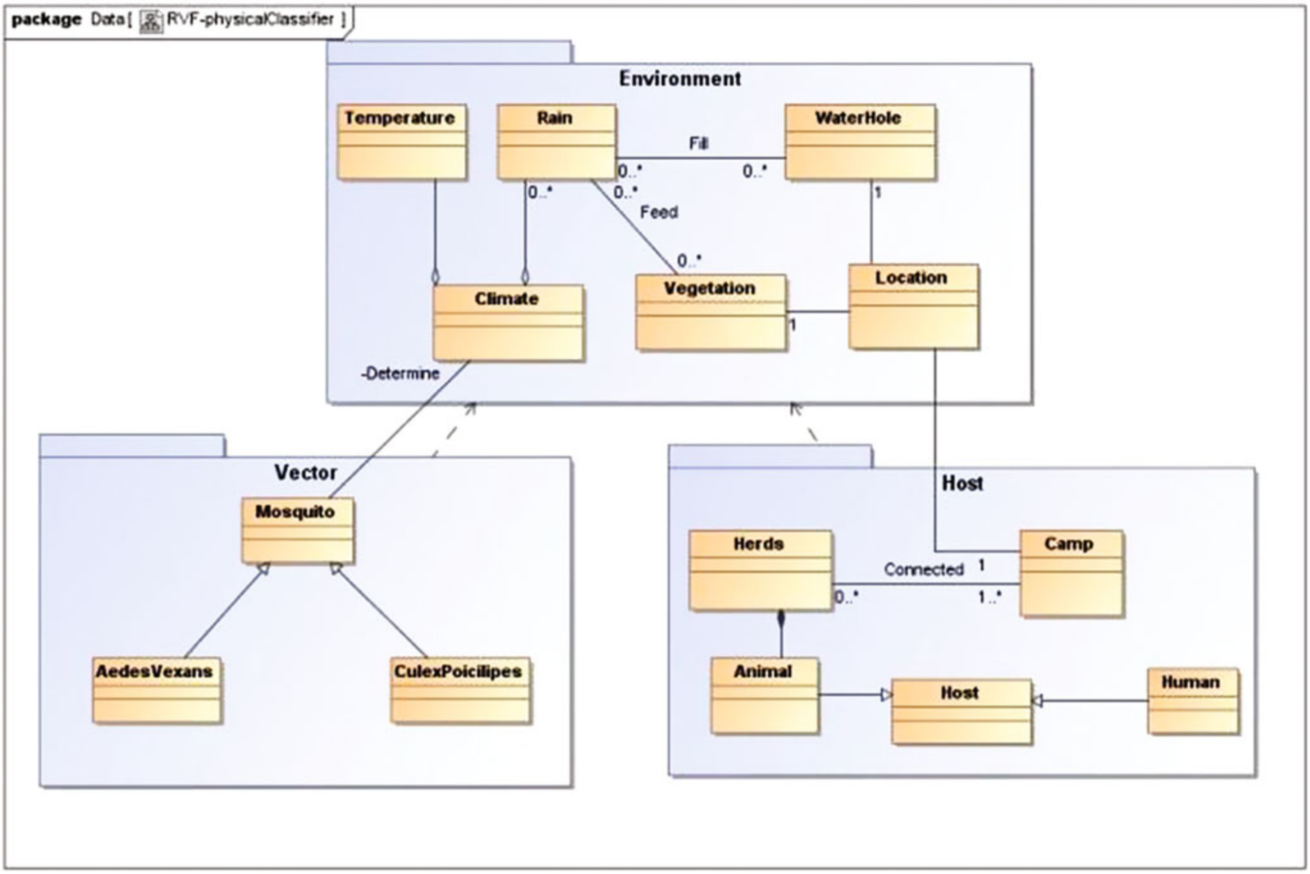
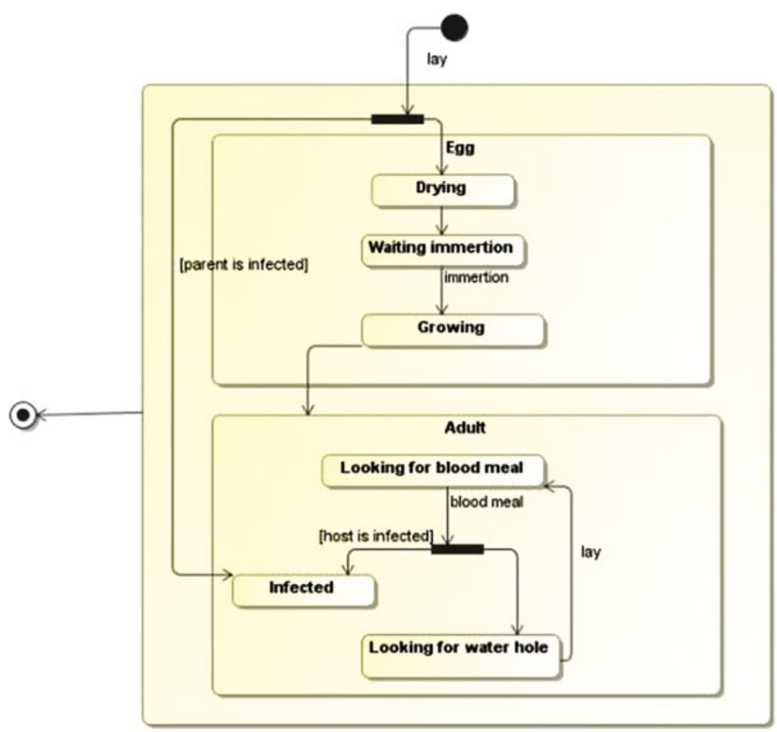


Fig. 5 Agent Physical classifiers of the system

Fig. 6 *Aedes vexans* states



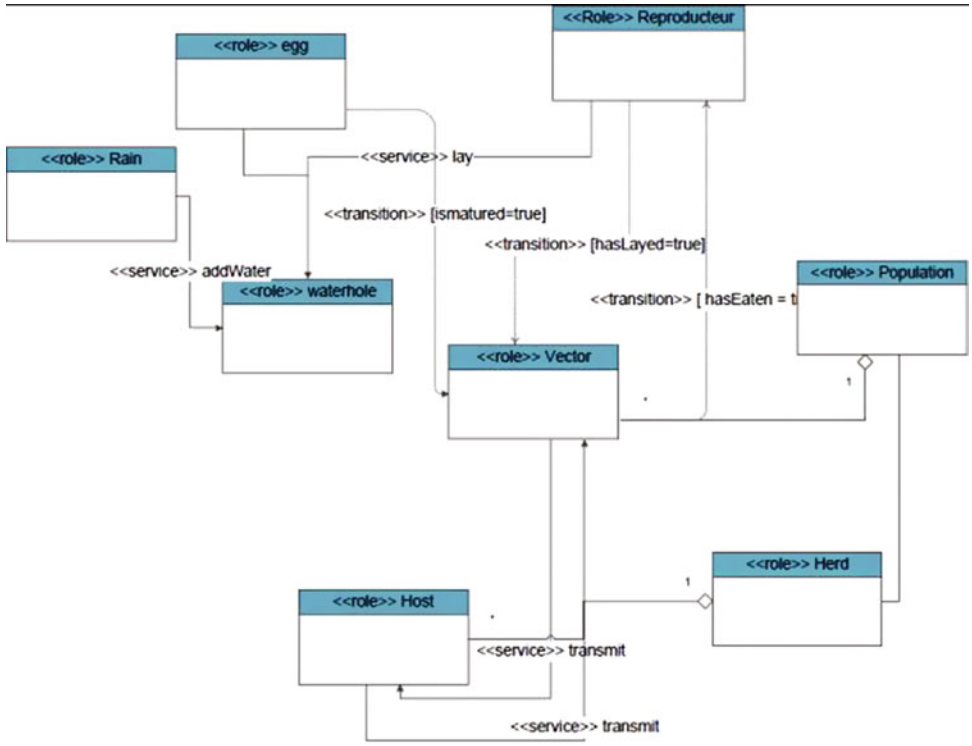
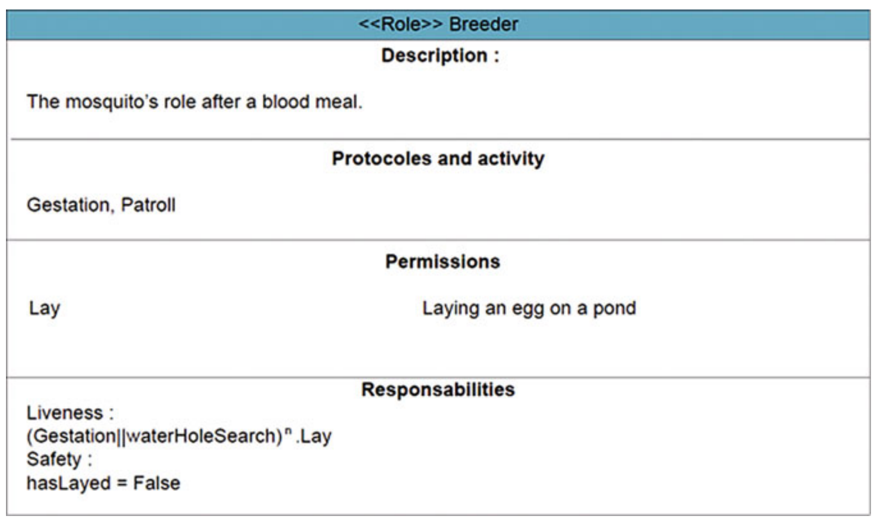


Fig. 7 UML class diagram


<<Role>> Breeder

Description :

The mosquito's role after a blood meal.

Protocoles and activity

Gestation, Patroll

Permissions

Lay Laying an egg on a pond

Responsabilities

Liveness :
(Gestation||waterHoleSearch)^n.Lay
Safety :
hasLayed = False

Fig. 8 Breeder role

– Service: which is a form of interaction between two elements which have the involved role;
– Transition: which represents the transition following a condition on the agent from a role to another.

Here, the roles can correspond to agent states, agent classifiers or agent organizations (Fig. 7).

It is fitting now to explain these roles in detail to specify the actions, the services and the responsibilities governing their dynamics. Thus, GAIA is used to describe them. The latter also give an idea of their guideline, the permissions given, the activities and protocols underlining this guideline (Fig. 8).

An agent with a breeder role at each time step is at gestation and looks for a water hole. It lays when it has performed n time steps corresponding to the gestation duration:

« (Gestation||WaterHoleSearch) n.Lay ».

There is going out of these role when the safety condition is no longer valid as it happens hasLayed = true, which corresponds to a laying.

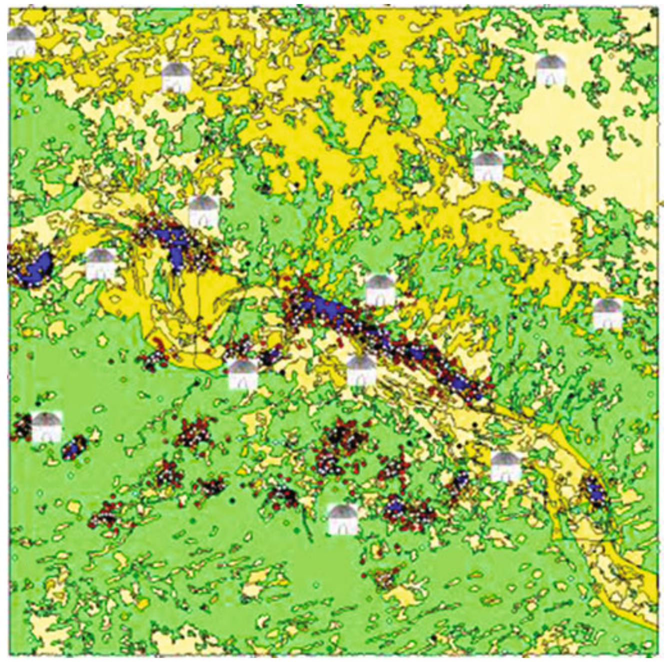


Fig. 9 Simulation environment with GIS background

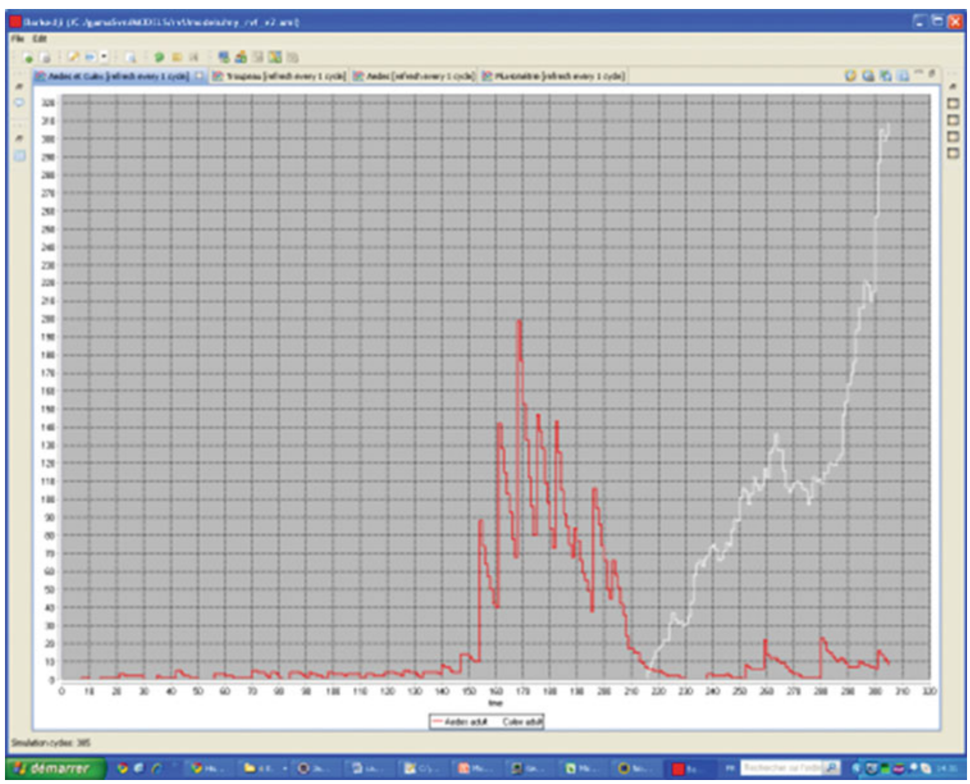


Fig. 10 Vectors evolution curves

Implementation

This model was implemented with the help of GAMA (http://code.google.com/p/gama-platform/) which is a multi-agent system simulation tool developed by MSI-IFI—Hanoï (Vietnam) laboratory. GAMA provides a set of tools which makes it possible to integrate a GIS, a meta model eabling the agent species to be described according to their states [FSM-based behavior (Finite State Machin Behavior)]. These implemented states correspond to the state diagrams available (Figs. 9 and 10).

Results of Simulation

The results of the simulation show the pluviometric factor impact on our model. The dynamic of the vectors is much more linked to the rain temporal distribution than the rain quantity. This does not mean that the amount of water fallen are not important. To start the activities, an efficient rain, (that is a quantity superior to 20 mm) is needed. Thus, maintaining ponds, the dynamic of latter depend on the amount received.

Concerning the monitoring on the viral flow by means of the vector dynamics and the infection rates of the herds, a correlation can be seen with specialist theories such as those of Ndione et al. [4]

Conclusion

In this paper, a methodology has been described which extends UP, a process tallying with the object approach. The extension key, for an agent approach is the use of UCD which pilot the design process but also the tests as in classical management application. Its assessment by means of a practical case has made it possible to gain time in the model development. However, the discussion remains open mainly in relation with the opportunity of use case diagrams in any agent model compared with an object model. These use case diagrams hopefully provide a sure means of piloting the project if the concepts behind make it possible to represent an overall view of the system. The question will then be for the future to check the assertion by testing it and then strengthen our proposal.

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Towards a Passive Gait: Modeling the Fully Actuated Humanoid NAO

P. Bohra and M. Bombile

Abstract

A simple dynamic model of the swing phase of the lower body of a NAO robot is developed and the frictional model for its hip joint is determined experimentally. Thereby a friction compensation technique is proposed to obtain a 'free' moving joint that is important for passive dynamic walking.

Keywords

Dynamic model • Friction model • Coulomb friction • Viscous friction

Introduction

SINCE the release of the NAO robot in 2008, extensive research has been conducted in achieving a robust omni-directional gait through careful control of foot and torso trajectories [1]. Achieving a stable gait for bipedal robots has been one of the more challenging fields of research in humanoid robotics [1]. A standard approach for realizing a stable gait is to generate trajectories that ensure the Zero Moment Point (ZMP) stays inside the foot contact. This requires control over all degrees of freedom of the biped, which makes the gait slow and 'un-human' like [2]. This is evidenced by the NAO robot, whose walking engine implements the same method.

An alternative approach is based on Passive-Dynamic walking, which was first introduced by McGeer in 1990 [3]. McGeer showed that a bipedal machine could walk on down-hill slopes without any active control or internal source of energy. Such passive walkers were based on pendulum models, where the swinging leg was represented as a regular pendulum and the support leg as an inverted pendulum. Several studies have been conducted on two dimensional motion, with and without knees [3]. Previous research at the University of Cape Town has been conducted to implement a stable rocking motion for NAO in the coronal plane, by using its intrinsic stability [4].

This paper aims to develop a simplified dynamic model of NAO's lower body, with the aim of developing a forward passive compass gait for NAO. However, these principles are generally not applied on fully actuated high degree of freedom robots such as NAO. Passive compass gaits have vastly been experimented on custom built robots with negligible joint friction and free swinging legs, such as the robots presented in [3, 5, 6]. To be able to apply passive gait theories such as those presented in [7, 8], on the simplified model of NAO, the robot joints are assumed to be frictionless with no damping. However, motor and gear dynamics at the joint level have to be accounted for in the model, because of the significant presence of friction. Once the motor dynamics are recognised, a compensation technique can be applied that will mimic a free swinging leg. Therefore enabling us to implement passive compass gait theories as in [7].

This paper is organized as follows: section "NAO the Humanoid" introduces the NAO robot. Section "Modeling the Lower Body" describes the parameters of the simple planar model of the NAO robot and the swing leg dynamic model is determined, with the added frictional component. A friction compensation technique is proposed in section "Friction Compensation". Experimental results for friction modeling are shown in section "Experimental Results", followed by conclusions.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_64, © Springer International Publishing Switzerland 2015

NAO the Humanoid

NAO is an autonomous humanoid robot, as shown in Fig. 1. It is designed by the French company Aldebaran robotics. NAO is a standard platform for the Robocup Soccer Humanoid League, ever since it replaced the AIBO quadrupeds in 2008 [10].

The humanoid weighs 5.2 kg and has a height of 0.57 m. There are 25 degrees of freedom (DOF) in the robot, with 5 DOF in each leg; 2 in the ankle, 2 in the hip and 1 at the knee. An additional degree of freedom exists in the hip joint, for yaw, which is enabled by the unique design of the coupled, inclined rotary axis joint. For actuation, coreless brushed DC motors are used. Each actuator is equipped with a Magnetic Rotary Encoders (MRE) for position feedback. NAO currently employs a Zero Moment Point (ZMP) based gait. ZMP is the point on the foot at which all active forces acting on the mechanism during the motion (inertia, gravitation, Coriolis, centrifugal forces and moments) are balanced [11]. The footstep planner of NAO calculates footstep trajectories that will satisfy the ZMP stability criterion. The ZMP stability criterion states that, for dynamic stability, the ZMP of the robot has to be contained within the convex hull of its feet [12].

Fig. 1 NAO H25 by Aldebaran Robotics [9]

Modeling the Lower Body

Figure 2 shows a simple straight legged model of a bipedal robot walking down a ramp. The robot has two rigid legs with point mass m. The point mass is located at the center of mass of the legs, which is a distance b from the hip joint. The hip joint has a mass m_H, which is actually the mass of the entire upper body lumped at the center of mass of the hip joint. The ankle joints of this simple model of NAO are considered to be stiff, which is similar to a human walking with ski boots. The leg cannot rotate about the ankle joint. The heel is located at a distance c and the toe at distance d from the ankle joint.

θ_s and θ_{ns} are angles of the swing leg and the non-swinging stance leg respectively, from the vertical axis intersecting the hip joint. The angle of the slope down which the robot is walking is ϕ.

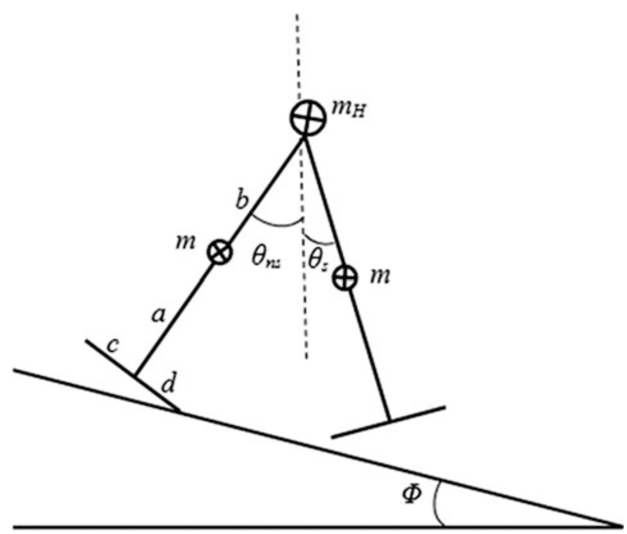


Fig. 2 Simple point mass model of NAO [8]

Assumptions

For the purposes of modeling, some characteristics of NAO were greatly simplified, leading to the following assumptions:

- There is no actuation in any of the joints.
- Motion occurs only in the sagittal plane.
- All the links are rigid and do not suffer any deformation.
- There is no slip, meaning that the friction between the foot and the ground is very large.
- All foot strikes are instantaneous and fully inelastic.
- There is no foot scuffing.

Model Parameters

The model is based on the simplest walking model developed in [13], with the addition of flat feet and fixed ankle joints instead of pointed feet. The model closely resembles the model proposed in [8]. The reason for choosing this flat foot model over the simplest point foot model is that, the

Table 1 Model parameters of NAO

Parameters	Values	Description
m_H	3.243	Mass at hip (kg)
m_1/m_2	0.977	Mass of leg (kg)
a	0.208	Distance between ankle and COM of leg (m)
b	0.041	Distance between hip and COM of leg (m)
c	0.025	Distance between heel and leg (m)
d	0.034	Distance between toe and leg (m)
θ_{ns}	θ_{ns}	Angle of non-stance leg with respect to the vertical
θ_s	θ_s	Angle of stance leg with respect to the vertical
ϕ	ϕ	Angle of slope

orientation of links and joints of the chosen model can easily be configured on NAO for testing. The model parameters and their NAO equivalent values are shown in Table 1

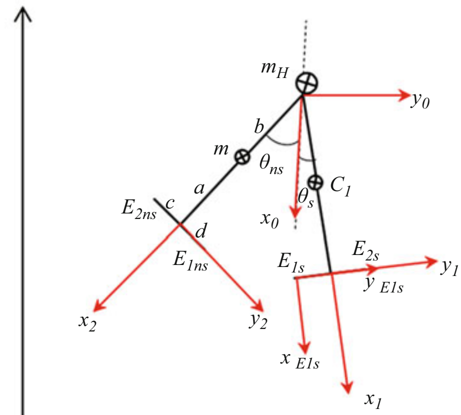


Fig. 3 Simple model in the inertial reference frame

Passive Gait Model

Dynamic equations of the swing phases can now be developed: There are two distinct swing phases and two transition phases in one complete step. Each swing phase has its respective dynamic equation. The first phase of the swing is where the supporting leg starts to rotate about its heel and the non-supporting leg starts to swing. The dynamic equation for the first phase of the swing is worked out here, as explained in [14]. In the swing phase, the robot behaves like a planar double inverted pendulum. Therefore the well-known Euler-Lagrange formulation, shown in Eq. (1) can be applied to derive the dynamic equation (2)

$$\frac{d}{dt}\left(\frac{\partial L(\theta,\dot{\theta})}{(\partial\dot{\theta})}\right)-\left(\frac{\partial L(\theta,\dot{\theta})}{(\partial\dot{\theta})}\right)=\tau \tag{1}$$

$$M(\theta)\ddot{\theta}+C(\theta,\dot{\theta})\dot{\theta}+G(\theta)=\tau \tag{2}$$

The right hand side of Eqs. (1) and (2) have $\tau = 0$ because it is assumed that the robot is completely passive and none of the joints are actuated during the walk. $M(\theta)\ddot{\theta}$, $C(\theta,\dot{\theta})\dot{\theta}$ and $G(\theta)$ are the inertia, Coriolis and gravitation components of the model. To derive these terms, firstly the inertial reference frame is assumed to be given and orientated in the standard manner as shown in Fig. 3

The Denavit-Hartenberg parameters, as explained in [14] are then determined and shown in Table 2.

The dynamic equation is determined as follows: From [14], we know that

$$L = K - P \tag{3}$$

Where L is the Lagrangian ; K, the kinetic energy and P, the potential energy.

Table 2 Denavit-Hartenberg parameters

Frame	a_i	α_i	d_i	θ_i
Stance foot	$a + b = l$	0	0	θ_s
Non stance foot	$a + b = l$	0	0	$\theta_{n,s}$

The kinetic energy is determined as:

$$K=\frac{1}{2}v^T Iv+\frac{1}{2}\omega^T I\omega \tag{4}$$

Therefore

$$\begin{aligned}K=\frac{1}{2}\big[&m_1(J_{VC1})^T J_{VC1}+m_H(J_{VO})^T J_{VO}\\&+(J_{\omega C1})^T\mathcal{I}_1 J_{\omega C1}+(J_{\omega 0})^T\mathcal{I}_0 J_{\omega 0}\\&+m_2(J_{VC2})^T J_{VC2}+(J_{\omega C2})^T\mathcal{I}_2 J_{\omega C2}\big]\end{aligned} \tag{5}$$

Where J_v and J_ω are, linear velocity and rotational velocity Jacobian of the links respectively and, $\mathcal{I}$ is the inertia tensor.

The potential energy of the swinging leg is determined by attaching the body referenced frame to the ground, which represents the moment of heel-strike as shown in Fig. 4.

$$\begin{aligned}P(\theta)=&\,m_1 g(cSin\theta_s+aCos\theta_s)+m_H g(cSin\theta_s+lCos\theta_s)\\&+m_2 g(cSin\theta_s+lCos\theta_s-bCos\theta_{ns})\end{aligned} \tag{6}$$

Since $G(\theta)=\left(\frac{\partial P}{\partial\theta_s}\right)$, it can be deduced that:

$$G(\theta)=\begin{bmatrix}[m_1 g(cCos\theta_s-aSin\theta_s)\\+m_H g(cCos\theta_s-lSin\theta_s)\\ \left[m_2 g\left(cCos\theta_s-lSin\theta_s\right.\right.\\ \left.\left.-bCos\theta_{ns}\right)\right]\end{bmatrix} \tag{7}$$

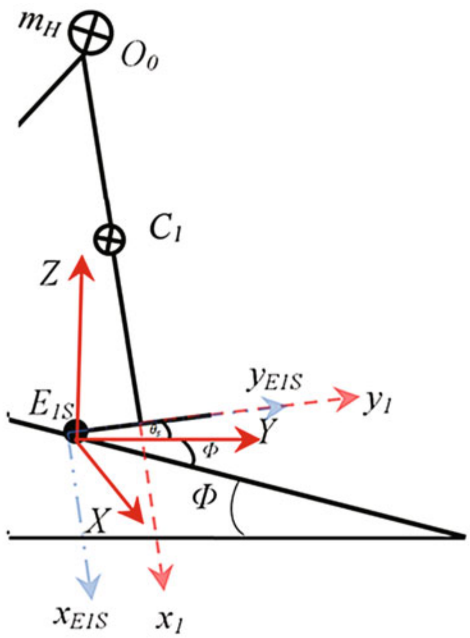


Fig. 4 Swing leg fixed to the ground

The inertia matrix will be:

$$M = m_1(J_{VC1})^T J_{VC1} + m_H(J_{VO})^T J_{VO} + (J_{\omega C1})^T \mathcal{I}_1 J_{\omega C1} + (J_{\omega 0})^T \mathcal{I}_0 J_{\omega 0} + m_2(J_{VC2})^T J_{VC2} + (J_{\omega C2})^T \mathcal{I}_2 J_{\omega C2} \tag{8}$$

The centrifugal and Coriolis terms are determined from the Christoffel symbols:

$$C_{ijk} = \left(\frac{\partial m_{jk}}{\partial \theta_i} + \frac{\partial m_{ik}}{\partial \theta_j} + \frac{\partial m_{ij}}{\partial \theta_k}\right) \tag{9}$$

Where m_{ijk} are the elements of the inertia matrix $M(\theta)$. Note that the terms where $i = j$ are centrifugal while the terms where $i \neq j$ are the Coriolis components [14]. The M and C terms are listed in Table 3.

Table 3 Elements of the Mass, Coriolis and Centrifugal force matrices

M_{11}	$m_2\left(a^2 - 2l^2 Cos\,(\Delta\theta) + 2lb - 2al + 3l^2 + b^2 + 2c^2 + 2acSin(\Delta\theta) - 2lcSin(\Delta\theta) + alCos(\Delta\theta)\right) + 2m_1\left(a^2 + c^2\right) + 2m_H\left(l^2 + c^2\right)$
$M_{12} = M_{21}$	$m_2\left(a^2 + 2l^2 - 2al - l^2 Cos\,(\Delta\theta) + acSin(\Delta\theta) - lcSin(\Delta\theta) + alCos(\Delta\theta) + 2lb + b^2 + c^2\right)$
M_{22}	$m_2(l^2 - 2al + a^2 + (-l - b)^2 + c^2)$
C_{111}	$m_2(a - l)(cCos(\Delta\theta) - lSin(\Delta\theta))$
$C_{121} = C_{211}$	$-\ m_2(a - l)(cCos(\Delta\theta) - lSin(\Delta\theta))$
C_{221}	$-\ m_2(a - l)(cCos(\Delta\theta) - lSin(\Delta\theta))$
C_{112}	$2m_2(a - l)(cCos(\Delta\theta) - lSin(\Delta\theta))$
C_{122}	0
C_{222}	0

where $\Delta\theta = \theta_s - \theta_{ns}$

Hip Joint Dynamics

Motor and Gear Train

NAO uses brushed DC coreless servo motors in series with spur and planetary gears, with a reduction ratio, $r = 201.3{:}1$, in its hip joint [15]. Thus the joint can be modeled as in Fig. 5 where J_a, J_g and J_l are motor, gear and load inertia respectively. τ_m and τ_l are motor and load torques. B_m is the motor damping, θ_m and θ_s are motor shaft angle and angle of the load shaft.

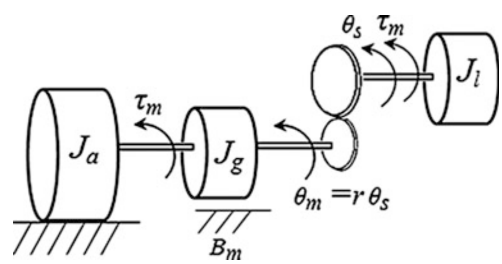


Fig. 5 Model of the hip joint [14]

The equation of motion of the system shown in Fig. 5 is

$$J_m\ddot{\theta} + B_m\dot{\theta} = \tau_m - \frac{\tau_l}{r} \tag{10}$$

where $J_m = J_a + J_g$

Joint Friction

The spur and planetary gears in the hip joint significantly increase the effects of viscous and Coulomb friction [16]. Viscous friction is caused by the viscosity of lubricants used in the joint. It is a velocity dependent component of friction, meaning that it increases as the velocity of the joint motor increases. It can be seen in Fig. 6 as the slope [17]. Coulomb friction on the other hand is independent of velocity and is always present. It opposes the armature torque of the joint motor. Because of the simplicity of the Coulomb friction model, it has commonly been used for friction compensation [13].

The friction model, as seen in Fig. 6 can be written as:

$$\tau_f = B_v\dot{\theta} + \tau_c \tag{11}$$

where

$$\tau_c = \tau_c sgn\left(\dot{\theta}\right) = \begin{cases} 0 & \text{if } \dot{\theta} = 0 \\ \tau_c^+ & \text{if } \dot{\theta} > 0 \\ \tau_c^- & \text{if } \dot{\theta} < 0 \end{cases} \tag{12}$$

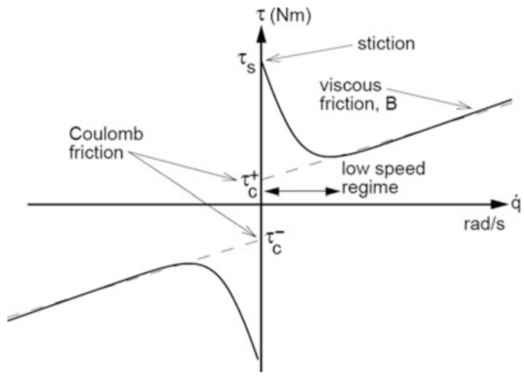


Fig. 6 Frictional components of the hip joint [16]

Another phenomenon that can be observed from Fig. 6 is stiction or static friction, that describes the friction force at rest [13]. Stiction is the torque necessary to bring the stationary joint to motion [16]. Adding the frictional and motor dynamics terms, as in [14], the dynamic model of the swing phase of NAO's leg becomes:

$$M(\theta)\ddot{\theta} + J(\theta)\ddot{\theta} + C(\theta,\dot{\theta})\dot{\theta} + G(\theta) + \mathcal{B}(\dot{\theta}) + \tau_c = \tau \quad (13)$$

Where τ_c, $J(\theta) = r^2 J_m$ and $\mathcal{B}(\dot{\theta}) = B_v\dot{\theta} + B_m\dot{\theta}$ are the additional terms brought about by the hip joint dynamics. The hip joint dynamics terms can lumped together as $J(\theta)\ddot{\theta} + B(\dot{\theta}) + \tau_c = \psi(\theta,\dot{\theta})$. Therefore the augmented formula can be written as:

$$M(\theta)\ddot{\theta} + C(\theta,\dot{\theta})\dot{\theta} + G(\theta) + \psi(\theta,\dot{\theta}) = \tau \quad (14)$$

In order to obtain a free moving hip joint, $\psi(\theta,\dot{\theta})$ will have to be compensated. First, the unknown terms have to be obtained. Henceforth, the lumped $\psi(\theta,\dot{\theta})$ term will be referred to as friction.

The hip joint friction can easily be measured, by working out experimentally the linear piecewise friction model in Eq. (12). This involves moving the joint at various velocities in the sagittal plane and finding the average torque [16]. The basis of this measurement technique is that it is able to determine the work done by the joint motor by measuring the motor's armature current i_a [18].

$$W = \int_{\theta_{s1}}^{\theta_{s2}} \frac{i_a d\theta_s}{\sigma} \quad (15)$$

where i_a is the armature current and σ is the motor torque sensitivity.

Equation (15) basically says that, the work associated with the motion of the hip joint along a defined path, θ_{s1} to θ_{s2}, can be measured. If a path is chosen for which the work depends on a particular parameter, it is then possible to solve for this parameter [18].

Stiction τ_s can be measured by increasing the armature current to the hip joint motor until the joint starts to move.

Friction Compensation

Friction compensation at the hip joint can be achieved based on the computed torque control as described in [16]. This will mimic a frictionless hip joint for a passive gait. The technique is proposed here and a proof of the concept provided.

Equation (14), gives the augmented dynamic equation with the added frictional component, where τ is the input torque:

$$M(\theta)\ddot{\theta} + C(\theta,\dot{\theta})\dot{\theta} + G(\theta) + \psi(\theta,\dot{\theta}) = \tau$$

From [16], the input torque to the hip joint will be:

$$\begin{aligned}\tau = {} & M(\theta)\{K_v(\dot{\theta}_d - \theta) + K_p(\theta_d - \theta) + \ddot{\theta}_d\} \\ & + C(\theta,\dot{\theta})\dot{\theta} + G(\theta) + \psi(\theta,\dot{\theta})\end{aligned} \quad (16)$$

where θ_d is the desired position and θ the actual position of the joint.

Therefore, equating (13) to (16) gives

$$\ddot{\theta} = K_v(\dot{\theta}_d - \theta) + K_p(\theta_d - \theta) + \ddot{\theta}_d \quad (17)$$

hence:

$$\ddot{\theta}_d - \ddot{\theta} + K_v(\dot{\theta}_d - \theta) + K_p(\theta_d - \theta) = 0 \quad (18)$$

Ideally, the error dynamics of the system will be

$$\ddot{e} + K_v\dot{e} + K_p e = 0 \quad (19)$$

where $e = \theta_d - \theta$.

For a passive gait, the desired dynamic of the system is

$$M(\theta_d)\ddot{\theta}_d + C(\theta_d,\dot{\theta}_d)\dot{\theta}_d + G(\theta_d) = 0 \quad (20)$$

$$M(\theta_d)\ddot{\theta}_d = -C(\theta_d,\dot{\theta}_d)\dot{\theta}_d - G(\theta_d) \quad (21)$$

for perfect tracking $\theta \to \theta_d$; therefore $\theta_d - \theta = 0$ and $\dot{\theta}_d - \dot{\theta} = 0$

therefore (16) will be:

$$\tau = M(\theta)\ddot{\theta}_d + C(\theta,\dot{\theta})\dot{\theta} + G(\theta) + \psi(\theta,\dot{\theta}) \quad (22)$$

substituting (21) in (22):

$$-C(\theta_d,\dot{\theta}_d)\dot{\theta}_d - G(\theta_d) + C(\theta_d,\dot{\theta}_d)\dot{\theta}_d + G(\theta_d) + \psi(\theta,\dot{\theta}) = \tau \quad (23)$$

Fig. 7 Armature current and joint position plotted against time

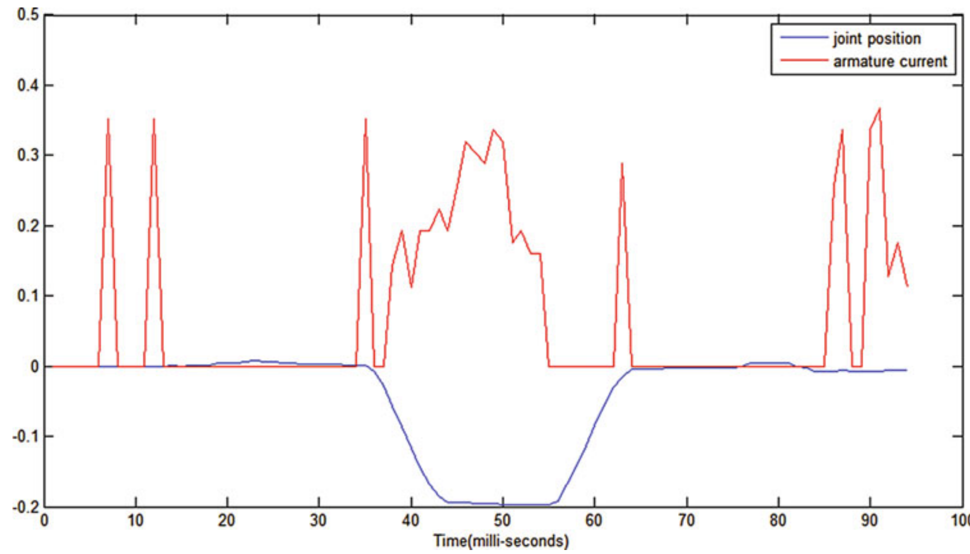
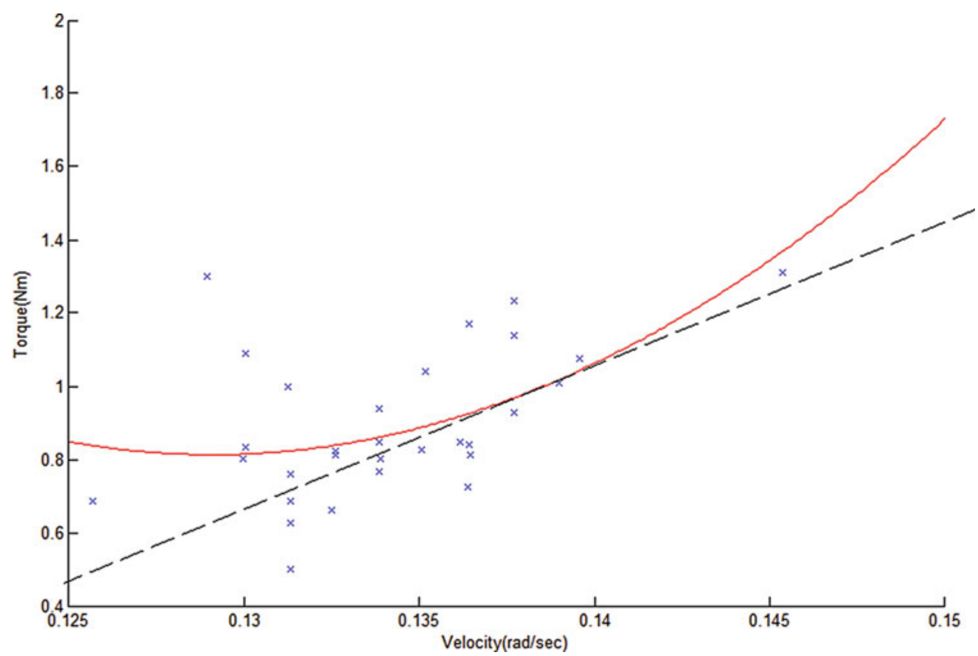


Fig. 8 Torque versus joint velocities



thus

$$\tau = \psi(\theta, \dot{\theta}) \tag{24}$$

Proving that an input torque, as expressed in Eq. (16), will cancel the friction brought about by the hip joint.

Experimental Results

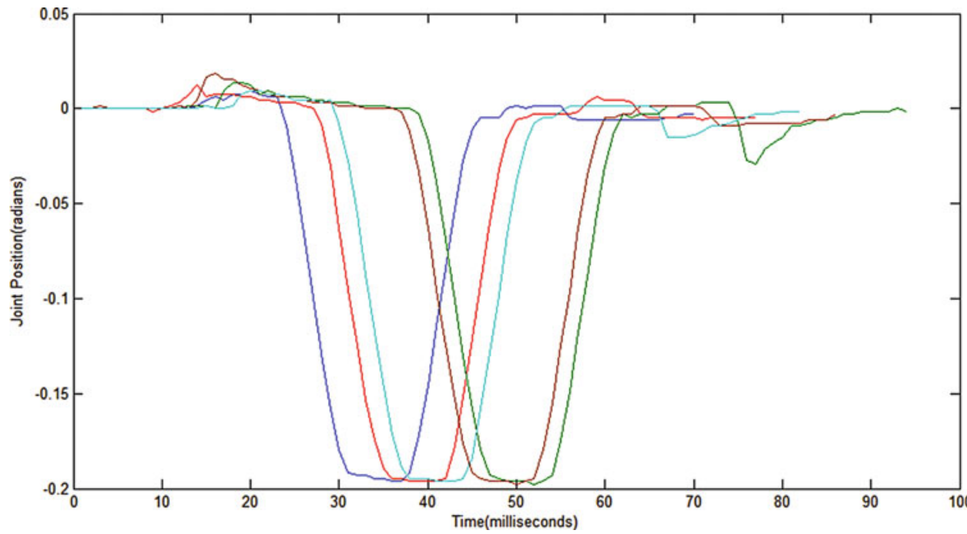
This section presents experimental results related to the identification of τ_c and B necessary to implement the compensation scheme given by Eq. (16) in order to achieve a passive gait. To determine the friction terms, τ_c and B, the hip joint was moved in the sagittal plane at a fixed velocity, and the joint motor armature current recorded. Typical results are shown as time plots in Fig. 7. The motor generated torque is directly proportional to the armature current:

$$\tau = K i_a \tag{25}$$

where $K = 19.4$ mNm/A is the motor constant.

Armature current readings are recorded for 30 positive joint velocities. The average torque to move the joint at a specific constant positive velocity is determined using Eq. (25). A torque versus joint velocity graph is plotted in Fig. 8. Unlike Fig. 6, the behaviour that Fig. 8 exhibits, consists of Coriolis, centrifugal, and frictional terms.

Fig. 9 Joint positions showing the symmetrical motion



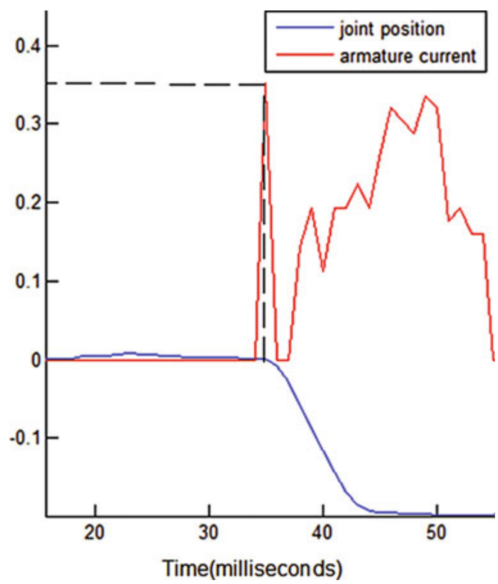


Fig. 10 Measurement of stiction

From Eq. (13), the motor inertia term $J(\theta)\ddot{\theta}$ will be zero as the experiment was done for a constant velocity. The gravitational term $G(\theta) = m_1 gbSin\theta_s$, the Coriolis and centrifugal terms $C(\theta, \dot{\theta})\dot{\theta}$ can also be considered zero, this is because the variation in joint position and velocity is small, therefore the torque due to the respective components will be negligible as compared to torque provided by the motor to swing the leg. Hence the dominant terms seen in Fig. 8 are that of τ_c and B.

Joint torques in the negative velocity region can be approximated by the virtue of a symmetrical motion in the negative and positive direction. The motion symmetry of the hip joint is clearly visible in the time plots of Fig. 9.

Table 4 Friction parameters of the hip joint

Parameters	Value
τ_s	1.140 Nm
τ_c	0.410 Nm
B	38.63 Nms/rad

As shown in Fig. 10, to determine static friction, the current at which the joint starts moving is recorded. The experiment is repeated 30 times and an average value of the initial current is determined.

The friction parameters are listed in Table 4 where $|\tau_s^+| = |\tau_s^-|$ and $|\tau_c^+| = |\tau_c^-|$

Conclusions

With an eventual aim of developing a planar passive compass gait for the NAO robot, a simple planar model for the lower body of the robot is developed using a well documented Euler-Lagrange method. The model consists of knee-less lower body with flat feet. To support the assumption of a free swinging leg, a model based friction compensation technique is proposed. The compensation technique required a model of the hip joint dynamics, which was developed. The dominant features in the model were that of Coulomb, viscous and static friction.

It should be emphasized that good tracking performances are required to achieve a passive gait under the compensation scheme equation (16). Therefore, future work will be devoted to the design of tracking controllers and implementation of the proposed scheme. A possible extension to passive walking with knees can also be envisioned.

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An Architectural Model Framework to Improve Digital Ecosystems Interoperability

S. Shervin Ostadzadeh, Fereidoon Shams, and Kambiz Badie

Abstract

For the last two decades, software architecture has been adopted as one of the main viable solutions to address the ever-increasing demands in the design and development of complex software systems. Nevertheless, the rapidly growing utilization of communication networks and interconnections among software systems have introduced some critical challenges, which need to be handled in order to fully unleash the potential of these systems. In this respect, digital ecosystems, generally considered as a distributed adaptive open socio-technical system, have gained considerable attention, since their scale is incomparable to the traditional systems. The scale of socio-technical ecosystems makes drastic changes in various aspects of system development. As a result, it requires that we broaden our understanding of software architectures and the ways we structure them. In this paper, we investigate the lack of an architectural model framework for digital ecosystems interoperability, and propose an architectural model framework to improve digital ecosystems interoperability based on complex system theory.

Keywords

Framework • Architectural Model • Digital Ecosystems Interoperability • Software architecture • Software systems

Introduction

In today's digital era, the development of solid digital network infrastructures drastically affects the economic resources of communities as well as their lifestyle. In various domains, such as healthcare/health-science, energy, social networks, and logistics, future applications require infrastructures that are more agile than those functional at the moment. Digital ecosystems have emerged with the goal of capturing the notion of such agile and adaptive infrastructures. For this purpose, digital ecosystem technologies enclose the entire spectrum of Internet related technologies. This ranges from the hyperlinked web towards pervasive internet applications, and from peer-to-peer systems to Grid middleware. It also includes Cloud services, agent technologies, sensor networks, and cyber-physical systems. Thus, digital ecosystem has become one of the main topics for business process digitalization.

As systems grow larger and more complex to become digital ecosystems, new requirements for software architectures emerge. The software architecture of a program or computing system is the structure(s) of the system, which comprise software elements, the externally visible properties of those elements, and the relationships among

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_65, © Springer International Publishing Switzerland 2015

them [1]. Based on this definition, it is inferred that software architecture characterizes the structure of a system. In general, architecture is the fundamental organization of a system embodied in its components, their relationships to each other, and to the environment, and the principles guiding its design and evolution [2].

According to the ISO 15704 standard [3], an architecture represents a description of the basic arrangement and connectivity of parts of a system (either a physical or a conceptual object or entity), which is expected to create a comprehensive overview of the entire system when put together [4]. It should be noted that handling this large amount of information is quite challenging and needs a well-developed framework. The problem is even intensified in the case of digital ecosystems, due to their scale and socio-technical characteristics. So far, various Information Systems Architecture (ISA) frameworks have appeared in literature: Zachman framework [5], FEAF [6], TEAF [7], ToGAF [8], and C4ISR [9] to name a few. Nevertheless, these frameworks fail to provide all the required support for digital ecosystems. Consequently, the inability of current ISA frameworks to meet these requirements necessitates a breakthrough research in the development of a socio-technical ecosystems architectural framework [10].

In this paper, we present an architectural model framework in digital ecosystems interoperability based on complex system theory. The proposed framework is assumed to be capable of addressing the requirements of such systems.

The rest of the paper is organized as follows. In section “Helpful Hints”, we present the required background and the problem definition. Next, we present an interoperability model overview in section “Interoperability Models”. The digital ecosystems interoperability model based on complex system theory is discussed in section “Digital Ecosystems Interoperability”. Finally, section “Conclusions” summarizes the contributions and sets the direction for the future work.

Helpful Hints

A digital ecosystem is a distributed adaptive open socio-technical system with properties of self-organization, scalability and sustainability inspired from natural ecosystems [11]. Digital ecosystems inherit concepts of open, loosely coupled, demand-driven, domain clustered, agent-based self-organized collaborative environments where species/agents form a temporary agreement for a specific purpose or goal. In such environments, the agents take precautionary measures and react for their own advantage. The adoption of ecological system concepts is the central characteristic of all digital ecosystems. This is achieved by bonding via collective intelligence and further collaboration instead of uncontrolled competition as well as ICT-based catalyst effects in various domains, to produce improved connected communities and solutions.

Ultra-Large-Scale Digital Ecosystems

In biology, an ecosystem is a community of plants, animals, and microorganisms that are linked by energy and nutrient flows and that interact with each other and with the physical environment. Rain forests, deserts, coral reefs, grasslands, and a rotting log are all examples of ecosystems. In 2006, SEI [10] published a report about some systems which were called as Ultra-Large-Scale (ULS) systems. ULS systems can be characterized as socio-technical ecosystems, whose elements are groups of people together with their computational and physical environments. These systems will go far beyond the size of current systems and system of systems by every measure, such as, the number of the lines of code; the number of people employing the system for different purposes; amount of data stored, accessed, manipulated, and refined; the number of connections and interdependencies among software components; and the number of hardware elements.

There are some characteristics of ULS systems that will be revealed because of their scale [10]: (1) decentralization; (2) inherently conflicting, unknowable, and diverse requirements; (3) continuous evolution and deployment; (4) heterogeneous, inconsistent, and changing elements; (5) erosion of the people/system boundary; (6) normal failures; (7) new paradigms for acquisition and policy. These characteristics undermine current, widely used, information systems framework and establish the basis for the technical challenges associated with ULS systems.

A complex systems is a system composed of interconnected parts that as a whole exhibit one or more properties (behavior among the possible properties) not obvious from the properties of the individual parts [12]. A system’s complexity may be of one of two forms: disorganized complexity and organized complexity [13]. ULS digital ecosystems are examples of disorganized complexity, since disorganized complexity is a matter of a very large number of parts [14].

It has been observed that current approaches fail to fully define, develop, deploy, operate, acquire, and evolve ULS systems, as described in SEI report [10]. ULS systems are considered as cities or socio-technical ecosystems, while our current knowledge and practices are geared toward creating individual buildings or species. This inconsistency points out the research direction that is crucial for reaching a proper solution to develop ULS systems.

Research Context

The challenges that have to be addressed when developing a ULS digital ecosystem span three different areas: (1) Design and Evolution; (2) Orchestration and Control; and (3) Monitoring and Assessment [10]. The research work presented here addresses the design area related to "design and evolution".

Fundamental to the design and evolution of a digital ecosystem will be explicit attention to design across logical, spatial, physical, organizational, social, cognitive, economic, and other aspects of the system. Attention to design is also needed across various levels of abstraction involving hardware and software as well as procurers, acquirers, producers, integrators, trainers, and users. A key area of research in design is thus the need for design of all levels of a ULS digital ecosystem.

Why Interoperability?

Broadly speaking, interoperability refers to coexistence, autonomy, and federated environment, whereas integration conventionally refers to the concept of coordination, coherence, and uniformization [4]. ULS digital ecosystems go far beyond the size of current systems and system of systems by every measure, including, the number of the lines of code; the number of people using the system for different purposes; amount of data stored, accessed, manipulated, and refined; the number of connections and interdependencies among software components; and the number of hardware elements [10]. These are instances of "Loosely-Coupled" systems. This means that the components in such systems can interact and are connected by a communication network; they can exchange services while continuing locally their own logic of operation. "Tightly-Coupled" indicates that the components are interdependent and cannot be separated. This is the case of a fully integrated system. Thus, two integrated systems are inevitably interoperable; however, two interoperable systems are not necessarily integrated.

Problem Definition

The scale of complexity and uncertainty in the design of digital ecosystems is so immense that resists the treatments offered by traditional interoperability methods. According to SEI report [10], complexity is a new perspective: "architecture is not purely a technical plan for producing a single system or closely related family of systems, but a structuring of the design spaces that a complex design process at an industrial scale will explore over time". Breaking up an architecture into design spaces and striving for a set of coherent and effective design rules would seem to imply a significant degree of control of the overall design and production process. Nevertheless, the design spaces, design rules, and organizations will be continually adjusting and adapting to both internal and external forces, which makes it difficult to handle them all.

The criticality of the research is justified by the fact that handling the large volume of information available in ULS digital ecosystems is only feasible by utilizing a well-developed interoperability framework. A newly proposed framework is expected to broaden a traditional interoperability framework to include people and organizations; social, cognitive, and economic considerations; and design structures such as design rules and government policies.

This research work centers on the development of an architectural framework to improve the interoperability of digital ecosystems. We pose the question that given the issues with the design of all levels of ULS architectures, how can one organize and classify the types of information that must be created and used in order to improve the digital ecosystems interoperability?

Interoperability Models

Since the beginning of the last decade, most recent work on architecture development is focused on careful planning and improving an enterprise interoperability framework. Conventionally, such a framework is primarily concerned with establishing a mechanism to describe the concepts, the problem and the knowledge on enterprise interoperability in a more structured manner. This section will survey some recent interoperability models.

LISI Reference Model

The LISI [15] (Levels of Information Systems Interoperability) approach developed by C4ISR Architecture Working Group (AWG) during 1997, is a framework to provide the US Department of Defense (DoD) with a maturity model and a process for determining joint interoperability needs, assessing the ability of the information systems to meet those needs, and selecting pragmatic solutions and a transition path for achieving higher states of capability and interoperability.

A critical element of interoperability assurance is a clear prescription of the common suite of requisite capabilities that must be inherent to all information systems that desire to interoperate at a selected level of sophistication [4]. Each level's prescription of capabilities must cover all four enabling attributes of interoperability known as PAID, namely, Procedures, Applications, Infrastructure, and Data.

The LISI approach is focused on developing interoperability in US military sector. It is also used as a basis to elaborate other interoperability maturity models such as Organizational Maturity Model [16] and Enterprise Interoperability Maturity Model in ATHENA Integrated Project [17].

IDEAS Interoperability Framework

The IDEAS Interoperability Framework was developed by IDEAS project on the basis of ECMA/NIST Toaster Model, ISO 19101, ISO 19119 and was augmented through the quality attributes and intended to reflect the view that "Interoperability is achieved on multiple levels: inter-enterprise coordination, business process integration, semantic application integration, syntactical application integration and physical integration" [18].

In the business layer, all issues related to the organization and the management of an enterprise are addressed. The business model is the description of the commercial relationships between an enterprise and the way it offers products or services to the market. The knowledge layer is concerned with acquiring, structuring and representing the collective/personal knowledge of an enterprise. The ICT system layer is concerned with the ICT solutions that allow an enterprise to operate, make decisions and exchange information within and outside its boundaries. The semantic dimension cuts across the business, knowledge and ICT layers. Quality attributes are a supplementary dimension of the framework. The considered attributes are: (1) Security; (2) Scalability; (3) Portability; (4) Performance; (5) Availability; (6) Evolution.

ATHENA Interoperability Framework

The ATHENA Interoperability Framework (AIF) [17] is structured into three levels: (1) The 'Conceptual' level is used for identification of research requirements and integrates research results; (2) The 'Applicative' level is used for the transfer of knowledge regarding application of integration technologies; (3) The 'Technical' level is used for technology testing based on profiles and integrates prototypes.

The AIF and the IDEAS Interoperability Framework are considered complementary [4]. At each level of AIF, one can use the IDEAS interoperability framework to structure interoperability issues into three layers (business, knowledge and ICT) and a semantic dimension.

European Interoperability Framework

The European Interoperability Framework (EIF) [19] aims at supporting the European Union's strategy of providing user-centered e-Government services by defining as the overarching set of policies, standards and guidelines, which describe the way in which organizations have agreed, or should agree, to do business with each other.

This EIF is defined as the overarching set of policies, standards and guidelines, which describe the way in which organizations have agreed, or should agree, to do business with each other. EIF provides recommendations and defines generic standards with regard to organizational, semantic and technical aspects of interoperability.

Organizational Interoperability Maturity Model

Clark and Jones [16] proposed the Organizational Interoperability Maturity Model (OIM), which extends the LISI model into the more abstract layers of command and control support. OIM extends LISI to cover organizational interoperability. Five levels of organizational maturity, describing the ability to interoperate, are defined. These include: (1) Independent; (2) Ad-hoc; (3) Collaborative; (4) Combined; and (5) Unified.

Layers of Coalition Interoperability

Layers of Coalition Interoperability (LCI) [20] is a model to deal with possible measures of merit to be used with the various layers of semantic interoperability in coalition operations. In the LCI, interoperability is definitely not only limited to the technical domain, but also is dependent on organizational aspects.

LCI tries different aspects to implementing the relations between the interoperability levels. It also stresses the important role of a common unified language for the integration of the interoperability levels from technical up to organizational.

Other Relevant Interoperability Models

In the United Kingdom, the e-Government Unit7 (eGU), has based its technical guidance on the e-Government Interoperability Framework (e-GIF) [21]. e-GIF mandates sets of specifications and policies for any cross-agency collaboration and for e-government service delivery.

The E-health interoperability framework [22] developed by NEHTA (National E-Health Transition Authority) initiatives in Australia, brings together organizational,

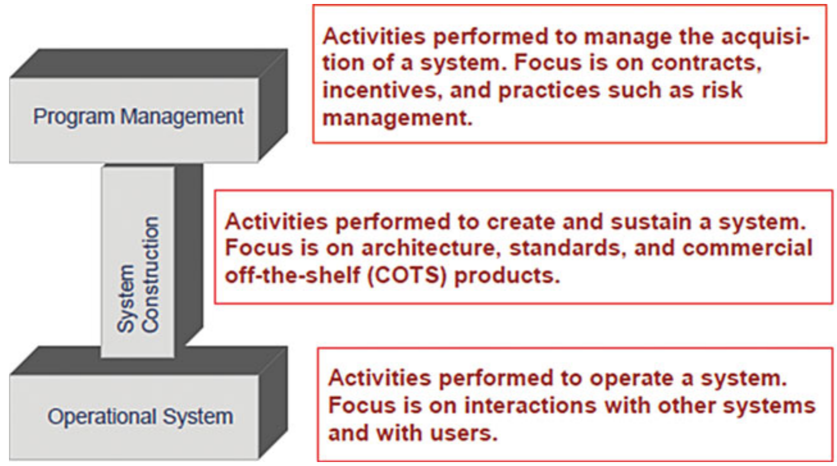

Fig. 1 System of systems interoperability [18]

information and technical aspects relating to the delivery of interoperability across health organizations.

The NATO C3 Interoperability Environment (NIE) [23] encompasses the standards, the products and the agreements adopted by the Alliance to ensure C3 interoperability. It serves as the basis for the development and evolution of C3 Systems.

The models previously discussed address a range of interoperability issues from technical to coalition organizational. SEI has developed the System of Systems Interoperability (SOSI) [24], which addresses technical interoperability (also covered by LISI, LCI, and NATO C3) and operational interoperability (also covered by OIM and LCI). However, SOSI goes a step further to address programmatic concerns between organizations building and maintaining interoperable systems. SOSI introduces three types of interoperability (see Fig. 1): (1) Programmatic, interoperability between different program offices; (2) Constructive, interoperability between the organizations that are responsible for the construction (and maintenance) of a system; (3) Operational, interoperability between the systems.

Digital Ecosystems Interoperability

As introduced in previous section, the SOSI can be considered as a significant initiative for digital ecosystems interoperability. However, as mentioned in SEI report [10], people will not just be users of a digital ecosystem; rather, they will be part of its overall behavior. In addition, the boundary between the system and user/developer roles will blur. Just as people who maintain and modify a city, may also reside in the city, in a digital ecosystem, a person may act in the role of a traditional user, or in a supporting role as a maintainer of the system health, or as a change agent adding and repairing the functions of the system.

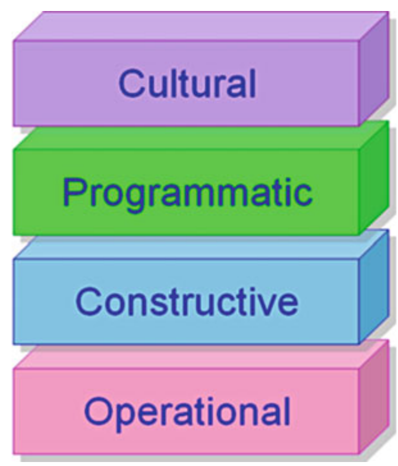

Fig. 2 Digital ecosystems interoperability model

Socio: The New Concern

Assuming that people are part of a digital ecosystem signifies that a new perspective has to be taken into account: culture. Figure 2 depicts an extension to the SOSI model in order to achieve socio-technical digital ecosystems characteristics.

The four layers of digital ecosystems interoperability model corresponds to the four layers of complex system theory model. In complex system theory, we can divide a system into four layers: (1) vital; (2) psyche; (3) social; (4) cultural [14].

The digital ecosystems interoperability model address a range of interoperability issues from operational to cultural. In order to achieve socio-technical interoperation among systems, a set of cultural, management, constructive, and operational activities have to be implemented in a consistent manner. These activities require to support adding new and upgraded systems to a growing interoperability web.

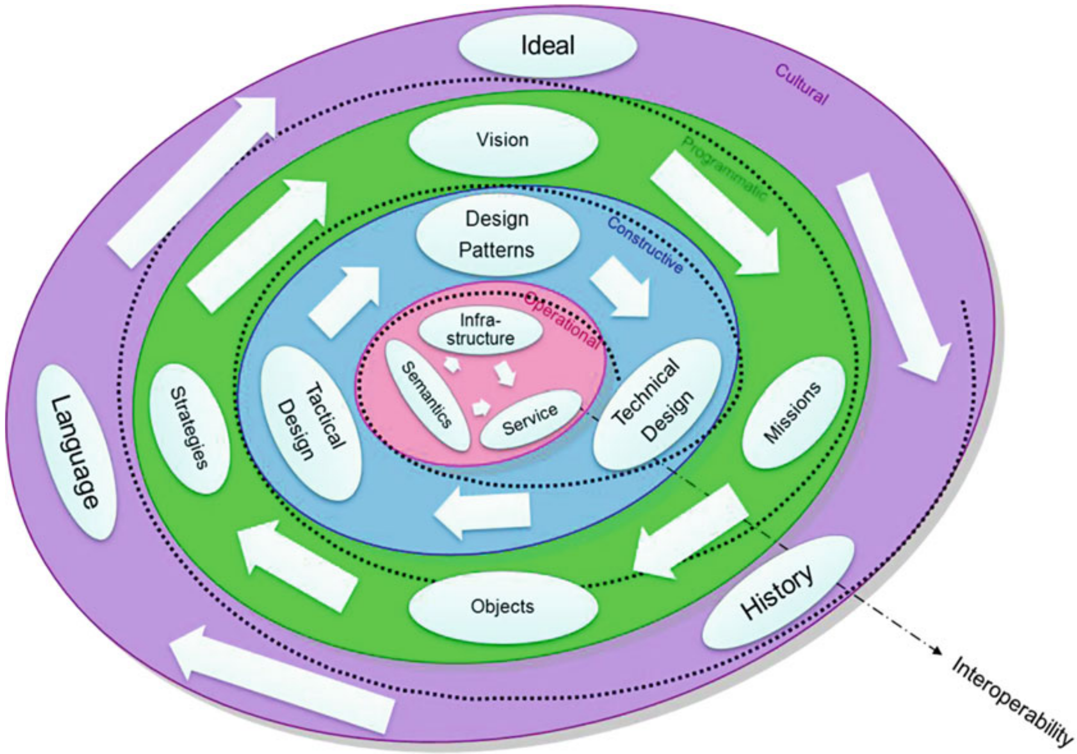

Fig. 3 Digital ecosystems interoperability model details [14]

- Operational issues define the activities within the executing system and between the executing system and its environment, including the interoperation with other systems.
- Constructive issues define the activities that develop or evolve system interoperability.
- Programmatic issues define the activities that manage the acquisition of system interoperability.
- Cultural issues define the activities that sustain the ULS system socio-technical characteristics.

As we mentioned earlier, the proposed model should be a spectrum of technologies and methods with software engineering, economics, human factors, cognitive psychology, sociology, systems engineering, and business policy. The Operational layer divided to three sub layers: (1) Infrastructure; (2) Service; (3) Semantics. The Constructive layer is divided to: (1) Tactical Design; (2) Technical Design; (3) Design Patterns.

The programmatic and cultural layers can also be divided to some sub layers. In the programmatic layer, we have four sub layers: (1) Vision; (2) Missions; (3) Objects; (4) Strategies. Finally, Cultural layer can be considered as combination of ideal, history, and language layers. Figure 3 depicts the details of digital ecosystems interoperability mode.

Digital Ecosystems Interoperability Framework

Zachman Framework (ZF) [5], originally proposed by John Zachman, is often referenced as a standard approach for expressing the basic elements of information system architecture, and is widely accepted as the main framework in ISA. Although some of today's successful ISA frameworks (including ZF) are used for enterprise systems architecture, the problem discussed in the previous section is inherently broader and deeper than current capabilities of ISA frameworks [25–27]. Figure 4 depicts our initiative proposed framework to improve interoperability based on complex system theory. In this work, we apply ZF as an initial start and try to enrich it by digital ecosystems interoperability model to support the special characteristics of socio-technical interoperability. The proposed framework should be a spectrum of technologies and methods with software engineering, economics, human factors, cognitive psychology, sociology, systems engineering, and business policy.

The proposed framework uses three basic dimensions: (1) The abstract dimension is based on six general questions required to understand interoperability; (2) The perspective dimension is based on interoperability concerns in an enterprise; (3) The final dimension is based on interoperability barriers in a socio-technical ecosystems.

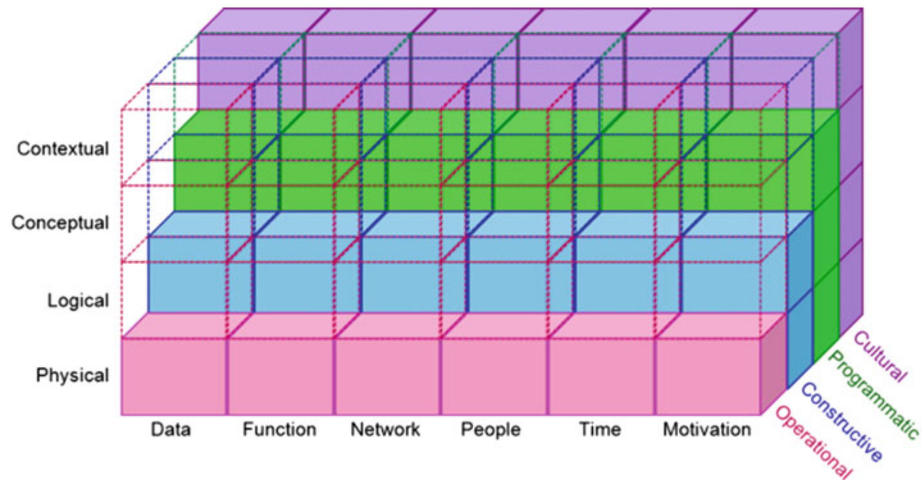

Fig. 4 Digital ecosystems interoperability framework. *Blank cells* are not supposed to be modeled

The interoperability abstracts define the contents of interoperations:

- Data (What?): The interoperability of data handles information finding and sharing from heterogeneous data sources. These data sources possibly exist within different machines running different operating and data management systems.
- Function (How?): The interoperability of function takes care of identifying, composing and making various application functions work together.
- Network (Where?): The study of interconnecting the internal networks of companies is essential in a networked enterprise. This facilitates the creation of a common network for the whole enterprise. This type of interoperability focuses on the geometry or connectivity of the system's physical nodes.
- People (Who?): It focuses on the people and the manuals and the operating instructions or models they use to interoperate their tasks/duties.
- Time (When?): It is concerned with the life cycles, the timing and the schedules used to interoperate activities.
- Motivation (Why?): It focuses on goals, plans and rules that prescribe policies and ends which guide the enterprise interoperability.

The interoperability perspectives define various concerns of interoperation:

- Contextual: It describes the artifacts that provide the boundaries for the interoperability.
- Conceptual: It focuses on the artifacts that conceptually define the interoperability from the enterprise owners' perspective.
- Logical: It describes the artifacts that design the way interoperability will be realized systematically, quite independently of any technologies.
- Physical: It focuses on the artifacts that define the interoperability implementation based on the general technological constraints being employed.

The interoperability barriers address a range of interoperability issues from operational to cultural. Together, the abstract, perspective and barrier dimensions constitute the digital ecosystems interoperability framework. The two dimensional matrix (abstract $\times$ perspective) defines the contents of interoperations that take place in various levels of system perspectives. The third dimension enables to capture and to structure the type of interoperation.

Conclusions

Achieving digital ecosystems interoperability involves changes to the way we define life, including acquisition practices and guidance, technologies, engineering and management practices, operational doctrines for both the usage and those who support the systems. Realizing this vision requires that we begin to define approaches and models in more concrete terms.

In this paper, an architectural framework to improve digital ecosystems interoperability was proposed. The framework presents a classification schema for descriptive representation of digital ecosystems and allows software architects to model various aspects of socio-technical interoperability. The goal is that the framework be used to complement a full-structural model within the socio-technical interoperability.

In the future work, one is expected to propose a methodology to help architectures model the framework cells.

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Towards Multi-gigabit OFDM Throughput on FPGA: Limitations and Challenges

Goce Dokoski and Aristotel Tentov

Abstract

This paper explores the possibility of achieving multi-gigabit data throughput over wireless networks, by using the FPGA technology. It focuses on the implementation complexity of an Orthogonal Frequency-Division Multiplexing (OFDM) modulation, as the most complex part of a physical layer in an 802.11 wireless network.

The calculations are performed for a typical OFDM system, as synthesized in the Xilinx ISE Studio for the most popular Xilinx FPGA architectures. The results present the upper processing speeds that can be expected from an FPGA implementation of an OFDM modem.

Keywords

OFDM • Throughput • FPGA • Fast Fourier Transform

Introduction

The newest 802.11 wireless standards such as 802.11ac, 802.15.3.c and especially the 802.11ad are aimed at raising wireless network throughput up to new limits. The goal is to utilize higher bands of the electromagnetic wave spectrum, and achieve multi-gigabit data transfer speeds [1–3]. Support from the underlying networking hardware in this case is a very important concern. It is clear that ASIC technology provides sufficient processing speed, although at the price of a longer and more expensive development process. FPGA technology on the other hand, is a very popular choice for networking hardware. It provides the flexibility needed in the ever-changing field of wireless standards and protocols, at the same time providing descent processing capabilities. It is also the most affordable choice for academic research.

The highest speeds defined by the 802.11 standards are achieved by using the OFDM type of modulation. Therefore its implementation on FPGA is chosen as the main topic for discussion in this paper.

It should also be noted that the most important processing part of the OFDM modulation is the Fourier transform, usually implemented as one of the many Fast Fourier Transform (FFT) algorithms.

In the following section, the paper presents the current state of the art. Afterwards, in the third section it gives a brief design-space exploration for OFDM implementation on FPGA. The performances that can be expected from an FPGA technology are also estimated. In the fourth section it elaborates the Fast Fourier transform. The fifth section presents the performance characteristics for a typical OFDM system, as synthesized in the Xilinx ISE Studio for several popular Xilinx FPGA boards. Then, according to the estimations of the second section, the maximally achievable throughputs are shown if parallel processing were used. The last section gives a conclusion of the obtained results.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_66, © Springer International Publishing Switzerland 2015

State of the Art

There are many FPGA implementations of OFDM modems and also many related publications, [4–13]. Xilinx FPGA boards are widely used in these publications. In [5] an OFDM modem is modeled and synthesized for Xilinx Virtex 2 that can work at a frequency of 100 MHz, whereas in [6] the same board achieves a working frequency of 200 MHz for a 128 point FFT/IFFT. In [7] an OFDM implementation is synthesized for Virtex 4, reaching a working frequency of 227.355 MHz. Reference [9] presents an 802.11a/g WLAN receiver implemented on Virtex-II proto board. Reference [10] presents FPGA design, validation, and implementation of an OFDM modulator for IEEE-802.16e. In [11, 12], OFDM implementations on the Xilinx Spartan 3 board are shown.

There aren't many publications that discuss the FPGA upper speed limits, or that give an overview of the opportunities for high-speed OFDM data throughput in wireless networking when using the FPGA architecture.

OFDM Design-Space Exploration

OFDM Computation

The basic OFDM computation scheme is shown in Fig. 1. It consists of three steps:

1. Data mapping. The input data array is mapped onto the OFDM carriers by using the Quadrature Amplitude Modulation (QAM), or Phase-Shift Keying (PSK) scheme;
2. Fourier transforms. On the transmitter side, the mapped symbols are treated as a wave frequency spectra, and are used to generate the wave to be transmitted by using IFFT; On the receiver side, the received samples are reverse-transformed into frequency domain data by using FFT;
3. Cyclic prefix insertion. The end of the resulting waveform is copied at its beginning;

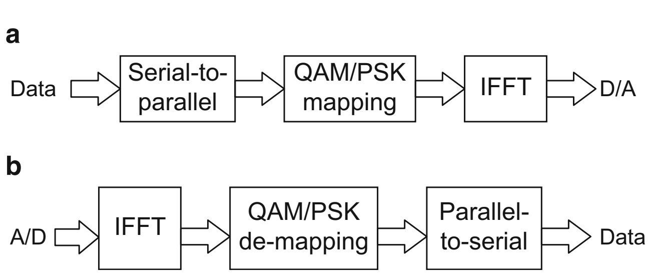

Fig. 1 The process of digital OFDM computation. (**a**) OFDM modulation on the transmitter side. (**b**) OFDM demodulation on the receiver side

The same steps in reverse order are performed at the receiving side.

Symbol mapping/de-mapping is easily implemented by using ROM/LUT slices. These are the basic FPGA building blocks so this implementation method is the most appropriate for the FPGA technology.

The cyclic prefix insertion is also easy to implement, and the delay it brings to the pipeline architectures is inevitable, regardless of the implementation.

Theoretical Speed Limit on FPGA

When implementing a single OFDM module, it is easy to estimate its upper theoretical speed limit. Assuming that the highest achievable frequency for the OFDM module on a particular FPGA board is $f_{\max}$, then this will also be the highest possible data input frequency. So if the maximal number of symbols that can enter the module per clock cycle—N_{SC} is equal to $f_{\max}$, then the maximally achievable data throughput $Th_{\max}$ will be N_{SC} multiplied by the number of bits that are carried by each symbol—N_{bS}:

$$Th_{\max} \leq N_{SC} \cdot N_{bS} = f_{\max} \cdot N_{bS}, \tag{1}$$

where $N_{bS} = \log_2 M$, for M-QAM mapping.

For example, a single OFDM module working at a frequency of 350 MHz can achieve a maximal throughput of 2.1 Gb/s.

This estimation only takes into account the data input throughput, and it is valid regardless of the implementation algorithm.

From Parallelism to Multi-gigabit Throughput

Given the FPGA working frequency limits, parallelism becomes the obvious step towards increasing the OFDM throughput. Putting several OFDM modules to work in parallel, allows for multiple increase in the OFDM computation speed. The only condition is parallel data input with double buffering. Each buffer's size B_{SIZE} should be equal to the number of OFDM modules N_{MOD} multiplied by the number of bits input to a module per clock cycle N_{bS}:

$$B_{SIZE} = N_{MOD} \cdot N_{bS} \tag{2}$$

The double buffering concept consists of filling one buffer with data, while the other one is providing the data to the OFDM modules, and vice-verse. The data throughput obtained by parallel modules is shown in (3). This is possible because of the parallel data input, and the increased number of bits that are processed per clock cycle.

$$Th_{\max} = N_{MOD} \cdot N_{bS} \cdot f_{\max} \quad b/s, \tag{3}$$

where N_{MOD} is the number of OFDM modules working in parallel.

Fourier Transform Implementation

The Fourier transform is most often performed by using one of the many Fast-Fourier Transform (FFT) algorithms, the most well-known of which is the Cooley-Tukey algorithm, [14]. There are many FFT algorithms and their variations, and their implementation can be categorized into three main categories:

- *Pipeline*, achieves maximal operation speeds, but has largest overall delay;
- Partially parallel - one stage of processing elements (butterflies) is used for several stages. This method gives medium balance among speed, resource usage and delay;
- Fully parallel—all stages are fully implemented. This method achieves high speeds with low delay, but also has biggest resource needs.

The Xilinx LogiCORE IP FFT v7.1 core generator, implements the Cooley-Tukey FFT algorithm, and offers these three implementation methods: *pipelined* (Streaming I/O), *partially parallel* (Radix-2 Lite) and *fully parallel* (Radix-4/Radix-2 Burst I/O), [15].

The most useful implementation for the 802.11 family of protocols is the pipelined option as it allows continuous data processing.

OFDM Synthesis Results

The basic OFDM system shown in Fig. 1 was implemented and synthesized in Xilinx ISE Studio v.14.5, by using the Xilinx LogiCORE IP FFT v7.1 generator for the FFT transformations. The pipelined (Streaming I/O) architecture was chosen for the Fourier transforms. The rest of the processing was programmed in VHDL.

The system was synthesized for several popular Xilinx FPGA architecture families, such as Artix, Kintex and Virtex. We chose two boards from each family—one with the smallest and one with the largest capacity, so that the range of implementation possibilities can be easily perceived.

The synthesis results for a single transmitter are shown in Table 1. The implementation characteristics of a basic receiver are similar and are therefore omitted.

As it can be seen in Table 1, the maximal working frequency for a single OFDM module varies from 350 – 540 MHz. According to (1), this means that the maximally achievable throughputs will range from 2.1 Gb/s up to 3.2 Gb/s, if a single OFDM module with a 64-QAM mapping were used.

In order to achieve higher speeds, with the same mapping, parallelization must be used. According to the percentages shown in Table 1, the boards with smallest capacities can fit from one up to six OFDM transmitters, whereas the largest capacity board can fit around 40. So the maximal throughput with this board potentially reaches more than 100 Gb/s.

The maximal throughputs achievable by the FPGA boards of Table 1 are shown in Fig. 2. These values are calculated according to (3), with appropriate values for N_{MOD} chosen according to the device utilizations from Table 1.

Table 1 Synthesis results for an OFDM module implemented on Xilinx FPGA boards

Board		Freq. (MHz)	Utilization (%) LUT slices	18 K Block RAM
Artix 7	xc7a100t	353.201	42.13	4.44
	xc7a200t	395.023	19.84	4.00
Kintex 7	xc7k160t	466.385	26.34	1.84
	xc7k480t	541.038	8.95	0.63
Virtex 5	xc5vlx50t	383.613	92.74	10.00
	xc5vlx110t	383.613	12.88	1.85
Virtex 7	xc7vx330t	466.385	13.09	0.80
	xc7v2000t	466.385	2.19	0.47

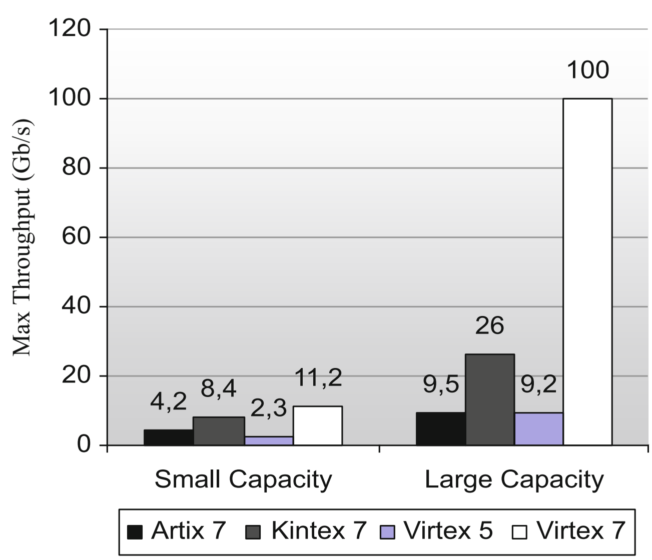

Fig. 2 Maximal throughput of OFDM transmitter by using parallelization on Virtex FPGA boards

Conclusion

The FPGA Design-Space for OFDM implementation is limited to a large degree by the FPGA architectural characteristics and operating speed. Therefore, the only trade-off available is the amount of utilized FPGA resources and their power consumption.

The results show that the newest boards from Xilinx have sufficient capacity and speed to perform OFDM modulation with throughput of up to hundred gigabits per second. These speeds however can only be accomplished by using parallelization, regardless of the implementation algorithm for a single core. The assumptions concerning parallel processing are theoretical, and a complete parallel implementation is needed for a definite confirmation. This is planned as a future work.

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Integral KNIT Cluster as a Source of Development Factors and Technological Frame for the Realization of Country Development Projects

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Abstract

The purpose of the paper is to describe how the structure of knowledge, innovation and technologies' cluster should be formed. The structure of the cluster is understood as a set of interacting components of the system. The authors synthesize the existing opinions and also reveal their own opinion about the possibility to determine the interaction among knowledge, innovation and technologies in the context of value creation or resource management, pursuing the preparation of the project for country universally sustainable development. The paper presents the original evaluation performed by the authors in order to optimally allocate the investment resources for knowledge, innovation and technologies among distinctive subsystems of universally sustainable development, as well as to optimally allocate the investment resources inside the cluster among the creation and development of knowledge, innovation and technologies. For the solution of the mentioned problem the methods of stochastic informative expertize and the adequate investment portfolio have been used.

Keywords

Cluster • Knowledge • Innovation • Technology • Development Projects

Introduction

Sustainability, as orientation of activity towards the today's needs satisfying, leaving for future generations the possibility to satisfy their needs as well, is the main concept of science capable of finding the solution for the relevant problem. Today the category of sustainability is highly demanding the adequate appreciation and engagement in science, as well as in practice [1–5].

Indeed, sustainable development and purposeful integration of science, innovation and technology are the main instruments for the world to overcome the current and potential future challenges [6–8]. The rapidly evolving globalization and the shaped demand for development sustainability of the country stimulate the continuous integration of knowledge, innovation and technology. However, the contemporary researches on sustainable development lack the quantitative measures and the proportionately unified concept and expression of sustainability. Moreover, the practice of measurement of the sustainability of a system or a process, especially when this system is a country, region or any other complex system, is in its beginning decision stage [9, 10].

Going deep into "The Post–2015 UN Development Agenda", prepared by the United Nations (UN) [11], one can see that for both the ensuring of global economic growth and the increase of growth potential of individual region or country, as well as solutions for such global problems as food, health or ecological unsustainable growth, can be successful only using the purposefully developed cluster of knowledge, innovation and technology (KNIT). We do not have any alternative for the purposeful development of science knowledge and space technology, if we consider the possibilities of global and space disasters.

The EU Future Programme 2014–2020 [12] quite similarly treats the future challenges and possible ways to avoid the consequences. As well as the Post-2015 UN

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DOI 10.1007/978-3-319-06764-3_67, © Springer International Publishing Switzerland 2015

development agenda, it strongly speaks for the science research shifting from interdisciplinary research towards problematic researches.

Both the first and second of the mentioned programs and a number of other UN and EU documents very responsibly describe the development policies and priorities, potential problems and their solutions, discuss the local and global challenges. And almost everywhere the unified approach to the development sustainability as a useful and measurable way to the perspective is seen. Also, the integral KNIT cluster structure appears to be the adequate and the most effective tool to achieve the state of sustainable development.

The following relevant problems that received a cautions attention should be distinguished among the many of projected science research and pragmatic KNIT structure improvement projects.

- what is the compatibility of boundless economic and demographic growth with the limited possibilities of the planet Earth;
- what are practically unmanageable or even artificially promoted negative consequences of globalization, could they ruin all benefits provided by globalization;
- what should be the concept and models for quantitative measurement of sustainable condition or development pertaining to the process or system;
- what are the mechanisms of KNIT cluster optimal structure formation in order to prevent the boundless sources of development factors from becoming unobtainable;
- what forms of integration and communication between different states should dominate in the period of post globalization.

In essence, these are the most complex global political problems, and their efficient solution can be the adequate broadening of KNIT cluster. However, the subjective interests and factors probably would dominate here. On the whole, the preparation of the active KNIT cluster allows understanding easily the main intelligent investment principles seeking for country development sustainability.

The Features of The Formation of Integral KNIT Cluster Structure

Cluster as the Category Integrating the Development Interests and Possibilities

If you look at the Oxford Modern English Dictionary [13], the main meaning of cluster is named as a "close group or bunch of similar things growing together". Cluster analysis or clustering is the task of grouping a set of objects in such a way that objects in the science group (called a cluster) are more similar (in some sense or another) to each other than to those in other groups (clusters).

The methodology of cluster analysis is common in many human activities, social organization and the cognition of natural nature processes and management spheres and its ways and tolls of practical use are very different. According to Vladimir Estivill-Castro [14], the notion of cluster cannot be precisely defined, which is one of the reasons why there are so many clustering algorithms.

At this article the authors are focused on the analysis of possibilities of the integral KNIT cluster while projecting the strategy of country universally sustainable development and providing the possibilities for implementation of such strategy. The functional interaction of the cluster components now is intended towards the solution of the key economic problem—the rational allocation of scarce resources.

Here the integral KNIT cluster has a double advantage in terms of many objects, but it is especially clear when we talk about the projection and implementation of individual country sustainable development. Firstly, the integral KNIT cluster is universally recognized as an inexhaustible source of the development factors [15–17]; and, secondly, the integral KNIT cluster can and should serve as a technology of the adaptive complex system [18–20] implementing the sustainable development strategies of countries with no abundant natural resources.

The supplement of the selected concept of cluster and developing its functional purpose should not cause incompatibility with the content and interaction of development of knowledge, innovation and technologies. The goals and irrational development ways of selected process or system are based on the historically long industrial experience of clusters, based on possibilities of knowledge, innovation and technologies.

There is no doubt that the structure of KNIT cluster depends on the object, and the necessary information for its cognition and management is generated using KNIT. In any case, the structure of KNIT cluster can be expressed only on the example of fuzzy set (Fig. 1). It is clear that without the assumption that the object is in the "focus" center, it is difficult to talk about Fig. 1, because it is the collection of

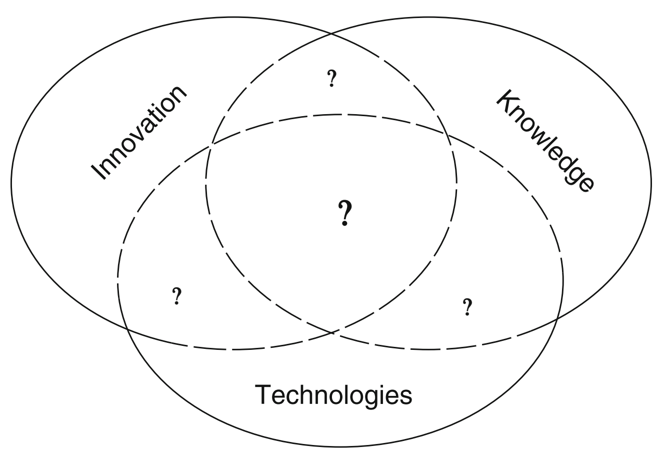

Fig. 1 The integral KNIT cluster

unrelated (not linked by the needs of the mentioned object) knowledge, innovation and technology digests. Also, there is no doubt that with the change of KNIT object, the content of cluster components and the structure of the cluster also changes.

Pragmatic Description of the Integral KNIT Cluster Structure

Often, for the initial understanding of KNIT cluster structure, the model of interaction of simplified military part units is invoked, while the military unit faces a previously unknown enemy. Then the role of intelligence unit the knowledge subsystem is assigned, that using all the possibilities helps creating an adequate image of the enemy, along with that forming a vision of what should be: the operating unit of military actions—the investment subsystem, and the unit of strategic actions—the technology subsystem.

In other words, the authors of the paper can generalize such a statement: the knowledge subsystem is the substance which highlights what we know and what we need to know, innovation—the substance ensuring the problem solving in the field of operational development, and technology—the substance mobilizing strategic implementation of objectives through knowledge and innovation. There is no doubt that the process of development transformation is an integral part of the development code, which includes the most important information about what happens if the subsystem is not performing its functions and development process stagnates.

In fact, the interaction of scientific knowledge, innovation and technologies even in a simple situation remains complicated and especially important problem of science efficiency management. The main focus of the experiment, which was intended to explain the specific problem by the specific evaluation and the obtained solutions, was intended for the formulation of the principles for cluster structure determination, based on the investment expenses, and for revelation of management possibilities of this structure, seeking to implement the purposes and functions of the cluster.

Integral KNIT Cluster as a Main Source of Universally Sustainable Development Factors Pertaining to a Country

Considering the development project of many countries, especially if they do not dispose the abundant natural resources, the idea is being unambiguously revealed that the main and inexhaustible resource for their development becomes an integral cluster of scientific knowledge, innovation and technologies. The concept "inexhaustible" in the last sentence requires a special attention. Since this factor is both naturally evolving and purposefully educated, there is probably no need to talk about its inexhaustibility. However, on the other hand, recognizing that future problems become more sophisticated, and negative processes in many areas of human existence obtain the catastrophic speed, we need to understand that even if the resource remains everlasting, but for many subjects, including individual countries, it may become unattainable.

There is no doubt that the integral KNIT cluster efficiency evaluation problem should become the object of exclusive attention of national and global science. Unfortunately, little work deals to propose a pragmatic solution for the latter problem. What should be the structure of the integral KNIT cluster, recognizing that the categories of knowledge, innovation and technologies mean the implementation of different functions, and the need for financial resources is also formed in different ways?

In our experiment the object of integral KNIT cluster is the projection of universally sustainable development pertaining to a country. The concept of universally sustainable development is quite extensively presented for the scientific community (see examples in references [21–24]), so there is no need to talk about its content and constructivism.

However, it is worth mentioning that the concept of universally sustainable development is based on the idea of the round table, which is composed of the four components (subsystems), bringing together the sustainability of development through attaining the respective targets [21]:

- 1st subsystem: social, economic, ecological targets;
- 2nd subsystem: educational, creative and religious targets;
- 3rd subsystem: political, integrative and managerial targets;
- 4th subsystem: energetic, financial and investment targets.

The idea of the round table highlights the fact that in the selection of final solution the interests of all development sustainability components or just the experts representing those interests should participate; otherwise there should be adequately formulated criteria. The idea of the round table helps to express a provision that there should be a possibility to quantify and coordinate these interests.

The title of this section appeals to the fact that the integral KNIT cluster should be a key source of the universally sustainable development factors. All the components of universal development sustainability require the help of KNIT for generation of required knowledge, innovation and technologies. It is evident that the cluster has to be adapted to meet the needs of a specific object—the projection of universally sustainable development pertaining to a country.

The Experiment of Integral KNIT Cluster Structure Optimization

Experimental Solutions to Optimize the Allocation of Investment Resources Among Development Subsystems of the KNIT Cluster and to Optimize the Structure of Integral KNIT Cluster

To formulate and solve the problems related to sustainable development and the interaction of its subsystems, one needs to invoke a whole of tools: the systems of knowledge information, decision management and uncertainty assessment, as well as stochastic models of quantitative decisions and expert valuation. An exceptional moment here is the assessment of individual problems, when using collected and generated information, the compatibility between the different aspects of development is found, and the fact that the expert assessment is focused on the methods of stochastic informative expertize. The system is focused on the formation of quantitative possibilities for interviews, while projecting or analyzing the development system of a country. Practically this means that the information about changes and emerging problems in any subsystem or component is sent to other subsystems, which, in turn, give their own response.

The countries or regions, for which there is no alternative besides the integral KNIT cluster as a source of development factors, should be interested in efficiency of economic KNIT cluster and especially the problems of rational use of investment resources. These problems could be solved by optimizing the investment resources among the possibilities of KNIT in separate development subsystems and by optimizing the structure of KNIT cluster itself. Further the experimental version to solve these problems is presented.

The universal sustainability of a country is measured by an exponential growth function:

$$y = e^{f(x_1)+f(x_2)+f(x_3)+f(x_4)} \tag{1}$$

where:
y—projected integral (gross) index of development sustainability;

x_i—the law defining how the marginal investment unit in the i^{th} component of development converts into potential component of integral sustainability index (i = 1, 2, 3, 4). (see Fig. 3);

$f(x_i)$—the function describing how a mentioned potential possibility is transformed into a real integral component of sustainability index in the context of first equation.

In this way we associate the possibilities of investment with the volume of development components sustainability growth and, in turn, with the subsystem of integral sustainability.

The first dependence indicates that in the absence of investment in a point in time, the growth possibilities of the index of the integral content also disappear. So, intelligent distribution of investment unit may be quite informative also about dynamics of sustainability.

Thus, the distribution of marginal unit between the development components of KNIT clusters, in order to obtain the maximum growth of integral sustainability index, will be perceived as the above mentioned criteria of problem—how to distribute the marginal investment unit among all the development subsystems of KNIT cluster to achieve the best growth of integral sustainability.

Similarly, we also have to formulate the problem of integral KNIT cluster structure optimization. Here the functional dependence is used:

$$y = e^{f(\bar{x}_1)+f(\bar{x}_2)+f(\bar{x}_3)} \tag{2}$$

where y—the effect which is given by the marginal investment unit among distributed components of KNIT cluster;

$\overline{x_i}$—the law describing the i^{th} component of contribution of marginal investment unit, forming the potential effect of i^{th} component;

$f(\overline{x_i})$—the function describing how the potential possibility is transformed into the real i^{th} component and its impact on general KNIT cluster.

$\overline{x_1}$—knowledge, $\overline{x_2}$—innovation, $\overline{x_3}$—technologies.

Thus, as in case of the above mentioned problem, we obtain that the $x_i$are the laws describing how the marginal i^{th} component of the cluster generates the potential benefit and $f(x_i)$ describes an algorithm how to evaluate the contribution for the real benefit of investment unit in the context of second function.

Here the criteria of optimization can be identified—it is the distribution of marginal investment unit among the knowledge, innovation and technologies.

Since the investment unit becomes the indicator of the potential sustainability in a particular subsystem of development, and measuring the integral development sustainability by the real sustainability indicator they are the stochastic processes, thus solving the problems of derived rational distribution of resources among the various development components of KNIT cluster and it is the optimal structure of development, the integral KNIT cluster structure establishment, we will use the technique of adequate investment portfolio stochastic optimization (Fig. 2).

General Results of the Evaluation

Practically, now it is impossible to gather the statistical data or the analytical estimates which lets to evaluate the (1) and (2) parameters. Therefore for the description of parameters

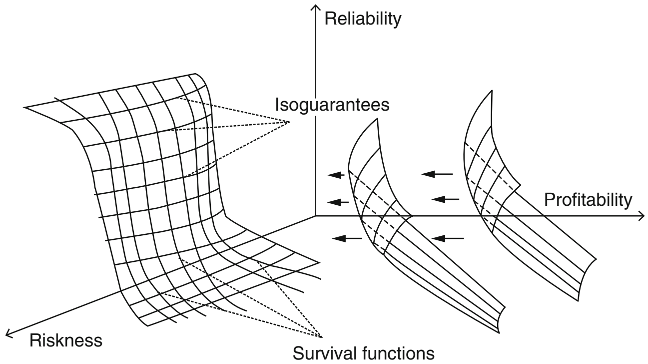

Fig. 2 The scheme of stochastic optimization problem solution, where a set of the potential solutions is the network of isoguarantees and survival functions, and the utility function is composed of the set of functions depending on the efficiency, reliability and riskiness of possibilities

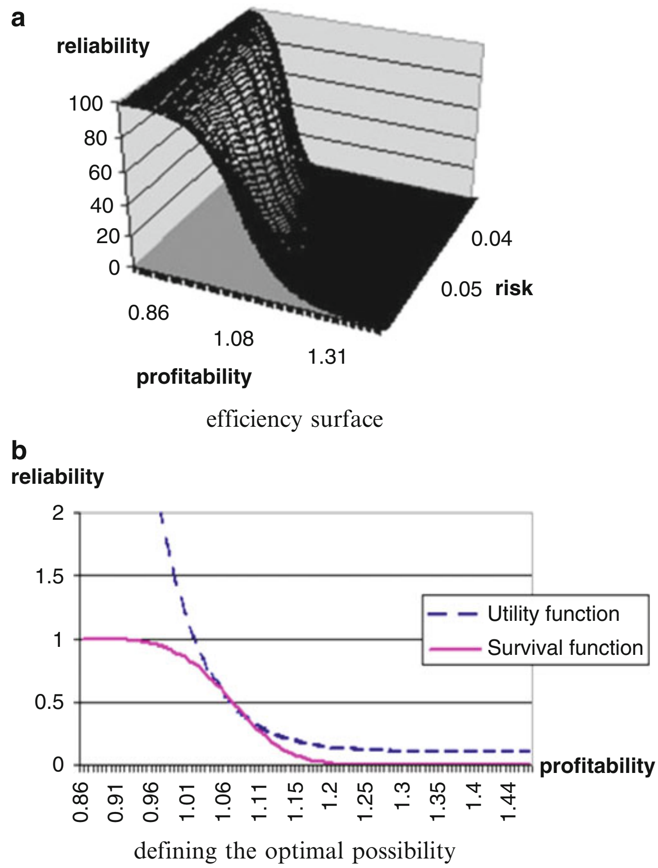

Fig. 3 Optimal allocation of resources among four subsystems. (**a**) efficiency surface (**b**) defining the optimal possibility (**c**) numerical characteristics of the optimal solution

necessary for the solutions of first and second task, the groups of experts were selected, who very actively used the principles of the stochastic informative expertize.

The experts suggest that rates of the laws or indicators $x_1, x_2, x_3, x_4, \overline{x_1}, \overline{x_2}, \overline{x_3}$ should be the probability distributions with the appropriate forms, probable averages and standard deviations.

x_1—is the normal distribution with the average α=0.03 and standard deviation σ=0.01 or simply N ($\alpha_1 = 0.05$, $\sigma_1 = 0.02$);

x_2—is the lognormal distribution: LN ($\alpha_2 = 0.065$, $\sigma_2 = 0.02$);

x_3—is the Gumbell distribution: GB ($\alpha_3 = 0.02$, $\sigma_3 = 0.009$);

x_4—is the Triangle distribution: TR ($\alpha_4 = 0.44$, $\sigma_4 = 0.015$).

Note. For the control of the expert evaluation and justify of decisions parallel were done the evaluations, when all x_1, x_2, x_3, x_4 are the normal distributions with the above mentioned parameters. The results of calculations are shown in Fig. 3.

Table 1 shows the optimal structure solution of KNIT cluster, when the experts have found that:

$\overline{x_1}$—$N(\alpha_1 = 0.04, \sigma_1 = 0.02)$

$\overline{x_2}$—$N(\alpha_2 = 0.04, \sigma_2 = 0.03)$

$\overline{x_3}$—$N(\alpha_3 = 0.07, \sigma_3 = 0.04)$

Thus the experiment is performed on finding the optimal allocation of resources, forming an integrated knowledge, innovation and technology cluster in order to promote the universally sustainable development in Lithuania.

Trying to directly identify and generate the knowledge, implemented technologies and cherished innovations for longer perspective would require the analysis of quite debatable problems since most already set universal component of sustainable development is social—economic. Here, the identification of technologies and innovations and assessment of their need of cost implementation provoked a lot of discussion problems.

Therefore, as a rather simplified scheme for solution of mentioned problem, we will use the model structure of innovative functions of the system submitted by [25]. For solution of the particular problem it is transformed and therefore becomes suitable for the analysis of possibilities of universally sustainable development. It is done with the help of the principles of stochastically informative expertise and the model of adequate portfolio for the optimal allocation of financial resources among four integrated components of universal sustainable development [26]. Very early results of the assessment are provided in Table 1.

Table 1 the optimal allocation of financial resources among the KNIT cluster components: knowledge, innovation and technologies

KNIT components		
Knowledge	Innovation	Technology
The marginal cost per unit of component		
0.19	0.30	0.51

e = 1.1007, p = 0.56, r = 0.331

From Table 1 we can see that 19 % of the investment funds should be allocated to knowledge subsystem, 30 %—to innovation subsystem and 51 %—to technology subsystem in order to achieve the optimal use of investment resources. The effect of such an utilization accounts for 1.1007, the reliability is 0.56, and riskiness of the mentioned effect is 0.331.

Conclusions

The performed analysis of scientific and practical literature, as well as the insights of the authors suggest that the integral KNIT cluster becomes the continuous source of development possibilities for the necessary survival in contemporary era of globalization. Also, the KNIT cluster serves as an adaptive complex system technology while projecting and implementing any development activity.

The functional flexibility and cost structure of KNIT cluster become particularly important optimization problems. The question arises if KNIT cluster can be an inexhaustible source of funds that is financially inaccessible to many subjects including individual countries and regions.

The experiment was performed on finding the optimal allocation of resources, forming an integrated knowledge, innovation and technology cluster in order to promote the universally sustainable development of a country. The particular marginal units of investments into each subsystem of the cluster were obtained. Such an investment allocation and implementation could be treated as a project intended towards the development of country sustainability.

Acknowledgment This research was funded by a grant (No. IEP-01/2012) from the Research Council of Lithuania.

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The Impact of Film-based Learning in Science Education

M. Hosťovecký and J. Štubňa

Abstract

Innovative education of natural sciences consider learning with dynamic image, films and 3D technologies. In the research article we discuss impact and importance of documentaries on the knowledge level of students and their reactions. The research article describes the importance of integrating films into education through documentary films. Evidence of the effectiveness of innovative elements mentioned are the conclusions of the research which is presented in this paper. Currently we continue this research in teaching by watching documentaries on cinema pre-selected topic.

Keywords

Education • Film-based learning • 3D films • Science education

Introduction

Fundamental changes to the greatest extent reflected in media literacy education and film viewers, because the expansion of the development of media and communication media is one of the typical examples of a dynamically changing attribute of social life, which must find its reflection in the school environment. Film and cinema arts are not just modern information and communications, but also a respected artistic one, on which pupils should receive systematic instruction that happens rather randomly and unsystematically without particular efforts to provide a deeper film not only as illustrative supplement the teaching of certain subjects as well as specific artistic expression [1].

Initially, computer technology was viewed as a way to improve students' access to instruction and instructors. The vision was that computer hardware and software would make teaching and learning better and more effective [2].

Scientific knowledge must now acquire every one that focuses on education and qualifications. Science therefore can no longer be focused solely on technology but also on man himself, to develop his intellect, and creative capabilities for the creation of material and spiritual prerequisites for its overall development.

The twenty-first century, we are watching the growing influence of the media, which spread into every corner areas of life as well as the individuals themselves. Globalization brings a tremendous change and opportunities for man and his individuality.

For educational system implies some need to develop creative abilities of pupils and students, their creative thinking and problem solving. There is a need to develop greater awareness, emotional intelligence, prosocial behaviour and shape the noble values such as respect, respect, equality, freedom (limited freedoms of others), goodwill, tolerance, trust, honesty, integrity, mutual assistance and cooperation. To make this possible, the school should become a humanistic and creative institution [3].

The world is perceived and interpreted by the media as well as their means. For the man of today it is increasingly important to understand the media—to understand their

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_68, © Springer International Publishing Switzerland 2015

function, typology, structure, principles on which they operate. Media, especially as modern and efficient means of sharing and shaping of personality, as in recent years increasingly in order of interest to those that deal with educating and training the next generation. We understand the media as a means of sharing a fact, then it is irrelevant what the possibilities and the impact of specific media (books, film, TV etc.) [1].

In terms of education, respectively. education and other related fields, it is useful to analyse the media in a broader context and to examine not only their social interaction, but also pedagogical aspects, especially how to use efficiently the system of education.

The pedagogy therefore a situation where you need to upgrade both the aim and the content of education but also all the resources with which it is possible to increase the effectiveness of educational and training process. Finding new methods of learning is therefore now as important as focusing the students on work productivity. To be at the school level of scientific development, and where it has also actively promote the application of such methods and forms of work, which will enable the rationalization and optimization of teacher and pupil. It should take into account the fact that in learning but also learning plays an important role and motivation and emotional—emotional relationship to certain activities [4].

Define the concept of culture is very difficult. Often referred to as the sum of material and spiritual values created by people throughout history. Their part is art that speaks to man in particular its aesthetic values, acting on his senses and enriches his personality. Art has a unique place in the process of education especially in primary schools.

Recently, the experiential learning disability as a certain theoretical perspective of educational resources, methods and forms that are not based from selling information, but the mediation experience [3].

The term “experiential learning” means a theoretical analysis of such disability and educational processes that work with inducing, analysis and reflection experiential events to gain experience transferable to the next life. For experiential education experience is always only a means, not an end) [4].

Experiential education is a philosophy and methodology in which educators purposefully working with students with direct experience and focused reflection in order to increase their knowledge, improve skills and clarifying your values. Experiential approach to teaching is very important that students first thing practically and actively survivors, did you tried etc. and then at a later stage yourself, or under the direction of the teacher returned to the acquired experience, they realized exactly what was happening during the activity and tried to get out of the lessons for the future.

Since the early 1990s of last century, traditional school became the centre of criticism not only from the educational sciences, but also teachers, parents and pupils. The main object of criticism was the lack of emphasis on developing pupil’s personality, lack of access individual teacher to pupil, overdraw curriculum revaluation of knowledge without application of skills compared to skills, values and attitudes etc. Develop primarily lower cognitive function, dominated mechanical acquisition of knowledge, and ability application was developed to a much lesser extent. Efforts to change in this area related to the total humanization of the school, which was carried out mainly through educational innovation. Their support and introduction to school is one of the important means of carrying out this reform. Under the pedagogical innovation understand certain changes in teaching practice to bring new approaches, strategies, methods, content, organization, while maintaining its original system. This word is synonymous with renewal. In practice, innovation often undertaken an initiative of the so-called “bottom-up” that is as individual innovation in the methods, forms, organization of teaching, which brought and still bring particular creative teachers [5, 6].

For innovative features in teaching teachers considered first activating, participatory methods, less common forms of organization. Innovative indicate any change curriculum content, even the use of ICT or multimedia design and teaching. Increasingly popular is the introduction of elements of proven innovative programs. Aware of the importance and need for innovation in teaching, particularly in the context of the new “needs time”, use this resource as a source of motivation and interest of students in learning (teaching issues). Apply the elements of good and innovative educational programs, regional education and culture, complemented by the content of articles on interesting subject matter, using the interactive whiteboard and other gadgets, and also that teachers themselves creatively produce.

Cinema as an institution, as we know it today with input, since its inception present reality. Well, this first film of the founders of modern cinema Lumiére brothers show real life. Their first film was called “Workers Leaving the Factory”, and was first introduced to the public on 28th December 1895 in the basement of the Grand Café on the Boulevard des Capucines Paris [7].

The beginnings of the film are associated with news reportage is a fact that emphasizes direct link former movie information with information newspaper, where the film was primarily informative function. In teaching film has an extremely important role [8, 9].

The film not only serves as motivation but also the bearers of information. His aim in education is to transfer visual information to the student through a dynamic image [10, 11].

For a huge step forward in the development of audio-visual techniques consider the discovery of a three-dimensional image, the so-called 3D. 3D technology is not as it first appears, achievement twenty-first century. More apt to attribute to them the label “remake”. When does look to film history and begins the era of 3D movies in parallel with 2D movies. Attempts to rotation of the first 3D film dates back to 1890. The true 3D projection was screened for audiences 10 June 1890 at New York’s Astor Theatre. The program was composed of three short films (the first American country, the other was cut from the movie Famous Players’ Jim the Penman and the third was a travelogue about Niagara Falls) [12].

Today, the term is mostly commonly associated with the entertainment and film industry, but has great potential as a useful tool in education. Students especially at a younger age do not understand the phenomenon, which can not be seen. Visual learning enhances their understanding of the way that knowledge through quite understand the meaning of its parts. Research related to the integration of 3D teaching in regular classes showed that over 85 % of students prefer just visual learning. From this perspective, it is just a three-dimensional representation of the image by the most economical and efficient way to communicate abstract and complex concepts [13, 14].

We should not forget that the more realistic 3D image, the more active the senses pupils. Up to a third of students participating in research at some point forgot that the picture is only projected and worked to “catch”. It is these parts of the course will be etched in their memory deepest, which is then reflected in test results. After 4 weeks, the pupils of the experimental classes perfectly remembered concepts screened in 3D, even using body language and gestures to the more detailed description of these concepts. Finally, it should also mention the impact of such non-traditional teaching on motivation, focus and discipline of students. Although 3D part of the lesson ended 96 % of pupils to keep its attention. Just been wondering what will happen next, what the unconventional teacher has prepared for them [15].

History of 3D Films

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The first attempts to rotation 3D film dates back to 1890. The true 3D projection was screened for audiences 10 June 1890 at New York’s Astor Theatre. The program was composed of three short films. The first was the American countryside, the other was cut from the movie Famous Players’ Jim the Penman and the third film—a travelogue about Niagara Falls) [6].

The first full-length 3D film was the 1922 melodrama Power of Love. It was a anaglyfitické design, i.e. film cameras filmed two upright side by side on viewing distance. Amplified first 3D film was an Italian film Nozze Vagabond of 1936. Breakthrough film became the first color film music in the Soviet production of Robinson Crusoe in 1947. Filmed the process called stereo-cinema first and it was not necessary to use anaglyph glasses. The process was developed S. P. Ivanov—Corrugated iron canvas reflecting the different screens on two separate images for the left and right eye. Among the most wonderful scene is reputed scene of wild cats, which is close to the camera. This scene was filmed to 5 days. At that time the scene was one of the most beautiful experiences. In this scene, the audience had the feeling that beast passed directly over their heads and disappeared at the end of the movie theatre [10, 16].

In the 1970s and 1980s to become a 3D movie attraction at amusement parks, but only with the development of technology companies are starting to spin IMAX 3D films, which are popular character—an educational film. Currently, 3D cinema has very little. In Slovak conditions were screened mainly action films, thrillers and comedies, yet there has been no naturalistic film or documentary. Since 1953 (the period from the golden era of 3D movies) was filmed in 31.8.2012 basically just over 200 films (Fig. 1), some of which were subsequently remanufactured to 3D, most prime minister in the world was in 2011, though still within first 5 years with a number of premier also occurs in 1953, which is referred to as the Golden Age.

In recent years this number has grown exponentially and the prognosis is possible to say that a similar nature will take another few years (see Fig. 2).

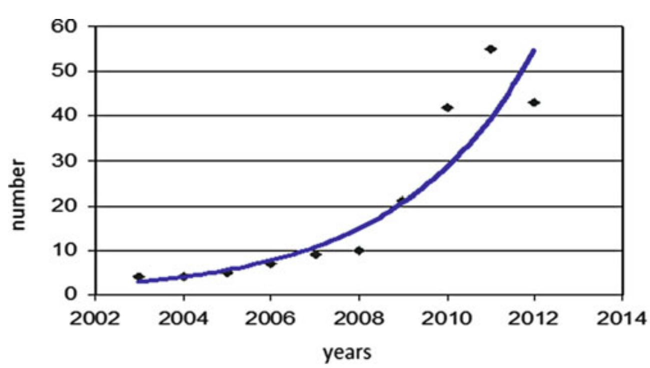

Fig. 1 Overview of the number of 3D movies by year. For the period 1953–2010

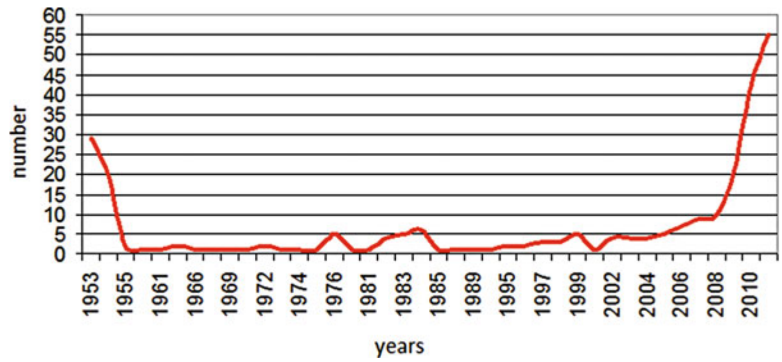

Fig. 2 FORECAST evolution of the number of 3D movies

Methods and Results of Research

To meet the target cells based on two questionnaire surveys. The first research is focused on the teacher factor, which greatly influences and creates conditions to establish a relationship to the natural sciences. In the second research, we focused on the factor student who graduated from activities associated with the projection of natural history film at the cinema.

The first research we have done in the months of January and February 2013 on a sample of 78 primary and secondary schools, of which 42 teachers were teachers at secondary school and 36 teachers taught in secondary schools. Participants were 16 teachers aged under 30 years, 27 teachers aged 30–40 years, 18 teachers aged 40–50 years and 18 of them were over 50 years of age. Most of them, namely the 61 respondents were female and the rest, ie 16 teachers were men. Most of them were one of the objects of approbation biology, and to 41, followed by mathematics (31), chemistry (24), geography (20) and Physics (12). The questionnaire aimed to determine the current status of the use of film teachers in teaching science in primary and secondary schools. He also aimed to identify potential options for further use in film school practice. Examined teacher's attitudes to film the movie in general but also as a means of learning. He was drawn to the manner and method of using film to lesson with regard to the activity and activities of students. The questionnaire was anonymous and had 43 entries. All items were scaled, teachers could choose from five point scale: always—very often—often—sometimes—never. Their task was to identify the range of frequency of occurrence of a phenomenon. For the purposes of Article 7 have been appropriate items:

- Item 10—During a visit with students teaching theatre.
- Item 17—During the teaching of pupils going on documentaries.
- Item 19—Do you practice teaching experience.
- Item 20—When using film or animation observe positive changes in pupils' behaviour.
- Item 30—screening of the movie you are trying to achieve an increase in motivation.
- Item 38—Film used to teach theoretical approach to the facts.
- Item 41—After showing the film, followed by discussion in class.

Items focus on creating a positive attitude towards science through film and theatre environment. Depending on the findings of the items we collected data enrol in pivot tables.

In the first case (Table 1), we investigated whether a teacher who practiced experiential teaching, visiting with students during class cinema . We found that almost half of the teachers (46.93 %), who practiced experiential teaching can be seen in the movie theatre as a sort of an innovative form of teaching that pupils can evoke experience as a result of the changing environment and the quality of picture and sound. In the second case (Table 2), we looked to what extent teacher who goes to the cinema with students during class chooses deliberately documentaries, which are often the bearer of reality and the present reality of the spectator, whose task is the enrichment and new information. In this case, we have come to find that more than half (57.69 %) science teachers tend to choose if it is just a documentary that student may provide new information for the development of knowledge and personality. In the third case (Table 3), we found that a teacher who takes students during class to the cinema, the film follows the students positive changes in behaviour. And at 86.96 %. It follows that the teacher can positively influence student film with all his emotional side, but it should be borne in mind that the teacher must have sufficient advance information about the selected movie to which it can facilitate the use of the methodology movie search for teaching process [10].

Also true that the film screened during class at cinema generally has a high motivational factor (Table 4) and up to 91.30 % cases. Assess the impact of dynamic projections on student motivation has been the subject of research, which took place in 2010 in Israel. The aim of this research was to determine to what extent affects dynamic projection pupil

Table 1 Does the teacher visit cinema, who practice experiencing teaching?

	Item no. 10			
Item no. 19	Never	Rarely	Often	Together
No answer	1	0	0	1
Never	0	1	0	1
Rarely	24	12	0	36
Often	19	7	1	27
Very often	6	2	0	8
Together	50	22	1	73

Table 2 Does a teacher choose documentaries, who goes to the cinema with the pupils?

	Item no. 17			
Item no. 10	Never	Rarely	Often	Together
Never	2	37	11	50
Rarely	0	7	15	22
Often	0	1	0	1
Together	2	45	26	73

Table 3 Prove the positive effects in behaviour For students under the influence watching movies in the cinema?

	Item no. 20					
Item no. 10	No answer	Never	Rarely	Often	Very often	Together
Never	5	4	22	14	5	50
Rarely	2	1	8	6	5	22
Often	0	0	1	0	0	1
Together	7	2	31	20	10	73

Table 4 Does have a screening of the film at the cinema during class incentive importance?

	Item no. 30						
Item no. 10	No answer	Never	Rarely	Often	Very often	Always	Together
Never	1	2	12	23	5	7	50
Rarely	1	1	8	8	2	2	22
Often	0	0	0	0	0	1	1
Together	2	3	20	31	7	10	73

motivation fourth and fifth year of primary school. Teachers develop their students to a Web page on which were located a few minute videos that explain the fun way hundreds of theoretical scientific concepts. Such a projection they were screened at least once a week and always was associated with the current curriculum. After research students showed significant progress in its self-evaluation, changes in attitudes and interests. It is often commented on the importance of science to man and its connectivity to everyday life. The film thus greatly increased their intrinsic motivation and deepened interest in natural phenomena and science itself [16, 17].

That the film can approach the theoretical facts are known, but if this film is screened in cinemas space, the

Table 5 Does have the impact of watching movies in the cinema higher effects of theoretical knowledge

	Item no. 38						
Item no. 10	No answer	Never	Rarely	Often	Very often	Always	Together
Never	0	4	18	18	7	3	50
Rarely	1	4	10	7	0	0	22
Often	0	0	1	0	0	0	1
Together	1	8	29	25	7	3	73

Table 6 Influence watching movies in the cinema, manifests higher interest pupils in discussing?

	Item no. 38						
Item no. 10	No answer	Never	Rarely	Often	Very often	Always	Together
Never	1	2	15	16	4	12	50
Rarely	0	10	1	1	2	8	22
Often	0	0	0	1	0	0	1
Together	1	12	16	18	6	20	73

theoretical knowledge from watching TV only amplify (Table 5), as confirmed by the finding in more than three-quarters (78.26 %). Other findings (Table 6), confirming the fact that the discussion after the film has views influence (86.96 %) to build pupils' attitudes to the subject of the film, resulting in the creation of the relationship towards science.

Discussion

The results of the questionnaire survey aimed at pupils come from the project in the Film Club Charlie centre in Bratislava from 23th October to 19th December 2012 in the school performances. Project was presented under the title "Let's go to the cinema (instead of school)." In designing the project, we tried to link the film showing in the cinema with gain and deepen the life of the animals from the natural history and biology.

For the project we have set out six key objectives:

1. Do with natural history film cinema showcasing how real educational institution.
2. Unconventional way of teaching to motivate students to focus on nature.
3. Using film to familiarize students with new information about the behaviour of animals.
4. Help students to get feedback on the film using the worksheet that you completed immediately after showing the film and subsequent discussion.
5. Clarify student life animals in the wild, but also around us.
6. Develop pupils' environmental awareness.

In the final stage performances, was added to the questionnaire. The research participated 155 students aged 10–15 years

Table 7 Results of the research

Item	Yes	No	Not know	No answer
Biology is one of my three favorite subjects	75 (48.39 %)	53 (34.19 %)	26 (16.77 %)	1 (0.65 %)
Did you like the film Love animals?	129 (83.23 %)	10 (6.45 %)	15 (9.68 %)	1 (0.65 %)
Would you like to see Love animals again?	76(49.03)	41 (26.45 %)	36 (23.23 %)	2 (1.29 %)
Did you enjoy today's program in theater	123 (79.35 %)	11 (7.10 %)	17 (10.97 %)	4 (2.58 %)

in three primary schools in Bratislava, where it was 77 boys and 78 girls. The questionnaire aimed to determine the use of film in teaching biology and detection of feedback students about film and design. Was anonymous and had nine items. Items were closed-response, in which the students could choose from three options: yes, no and I do not know. While the purpose of this article is useful four items (Table 7).

Conclusion

Watching films never replace real reality, but it can be a very close approach. One of the main objectives of scientific and technological progress, just close as possible to reality anytime and anywhere. This way we can comfort such home, school, cinema exploring close but distant world. More and more realistic picture and sound for film viewers create more and more realistic feelings that generate both positive as well as negative feelings. Properly channelled these emotions can help the student see the world, to train him to respect for self and the outside world. It is therefore necessary that the education of pupils with film clearly entering a teacher who largely coordinator is to train the student in a spirit of understanding the world around moral ethical principles.

Acknowledgment The publication of this paper was supported by grant KEGA 119-005TVU-4/2010

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An Approach Based on BPMN to Detail Use Cases

Adriana Herden, Pedro Porfirio M. Farias, and Adriano Bessa Albuquerque

Abstract

Use cases have become an important technique for capturing user requirements in software development projects. Likewise, increasingly adopted the workflow models represent the business processes of a company. Recently the standardization of BPMN v2, by the OMG, has given rise to modeling tools and execution of business processes that enable rapid prototyping applications, integrating the requirements to business. One of the most critical problems of software development is to create solid models and consistent with the company's business. Thus, this work presents an approach for developing systems based on the detailed use cases in BPMN in order to refine and validate executable prototypes produced during the development.

Keywords

Use Case Specification • BPMN • Software Process • Activity Diagram

Introduction

The system specifications are the user's requirements transformed by the activities of requirements engineering. According to [1] the main problems of system development happen in the specification and management of these requirements. It is known that at each stage of requirements engineering documents are produced by techniques such as interviews, scenarios, use cases, diagrams, prototypes, among others. The application of these methods seeks to reduce the distance between the views of users and developers. This is no different in collaborative systems because workflows are modeled along with users, and decisions often depend on the validation of one or more people involved in the business process. According to [2] a business process is a series of activities that are undertaken to achieve a particular business goal.

Currently, there is a tendency to combine the concepts of Business Process Management (BPM) and its collaborative aspects to the software engineering process, from the initial stages of development. It is known that the business processes modeled in Business Process Model and Notation (BPMN) support the execution itself through Business Process Management System (BPMS) tools. Such tools enable a rapid prototyping of the business process models for requirements validation, which are also used to develop systems in the context of agile methods. In addition, BPMN is widely used to represent high-level process responsible for the integration of applications in the context of Service Oriented Architecture (SOA).

The authors [3] describe an integrated framework with the aim of supporting the inclusion of SOA, BPM and EA (Enterprise Architecture) within the RUP (Rational Unified Process). In the authors view the software development process depends of business model more flexible, which requires the use of technologies such as SOA and BPM to support the changes and integrate services. Some suggested activities are: "Identify Business Processes" and "Process Automation Explore" at the phase of Conception. The results showed that these methodologies have additional features,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_69, © Springer International Publishing Switzerland 2015

and if used in an integrated manner to provide real benefits to the development.

This article proposes an alternative path is exploited to describe use cases. Normally use cases are detailed in a textual description and, optionally, complemented with UML activity diagrams. In this proposal the use cases will be detailed using BPMN in the context of an Agile Methodology of software development.

The paper is organized as follows: Section "Business process management" describes business process management; Section "BPMN and UML" comparisons between BPMN and UML; Section "Approach based on BPMN to detail use cases" the proposed approach, Section "Evaluation of the approach" the application of the evaluation approach, finally, Section "Conclusions" describes some conclusions and future work.

Business Process Management

According to [4] BPM is defined as "supporting to business processes using methods, techniques and software to design, enact, control and analyze operational processes involving humans, organizations, applications, documents and other sources of information." BPMS is "a generic software system that is driven by explicit projects of process in order to execute and manage operational business processes."

Business processes represent a vision of ordered work activities, and also the high level of the operation and structure of companies. In this context, the management of these processes emerges as a competitive advantage for organizations that formerly sought only to computerize their tasks without any use of services, or a network of workflows within the company.

According to [4] in the seventies there was a consensus in the development of systems with respect to data modeling, guided by the guidelines of the relational model and entity-relationship model. However, there were different concepts regarding to the form of representation of business processes. To reach a process-centric vision of the company was not always an easy task, because usually companies classify their model of administration by departments.

Therefore, workflow systems that initially emphasized only the office automation with little support process analysis, underwent several changes between the 1970s and 1990s. To increase its applicability they began to support the changes inherent to business processes, and provide a greater flexibility for modeling activities, enact, analysis and management. Besides allowing the participation of stakeholders, such as workers and decision makers in activities involving the change, as well as innovation.

Figure 1 shows the tasks of BPM segundo Van Der Aalst (2013) [17].

Recently, [5] complements his definition saying that "BPM is a discipline that combines the knowledge of information technology, and knowledge of management of science and apply it in business processes." Elsewhere, it still says that BPM can be seen as an extension of systems workflow (Workflow Management).

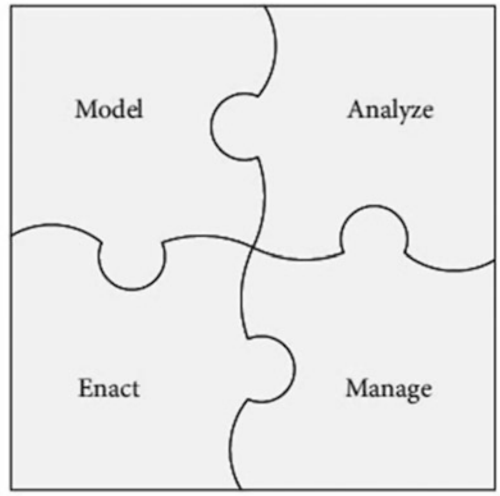


Fig. 1 Activities of BPM, Source: van der Aalst (2013)

Therefore, it is believed that the adoption of BPM provides to decision makers a greater control over the productivity of your employees, because it is possible to measure using the process models that run on a particular project, the time and cost associated with activities performed.

It can be highlighted that the use of BPM is not always adequate [6]. There are cases in which the formalism is not necessary, and there are processes which are unpredictable are so particular that the process modeling becomes unnecessary. Or, in some case, the technical advantages offered by BPMN are not used, for example: the mapping to BPEL (Business Process Execution Language).

BPMN and UML

In order to support the simulation of business processes, initiatives arise aiming to use BPM and BPMS in companies through the process modeling using the notation Business Process Management Notation (BPMN).

According to [7], the main goal of BPMN is to provide a notation that is understandable by all business users, including business analysts that create an initial design of the processes, the technical developers responsible for implementing the technology that supports the execution of such processes, and finally, to the business people who manage and monitor these processes. In addition, one of the advantages of the models represented in BPMN is that these can be executed through the mapping to BPEL.

In [8] the authors prepared a survey of methodologies for service-oriented development, which identified characteristics before comparing existing approaches. Among the methods studied the authors cite BPMN to BPEL as an approach for expressing business processes in an abstract model, which is automatically mapped on a BPEL description language being executed by a process engine. The authors also believe that agile methods like XP (Extreme Programming) are successfully used in SOA projects.

The BPMN v.1 was adopted as a modeling standard by the Object Management Group (OMG) in 2006. This aims to provide a graphical expressivity to facilitate the understanding of its elements, both to the end user, and for the domain experts. Besides offers technical advantages such as, support for service-oriented computing. The current version of BPMN v.2 was adopted by OMG in 2011 [9].

BPMN is based on a review of other notations and methodologies, particularly UML Activity Diagram, UML EDOC Business Processes, IDEF, ebXML BPSS, Decision Activity Flowchart (ADF), RosettaNet, LOVeM and Event Driven Process Chains (EPC) [9].

There are works that study the relationship between the elements of the already established UML and the elements of BPMN notation in the context of systems development. In the article [10] proposes an evaluation of modeling languages for business processes (UML 2 and BPMN), from the perspective of readability of the models created from the perspective of end users. The evaluation showed that these two languages have no significant differences to its users. However, [11] believe that the transformation of BPMN diagrams for UML Activity Diagrams is required for the development of systems, since UML is better supported by modeling tools and are easier to read from end users.

To [12] the extent to which the number of use cases increases it becomes more complex to obtain an overview of the system and understand the dependencies and execution order of use cases. So, these authors suggest an approach for generating models represented in BPMN as a means of restoring overview of the use cases, and sort them according to their pre and post conditions.

To [13], the OMG adopted BPMN instead of UML Activity Diagram as primary standard for modeling business processes. The authors conducted an empirical comparison with both techniques in order to evaluate the applicability of modeling languages. The results indicate that the UML Activity Diagram is at least as useful as BPMN, since there are no significant differences of the effectiveness, efficiency, and user satisfaction.

The author of [14] conducted a study to identify capabilities and restrictions of BPMN and among the several findings the author considers that any modeling language must offer adequate expressiveness of the representation and complexity of graphical elements. The results of his study he observed that most of the BPMN modelers are autodidacts, with little advanced skill in modeling, and receive little training in notation. Furthermore, BPMN provides a limited support for decomposing complex scenarios into smaller models.

In another study [15] assesses the complexity of modeling methods BPMN and UML. He found that BPMN has very high levels of complexity when contrasted with UML. Further suggests that BPMN is significantly more complex than any UML Activity Diagram or a state diagram in terms of objects, properties and relationships, as well as the ratio of these components.

In [16] believes that IT architects are forced to understand the particularities of BPMN to develop IT solutions aligned with the business processes of the company. Then, proposes a Model-Driven Engineering (MDE) that performs the automatic translation of BPMN models to UML Activity models through the transformation language ATL. In the context of this study the transformation facilitated the alignment between processes.

There are similarities between the representations elaborated (UML Activity Diagrams), and BPMN (Business Process Diagram), for example, the behavior of processes. One of the possible reasons for the increased complexity of BPMN referenced in [15, 16] work is the difference in purpose when compared to the UML Activity Diagram. While the purpose of the UML Activity Diagram is to be a model in the analysis phase, in contrast the purpose of BPMN includes the generation of executable code, thus requiring a higher level of detail.

Depending on the purpose of BPMN be wider than the UML Activity Diagram and there is a growing number of tools that support the generation of executable code from its specification, this paper proposes, then, an approach of integration between the use cases and BPMN in the context of agile methodology.

Approach Based on BPMN to Detail Use Cases

In general, the use cases are detailed textual form. This textual detailment can cause several problems. The first is the ambiguity contained in the texts that can generate conflicts between the views of users and developers. Furthermore, the textual representation can be more summarized as much as more verbose. This uncertainty about the granularity of specification is a problem pointed out by software estimation technics, such as Use Case Points (UCP), which is based on use case descriptions to estimate the effort of system development.

An alternative to textual descriptions is the use of a graphical but more formal notation to construct a model to detail the use cases. The BPMN is a good choice to detail the use cases for four reasons:

- The graphical aspect can improves the visibility and easiness of design the models. UML Activity Diagram also has this advantage. It is used to clarify but not avoid the use of textual descriptions.
- The formal aspect permits to reduce the ambiguity of the model
- The formal aspect also standardizes the granularity of the model so it may be more appropriately used to estimate the development effort of software.
- Modern BPMS, that use the version 2 of BPMN, can generate executable code directly from the model.
- BPMN may avoid the use of textual descriptions so can be used to specifying requirements in the context of Agile Methodologies

The proposed approach in which use cases are detailed in BPMN is contained in one of the stages of the process AgilePDD. The AgilePDD is been tested and is a software process that adopts the philosophy of agile methods and uses techniques related to BPM and SOA in their life cycle. Its main phases are: *Scope Definition*, ***System Prototyping***, *Sprint Producing*, *Enactment*, *Monitoring and Optimization*.

The specification of use cases is included in the *Scope Definition* (the first phase of AgilePDD). Next is done the phase of **System Prototyping** that includes as sub phase the ***detailing of the use cases*** using BPMN.

The phase of **System Prototyping** is iterative and incremental. In each cycle aims to present to the user an executable code that allows him to visualize the functioning of the system under development. This phase has, according to Fig. 2, the following activities:

- **Use Case Description using BPMN:** activity that describes, as a BPMN process, the flow of activities that comprise the scenarios of use cases. It is considered that at this stage the developer will uses a BPMS;
- **Data Definition:** activity that identifies the data that will flow in the process. These data may be internal to an activity or may have global scope carrying information from one activity to another;
- **Screens Definition:** activity that defines the screen elements and their layout for each activity of the BPMN diagram that requires user iteration;
- **Prototype Generation:** activity that, using a BPMS, generates a prototype having as input the BPMN diagram, data definitions and screens definitions;
- **Validation with the User:** activity that confirms with the users if the prototype corresponds to their expectations about the system.

The activities of the phase **System Prototyping** are performed by business analysts in close collaboration with users. The *Sprint Producing* phase is also iterative and incremental and is normally used in agile methodologies. In AgilePDD this phase has two activities: "Integration with Existing Data and Processes" and "Automation of Activities". These two activities are more technical and, therefore, must be performed by systems analysts and programmers and, also, must be validated by users. The phases of *Enactment, Monitoring and Optimization* is related with the maintenance and evolution of the system.

Table 1 presents a mapping between elements of the use cases and BPMN.

Each use case is related to a Pool and Lanes inside the Pool can represent each of the actors of the use case. Finally, the scenarios of use cases are mapped to task flow in each BPMN diagram. Alternate flows generally start from a gateway or from error events.

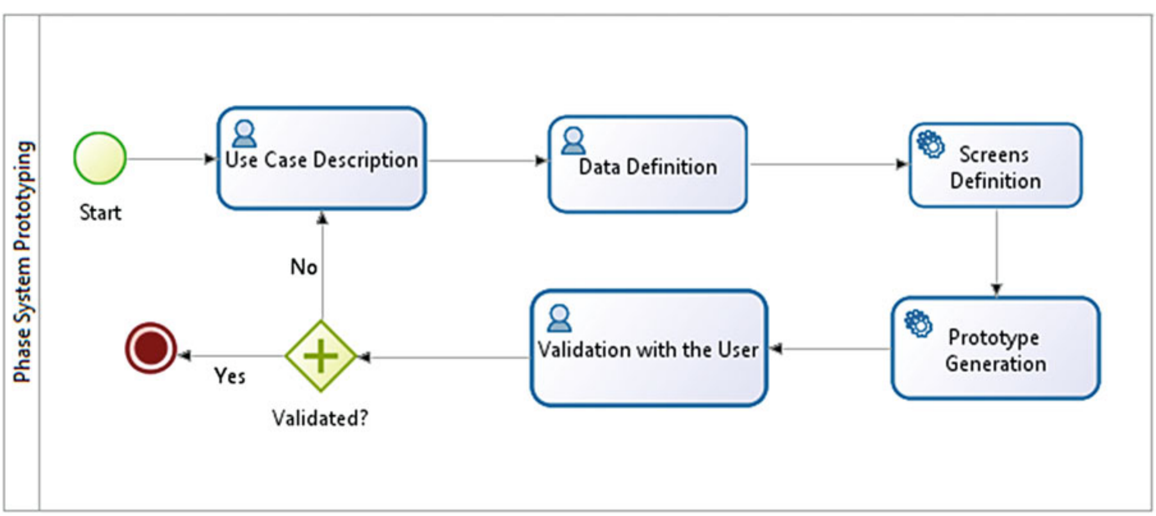
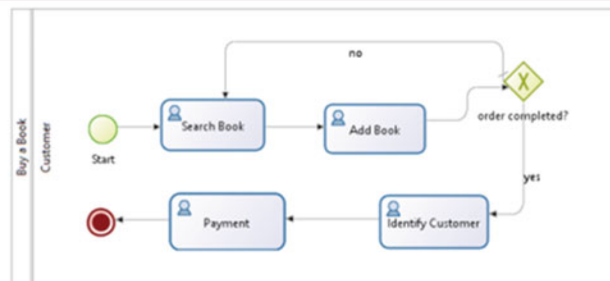


Fig. 2 System prototyping phase (AgilePDD)—Detailing use cases

Table 1 Mapping between BPMN and use case elements

	
	
Main and Alternative Scenarios	

Example

In order to clarify the use of the proposed approach the Fig. 3 presents a sample related to a "Bookstore".

Normally, a use case is described textually informing a sequence of steps that constitute the main scenario and the conditions that make the flow to follow to the alternative scenarios. In Table 2 it is possible to read the description of the use case "Buy a Book".

The graphical representation of BPMN allows improving the visualization of the situation described in the use case. In addition, the diagrams can be used to generate executable prototypes. Figure 4 shows the Business Process Diagram for the use case "Buy a Book".

The diagram of Fig. 4 will be complemented with the specification of global data and local data, as well, for each task, will be incorporated the screens that allow the user interaction. With these specifications can be generated an executable prototype that will be validated by the user.

Evaluation of the Approach

For the evaluation of the approach proposed, a very simple questionnaire was applied to a group of 22 people with similar job profile, working in public and private companies. The participants perform functions such as business analysts, systems analysts and project manager, usually with two or more years of experience. The questionnaire consists of three questions only aimed to identify which of the notations offers better usability to the analysts.

In each question a characteristic of the notation has been addressed (expressiveness, complexity, graphic design), and, for each characteristic, the participants selected which is, in your opinion, the best representation between

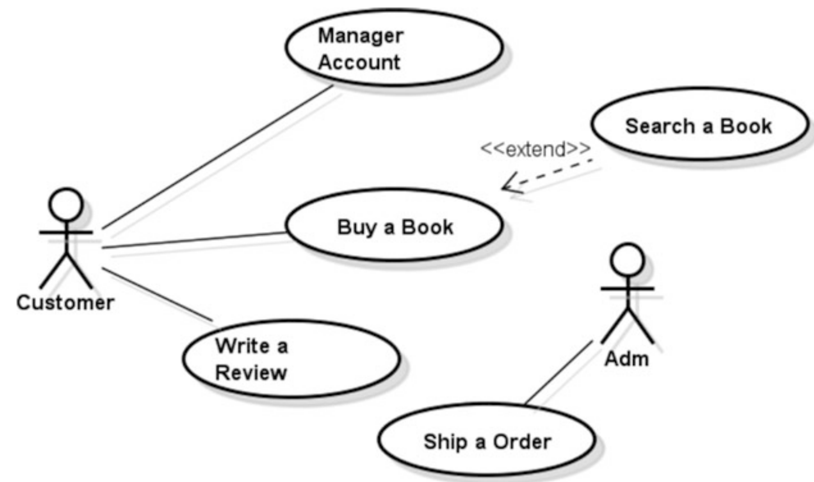


Fig. 3 Use case diagram—Bookstore

Table 2 Description of use case buy a book

Use Case	**Buy a Book**
Actor	Customer
Description	This use case describes the steps taken by you to order a book.

Main Scenery
While the order is not completed 1. Search a Book 2. Add Book 3. Identify Customer 4. Make the Payment

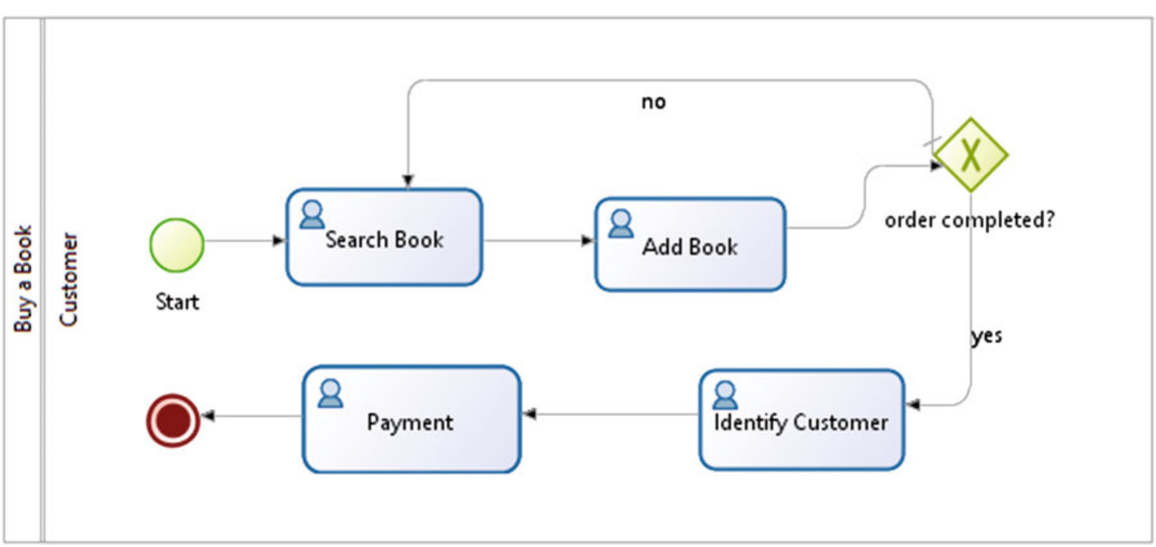


Fig. 4 Business process diagram—buy a book

UML—Activity Diagram versus BPMN—Business Process Diagram.

Analysis and Interpretation of Collected Data

The objective of the evaluation was to identify which of these notations is the best with respect to the perspective of usability in the opinion of professionals with some experience. Tables 3, 4, and 5 respectively show the results of the questionnaire questions.

In the analysis of question 1, Table 3, it is observed that there was no significant difference in the results. Probably one of the causes may have been the habit of using the UML Activity Diagram, and the inceptive use of the BPM by software houses.

In the analysis of Question 2, Table 4, it is observed that 64 % of participants considered that BPMN is simpler compared to UML Activity Diagram despite it is already being used by most professionals for over ten years. This result may be happened because the BPMN was proposed considering the experience of years in the using of UML Activity Diagrams. As the Business Process Diagram—BPMN has more elements than the UML Activity Diagram, can be considered that it is a notation with greater expressive power. This greater expressiveness facilitates the modeling in the situations for which there are appropriate elements in BPMN, such as the inclusive gateway, that are not easily modeled in UML Activity Diagram.

In the analysis of Question 3, Table 5, can be observed that, despite the greater expressiveness of BPMN and the possibility of generating executable prototypes, the result was similar for the two notations. Probably because the wider use of BPMN has yet been to model business processes, while the UML Activity Diagram has, for a long time, been used to model systems.

Table 3 Results of question 1

The detailment of the use cases is easier for which notation?	
UML—Activity diagram	50 %
BPMN—Business process diagram	50 %

Table 4 Results of question 2

What notation is simpler for the user? (Complexity)	
UML—Activity diagram	36 %
BPMN—Business process diagram	64 %

Table 5 Results of question 3

Notation which has a set of more appropriate graphics to represent the system? (Graphic design)	
UML—Activity Diagram	50 %
BPMN—Business Process Diagram	50 %

Conclusions

Agile methods prioritize to produce executable artifacts and avoid producing textual descriptions that increase the time required for system development. Following this philosophy, was proposed an approach of detailing the use cases using BPMN that is inserted into the phase of **System Prototyping** in the context of an agile development methodology.

The graphical appearance of BPMN allows its elaboration by business analysts as well as facilitates the visualization of the flow of activities and participation of users in specifying processes.

Using BPMS, information can be added to the BPMN processes (the specification of the data manipulated and the screens definitions) enabling the generation of executable prototypes of the system.

As future work, will be realized more experiences using the proposed approach to evaluate all the phases of AgillePDD. Furthermore, it is intended that the processes specified under this approach could serves as input to estimate the effort in developing systems.

Acknowledgment This work has the support of the University of Fortaleza (UNIFOR) and the Federal Technological University of Paraná (UTFPR).

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Design of a SCADA System to Prevent Freezing in Oil Pipes

M. Ghassoul, S.R.A. AlMishal, B.A. Ali, and A.A.N. Thabit

Abstract

This paper presents a SCADA system based on Siemens WINCC implemented on a Web server to drive an Allen Bradley PLC to control temperature in two tanks, the first tank simulates a tank for collecting oil from several locations with different temperatures, and the second tank simulates an oil tank which is heated and used to defreeze the pipeline during winter. A tuned cascade PID is used to control the process. This has two advantages; firstly, this controller saves on fuel consumption and secondly, it eliminates the effect of feed tank temperature. The system could be controlled locally or remotely through the web server. The system could also send text messages using GSM to an IPHONE.

Keywords

Freezing • Oil • Pipes • Temperature • Tank • Web server • GSM • IPHONE

Introduction

One of the major problem facing companies exploiting oil in the middle east region is pipe freezing over the winter where temperature plunges to below freezing. This phenomenon freezes pipe carrying oil. The oil, in turn, freezes. This renders its transportation very difficult. What the companies generally do is to place several oil heaters in different locations to heat up oil which circulates around the pipes to prevent them from freezing. Unfortunately those heaters are still controlled manually, based on relays driving pneumatic equipment. With their aging, they are becoming more and more expensive to maintain, inefficient on fuel consumption, difficult to control, and most of them are obsolete. Another problem is in collecting oil from several different wells, located at different locations. When oil is collected in a single tank, each well contributes with an amount of oil different in quantity and temperature, depending on the distance and the under soil temperature. This contributes to temperature disturbance. If a classical PID alone is used, a big amount of heating oil will be wasted due to the temperature disturbance inside the tank as a result of the temperature disturbance contributed by different distant wells. A better way of controlling the system efficiently is the use of PID cascade controller system, based a modern digital control system. This paper presents an up to date design of an efficient automatic cascade PID control system, using a Supervisory Control And Data Acquisition (SCADA) based on a programmable logic controller(PLC). The PLC used is the RSLogix5000 interfaced to a Siemens WINCC Human Machine Interface (HMI), through RSVIEW OLE for Process Control (OPC) web server. The system could be supervised either locally or remotely using an internet router or radio frequency.

System Design

As seen in Fig. 1, the system consists of two tanks, tank1 (TK1) represents the feed tank which feeds the heater with the returned water from tank2 through pump (P1). The water flow is controlled by the solenoid valve (SOV1). The temperature of the liquid in the tank is picked up by a resistor dependent detector (RTD) TT1. And the level is measured using a differential pressure transmitter (DPT). All the actuators and transmitters are connected to the PLC.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_70, © Springer International Publishing Switzerland 2015

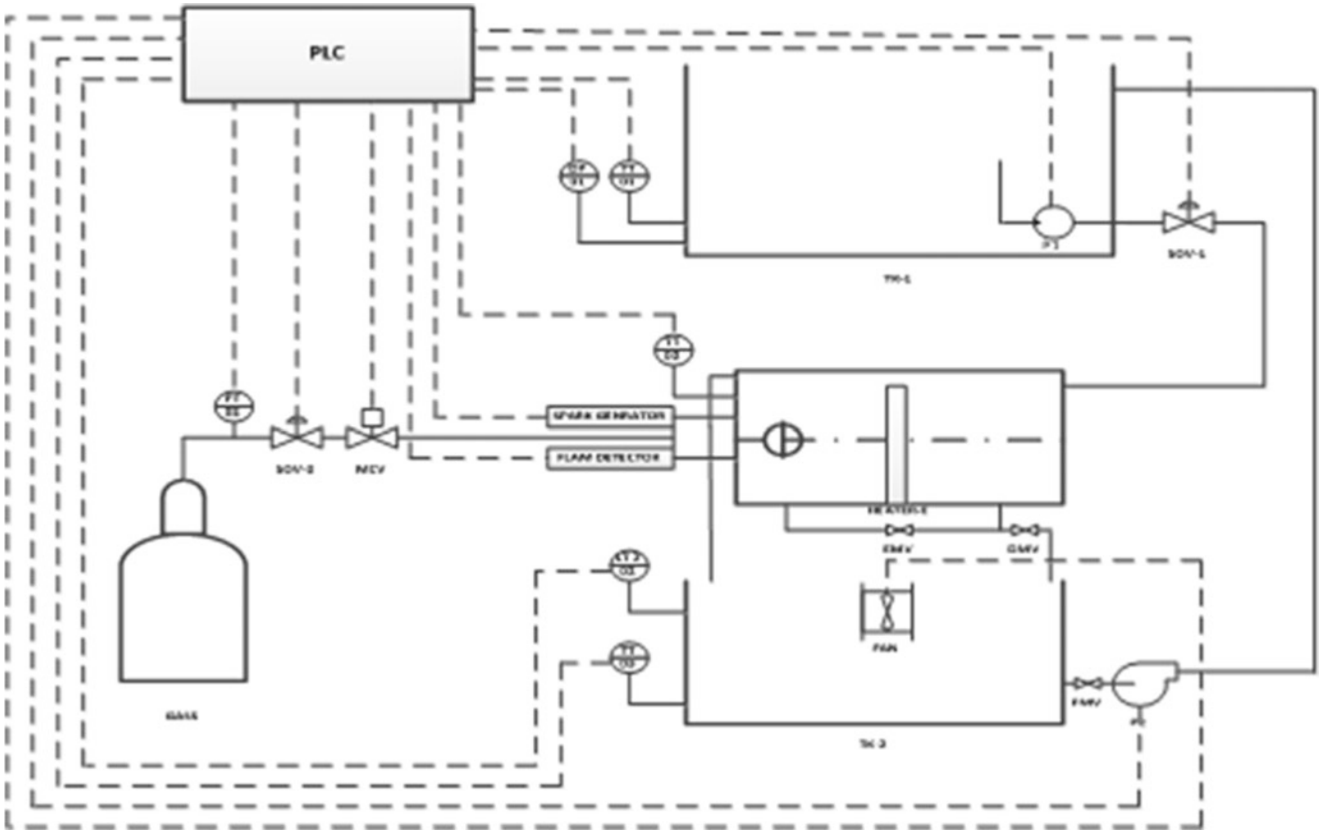


Fig. 1 PI&D of the control system

The second part of the project is the heater (E). It is the heart of the process. It is divided into two sections. A hollow metallic core crosses the two sections, as a heating chamber, which plays the role of the heat exchanger. A gas burner goes underneath the chambers. The gas feed to the burner is controlled by a proportional motorized control valve (MCV) to heat up the water to the desire set point. The water temperature is measured using a thermocouple sensor connected to the temperature transmitter (TT2). The flame inside the heater core will be automatically started by a spark generator (SG), and for safety reason, a flame detector (FD) is used so that in the absence of the flame, the gas supply will be cut off completely and instantaneously by gas solenoid valve (SOV2). The fuel gas is provided by a gas cylinder (GASS). The gas cylinder pressure is kept checked by a pressure transmitter (PT02).If the gas pressure goes below a certain level, the solenoid valve (SV02) automatically shuts off.

At the bottom of the heater there are two manual valves one used as equalizer (EMV) for both sections. if the valve is closed, the water builds up until it overflows into the adjacent section; and if opened, both sections will fill in simultaneously. The second valve is the drain manual valve (DMV) used to drain the water from the heater to the second tank. Tank2 (TK2) is used as a disturbance tank, where it simulates the collection of crude oil from different wells, with different temperature, depending on the location, and geography. The top of the tank is fitted with a fan (FAN), which plays the role of heat exchanger, where the output of tank1 cools down after releasing its heat to the crude oil. The temperature is read to the PLC through the temperature transmitter (TT3), which in turn, uses a resistor temperature dependent (RTD) to pick up the temperature. A manual valve (DMV) is mounted just before the water pump (P2). It could be closed during maintenance routine. A level transmitter (LT2)is used to measure the level, in order to prevent any

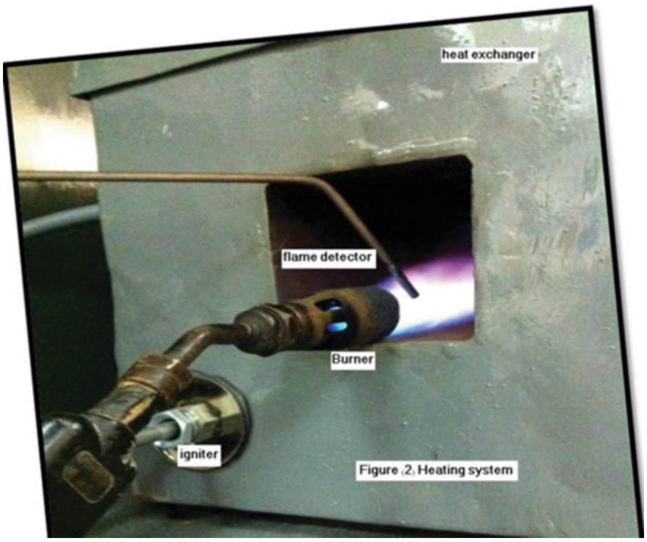


Fig. 2 Heating system

overflow. The system was modeled and gave the following transfer function (Fig. 2).

$$\frac{T'_{out}(s)}{Q'_{comb}(s)} = \frac{3.06}{360\ s + 1}$$

Where $T'_{out}(s)$ is the output temperature and $Q'_{comb}(s)$ is the combustion heat rate.

Interfacing the Process to the PLC

The PLC used is Allen Bradley Logix 5561. Beside the power supply and Control Logix processor, five interfacing modules are used, two as digital input and output, one as 4–20 mA input module, one as 4–20 mA output module and an Ethernet/IP communication module. This module is connected to a workstation to visualize the process and to a remote terminal unit(RTU) to send SMS messages to an IPHONE. Figure 3 shows the hardware, including wireless communication.

Software

The SCADA control system is built around Siemens Windows Control Center (WINCC). It enables the visualization of production flows, operating process, machines and systems in all areas. WINCC is interfaced to AB LOGIX PLC, through an OPC server, so it could access directly the internal and external tags and message configuration and can also use the communication parameters reducing development and time consuming multiple entries of data base configuration. This then enables WINCC to be integrated more closely with the AB controller, reducing engineering and lifecycle cost and enabling the integration of diagnostics functionality that can support maintenance operations. Figure 4 shows the HMI to the process. Figure 5 shows one of the faceplates, which displays the process value, setting up the range, setting up the alarms limit and enabling the alarms, maintenance override.

Results and Discussion

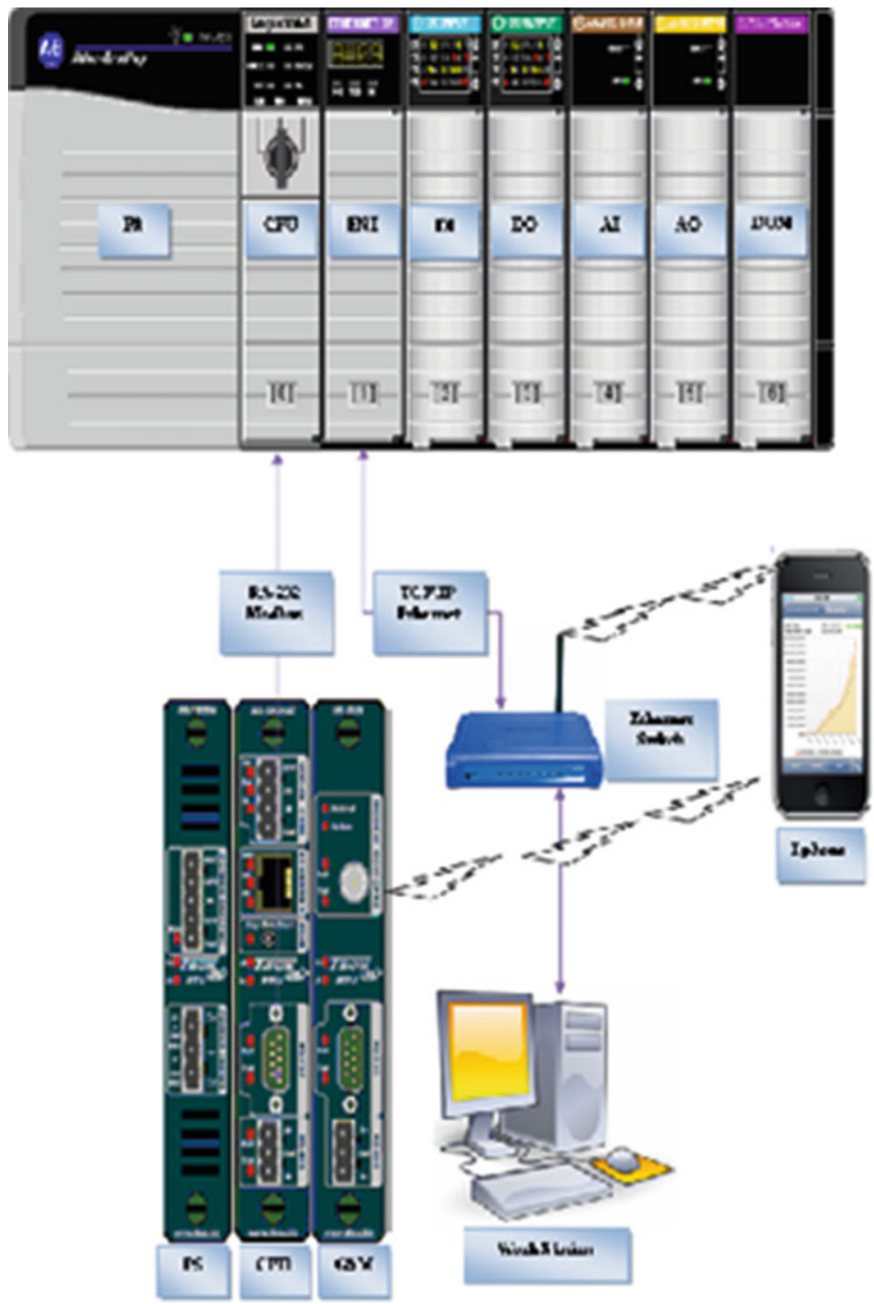
The system was first tested using a tuned PID and gave the response shown in Fig. 6. It is clear that the PID action does not take into account the main disturbance which is the feed temperature. That explains when excited with a step response and with the increase in feed temperature, the heater outlet temperature (H_VC) increases even after reaching the set point. To improve the system performance, a tuned cascade controller was implemented. This has two advantages. Firstly, it saves on fuel consumption and secondly, it eliminates the effect of the feed tank temperature.

The response of the tuned cascade PID controller is shown in Fig. 7. As the feed temperature (T1-TEMP) increases, the controller takes action to adjust the control variable, so that it starts to decrease to maintain the outlet temperature constant.

On top the benefits discussed above, this SCADA system has added the following advantages.

1. Easy to troubleshoot any problem through the provided diagnostic function
2. Historical data of the system could be archived by the SCADA system
3. The possibility of connecting the system to a third party control system with relative ease such as a DCS for instance
4. The process could be visualized remotely through a web server or SMART phone such as an IPHONE
5. All the hardwired annunciators, meters buttons could be replaced by soft ones.

Fig. 3 PLC hardware system



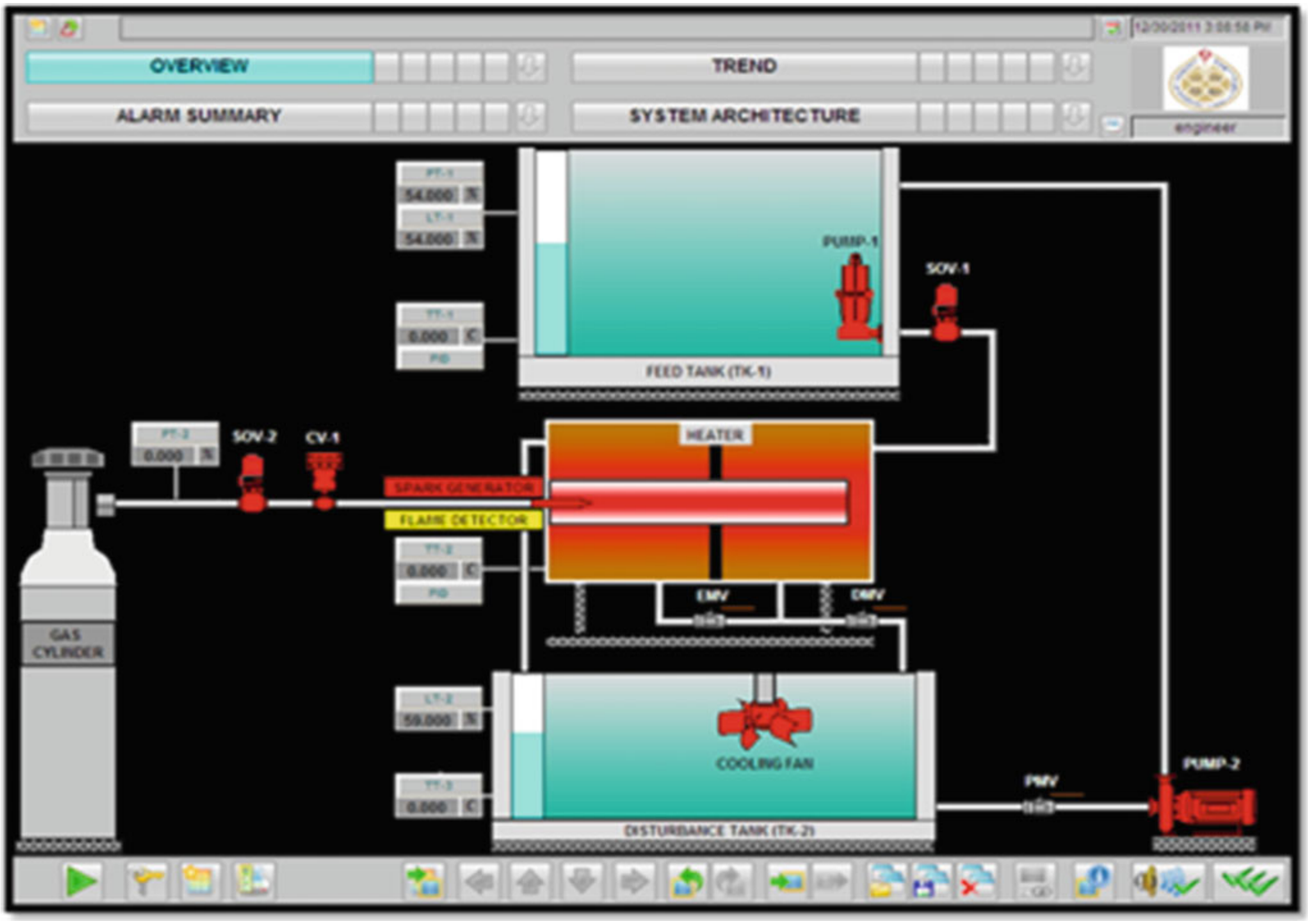


Fig. 4 Shows the system visualization

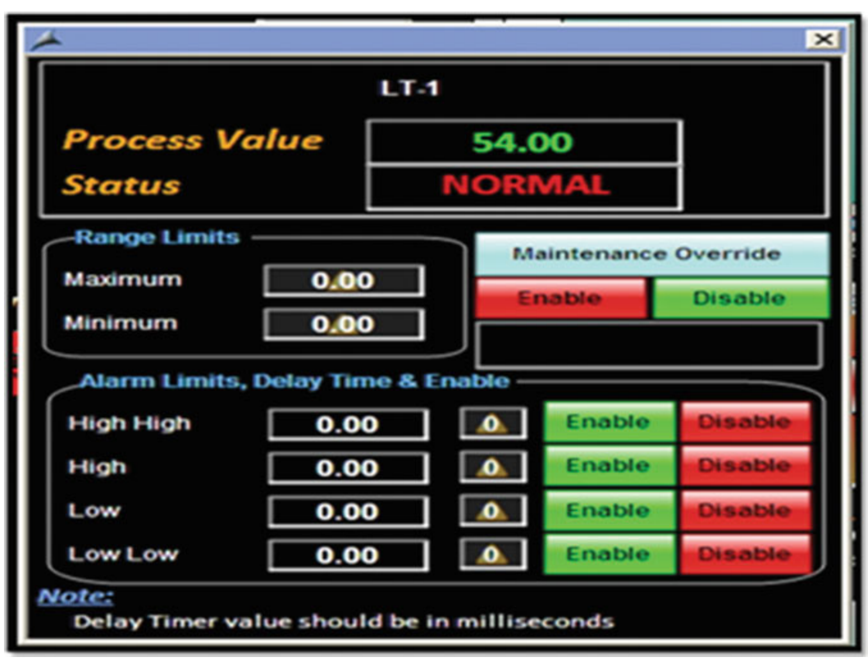


Fig. 5 Shows one of the faceplates used in controlling the system

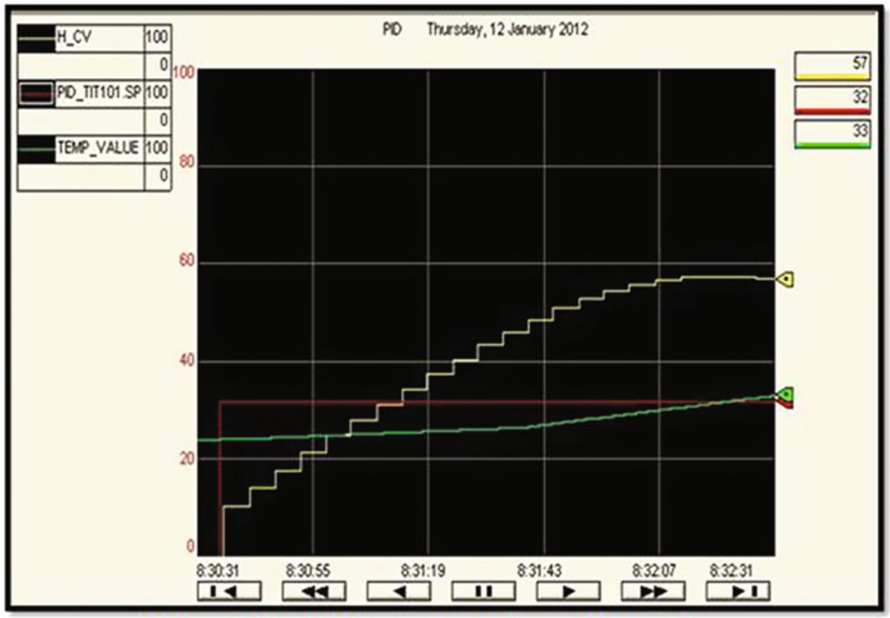


Fig. 6 Shows the output temperature without cascade

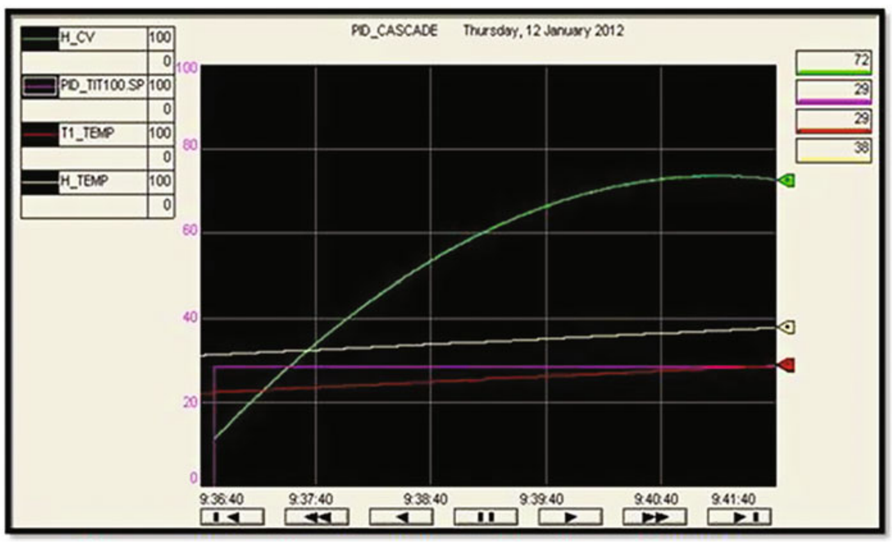


Fig. 7 Shows the temperature output when a tuned cascade PID is used

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Optimizing Next-Generation Mobile Networks Using Frequent Sequential Pattern Mining

Zachary W. Lamb and Sherif S. Rashad

Abstract

With the increasing number of mobile users and the high volume of traffic that exists in modern mobile networks, it is evident that new methods for uncovering the patterns that exist within the usage data need to be developed. We present an algorithm called Prefix-based Mobility Mining (PRIMO) that combines frequent sequential pattern mining and association rules to build a system that can be used to improve traffic handling and predict the future actions of users in a network. We also introduce an application called AndroidMiner that explores the applicability of pattern mining to the mobile computing environment. When a mobile user interacts with a network they are often in contact with multiple cell towers. Employing the sequential nature with which the identifiers of these towers can be collected, we can effectively generate frequent sequences that represent the towers a user or group of users are most frequently interacting with. By representing the interactions of mobile users with networks as frequent sequences, we can effectively develop an understanding of traffic patterns that exist within a mobile network. An understanding of the trends of mobile networks can easily be applied to load balancing or improving location-based services.

Keywords

Data Mining • Frequent Sequential Pattern Mining • Mobile Communications • Multithreading

Introduction

The increasing capabilities of mobile devices along with their widespread use has led to the generation of vast amounts of mobility data that can be used to accomplish several tasks. One such task would be to optimize the resources of an existing network by discovering trends from the usage data of numerous users. Discovering the frequent trends of users would enable the development of new techniques for traffic handling and load balancing. Such techniques could exploit the predictability of users by generating rules associated with their behavior.

Typical interactions with mobile networks occur between the users, who could be the network via a smartphone or laptop, and the network access points, which are often cell towers. By collecting the data that are generated from these interactions we can easily obtain information about the time and duration of the connection, as well as the location of the user. Using this information, which is easily collected and stored in a sequential manner, we can discover the frequent usage patterns of individual users or clusters of similar users. Not only could the discovery of frequent patterns of users be used to assist in optimizing a network, it could be used to improve location-based services (LBS's) by aiding in the understanding of how users interact with the network and move between different networks.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_71, © Springer International Publishing Switzerland 2015

This paper focuses on current frequent sequential pattern mining techniques and applies various methods to optimize their efficiency and accuracy for mining the data of users and their interactions with networks. We first chose a current algorithm to work with and evaluate its performance and applicability to mining sequential networks usage data. We then propose a new version of the algorithm that is more efficient than previous implementations. In addition to creating an optimized sequential pattern-mining algorithm, this paper presents a method for quickly generating association rules from the results of the pattern-mining algorithm. Our technique takes advantage of the information and sequences generated by the frequent sequential pattern-mining algorithm to construct association rules. This sharing of information eliminates the need to perform redundant calculations, such as collecting the supports for each element, and ultimately leads to a reduction in the total amount of time required to generate both frequent sequences and association rules.

Related Work

Sequential pattern mining can be broken into two categories based on the techniques used by algorithms: candidate generation and testing and pattern growth. Mining mobile user behavior cannot be as easily split into well-defined categories. Various approaches have been explored in attempts to effectively understand the trends of mobile users. These approaches range from cluster-based temporal patterns as presented in [1], to mining trajectory patterns as introduced in [2].

Pei et al. introduced the PrefixSpan algorithm [3], which is an improvement upon past pattern mining techniques that utilizes pattern-growth to efficiently mine frequent sequential patterns. This approach partitions the database into smaller projected databases and grows patterns by discovering locally frequent items in each projected database. Thus, pattern-growth eliminates the computationally intensive task of candidate generation and testing since every permutation of items need not be examined. The algorithm attempts to improve upon past pattern growth techniques by introducing ordered growth and reducing projected databases. Later in this paper we introduce our optimized version of this algorithm and present our comparisons of performance between the two implementations.

Cao, Cong, and Jensen [4] define a users trajectory as a sequence of GPS records that are ordered by time. They use the sequential format of the GPS records to discover significant locations and any correlations that exist between the locations. By using techniques similar to those proposed in [4] and the sequential nature of network transactions we can discover comparable information from users interactions with networks and treat the users movements between cells as trajectories.

Giannotti et al. in [2] introduce trajectory patterns that represent the frequent behaviors of moving objects. These patterns were discovered through patterns mining techniques that rely on spatio-temporal information in the data. Our technique, which uncovers the frequent behaviors of users in a network, does not require any spatio-temporal information. Instead we can rely on the sequential manner with which the data is stored to develop an understanding of how the users interact with and move throughout the network.

Wang and Han in [5] present an efficient algorithm called BIDE to mine frequent closed sequences. They define a closed sequence as a sequence that has no supersequence with the same support. This idea is applied to our algorithm by eliminating redundant patterns. Eliminating these patterns from our results will provide accurate and more compact results and reduce the computational overhead of discovering redundant information.

Our method, which uses a multithreaded structure, allows for multiple threads to be executed simultaneously much like a parallel algorithm. Many algorithms have been introduced to exploit the benefits of parallel programming. One algorithm that demonstrates the effectiveness of parallel implementations of pattern mining algorithms is the Par-CSP, which was introduced by Cong et al. in [8]. The Par-CSP algorithm which is an extension of the BIDE algorithm. Par-CSP decomposes the BIDE algorithm into separate tasks and executes the tasks in parallel. We propose a technique to decompose the tasks into separate threads that can then be executed concurrently.

Background

Problem Definition

Let S be a sequential database, the task of frequent pattern mining can be defined as the discovery of all *subsequences* α such that the support for sequence α is greater than the *minimum support threshold* supplied by the user. We extend the definition of a sequence so that every sequence is also a *closed-sequence*. A *closed-sequence* is a sequence α for which there exists no *superseqence* with equal or greater support.

Definition 1. ***Subsequences and Super sequences:*** *Given the two sequences* $\alpha = <a_1 a_2 \ldots a_n>$ *and* $\beta = <b_1 b_2 \ldots b_m>$*. α can be defined as a subsequence β, which is denoted as* $\alpha \subseteq \beta$*, if there exist integers* $1 \leq j_1 < j_2 < \ldots < j_n \leq m$ *such that* $a_1 \subseteq b_{j1}$, $a_2 \subseteq b_{j2}, \ldots, a_n$

$\subseteq b_{jn}$. *If true,* β *called a super sequence of* α*, denoted as* $\beta \supseteq \alpha$.

Definition 2. ***Prefix:*** *Given the two sequences* $\alpha = <a_1a_2\ldots a_n>$ *and* $\beta = <b_1b_2\ldots b_m>$*, such that* $m \leq n$*. Sequence* β *can be called a* *prefix* *of* α *if and only if:* $b_i = a_i$ *for* $i \leq m\text{-}1$ *and* $b_m \subseteq a_m$.

Definition 3. ***Suffix:*** *Let* $\alpha' = <a_1,a_2\ldots a_n>$ *be the projection of* α *w.r.t. prefix* $\beta = <a_1a_2\ldots a_{m-1}a'_m>$ $(m \leq n)$*. Then sequence* $\gamma = <a''_m a_{m+1}\ldots a_n>$ *is called the postfix of* α *w.r.t. prefix* β*, denoted as* $\gamma = \alpha/\beta$*, where* $a''_m = (a_m - a'_m)$.

Definition 4. ***Projection:*** *Given sequences* α *and* β*, such that* $\beta \subseteq \alpha$*. A subsequence* α' *of sequence* α *is called a projection of* α *w.r.t.* β *if and only if:* α' *has prefix* β *and there exist no super-sequence* α'' *of* α' *such that:* α'' *is a subsequence of* α *and also has prefix* β.

Definition 5. *The* ***Minimum Support threshold*** *is a user-defined value that represents the frequency with which a sequence appears in the sequence database*.

Definition 6. *A* ***Closed Sequential Pattern*** *is a frequent pattern* α*, such that minsup of* $\alpha \geq$ *the user-defined minsup and there exists no supersequence of* α *that has minsup* $\geq$ *the minsup of* α.

Sequential Algorithm: PrefixSpan

In developing our own optimized algorithm we started with the PrefixSpan algorithm [3]. We chose PrefixSpan for its efficient memory usage and lack of candidate testing and generation. PrefixSpan can be broken into the following steps:

1. Find all length-1 frequent sequential patterns. The algorithm scans the database once and calculates the support associated with each item.
2. Partition the search space. The complete set of sequential patterns is then partitioned based on the prefixes discovered in step one.
3. Find all subsets of sequential patterns. The algorithm constructs the projected databases and recursively mines each of them to discover the full set of frequent patterns by combining the prefixes with the projected suffixes.

Since each of the prefixes will have its own projected database and will be mined separately from the other prefixes, we determine that the algorithm could easily be broken into multiple threads. Splitting the computations for each prefix into multiple threads would allow the threads to be executed simultaneously. Thus creating an algorithm that performs similarly to algorithms that have been implemented in parallel. One reason for using threads instead of parallel programming can be found in the complexity of a multithreaded program. Multithreading creates separate tasks that require less synchronization by the programmer than parallel programming. The threads are created and started when the programmer specifies, but the compiler has more control over whether multiple threads will be executed concurrently. An additional reason that led us to chose multithreading over parallel programming is due to the varying hardware of different machines. Parallel programs are intended for machines with multi-core processors or even multiple processors. While also seeking to exploit the multi-core architecture of modern computing platforms, multithreading can still efficiently operate on machines with a small number of cores.

Proposed Method

Approach

Our method seeks to discover the frequent patterns of users interactions with networks and to generate rules that could be used to make predictions about future traffic within the network. In order to generate both the frequent patterns and rules associated with them we implement two algorithms. The two algorithms are incorporated into one algorithm, which we call PRIMO. Instead of implementing the algorithms to work independently, PRIMO uses a method that allows the two algorithms to share information and work together to eliminate the need for redundant calculations. For example, each algorithm will need to calculate the support for each item in the database. Instead of performing the calculation multiple times the results of the first algorithm to perform the calculations can share its results with the second. By generating both frequent patterns and association rules we are left with a more comprehensive understanding of the hidden trends of the network than would simply mining only the sequential rules form the database. One example of how we can benefit from keeping the sequential patterns instead of only generating sequential rules can be found in what the patterns represent. The frequent patterns will essentially show a cluster of cell towers that the users are most frequently in contact with. Knowledge of what towers the users are most frequently interacting with can prove very useful when a network needs to make decisions about what towers should be providing resources to a user.

PRIMO

In optimizing the PrefixSpan algorithm for mobility data as well as deployment into the mobile environment, there were two obvious areas that could use improvement. The first area we address deals with the accuracy and compactness of the results and the second to reduce the run time needed to discover all frequent patterns from a dataset.

The steps of PRIMO are similar to those of the PrefixSpan algorithm that was mentioned earlier in this paper with a few exceptions. The first of which deals with the prefixes and their projected databases. When the algorithm discovers a prefix and begins to mine the locally frequent sequences from the database PRIMO separates these tasks into threads, so that multiple prefixes can be mined concurrently. The next difference lies in the pruning technique we use to remove any redundant sequences. Once a new sequence is added to the database of frequent sequences the algorithm checks to see if it is a closed-sequence. To do this, the algorithm checks if there exists a supersequence of the new sequence that has equal or higher support. A third difference is that the algorithm also generates association rules based on the frequent sequences. The generation of these rules is done with our implementation of the RuleGen algorithm [10].

To eliminate redundant patterns and rules from our results we apply the concept of closed-sequences. Reference [5] defines closed sequences as sequences for which there exists no supersequence with equal or greater support. By removing these redundant patterns form our results we are left with a more compact and accurate set of frequent sequences and association rules. To remove any redundant patterns from our results we use a simple pruning technique that checks for any non-closed sequence and removes it from the current set of frequent sequences.

The next area our implementation seeks to improve upon is the runtime of the algorithm. To do this we take advantage of the capabilities of modern compilers by explicitly creating separate threads in our implementation. Threads are processes that can execute independently of one another and even run simultaneously. Today's modern compilers can discover these separate threads and execute them concurrently, just as an algorithm written in parallel would simultaneously execute multiple tasks on separate processor cores. To break the algorithm into separately executable tasks we use a technique similar to that mentioned in [8] for task decomposition. The benefit of using threads over parallel programming is the lack of modification to the code and small amount of synchronization the implementer needs to be concerned with.

To decompose the algorithm into separately executable tasks we first identify any tasks that are repeated or that have little dependence on other processes. The first set of tasks we identified for decomposition exist when the algorithm is first started. To begin execution the algorithm needs to setup databases, initialize variables, and read the dataset file. Reading the dataset into main memory will be the biggest task of those mentioned above, so this task is split from the others and executed in its own thread. The remaining initialization tasks are put into their own thread and the two can execute simultaneously.

Another, more significant, set of tasks we targeted for multithreading was the mining of prefixes. Each discovered length-1 frequent sequence will have its own projected database that is mined to discover the full set of locally frequent patterns. Each of these tasks are separate from one another, so we split them into separate threads. While it would seem that creating a thread for every prefix would give the best results, that's not always the case. When using multiple threads we need to keep track of which threads have been executed and remove that task from the list of remaining tasks. A higher number of threads require a larger amount of synchronization that needs to be maintained in order to reduce the possibility of performing the same task more than once.

An additional feature of our algorithm is its ability to produce association rules along with the frequent sequential patterns. To do this we use our implementation of the RuleGen algorithm that was introduced in [10]. Once the frequent patterns have been discovered the process of generating rules from them is fairly simple. The rules are generated based on the occurrence of items together and the minimum support that is supplied by the user. For example, if the sequence (A) occurs if in 2 sequences, while (AB) occurs in 2 sequences we can say that if B occurs there is a 50 % chance that A will also occur. By using this relationship we are left with the rule B $\Rightarrow$ AB, which has a support of 50 %.

Using the sequential rules derived from the frequent patterns we were able to construct a simple classifier that could make predictions about any new input sequences. This classifier was used to test the accuracy of our rules by providing it with new input sequences and observing if the new sequence could be correctly classified as an existing user. The ability to determine if usage patterns are those of an existing user could easily be applied to improving the resource allocation of a network. For example, an established and valid user could be given priority over an unidentified user that is requesting resources from the network.

Mobile Platform

In addition to developing the PRIMO algorithm we also propose a smartphone based data collection and pattern mining platform called AndroidMiner. AndroidMiner combines the data collection and mining phases into one platform. An implementation of a data collection application, which we called Collector, was incorporated into an application that runs the PRIMO algorithm to create the AndroidMiner application. Combining the data collection and mining process into one platform reduces the need for specialized hardware. The smartphone phone can effectively collect the data, format the data, and has the abilities to mine moderately sized datasets. Some limitations are present in the mobile environment, but new techniques can be developed to increase the efficiency of smartphone based data mining.

To collect the data of users we simply use the built-in capabilities of modern smartphones. The data collection portion of the application can collect various types of user-associated data such as network information, location, speed, accelerometer readings, time and date, and many other parameters. Collecting the data requires very little resources and can easily be done with the smartphone. One potential problem that may be encountered when collecting data on a smartphone could arise from the amount of storage the device has. If the platform is used to collect data for a very long period, this may impact the users experience or the device may simply run out of storage space. To prevent this from happening, the application checks if there is sufficient internal storage before beginning the collection process. If there is not enough internal storage the application can check for external storage, such as an SD card, and setup it's database in this external directory. While possible in extreme cases, exhausting the smartphones storage is unlikely as the data stored by the application is compact and requires very little space for thousands of records.

The second half of the AnrdoidMiner application is an implementation of the PRIMO algorithm. The Android operating system, which is written in the Java programming language, allows applications to take full advantage of the capabilities of Java. Thus very few modifications had to be made the algorithm to adapt it to the mobile platform. Limitations and restrictions were needed to keep the algorithm from exceeding the smartphones resources. One such restriction was put on the amount of memory or RAM that was used by the algorithm. The Android operating system limits the amount of heap space that can be used by an application. This limit is one factor that weighs against the AndroidMiner application. The limited amount of memory reduces the size of the datasets that can be mined and the level of minimum supports that can be used. This leads to one of the shortcomings of the AndroidMiner application, which is that the application can only work with reasonably sized datasets. Another potential problem with the application is that it is very difficult to determine how much memory a run of the algorithm will need for a particular dataset. The amount of memory used depends not only on the size of the dataset but also it's complexity. For example, if a dataset has numerous length-1 frequent sequences, the algorithm will need to project a database for each of the sequences. This projection can use a large amount of memory even if the database is not large.

Experimentation and Results

Performance Evaluation of the PRIMO Algorithm

In order to evaluate the efficiency of the PRIMO algorithm extensive performance testing was done to compare its performance to the standard implementation of the Prefixspan algorithm. The PrefixSpan algorithm was implemented according to the description given in [3].

All personal computer based testing was done on a 2.2GHz Intel i7 Mac with 8GB of main memory. All implementations were written using the Java programming language in the NetBeans IDE. Testing and profiling was also done using the NetBeans IDE. The Java Virtual Machine for each run was allocated a maximum heap space of 4GB.

For the data used in our performance evaluation, we use a real world dataset that was collected by Rahmati and Zhong in [6] and that was downloaded from [9]. The dataset represents mobile network data collected by participants on the Rice University campus in Houston, Texas. The dataset represents the traces of ten individual users. The participant identifiers are randomized and the WIFI and Cell Tower identifiers have been anonymized. To reduce the number of features of the dataset and to work with the information most relevant to our work, we use a simple feature reduction technique to remove some of the less meaningful features. The features we are most interested in are the identifiers of the WIFI access points and Cell Towers. Each of these items had other information, such as security properties, which we removed for this particular study.

Figure 1 shows a performance comparison of the PRIMO and PrefixSpan algorithms. The run times for the corresponding minimum support thresholds were determined by taking calculating mean of multiple runs of the algorithms at each support level to ensure the results were accurate. The dataset for this test was for a specific user from the eleven users in the dataset from [6], which we identify as USER1. The USER1 dataset consisted of approximately 225,000 records, each of which contained sequences that ranged from 1 to 23 items in length.

The results in Fig. 1 show that for each minimum support threshold used in testing with the USER1 dataset, the PRIMO algorithm outperforms the standard implementation of the PrefixSpan algorithm. Comparisons of the results generated by each algorithm, verifies that PRIMO maintains the same level of accuracy as PrefixSpan and has fewer redundant patterns.

Further evaluation of the performance of PRIMO was done with a dataset of smaller size. The dataset used for the second round of testing were the records from another

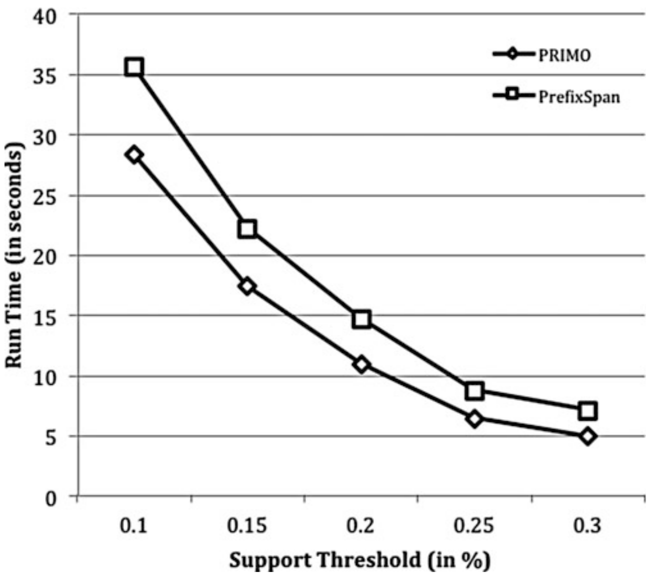

Fig. 1 Performance of the PrefixSpan and PRIMO for USER1 dataset

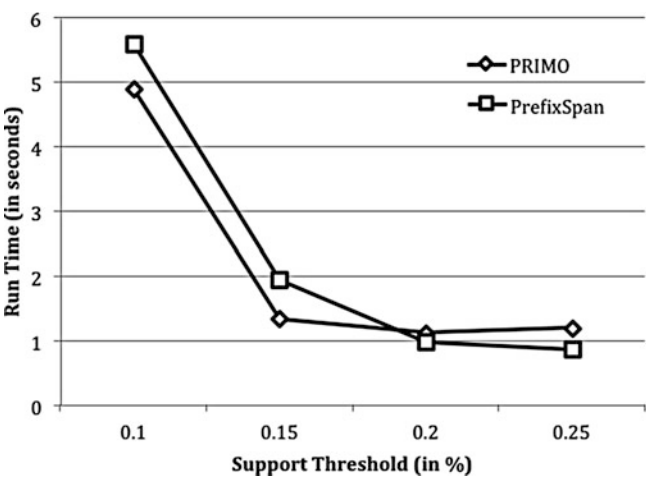

Fig. 2 Performance of the PrefixSpan and PRIMO for USER2 dataset

user from [6], which we refer to as USER2. The USER2 dataset consisted of approximately 26,000 records, each of which contained sequences that ranged from 1 to 24 items in length. Figure 2 displays the results that were obtained when executing each algorithm with varying minimum support thresholds.

The results of testing with the USER2 dataset revealed one limitation of the PRIMO algorithm. When testing higher minimum support thresholds and a smaller number of records, the standard implementation of the PrefixSpan outperformed PRIMO. We formulate that this is due to the overhead associated with the added complexities of the multithreaded structure we use to implement our algorithm. Although multiple threads have proven very useful when doing large computations the benefits diminish as the minimum support increases. Figure 2 also demonstrates the effects of a high support threshold on the PRIMO algorithm. When the minimum support reaches 18 % the algorithm no longer outperforms PrefixSpan. One possible solution to this would be reducing the number of threads that are created for executions that use small datasets and have high minimum supports. To apply this solution to the algorithm a simple classifier could be constructed to rank the datasets along with their minimum support thresholds by the predicted amount of computation time needed to generate the results. A system such as this could be applied to the PRIMO algorithm and be used to determine whether the algorithm should divide the computations into many threads or only a few. Several techniques have been developed to determine the optimal number of threads for a job, such as the work of Pusukuri, Gupta, and Bhuyan in [7]. They use a method that observes the programs execution at the operating system level to determine the optimal number of threads. Although adding a system to determine an optimal number of threads will introduce some overhead, overall it may increase the efficiency of the algorithm. One way to prevent the overhead of such a system from being added to the mining phase would be to include it in the data pre-processing task. After the data has been pre-processed the classifier could analyze it and predict the optimal number of threads and inform the mining algorithm of this predicted value.

AndroidMiner

In addition to determining the performance of PRIMO, testing was done to determine the applicability of pattern mining on the mobile computing platform. Comparisons were done between the executions of the PRIMO algorithm implemented on a standard Mac personal computer and a smartphone running the Android operating system. The dataset used for these tests was the same USER2 dataset that was used in the second round of performance evaluations.

Figure 3 shows a performance comparison of the PRIMO algorithm implemented on same personal computer used for earlier testing and a smartphone, which has a 1GHz processor, 1GB of ram, and is running the Android Jelly Bean operating system. Smartphone-based testing was done, not with expectations of better performance, but to determine the applicability of data mining in the mobile environment. As expected, the Fig. 3 clearly demonstrates that the computer-based implementation outperforms the smartphone-based version. This result was expected since the personal computer has a much faster processor than the smartphone and has more main memory.

While the performance of the smartphone was lacking in comparison to that of the personal computer, the results

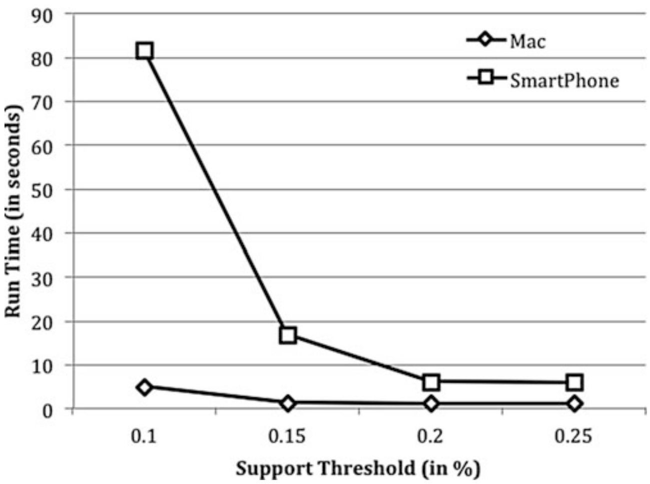

Fig. 3 Performance of the computer and smartphone-based implementations of PRIMO algorithm with USER2 dataset

demonstrate that smartphones are capable of preforming complex data mining processes. Observing Fig. 3 we can see that the only vast gap in performance exists for the execution that used a 10 % minimum support level. The main reason for this is the slower processor of the smartphone. When running the algorithm with a low minimum support the process becomes very computationally intensive. While a major difference exists in the run times of the two platforms, the results generated by each were equivalent. Thus showing that a smartphone-based data-mining platform is plausible and could be useful in mining the data of an individual user, but would require some optimizations to reduce the run times when working with very low minimum support thresholds.

Conclusion

In this paper we introduce the PRIMO algorithm and a smartphone application called AndroidMiner. Furthermore we evaluated the performance of the PRIMO algorithm and tested its applicability to the mobile computing environment using the AndroidMiner platform. In addition to testing we introduce the concept of combining algorithms to develop a better understand of the dataset. We also introduced a method for decomposing data mining tasks into separately executable threads that can be executed in parallel. Since implementing the multithreaded structure required few changes to the algorithm, this approach could potentially be applied to other algorithms, assuming they can be decomposed into separate tasks.

Our experimental results show that PRIMO outperforms the PrefixSpan and maintains the same level of accuracy in its results. Further testing demonstrated that the task of data mining can be accomplished with the limited resources of the smartphone platform. Although we determine that data mining on a mobile platform is possible, it was determined that techniques need to be developed to improve the mining process when working with the limited resources of a smartphone.

One possible extension of our technique would be to allow hundreds or even thousands of mobile devices to work together and combine results in a distributed environment. Also, techniques to dynamically determine the optimal number of threads could be added to the algorithm or to the pre-processing phase. This will also increase the performance when the algorithm is working with a large dataset, since the optimal number of threads may differ from the number of threads the algorithm creates.

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A Critical Review on World University Ranking in Terms of Top Four Ranking Systems

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Abstract

Now-a-days ranking of universities and institutions has become an appealing topic to study or research, and it has got wide attention to all over the world to recognize the top higher education institutes. Therefore, study on the strategies of the ranking system is vital to ensure the acceptability. There are number of strategies have been developed to rank higher education institutions worldwide. This Study has focused to critically evaluate the potential shortcomings of the top four widely accepted ranking systems. These are the Times World University Rankings, QS World University Rankings, Academic Ranking of World Universities (ARWU) and Webometrics Ranking. We critically reviewed and analyzed these four higher education ranking systems to identify potential shortcomings in their strategies. Based on our investigation, it was observed that none of these ranking systems can provide satisfactory evaluation in terms of their construct validity and other parameters related to disputation. Nevertheless, these ranking systems are the most popular for what they have been doing over the decades but unfortunately each and every one of them has to some extent lacking as far as ranking excellency is concerned. Lack of availability of data and publications through which ranking is done is one major obstacle faced to determine the authenticity of ranking systems. Overall observation of these four ranking systems reflects the fact that generic challenges include adjustment for institutional size, differences between average and extreme, defining the institutions, measurement of time frame, credit allocation, excellency factors as well as adjustment for scientific fields. Misinterpretation of measurement data is also responsible for some of the ranking disputes. We have proposed a number of recommendations that could address the identified inadequacy and considerably improve the ranking system as well as incorporate more participation of higher education institutes form developing world.

Keywords

Higher Education • Ranking • THE • ARWU • QS • Webometrics

Introduction

World university ranking has become a very appealing topic in the recent years. It has indeed made higher studies accelerate and boost up to a certain level. Various kinds of proposals were made by different ranking organizations to evaluate and determine performance of universities around the world. Literally, a widely accepted ranking system would help students in choosing their right institution to pursue

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_72, © Springer International Publishing Switzerland 2015

higher degrees, achieve funding facilities, help new researchers to carry on their research activities; even it helps develop a constructive competition among the institutions. There are a number of ranking systems exists, which includes, The Shanghai Jiao Tong University 'Academic Ranking of World Universities (ARWU)' [1], the Times Higher Education Supplement 'World University Rankings (THE Rankings)' [2, 3], Quacquarelli Symonds (QS) 'World University Rankings' [4, 5], Webometrics Ranking of World Universities 'Webometrics Ranking' published by Spanish National Research Council (CSIC) [6], The Guardian Higher Education Network, UNESCO Rankings and Accountability in Higher Education, US News Education, UNIVERSITAS 21, and NTU Ranking.

From the above ranking systems, we have selected top four (THE, QS, ARWU and Webometrics) widely accepted ranking systems to critically evaluate the potential shortcoming in their strategies. Based on the analysis, we identified a number of inadequacies in each of the ranking systems and made a list of recommendations to address them. We also highlighted several issues that could significantly improve the ranking system as well as incorporate more participation of higher education institutions form developing countries.

Review and Analysis

World university ranking has recently become a point of interest in different concerned organizations and institutes, which have been doing rankings over a period of years. We have critically reviewed the four most publicly visible ranking systems, the Academic Ranking of World Universities (ARWU) [1], the THE World University Rankings [2, 3], QS World University Rankings [4, 5], and Webometrics Ranking [6]. We have assessed the construct validity for educational and research excellence, and the measurement validity of each of the existing ranking criteria. Based on the assessment, we have tried to identify generic challenges and limitations in each of these systems for ranking of universities and institutions worldwide.

THE

The Times Higher Education World University ranking is one of the most widely accepted organization for ranking educational institutes and universities, powered by the Thomson Reuters [2, 3]. It is the only ranking table to judge world universities having focused on all of their core missions such as teaching, researches, knowledge transfer, and international acceptance. THE use the following indicators and weights to rank the universities around the world—40 % is determined by peer review, 10 % on ranking by major (mainly international) graduate recruiters, 20 % by citations (per capita) of published academic papers, 20 % by teaching staff: student ratios, and 10 % by 'international orientation' [3, 7]. By evaluating these parameters a "z-score" is developed by which afterwards the ranking is done, where "z-score" is a scoring table that standardizes scores from different criteria which is a very unique feature of THE [3].

QS

In cooperation with Times Higher Education (THE), QS has been issuing world university rankings since 2004 though eventually it severed its tie with THE in 2010. The following certain indicators and their weightage of QS World University Rankings—Academic peer review (40 %), faculty student ratios (20 %), citation per faculty (20 %), employer reputation (10 %), international student (5 %) and international faculty (5 %). Academic peer review is given the highest (40 %) weightage in this ranking method [5]. However in regional ranking like QS University Ranking: Asia, some of the weightage of the indicators were revised and giving 30 % weightage to Academic peer review [7]. Citation per faculty was divided into two categories—'Papers per Faculty' and 'Citations per Paper' and each category was given 15 % weightage.

ARWU

To rank universities globally, Shanghai Jiao Tong University established The Shanghai Ranking and its formal name is Academic Ranking of World Universities (ARWU). This ranking is heavily focused on the natural sciences over the social sciences or humanities which has opened the door for criticism. Research over the quality of instruction that also questions the integrity of the ranking [1]. It is more logical, invariable and it has achieved the worldwide appreciation because of its objective methodology [8]. Certain indicators and their weightage of ARWU are Quality of education (10 %), Quality of faculty (40 %), Research output (40 %) and Productivity (10 %).

Webometrics

Webometrics Ranking of World Universities is also known as Ranking Web of World Universities. Its ranking strategies are different compare to THE, QS and ARWU ranking systems. Webometrics indicators focuses on both the volume of web articles and influence & visibility of the publication. It depends on the external in links which is the base of the system [6]. Four indicators of Webometrics were obtained from the quantitative results provided by the main search engines as follows: Visibility or outer links (50 %), Size or web pages (20 %), Rich files (15 %), and Scholar (15 %).

Ranking Validity Assessment

THE apparently is doing a part of its ranking based on peer review. What THE do is simply asks experts to rank top 30 universities in their area, which might not be as reliable as it should be [2]. However the process of selecting the experts is not clearly visible. Without the transparency of any part of the methodology, evaluating excellence can be questionable. About the international character, THE believes having good amount of international students is not always the indicator of excellence. The ratio of student to faculty is another issue with transparency in THE. Another feature of THE is "Hundred under 50" ranking. This is a ranking for those universities which came up within 100 positions having an age of 50 or less. This ranking system is emphasizing on the future potential as well as present excellence of new and young universities.

Initially **QS** used to just ask people about their preferences about universities in different fields. These fields included technology, biomedicine, science, arts, humanities and social sciences. Additionally 2012's ranking has another data set which initially began in 2005. This is a survey of employers who are involved in graduate recruitment and this year ranking includes over 25,000 individuals' data. This indicators with four others sum up about 50 % of the prospective score for each of the universities. One of the indicators involves enlisting 5 years of academic references for the papers published there. QS Stars Methodology [9] are a rating system that allows students to get a clear picture of an institution's qualities, looking at everything from the employability of graduates, to sports facilities and community engagement.

Although the initial purpose of **ARWU** was to find the global standing of top Chinese universities, it has attracted a great deal of attention from universities, Governments and public media worldwide. ARWU or Shanghai Rankings have been reported by mainstream media in almost all major countries. Hundreds of universities cited the ranking results in their campus news, annual reports or promotional brochures. A survey on higher education published by The Economist in 2005 commented Shanghai Ranking as 'The most widely used annual ranking of the world's research universities' [1]. One of the factors for the significant influence of Shanghai Ranking is that its methodology is scientifically sound, stable and transparent [10].

Link analysis is considered as a powerful tool than citation analysis or global surveys, therefore **Webometrics** uses it for quality evaluation. Design and weighting of indicators are structured in terms of what was found in search engine based ranking.

Table 1 depicted the critical analysis and findings in respect to construct validity and measurement validity of THE, QS, ARWU and Webometrics ranking systems. According to the Table 1, it was observed that THE ranking system has much stronger construct validity for peer review. Citations per faculty are moderate. The graduate recruiter parameter is negatively marked in the table due to not having enough emphasize on this parameter. QS On the other hand, is not up to the mark in their employee reputation indicator as far as its construct validity is concerned. However, it has a good academic peer review background. International student parameter is negatively marked in the table for QS because of having disputed factors related to the scoring for international students. ARWU, is very good at the indicator of Faculty, Nobel/Field for its construct validity. However, number of articles they count is not good enough for its construct validity. Webometrics is very good in construct validity whereas result is almost unknown for Webometrics in measurement validity. Overall this table compares between construct and measurement validity for different indicators in respective ranking systems. It also shows relative analysis for the indicators so that it helps to identify which indicator needs to be improved.

Generic Challenges in Institutional Rankings

Adjustment for Size. Larger institutions may have more papers, citations, award-winning scientists, students, weblinks and funding. For example, the QS Rankings adopted reputational survey and quantitative indicators, both exhibiting 50 % in the weighting play an important role for their adjustment of size [7]. THE also follows institutional size, which they believe, the more is the size of the institution, the more is the possibility of greater teaching and broader research output. THE has 30 % of their weight on each of the indicators of Teaching and Research [3]. The weighting percentage has remained the same in the intended and the previous weights for indicators for adjustment of size in the indicators. On the other hand we can observe in the Shanghai ranking that sum of the five indicators (quality of education, AWA , HiCi, N & C and PUB) divided by the number of full time equivalent (FTE) academic objects [1, 11]. When the number of FTE academic stuff could not be collected then this criterion is ignored. Based on the size of the degree-seeking student body where an FTE number is not provided or available, one will be estimated based on common characteristics of other institutions in the country or region in question.

Averages and Extremes. The overall ranking systems those currently exist do not do equal justification to all the universities. Averages and extremes are an important issues

Table 1 Construct validity and measurement validity of discussed ranking systems

Ranking education	Indicators	Construct validity Research	Education	Measurement validity
THE	Peer opinion	+++	+++	–
	Graduate recruiter	–	+	–
	International faculty	+	+	?
	International students	–	+	?
	Student-faculty ratio	–	+	?
	Citations per faculty	++	–	+
QS	Academic peer review	+++	+++	–
	Faculty student	++	+	?
	Citation per faculty	++	–	?
	Employer reputation	+	+	–
	Employer reputation	–	+	+
	International student	–	+	+
ARWU	Alumni, Nobel/Fields	–	–	++
	Faculty, Nobel/Fields	+++	+	++
	Faculty, highly-cited	++	+	++
	Nature/Science articles	++	–	+++
	Number of articles	–	–	+
	Size	–	–	–
Webometrics	Visibility	+++	+	?
	Activity	+++	+	+

–, poor; +, low/modest; ++, good; +++, very good; ?, unknown (insufficient detail provided on the reliability of databases)

behind this natural fallacy. For instance, an institution having a department with some score 10/10 and another department with 0/10, the gross average becomes 5/10 which doesn't seem pretty appropriate marking. Nevertheless, both the "Extreme" and the "Average" cases are useful in order to know different queries depending on some particular scenarios [7]. Some ranking systems only focus on extreme issues disregarding all other average parameters. However while focusing on only extreme cases, along with the best extremes, the worst extremes should also be considered. So, that no injustice is likely to be done on institutions having lower ranks. If we only focus on best extreme cases, a university could go further high on the ranking just because of having some particular qualities which might really not represent the overall scenario of the university or institute. Therefore considering worst extremes will prevent from having any extra benefits for only some good qualities. Average cases are also necessary for most of the circumstances.

Defining the Institutions. Defining an institution is a bit complex and not really as straightforward as it seems to be. Size and nature of any institution depends on the fact that if the institution is clustered into sub-units or compact into one single unit as a whole. Different institutions have different scenario. So, on aggregate ranking it is more likely to over-rank or under-rank any institution if not defined properly. Thus, categorizing many different universities around the globe is not an easy task. Conversely, generalize these institution on one single scale is even more complicated. Therefore, an appropriate approach should be developed for defining different institutions effectively.

Measurement Time Frame. Time duration is also an important factor as far as university rankings are concerned. One real life example could be citation impact. It would need specific time duration to determine the importance of citation of any institution. But taking too long might otherwise be irrelevant to the institutions current state. However, time duration could, to some good extent, is less of a concern to those large institutions having a good strong background. Achieving or losing some influential scientists or a team might really not change their overall scenario. Mathematical data on various concerned papers show that large groups persist longer, in case they are able to dynamically change their membership. On the other hand, smaller groups develop themselves when their combination remains as it is. Thus, it reflects to the fact that smaller institutions would have significant changes if any modest changes occur.

Credit Allocation. Credit allocation focuses on how much credit should be given to any particular parameter for ranking. It is very critical in some cases. For instance, credit allocation for a Nobel Prize winner or any other exalted price winning is a concern on the fact that whether where they accomplished their projects versus where those winners are working currently. Another critical task is to allocate credit for some job that needs multiple scientists and institutions. No specific scientist and institution could be assigned all the credits. Like a paper, equally cited but authored by several institutions versus a paper authored by one institution. In this case several institutions authored paper should be given more credit [7]. Institutional definition also influences credit allocation. There is no systematic way until now through which credit could be allocated uniformly.

Overall Findings

Based on the review and analysis (Table 1) of the four ranking systems, we observed a number of the significant points those can be considered as shortcoming and limitations of these ranking systems. None of the ranking systems are perfect in a way or other. Each of them has inadequacy and weakness. Followings are the list of concerns that came out in respect to overall as well as each individual system with the perspective of effective ranking system for higher education institutions.

1. The ranking systems are not providing transparency about their assessment methods. Therefore it limits the proper evaluation and comparison of their qualities and shortcomings. Here we listed a number of reasons for considering intransparency:
   (a) All the ranking systems do not provide information and data sources. So it raises a question about the quality and integrity of the ranking.
   (b) The marking in most of the criteria heavily depend on the survey. So if the survey is not just or limited to a few numbers of people it may not show us the real picture and result in a faulty ranking.
   (c) Every ranking uses different values for weight in different criterion. As there is no ideal method, different weighted system may favor different institutions.

   All this reasons result in a substandard ranking system.
2. The indicators used in ranking the universities, are to some extent confined into some traditional ideas. The indicators of THE are: Research, Teaching, International outlook, Citation and Industry income. The indicators of QS are: Academic peer review, Faculty student, Citation per faculty, Employer reputation, International student and International faculty. The indicators of shanghai are: Quality of education, Quality of faculty, Research output, Productivity. Finally the indicators of Webometrics are: Visibility, Size, Rich Files and Scholars. For research related indicators, all four ranking give equal importance to all the subjects. But rather research related indicator should give extra importance and value to the fields such as quantum physics, astronomy, modern chemistry, computer science and technology etc.
3. Developed countries have ample opportunities such as: sufficient budget, and necessary resources and research opportunities, therefore they produce significance research outcomes. On the other hand, developing countries have inadequate budget, insufficient resources and they lack of expert people for research. Hence they are lagging behind significantly in scientific research and resources. This reason is also significantly affect the developing countries higher education institution and universities to compete in international ranking.
4. With the advancement of Internet technology, now-a-days plagiarism has become an important issue [12]. People are plagiarizing so frequently. Plagiarism is not having originality and academic dishonesty. Plagiarism is the act of taking another person's writing, even idea and passing it off as their own. There are many reasons why people do plagiarism. The reason can be not having enough time to think about and write the paper, without hard work wanting to get a better grade, feeling that the course is irrelevant to their career plans and hence not worth their time or effort, insecurity about their own creativity and writing ability, not having proper ethical values, struggles with a second language [13–15]. No ranking system has addressed plagiarism yet. The ranking system should introduce it to make the differences in the university ranking.

In the following we listed the shortcoming and limitations of each individual ranking system.

THE

1. THE asks experts to select top 30 universities in their respective area. But the selection process of the experts is entirely unclear [3].
2. According to THE, institutions having large amount of international students deserve better position in the ranking list, which in reality doesn't necessarily mean that having international students is always a positive issue. It depends much on the country the institution is situated in.
3. Number of ranking parameters is still disputed as far as different settings on different institutions are concerned. For example, Nobel or any other prestigious prize winning by any staff or any institution.
4. The reputation table ranks institutions according to an overall measure of their esteem that combines data on their reputation for research and teaching [3].

QS

1. The QS World University Rankings have been criticized by many for placing too much emphasis on peer review, which receives 40 % of the overall score [7].

2. There are still many issues remaining to be dealt with regarding citations in different ranking systems as the arts and humanities generate less citation compared to scientific journals and other publications.
3. From the beginning, the QS Rankings have relied on reputational indicators for half of its analysis [5, 16].
4. The process of calculating return questionnaire for university reputation, QS Rankings failed to control the number and qualification of questionnaire, thus leading to a selection bias [7].

ARWU

1. Five of the six criteria used by ARWU are counting criteria (prizes and medals, highly cited researchers, papers in N&S, papers indexed by Thomson Scientific). Hence, it should be no surprise that all these criteria are strongly linked to the size of the institution.
2. Their process of measuring 'academic excellence' has discrimination between top ranked and lower ranked universities [17].
3. This ranking claims that it uses carefully selected objective criteria which are based on internationally comparable data. But as they do not make this data publicly available it is not possible to check the authenticity of this data [17].

Webometrics

1. Promoting Web publication was the original aim of the Webometrics. Webometrics's primary targets were supporting Open Access initiatives, electronic access to scientific publications and to other academic materials.
2. Often universities change their web domain keeping the older one valid as well. This is considered as a bad practice and such institutions are penalized [6].
3. On which basis universities are being ranked is not clear to the audiences. So there is still some misunderstanding and misconception.
4. There are prejudices in the coverage of searchers' databases. And some methodological problems are also there.

Recommendations

Overall Recommendation

On the basis of the investigation which was made in this study, we come up with a number of recommendations to address the limitations and make the existing ranking systems more robust. Recommendations are given on the basis of the common problems found in common in the four ranking systems as well as specific ranking system investigated this study. The significant errors of each ranking systems are highlighted and the potential recommendations are provided as follows to avoid the identified problems of the ranking systems.

1. Raw data and analytical parameters should be made public to make the ranking process more authentic and trust worthy. Another important fact is that the process of selecting experts should be transparent as they play important role to decide the ranking process and procedure.
2. Innovative non-traditional ideas for indicators should be achieved by research. Traditional indicators discriminate the institutions at developed and developing countries.
3. Developing countries are lack of research & development. Ranking system could take consideration about the fact that the developing countries need alternative ranking system for themselves like QS Asian University Rankings and the QS Latin American University Rankings.
4. The optimistic way to avoid plagiarism is to understand what it is. Then steps should be taken to avoid committing either accidental or intentional plagiarism. Catching those students who plagiarize may stop other to start practicing plagiarism. Ranking system may adopt the step that if any paper is found with plagiarism from a university then that university will get negative point and they will lag behind in university ranking. So the faculty members and the students of respective universities will be concerned about not getting negative point.

THE

1. The selection process of the experts should be made transparent and should be added to the website also in user friendly manner so that visitors in the website can also possess a clear idea on how experts are chosen.
2. Having international student is obviously a positive marking in a viewpoint of a university, but that's not the end of the world. International student admission is not only concerned with the university, it's far beyond. Political stability, government relations between the countries students are transferred to or from all are co-related behind this issue. So, before regarding it as a strong parameter for ranking THE ranking should just make sure they account on those things. For example, getting visa to study UK itself is getting tougher day by day for the third world countries recently.
3. Any disputed parameters may be ignored or redesigned while ranking. The expert selection method being

non-transparent should not come along with other parameters for example.

4. The reputation table ranks on the basis of overall esteem of an institution which mainly focuses on teaching and research, but the overall reputation should be an issue with other facilities in an institution as well.

QS

1. Most controversial part of the QS World University Rankings is their use of an opinion survey referred to as the Academic Peer Review. As the survey for peer review asks active academicians across the world, it actually gives a clear picture for a certain university's condition in the job and academic sector. At the same time other indicators are important as well. So, all the indicators should get enough weighting.
2. Global performance factors like international collaboration among universities or scholars should be considered and evaluated more precisely in the ranking systems for better understanding of a university's worldwide position.
3. Countries from the Commonwealth of Nations contributed 32 % of the return questionnaire, while Asian countries, limited in only a few countries, occupied 22 %. So, the contribution of the other countries should be increased.
4. QS Rankings have relied on reputational indicators for half of its analysis, it should give more emphasize on the qualitative and quantitative indicators those are more practical.

ARWU

1. The evaluation criterion of ARWU involves several arbitrary parameters and many micro-decisions that are not documented. So they must make raw data publicly available so that it is possible to analyze the robustness of the final ranking with respect to these elements.
2. Five of the six criteria used by Shanghai ranking are counting criteria (prizes and medals, highly cited researchers, papers in N&S, papers indexed by Thomson Scientific). So they must introduce some new criteria which not only depend on size and quantity but rather grade quality.
3. The criteria ARWU uses to measure the academic excellence contribute almost to nothing in the ranking of institutions that are not in the top 100. So they may introduce effective criteria that will also measure the universities that are not in the top shortlist.

Webometrics

1. Institution's authorities should also enforce an adequate web policy, or promote the electronic publication along with their academic excellence to gain higher position in the ranking.
2. Quality of education provided and academic prestige are correlated well by Webometrics ranking. But other non-academic variables need to be taken into account also.
3. Google's Page rank algorithm is a tool which is used by Webomatrics. But there is a chance of ranking all the population of the higher education institutions with their own web domain. Using several search engines has been proved an effective approach to reduce this issue.
4. Webometrics can reduce the misunderstanding and misconception about it by publishing the details of indicators along with the rankings.

Conclusion

The evaluation and analysis is made on the basis of THE, QS, ARWU and Webometrics which are the most popular ranking systems. It's thereby seen none of the ranking systems are perfect as far as transparency as well as other concerned. Several recommendations were proposed in this study to make these ranking systems more robust and transparent to the end level users.

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Multi-agent Architecture for User Adaptive Information Retrieval Systems

Juan J. Pacheco-Reyes, Oleg Starostenko, Vicente Alarcon-Aquino, and Jorge Rodriguez-Asomoza

Abstract

This paper presents the proposed architecture that provides platform for implementation of intelligent software agents (Multi-Agent System) used for information retrieval based on user-oriented model. The existing architectures based on multi-agents as usual lack the ability for adaption to user requirements. Thus, the principal goal is to develop standard reusable scalable infrastructure based on intelligent multi-agents that incorporates particular user model providing relevant information to user queries taking into account his profile, operational patterns and preferences. The evaluation of the proposed architecture has been done comparing it with existing approaches taking into account relevant criteria such as used communication standards for information exchange between agents, available scalable layered structure and ability of adapting information or services to user habits, personal data and requirements. This approach has sufficient merit to be used as a reference for development of applications for user-oriented and adaptive information retrieval systems.

Keywords

Adaptive systems • Information retrieval • User preference model • Multi-agent architecture

Introduction

Since the beginning of Internet the quantity of information presented in Web grows exponentially. The problem now is how to find and to retrieve information that users are really seeking. Today anyone generates new information in Web by simply taking a picture, recording a video with your phone or electronic tablet and shares it via Facebook, Twitter, YouTube, blogs, etc. as well as writes notes anywhere and anytime. The diversity of content and the fact that each user connected to network have their own preferences and objectives have motivated to develop mechanisms for access and adapting information to user's needs.

To achieve adaptation of required information it is important to have appropriate infrastructures and systems to handle user-oriented preferences for fast processing and recovery only the most relevant data to user. Nowadays, there are some available approaches that use intelligent agents design methodologies such GAIA, ROADMAP, Tropos, ADELFE and others. They may be used for development of multi-agent systems however, previously a standard reusable scalable infrastructure must be proposed in order to incorporate particular user model to information search and retrieval system providing in this way adapting that information to user profile, operational patterns and preferences.

The paper is organized as follows. Section "Intelligent Agent Architecture" presents taxonomy of intelligent agents as well as description of methodologies used for agent development. Section "Information Retrieval Approaches"

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_73, © Springer International Publishing Switzerland 2015

describes analysis of existing approaches for development of user-adaptive information retrieval systems. Section "Existing Adaptive Systems and Proposed Generalized Multi-agent Architecture" discusses known efficient multi-agent infrastructures and their comparison with the proposed architecture. Finally, the evaluation of multi-agent architecture and contributions of the paper are presented in conclusions.

Intelligent Agent Architecture

According to Russel and Norving [1] *an agent is anything that can be considered as something that perceives its environment through sensors and acts on through actions* (see Fig. 1).

Currently, there are various definitions regarding what an intelligent agent is and which environment is appropriate for specific agent. According to Gunga an intelligent agent is: *an autonomous system that shows a certain degree of independence in its environment* [2]. In other words, intelligent agents are physical or virtual entities that take decisions autonomously. In our case we are only interesting in virtual intelligent agents. These types of agents can represent real-world objects such as atoms, processes, people, animals, wizards, etc. The environment is the virtual space, where intelligent agents interact. A distributed virtual environment has its resources (processing time, memory space), its objectives (maximize production, reduce risks) and its decisions based on pre-established rules, reacting and adapting itself to situations or environmental conditions.

There are different types of intelligent agent architectures that determine items, which agent may use reacting to stimuli, act and communicate with each other. These architectures define a structure of interacted modules to achieve their required functionality. However, there is not unique standard methodology that allows construction of intelligent agents.

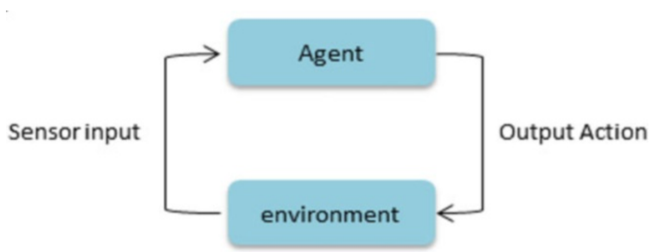

Fig. 1 Representation of agent concept by sensor-act interaction in particular environment

Agents may be classified considering the type of employed information processing particularly, grouping them as reactive, deliberative and hybrid agents. They also differ by the knowledge obtained through interaction of their very basic actions. Some other classifications are resumed in Table 1.

During interaction of agents in a heterogeneous environment many problems arise due to complexity of their management and coordination as well as increasing unexpected situations, security holes, etc. Therefore, to propose a solution it is necessary to adhere to the methodologies of agent-oriented design. In Fig. 2 some well-known methodologies are presented.

In many cases, a single agent cannot solve whole problem of information access and retrieval therefore, several agents need to interact with each other making too complex their management. These types of systems need other methodologies such as MaSe and ADELFE, which are designed to create multi-agent systems.

In order to manage two or more agents the appropriate communication between them must be established defining a used protocol on both the syntactic and semantic levels.

In our proposal the used protocols are based on some standards of Foundation for Intelligent Physical Agents (FIPA) [3]. These protocols are defined by the FIPA as it is described in Table 2.

Information Retrieval Approaches

Originally, the information retrieval systems have been developed to help manage large amounts of scientific literature. Currently, many universities, companies and public libraries utilize these systems to provide access to books, journals and other documents. Many of these systems are based on techniques that provide efficient textual and visual information retrieval using approaches such as Boolean model, probabilistic model, vector model, fuzzy logic, neural networks, Bayesian networks, structured text, etc.

Table 1 Taxonomy of intelligent agents

Individual agent systems		Multi-agent systems	
local agent	Network agents	DAI based agents	Mobile agents
Personal assistants Help assistant agent Meeting Planner	e-mail agents Information retrieval agents Process automation agents	Distributed problem resolution	Telecommunications Personal communication Network management Services on demand e-mail Marketing

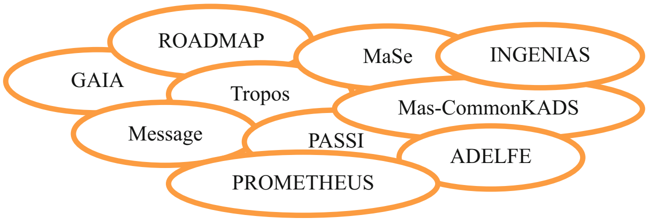

Fig. 2 Well-known agent-oriented methodologies

Table 2 FIPA communication protocols

Protocol	Description
Request	Request to agent to perform an action
Request when	To an agent is asked performing action upon an event
Query	Request to an agent to send report
Contract Net	An agent operates as administrator and requests other agents to perform a task. The administrator receives proposals and selects the best ones to be performed
Brokering	An agent acts as intermediary one between two agents
English auction	Several agents participate in auction that starts with a lower price and rise progressively
Dutch auction	Several agents participate in auction that starts with a higher price and progressively is declined
Recruiting	Similar to brokering, but the answers in this case goes directly to the applicant
Propose	The initiator agent proposes the creation of a task to other agents, which they may accept
Subscribe	An agent asks to be notified if a certain condition is true
Iterated Contract Net	Net Contract Extension with multiple iterations of bidding

Table 3 Sample terms and document identifiers

ID	Terms	Document ID
T1	Library	{1, 3, 6, . . .}
T2	Cinema	{4, 7, 12, . . .}
T3	News	{1, 2, 3, 4, . . .}
T4	Vacations	{1, 3, . . .}
T5	Videos	{1, . . .}

For example, the Boolean model uses logical operators such as *AND*, *OR*, *NOT* and *XOR* to combine multiple queries (terms to be found) and relevant documents during searching process. In Table 3 the example of information retrieval is shown for case, where queried terms are found in some documents using *AND* operator. The term *T1* indicates that it appears in the documents *1*, *3* and *6*, whereas the term *T5* only appears in the document *1*.

Some tests were made on 20,000 records with information about press, radio and television news using on-line Blazquez's simulator [4] that provides retrieving information with Boolean method. The obtained results are shown in Table 4.

Table 4 Information retrieval results using Boolean model

ID of query	Boolean operator	Retrieved documents	Runtime (μs)
1	Policy *AND* National	45	0.10376596450
2	Policy *OR* National	1,437	0.19587302207
3	Policy *XOR* National	1,392	0.19147014617
4	Crisis *AND* Economy	49	0.09648394584
5	Library *AND* sport	3	0.08790493011
6	Elections *AND* Europe	7	0.13520288467
7	Museum *NOT* Library	175	0.09561800956
8	Europe *AND* Labor *OR* Crisis	395	0.20249605178

The model has some disadvantages. For example, it does not take into account the frequency with which terms appear in documents as well as it does not provide information if a document is relevant or irrelevant.

This problem may be solved in systems that use semantic analysis of document content or, for example, in adaptive intelligent systems, where automatically detected user patterns, habits or previously defined preferences are taken into account in searching and retrieving processes.

This problem may be solved in systems that use semantic analysis of document content. Additionally, in adaptive intelligent systems automatically detected user patterns, operational habits or previously defined preferences are taken into account in searching and retrieving processes.

As usual, adaptive system includes a user model that consists of explicit representation of certain user features. Each model is created for particular user and it is constantly updated to keep available the current state of corresponding user.

There are a lot of approaches of Artificial Intelligence that have been used for development of adaptive systems, since they have to perform various tasks in the process of adaptation and take relevant decisions that must minimize user effort during information seeking and must maximize user degree of satisfaction.

As it has been mentioned earlier, development of software agents is important way used in adaptive systems because they are able to react immediately to events in distributed environment. One of implementation techniques used for this proposal is dynamic programming, i.e. using programming languages that may be able to update the application while it is running. Additionally, it is important to apply some approaches based on decision theory that provides a combination of utility theory (description of user preferences) and the theory of probabilities (representation of uncertainties), as well as techniques such as learning by reinforcements and probabilistic networks also known as Bayesian networks and decision networks.

There are many applications that currently are used, for example, in analysis of crime and detection of fraud, in shopping support systems, in manufacturing, in demand video systems, image processing, education (Adaptive Hypermedia Systems), and others.

Existing Adaptive Systems and Proposed Generalized Multi-agent Architecture

There are a lot of scientific reports about agent-oriented infrastructures that adopt different approaches for generation of adaptive systems. Some more relevant systems and architectures are described below.

(A) The *Dynamically Adapting BDI Agent Architecture based on High-level User Specification* is based on framework of Personal Assistance Software and it handles a high level of adaptation of application using predefined set of rules and actions [5]. This research is the most similar to our proposal; however, it is based on client-server application, while our proposal is multiplatform Web service guaranteeing its implementation in different environments.

(B) The *Agent Based Information Retrieval System Using Information Scent* is implemented on architecture that allows design of personalized user information retrieval systems using clues or traces, which recommend other sites that are similar to user query. However, it is not clear how this information fits to user profile, i.e., the system recommends similar sites only at a time, when user performs a search without taking into account the searching history [6].

(C) The *Intelligent System: Recommendation for Learning Objects* represents implemented recommender based on intelligent agents. The main function of this prototype is to retrieve documents on base of user profile [7]. The limitation of this project is that, the framework does not work with other Web applications and it has no structured scalable layered architecture. This architecture serves only to develop information systems for recommendations without detection of preferences and needs of particular user.

(D) The reported *U-Learning within a Context-Aware Multiagent Environmen*t is based on multi-agent architecture and context sensitive learning scenarios. Implementations of this architecture will serve for exchange of information and knowledge of public interests. The project itself looks like a social network, where groups of people share their knowledge. The architecture does not show records of history of each user [8].

(E) Described in [9] *an E-Commerce Applications Based on the Multi-Agent System* implements a distributed computing architecture for business, where a group of agents work as mediators for purchase of products. The used multi-agent architectural elements are: *Resource, User Interface Agent, Resource Agent, Host Agent, and Mobile Agent.* The architecture does not handle search adapted to needs of users, where they can at least have recommendations for product.

(F) Proposed in [10] *Architecture of Multi-agent Systems for the Health Sector* is implemented on JADE and provides FIPA standard definitions. Although these elements are part of our proposal, this project is focused on the health sector and therefore lacks necessary elements for architecture that can be used for adapting information.

(G) A *System for Semantic information management of graduated* is based on architecture of multiple agents: interface, semantic and crawler wrapper agents. The interesting point of the proposed system is its ability to support ontologies and management of stored information about each user. However, this architecture is far from having elements that allow adaptation of information [11].

(H) The project called *Cognitive Multi-agent Systems for Integrated Information Retrieval and Extraction over the Web* presents very interesting proposal based on a multi-agent architecture consisting in extracting information from Web pages with similar content and structure [12]. The use of ontologies to recognize pages provides extraction of all the possible information. A main feature of this architecture is reduction of nonsensical information considered as a trash. It is clear, that this project presents search engine and does not consider user preferences.

(I) The *Similarity-Based Resource Retrieval in Multi-agent Systems by Using Locality-Sensitive Hash Functions* presented in [13] uses encryption algorithms such as SHA-1 based on hash functions and cryptographic message digest algorithms as MD4 or MD5 to authenticate if documents are original. The used architecture consists of three layers: LS-Agents, Agents and resources. The project is focused on verifying the integrity of information and based on a multi-agent architecture, however, it does not consider requirements for adaptation of information to user.

(J) Finally, *Real-Time Agent Architecture* is presented in [14], where agents as actors interact together to achieve goals under time constraints. The architecture provides layered structure of the CORBA standard, which allows different programs written in different programming languages run on any computer. The used layers are: Real-Time Agent Communication Layer, Real-Time CORBA Services, Real-Time ORB and Real-Time Operating System. In conclusion, this architecture serves as basis for deploying applications, where multiple agents may interact and perform tasks in real time.

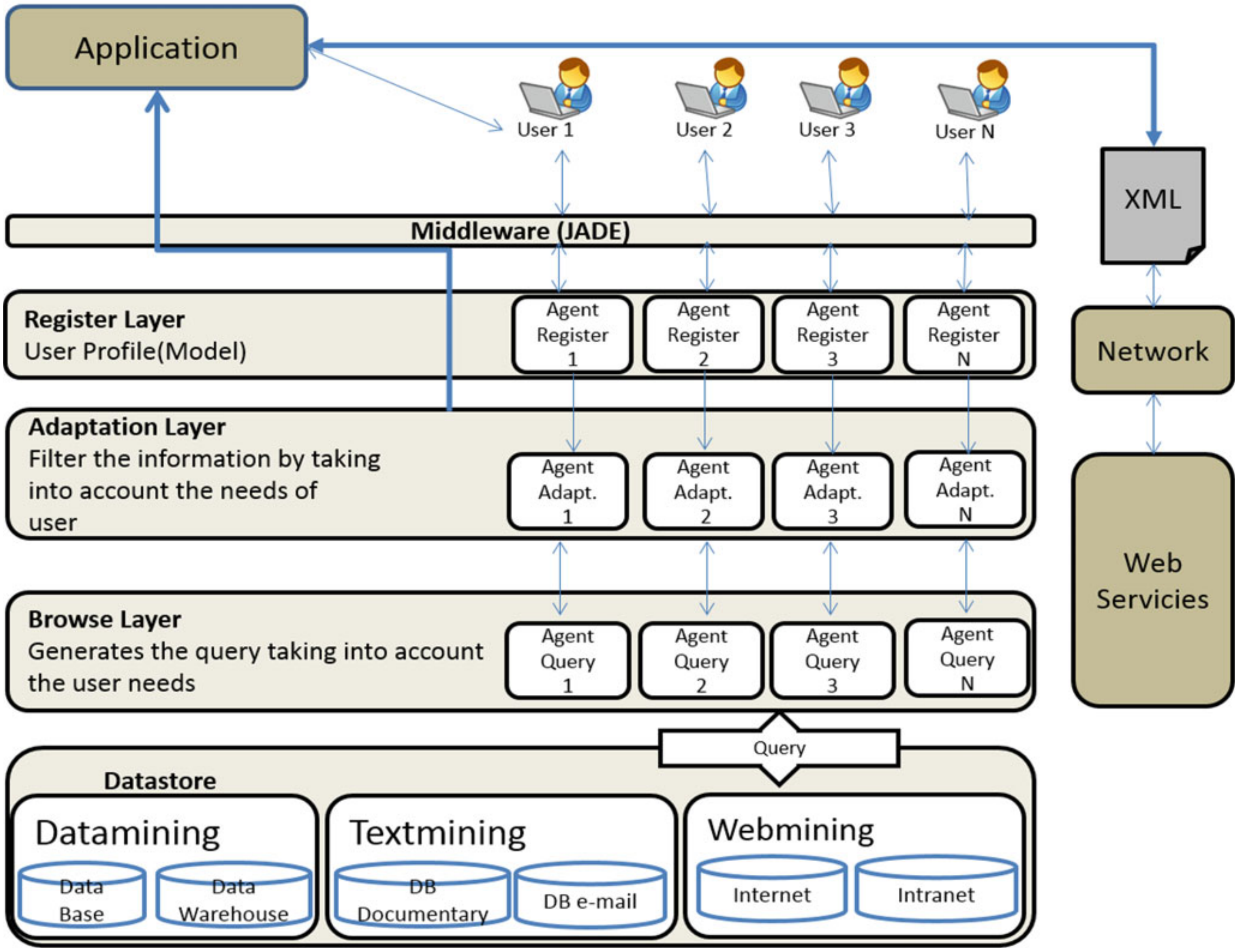

Fig. 3 The proposed architecture for adaptive information retrieval system based on intelligent multi-agents

After analysis of recent reports about existing adaptive systems, the proposed layered architecture based on intelligent multi-agents may be introduced as it is shown in Fig. 3. The principal goal is to propose standard reusable infrastructure based on intelligent multi-agents that run within distributed environment incorporating particular user model. These agents must interact with each other for the sole objective to provide relevant information to user taking into account his profile, operational patterns and preferences.

The *Application Layer* provides the graphical user interface that allows access to services of other layers reducing in this way the complexity of system.

The *Middleware* based on JADE template engine is widely used platform to implement the agent paradigm. Another important complementary middleware that has been proposed to use is JADEX reasoning engine as agent-oriented framework on XML and Java.

To achieve interoperability of used technologies or applications, the project is implemented as *XML Web Service*. It allows heterogeneous systems work together helping to bridge the differences between them on the levels of incompatible operating systems and programming languages.

The *Register Layer* is responsible for recording user information and activities, among which there are most visited sites, browsing time at each site, topics of interest, etc. At the same time the layer is assessed whether the recommended information for user is appropriate to his profile and is of his interest updating dynamically the preferences if necessary.

The *Adaptation Layer* is responsible for managing the input data flow by information filtering according to established preferences relevant to user. This layer works as intermediary agent between the *Register* and *Browse* layers. It provides application of selected *User Model* (structured data) analyzing and preparing those data to be sent to the *Browse layer*. At the same time layer receives the searching results from *Browse layer* and applies semantic

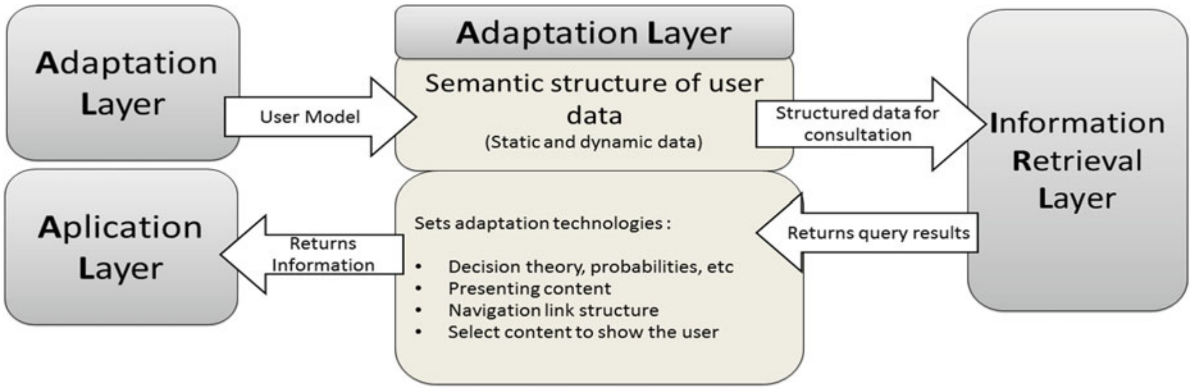

Fig. 4 Block diagram and main functions of the *Adaptation Layer*

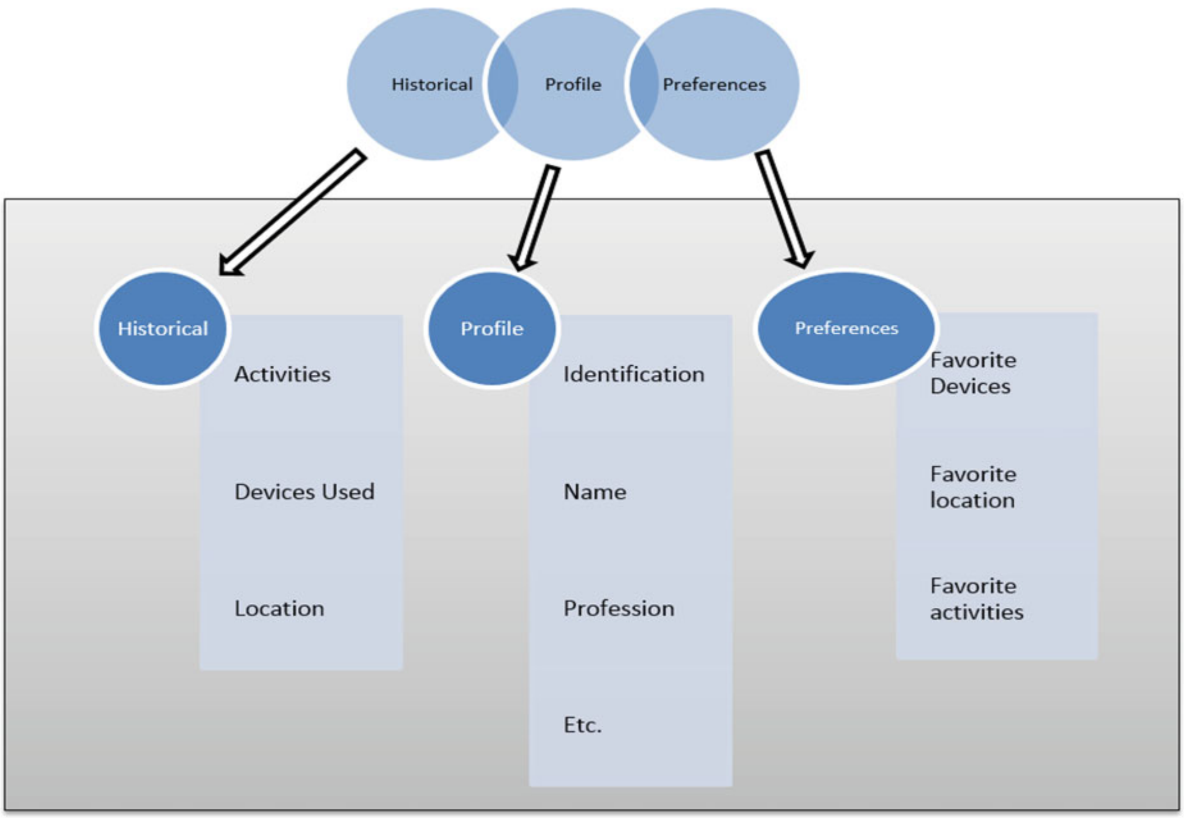

Fig. 5 Structure of *User Model*

filtering to deliver only relevant user-oriented information to the *Application layer*. The block diagram of *Adaptation Layer* is presented in Fig. 4.

User Model stores relevant information about user. It includes description of user personal data, preferences, habits, goals, track records, socio-cultural aspects, possible disabilities, experiences, operational abilities, etc. One example of *User Model* structure is presented in Fig. 5.

Browse Layer receives information from *Adaptation layer* and generates structured queries taking into account user profile and preferences. It sends either all necessary parameters to search engines or runs information retrieval algorithms returning results to the *Adaptation layer*.

Finally, *Data store* is the collection of data that may be classified according to certain fields (enterprise, organization, universities, manufacturing, etc.) or oriented to particular user needs such as specific understandable structures, textual or visual documents or perhaps, content of Web pages, etc.

Table 5 Reviewed architectures for adaptive multi-agent systems

Reviewed architecture	Communication standards	User profile	Layered structure	Adaptability	Web service
A	–	X	X	X	–
B	–	–	–	–	–
C	X	X	–	–	–
D	–	–	–	–	–
E	–	–	–	–	–
F	X	X	–	–	–
G	–	–	–	–	–
H	–	–	–	–	–
I	–	–	X	–	–
J	X	–	X	–	–
Proposed	X	X	X	X	X

Evaluation of Proposal and Discussion

In order to evaluate the proposed architecture, the comparison of principal features of existing approaches for development of multi-agent adaptive systems has been done as it is presented in Table 5. In the table some relevant criteria have been taken into account such as used communication standards for information exchange between agents, user profiles, personal data and preferences, available scalable layered structure, ability of adapting information or services and system implementation as Web service. The reviewed in section "Existing Adaptive Systems and Proposed Generalized Multi-agent Architecture" approaches are labeled in Table 5 with the same letters (A, B, . . . J).

According to Table 5, the existing approaches and systems lack many elements that we propose in our architecture, particularly: scalable layered structure, independent management of information on each layers, implementation as Web service and ability of adaptation to user requirements. This does not mean that mentioned projects are not efficient, just they were designed for specific tasks [15].

The proposed architecture is designed as generalized structure for multi-tasks and implemented as multi-agent system with ability to manage user-oriented information and services.

Conclusion

After analysis of well-known approaches for development of information retrieval systems it has been detected that existing architectures based on multi-agents usually lack the ability to adaption to user-oriented requirements. The proposed generalized architecture supports FIPE standard as compatible reusable scalable infrastructure based on intelligent multi-agents that run within distributed environment incorporating particular user model that takes into account user learning software patterns, operational habits and preferences.

This approach has sufficient merit to be used as a reference for development of applications for user-oriented and adaptive information retrieval systems based on intelligent multi-agents. The proposed architecture may serve to exchange data and information by software agents based on XML Web services transparent to user, establishing adaptation layer as additional benefit in efficient information searching engines, which operates without nonsensical results.

Acknowledgment This research is sponsored by European Grant: Cloud computing-based Advisory Services towards Energy-efficient manufacturing Systems (EU FP7 PEOPLE IRSES) and by Mexican National Council of Science and Technology, CONACyT, project #154438.

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Case Study: Challenges and Issues in Teaching Fully Online Mechanical Engineering Courses

Sara McCaslin and Fredericka Brown

Abstract

Every year more engineering programs are looking at online courses as a way to expand their programs and facilitate the educational goals of working professionals. This case study summarizes specific challenges faced by two faculty members in preparing and presenting six mechanical engineering classes, all core classes at either the graduate or undergraduate level, in a fully online format. The challenges discussed involve course preparation and planning, interaction with and among students, lack of student preparation, and exams.

Keywords

Distance education • Online education • Engineering education • Mechanical engineering

Background for the Case Study

This paper presents a case study in the preparation and presentation of engineering graduate and undergraduate courses in a completely online format.

The mechanical engineering courses listed in Table 1 were taught as fully online courses at The University of Texas at Tyler. These courses represent one core class at the undergraduate level and three four core classes at the graduate level. The faculty members involved taught the same classes in a face-to-face setting in previous semesters. All courses, while fully online, did include students who were on-campus. For the graduate level courses there was a mix of working professionals and traditional, full-time graduate students.

All course material was delivered to the students through the course management software *Blackboard*. All assignments and exams were submitted electronically through *Blackboard*. Communication with faculty was primarily accomplished through email, but phone calls and discussion board posts were also used as needed.

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Issues in Course Preparation

Workload

The initial perception of many faculty seems to be that online classes require significantly more work. In 2000, Visser performed an empirical study comparing delivery of the same class delivered online versus face-to-face, and found that the ratio of preparation time to contact hours was almost double for the online course [1]. Later in 2004, Hislop and Ellis came concluded that face-to-face teaching actually required more time per student [2], as did Van de Vord and Pogue in 2012 [3]. However, the latter admitted that there were some aspects of online teaching that did require more time per student.

It is doubtful that any certain rule would apply to faculty members because of varying levels of skill and experience with the software and technology used to prepare an online course, even though they may have the same level of teaching experience in general. Furthermore, yearly updates and advances in both software and technology can have both a positive impact and a negative impact on the time required. New software may save time, but there will be a time-consuming learning curve involved.

K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_74, © Springer International Publishing Switzerland 2015

Table 1 Classes involved the case study, where G is graduate level course and UG is undergraduate level course

Semester	Course	Instructor	Students	Level
Fall 2012	G Core #1	B	13	G
	G Core #2	A	13	G
	UG Core #1	A	9	UG
Spring 2013	G Core #3	B	14	G
Fall 2013	G Core #4	B	8	G
	UG Core #1	A	12	UG

To reduce the time involved in preparing an online course, it is very important to be thorough with planning out the course before the semester starts.

Course Planning

Both professors, based on their experience, strongly feel that the best approach to preparing an online graduate course is to divide the subject material into modules. Each module should have assigned to between five and seven learning objectives. The module should then be broken down into "chunks" of material, or sub-modules. The reasoning behind this is quite practical: Smith has pointed out that most online students (especially those who are working professionals) will not have uninterrupted time to work on a web-based course [4]. These "chunks" are short sub-modules ranging from 5 to 15 min with each focused on a single topic or learning objective [5].

In preparing materials for their classes, both faculty members did their best to follow these guidelines. This modular approach can pose a considerable challenge to faculty who are accustomed to filling 50-min time slots with material. However, note that just because the material is broken into smaller chunks does not mean that any additional time is required. In fact, in the long run, this may save time for the online faculty member who needs to update course material. Rather than rerecording an entire 50 min lecture, there may just be a short 5 min lecture that needs to be updated while the rest can remain .

Ideally each online module should include the presentation of new material (which could be in the form of a document, slides, or a streaming video), an assignment, and a short quiz to be used for self-assessment [4]. For each module, both faculty members aimed to provide a short guide listing the learning objectives, contents of the module, and references to the textbook or other supplemental material. This provides the students with a roadmap for accomplishing the tasks for the module.

One aspect of preparing videos that is often forgotten is the time required to record, edit, produce, and upload the videos to a course management system. Links must be then provided to the students so they can access the videos. This process does take more time than merely writing and presenting a lecture live to a classroom full of students, and upload times (depending on available internet speed) can lead to frustration for the faculty.

Presenting the Material

Another challenge in converting a traditional course to an online course involves selecting the best means to communicate the material to the students. Many faculty members will continue to use the traditional slide-based delivery with narration, thus substituting a video recording for a lecture. Online course delivery, however, allows for far more flexibility and makes it possible to combine multiple tools to present the material as clearly as possible. One of the benefits of online teaching is the ability to present concise lectures supplemented by additional materials, both those created by the instructor and those available on the web.

Camtasia is one of many video editing packages that allow faculty to record narrated *PowerPoint* lectures and screen recordings. Both faculty used *Camtasia* for presenting narrated lectures, as well as "how-to" videos for engineering software packages or examples. *Jing* is a free screen capture tool that allows the user to quickly capture equations, images, etc. from other documents and paste them into a presentation or another document. Software packages such as *Softchalk* enable instructors to include slideshows, audio, video, images, widgets, and embedded self-assessments in a course module [6]. Both faculty had issues with the learning curve involved, but not so much in using *Softchalk* but integrating *Softchalk* content into Blackboard.

For presenting derivations or example problems, the *LiveScribe* pen can combines digital capture of handwriting with audio narration at far less cost than a tablet PC [7]. Moore et. al. describe its use as follows: "The smartpen technology works differently from other digital pens in that it records both the written word on the page and audio simultaneously, which can then be played back by tapping the handwritten marks on the page or saved as recordings that can be transferred to the computer as a viewable movie and played back" [8]. This technology already has a proven track record in engineering education [9, 10], and one author in particular received very good feedback from students regarding this approach to working out example problems in continuum mechanics.

Core graduate courses 1, 3, and 4 used a combination of narrated *PowerPoint* or *SlideRocket* slides, how-to videos recorded using *Camtasia* in addition to *YouTube* videos. Example problems and some derivations were presented using the *LiveScribe* pen. The core graduate course 2 and undergraduate core course 1 combined *PowerPoint* slides supplemented by *YouTube* videos, with example problems worked and recorded using *Camtasia*.

Interaction and Feedback

Interaction Between Faculty and Students

Ubell reports that one faculty member teaching an online engineering course (of similar size to those that formed this basis of this paper) likened the process of online teaching to teaching through a straw [11]. There is no opportunity to read body language, recognize tone of voice, or pick up on other non-verbal indicators of student learning. The only medium of communication is electronic, which is typically via a course software platform and email. This may be one of the most challenging aspects of online teaching for those of us who are used to looking at our students while we lecture, watching for signs of recognition or confusion. Too often, the authors received feedback from students regarding confusion or frustration after an assignment or exam had been administered (and sometimes not until the end of the semester).

Student-instructor interaction is an extremely critical issue for a successful online course. Some students perceive student-instructor interaction to be more difficult in an online setting [12], unable to really seek out help unless they can visit with the instructor face-to-face. In addition, students tend to get frustrated waiting for an email response to their question, especially compared to the instant feedback they can often receive in a face to face to class [13]. There is a documented need for the instructor to take a proactive role in communicating with the students, rather than waiting for the student to contact the professor [13].

To reduce the number of inquiries, faculty members should be as clear as possible when giving instructions. Smith identified five key considerations for online faculty [4]:

1. Provide detailed instructions and do not skip steps
2. Be proactive by emailing students regularly during the duration of the course
3. Tell students what the expected response time will be for email
4. Define times when you will be available for immediate email responses
5. Develop self-assessment tools will help students determine more quickly when they need to seek assistance

The experiences related to this case study have proven the usefulness of these guidelines. One faculty member found that just sending out a general email to the entire class asking how things were going would get responses from students who might be self-conscious about initiating a request for help.

Note also that self-assessment tools are a powerful means of allowing students to gauge their level of understanding, and the authors noticed that when self-assessment tools were not available the students did not realize whether or not they were confused.

Interaction Among Students

The importance of encouraging student interaction is vital to student success in the course, as demonstrated by Dixson [14], who also pointed out the need for instructors to require student interaction as part of the course requirements [15].

To aid in student perception of being part of a class, in this case study they were required to post to a discussion board as part of their course grade and were presented with a rubric that would be used for grading their discussion board activities. In most instances, the student would post an original post and be required to comment on posts by at least two other classmates. This gave students the opportunity to interact with their classmates both socially and academically.

Many students, however, either posted at the last minute or fail to post at all, seriously affecting the level of interaction in the course. One instructor noted that this issue was the most serious with students who did not speak English as their native language. Much like in a traditional classroom, it is difficult to enforce class participation.

However, it should be that noted that many faculty members should possibly follow the lead of Hrastinski, who points out that online learner participation is not "synonymous with talking or writing", such as discussion board, email, blogs, or wikis [15]. In Hrastinski's literature review, online learner participation was defined as "*. . . a process of learning by taking part and maintaining relations with others. It is a complex process comprising doing, communicating, thinking, feeling and belonging, which occurs both online and offline*." [14].

Alternative means of active learning and class participation include group papers and/or projects, chat rooms, and peer review of assignments [15], or audio recordings submitted by students that present reviews of papers or summaries of topics [16]. Other researchers have suggested the use of blogs, *Facebook*, *Twitter* to increase the online social presence of both the students and the instructor [17].

Student Preparation

Another challenge faced in all courses was a lack of student preparation with regard to advanced mathematics. In such cases, the instructor would post supplemental videos or web links covering the necessary background material. Those without the background had to take more time to teach themselves the concepts/material to be able to apply it to the subject material in the course.

Similar issues arose with the use of software such as *Matlab* or *Mathematica*. Again, supplemental material was posted to help students get up to speed on the additional skills they needed. However, this did produce a heavier workload for ill-prepared students.

One similarity between online courses and face-to-face courses was the lack of student initiative in reading assigned material, which is another form of student preparation. This became most obvious in courses where, as part of the module instructions, students were required to indicate whether or not they had read the textbook material. Their responses indicated that they did so, but their class performance indicated the opposite. As with the traditional classroom setting, there is no way to guarantee that a student will do their part in reading the textbook, viewing course materials, and exploring supplementary materials.

Exams

The classic approach to administering exams in distance education is the use of proctors, which pre-dates the availability of taking exams online. For proctored exams, there is a limited time window for the student to arrange to take the exam and a time limit for the exam itself. The proctor serves to prevent academic dishonesty by taking responsibility for administering the exam, making sure that the exam instructions are followed, verifying the identity of who is taking the exam, enforcing a time limit, returning the exam to the faculty member, and verifying that no unauthorized materials were used during the exam. This allows the student to take the same type of exam as on-campus students would take.

Proctors can be testing centers at other educational institutions or even staff members at the student's workplace, but it is vital that proctors be approved by the faculty member.

Students in undergraduate core course 1 and graduate core course 3 were required to submit proctor request forms at the beginning of the semester. Such forms document the proctor information, provide contact information for the proctor, and allowing the instructor to verify that the proctor selected is appropriate. Many students did not submit their proctor forms in a timely manner. Issues arose in undergraduate core course 1 with regard to requests for an inappropriate proctor (such as a family member) and failure to verify that the proctor was available to give the exams on the during the allowed time window.

Fully online exams are an attractive alternative for busy faculty teaching distance education courses. However, documented issues for online exams general courses include access issues involving forgotten or confused passwords, incorrect user names, internet connection problems, power failures, and students contesting the results of the exam [15]. The instructors involved in this case study experienced many of these same issues with fully online exams.

In core undergraduate course 1, several on-campus students were taking the course online. Non-graded conceptual quizzes for self-assessment were made available online, and the first two exams were also online. For the online exams, the students uploaded their answers into *Blackboard*. The problems assigned were traditional fill in the solution type typical of engineering coursework. The exams given online were delivered via *Blackboard*. The students were allowed to log into the system at any time during a 24 h period and had 1 h to complete the exam once they logged in.

The students were caught cheating and using the solution manual when taking the fully online exam. To prevent this from happening again, the exams in that class from that point on were proctored exams, with the on-campus students proctored by the instructor.

For graduate core course 3, fully online exams with a 1-h time limit were posted and the students had 24 h in which to submit them through Blackboard. No students were caught cheating.

In core graduate course 3, a take-home exam was accompanied by a fully online conceptual exam. A time limit of 30 min was set for the online exam, and to ensure that the students could not access online resources during the exam they were required to download *Respondus Lockdown Browser* [18]. The instructor intended for the time limit to discourage the students from accessing offline or online resources.

One student was unable to download the software because of company policy. One student expressed worry that he would not be able to complete the timed portion of the exam, while another student expressly stated that he ran out of time to complete the exam. From a faculty perspective, there was a learning curve involved in creating the online exam and setting it up correctly for use with *Respondus*.

An alternate approach to testing is take-home exams, where the students are given access to any type of resource except their classmates. To develop an exam that is sufficiently complex to measure student performance in a "free for all" atmosphere is time consuming and challenging. It eliminates the need for a proctor, but does not fully prevent academic dishonesty. These exams can also result in excessive grading time, due to the complexity of the problems assigned. This was the experience of the faculty member who used such exams in graduate core courses 1 and 2. Collusion was not detected, but a later review of the exam materials revealed use of online sources without properly referencing them.

Conclusions

Based on the combined experience of teaching a total of six online courses since Fall of 2012, the faculty involved in this case study make the following recommendations:

- It is important to recognize that preparing an online course may take a greater initial investment of time than teaching a face-to-face course. The planning that goes into preparing the course saves time in the long run, and reduces the level of stress experienced by faculty. This involves thinking the course through from beginning to end with an emphasis on breaking the material into small "chunks" rather than 50 min lectures [5].

- Be open to the idea that effectively teaching the course material may require multiple methods of presentation. This can include lecture videos, demonstrations, how-to videos, links to *YouTube* videos, or links to sites with additional information on a topic. An online course is not just recording a traditional lecture and posting the video online.
- For students without sufficient preparation for the course, make supplemental material available or provide the students with links to reliable sources. This will reduce the frustration of the students, and the faculty responsible for teaching the course. In addition, the students will benefit far more from the course.
- Faculty are responsible for being proactive in communicating with the students [13]. To encourage questions about the material, and aid students in determining when they should ask for help, provide means of self-assessment that has no grade associated with it.
- Based on the experiences in this case study, proctoring of exams seems to be the best way to minimize the chances of student dishonesty and the most efficient means of testing. Trenholm suggests proctoring of exams by individuals or test centers for fact- and math-based courses, such as those in this case study [14].

Teaching an online course can be daunting at first, but planning, common sense, and continued research into best practices as software tools evolve can make it an exciting experience for both faculty and students.

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Finding Specifications of While Statements Using Patterns

Aditi Barua and Yoonsik Cheon

Abstract

A formal correctness proof of code containing loops such as while statements typically uses the technique of proof-by-induction, and often the most difficult part of carrying out an inductive proof is formulating a correct induction hypothesis, a specification for a loop statement. An incorrect induction hypothesis will surely lead to a proof failure. In this paper we propose a systematic way for identifying specifications of while statements. The key idea of our approach is to categorize and document common patterns of while statements along with their specifications. This is based on our observation that similarly-structured while statements frequently have similarly-structured specifications. Thus, a catalog of code and specification patterns can be used as a good reference for finding and formulating a specification of a while statement. We explain our approach using functional program verification in which a program is viewed as a mathematical function from one program state to another, and a correctness proof is done by comparing two functions, the implemented and the specified. However, we believe our approach is applicable to other verification techniques such as Hoare logic using pre- and post-conditions.

Keywords

Code and specification patterns • Functional program verification • Intended function • Specification • While statement

Introduction

Functional program verification is a formal verification technique originated from the *Cleanroom Software Engineering* [1], in which a program is viewed as a mathematical function from one program state to another. In function program verification, a correctness proof is done by comparing the function implemented by a program called a *code function* with its specification called an *intended function* [2, 3]. For this, each section of code is annotated with its intended function. If a section of code consists of only simple statements and control structures such as assignments, sequences and branches, its code function can be calculated directly and compared with its intended function. However, if it contains loops such as while statements, it is mostly impossible to calculate its code function directly, thus its proof is done by using the technique of proof-by-induction. Applying the inductive proof rule of while statements is in most case rather straightforward, but finding a correct induction hypothesis, the intended function of a while statement, is not and often is the most difficult part of the proof. And there is no systematic way of formulating a good intended function for a while statement, and programmers mostly relies on their intuitions, insights, or previous experiences to formulate one. Nevertheless, it is vital to formulate a correct intended function for a while statement, as an incorrect induction hypothesis will definitely fails an inductive proof.

One possible way to help the programmers find correct intended functions for while statements is to provide them

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_75, © Springer International Publishing Switzerland 2015

with a catalog of sample while statements along with their intended functions. The samples in the catalog provide patterns of while statements and their intended functions that can be matched to one's own code and thus can be instantiated to derive one's own intended functions. If a while statement matches a code pattern in the catalog, its intended function will have a similar structure as that of the matched code in the catalog. That is, similarly-structured while statements have similarly-structured intended functions.

In this paper we describe our approach for identifying such patterns of while statements based on loop conditions and loop bodies, documenting them in a pattern catalog, and applying them to find intended functions of while statements. However, there are conflicting requirements for being a good pattern. A pattern should be as general as possible to be widely applicable and usable, but at the same time it should be as specific as possible to be meaningful in deriving an accurate intended function. We explain how we address these conflicting requirements. Like software design patterns that describe reusable design solutions to recurring problems in software design [4], our specification patterns also provide other benefits by allowing one (a) to capture and document program specification knowledge, (b) to support reuse in specification and boost one's confidence during program verification, and (c) to provide a vocabulary for communicating one's specifications and proofs. We explain our approach using functional program verification. However, we believe that our approach is equally applicable to other verification techniques such as Hoare-style axiomatic verification using pre- and post-conditions.

This paper is structured as follows. Section "Functional Program Verification" provides a brief overview of functional program verification including the notation for writing intended functions. Section "Intended functions of While Statements" describes the problem of finding and formulating intended functions of while statements. Section "Patterns of While Statements" explains our approach for documenting and cataloging patterns of while statements along with their intended functions, and Sect. "Pattern Applications" illustrates some of our patterns by applying them to examples and provides a preliminary evaluation of our approach. Lastly Section "Conclusion" concludes this paper with a concluding remark.

Functional Program Verification

Functional program verification is a program verification technique originated from the *Cleanroom Software Engineering* [1]. The main idea behind functional program verification is to view and model a program as a mathematical function that maps one program state, an *initial state*, to another, a *final state*. The specification of a program called an *intended function* defines this mapping of states by describing the expected final state in terms of the initial state. Program verification is done by comparing the intended function of a program with its *code function*, the actual function implemented by the program. For this, each section of a program is documented with its intended function (see Fig. 1).

An intended function is written using a *concurrent assignment* notation of the form $[x_1, x_2 \ldots, x_n := e_1, e_2, \ldots, e_n]$ stating that each x_i's new value in the final state is e_i evaluated concurrently in the initial state [2]. For example, the intended function f_1 in line 1 describes the behavior of whole code and asserts that the final value of r is the number of positive values contained in the array a. On the other hand, the intended functions f_2 and f_3 in lines 2 and 6 specify the sections of code in lines 3–4 and 7–15, respectively. In f_3, the keyword *anything* indicates that one doesn't care about the final value of the loop variable i. In this paper we write intended functions semi-formally by using the Java expression syntax and mathematical symbols such as $\sum$. There is also a formal notation for writing intended functions [4].

Once each section of code is annotated with its intended function, its correctness can be proved by comparing its code function with its intended function. This proof can be performed in a modular way by using the intended functions of lower level code in the proof of higher level code. For example, in order to prove the correctness of the code shown in Fig. 1, we need to prove (a) the functional composition of f_2 and f_3 is correct with respect to f_1 and (b) both f_2 and f_3 are correctly implemented or refined by their code. If a section of code consists of only assignments, sequences, and branches, its correctness proof is often straightforward, as its code function can be directly calculated. For example, the code function for lines 3–4 is the same as its intended function, f_2. However, the proof of a loop such as a while statement is a bit involved, as there is no direct way of calculating its code function. It is done by using proof-by-induction [2].

```
1:  // f1: [r : = ∑_{i=o…a.length–1}(a[i] > 0 ? 1 : 0)]
2:    // f2: [r, i := 0, 0]
3:      r = 0;
4:      int i = 0;
5:
6:    // f3:  [r, i := r + ∑_{j=i…a.length-1}(a[j] > 0 ? 1: 0), anything]
7:      while (i < a.length) {
8:        // f4: [r ,i :=  a[i] > 0 ? r + 1 : r,  i + 1]
9:          // [r := a[i] > 0 ? r + 1: r ]
10:           if (a[i] > k)
11:            // [r := r + 1]
12:               r++;
13:         // [i := i + 1]
14:           i++;
15:     }
```

Fig. 1 Code annotated with intended functions

For example, the correctness of code in lines 7–15 with respect to its intended function f_3 requires three sub-proofs: (a) termination of the loop, (b) a basis step of proving that when the loop condition doesn't hold an identity function (i.e., no state change) is correct with respect to f_3, and (c) an induction step of proving that when the loop condition holds the composition of f_4 and f_3 is correct with respect to f_3. The basis and induction steps are for when the loop makes no iteration and one or more iterations, respectively.

Intended Functions of While Statements

In order to apply functional programming verification effectively, it is important to formulate a correct intended function for the section of code to be verified. If the intended function is incorrectly formulated, the proof will fail even if the code is indeed correct. This is particularly true for the verification of loops such as while statements, as their proofs are done inductively and their intended functions become induction hypotheses (see Sect. "Functional Program Verification"). With a wrong induction hypothesis, an inductive proof will fail.

However, formulating and defining a good intended function for a while statement is not easy. It is often the hardest part of formal program verification, and there is no systematic way of doing it. One difficulty is that a loop typically computes a more general function than the one needed. A loop is seldom used by itself in isolation but is preceded by an initialization, which together with the loop computes something useful. For example, the while statement in lines 7–15 of Fig. 1 doesn't calculate the number of positive values contained in the array *a*, but when the loop variable *i* is set to 0 it does. A loop in isolation doesn't do a computation but completes it; an initialization (e.g., setting *i*) determines where the computation starts. An intended function of a while statement should be written in such a way that it captures the completion of a computation regardless of where the computation starts. It should be a correct generalization of the intended function for the code containing both the initialization and the loop, and at the same time it should be specific enough to capture the accurate result of the computation.

Formulating an intended function for a while statement requires a programmer's insight, practice, and experience [2]. The problem of finding an intended function for a while statement is similar to that of finding a loop invariant in Hoare logic. A loop invariant should be general enough to hold on each iteration of the loop and specific enough to lead to a post-condition when the loop terminates. Even if there is no known work done on systematically finding intended functions for loops, many researchers have studied this similar problem of finding loop invariants and proposed various static and dynamic techniques based on pre-conditions, post-conditions, loop executions, and theorem proving (cf., [5–7]).

Patterns of While Statements

One way to figure out a correct intended function of a while statement is to look at other while-loops that have similar code structures. If two while loops have similar code structures, their intended functions are likely to have similar structures too [2]. Therefore, if we know the intended function of one, we may be able to derive that of the other from the known one. For this, we can develop patterns of while loops along with their intended functions based on the code structures of while loops including loop conditions and loop bodies, and these patterns can be used as a reference for formulating an intended function for a while loop (see Fig. 2). For this pattern-based approach to work effectively, we need to identify and accumulate a large number of patterns to cover a wide range of while loops appearing in application code. And each pattern should be as general as possible to be widely applicable to loops written in many different ways. At the same time it should be as specific as possible to derive an accurate intended function when applied to a particular loop. In any pattern-based approach, properly documenting patterns is crucial. Each pattern should be documented in such a way that it is easy to determine its applicability, to instantiate it for a particular application, and to derive an actual intended function from it. Patterns need to be classified and organized to be presented in a *pattern catalog* that can be easily looked up and matched for by programmers. Below we explain how we address these requirements for our patterns.

Pattern Documentation

We document our patterns using a format similar to that of software design patterns [4]. Each pattern has a name, purpose, description, structure, applicability, variations, related patterns, and examples. Figure 3 shows one of the simplest patterns that we identified and documented in this format. A pattern has a *name* to uniquely identify it. Then, its

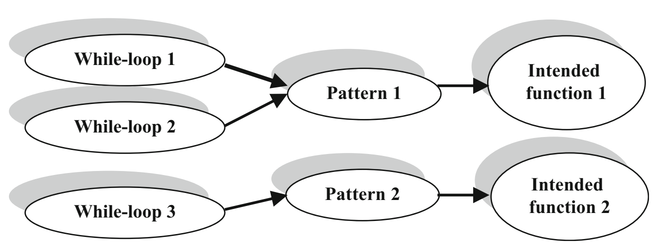

Fig. 2 Pattern-based identification of intended functions

Name: Indexed Accumulating
Purpose: Accumulate elements of a sequence
Description: A loop combines the values of a sequence to a single value by using various accumulation operations such as addition, multiplication, and concatenation. An index is used to iterate over the elements of a sequence. The result is of the same type as that of the elements of the sequence.
Structure:

```
[r, i := r ⊗ Σ⊗j=i..N s@j, anything]
while (B(i)) {
   [r, i := r ⊗ s@i, E(i)]
}
where
 s: sequence whose elements are accumulated
 r: result variable accumulating elements of s
 i: index and loop variable
 s@i: i-th element of s
 ⊗: accumulation operator such as +, -, and *
 B(i): Boolean expression with a variable i
 E(i): expression with a variable i
 N: last i prior to loop termination such that B(i)
```

Applicability: arrays, strings, indexable collections, etc.
Variations: Conditional Accumulating, ...
Related patterns: Iterated Accumulating, …
Examples:

```
[r, i := r + Σj=i..a.length-1 a[i], anything]
 while (i < a.length) {
   [r, i := r + a[i], i + 1]
    r = r + a[i];
    i++;
 }
```

Fig. 3 Indexed accumulating

purpose is stated briefly. The *description* section explains the pattern and is followed by the *structure* of the code along with its intended function. The structure is given as a skeletal annotated code; as shown, the body of a while loop can be abstracted to an intended function to be applicable to a wide range of implementation variations. The *applicability* section lists different contexts in which the pattern can be applied. A pattern can have *variations* and *related patterns*. Lastly the *examples* section shows sample loops matching the specified pattern.

The pattern depicted in Fig. 3 is named *Indexed Accumulating*. It describes a while loop that iterates over the elements of a sequence using an index and accumulates them by using a binary accumulation operator such as +, *, and string concatenation. In the loop body abstracted to an intended function, the element of the sequence s at the current index i (i.e., $s@i$) is accumulated to the result variable r using an accumulator operator $\bigoplus$ (i.e., $r := r \bigoplus s@i$), and the index variable i is set to a new value $E(i)$, an expression written in terms of i. The loop iterates as long as the loop condition $B(i)$, a Boolean expression written in term of i, holds. The intended function of this loop states that the final of the result variable r is its initial value accumulated or combined with all the elements of s starting at index i to index N, where N is the value of i just before the loop condition becomes false. This pattern is applicable when the sequence is an array, a string, and an index-based collection like a Java List class. It has several variations including Conditional Accumulating in which an element is accumulated only if it satisfies a certain condition, e.g., being a positive value. If a sequence or collection provides an iterator (cf. Iterator pattern in [4]), its values can be merged or accumulated by using its iterator operations rather than indexing, and the Iterated Accumulating pattern is for such loops.

```
[r, i := r + ∑j=i..N C(s@j) ? 1: 0, anything]
while (B(i)) {
   [r, i := r + (C(s@i) ? 1: 0), E(i)]
}
```

Fig. 4 Indexed conditional counting

```
[r, i :=(∃j=i..N C(s@j)) ? (s@k s.t. k∈i..N ∧C(s@k)): r, anything]
while (B(i, r)) {
   [r, i := C(s@i) ? s@i :r, E(i)]
}
```

Fig. 5 Indexed searching

Sample Patterns

To identify patterns, we studied a wide range of while loops from several different sources including computer programming textbooks, class programming assignments and projects, and well-known open source software. Through this study we were able to identify patterns of recurring while loops and documented them as specification patterns by generalizing their source code structures and formulating their intended functions. Below we describe a few representative patterns that we identified and documented.

One related pattern of the Indexed Accumulating pattern described previously is a pattern named Indexed Conditional Counting (see Fig. 4) As hinted by its name, it represents a loop that counts the number of elements contained in a sequence that meets a certain condition. Its structure is very similar to that of the Indexed Accumulating except that instead of accumulating the elements of a sequence it accumulates 1's for elements that satisfies a certain condition, thus counting the occurrences of elements in a sequence that satisfies the condition. In the pattern, the notation $C(s@i)$ denotes a Boolean expression that checks if the i-th element of the sequence s satisfies a certain condition; if there is no such a condition imposed, the loop calculates the cardinality of the sequence.

Another recurring pattern of while loops is searching for an element in a collection. For example, a while loop may look for any negative value contained in a list. We named this pattern Indexed Searching (see Fig. 5). The intended function states that the final value of the result variable r is the element of the sequence s at index j, denoted by $s@j$, if the element at index j satisfies the searching condition C, i.e., $C(s@j)$; if there is no such a j, it is the initial value of r. Note that the loop condition $B(i, r)$ may refer to the result variable r to allow an early termination of the loop, e.g., as soon as an element is found.

A loop is also frequently used to select or collect elements from a collection. We documented this use of while loop as a family of patterns, and one particular pattern named Iterated Collecting is shown in Fig. 6. This particular pattern is for selecting elements of a collection by accessing them using an iterator and transforming them to construct a new collection. Since there are many different implementations of iterators, we abstract away from implementation details of iterators in our pattern documentation by introducing abstract iterator operations such as *hasNext*, *current*, and *advance*. There are several variations of this pattern, e.g., collecting only those elements that meets a particular selection criterion and selecting elements without transforming them.

All the patterns introduced and described so far are for accessing and manipulating elements of collections such as arrays, strings, sequences, streams, and files. However, a loop doesn't need to access or manipulate a collection, and another typical use of it is to iterate an indefinite number of times. A while statement, for example, can be used to calculate the factorial of a positive number. Figure 7 shows a variation of the Accumulating pattern described earlier along with its instantiation for factorial code. In fact, it is a generalization of the Indexed Accumulating pattern in that each occurrence of reference to the i-th element of the sequence, $s@i$, is abstracted and generalized to an expression $E(i)$. We will discuss more on pattern generalization and specialization in the following subsection.

```
[r, i := r ∪ {e ∈ C_i • E(e)}, anything]
while (B(i.hasNext())) {
   [r, i := r ∪ {E(i.current())}, i.advance()]
}
where
  ∪: collection merge operation such as union, concatenation, etc.
  i: iterator of the collection whose elements are to be collected.
  i.hasNext(): true only if the iterator i has more elements
  i.current(): current element of the iterator i
  i.advance(): move to the next element of the iterator i
  C_i: all elements available from the iterator i
```

Fig. 6 Iterated collecting

```
[r, i := r ⊕ Σ⊕ j=i..N E1(j), anything]
while (B(i)) {
   [r, i := r ⊕ E1(i), E2(i)]
}

// Instantiated with bindings
//   × for ⊕, ∏ for Σ⊕, 1 for N, i > 0 for B(i), i for E1(i),
//   i – 1 for E2(i)
// [r, i := r × i × (i-1) × (i-2) × ... × 1, anything]
while (i > 0) {
   // [ r, i = r × i, i – 1]
    r = r * i;
    i--;
```

Fig. 7 Variation of the accumulating pattern

Pattern Classification and Hierarchy

While analyzing many different while loops, we soon learned that the structure of a pattern is determined by three factors: (a) how the value to be manipulated is obtained, (b) how the value is manipulated, and (c) how the termination of the loop is determined. These three factors are mostly orthogonal, and thus most combinations of them produce new patterns (see Fig. 8). For example values can be retrieved from collections like arrays, strings, streams, and files using indices, iterators, or in ad-hoc fashions, or they can be created on the fly without retrieving stored ones. There are many different manipulations of values possible, e.g., accumulating (conditionally or unconditionally and with or without transformation), searching, counting, selecting, and collecting. The loop conditions may be written in terms of indices, iterators, values being manipulated, and others. Therefore, we can define our patterns compositionally by picking up one particular possibility for each of these three factors. For example, the Indexed Searching pattern is a composition of an index-based acquisition, a search manipulation, and an index-based termination.

As mentioned previously, one key requirement of patterns is to make them as general as possible and at the same time as specific as possible. This is to make patterns as widely applicable as possible and at the same time to derive accurate and detailed intended functions upon their applications. To address this requirement we generalized and specialized patterns to produce a pattern hierarchy. The idea is to have abstract or general patterns to cover a wide range of while loops but with coarse-grained intended

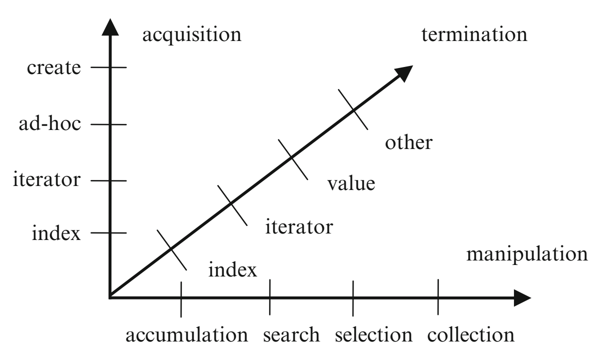

Fig. 8 Orthogonal factors of patterns

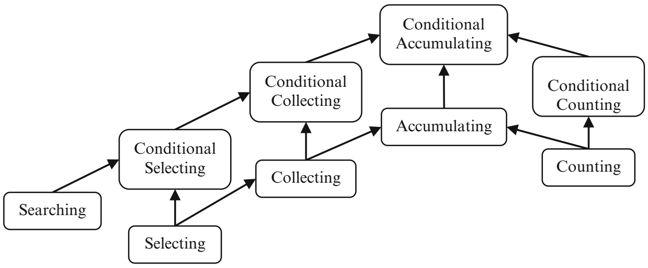

Fig. 9 Pattern hierarchy

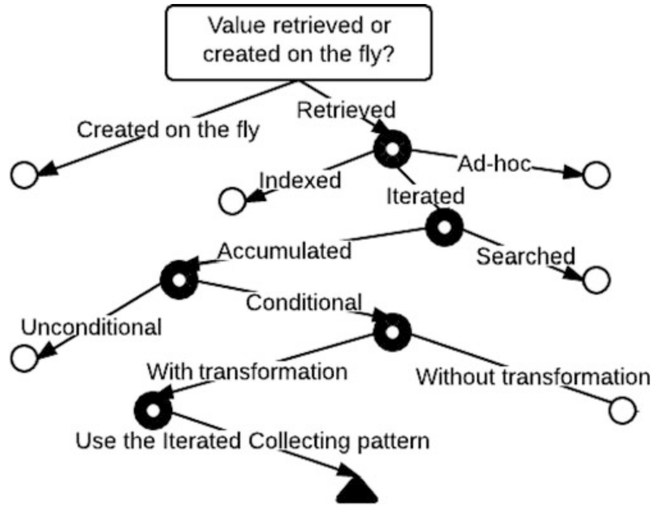

Fig. 10 Using patterns

functions. Concrete or specialized patterns will cover a limited range of while loops but will provide more accurate and detailed intended functions. In the previous subsection, for example, an index-based access of elements was denoted by an expression $s@i$, where s is a sequence and i is an index, and an iterator-based access was denoted by an expression *i.current*(). We can unify these two expressions to come up with a more abstract expression $E(i)$ and use this abstract expression to define a pattern, thus resulting in a more abstract, general, and widely applicable pattern. Applying such a pattern, however, requires more work, as a correct instantiation of an abstract expression like $E(i)$ may not be straightforward for both matching a pattern and deriving an intended function from the matched pattern.

Figure 9 shows a simplified version of a pattern hierarchy focusing on the manipulation of values, not their acquisitions or loop termination. The *Selecting* pattern, for example, is a special *Conditional Selecting* without any imposed condition as well as a special *Collecting* without any transformation of values. As mentioned earlier and shown in the figure, a higher level pattern such as *Conditional Accumulating* is more general and thus has an intended function applicable to a wide range of while loops. A lower level pattern such as *Selecting* is more specific and thus has a more detailed intended function with a narrow scope of applications. A general guideline is to match patterns starting from the root of the tree and move downward to find as specific pattern as possible.

Pattern Applications

We conducted a preliminary experiment to evaluate the effectiveness of our patterns by applying them to industrial strength open-source code. We chose Apache HTTP Server 2.0.65 that has about 486 C files with over 1,500 while loops [8]. We picked up a dozen different while loops from the Apache source code and applied our patterns. For pattern matching we used a decision tree similar to the one shown in Fig. 10. The decision tree allows us not only to perform pattern matching systematically and semi-automatically but also to identify general patterns first and then move toward more specific ones. Below we describe a few simple but interesting while loops from the Apache source code to illustrate applications of our patterns.

The most common use of while statements in the Apache source code is to manipulate pointer-based data structures such as linked lists. Shown below is one such a while loop that is simple but shows an interesting aspect of the application of our patterns. It traverses a linked list pointed to by *f* and changes the *r* field of each node if its current value is equal to *from*.

```
while (f) {
  if (f->r == from)
     f->r = to;
  f = f->next;
}
```

Even if the loop itself is very simple, specifying its behavior is a bit involved because it may mutate not only a single node pointed to by *f* but also potentially all the nodes reachable from *f*. Thus, its intended function needs to capture the side effect caused to the whole list, not just to a single node denoted by the pointer *f*. For this, we introduce a notation to denote the whole list and manipulate a pointer-based list data structure abstractly.

$L_{<f}$: list consisting of all the nodes preceding f; whole list if f is null

$L_{f>}$: list consisting of f and all the trailing nodes; empty if f is null

$\langle\rangle$: list comprehension, e.g., $\langle f \rangle$ for a singleton list consisting of f

+: list concatenation, e.g., $L_{<f} + \langle f \rangle$

$f\{r := e \text{ if } b\}$: node f with its *r* field set to *e* if *b* is true

Using this notation we first calculate and document the intended function of the loop body as follows.

```
while (f) {
  // [L<f, f := L<f + ⟨f{r := to if r == from}⟩, f–>next]
    if (f–>r == from)
        f–>r = to;
    f = f–>next;
}
```

Note that the intended function of the loop body can also be written as [f- > r, f := (f- > r == from) ? to : f- > r, f- > next]. However, this formulation doesn't capture the side effect of the statements in terms of the whole list and thus will make it difficult to prove the correctness of the whole loop. Once the intended function of the loop body is formulated and documented, we can match the annotated code to one of the patterns. Using the decision tree mentioned earlier we can match it to the Iterated Collecting pattern (see Fig. 10). Although this matching doesn't seem possible at first, it should be apparent with the following unification of iterator operations.

Pattern	Code
i.current()	f
i.advance()	f = f- > next
i.hasNext()	!f

Once a matching pattern is found, we can instantiate its intended function to derive the intended function for the code as follows, where the notation $I(x/y)$ denotes replacing every free occurrence of y in the intended function I with an x.

[r, i := r ∪ {e ∈ C$_i$ • E(e)}, anything] (r/L$_{<f}$, i/f, L$_{f>}$/C$_i$, +/∪, ⟨⟩/{}, E(e)/e{r:= to if r == from}) ≡
[L$_{<f}$, f := L$_{<f}$ + ⟨n ∈ L$_{f>}$ • n{r:= to if r == from}⟩, anything]

The derived intended function states that every node reachable from *f* now has a new final value (*to*) for its *r* field if its initial value is *from,* and it matches our informal understanding of the loop.

Another while loop from the Apache source code is shown below. It iterates over the nodes of a list to check if there is a node with a particular name. The code matches our Iterated Searching pattern, and the matching produces the intended function annotated in the code. The Iterated Searching pattern is similar to the Indexed Searching pattern described in the previous section except that elements are accessed using an iterator. As before we need to refer to the whole list, and for this we introduce a special notation L_{filter} to denote the list consisting of the node pointed to by *filter* and all the nodes reachable from it.

```
/* [found, filter := (∃n∈L_filter • f(n)) ? true: found, anything] where
 *   L_filter = list consisting of filter and all nodes reachable from it
 *   f(n) = !strcasecmp(name, n->frec->name) */
while (!found && filter) {
  // [found, filter  := f(filter) ? true : found, filter->next]
    if (!strcasecmp(name, filter->frec->name))
        found = true;
    filter = filter->next;
}
```

In the Apache source code, we also found while loops that refer to arrays. The following is one such a loop, and its loop body is annotated with its intended function written using a conditional concurrent assignment of the form $[B_1 \rightarrow A_1 | \ldots B_n \rightarrow A_n]$ that specifies different functions (A_i's) based on conditions (B_i's) [2].

```
while (name[i] != '\0') {
  /* [C(i) → i := i + 2
   *  | ¬C(i) → w, i, name[w-1] := w + 1, i+ 1, name[i]]
   * where C(i) is the if condition below. */
    if (name[i] == '.'  && IS_SLASH(name[i+1])
      && (i == 0 || IS_SLASH(name[i–1])))
      i += 2;
    else
      name[w++] = name[i++];
}
```

Its intended function is not obvious, but our decision tree and patterns can guide us to it. For example, the values to be manipulated in the loop are *retrieved* from an array using an *index*, selected on a *condition*, and stored *without being transformed*. This leads us to the Indexed Conditional Selecting (or its generalizations) as a possible pattern. The pattern's intended function and our insight on the code lead us to the following intended function, where $s[i..j]$ denotes a substring of s from index i to j, inclusive.

[name, i, w := name[0..w-1] + shifted name[w..], anything,
w + num of shifted chars]

For more rigorous formulation, one need to state precisely what one means by a *shifted name*[*w*..]; informally, it's the suffix of *name* starting from index *w* in which the characters of *name* starting from index *i* that doesn't match the specified patterns (e.g., "./") were shifted over those characters matching the patterns.

Even if our experiment is limited in the number of while loops, the applications and the implementation languages that we considered, it showed a promising result. We often were able to systematically derive very detailed and precise intended functions. Most non-trivial loops, however, required a varying degree of insight and work to come up with accurate intended functions for them. Nevertheless, the pattern decision tree helped us to analyze the code systematically and the matched patterns helped us find correct intended functions for these loops by providing skeletal intended functions. Our patterns also detected certain mistakes that we made when formulating intended functions for both the loop body and the whole loop—e.g., documenting side-effect caused only to a single node, not to the whole data structure. In fact we made such mistakes several times during this experiment, and our patterns exposed and revealed them. We also learned a few shortcomings of our patterns through this experiment. Our patterns, for example, don't capture a certain common use of while loops very well, e.g., mutating the collection data structure that is being iterated over. These loops can be certainly abstracted to selecting or collecting patterns as done in this paper, but more direct and concrete patterns would be preferred and appreciated by the users of our patterns. Some of our patterns may be further specialized to address language-specific features and constructs such as C/C++ pointers and pointer-based data structures.

Conclusion

In this paper we proposed specification patterns to address the problem of formulating specifications of loop statements such as while statements, which is recognized as one of the most difficult part of formal program verification. Our approach was initially inspired by software design patterns [4], however the key difference and thus contribution of our patterns compared with design patterns and other specification patterns (e.g., [9]) is that our specification patterns are *compositional* and *hierarchical*. Each pattern consists of three orthogonal components—value acquisition, value manipulation, and loop condition—and thus is assembled by selecting an appropriate combination of these building blocks. Our patterns are classified into a pattern hierarchy. A generalized pattern is applicable to a wide range of loops, but its specification is more abstract. A specialized pattern, on the other hand, is more specific with limited applicability but provides a more accurate specification. The pattern hierarchy allows one to match patterns starting from more general ones to more detailed. The work reported in this paper is an on-going research, and thus the number of patterns identified and documented is limited. Nevertheless, a preliminary experiment showed a promising result in that the patterns were able to derive specifications for a representative set of while loops found in well-known open source code.

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High-Linearity MEMS Accelerometer with Lateral Comb Finger Groups

Xingguo Xiong and Huanyu Li

Abstract

Most capacitive MEMS (Microelectromechanical Systems) accelerometers sense capacitance gap change due to inertial force. Since capacitance is inversely proportional to capacitance gap, this results in nonlinearity of its output response. In this paper, a high-linearity and high-sensitivity capacitive MEMS accelerometer based on overlap length change instead of capacitance gap change has been reported. Since capacitance is directly proportional to the overlap length, the proposed accelerometer has inherent high-linear relationship between its differential capacitance output and input acceleration. The proposed accelerometer utilizes threefold beam structure, which leads to improved device sensitivity. The working principle of the accelerometer is discussed. Theoretical analysis was performed to guide the device design optimization. The proposed MEMS accelerometer is to be fabricated with bulk-micromachining using silicon DRIE (deep reactive ion etching) and silicon-glass anodic bonding techniques. ANSYS simulation is used to verify the function of the designed MEMS accelerometer.

Keywords

Microelectromechanical systems (MEMS) • Accelerometer • High-linearity • High-sensitivity • ANSYS simulation

Introduction

Microelectromechanical Systems (MEMS) integrates both electrical and mechanical components in the scale of microns (1 μm = 10^{-6} m) into a single chip [1]. The remarkable character of MEMS is its extremely small size. Because of this small size, the manufacturing process can utilize the batch fabrication processes. Therefore, the cost of MEMS devices can be significantly reduced. Furthermore, MEMS devices also have the advantages of low weight, high energy efficiency, high resolution and high functionality, etc. MEMS devices are widely used in automobile industry, light display, optical/RF communication, biomedicine, consumer products and many other applications. Among them, inertial MEMS for navigation system is an important sector. MEMS accelerometers have been successfully used for automobile air-bag deployment system, oil drilling and aerospace/sailing navigation system. Various MEMS accelerometers have been developed. Among them, capacitive MEMS comb accelerometers have been very popular due to their easy operation and compatibility with CMOS technology [2–7]. As an example, ADXL-50 surface-micromachined poly-Si comb accelerometer has been the world's first commercial MEMS device used for air-bag application [3]. Most MEMS capacitive accelerometers utilizes capacitance gap change to sense the acceleration [2–7], which inevitably introduces nonlinearity in device operation. The differential capacitance change of the accelerometer can be treated as approximately linear to the displacement of the movable fingers only if the displacement is much smaller than the static capacitance gap. Furthermore,

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_76, © Springer International Publishing Switzerland 2015

for capacitive MEMS accelerometers with movable fingers moving perpendicular to the fixed fingers, the maximum displacement is limited by the capacitance gap, which cannot be very large. Large displacement of movable fingers may exceed the small-deflection approximation of the device and further worsen the linearity. In order to improve the linearity, several capacitive MEMS accelerometers based on overlap area change have been reported.

In [8], a lateral capacitive MEMS accelerometer based on area-changed capacitance detection scheme is reported. The left and right fixed electrodes are on the bottom of movable electrodes. During the working mode, the movable electrodes move laterally to the bottom fixed electrodes due to inertial force. Hence the capacitance gap is not changed, but the overlap area between movable plates and bottom left/right fixed plates is changed. The device is fabricated with silicon bulk-micromachining. Deep RIE and wafer bonding techniques are utilized in the fabrication. However, since both left and right fixed electrodes are in the bottom of movable electrodes, the electrostatic force acting on the movable plates in capacitance sensing circuitry cannot be balanced. It may attract the movable plates to move down toward the substrate. In [9], a surface-micromachined MEMS comb accelerometer with branched comb fingers and stiffness tuning capability is reported. The stiffness of the sensor structure can be reduced by providing the electrostatic negative stiffness along sensing direction. The comb accelerometer works in force-balanced capacitive sensing mode. Electrostatic force is used to balance the inertial force during acceleration sensing, so that the movable fingers remain unmoved in the working mode. By measuring the required voltage to balance the inertial force, the acceleration can be calculated. Since the device uses poly-silicon surface-micromachining, the poly-silicon is limited to 7 µm and the device capacitance is small.

In this paper, a capacitive MEMS comb accelerometer based on area change instead of gap change is proposed. Since capacitance is directly proportional to the overlap area, the device has inherent high-linearity. The accelerometer utilizes lateral tree-shape comb fingers to sense the acceleration. The movable fingers move in parallel instead of perpendicular to the fixed fingers. This induces overlap length (and area) change instead of the capacitance gap change between movable and fixed fingers. This ensures that the differential capacitance change is always linear to the displacement of movable fingers, hence it is also linear to the acceleration input. Since the movable fingers move laterally to fixed fingers, it can accommodate large displacement without the limitation of capacitance gap. Furthermore, the proposed accelerometer has threefold flexible beams, which leads to improved device sensitivity. The device is based on Silicon-on-Glass (SoG) structure. Glass is used as substrate so that the parasitic capacitance between the silicon device and substrate can be avoided. Single crystal silicon is used as structure material of the device to avoid residual stress of poly-Silicon. Silicon deep reactive ion etching (DRIE) is used to pattern the silicon microstructure so that device thickness can be increased to hundreds of microns. This leads to increased device capacitance for easy signal sensing. The device has 12 comb finger modules (6 modules on the top and 6 modules on the bottom). Each module consists of a stem with 14 movable fingers in the left and 14 movable fingers in the right. There are also 15 left fixed fingers and 15 right fixed fingers interdigitated with the movable fingers. Together the constitute differential capacitance. Unlike the traditional comb accelerometers where the movable fingers are perpendicular to the central mass, the movable fingers in this accelerometer are in parallel to the central movable mass. The device utilizes overlap area change between movable and fixed fingers to sense acceleration, which eliminates the nonlinearity of the traditional comb accelerometers. The central movable mass is suspended by four threefold beams connected to the anchors. The threefold design effectively increases the overall length of the beams. As a result, the beams are more flexible compared to single-fold beam design. Without the need of enlarging the device area, the threefold beam structure greatly enhances the sensitivity of the device. The working principle of the proposed high-linearity MEMS comb accelerometer is analyzed. Based on the analysis, the design parameters of the accelerometer is decided. ANSYS simulation results are used to verify the function of the accelerometer design. The fabrication flow of the MEMS accelerometer is also suggested. The proposed comb accelerometer can be used to sense acceleration along X direction with excellent linearity.

Device Design and Analysis

Differential capacitive sensing has been a popular sensing mechanism for MEMS accelerometers due to its easy implementation and good compatibility with CMOS circuits. Most capacitive MEMS accelerometers utilizes capacitance gap change to sense acceleration. As shown in Fig. 1a, two capacitance plates (with width W and length L) originally has gap d_0 between them. Due to inertial force, the movable plate moves perpendicular to the bottom fixed plate. Assume the movable plate moves up/down by displacement x, the corresponding capacitance change is:

$$\begin{aligned}\Delta C = C' - C_0 &= \frac{\varepsilon WL}{(d_0 \pm x)} - \frac{\varepsilon WL}{d_0} \\ &\approx \frac{\varepsilon WL}{d_0} \cdot \left(1 \mp \frac{x}{d_0}\right) - \frac{\varepsilon WL}{d_0} = \mp C_0 \cdot \left(\frac{x}{d_0}\right)\end{aligned} \tag{1}$$

From Eq. (1), we can see that displacement x is in denominator of capacitance equation. As a result, the capacitance

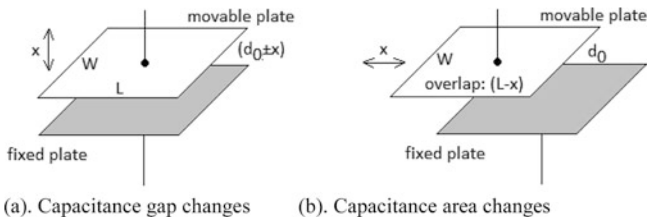


Fig. 1 Comparison of capacitive sensing mechanisms

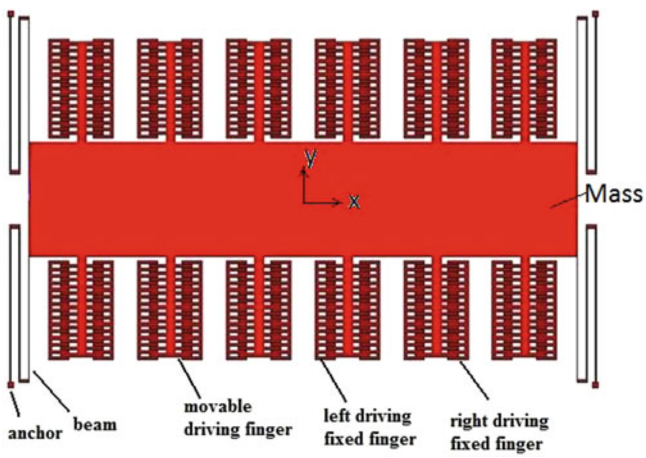


Fig. 2 Structure diagram of high-linearity MEMS comb accelerometer

change is not directly proportional to displacement. This results in inherent nonlinearity of the accelerometer. However, if the movable plate moves laterally with respect to the fixed plate, as shown in Fig. 1b, the corresponding capacitance change becomes

$$\Delta C = C' - C_0 = \frac{\varepsilon W(L-x)}{d_0} - \frac{\varepsilon WL}{d_0} = -\frac{\varepsilon Wx}{d_0} \quad (2)$$

which is directly proportional to the displacement x. This leads to inherent high linearity of the MEMS accelerometer.

The proposed high-linearity MEMS comb accelerometer design is shown in Fig. 2. As seen from the figure, a central seismic mass is suspended by four threefold beams connected in parallel. The other ends of the folded beams are anchored to substrate. There are 12 tree-shape modules of lateral movable comb fingers connected to the seismic mass: six modules are on the top and other six modules are on the bottom. Each tree-shape module consists of a central stem connected with 14 left and 14 right lateral movable fingers. The movable fingers are in parallel instead of perpendicular to the movable mass. There are also 15 left and 15 right fixed comb fingers overlapped with the movable fingers. Together they constitute differential capacitance. All the left comb fingers are connected together, and all the right comb fingers are connected together. Each movable finger is located in the middle between both fixed comb fingers in its top and bottom.

When there is no acceleration input, the overlap length of left and right capacitances are equal. As a result, left capacitance equals to right capacitance. The differential capacitance output $\Delta C = C_1 - C_2 = 0$. However, if there is acceleration input along X direction, the central movable mass experiences inertial force $F_{inertial} = -Ms \cdot a$, hence the beams bend toward -X direction. The movable fingers move together with the seismic mass. As a result, the overlap length between the movable fingers and left/right fixed fingers are also changed: one increases and the other decreases. This results in linear change of the left and right capacitances. By measuring the differential capacitance change, the input acceleration can be derived. This is the working principle of the accelerometer.

In order to increase the device sensitivity, the proposed comb accelerometer utilizes threefold beam structure. Such multi-folded beam design can increase the effective length of the beam without increasing the overall device size. Longer beam width means the beam is more flexible, hence deflects more under same inertial force. This results in larger sensitivity of the accelerometer. MEMS beams can be approximately treated as springs. Once its effective spring constant is calculated, the expected deflection of the beam due to inertial force can be predicted by Hooke's law.

Each folded beam consists of three beam sections connected in series. All the three beam sections have the same width (W_b) and thickness (t_b), but the length of three beam sections (starting from anchor to the mass) are L_{b1}, L_{b2} and L_{b3} separately. The width and length of central seismic mass are W_m and L_m. The width and length of the stem of movable finger module are W_s and L_s. The width and length of each movable finger are W_{mf} and L_{mf}. The width and length of each fixed finger are W_{ff} and L_{ff}. The overlap length between a fixed finger and movable finger is L_{ov}. The capacitance gap between movable and fixed finger is d. The device thickness is t ($t = t_b$). Assume there are N_t modules of tree-shape movable fingers (in our case $N_t = 12$). Each module contains N_{mf} left movable fingers and N_{mf} right movable fingers (in our case $N_{mf} = 14$). The Young's modulus of silicon material is E = 190GPa. The density of silicon material is $\rho = 2.33 \times 10^3$ kg/m^3. The dielectric constant of air is $\varepsilon = 8.85 \times 10^{-12}$ F/m.

Each threefold beam consists of three sections connected in series. Each section of beam can be treated as double-clamped beam model. The spring constant of the first, second and third section (starting from the anchor end) can be calculated as

$$K_{b1} = \frac{12EI_{b1}}{L_{b1}^3} = \frac{Et_bW_b^3}{L_{b1}^3} \quad (3)$$

$$K_{b2} = \frac{12EI_{b2}}{L_{b2}^3} = \frac{Et_bW_b^3}{L_{b2}^3} \quad (4)$$

$$K_{b3} = \frac{12EI_{b3}}{L_{b3}^3} = \frac{Et_bW_b^3}{L_{b3}^3} \quad (5)$$

Three sections of the beams are connected in series. The spring constant of one threefold beam is

$$K_{fb} = \frac{1}{(1/K_{b1}) + (1/K_{b2}) + (1/K_{b3})} \quad (6)$$

Four folded beams are connected in parallel. Thus the total spring constant of the accelerometer is

$$K_{tot} = 4K_{fb} \quad (7)$$

The movable sensing mass M_s include the central seismic mass, N_t stems, and $2N_t \cdot N_{mf}$ movable comb fingers. It can be calculated as

$$M_s = \rho V = \rho\left(W_mL_m + N_tW_sL_s + 2N_tN_{mf}W_{mf}L_{mf}\right) \quad (8)$$

Without acceleration input, left capacitance equals to right capacitance. The static capacitance of the accelerometer is

$$C_1 = C_2 = C_0 = \frac{2N_tN_{mf}tL_{ov}}{d} \quad (9)$$

If there is acceleration a along X direction, the movable mass experiences inertial force

$$\begin{aligned} F_{inertial} &= -M_sa \\ &= -\rho a\left(W_mL_m + N_tW_sL_s + 2N_tN_{mf}W_{mf}L_{mf}\right) \end{aligned} \quad (10)$$

The resulted displacement of movable mass and fingers is

$$\begin{aligned} x &= \frac{F_{inertial}}{K_{tot}} = -\frac{M_sa}{K_{tot}} \\ &= -\frac{\rho a\left(W_mL_m + N_tW_sL_s + 2N_tN_{mf}W_{mf}L_{mf}\right) \cdot \left(2W_{b3}^3L_{b1}^3 + W_{b1}^3L_{b3}^3\right)}{4Et_bW_{b1}^3W_{b3}^3} \end{aligned} \quad (11)$$

The left and right capacitances become

$$C_1' = \frac{2N_tN_{mf}t(L_{ov} + x)}{d} \quad (12)$$

$$C_2' = \frac{2N_tN_{mf}t(L_{ov} - x)}{d} \quad (13)$$

The resulted differential capacitance change is

$$\Delta C = C_1' - C_2' = \frac{4N_tN_{mf}t \cdot x}{d} = \frac{4N_tN_{mf}t \cdot M_s \cdot a}{d \cdot K_{tot}} \quad (14)$$

From the equation we can see that the resulted differential capacitance change ΔC is directly proportional to the input acceleration a. Thus the proposed accelerometer has inherent high-linearity.

The displacement sensitivity of the accelerometer is defined as the displacement of movable fingers in response to 1 g (g is gravity acceleration, 1 g = 9.8 m/s^2) acceleration input along its sensitive direction. Thus the displacement sensitivity of the accelerometer is

$$S_d = -\frac{M_sg}{K_{tot}} \quad (15)$$

The capacitance sensitivity is defined as the differential capacitance change per 1 g acceleration input. For the proposed accelerometer, its capacitance sensitivity is

$$S_c = \frac{4N_tN_{mf}t \cdot M_s \cdot g}{d \cdot K_{tot}} \quad (16)$$

The resonant frequency of the accelerometer can be calculated as

$$f_0 = \frac{1}{2\pi}\sqrt{\frac{K_{tot}}{M_s}} \quad (17)$$

Device Design Optimization

Based on the above analysis, we can decide the design parameters of the accelerometer to meet the given specifications. For device design optimization, it is important to understand the relationship between device performance and various design parameters, so that we know in order to achieve certain performance, what design parameter (s) we should adjust and whether we should increase or decrease them. As shown in previous analysis, the spring constant of beams are directly linked to the device sensitivity and its resonant frequency. To improve the device sensitivity, we need to decrease the spring constant. The spring constant is a function of beam width and length. The relationship between the device spring constant and beam length for different beam width ($W_a < W_b$) is shown in Fig. 3. From Fig. 3, we can see that the spring constant of the accelerometer decreases with the length of folded beam. Furthermore, the spring constant increases with beam width. Decreasing beam width can quickly decrease the spring constant, hence increase the device sensitivity.

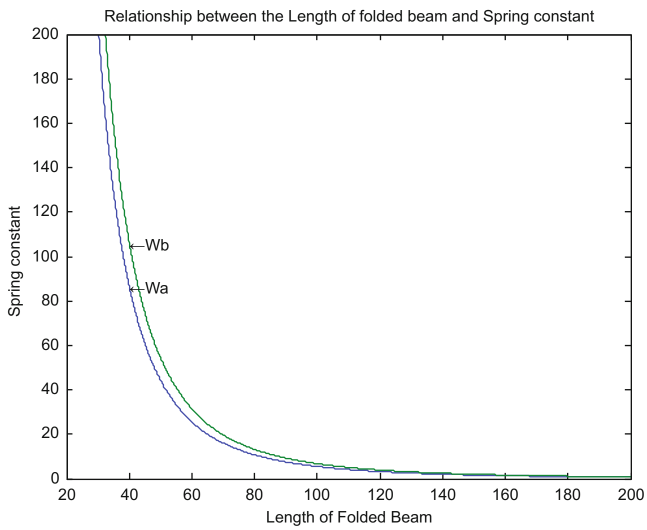


Fig. 3 Relationship between the length of folded beam and Spring constant

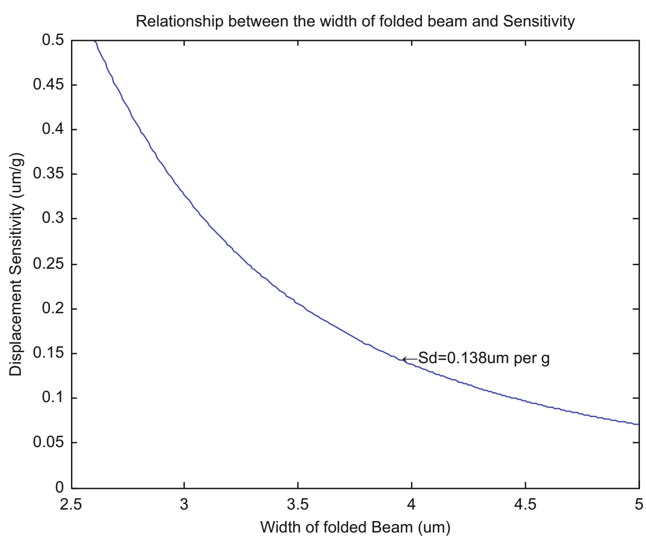


Fig. 4 X-axis displacement sensitivity versus the width of folded beams

The relationship between the device sensitivity and the beam width is further shown in Fig. 4. This can help us in deciding the optimized beam width to achieve certain device sensitity. For example, if we need displacement sensitivity to be $S_d = 0.138$ μm/g, from the curve, we can find that the beam width needs to be set as $W_b = 4$ μm. Unlike beam length, adjusting beam width does not increase the overall device area. Thus it is an effective way to adjust the beam width to achieve the requirement of device sensitivity. However, the beam width cannot be too small (e.g. <1 μm), otherwise the beam may be too fragile and cause difficulty in fabrication.

Based on theoretical analysis, a set of optimized design parameters of the MEMS accelerometer are achieved, as shown in Table 1. The thickness of device is 80 μm and the capacitance gap between two neighboring movable and fixed fingers is 2 μm. The device contains 12 tree-shape modules. Each tree-shape module contains 14 left movable fingers and 14 right movable fingers, as well as 15 left fixed fingers and 15 right fixed fingers. Each movable finger has overlap capacitance with the fixed fingers in the top and bottom. All the left capacitance (capacitance between movable fingers and left fixed fingers) are connected together, and all the right capacitance (capacitance between movable fingers and right fixed fingers) are connected together. Thus the device has two (left and right) total capacitances and they are differential to each other.

Table 1 The optimized design parameters of the MEMS accelerometer

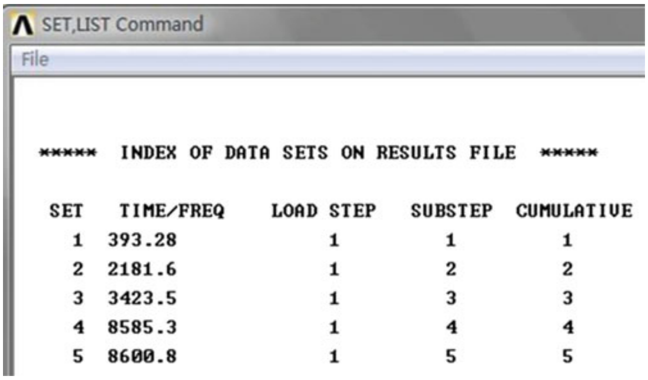
Design parameters	Values
Folded beam width W_b	4 μm
Folded beam length (1st section) L_{b1}	550 μm
Folded beam length (2nd section) L_{b2}	540 μm
Folded beam length (3rd section) L_{b3}	430 μm
Mass width W_m	400 μm
Mass length L_m	1920 μm
Movable finger width W_{mf}	10 μm
Movable finger length L_{mf}	60 μm
Fixed finger width W_{ff}	10 μm
Fixed finger length L_{ff}	60 μm
Static overlap length L_{ov} between fixed/movable fingers	20 μm
Device/beam thickness ($t = t_b$)	80 μm
Number of tree-shape modules N_t	12
Number of left or right movable fingers per module N_{mf}	14
Overall device area (width × length)	800 μm × 2200 μm
Gap between movable and fixed fingers d_0	2 μm

ANSYS Simulation

ANSYS FEM (Finite Element Method) simulation is used to verify the function of the designed high-linearity MEMS accelerometer. Based on the device structure symmetry, 2D model is used for the accelerometer.

Resonant Frequency Simulation

ANSYS vibration mode simulation is used to verify the dynamic performance of the designed MEMS accelerometer. By setting the displacement of anchors to be zero, the first 5 vibration modes of the accelerometer are extracted in ANSYS simulation. The resonant frequencies of the first 5 vibrational modes are shown in Fig. 5.


SET,LIST Command
File

***** INDEX OF DATA SETS ON RESULTS FILE *****

SET	TIME/FREQ	LOAD STEP	SUBSTEP	CUMULATIVE
1	393.28	1	1	1
2	2181.6	1	2	2
3	3423.5	1	3	3
4	8585.3	1	4	4
5	8600.8	1	5	5

Fig. 5 ANSYS simulation of resonant frequencies of first five vibrational modes

To decide which vibrational mode is the working mode of the accelerometer, the deformed shapes of each vibration mode is plotted and animation is used to observe its vibration behavior. The deformed shape plots of the first two vibration modes are shown in Figs. 6 and 7 respectively. Note that in the figures, only the movable parts and the anchors are plotted in the device model. The fixed comb fingers do not contribute to the resonant frequency of the movable parts, thus they are not shown in the simulation.

From the deformed shape plots, we can see that in the first vibration mode, the movable mass and movable fingers move horizontally toward the left and right. The folded beams deflects along horizontal direction. This is exactly the working mode of the accelerometer. Its resonant

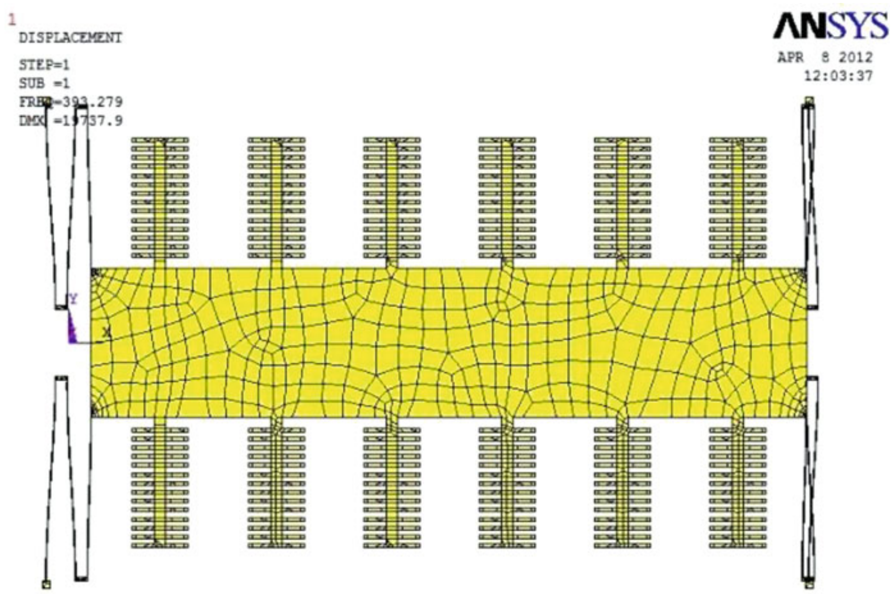


Fig. 6 Deformed shape of the accelerometer in the first vibration mode

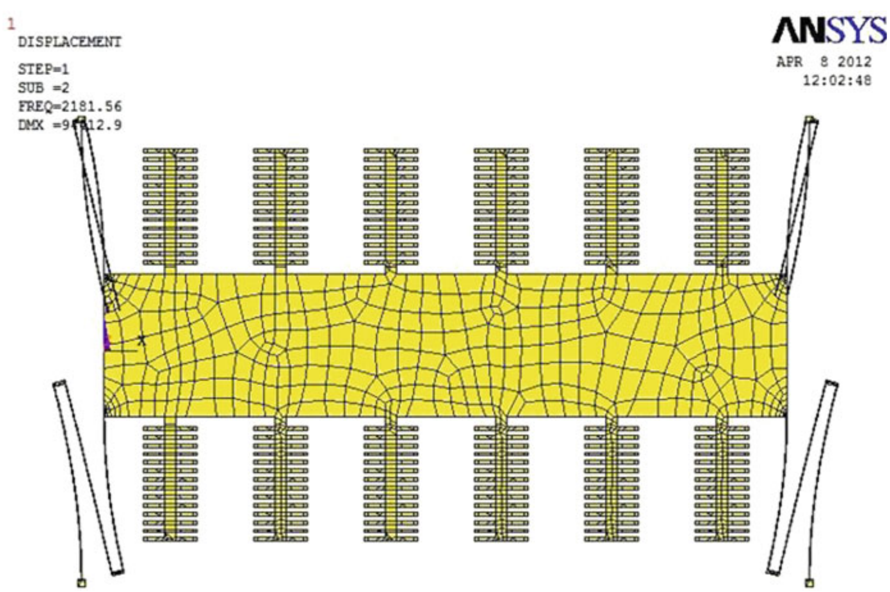


Fig. 7 Deformed shape of the accelerometer in the second vibration mode

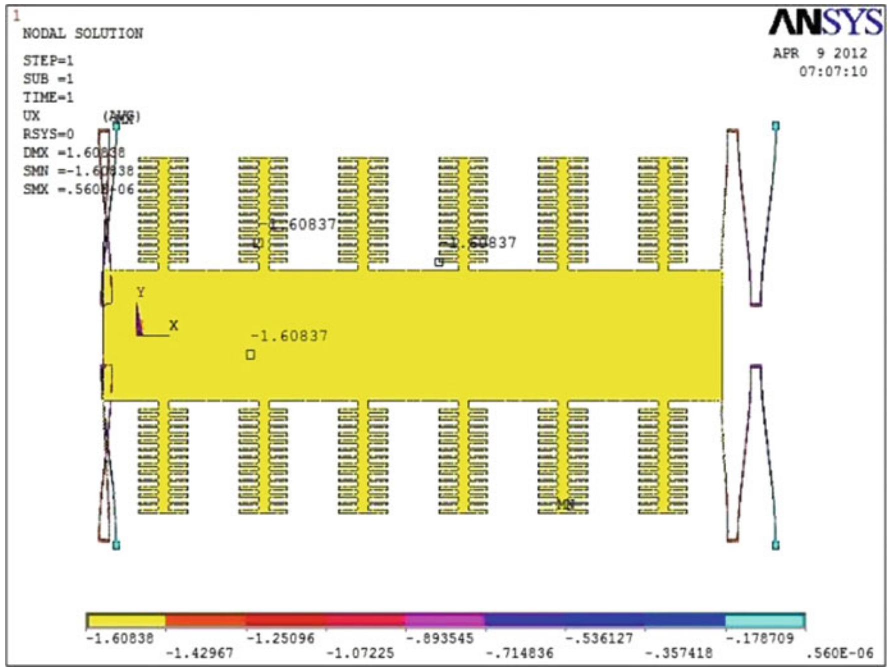


Fig. 8 Sensitivity analysis when experiencing a X direction acceleration of 1 g

frequency is found to be $f_0 = 393.279$ Hz. In the second vibration mode, the folded beams tilt left and right with resonant frequency of $f_0 = 2181.56$ Hz. Apparently this is not the working mode of the accelerometer. The other higher order vibration modes are also verified one by one and they are not the working modes of the accelerometer. Furthermore, from the resonant frequency results in Table 2, we can see that the resonant frequency of the first mode ($f_0 = 393.279$ Hz) is far away from the second vibration mode ($f_0 = 2181.56$ Hz). This verifies that the working mode of the designed accelerometer is stable and higher order vibration modes are not easy to be coupled into the working mode.

Displacement Sensitivity Simulation

Displacement sensitivity of the accelerometer is simulated in ANSYS. Input acceleration (1 g) is applied to the device in + X direction. The resulted displacement along -X direction is plotted in contour plot, as shown in Fig. 8. Different colors are used to represent different values of the displacement. The displacement of certain spots are quoted and marked in the figure. From the figure, we can see that for 1 g acceleration input, the resulted displacement of the movable fingers is $x = 1.60837$ μm. Thus the displacement sensitivity of the accelerometer is

$$S_d = 1.60837 \mu m/g \tag{18}$$

The percentage differential capacitance change with respect to static capacitance C_0 per 1 g acceleration input is

$$\frac{\Delta C}{C_0} = \frac{x}{L_{ov}} = \frac{1.60837 \mu m}{20 \mu m} \times 100\% = 8.04\% \tag{19}$$

This indicates the sensitivity of the accelerometer is very good. Such a percentage change of capacitance is easy for the signal sensing circuit to sense. Furthermore, the differential capacitance change in response to acceleration input a along X direction can also be expressed as

$$\Delta C = C_1' - C_2' = C_0 \cdot \left(\frac{x}{L_{ov}}\right) = C_0 \cdot \left(\frac{M_s \cdot a}{L_{ov} \cdot K_{tot}}\right) \tag{20}$$

It is directly proportional to the displacement x and acceleration input a. This verifies that the accelerometer has inherent linearity due to its sensing mechanism. We should see a linear curve between the differential capacitance output and acceleration input.

Stress Simulation

During the working mode, the beam deflects due to inertial force experienced by the seismic mass. Furthermore, the accelerometer may experience acceleration overload during some unusual conditions (e.g. falling to the ground, shaking environment, etc.). The deformation of the beams inevitably induces stress on the beams. If such stress goes beyond the material strength of silicon, the beams may be broken. Thus it is important to perform ANSYS stress simulation to find out which part of the device experiences maximum

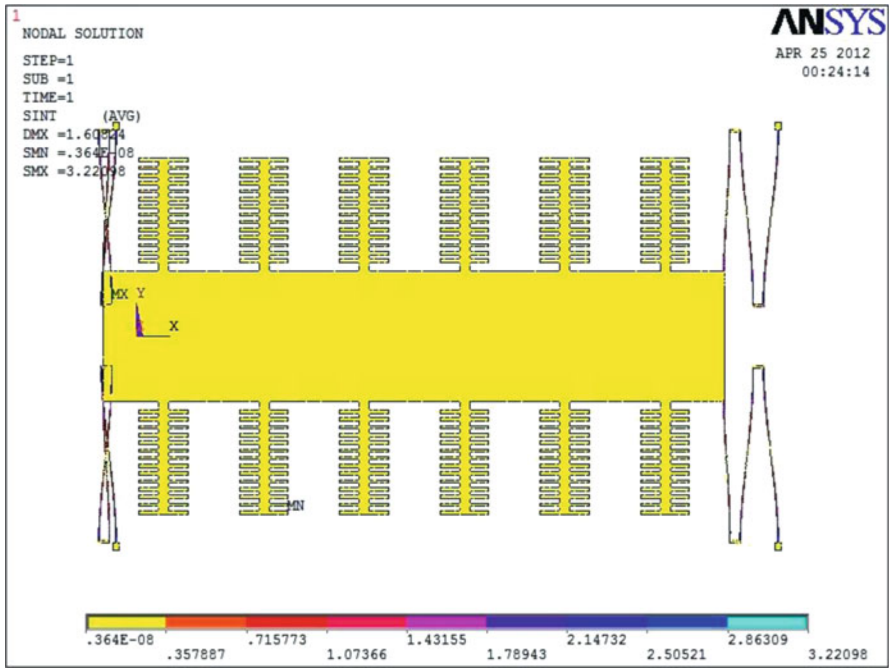


Fig. 9 Stress distribution when experiencing a horizontal acceleration of 1 g

stress during working mode, and whether the maximum stress causes failure of the device. Stress of the accelerometer under the deflection in working mode is simulated in ANSYS. Fig. 9 shows the stress distribution of the accelerometer when experiencing a horizontal acceleration of 1 g. From those two figures, it is clearly shown that the stress is lager at the position where it is close to the anchors or close to the connection of the two beam sections. We further verified that the simulated maximum stress experienced by the beams is well below the material strength of silicon. Thus it is safe for the device operation.

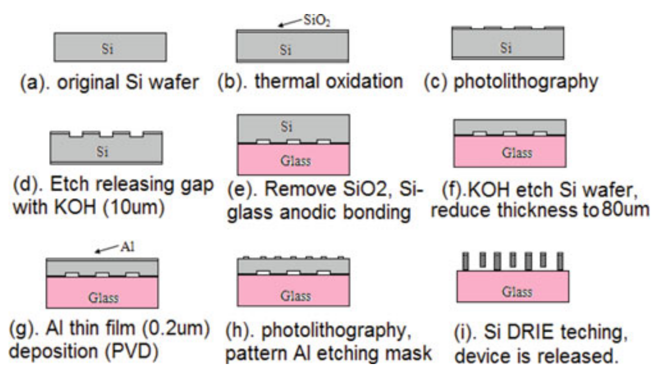


Fig. 10 The fabrication sequence of the accelerometer [10]

Device Fabrication

The proposed bulk-micromachined MEMS accelerometer has silicon-on-glass (SoG) compound structure. The fabrication sequence of the bulk-micromachined comb accelerometer is shown in Fig. 10. The key techniques used are silicon-glass anodic bonding and Deep Reactive Ion Etching (DRIE). By using glass as substrate, the parasitic capacitance between the device and substrate can be eliminated, which eases the signal sensing. Single crystal silicon instead of poly-silicon is used as structure material, so that residual stress inside the material can be avoided. By using silicon DRIE etching, the device thickness can be increased to be more than 100 μm. This further increases the device capacitance, hence make it easy for the signal detection. The bottom of silicon wafer was pre-etched with releasing channels (~10 μm deep) under the corresponding movable parts. No sacrificial layer is needed for the releasing of movable microstructure. This can avoid stiction problem in surface micromachining. After bonding with glass, the silicon wafer needs to be etched down with KOH etchant to reduce the thickness of the wafer. Proper etchant concentration and temperature should be chosen so that the etched surface is smooth enough for the following photolithography steps. The fabrication flow of the accelerometer is similar to the processes introduced by the authors in previous work [10].

Conclusions and Future Work

In this paper, the design and simulation of a high-linearity MEMS accelerometer with lateral comb finger groups is proposed. The accelerometer utilizes capacitance overlap area change instead of gap change to sense the acceleration. As a result, it has inherent linearity in its output response.

The accelerometer uses four threefold beam design to further improve its sensitivity. The working principle of the accelerometer is discussed. Theoretical analysis is performed on the accelerometer to predict its sensitivity and resonant frequency. Based on the analysis, curves are plotted to assist the device design optimization. ANSYS FEM simulation is used to verify the performance of the designed accelerometer. The vibration modes, displacement sensitivity and stress distribution are simulated with ANSYS. The fabrication flow of the accelerometer is also proposed.

In the future work, we will analyze the fringing capacitance effect between the movable and fixed comb fingers, and see how it should be included in the capacitance output of the accelerometer. We will also fabricate the accelerometer using commercial MEMS foundry service, so that we can measure the actual output responses of the fabricated device and compare them with the theoretical prediction and ANSYS simulation results.

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A Quality-Driven Approach for Ranking Web Services

Eyhab Al-Masri

Abstract

The proliferation of Web services over the Web today has led to an increasingly complex and time consuming task for effectively discovery relevant services. One approach to improve the discovery process is considering the behavior of such services under certain conditions or considering their overall quality in accomplishing the expected functionality. In this manner, clients utilize their time more efficiently to find services of interest. In this paper, a quality-driven approach for effectively finding relevant Web services, which we call Quality of Web Service (QWS) is introduced. Results from our proposed solution demonstrate high success rate in discovering Web services of interest and that our quality-driven ranking technique outperforms current state-of-the-art keyword-based methods by at least 36.25 % in terms of precision improvement.

Keywords

Quality of service • QoS • Web services • UDDI extension • Ranking • Ranking web services

Introduction

IN recent years, Web service technologies have demonstrated usefulness and promising capabilities to the deployment of understandable applications used for business-to-business integration. Despite this abundance, a large number of Web services will eventually compete in providing the same functionality.

The popularity of Web services, a key factor in distinguishing services from one another, is significantly achieved by non-functional QWS properties such as availability, usability, performance, integrity, among others. It would be desirable if existing standards applied for publishing, discovering, and using Web services have the ability to incorporate QWS parameters as part of the registration process (i.e. publishing process) while continuously regulating or monitoring revisions to QWS information as a result of any related Web service updates.

To address the above issues, this paper introduces a mechanism that extends our Web Service Broker (WSB) framework [4] by offering QWS support for Web services and uses a combination of service parameters as constraints when searching for relevant or appropriate business applications. Our solution has been tested and results show a high success rate of having the correct or most relevant Web service of interest within top results. Results also demonstrate the effectiveness of using QWS parameters as constraints when performing a search request and as elements when outputting results and that discovery results may significantly be influenced by the choice of QWS parameters used as constraints. Incorporating QWS properties when finding Web services of interest provides adequate information to service requestors about service guarantees and gives them some confidence as to the quality of Web services they are about to invoke.

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_77, © Springer International Publishing Switzerland 2015

The rest of this paper is organized as follows. Section "Related Work" discusses the related work. The proposed solution is discussed in Sect. "Quality of Web Service Manager (QWSMan)". Quality-Driven ranking is discussed in Sect. "QWSMan Computation and Ranking". Experimental results are discussed in Sect. "Experiments and Results", and finally the conclusion is discussed in Sect. "Conclusion".

Related Work

Researchers on separate efforts attempted to add Quality of Service (QoS) support within UDDI registries. The authors in [6,7] proposed QoS support within UDDI registries through tModels. Technical models or tModels are used to describe the technical information about Web services and is part of the current UDDI implementation [16]. To achieve QoS support, Blum used a tModel for QoS information within bindingTemplates in which each QoS metric is represented by keyed Reference, a general-purpose structure for a name-value pair [16].

Ran proposed a system for adding QoS information in UDDI registries using a QoS certification framework in which a QoS Certifier verifies QoS claims provided by a service provider [13]. Although Ran's solution may provide QoS support for Web services' discovery, it has several limitations such as the redundancy of performing QoS measurements which first have to be supplied by the service provider at the time of registration, and then these QoS measurements will eventually be performed by a certification authority.

In [14], the authors proposed QoS support for service-oriented middleware (SoM). In this model, the middleware monitors QoS metrics for Web services automatically, and four QoS properties were identified: time, cost, reliability, and four QoS properties were identified: time, cost, reliability, and fidelity. In a similar effort, the authors in [10] proposed a model for identifying services based on QoS guarantees. However, in both of these proposed solutions, the authors did not take into consideration QoS properties from various dimensions and only focused on a small set of QoS attributes. In addition, the authors do not provide an actual implementation of the proposed systems or how QoS metrics are conducted.

Although there have been numerous efforts that aimed at providing QoS support for Web services [16], very little research has been conducted to explain how the collection of service qualities can be achieved in a transparent and fair manner. In addition, there is a high probability that service providers may perform changes or updates to a Web service and as a result QoS values may also be affected. The proposed solution provides an active monitoring tool that constantly collects the most recent and up-to-date QoS values and ensures that QoS computations are performed in an open and dynamic manner.

A desirable solution is a trusted service broker that continuously monitors and performs QoS metrics for registered Web services and provides a QoS-aware discovery. In this paper, we design and develop a monitoring scheme called the Quality of Web Service Manager (QWSMan) that extends our Web Service Broker (WSB) framework [1–3]. QWSMan continuously monitors the QWS of Web services from accessible Web service resources (i.e. UDDI Business Registries or UBRs, search engines, etc.) and assess their behavior in accomplishing the expected functionality.

Quality of Web Service Manager (QWSMan)

In this section, we present the architecture of the QWS Manager (QWSMan) monitoring scheme and explain the functionality of all of its components. The architecture of QWSMan proposed solution is shown in Fig. 1. QWSMan is part of our larger framework called the Web Service Broker (WSB) [2] which resides on a network of servers.

As shown in Fig. 1, through the WSB framework, service providers begin with publishing service information in the UDDI registry or make them accessible via search engines (Step 1). Once a service is published, the WSB has the capability of discovering newly added or updated services and sends a notification to QWSMan which begins collecting service QWS metrics such as response time, maximum throughput, availability, accessibility, among others (Step 2). After finalizing QWS information, QWSMan provides an access point (or URL) of an XML-based file that contains QWS information about the Web service, and

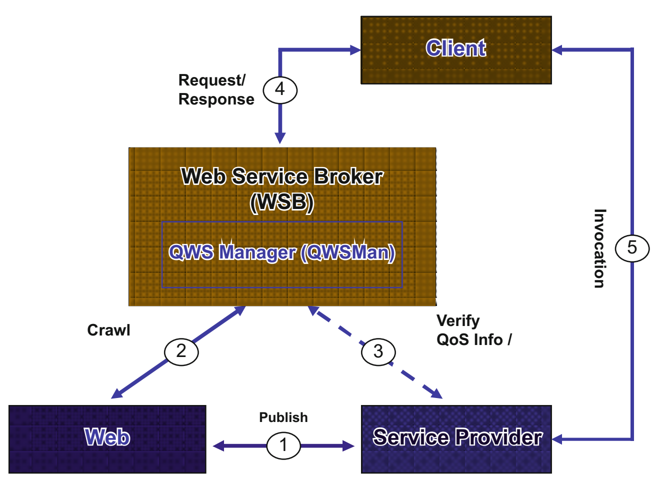


Fig. 1 Proposed QWS Manager (QWSMan) architecture within the WSB framework

issues a notification to the contact information (i.e. email) to the service provider, if necessary (Step 3). QWSMan also provides an interface for clients to discover Web services based on selectively choosing QWS parameters as constraints. In this case, the QWSMan takes client requests and collects the necessary information from within the WSB and prepares a response message back to the client containing a ranked list of pertinent Web services (Step 4). Once clients receive the response message, they can invoke services of interest (Step 5).

QWS-Based Discovery

A service provider may greatly influence how QWS metrics are generated to obtain the most suitable results, and therefore, may publish inaccurate or false information. In addition, service providers have to regularly update QWS information within UDDI registries to maintain accuracy. Automating QWS measurements and storing them in UDDI registries is not currently supported. Furthermore, allowing clients to rate Web services and publicizing this type of information can provide new clients with valuable information on the overall ranking of Web services.

Challenges When Searching for Web Services

Associating QWS information within UBRs is one of the chief challenges for discovering relevant Web services. Furthermore, searching Web services based on QWS metrics is essential since it provides clients with the ability to make accurate decisions. For example, a developer that wishes to choose between various implementations of a Web service may be influenced by information such as performance, reliability, accessibility, to name a few. To elaborate on this problem, consider for example the following statements:

- *Display all Web services offered by business entity A that have response time of greater than 390 ms but less than 500 ms, or*
- *Display all binding templates of Web service A that have high performance, or*
- *Find all binding templates of Web service A that have medium performance and SOAP faults between 10 and 15, or*
- *Were there any binding templates that are offered by business entity A that have more than 50 SOAP faults within the last year?*

Many of the above statements may resemble simplicity when performing lookup or search queries; however, current UDDI standard fails to provide answers to these statements due to many reasons most notably that current UDDIs do not offer complete QoS representation or support for Web services, therefore, searching UDDIs with current APIs is not offer complete QoS representation or support for Web services, therefore, searching UDDIs with current APIs is not sufficient

QWS-Based Discovery

A service provider may greatly influence how QWS metrics are generated to obtain the most suitable results, and therefore, may publish inaccurate or false information. Furthermore, search engines are not well positioned to address service discovery issues since their retrieval algorithms were specifically exploited for the purpose of finding information within Web pages, and not Web services [2].

In order to accomplish the task of offering QWS information as part of the discovery process, a trusted third party service broker is desirable which is capable of collecting QWS information about Web services in an open and transparent manner. The QWSMan serves as a trusted third party service broker that can continuously monitor and collect QWS information about Web services from accessible UBRs and search engines in a timely fashion. Therefore, what is desirable is to have a strutted third party (i.e. service broker) that regulates the support of QWS for Web services by computing QWS information using measurement tools (i.e. WebPartner or ParaSoft SOATest), and generating results in an open, dynamic, and transparent manner and that can be pointed to by service registries by the tModels.

QWS Parameters

Defining QWS parameters for Web services has been a wide area of research and has mainly focused on the nonfunctional requirements such as network research community [5,11,15]. The WSB uses the following QWS properties:

Response Time (QWS_{rt}): Response time is the time taken to send a request and receiving a response. The response time is measured using the following formula:

$$QWS_{rt} = RT_c - RT_r \tag{1}$$

where RT_c represents the time a response is received and RT_r represents the time the actual request has taken place

Maximum Throughput (QWS_{tp}): Maximum throughput is the maximum number of requests that can be handled or processed at a given unit time. The following formula is used:

$$QWS_{tp} = \frac{\mathrm{Re}q_c}{T_s} \tag{2}$$

where Req_c represents the total number of requests completed, and T_s represents the unit time (i.e. 1 min).

Availability (QWS_{av}): Availability is defined as the ratio of the time period when the Web service is available or ready for use. It can be measured as [9]:

$$QWS_{av} = \frac{upTime}{totalTime} \tag{3}$$

where uptime represents the total time the system exists during the time when the measurement is taking place, and the totalTime represents is the addition of the upTime and the downTime. The downTime represents the total time the system has been unavailable during the time when the measurement is taking place.

Reliability (QWS_{rb}): Reliability defines the ability of a Web service to successfully perform the requested functionality, that is, the number of unsuccessful invocations during the period at which the measurement is taking place. It can be measured by:

$$QWS_{rb} = \frac{R_f}{Total_{req}} \tag{4}$$

where R_f represents the number of requests the service was able to perform the requested functionality, and $Total_{req}$ represents the total number of requests.

Best Practices (QWS_{bp}): Best Practices measures the degree to which a Web service is conformant with WS-I profile guidelines [18]. Best practices can be calculated using the following formula:

$$QWS_{bp} = \left(\frac{BP_{pass}}{BP_{total}}\right) \times 100 \tag{5}$$

where BP_{pass} represents the number of tests that a Web service has passed during evaluation, and BP_{total} is the total number of tests conducted for measuring the best practice.

Compliance (QWS_{cp}): Compliance measures whether a Web service is in compliance with a given set of standards. Compliance can be measured using existing standards analysis tools such as MindReef's SOAPTest. It is measured using the following formula:

$$QWS_{cp} = \left(\frac{CP_{pass}}{CP_{total}}\right) \times 100 \tag{6}$$

where CP_{pass} represents the number of tests that a Web service has passed during evaluation, and CP_{total} is the total number of tests conducted for measuring the compliance practice.

Documentation (QWS_{dc}): Documentation defines the extent to which a Web service interface contains an adequate level of documentation. That is, the amount of textual documentation in description tags including service, port, and operations. It is measured using the following formula:

$$QWS_{dc} = \frac{WSDL_{doc}}{WSDL_{total}} \tag{7}$$

where $WSDL_{doc}$ represents the total number of documented tags (i.e. tags containing textual data) and $WSDL_{total}$ represents the total number of tags that can be documented (i.e. can have documentation tags).

QWSMan Computation and Ranking

In order to measure the relevancy of Web services to a given query, QWSMan uses a Web Service Relevancy Function (WsRF). WsRF is used to measure the relevancy ranking of a particular Web service ws_i. Table 1 shows the units used for the QoS parameters used in QWSMan to measure WsRF and by clients as preferences.

Clients via a GUI can submit their requests to QWSMan which will process these requests and computes the WsRF for all available services related to the search query. A Web service with the highest calculated value for WsRF is the most desirable and relevant for the client based on his/her preferences. Assuming that there is a set of Web services that share the same functional attributes and QWS attributes (from Table 1), where, WS ($WS = \{ws_1, ws_2, ws_3, \ldots, ws_i\}$) and P ($P = \{p_1, p_2, p_3, \ldots, p_j\}$), a QWS-based computation algorithm determines which ws_i is relevant based on QWS constraints provided by the client. Using j (same as Reference ID from Table 1) criteria for evaluating a given Web service, we obtain the following matrix WsRF. Each row represents a single Web service ws_i, while each column represents a single QWS parameter.

Table 1 Client preferences used as QWS constrain0ts for WsRF calculation

ID	QWS parameter	Unit
1	Response time (QWS_{rt})	Milliseconds
2	Throughput (QWS_{tp})	# Requests/Min
3	Availability (QWS_{av})	%/3 Days
4	Accessibility (QWS_{ac})	%/3 Days
5	Best practices (QWS_{bp})	%
6	Compliance (QWS_{cp})	%
7	Documentation (QWS_{dc})	%

$$E = \begin{bmatrix} q_{1,1} & q_{1,2} & \cdots & q_{1,j} \\ q_{2,1} & q_{2,2} & \cdots & q_{2,j} \\ . & . & . & . \\ . & . & . & . \\ . & . & . & . \\ q_{i,1} & q_{i,2} & \cdots & q_{i,j} \end{bmatrix} \tag{8}$$

Due to the fact that QWS parameters vary in units and magnitude, the matrix E must be normalized to be able to compute the WsRF and perform QWS-based ranking. Normalization provides a more uniform distribution of QWS measurements that have different units. In order to calculate $WsRF(ws_i)$, we need the maximum normalized value for each P_j column. Let N be an array where $N = \{n_1, n_2, n_3, \ldots, n_d\}$ with $1 \leq d \leq j$ such that:

$$N(j) = \{q(1,j), q(2,j), q(3,j) \ldots q(m,j)\} \tag{9}$$

with $1 \leq m \leq i$ where $q_{m,j}$ represents the actual value from the WsRF matrix in Equation (8). Each element in WsRF matrix is compared against the normalized maximum in its corresponding column based on the following equation:

$$h_{i,j} = \frac{q_{i,j}}{\max(N(j))} \tag{10}$$

where $h_{i,j}$ measures the distance of $q_{i,j}$ from the maximum normalized value in the corresponding QWS property group or j column.

In order to allow for different circumstances, there is an apparent need to weight each factor relative to the importance or magnitude that it endows upon ranking Web services based on QWS parameters. Therefore, we need to define an array that represents the weights contribution for each P_j where $w = \{w_1, w_2, w_3, \ldots, w_j\}$. The values of these weights are fractions in the sense that they range from 0 to 1. In addition, all weights must add up to 1.0. Each weight is proportional to the importance of a QWS parameter to the overall Web service relevancy ranking. The weights are obtained from the client via a client interface. Introducing different weights to Equation (9) results in the following equation:

$$h_{i,j} = w_j \left[\frac{q_{i,j}}{\max(N(j))}\right] \tag{11}$$

Applying Equation (11), we get matrix which is shown below:

$$E' = \begin{bmatrix} h_{1,1} & h_{1,2} & \cdots & h_{1,j} \\ h_{2,1} & h_{2,2} & \cdots & h_{2,j} \\ . & . & . & . \\ . & . & . & . \\ . & . & . & . \\ h_{i,1} & h_{i,2} & \cdots & h_{i,j} \end{bmatrix} \tag{12}$$

Once each Web service QWS value is compared with its corresponding set of other QWS values of the same group, we can calculate WsRF as follows:

$$WsRF(ws_i) = \sum_{j=1}^{N} h_{i,j} \tag{13}$$

where N represents the number of Web services from a given set. To demonstrate how WsRF works, we will consider a simple example in which a client assigns weights to QWS properties listed in Table 1 as follows:

$$w_1 = 0,\ w_2 = 0,\ w_3 = 0,\ w_4 = 0,\ w_5 = 0,\ w_6 = 0,\ and\ w_7 = 1.$$

From the weights assigned, it is clear that the last weight represent the most important QWS property to this client is w_1 or Documentation (from Table 1). The importance level corresponding to each QWS varies since these properties vary in unit.

Experiments and Results

To effectively evaluate our quality-driven discovery and ranking solution, it is essential to have a sufficient amount of real Web service implementations. In that respect, we have taken advantage of the Web services collected using our Web Service Crawler Engine (WSCE) of the WSB framework [2,3] and implemented our proposed QWSMan monitoring scheme to continuously monitor their behavior over a six-day period. The WSCE data consists of 5,077 real Web service implementations that were collected from various Web service resources including UBRs, search engines and service portals.

Rate of Serviceability

Having valid WSDL documents does not necessarily imply that Web services are serviceable. To determine the serviceability of the collected Web services, various techniques must be applied to invoke operations within the defined Web service interface. That is, without determining whether a Web service is functional or not, a client is deemed to invoke a Web service without any knowledge on its serviceability.

The QWSMan monitoring strategy maintains a level of confidence for WSB to provide clients with Web services that are functional. Using QWSMan's monitoring strategy, we determine an intriguing result and that is 49 % of the available Web services are considered serviceable. We can also conclude that using available Web service resources, the success rate for finding a serviceable Web service is extremely low as presented in Table 2.

Table 2 QWSMan serviceability rate summary

	UBRs	Search engines	Total
Serviceable	225	2,282	2,507
Not Serviceable	1,180	1,390	2,570
Total	1,405	3,672	5,077

We also observed the number of errors reported by QWSMan series of tests and based on the reported information from measurement and simulation tools. We found that 56 % of the Web services monitored had less than five errors returned by the various assessment tests including WSDL-related (i.e. best practices and documentation QWS parameters) and execution-related (i.e. response time and availability QWS parameters).

Reported errors exhibit an exponential distribution, as seen in the graphic error in one QWSMan cycle. We also note that although the majority of Web services that have low or no error ratio during our QWSMan assessment cycle, the number of Web services that are not serviceable is high.

Web Service Discovery Query Goals

At the Web level, several research studies have been conducted on identifying the goal of users during a Web search in which the goal is used to improve the page ranking and answer presentation [8]. As part of this endeavor, we conducted a survey to determine the goals of clients when discovering Web services.

Designing a survey is a crucial element in collecting reliable results from our user study. Therefore, we first evaluated the appropriateness to directly determine the degree of complexity for query variations with respect to QWS. For this purpose, we ask participants to become acquainted with the concept of QWS [12], determine the goal of the user and then ask three questions relating to: (1) types of service discovery goals, (2) quality-driven discovery goals, and (3) QWS parameters variations. The survey consisted of 144 valid responses.

Given the results from our survey study, we determine that the goal of Web service clients can be classified into at least two main categories: exploratory, and informational.

For exploratory queries, clients typically presume that any Web service to be the "best" answer and they are willing to discover multiple results. However, using information queries, clients typically presume that not all Web services that match a query term can be considered to be the "best" answer and they are willing to discover only quality results. To that respect, we implemented two ranking algorithms that would accommodate the findings from our survey as presented in Table 3.

Table 3 Summary of the two ranking algorithms

Algorithm	Description
KRank	KRank processes service search queries without taking into consideration any QWS metrics (non-quality-driven). KRank represents exploratory queries.
QRank	QRank processes service search queries with taking into consideration QWS metrics (quality-driven). QRank represents informational queries.

Ranking Evaluation

Service search query terms that we have defined for our evaluation are shown in Table 4. Precision and recall are used in our experiments to determine the usefulness of our ranking mechanism, and how well our system retrieves relevant Web services based on given service search criteria. In addition, we use precision and recall to compare our ranking algorithms including KRank and QRank with other systems such Yahoo and Google search engines. To be able to conduct comparative assessments with other systems, we use TREC methodology for calculating precision and recall at various cut-off levels to compare the performance of our retrieval system [7,17].

Figure 2 presents two recall-precision curves: (1) exploratory (keyword-based) denoted by KRank, and (2) informational (quality-driven) denoted by QRank corresponding to average precision values at each recall level. That is, for each precision value at a particular cut-off recall level, a precision score is computed by summing all precision values of this level for all queries then dividing by the number of queries, which in this case is 10.

As demonstrated by the recall-precision curves in Fig. 3, QRank performs better than KRank. That is, the quality-driven ranking (QRank) algorithm has higher precision values at very low recall levels compared with the keyword-based (KRank) algorithm. QRank performs above average in all cases while KRank performs below average at all cases. Results from Fig. 3 demonstrate that using a quality-driven ranking strategy outperforms regular keyword-based ranking.

The average improvement using QRank over KRank is 24 %. This percentage number represents an increase in the level of precision which indicates that the ranking strategy has improved the performance by approximately 24 % or that using QRank yields more relevant results than that of KRank.

Having average recall-precision curves are very useful for performing comparative analysis of distinct ranking algorithms over a set of queries. Figure 3 shows the average precision versus recall curves for ranking performance for all of the ten queries we have used in our experiments. Figure 3 also presents the precision values at 11 point cut-

Table 4 Ten queries used for the ranking evaluation of KRank and QRank

Query	Service need	Query	Service need
Zip	Finding zip codes information	Currency	Exchange rates for currencies
News	Getting latest world news	Email	Validating email addresses
Map	Locating geographical information on maps	Phone	Lookup phone numbers
Stock	Getting stock information	SMS	Sending SMS messages
Address	Getting address information	Weather	Getting weather information

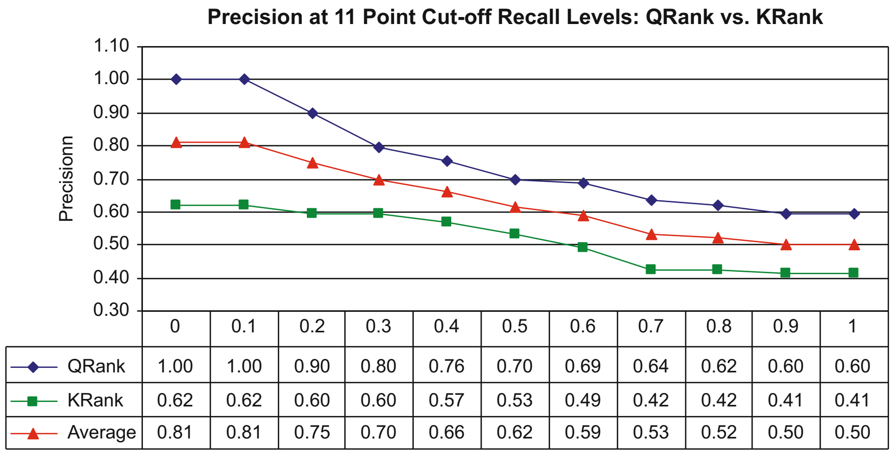


Fig. 2 Average recall-precision curves of all queries for KRank & QRank

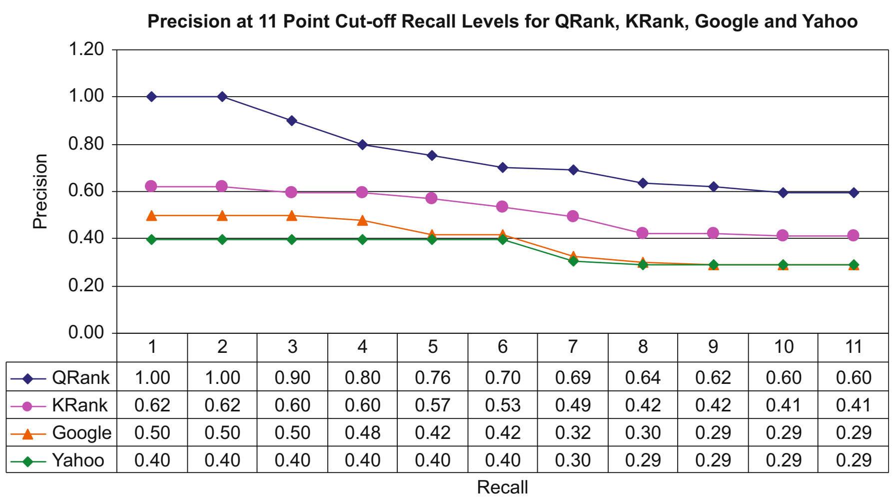


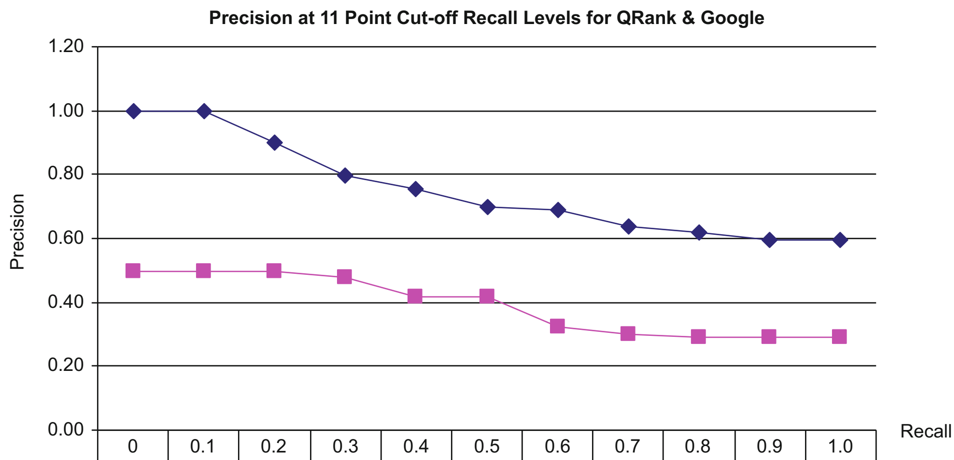
Fig. 3 Average recall-precision curves of QRank, KRank, Google & Yahoo

off recall levels for QRank (quality-driven), KRank (keyword-based), Google and Yahoo search results.

Figure 4 shows the recall-precision curves for all queries and average precision improvements. The average improvement in precision for all queries is 36.25 % which verifies the R-Precision experiment using the precision histogram. This significant improvement in precision demonstrates the usefulness of using quality-driven ranking algorithm versus Web-based ranking algorithms. That is, quality-driven ranking has a significant improvement over search engines such as Google in terms of precision.

Conclusion

Results from our ranking scheme demonstrate usefulness of quality-driven service discovery when compared to regular keyword-based discovery techniques. In fact, our quality-driven ranking performs much higher than current state-of-the-art Web-based search methods such as Google and Yahoo by approximately 36.25 % and 40.39 %, respectively. This high increase in precision improvement reflects the effectiveness of associating



	0	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1.0
QRank	1.00	1.00	0.90	0.80	0.76	0.70	0.69	0.64	0.62	0.60	0.60
Google	0.50	0.50	0.50	0.48	0.42	0.42	0.32	0.30	0.29	0.29	0.29
Improvement	0.50	0.50	0.40	0.32	0.34	0.28	0.37	0.34	0.33	0.31	0.31

Fig. 4 Precision at 11 point cut-off recall levels of all search terms for QRank & Google

quality of Web service into the discovery process. Providing clients with ways to assess the behavior of Web services helps them determine Web services much more efficiently. That is, quality-driven discovery and ranking of Web services becomes a crucial element for differentiating between Web services, and hence outperform keyword-based ranking algorithms and those exploited for finding pertinent information on Web pages.

Acknowledgment We would like to thank Q. H. Mahmoud for his support and assistance. This research was supported in part by the Natural Sciences and Engineering Research Council of Canada (NSERC) Discovery Grant No. 045635.

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RFID-Based Approach for Monitoring Patient's Health Inside Hospitals

Eyhab Al-Masri and Mohamed Hamdi

Abstract
While many technologies have been adopted in hospitals for preventing adverse events that could result in patient's injury or death, detecting the precursors at early stages is yet a common problem and is often neglected due to improper patient monitoring mechanisms. Traditional continuous patient monitoring equipment (i.e. in ICU) lack wearability, portability and cost effectiveness. In this paper we introduce RFIDTrack, a mobile monitoring solution that allows wearable, continuous observation of many patient vital signs to be used inside hospitals. RFIDTrack offers a robust distributed system that is capable of alerting medical staff (i.e. nurses or doctors) in case of early detection of adverse events. The paper presents some experimental validation results, and analysis of the proposed ideas.

Keywords
RFID • Tags • Sensors • Hospitals • Electronic monitoring • Patient monitoring • Administering patients

Introduction

IN recent years, hospitals have become under pressure due to the increasing demand in improving the safety of their patients while minimizing the operational costs as much as possible. Reducing medical errors and instantly responding to adverse emergency situations are some of the contributing factors that have led healthcare organizations attempt to leverage existing technologies to meet these demands.

Patient safety can be jeopardized due to many reasons. For example, it is estimated that between 44,000 to 98,000 people in the USA died as a result of a potentially in-hospital medical errors in 1997 [13,14]. This number is such a large figure in the United States, for example, that attributes to deaths more than other factors such as motor vehicle accidents, breast cancer, AIDS, among others [1].

Furthermore, monitoring patients' vital signs can become costly and time consuming. For example, infants and pediatric patients who undergo heart surgeries must be observed very closely and continuously for couple of days. This requires competent hospital staff to observe patients' vital signs (i.e. temperature, oxygen saturation, heart rate) and document them at least every 5 min in a time-based record [2]. Documentation shall include name, time of administration, and perhaps dosage of all drugs administered. Such documentation is subject to human errors which could potentially lead to adverse outcomes.

There have been numerous technologies that have been adopted in the area of patient monitoring. However, it is believed that Radio Frequency Identification (RFID) is developing at a rapid rate and is believed to be the next generation innovation for automatic data collection, object identification, and asset tracking [3]. Today, many healthcare institutions are becoming increasingly supportive for adopting RFID technology as a major potential

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_78, © Springer International Publishing Switzerland 2015

contributor to patient safety across the healthcare industry. For example, BearingPoint, a system integration firm together with efforts from the National Alliance of Health Information Technology conducted a “RFID in Healthcare Survey.” Seventy percent of the participants cited patient safety as the key benefit for adopting RFID technologies inside hospitals. The survey concluded that RFID technology is widely used in a variety of applications within healthcare organizations, including patient flow management, access control, and security. Furthermore, the survey also concluded that RFID deployment would reach the usage of real-time medical equipment and patient-safety systems such as identification and medical administration [4].

RFID has become widely accepted across healthcare organizations due to its robust application capabilities. That is, RFID-based systems are becoming the natural solution inside hospitals due to its widespread use across many other applications including logistic operations, inventory management, industrial automation, access control, among many others. RFID technology can significantly impact the healthcare industry by improving patient care and safety, optimizing workflows, reducing operational costs, real-time administration of patient vital signs, medical equipment asset management, among many others [5].

RFID is an identification technology that uses radio waves. It consists of multiple parts: (1) RFID identifier (or tag—sometimes called responder), and (2) an RFID reader, and (3) a data system for data processing. RFID tags include an antenna and small memory that is used to store data. Such data is used to identify the tags and content of this tiny memory can be either read or written through an RFID reader. Although this technology is comparable to the bar code technology, however, RFID system reader can read and write data from and to tags without a line of sight. Furthermore, information can be rewritten as many times on many RFID tags. In addition, an RFID reader is capable of reading several tags simultaneously [6].

Related Work

There have been a number of RFID-based solutions applied in hospitals. In [7], the authors introduce a system for improving patient’s safety in healthcare particularly in medication care by using RFID and NFC (Near Field Communication). In [8], the authors proposed a mobile healthcare service system using RFID. The system provides a platform that uses RFID and mobile devices for positioning and identifying persons and objects inside and outside hospitals when disease takes place. The authors in [9] introduced an application of RFID to improve patient safety in observation unit of hospital emergency departments.

In [10], the authors introduced AGALZ (Autonomous aGent for monitoring ALZheimer patients) using RFID. The authors in [11] discuss a system that uses RFID to tracking objects. The authors in [12] examined the social dimensions of RFID system in hospitals in which they conclude that implications of RFID systems, such as privacy concerns [16] and work intensification for nursing and other hospital staff, should be taken into consideration from the outset, particularly during the design and implementation of the technology.

In this paper, we discuss the use of RFID technology for administering, observing patient vital signs continuously while alerting medical staff in case of early detection of precursors of adverse events. Hence, we developed RFIDTrack, a system for accomplishing these tasks in such a way that utilizes latest RFID, Web, and mobile devices technology to create a real-time administration of patients’ throughout hospitals. Whether patients are in ICUs, emergency units, or other areas inside a hospital, RFIDTrack is capable of tracking patients’, monitoring their vital signs (i.e. temperature, heart rate, blood pressure, glucose level, among others) in real-time, and alerting medical practitioners in case of emergency signs exhibited by the data RFIDTrack continuously collects.

This paper is organized as follows: section “Introduction” provides an overview of the importance of patient monitoring and the applicability of RFID to be integrated inside hospitals; section “Related Work” discusses some of the related work in the area of RFID across the healthcare industry; section “RFIDTrack Architecture” introduces our RFIDTrack architecture, design challenges, and the advantages of using RFID technology; and finally section “Conclusion” discusses the conclusion and future work.

RFIDTrack Architecture

Advancements in the area of information technology over the past few years have caused significant changes to the way information is collected, gathered, and disseminated. However, a fundamental question of feasibility remains a challenge. For example, one must determine whether using RFID inside hospitals would be feasible or not. Then, the practicality and cost effectiveness can then be determined.

To determine the feasibility of the RFID system inside hospitals, we need to find the parties involved. Table 1 summarizes the various parties and their tasks.

As shown in Table 1, biomedical engineers need to develop a system based on the specifications and requirements provided by the clinical staff. Hence, we need to identify the requirements of the RFIDTrack system first prior to establishing the design and implementation of the system. Table 2 shows a list of functional requirements

Table 1 User-groups using RFIDtrack

ID	Group	Task
1	Biomedical engineers	Developing physical equipment for the monitoring system
2	Software engineers	Developing the health informatics framework for the monitoring system
3	Clinical staff	Analyze captured data
4	Funding bodies	Provide necessary resources
5	Participants/users	Patients inside hospitals

Table 2 Functional requirements of RFIDTrack

ID	Functional requirement
1	RFIDTrack need to report the basic patient's vital signs including: body temperature, heart rate, blood pressure, and geographic location
2	RFIDTrack need to alert clinical staff in case of emergencies via transmission means such as text messages or phone rings
3	RFIDTrack must be capable of collecting data in real-time iterations from patients' sensors within a reasonable time fame set by clinical staff
4	RFIDTrack must be able to analyze patients' vital signs for a time period and averages out the results
5	RFIDTrack need to capable of being integrated into existing health informatics solutions

that RFIDTrack must implement based on a conducted survey during 2013 [8] that included eight healthcare practitioners in the area of adult and pediatric heart surgery, four consultants specialized in biomedical collection of data, and two executives of industry-leading biomedical equipment suppliers.

Design Challenges

RFIDTrack is a system that allows patients to wear multiple sensors to form a sensor network for capturing different types of information about a patient. The sensors transmit data continuously collected from the patient's body to an RFID reader. The reader transmits this data to a centralized server which captures this data and stores it into a centralized database. The server continuously analyzes the data and determines any precursors of adverse events in which it immediately sends an alert message to clinical staff registered in the RFIDTrack system to receive alerts. Figure 1 shows an overview of the different RFIDTrack components.

In order for the RFIDTrack to be feasible, it is imperative to analyze the data obtained from the various sensors and determine whether the captured data is meaningful. There are several factors that may influence the validity of the data such as securing the sensors at the relevant points in the participant's body, each sensor must be sensitive enough to pick up subtle tilting of the body while at the same time not making too sensitive to pick up any vibrations of movements from other sources. The RFIDTrack sensors also must be suitable for specific applications. For example, the size of RFID sensors that collect temperature must be applicable to both adults as well as infants.

In addition to the requirements presented in Table 2, other important questions may arise and must be also taken into consideration such as how the capture of data will be transmitted from the RFIDTrack reader to the RFIDTrack server, and any sensor interference that could potentially be generated having more than one sensor (e.g. when one RFID sensor comes into close proximity with other sensors). Answer to many of these questions relies on the use of the technology that will be utilized for sending and capturing data. In the next section, we discuss the conceptual point of RFID technology that will have an answer to many of the requirements in Table 2 in addition to other technical questions.

Advantages of Using RFID

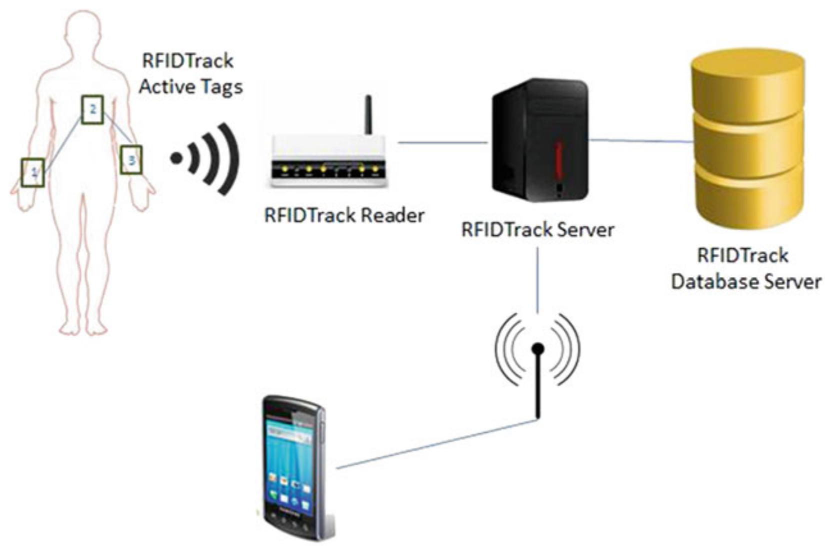
RFID is a technology that, as the name suggests, identities objects using radio frequency signals. It is essentially perceived as an electronic barcode. In general, RFID involves "tags" that identify an object and "readers" that communicate with the tags. There are different types of tags such as active or passive which mean whether the tag requires a built-in battery to be powered on to transmit and receive data. The battery in such tags serves the purpose of reading and transmitting from and to a longer distance. Passive RFID tags do not have internal batteries, and therefore read and transmit data in shorter distance compared to active RFID tags.

RFID systems operate on a number of frequency ranges such as Low Frequency (LF) at 135 kHz, High Frequency (HF at 13 MHz, Ultra High Frequency at 900 MHz, Microwave at 2.4 GHz and up to 10 GHz. RFID is capable of acting more than a simple identifier. For example, active RFID tags have an internal battery which could be used to power biomedical sensors such as body temperature or heartbeat rate. In this manner, active RFID tags can be used as sensors as well identification of patients. RFIDTrack uses active RFID tags which are linked at the time of patient registration to the hospital and are used to relate captured data to patients.

RFIDTrack Architecture

RFIDTrack consists of multiple components: (a) RFID tags (or RFID sensors), (b) RFID Reader(s), (c) RFID Server, and

Fig. 1 High-level architectural view of RFIDTrack system



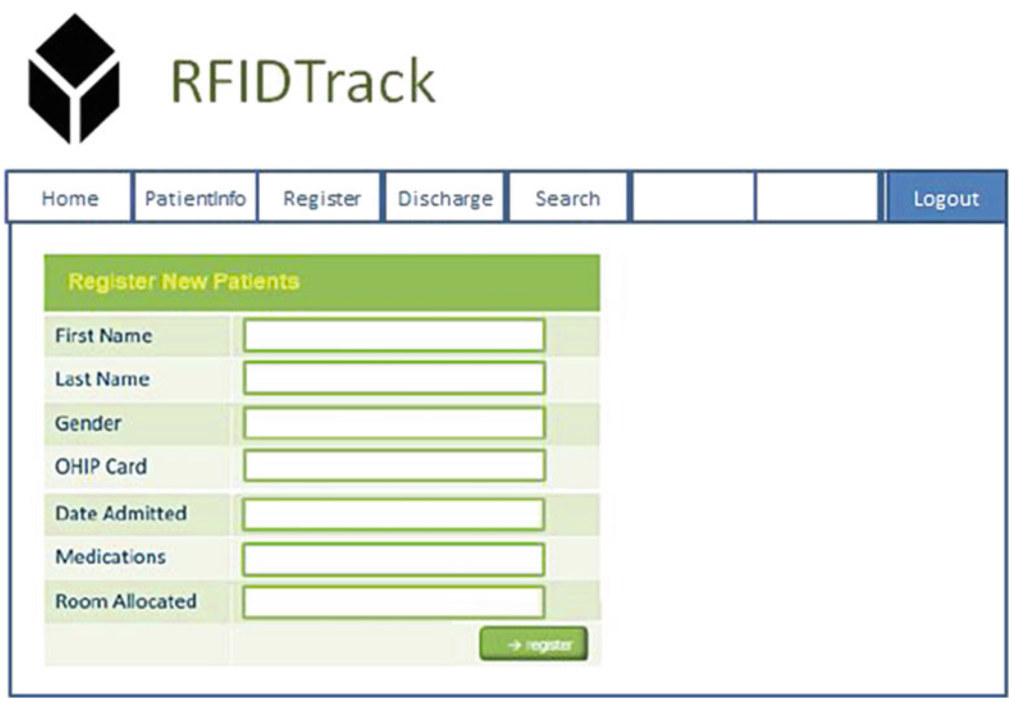


Fig. 2 RFIDTrack interface for registering hospital patients

(d) RFID Database as shown in Fig. 1. The RFIDTrack tags used in this system were engineered in cooperation with Summit Automation [9].

RFID tags can simultaneously transmit data to the reader. The RFID reader then captures this data and passes this information through a physical connection to the RFID Server. The RFID server retrieves the incoming data and identifies the patient to which the data belongs to. Every data packet that arrives at the RFID server contains data on the unique identifier used to identify the RFID tag or sensor. Hence, patients who are registered into the hospital would use RFID database to store which tags have been assigned to them as shown in Fig. 2. Clinical staff is responsible for performing this operation at the time of admitting patients or when RFID tags are required.

Once RFID Server captures the data, it is stored in the RFIDTrack Database. An AJAX-enabled RFIDTrack application is then used to continuously monitor the incoming

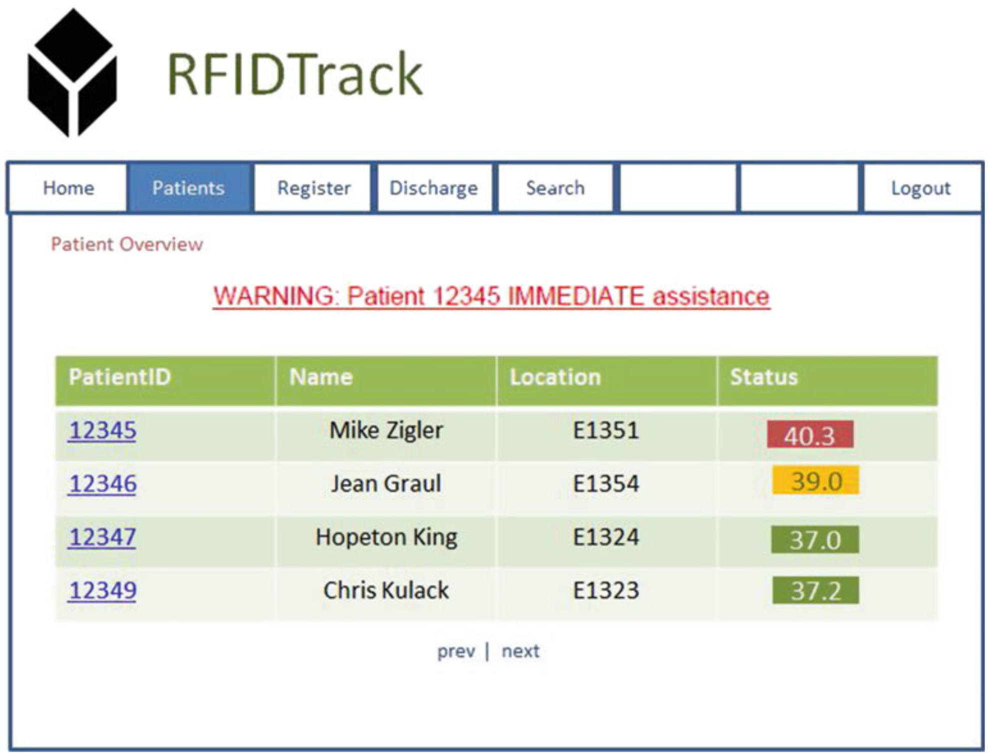


Fig. 3 AJAX-enabled interface of the RFIDTrack frontend

data for each patient. Multiple algorithms have been implemented in RFIDTrack. For example, an algorithm that continuously monitors all registered patients and monitors their temperature in real-time and applies certain techniques to determine whether a patient is showing precursors of adverse events or emergency situations. Figure 3 shows an example of the real-time monitoring of patients using RFIDTrack.

The use of the Asynchronous JavaScript and XML (AJAX) technology enables the real-time processing and display of the data captured from various patients. Since RFIDTrack reader continuously captures data and sends it to the RFIDTrack Server, we can then support the administration of multiple patients at the same time. Currently, RFIDTrack is being tested and can handle up to 200 patients at the same time. Additional RFIDTrack readers may be required to handle additional patients for larger hospitals.

One advantage of using the simple interface such as the one shown in Fig. 3 is that this interface can be integrated to a large graphical display inside emergency units, for example, in which clinical staff would continuously monitor all patients statuses in real-time. This allows more than person to be involved in the monitoring process. The RFIDTrack display interface is set to blink the status of patients considered to be in critical conditions (i.e. Mike Ziegler in Fig. 3) and can make an alerting sound that could be outputted through the graphical display sound port (if it supports audio output).

Furthermore, RFIDTrack allows clinical staff or administrators to register patients and assign particular RFIDTrack tags based on the need for which vital signs need to be monitored. RFIDTrack tags are wearable sensors that can be placed using hand wrists or can be attached to parts of the human body (i.e. abdomen) to detect these vital signs. RFIDTrack tags are equipped with internal batteries that can last for 6–9 months of continuous daily usage. Internal batteries can always be swapped with new ones when necessary.

The RFIDTrack system was implemented using various Web technologies including AJAX, Microsoft Active Server Pages .NET using Visual Basic .NET and runs on a Microsoft Windows 2008 Server. RFIDTrack is designed in such a way that it can be used as a software-as-a-service (SaaS) in which other electronic patient software applications can utilize RFIDTrack software as a service which can be easily integrated into existing systems. This allows RFIDTrack to be extensible to other currently in-use healthcare application with minimal changes or setup required.

RFIDTrack Operating Scenarios

In order to test the applicability and feasibility of the RFIDTrack system, the system has been applied inside a hospital [8] and was applied on pediatric patients. Pediatric patients require special care particularly in situations such as not to wake infants after a heart surgery is performed. The temperatures of their bodies change frequently and without

Fig. 4 Real-time alerting for patients in the RFIDTrack who have shown precursors of adverse events

having careful monitoring or administration, a pediatric patient can easily pass out if no attention is given to their vital signs continuously and frequent periodic monitoring. This would require a large number of dedicated clinical staff which is not cost effective. Using RFIDTrack, it is possible to significantly reduce the number of clinical staff required to frequently check on patients' statuses since this process can be administered automatically and accurately. RFIDTrack results such as those shown in Fig. 3 were sent to multiple displays across the hospital to provide a more real-time display of the patients' statuses to various clinical staff inside the hospital.

To control the accuracy of the captured data, a refresh rate of 1 min was applied to RFIDTrack and an average of the last 100 readings was used. Averaging the temperature provides more accurate results since pediatric patients' temperature may fluctuate suddenly and then return back to normal which would cause false alert.

The alert mechanism applied using RFIDTrack is by sending a text message to a mobile device registered with the patient's clinical staff with the latest alert information. The causes of the alerts can be, for example, a steady rise in the temperature of the patient's body for the last few minutes based on the RFIDTrack tags captured data. In this case, RFIDTrack attempts to alert clinical staff of patients' vital signs if precursors of adverse events or conditions can be detected. RFIDTrack uses three main detecting algorithms: (a) one for detecting abnormal heartbeats, (b) detecting body temperature, and (c) detecting the patient gravitational positioning. An example of visually alerting medical staff is shown in Fig. 4.

The gravitational positioning is detected using an RFIDTrack tag that contains an accelerometer which enabled us to determine the location of the tag in a given time. Over a period of time, multiple location points are recorded. Hence, we can then determine any change in the position between two different time periods. That is, we can then determine the velocity the tag may have experienced. We applied an algorithm to detect such velocities and determine whether for example the patient is moving, fell down, fainted, among other possibilities. This is extremely useful for elderly patients who may not have support and may, for example, fall down without clinical staff acknowledging such event.

The accuracy of the RFIDTrack monitoring techniques were fine-tuned over a period of testing during this trial. However, the system was tested on multiple patients and shows a state-of-the-art monitoring technique that can be used efficiently inside hospitals. RFIDTrack utilized 25 RFID tags and 2 RFID readers. Over a period of 1 month, the accuracy of the RFIDTrack was apparent and 99.5 % of all precursors of adverse events were handled properly and on timely manner on all patients who used RFIDTrack tags.

Conclusion

RFID systems have been applied in the healthcare industry for mainly identifying patients and objects. In this paper, we presented RFIDTrack, a system that is capable of identifying objects and enabled the administration of patients' vital signs in real-time. By utilizing modern information and Web-enabled technologies, RFIDTrack extends the use of RFID beyond simple identification means. RFIDTrack uses RFID-based tags that are capable of detecting patients' temperature, heartbeats, and gravitational position. By continuously monitoring captured data, it is possible to improve patients' safety and prevent medical negligence or errors from taking place [15]. We also presented operating scenarios for testing our RFIDTrack solution.

One possible improvement for this system is to enhance the detecting of the gravitational location to include other features such as idle, involuntarily movements, sleeping, among others. We are also working on integrating additional RFID-based sensors to be utilized in this system. Another improvement is to build a cloud-based RFIDTrack framework to extend its applicability to other electronic patient systems inside hospitals.

Acknowledgment We would like to thank R. Simonetti, S. McIntyre, G. Bishay, M. Weinerich, and A. Mondal for their efforts and work in this topic at UOIT. This research was partially funded by iTEK Solutions Inc.

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An Evaluation of N-Gram Correspondence Models for Transliteration Detection

Peter Nabende

Abstract

Transliteration detection (TD) is a natural language processing (NLP) subtask that is used to find matching Named Entities (NEs) from parallel or comparable text where each language text is in a different writing system. The task is aimed at building high quality transliteration lexicons for improving performance in cross language applications like machine translation (MT) and cross language information retrieval (CLIR). Recent evaluations of TD methods (for example those from the NEWS 2010 transliteration mining shared task (Kumaran et al., 2010)) underscore the need for more methods that can improve TD performance. This paper contributes to this need by evaluating the use of source-target language n-gram correspondences for TD. We present TD experiments that use three different classes of n-gram correspondence models on standard transliteration datasets from the 2009 and 2010 shared tasks on transliteration generation (Li et al. (Report of NEWS 2009 machine transliteration shared task. Proceedings of the 2009 Named Entities Workshop: Shared Task on Transliteration, 2009); Li et al. (Report of NEWS 2010 transliteration generation shared task. Proceedings of the 2010 Named Entities Workshop, 2010)). Results show relatively significant TD performance improvements between the use of lower order and higher order n-gram correspondence models, and between the different classes of n-gram correspondence models. We show that our TD experimental setup is more complex than that in related work (Li et al. (Report of NEWS 2010 transliteration generation shared task. Proceedings of the 2010 Named Entities Workshop, 2010)) in terms of the search effort required, and that the best TD performances from the n-gram correspondence models in this paper are comparable to those from state-of-the-art methods. Our results also show that the n-gram size that should be used for developing high quality transliteration models in different languages and writing systems varies. Our work therefore serves to provide preliminary insight to the n-gram sizes required to model related high quality TD models for different language pairs and writing systems.

Keywords

Named entity matching • N-gram correspondence models • Transliteration detection

Introduction

Transliteration detection (TD) involves the search for matching named entities from bi or multi-lingual text where the source and target languages use different writing systems. The aim of the TD task is to build high quality transliteration lexicons that can be used for developing transliteration

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_79, © Springer International Publishing Switzerland 2015

generation systems and for increasing the coverage and quality of cross language applications such as MT and CLIR. The TD task is very beneficial since it guarantees the automatic handling of unknown Named Entities (NEs) across different languages and writing systems.

There are several methods that have been evaluated for the TD task. The most common recent evaluations are associated with the 2010 Named Entities Workshop shared task on transliteration mining (NEWS 2010) [4]. Despite the numerous TD methods that have been developed, TD literature shows that there is no 'silver bullet' method for achieving best TD performances for all existing languages and writing systems. In fact, recent TD research [4, 5] shows that most of the TD methods achieve best performances on transliteration datasets associated with usually only one or two language pairs. Of course this can be attributed to stark differences in the fundamental properties between different languages and writing systems. Thus, TD research is still growing with the main objective of identifying TD methods and developing models that can improve the quality of resulting transliteration lexicons. In this paper, we explore the use of a relatively simple method (as compared to most state-of-the-art methods) that uses source-target language n-gram correspondence information in the TD process. An n-gram refers to a contiguous sequence of n characters or symbols from a given word (in our case a named entity). An n-gram correspondence therefore has a pair of n-grams where an n-gram in one writing system corresponds to another n-gram in a different writing system.

The n-gram correspondence method that is proposed in this paper is based on the observation that in most cross language information processing applications, there is a high likelihood of using corresponding n-grams to accurately process information across the languages involved. For the TD task, we argue that it is possible to identify transliteration units (which are typically associated with pronunciation units) that correspond in two different languages and that can be used to convert an NE from one language to an equivalent representation in another language that uses a different writing system. For example, in the conversion of the name "Peter" in English to 彼 得 'Bǐde' in Chinese, it can be noted that the substring "Pe" in the English representation corresponds to 彼 'Bǐ' in Chinese whereas "ter" corresponds to 得 'de'. We observe that such correspondences can be found in different NEs that have similar pronunciations at similar positions. For example 'Pe' in "Petya" corresponds to Chinese 彼 'Bǐ' in 彼佳 'Bǐ jiā' just like in the case for "Peter" where 'Pe' is at the start position of each NE in this particular example. The challenge then is to identify all possible and plausible n-gram correspondences that we can use to estimate the level of relationship between proposed transliteration pairs in the TD task. A natural approach would require a manual process for identifying n-gram correspondences for use in the TD task. But a manual process would necessitate the use of high expertise in the languages involved and it is generally infeasible. This paper considerably overcomes this limitation by using a simple automatic method of collecting possible n-gram correspondences from training data and using them in the TD task.

The rest of the paper is structured as follows: in the next section we provide an overview of related work; in the section "N-Gram Correspondence Models", we describe the proposed n-gram correspondence models method; in the section "Transliteration Detection Experiments", we present the TD experimental setup and discuss TD performance results from the application of several n-gram correspondence models; in the section "Discussion of Results", we discuss our TD results from a comparative point of view with related work; and in the section "Conclusion and Future Work", we conclude and establish some pointers for future work.

Related Work

The process of detecting cross-language transliterations generally involves the search for corresponding NEs from a collection of candidate NEs between two or more languages in different writing systems. The main differences in the transliteration detection approaches are typified by the following [1]: the data source for obtaining candidate NEs in each of the languages; the techniques for detecting candidate NEs; and the techniques for matching and extracting potential transliteration pairs. The complexity of the transliteration detection task depends on the requirements for the task and the languages involved. For example, in some cases Chinese language text presents more challenges for transliteration detection because of extra work that is required to first segment sentences into a set of separate word entities [6]. Most of the methods in TD literature specify the cross-language NE matching step in a similar manner but only differ in the models that are used for estimating the level of similarity between candidate transliteration pairs. Several probabilistic methods use a noisy channel model (NCM) approach to model transliteration but differ in the probability distributions that they use for the NCM specification. Other approaches use an edit distance-based measure such as the Levenshtein edit distance that is computed from a common representation for the languages involved in the transliteration similarity estimate. Other approaches use a dynamic programming algorithm to evaluate alignments between potential transliterations and use an Expectation Maximization (EM) algorithm to derive transliteration models.

There are several limitations that are associated with most of the existing transliteration similarity estimation methods as compared to the n-gram correspondence models method. First of all, probabilistic methods would be limited in the size of context that they can model for estimating transliteration similarity. Probabilistic methods would require huge amounts of data to facilitate the derivation of probabilistic transliteration models that are not over fitted. Dynamic programming methods are also limited by the amount of context they can model for the same task. The use of Edit-distance-based methods to estimate transliteration similarity can only be simplified if we have a common intermediate written representation for the different languages. However, the process of converting transliteration data to one intermediate representation is very difficult and there is a high likelihood of generating several errors in the final representation.

The n-gram correspondence models approach overcomes most of the limitations above with hardly any additional effort. The use of n-gram correspondences where the order of n is greater than 1 already involves the representation of context. Moreover, there is no limit to the n-gram sizes (hence context sizes) that can be evaluated in our framework. As opposed to edit distance-based methods, our method does not require the use of an intermediate representation during the collection of n-gram correspondences from training data and during the evaluation of n-gram correspondences from testing data.

N-Gram Correspondence Models

N-gram correspondence models constitute n-gram correspondences that are obtained from training transliteration data. The n-gram correspondences are exploited to identify transliterations from unseen (testing) data. In this section, we describe how the n-gram correspondences are acquired and how they are exploited for estimating the similarity between candidate transliterations.

Collection of N-Gram Correspondences from Training Transliteration Data

In theory, we can construct an infinite number of n-gram correspondence models including those that have p-to-m (where $p \geq 1$, $m \geq 1$, and $p \neq m$) and m-to-m n-gram relationships between the source and target languages. In this paper, we consider only the latter case (m-to-m) where the size of the source language n-gram (m) is equal to the size of the target language n-gram in the correspondence. We limit our investigation to the case where $m = 2$ (2-to-2 correspondences) through to $m = 6$ (6-to-6 correspondences).

The n-gram correspondences are extracted from training transliteration data that comprises matching NEs across the different writing systems. Starting from the first matching NEs, we extract the correspondences until the last matching NEs. Consider the case where we are supposed to extract Russian-English tri-gram (3-to-3) correspondences from an instance of the Russian-English training dataset with the following NEs: Петр (Russian) and Peter (English). Given the two NEs, we extract the tri-gram correspondences at similar positions in the forward direction (that is from the start to the end of the two NEs) and then in the backward direction (from the end to the start). In the forward direction we extract the following tri-gram correspondences:

$\text{3-grams}_{\text{forward}} = \{\text{Пет-Pet, етр-ete}\}$

It is clear from the set above that we can only extract n-gram correspondences up to when we reach the end of the NE with the smaller number of characters (Петр in this case). In the backward direction we extract the following:

$\text{3-grams}_{\text{backward}} = \{\text{етр-ter, Пет-ete}\}$

The tri-gram correspondence list for this NE pair will therefore have four tri-gram correspondences. It is important to note that using both the forward and backward direction ensures that we capture useful n-gram information at the start and end of the NE that is longer in character length (Peter in this case). We can see from the extractions above that had we used only the forward direction, we would not be able to capture the last tri-gram ('ter' from 'Peter'). Likewise, had we used only the backward direction, we would not be able to capture the first tri-gram ('Pet').

The extraction process here is, however, possible if and only if there are no NEs from the involved languages whose character length is less than the n-gram size. To overcome this limitation for cases where the character length of some NEs is less than the n-gram size, we add to the NEs 'dummy' characters so that the length of every NE in the training and testing datasets is equal to or greater than the n-gram size. The number of 'dummy' characters that are needed is determined by analyzing the case for an NE that has only one character. For instance, for the tri-gram case, we would need two dummy characters (which from now on we denote by the hash symbol '#') to make the length of the NE the same as the size of the tri-gram. Since we extract n-gram correspondences in both directions, the 'dummy' characters are added before the start and after the end of each NE (that is A → ##A##). For a 5-gram correspondence search, we would need four dummy characters as prefix and suffix (A → ####A####). After adding the necessary 'dummy' character(s) to each NE in training data, the n-gram correspondences are extracted into a list as described above for the example of Петр and Peter. The n-gram correspondence extraction process is summarized in Algorithm 1 below. In Algorithm 1, pos refers to the position at which we extract n-grams in the source and target language

Algorithm 1: ExtractN-gramCorrespondences(L1, L2)

```
//Input: Two lists of correct transliteration matches: L1, L2
LC = { };     //LC is list of n-gram correspondences
//Extract
foreach correct transliteration match from L1 and L2 do
   //Forward direction
   pos = 1;     //position in NEs from L1 & L2
   while pos <= min(length(NE_SRC), length(NE_TRG)) – (N – 1) do
      extract SRC and TRG n-grams at pos in NEs from L1 & L2;
       if SRC-TRG n-gram correspondence does not yet exist
          in LC, append it to LC
       pos++;
   //Backward direction
   pos_end1 = max(length(NE_SRC), length(NE_TRG)) – (N – 1);
   pos_end2 = min(length(NE_SRC), length(NE_TRG)) – (N – 1);
   while pos_end2 > N – 1 do
   extract SRC and TRG n-grams at pos_end1 and pos_end2;
      if SRC-TRG n-gram correspondence does not yet exist
          in LC, append it to LC
      pos_end1– –;    pos_end2– –;
return LC
```

NEs; NE_{SRC} and NE_{TRG} refer to the source and target language NE respectively (e.g. Петр (Russian) and Peter (English) respectively); N represents the size of the n-gram (e.g. $N = 3$ for tri-grams); and length (NE_{SRC}) and length (NE_{TRG}) refer to the character lengths of the source and target language NE respectively. Note that from our description, we do not capture n-gram correspondence statistics in the resulting n-gram correspondence list. The models with statistical information would constitute the same n-gram correspondences but with statistical information (such as their frequencies of occurrence in training data) for each n-gram correspondence.

Transliteration Similarity Estimation

The transliteration detection task requires the identification of highly similar source-target language NEs from a collection of bilingual unstructured text. Here, the notion of similarity requires us to define a similarity estimation function that takes as input proposed candidate bilingual NE pairs and computes a similarity value for them using an n-gram correspondence model. The n-gram correspondence model can be derived from a list of n-gram correspondences from the subsection "Collection of N-Gram Correspondences from Training Transliteration Data" above.

We first extract n-gram correspondences from the proposed bilingual NE pair in a manner similar to that for extracting correspondences from transliteration matches from training data as described in subsection "Collection of N-Gram Correspondences from Training Transliteration Data" above. We then check to see whether each n-gram correspondence from the proposed bilingual NE pair exists in the n-gram correspondence model. If an n-gram correspondence from the proposed bilingual NE pair exists in the n-gram correspondence model, the similarity estimation function improves the similarity value for the NE pair according to a specific modification scheme. Otherwise, if an n-gram correspondence does not exist in the n-gram correspondence model, the similarity estimation function does not modify the similarity value for the NE pair. There are several modification schemes that can be used to update the similarity value of proposed NE pairs in the similarity estimation process. In this paper, the modification scheme is such that we simply increment the similarity value of the proposed NE pair by 1 if the NE pair's n-gram correspondence exists in the n-gram correspondence model. Although we use only one type of modification scheme, we investigate three classes of n-gram correspondence information from the proposed NE pair for estimating their similarity. In the first case, the similarity estimation function estimates the similarity value for the proposed NE pair only in the forward direction. That is, the similarity estimation function uses only n-gram correspondences that have been extracted from the proposed bilingual NEs in the forward direction to compute their similarity value. In the second case, the similarity estimation function estimates the similarity value for the proposed NE pair in the backward direction. That is the similarity estimation function uses only n-gram correspondences that have been extracted from the proposed NE pair in the backward direction to compute their similarity value. In the third case, the similarity estimation function estimates the similarity value using n-gram correspondences that have been extracted from the proposed NE pair in both the forward and backward direction.

After checking all n-gram pairs, an overall similarity value is associated with the proposed NE pair. The similarity estimation function computes similarity values for each source language NE against each target language NE from the unstructured bilingual text. The target language NE that posts the highest similarity value against a specific source language NE is proposed as a true transliteration of the source language NE by the transliteration detection system. The proposed transliteration pairs are then used to evaluate the accuracy of the transliteration detection system.

Transliteration Detection Experiments

Transliteration Data

We use standard transliteration data for seven language pairs from the 2009 and 2010 NEWS transliteration generation shared tasks [2, 3] to investigate the TD quality of the proposed n-gram correspondence models. The language pairs where each language uses a different writing system include: Bengali-English (Be-En), Chinese-English (Ch-En), Hindi-English (Hi-En), Kannada-English (Ka-En), Russian-English (Ru-En), Tamil-English (Ta-En), and Thai-English (Th-En). This transliteration data has been made available as standard data for evaluating machine transliteration systems. For simplicity, the first language in the language pair represents the source language whereas the second language refers to the target language. Each of these datasets is divided into training and testing datasets. In the training datasets we use only single source and target language matching NEs. Therefore, there are no whitespace characters that are usually associated with full person NEs. Most of the datasets need no further pre-processing apart from the Chinese-English dataset. In a pre-evaluation of Chinese-English TD performance using the original simplified Chinese orthography, we got below 1 % TD accuracy for all proposed m-to-m n-gram correspondence models. This very poor TD performance on the original Chinese-English dataset is not surprising since the Chinese language uses a semanto-phonetic/logographic writing system (where symbols often represent both sound and meaning) in contrast to the English language that uses a phonemic alphabet (where each letter represents one or more phonemes for both consonants and vowels). At the most basic level, we expect to associate each simplified Chinese symbol with more than one English letter. This may necessitate the use of 1-to-m Chinese-English n-gram correspondences for detecting transliterations between Chinese and English. In the Chinese-English example of 彼 得 'Bĭde' and Peter, we see that each Chinese symbol corresponds to at least two English letters, i.e. 彼 ↔ Pe and 得 ↔ ter. It is therefore not plausible to employ m-to-m n-gram correspondences for the Chinese-English TD task. The approach we take to make it effectively possible to apply m-to-m n-gram correspondences for the Chinese-English TD task is to first convert the Chinese side of the Chinese-English dataset to a romanized representation using the Pinyin romanization system. With this approach, 彼 得 gets converted to Bĭde using the Pinyin system and a comparison of Peter to Bĭde can yield reasonable reasonable m-to-m n-gram correspondences. For example, we would get the following tri-gram correspondences: Bĭd-Pet, ĭde-ete in the forward direction, and ĭde-ter, Bĭd-ete in the backward direction.

All the Indic languages (Bengali, Hindi, Kannada, and Tamil) and the Thai language use syllabic alphabets. Symbols in syllabic alphabets represent consonant-vowel sequences such that each symbol is based on a consonant letter and the vowel representation is secondary. This is in contrast to the Latin alphabet for English where vowels have equal status to consonants. Clearly, if we are to use m-to-m n-gram correspondences for estimating transliteration similarity between NEs with a syllabic alphabet representation and NEs with a phonemic alphabet representation, it makes more sense to convert the original syllabic representations to romanized representations to ensure a common basis for comparison. However, in a pre-evaluation of TD performance on the datasets for these languages, we got above 50 % TD accuracy for most of them. We therefore decided to investigate TD performance using only the original alphabetic representation of these languages. Because of space constraints, we propose the investigation of TD performance using romanized representations for these languages as future work. The Russian language uses a phonemic alphabet and therefore, we use the original orthography in the Russian-English TD task.

Table 1 shows the total number of NE pairs per language pair that were used for experimentation. For each language pair, we select 90 % of each dataset for training and the rest (10 %) for testing. The training data for each language pair exists in two files such that there are corresponding NEs at the same line in both files.

It is important to note that in our TD tasks, the system searches thousands of target language NEs to propose matching transliterations for the source language NE. This makes the search space in our task larger than that in related work (for example in [4] where transliterations are only detected from corresponding Wikipedia topics).

Evaluation of the TD Quality of the N-Gram Correspondence Models

We evaluate TD performance associated with using the proposed n-gram correspondence models using standard TD evaluation metrics: TD accuracy and Average Reciprocal Rank (ARR). TD accuracy measures the proportion of target language NEs that are correctly detected as transliterations of the source language NEs.

$$\text{TD Accuracy } = \frac{1}{N}\sum_{i=1}^{N} 1 \text{ if } \exists r_{i,j} = c_{i,1}; 0 \text{ otherwise} \qquad (1)$$

where $r_{i,j}$ is the jth correctly matching target NE for the ith source NE in test data, $c_{i,1}$ is the first candidate target NE returned by the TD system, and N is the size of the test data.

Table 1 Total number of NEs per language pair

Language pair	Number of NE pairs
Bengali-English (Be-En)	15,000
Chinese-English (Ch-En)	37,753
Hindi-English (Hi-En)	19,233
Kannada-English (Ka-En)	16,063
Russian-English (Ru-En)	7,840
Tamil-English (Ta-En)	16,285
Thai-English (Th-En)	31,367

Average Reciprocal Rank is used to determine the average of the reciprocal ranks at which the correct target matching NEs were identified for the source NEs. ARR closer to 1 implies that many correct transliterations are detected close to the top of the best list proposed by the TD system. If a target language candidate NE matches one of the target language reference NEs in the *j*th position in the best list, its rank is *j* and its reciprocal rank is 1/*j*. Here we would like the value of j to be as minimal as possible. Equation (2) shows how ARR is determined.

$$ARR = \frac{1}{N}\sum_{i=1}^{N}\left\{\min_{j}\frac{1}{j} \text{ if } \exists r_{i,j}, c_{i,k} : r_{i,j} = c_{i,k}\right\} \quad (2)$$

Using the metrics above, we perform stratified 10-fold cross-validation (CV) to evaluate each of the n-gram correspondence models in the previous section. In stratified 10-fold cross-validation, we subdivide each dataset into ten equally sized datasets such that each of the ten datasets is used as testing data while the rest is used as training data.

Discussion of Results

Table 2 shows the TD CV accuracy and CV ARR from the use of n-gram correspondence information in only the forward direction. Table 3 shows the TD CV accuracy and CV ARR from the use of n-gram correspondence information in only the backward direction. Table 4 shows the CV accuracy and CV ARR from the use of n-gram correspondence information in both the forward and backward directions. The TD performance results in Tables 2, 3, and 4 suggest that TD CV accuracy and CV ARR generally increase from lower order n-gram correspondence models (2-gram and 3-gram) to higher order n-gram models (4-gram, 5-gram and 6-gram). It is clear from Table 2 that 2-gram correspondence models are not at all useful for transliteration detection on all language pairs. From the 2-gram to the 3-gram correspondence models, we see a significant improvement in TD CV accuracy and CV ARR. However, we get mixed TD performance results from the use of 3-gram to the use of 4-gram correspondences. Tables 2 and 4 TD performance results suggest that the use of 3-gram correspondences benefits only the Russian-English TD task where we use n-gram correspondences from only the forward direction and in both directions respectively. In contrast, Table 3 TD performance results suggest that the use of 3-gram correspondences in only the backward direction benefits not only the Russian-English TD task but also the Hindi-English and Kannada-English TD tasks. From Tables 2, 3 and 4 it is also clear that the use of 3-gram correspondence information is generally useless for the Chinese-English and Thai-English TD tasks. The 4-gram correspondence model's performance is better than the performance of the other n-gram correspondence models as it posts best results on 5/7 language pairs in all cases. The 5-gram correspondence model does not post best CV accuracy and CV ARR results on all language pairs and in all cases but its performance is significantly better than that of the 3-gram correspondence model when we use n-gram correspondences from only the forward direction and from both the forward and backward directions. Table 3 results suggest that the 5-gram and 6-gram correspondence models are useful if we consider n-gram correspondences from only the backward direction. The 6-gram correspondence model, however, achieves best CV accuracy and CV ARR results on 4/7 language pairs in all cases.

Table 2 Cross validation transliteration detection accuracy and MRR for different n-gram correspondence models using forward direction

N-gram models	Language pairs	CV top-1 accuracy	CV MRR
2-Grams	Be-En	2.0	3.4
	Ch-En	0.3	0.7
	Hi-En	5.5	8.1
	Ka-En	0.3	0.8
	Ru-En	16.1	22.6
	Ta-En	1.0	1.9
	Th-En	0.2	0.6
3-Grams	Be-En	65.6	71.5
	Ch-En	23.5	29.5
	Hi-En	75.5	80.1
	Ka-En	57.2	65.2
	Ru-En	**80.8**	**83.1**
	Ta-En	50.4	59.4
	Th-En	26.1	32.2
4-Grams	**Be-En**	**68.8**	**74.4**
	Ch-En	**41.4**	**47.7**
	Hi-En	78.1	82.2
	Ka-En	67.2	73.9
	Ru-En	78.6	81.3
	Ta-En	65.7	72.1
	Th-En	40.4	46.9
5-Grams	Be-En	66.5	72.6
	Ch-En	41.2	47.3
	Hi-En	78.6	82.6
	Ka-En	68.3	74.3
	Ru-En	77.1	80.1
	Ta-En	67.2	72.8
	Th-En	43.4	49.9
6-Grams	Be-En	66.1	72.4
	Ch-En	39.2	45.5
	Hi-En	**79.3**	**83.1**
	Ka-En	**68.6**	**74.6**
	Ru-En	76.7	79.6
	Ta-En	**68.4**	**73.8**
	Th-En	**45.4**	**51.4**

Bold formatting indicates best performance in this set of models

Table 3 Cross validation transliteration detection accuracy and MRR for n-gram correspondence models using backward direction

N-gram models	Language pairs	CV top-1 accuracy	CV MRR
3-Grams	Be-En	63.4	69.6
	Ch-En	28.6	34.8
	Hi-En	**65.6**	**72.1**
	Ka-En	**65.0**	**71.9**
	Ru-En	**74.6**	**79.0**
	Ta-En	58.6	66.5
	Th-En	36.6	43.5
4-Grams	**Be-En**	**63.6**	**69.7**
	Ch-En	**44.8**	**51.1**
	Hi-En	62.5	68.7
	Ka-En	63.0	69.0
	Ru-En	67.7	72.6
	Ta-En	**63.9**	**69.8**
	Th-En	**53.1**	**58.7**
5-Grams	Be-En	52.5	60.1
	Ch-En	31.3	35.7
	Hi-En	54.6	61.8
	Ka-En	53.7	60.7
	Ru-En	56.2	62.4
	Ta-En	51.5	59.2
	Th-En	43.9	49.8
6-Grams	Be-En	48.5	56.7
	Ch-En	15.4	18.4
	Hi-En	52.2	59.8
	Ka-En	52.2	59.4
	Ru-En	47.2	55.3
	Ta-En	49.4	57.0
	Th-En	36.3	43.1

Bold formatting indicates best performance from this set of models

Table 4 Cross validation transliteration detection accuracy and MRR for different n-gram correspondence models using both forward and backward direction

N-gram models	Language pairs	CV Top-1 Accuracy	CV MRR
3-grams	Be-En	73.2	78.1
	Ch-En	29.6	36.1
	Hi-En	80.7	85.2
	Ka-En	70.3	76.7
	Ru-En	82.7	85.0
	Ta-En	62.7	70.5
	Th-En	35.7	42.8
4-grams	Be-En	80.2	84.3
	Ch-En	53.6	59.8
	Hi-En	86.7	90.2
	Ka-En	79.7	84.8
	Ru-En	81.9	84.4
	Ta-En	79.1	84.0
	Th-En	58.3	64.4
5-grams	Be-En	79.0	83.4
	Ch-En	51.5	57.5
	Hi-En	87.3	90.5
	Ka-En	79.3	84.6
	Ru-En	80.3	82.8
	Ta-En	80.1	84.8
	Th-En	60.2	66.2
6-grams	Be-En	79.0	83.5
	Ch-En	48.8	55.1
	Hi-En	87.7	90.6
	Ka-En	79.6	84.6
	Ru-En	79.9	82.3
	Ta-En	80.7	85.3
	Th-En	61.2	66.8

Shaded formatting indicates best performance compared to results in Tables 2 and 3

A comparison of TD results across Tables 2, 3, and 4 shows significant differences. Tables 2 and 3 results shows that the use of m-to-m n-gram correspondences in only the forward direction results in significantly better TD accuracy and ARR than the use of m-to-m n-gram correspondences in only the backward direction. However, Table 4 results suggest that the use of m-to-m n-gram correspondences from both directions results in significantly better TD accuracy and ARR than the use of n-gram correspondences from only one direction.

Conclusion and Future Work

We have explored the use of several m-to-m n-gram correspondence models for transliteration detection. Results show variations in transliteration detection performance from lower-order n-grams to higher-order n-grams. However, there is a general improvement in TD performance as we increase the size of n-grams for the best case of using n-gram correspondence information in both the forward and backward direction. TD results, however, show that the n-gram correspondence models achieve TD accuracies above 80 % on only 4/7 language pairs. Moreover this performance is achieved in only the case where we use n-gram correspondence information from both the forward and backward directions. Our results prove that the use of only forward direction makes the TD system lose out on n-gram correspondence information from the backward direction and vice versa. Most of the n-gram correspondence models post competitive TD accuracies compared to state-of-the-art performances taking into consideration the fact that the search space for transliterations in our TD task is larger than that in related work.

For future work, we note that the approach proposed in this paper can be extended in several directions. An exploration of combining results from the different n-gram models should be interesting. Another direction is to investigate the use of p-to-m n-gram correspondences where $p \neq m$.

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Design of an Optimized Crane Location System for Construction Sites

Soly Mathew Biju, AbdulLatif Tchan, Marina Helmy, and Kirolos Ayad

Abstract

Tower cranes, have always the centerpiece of production, hoisting and transporting of a variety of loads on a construction site. Occasionally tower cranes operate with overlapping work zones and Crane positioning in any construction site follows a complicated procedure. Factors that have to be taken under consideration are building size and type, obstruction type, material yard location. Each project is unique and different from the other and thus location of the cranes in every situation.

A new developed application named Crane Locator is designed to facilitate the crane locating process by converting the engineer's expert knowledge to computerized System embedded knowledge. The application gives engineers the right location of where the crane should be positioned, the number, model(s) and type(s) of cranes.

Keywords

Location System • Optimization • Construction Sites • Tower cranes • Computerized System • application

Introduction

Cranes are considered to be one of the most important equipment used daily on a construction sites due to their key role in performing lifting tasks. The scale of investment in choosing a crane emphasizes the importance of the crane selection process. Thus, careful attention to such selection should be considered owing to the huge price that may be paid in case of mistakes.

The ability to predict and make positing decisions of cranes grows out of knowledge and experience gained during many years of work on construction sites; it requires expertise in the field and that takes time and effort. Since it's a human made decision, making errors is quite probable. Crane positioning is a critical matter and any wrong decision taken would lead to dramatics consequences not only to the cost and time but also leading to serious accidents at construction sites. Using the Crane Locator application, civil engineers can easily locate tower cranes without worrying about the possibility of human errors and accidents.

The application gives the user the location, number, model and type of the tower crane(s) to be used at a construction site. Currently, the procedure of locating the right location is done manually by expertise. However, with the Crane Locator application an optimized solution is determined based on given a set of parameters. Engineers' knowledge and experience were utilized during the process of the design and the implementation of the Crane Locator application. Required rules and procedures for locating cranes are coded in this system for better solutions. The purpose of the system is to facilitate the manual procedure carried out in locating the tower cranes required for a job; thus saving time, effort and money for engineers and decreasing fatal incidents.

One of the main issues concerning the location of cranes is obstructions. Obstructions could be neighboring buildings, roads, offices, pipelines or other cranes, all these should be

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_80, © Springer International Publishing Switzerland 2015

avoided by the crane unless the authorities have permission from the authority; The Crane Locator application takes into account of such an issue: it asks the user if there is permission or not. If there is not, it avoids the obstruction area that is covered by cranes according to the obstruction type, because different obstructions have different avoiding techniques and thus cranes will have different activation areas.

Existing Research

Frans Van Gassel and Wim Schaefer [2, 3] in their paper have described a decision Support System for a crane plan as part of generic construction plan.

A crane planner is developed by making use of Delphi 3.0 environment, the application runs on Windows 95 computers. Crane planner provides the user with the position of crane on a construction site, specification of selected crane, pick up and deposit sites for building parts.

The crane Planner has four model elements:

1) Knowledge base: where it stores all the knowledge that we gathered from the interviews that were held with engineers and from literature, this knowledge help in making the decisions.
2) Procedures: determines crane effort, type, position and costs.
3) Data Processing: The program consists of 90 screens for the user interface.
4) User-Interface: The program's outputs a 3-D drawing of the building and materials that have to be transported by the crane, 3-D drawing and the selected crane.

The application is useful as it provides a drawing of cranes by the computer which will be waste of time if done by hand, it's helping the engineers in taking the right decision, it provides tips and advices for the user which is very useful to enlarge their knowledge.

Javier Irizarry, Ebrahim P Karam [1] in their paper have examined the existing work done and practices by the engineers of crane selection used on high-rise buildings projects. The author studied the factors that affect the selection and the display of the process in a computerized way.

- The user inputs to the program are: the shape of the building, the type of construction, material specifications, and site constraints. The output results are: type number, position of crane(s), cost and time productivity of crane(s).
- Researchers have classified factors affecting crane selection into technical factors, contractual factors and economic factors.
- The system is based on two modules; Knowledge based module in which information has been collected from expertise related to crane selection process and Case Based module in which it has information on previous cases representing already constructed buildings with pre-selected crane(s). The system has embedded algorithms for calculating and validating graphics to support the crane selection process.

The System also gives the user more different alternative choice for crane location and type.

Alkass, S, Alhussein, M and Moselhi, O [4] have presented a new methodology for integrating the GIS (geographic information systems) with BIM (building information modelling); GIS: is applied to evaluate the objects surrounding the building in the most intellectual way. BIM is engaged to develop building models to visualize the results from the GIS into a 3D world.

The solution is intended through various steps, calculating the lifting radius, generating the feasible task areas for each of the pre located cranes, listing and ranking the tasks according to the frequency of the task and then according to these calculations the location is designed and the number of cranes.

Design

The user has to define each of the objects inside the blueprint in an output screen that shows him/her the detected shapes in the blueprint. If there are obstructions defined by the system then the user has to enter the permission of each as true or false. Obstructions' heights are also required for the system in order to locate the crane appropriately. Finally the desired max capacity the crane will carry during the life time of the project (Fig. 1).

Figure 2 is the first input screen that asks the user to enter the connection name to connect to database, the scale ratio to map the output, maximum weight of the crane required and the he boundary limit of the building. Generate will output the drawing of the construction site.

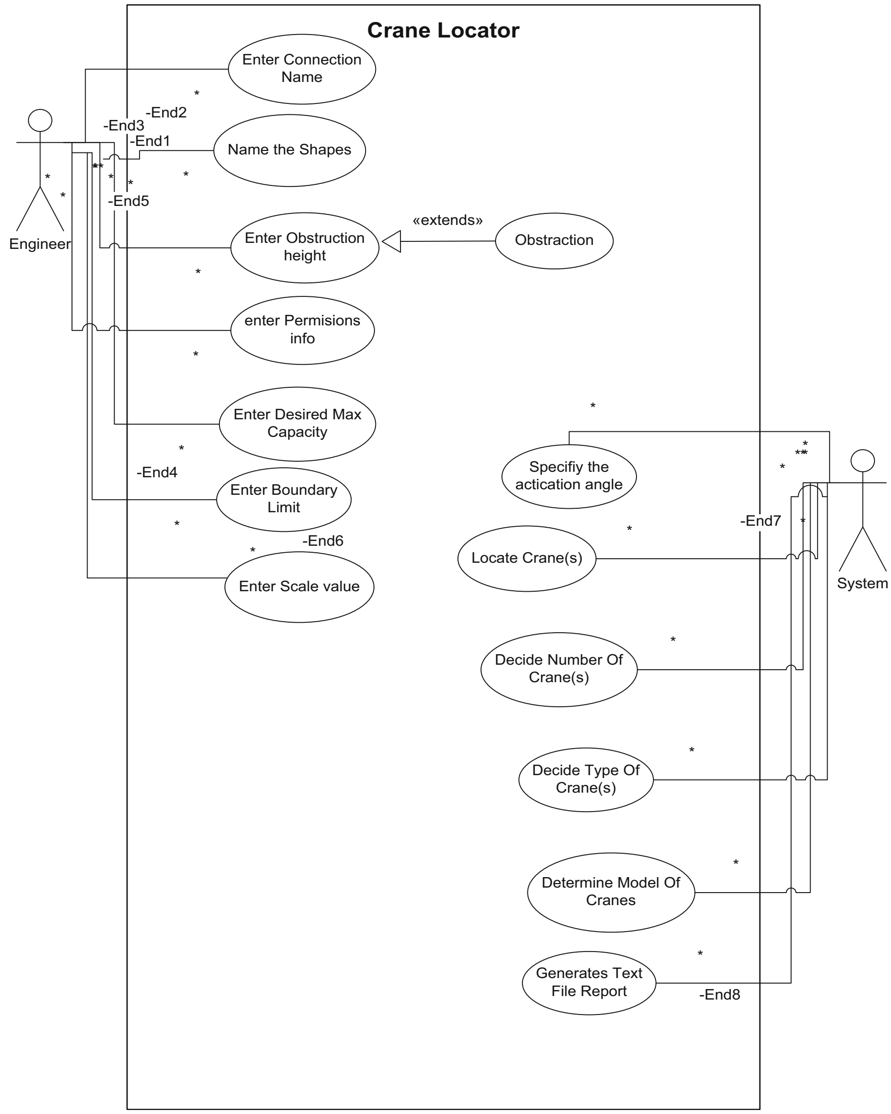
Figure 3 shows how objects to be placed in the construction site.

Algorithms:

Methodology for Locating the First Crane Without Obstruction

The first criteria is to locate the first crane so as to completely cover the material yard. To meet this criteria the side of the building closest to the material yard had to be located. The algorithm used was to compare each point of the building to all the material yard points and the closest two points at the side of the building is selected. After the side of the build is decided upon, the program generates random point along it using the equation of a line, then finds its mirrored image that lies on the boundary limit on

Fig. 1



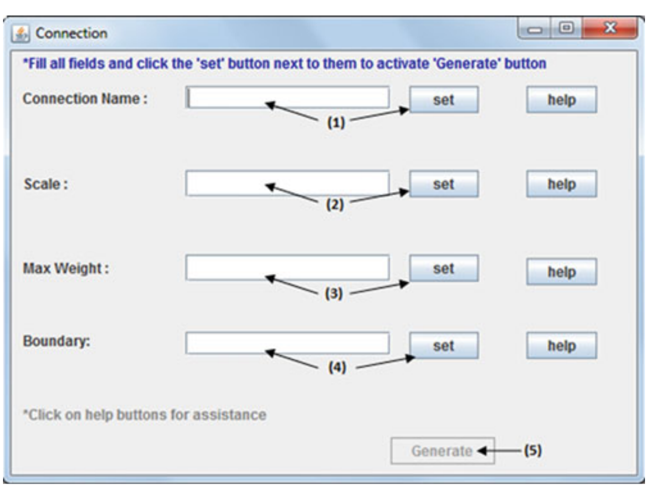


Fig. 2 Input screen

both sides then it removes the mirrored point which is inside the building, the remaining mirrored points and the original random point make up a line and the program generates a random point on this line to be the center of the first crane. Generating random points is useful in this algorithm as there could be a number of valid locations for the crane.

To choose the radius of the crane, the distance between the crane center and all the points of material yard is measured, and the greater (furthest point) distance is chosen to be the crane radius, this is done to ensure that the material yard is covered 100 %.

Methodology for Locating the Second Crane Without Obstruction

Second Crane will be placed on the construction site only if the building is not completely covered by the first crane. The

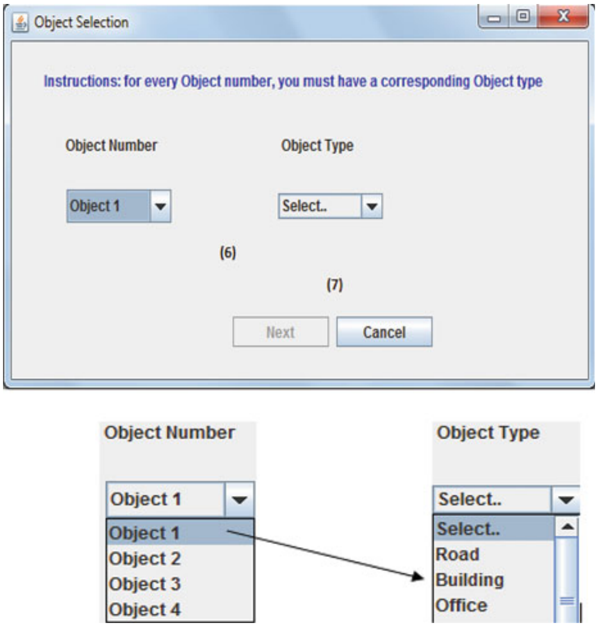


Fig. 3 Object selection screen

methodology for choosing the center of the second crane is the same as that for the first crane by generating random points on the preferable side. First check is whether the building is covered, if that is not the case the number of uncovered building sides are obtained, if the number is even and equal 2 then the crane is chosen to be placed on the first non-covered side (preferred side, if the number is greater than 2 then we divide by 2 and locate the crane on that side with index (result from division) to be the preferred side. If the number of sides is equal to 1 then it is decided to place the crane on that remaining uncovered side(preferable side), if the number is greater than 1 then it is divide by 2 and the crane is located on that side with index obtained as a result of the division to be the preferred side.

Avoid Obstructions Methodology

This function is designed to take care that the crane avoids hitting the obstruction or overlap it depending on its location with respect to the crane location and the permission given to that obstruction (whether its approved to overlap it or not). The function checks through the vector containing the coordinates of the obstructions. And with respect to the crane radius the function checks the angles between each two points inside the vector with those points and it retrieves the first point and the second point for the highest angle. The sector thus formed will be the sector of the circle to be avoided or considered as dead Zone for the crane (sector the crane cannot reach).

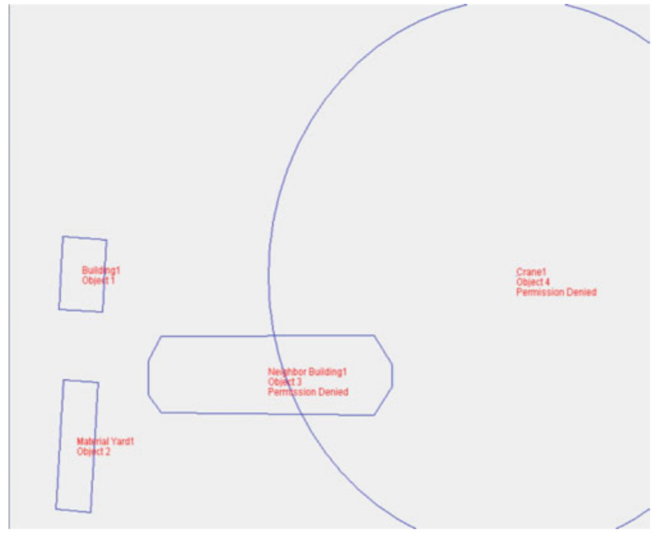


Fig. 4 Final output screen

This function treats all the special cases because the shape can be in the first and the second quadrants of the same circle at the same time, and the functionality for getting the point angle with respect to the 0 value, according to which the lowest point in the shape will have the highest angle ($270 < \text{ANGLE} \leq 360$), and this case solution has been taking care by special case function.

Avoid Pipeline Algorithm: Pipeline is type of an underground obstruction; Cranes can't be placed over the area if the user doesn't have permission.

The algorithm detects if the crane's center point is placed over the pipeline or not, if it's placed over it then it calls the function in order to get another random point and recursively checks if the new random point is inside the pipeline or not. If not then it will return the location for the cranes.

Figure 4 shows the construction site with the positions of the crane and the area it can over along with all the objects on the site.

Conclusion

Crane Locator assists engineers in the stage of locating tower cranes setup and deciding on the type and number of cranes to be used in a specific construction project. There are several factors that are involved including: building's dimensions and type, material yard's location with respect to the building's location, load's capacity that have to be lifted by crane from the material yard to the building, obstruction's type whether above ground or underground, obstruction's location proportion to the building's location and obstruction's permission. Each of these factors has a significant effect on the decision that has to be taken. Users' inputs should be accurate as it will affects the calculation to locate crane(s).

The program contains embedded algorithms and methodologies for various cases: whether the construction site has obstructions, whether the obstruction is above ground or underground, etc. Thus the program will take a turn on applying different methodologies depending on the user's input. As there is not just one right location for cranes in the construction site, Crane Locator gives the user couple of choices thus allowing the user to decide on which one is more applicable for the case and more suitable for the current project. Thus giving variety of options to the engineer and not limiting to one possible solution.

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Route Tracking of Moving Vehicles for Collision Avoidance Using Android Smartphones

Ehab Ahmed Ibrahim, Said El Noubi, and Moustafa H. Aly

Abstract

Android Smartphones are widely used nowadays. They have a lot of capabilities that made our life easier and more comfort. They opened the area of Android programming used to develop applications suitable for these smartphones. Our target, in this paper, is to propose an application that helps in collision avoidance between moving vehicles (especially trains) by tracking their routes. This application is a modified version of Route Tracker Application [Harvey and Deitel (Android for programmers an app-driven approach, 2011)], we use the Route Tracker Application on each smartphone that installed in the moving vehicle, to track its route. Then we save all trains points in a database, construct a map shows these points. In this map, a different color is assigned to each vehicle to distinguish it. This map is shown on a smartphone found in a control room. Using points in this map, the distance between vehicles is calculated (distance between points of vehicles). When two vehicles, for example, become too closer to each other, the inspector, in the control (monitoring) room, must take an action, he generates an alarm to warn the drivers of both vehicles so they can adjust their speed to avoid collision.

Keywords

ADT • AVD • DDMS • KML • GPX • JSON

Introduction

The Android operating system [1–4] was developed by Android, Inc., which was acquired by Google in July 2005. In November 2007, the Open Handset Alliance™—a 34-company consortium initially and 81 now was formed to develop Android, driving innovation in mobile technology and improving the user experience while reducing cost. Android is used in numerous smartphones, e-reader devices and tablet computers.

One benefit of developing Android applications is that it is open source and free operating system. This allows to view Android source code and see how its features are implemented. One can also contribute to Android by reporting bugs or by participating in the open source project discussion groups.

Android applications are developed with Java [5, 6]; the world's most widely used programming language. Java was a logical choice for the Android platform, because it is powerful, free and open source. Java is used to develop large-scale enterprise applications to enhance the functionality of web servers and to provide applications for consumer devices (e.g., cell phones, pagers and personal digital assistants) and for many other purposes.

It enables to develop applications that will run on a variety of devices without any platform-specific code. Experienced

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K. Elleithy and T. Sobh (eds.), *New Trends in Networking, Computing, E-learning, Systems Sciences, and Engineering*, Lecture Notes in Electrical Engineering 312,
DOI 10.1007/978-3-319-06764-3_81, © Springer International Publishing Switzerland 2015

Java programmers can quickly dive into Android development, using the Android (Application Programming Interfaces) APIs and others available from third parties.

Java is object oriented and has access to powerful class libraries that help in developing applications quickly. One can write applications that respond to various user-initiated events such as screen touches and keystrokes. In addition to directly programming portions of applications, one may also use Eclipse to conveniently drag and drop predefined objects such as buttons and textboxes into place on your screen, and label and resize them. Using Eclipse [7] with the Android Development Tools (ADT) Plugin, one can create, run, test and debug Android applications quickly and conveniently, and one can visually design his user interfaces.

Application Model

Our developed application depends mainly on Google Maps and Route Tracker Applications (discussed in next sections). We install a smartphone in each moving vehicles, on the phone we displaying a map that shows the vehicles moving points. Another map is used to show points of all vehicles. We used Google Maps Application to perform the following tasks:

1. Show a Google Map on each smartphones.
2. Locate the vehicles points on the map.
3. Add markers to these points (one marker for each point with the ability to use different markers colors for different points).
4. Draw a line that connects all these points together (route path).

At this stage, we completed an easy but very important task. Now, we become familiar with Google Maps and can deal, easily, with it. Then we move to the Route Tracker Application. This application targets to track only one route (for one moving vehicle).

To avoid collisions, we need to track routes for more than one vehicle (at least two), so we need to modify the Route Track Application to fit in our situation. We install the Route Tracker application for each smartphone, then collect the points of each vehicle (these points are the Route Tracker output) and save them on a database.

Finally, we show these points on a map that can be accessed by the vehicle driver, thus all drivers have an overview of the routes of other vehicles. If any sudden change occurs in any vehicle path, it updated automatically to all vehicles, so drivers can take a fast action towards this change.

Google Maps Application

Google Maps is one of the many applications bundled with the Android platform. In addition to simply using the Maps application, one can also embed it into his own applications and make it do some very cool things. This Android Google Maps application [8] shows how to use Google Maps in Android applications and how to programmatically perform the following:

- Change the views of Google Maps (street and satellite view).
- Display a Particular Location on the map.
- Display the Zoom Controls.
- Obtain the latitude and longitude of locations in Google Maps.
- Perform geocoding and reverse geocoding.
- Add multiple markers to Google Maps.

Obtaining a Maps API Key

One needs to apply for a free Google Maps API key before he can integrate Google Maps into his Android application. API version 2 is already in use today but for application in this paper we use API version 1.

Displaying the Map

To display the Google Maps in Android application, modify the main.xml file. One shall use the <com.google.android.maps.MapView> element to display the Google Maps and modify main class to extend from the MapActivity class, instead of the normal Activity (Fig. 1).

Displaying a Particular Location

By default, the Google Maps displays the map of the United States when it is first loaded. However, one can also set the Google Maps to display a particular location.

Getting the Location That Was Touched (Geocoding)

After using Google Maps for a while, one may wish to know the latitude and longitude of a location corresponding to the position on the screen that one has just touched. Knowing this information is very useful as one can find out the address of a location, a process known as Geocoding.

Adding Multiple Markers to Google Map

The objective is to add multiple markers to Google Map Application for Android. One may take a scenario where having a list of coordinates of different points and wants to display a marker on each point.

Fig. 1 Displaying Google maps Android Application

One must add a class that extends ItemizedOverlay which consists of a list of OverlayItems. This handles sorting north-to-south for drawing, creating span bounds, drawing a marker for each point, and maintaining a focused item. It also matches screen-taps to items, and dispatches focus-change events to an optional listener.

Drawing a Path or Line Between Several Locations

The objective is to draw a line between two points on Google Maps let's say point "A" and point "B". Repeat the process to draw a line between point "B" and point "C".

Route Tracking of Moving Vehicles

First, to run this Route Tracker application or to create your own application using the Google Maps API, you'll need to obtain a unique API key from Google. Applications must be signed with a digital certificate before they can be installed on a device. When you are building and testing applications, the ADT Plugin handles this automatically by creating a debug certificate and using it to sign your applications.

An Android device (such as smartphone) must be installed on the moving vehicle that one wants to track its route. This smartphone must have internet access to receive the map images, to acquire a GPS signal, the smartphone must have line-of-sight with the GPS satellite (the signal can take several minutes).

Once the Route Tracker application is running on the smartphone, and a GPS signal is received, a Toast appears on the screen saying that the GPS signal has been acquired. At this point, touch "Start Tracking" button.

As the vehicle moves, its route is marked with a red line. Open the application's menu and touch the Satellite item to display a satellite image rather than a standard street map.

One can switch back to a street map by selecting the menu's Map item. When the vehicle is stopped (have finished its route), touch "Stop Tracking" button. Touching "Start Tracking" again erases your route from the map and starts tracking a new one (Figs. 2, 3, and 4).

Then, one can extend the process of route tracking to more than one moving vehicle (two or more). For each vehicle, a smartphone must be installed and connected to internet and GPS Satellite.

We can use this approach in railway stations for tracking the routes of trains. For example, take a scenario of two trains tracking. A map is displayed on a smartphone on both trains. The first train positions are drawn on the map as blue balloons. The map is updated periodically with each point the train reaches. For the second train, points are drawn as a red balloons. The map on each train is showing its route. When the other train reaches a station (new point), the map is updated to show both train locations on the same map. So each train driver has an information about the current state. He can take a decision if any unusual event occurs. For example, if second train becomes closer, he must slow down and send alarm to this train. Also if one train is stopped suddenly, the train driver must send an alarm to all other trains to pay attention. Another situation is when a train gets close to a station or a crossing, the driver must be aware of that to slow down its speed.

A third map is displayed on a smartphone on a control room showing trains routes. The inspector on the control room also has the ability to take action such as open or close station barrier, send a signal to all other trains to slow down

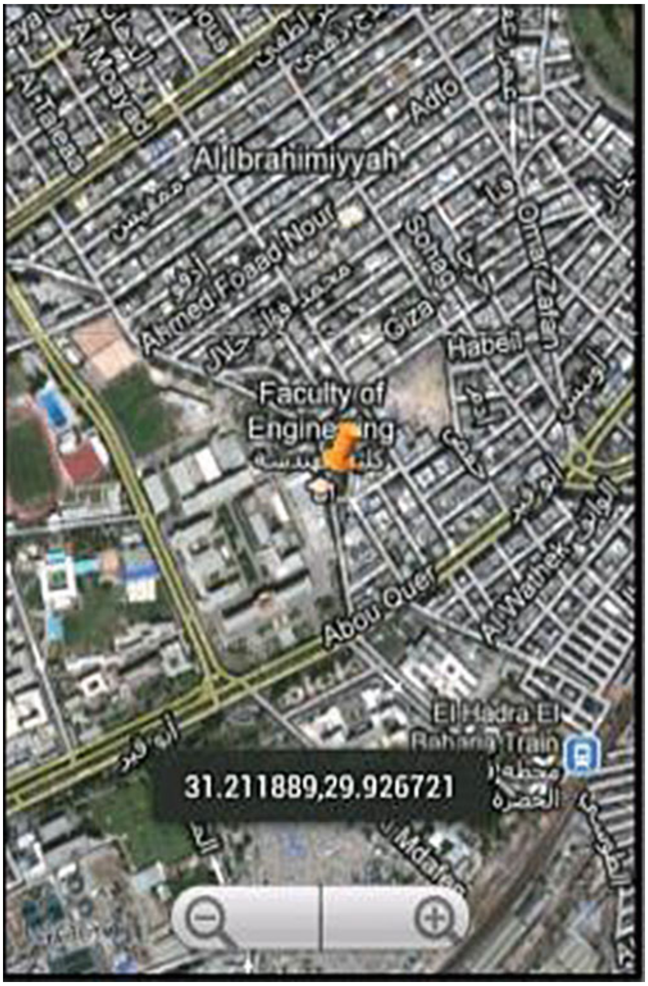

Fig. 2 Adding markers and geocoding

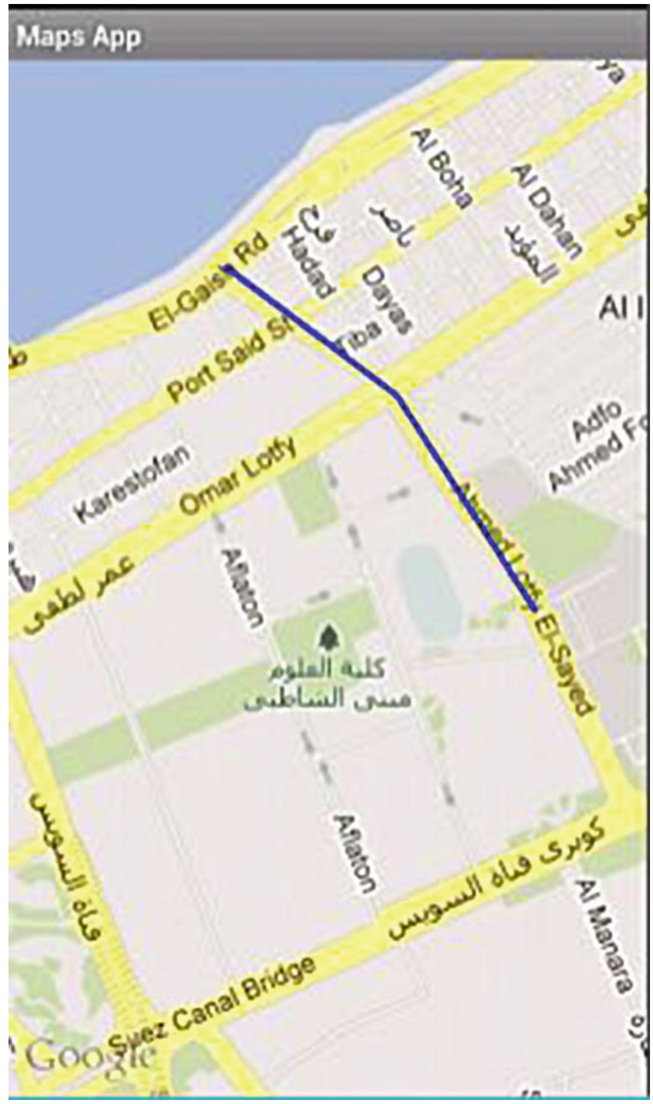

Fig. 3 Drawing a route path between three points

or completely stop until a crossing situation is perfectly handled. By this procedure, one can avoid (or at least decrease) collisions between trains and improve the safety factor of the train railway network.

Simulation Procedure

This simulation is carried out in an **A**ndroid **V**irtual **D**evice (**AVD**) [9] which is configured to use the Google APIs.

Sending GPS Data to an AVD

The Android emulator enables to send GPS data to an AVD. So, one can test this application without an actual Android smartphone. One can send a single point (latitude and longitude) manually Using **D**alvik **D**ebug **M**onitor **S**erver (**DDMS**) [10]. Actually, sending only one point is not suitable for tracking moving vehicle. So, one needs a series of points to really simulate a GPS data.

One can use a (**K**eyhole **M**arkup **L**anguage) [11] file (file with kml extension) that can be generated by simply drawing the desired routes on Google Earth software, save it in kml format and then convert this Google Earth routes to a format suitable for Eclipse Android ADT plugin using google earth to android ADT tool (http://ge2adt.applicationspot.com/).

This method has a drawback that the kml file contains only points but without their times (points are located on the Map all at once). So, one needs another way to take the factor of time into consideration so as to perfectly simulate GPS data.

To do so, one shall use a file containing GPS data in GPS Exchange Format. Such files typically end with the gpx extension and are called GPX files that one can load and "play" from the ADT Plugin DDMS perspective.

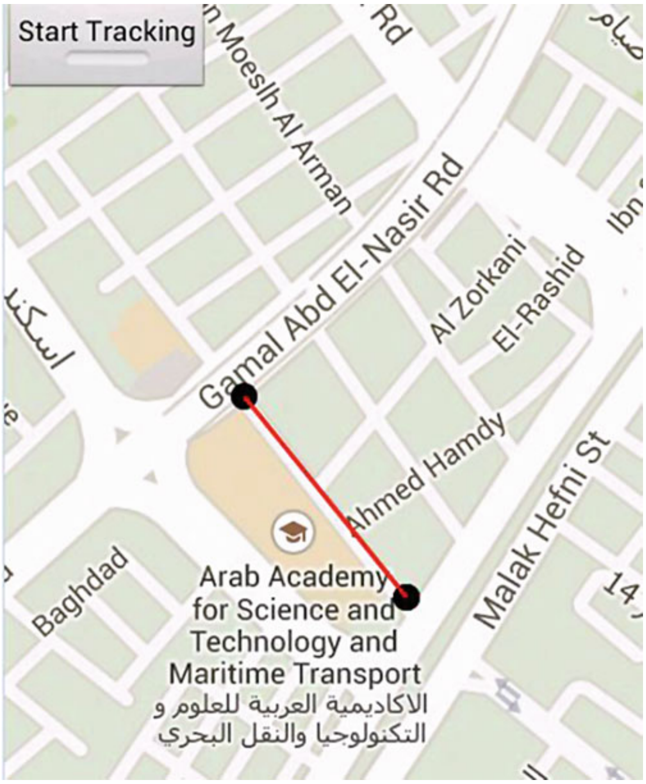

Fig. 4 Route path of one vehicle (street view)

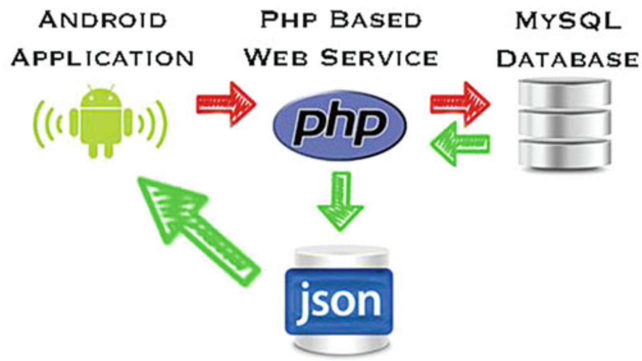

Fig. 5 Android—MySQL communication process

To generate these files, one can do the same steps of creating kml file, draw the desired routes on Google earth, save it in kml format and then convert this kml file format to gpx file using gpsbabel tool (www.gpsbabel.org). Note that, Android emulator uses GPX version 1.1 format data (Version 1.0 is not supported).

For our case of tracking two (or more) trains, we will have more than gpx file (one for each train) and need to add a point for each train according to its gpx file. To work with all these files we need to organize them and save their points on a database. This can be performed using a WAMP server.

WAMP is acronym for Windows, Apache, MySQL and PHP [12, 13]. WAMP software is one click installer which creates an environment for developing PHP, MySQL web application. By installing this software you will be installing **Apache**, **MySQL** and **PHP**.

The Android application and a remote MySQL database cannot communicate with each other directly without interpreter. This interpreter will be PHP. The process in a visual form would look like this (Fig. 5):

To accomplish our developed Route Tracking application we must go through these steps:

1. Creating MySQL database containing one tables with fields (point id, point latitude, point longitude, train number, and point time).
2. Connecting to MySQL database using PHP. Using PHP statements to export the table data into an XML format that map can retrieve through asynchronous JavaScript calls.
3. Adding a row in MySQL database corresponding to a new point, storing its latitude, longitude, train number and time of point generation.
4. Reading all rows from MySQL database to display all points on the map.
5. Creating the Android application that connect all these components together.

Simulation Results

In this paper, we simulated a crossing scenario of two trains. A hypothetic route is chosen for each train (a route of four points or stations). The present Android application is a modified version of Route Tracker application discussed in section “Google Maps Application”. In each train we run the ordinary Route Tracker application, save points to MySQL database. The map shows the route path until other train adds another point in the database. At this point, the map changes (using intent) to another view showing points of both trains. After a sufficient period of time (permits the driver to understand other train situation) the map returns again to its original view (original train route path).

Another map is shown on the central room. It is a webview activity, its source is a PHP file that reads all points from database and locates them on a Google Map distinguishing between train one and train two points by using different markers for each train. It always shows both trains routes.

The distance between trains is calculated using information about the trains’ points (latitude and longitude) which are read from database. We use the Haversine form to calculate the distance from points coordinates (Figs. 6, 7, 8, and 9).

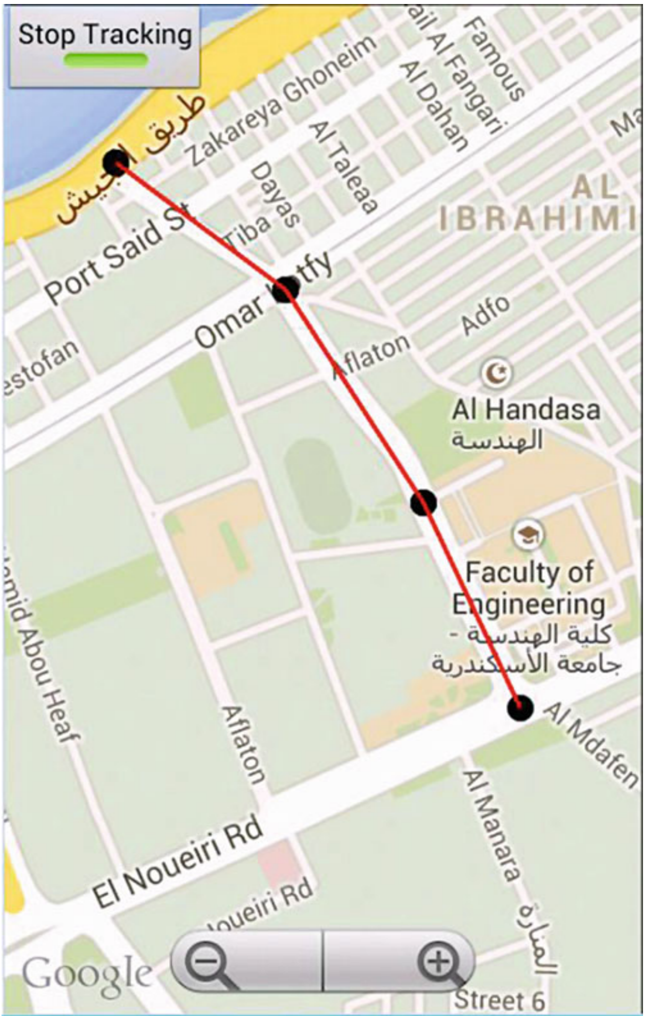

Fig. 6 Route path of train 1 (Map 1 view)

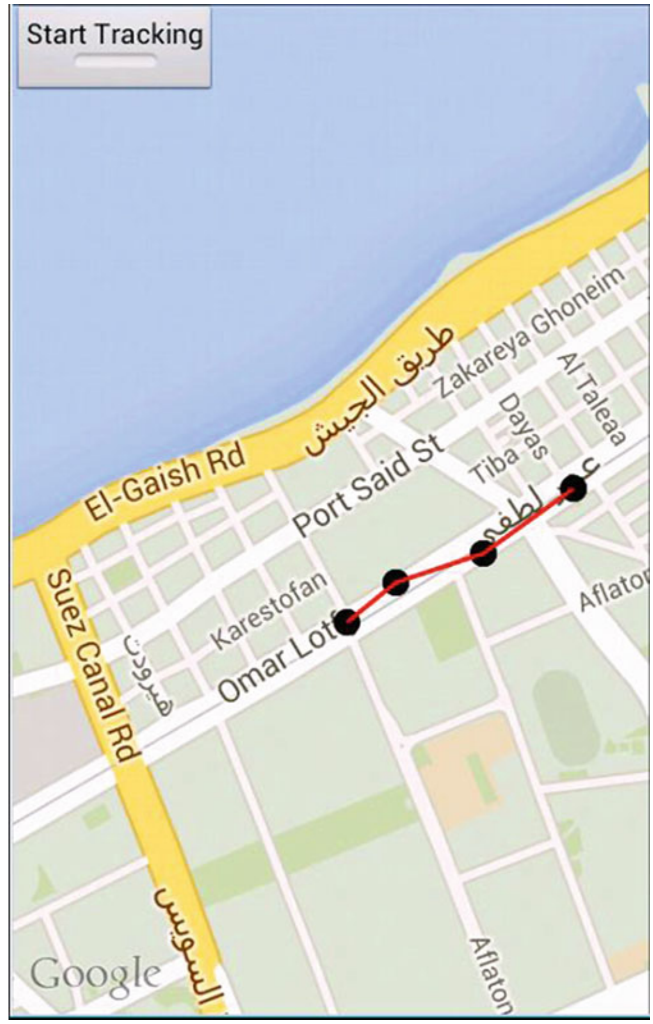

Fig. 7 Route path of train 2 (Map 2 view)

We put a distance of 5 km as a threshold. If the distance between the two trains becomes less than 5 km, an alert is generated (as shown in Fig. 8). The distance between trains is calculated periodically with each point update, and the check of 5 km is also repeated until all points are read from database.

Conclusion

In this paper, an application is developed that aims to participating in avoiding or decreasing the rate of collision between moving vehicles (in our simulation we worked on trains because the number of collision between trains is very large indeed). To achieve our goal, we start with Google Maps Application that, simply, displays a google map on Android phone, then, zoom controls are added to the map.

After that, we add markers for particular locations, and draw a line connecting the points drawing on the map. Then we move to Route Tracker Application that is already developed. We Display a map showing the route path of each train on a smartphone located in the train. This map is updated with points of the other train periodically. As Route Tracker Application is developed for tracking path for only one train, we need to modify this application to be suitable for tracking two or more trains.

To do so, we save the points of each train route in MySQL Database to construct a map showing routes of all trains (by locating points of all trains on this map). This map is displayed on a central (control) room.

If the two trains become close to each other (we took a distance of 5 km as a threshold), the smartphone generates an alarm to warn the drivers. The inspector on the control room must take an action based on this alarm. He must inform both drivers to adjust their speed to avoid collision.

Using this technique, we can decrease the collision rate between trains and improve its safety factor.

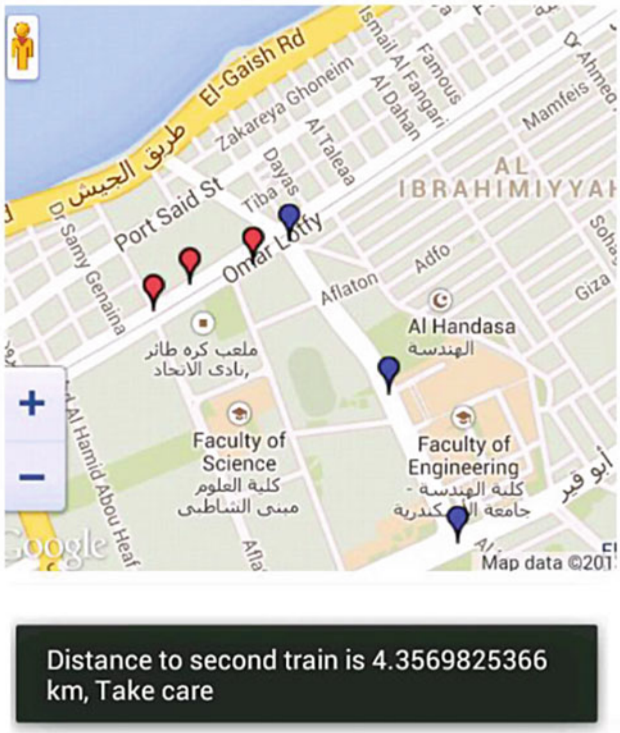

Fig. 8 Alarm generated when distance is less than 5 km, action is required (Control Room Map)

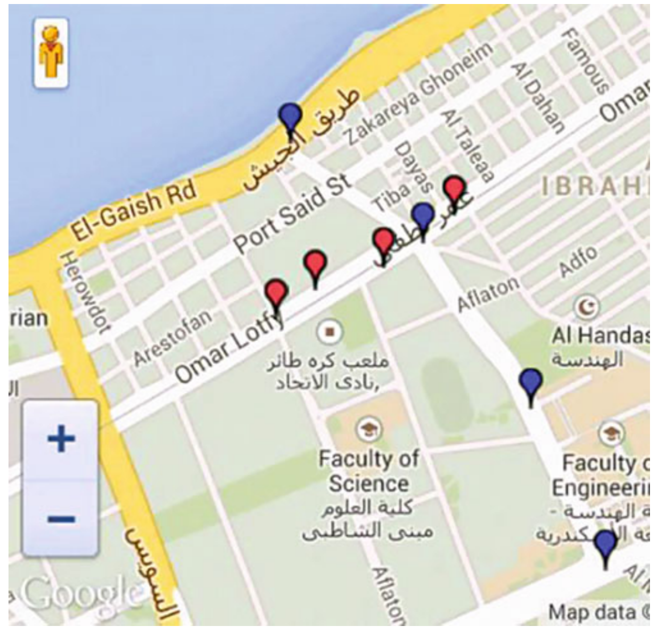

Fig. 9 Action is taken, each train completes its path (Control Room Map)

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