Evaluation of SIP Call Setup Delay for VoIP in IMS

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Abstract. IP Mulimedia Subsystem (IMS) is an architecture that can provide innovative solutions for multimedia services deployment regardless the type and network topology. New delivered value added services require significant changes to the multi-services IP network design, including the activation of multiple functions such as quality of service (QoS) capabilities, security mechanisms, multicast routing, etc. To ensure that services delivery meets the high expectations of end users, factors affecting QoS or user's quality of experience (QoE) must be properly considered. This paper will give a study of a the 'SIP Call Setup Delay' metric which will serve us to accomplish the QoE measurement in the case of VoIP sessions carried over IPv4 and IPv6 IMS networks.

Keywords: QoE, QoS, SIP, IMS, Call Setup Delay, IPv4, IPv6.

1 Introduction

The convergence phenomenon, modeled by the IMS (IP Multimedia Subsystem) architecture that started some years ago and that is accelerating will gradually remove the traditional boundaries between fixed and mobile communication systems. The success of IMS depends on satisfying the high expectations of end users. This led to the birth of the Quality of Experience (QoE). QoE is different from QoS which focuses on measuring performance from a network perspective. QoE is the term used to describe user perceptions of the service performance. On the other side, QoS is the ability of the network to provide a service at an assured service level.

The paper is organized as follows: we present the main components of the IMS architecture in Section 2. Then, we will introduce the concept of QoE and its correlation with QoS in section 3. In Section 4, we present our IMS testbed based on the "Open IMS Core". Section 5 describes our QoE evaluation of SIP Call Setup Delay and results. Finally, we present some conclusions in Section 6.

2 IMS Architecture

The IP multimedia subsystem (IMS) is rapidly becoming the de facto standard for real-time multimedia communications services. IMS standardization defines

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open interfaces for session management, access control, mobility management, service control, and billing. This enables service providers to offer Session Initiation Protocol (SIP) communication services with more features and more flexibility than legacy services provide by circuit switched networks.

Fig. 1 presents a general overview of the IMS architecture [1] where one of its main characteristics is the separation between its different layers. Depending on the environment where IMS is being deployed, there are several access network alternatives like, for instance, UMTS (Universal Mobile Telecommunications System). The transport layer in IMS is in charge of providing IP connectivity to terminals, and allowing signaling and media exchange. The control layer constitutes the core of the IMS, and it is in charge of providing session control through call routing and policy enforcement. Finally, the service layer provides multimedia services to the overall IMS network.

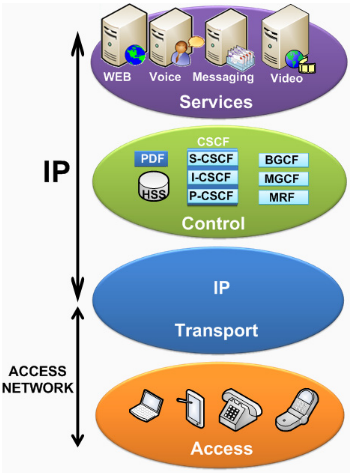

Fig. 1. IMS architecture overview

3 Introduction to QoE

3.1 Defining QoE

There are different definitions of QoE across current ETSI, ITU and other literature. ETSI TR 102 643 defines Quality of Experience as:"*A measure of user performance based on both objective and subjective psychological measures of using an Information and Communication Technologies service or product*" [2]. QoE takes into account technical parameters (e.g. QoS) and usage context variables (e.g. communication task) and measures both the process and outcomes of communication (e.g. user effectiveness, efficiency, satisfaction and enjoyment). Objective psychological measures do not rely on the opinion of the user (e.g. task completion time measured in seconds). While Subjective psychological measures are based on the user opinion (e.g. the perceived quality of a medium).

QoE is a concept comprising all elements of a subscriber's perception of the network and performance relative to expectations. The following table describes elements of user experience in the case of telephony service and the level of quality expectations for those elements [3].

Table 1. Elements of user experience for telephony services

Element of User's Experience	*Expectations for Level of Quality*
Reliability	Works every time
Availability	Always available
Call Completion	Completed successfully
Connect Latency	Rings in seconds
Voice Quality	Good as the PSTN
Speech Latency	Imperceptible
Services	All functioning properly
Billing	Completely accurate

3.2 QoE and QoS

As presented in Fig. 2, QoS is a performance measure at the packet level from a network point of view while QoE is the overall network performance from the user point of view. QoE is a measure of end-to-end service performance from the user perspective. For instance, QoE focuses on user-perceived effects [4], such as Call completion rate or Call setup time, whereas QoS focuses on network effects such as end-to-end delays, jitter or Packet loss.

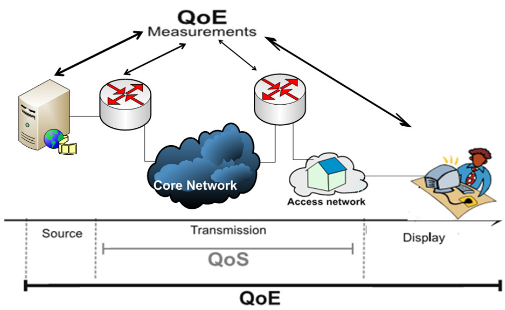

Fig. 2. QoE and QoS

3.3 Related Works

Some researches and development have been done in the field of QoE. Their main purposes can be classified in four categories: network planning, service and QoS provisioning, QoE and QoS monitoring and optimization. Four main approaches are used for QoE evaluation, which are:

1. QoE as an extension of QoS which use QoS perception models for network and service.
2. Objective cognitive schemas which use Human-Computer Interaction.
3. Subjective User Studies based on Marketing, and Business Models.
4. Subjective and Objective QoE which is an ITU-T Approach used for multi-media service and products.

4 IMS Testbed

4.1 Testbed Overview

The testbed is built based on the "Open IMS Core" [5] which is an Open Source implementation of IMS Call Session Control Functions (CSCFs) and a lightweight Home Subscriber Server (HSS). The figure below provides an overview of our experimental testbed.

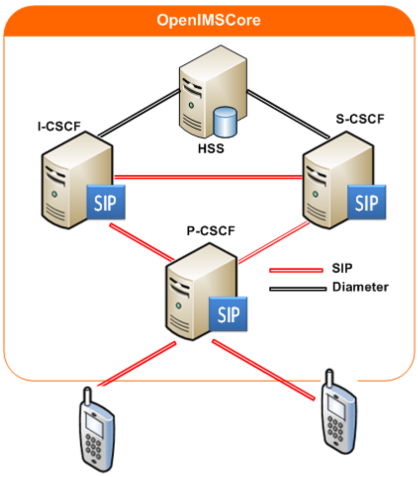

Fig. 3. Testbed Architecture

4.2 IMS Call Flow

The Fig. 4 depicts the SIP messages that are exchanged during VoIP Signaling at the IMS Access Network level. The signaling flows for call setup starts with a SIP invite request and finish with a 200 OK SIP response.

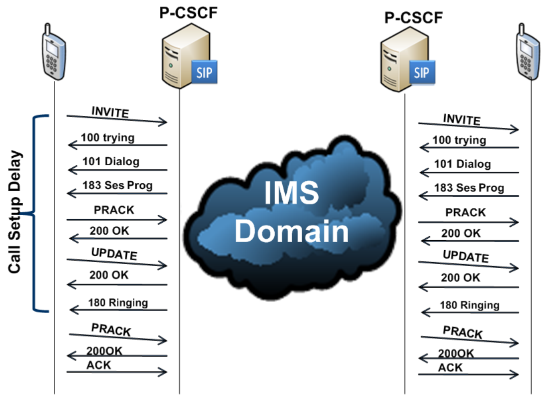

Fig. 4. Signaling flows for Call Setup in Access Network of IMS

5 Call Setup Delay Evaluation in IMS Environment

5.1 Definition of Call Setup Delay

The Call Setup Delay also known as the Post Dial Delay (PDD) is defined as the elapsed time between sending the initial INVITE request and receiving a 180 RINGING response. It is considered as one of the required parameters for QoE evaluation. The ITU-T recommendation defines the mean value of Call Setup Delay being equal to 800ms and the maximum value being equal to 1500 ms [6].

5.2 Theoretical Analysis of SIP Call Setup Delay

Theoretically, when requesting a VoIP services, in IMS the Call Setup Delay is defined as the summation of the Serialization delay (Dsip), Propagation delay (Dp) and Queuing delay (Dq) for exchanged SIP signaling messages during the call setup phase.

$$Dsip = Ds + Dp + Dq \ . \tag{1}$$

Where, the Serialization Delay is defined as the time it takes to send all the bits of a SIP message to the physical medium for transmission across the physical layer. The Propagation Delay is the time it takes for a SIP message bit to cross the physical link from end to end. Finally, the Queuing Delay is defined as the time the SIP message spends in the queue of the system.

In our SIP Call setup Delay's evaluation, we have only considered serialization delay, and we focused on the case where SIP signaling messages are exchanged via a Radio Access Network. The serialization delay formula is then defined as:

$$Ds = \frac{Msip}{Rlinq} \ . \tag{2}$$

Where, Msip is the length of SIP messages, and Rlink is transmission bit rate of Access link.

5.3 Numerical Results

For QoE evaluation, we need to define packet sizes for different SIP methods and responses. The table below lists the size for each SIP message established from packets captured by using Wireshark [7] in our experimental test bed. Also the number of exchanged messages, via the Radio Access Network for each SIP message is presented.

Table 2. Size and number of SIP messages

SIP message	*Numbers*	*Size(bytes)*
INVITE	2	1418
100 trying	2	555
101 Dialog	2	728
183 Session	2	1249
PRACK	2	1068
200 OK	2	451
200 OK with Session	2	917
UPDATE	2	1008
180 Ringing	2	740

Before transmitting SIP message, each message is encapsulated within a UDP datagram. This datagram is then encapsulated within an IPv4 or IPv6 packet. For this, the IPv4 layer adds 20 bytes header, and IPv6 adds 40 bytes header while UDP protocol adds an 8 byte header. Thus, table 3 shows the size of the SIP message including the UDP/IPv4 in the case VoIP calls over IPv4 and UDP/IPv6 header in the case VoIP calls over IPv6.

Table 3. Size of SIP messages including the UDP/IPv4 and UDP/IPv6 header

SIP message	*Size including UDP/IP4 header*	*Size including UDP/IP6 Header*
INVITE	1446	1466
100 trying	583	603
101 Dialog	756	776
183 Session	1277	1297
PRACK	1096	1116
200 OK	479	499
200 OK with Session	945	965
UPDATE	1036	1056
180 Ringing	768	788

By using equation number (2) and input data of table 3, table 4 shows the approximate Call Setup Delay for different values of bit rates in the case of VoIP calls transported over IPv4 and IPv6. Thus, for 9,6kbps bit rate the Call Setup

Delay is 13977 ms in the case VoIP calls over IPv4 and 14277 ms in the case VoIP calls over IPv6, while for 384kbps data speeds the Call Setup Delay value is then reduced to 349ms in the case VoIP calls over IPv4 and to 357ms in the case VoIP calls over IPv6.

Table 4. Call Setup Delay in IPv4 network versus IPv6 network

$Speed(kbps)$	$CSD\ IPv4(ms)$	$CSD\ IPv6(ms)$
9,6	13977	14277
14,4	9318	9518
56	2396	2447
128	1048	1071
220	610	623
260	516	527
384	349	357
1000	134	137

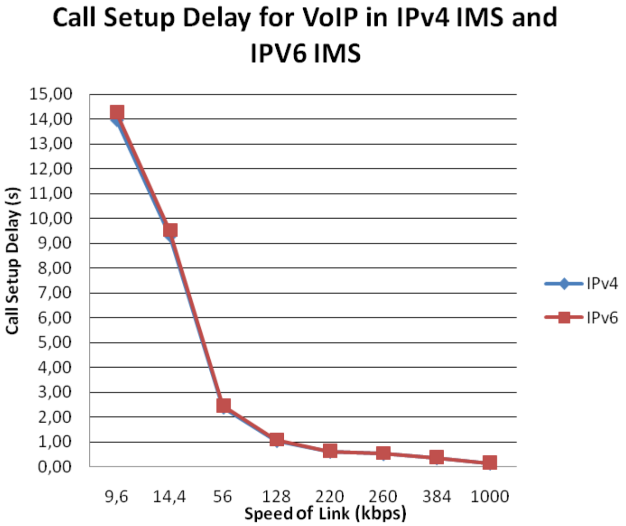

Fig. 5. Call Setup Delay for VoIP in IPv4 IMS and IPv6 IMS

Table 4 shows numerical results of the Call Setup Delay values depending on the version of the used IP protocol, the size of exchanged SIP messages, the number of these messages, and the bit rate of the wireless access link. By using results of table 4, Fig. 5 shows that the Call Setup Delay is conform to the ITU-T recommendation only for radio access link with a minimal bit rate equal to 128 kbps. Also, it is important to note that, for radio access links with a bit rate less than 128kbps, the obtained Call Setup Delay values when using IPv6 are

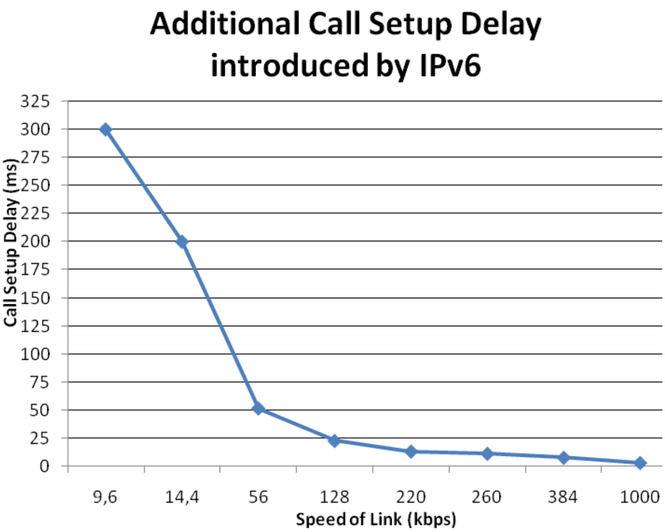

Fig. 6. Additional Call Setup Delay introduced by IPv6

greater than those obtained for IPv4. Thus, as shown in Fig. 6 from 128 kbps data speed, we can make VoIP calls over IPv6 in IMS without impacting the Call Setup Delay.

According to our results, we can propose various solutions in order to reduce SIP Call Setup Delay value. Our main propositions are:

- Compression of SIP messages using SigComp mechanism. This mechanism reduces the size of SIP messages and therefore contributes to the reduction of the Call Setup Delay.

- Optimization of SIP signaling flow in order to use a fewer number of SIP messages when establishing the VoIP session.

- Header compression contributes to the reduction of the total size of transmitted headers for each SIP message. The original headers should be decompressed at the reception end point.

6 Conclusion

QoE assessment is not an easy task. Indeed, their evaluation should take various parameters into consideration. The main challenges are development and standardization of objective and subjective QoE metrics.

In this paper, we presented an analytic model to evaluate the SIP Call Setup Delay for VoIP in IMS, we have measured SIP Call Setup Delay in IPv4 network and compared them with SIP Calls Setup Delays in the case of an IPv6 encapsulation. The results show that for radio access links with a bit rate less than 128kbps, the obtained Call Setup Delay values when using IPv6 are greater than those obtained for IPv4. Also, the Call Setup Delay is conform to the ITU-T recommendation only for radio access link with a minimal bit rate equal to 128 kbps.

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