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# Quality of service management for IPTV services support in VANETs: a performance evaluation study

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**Abstract** In order to deliver a qualitative Internet Protocol Television (IPTV) service over vehicular ad hoc networks (VANETs), a quality of service (QoS) mechanism is needed to manage the allocate of network resources to the diverse IPTV application traffic demands. Unlike other mobile network, VANETs have certain unique characteristic that presents several difficulties in providing an effective QoS. Similarly, IPTV requires a constant stream for QoS which at the moment is quite difficult due to the inherent VANET characteristics. To provide an effective QoS that will meet the IPTV application service demands, VANETs, must satisfy the compelling real-time traffic streaming QoS requirement (i.e., minimum bandwidth allocation, packet loss and jitter). In this report, we evaluate via simulation the feasibility of deploying quality IPTV services over VANETs, by characterizing the association between the IPTV streaming quality determining factors (i.e., throughput, delay, loss, jitter) and the IPTV quality degradation, with respect to node density and node velocity. Furthermore, we used an objective QoS metric (Media-Delivery-Index) to identify, locate and address the loss or out-of-order packet. We outline how, using these information's can support in shaping network parameters to optimize service flows. The implementation assures a priority for handling IPTV traffic, such that maximise the usage of VANETs resources, and opens the possibility that loss and delay can be minimised to a degree that could guarantee quality IPTV service delivery among vehicle in a vehicular network system.

**Keywords** MDI · IPTV · QoS · VANETs

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## 1 Introduction

In today's world, the rapid increase in the number of vehicles along with overloaded transportation infrastructure has lead to the unavoidable traffic congestion, road accidents and transportation delays we are experiencing day in day out. Many approaches have been proposed as an answer to all of these mentioned problems, advances such as construction of better roads and highways, infrastructure as well as applying safety applications, but so far it seems as if nothing has been done. The demand to minimize or, if possible, get rid of these problems (i.e., road traffic congestion and accidents) has contributed to the development of an intelligent vehicular network technology known as vehicular ad hoc networks (VANETs) [1]. VANETs is a technology that establishes a network from vehicles with embedded wireless equipment; it's a singular case of Mobile Ad-Hoc Networks (MANETs). VANETs differ from Infrastructure based networks such as cellular networks in the demanded equipment to form a transportable network. It is believed that if properly deploy VANETs will go a long way to minimise road traffic congestion and accident problems being faced daily on our roads. VANETs itself is a network and it's made up of roadside infrastructure and vehicles. It uses the wireless technology such as 802.11p wireless standards, General Packet Radio Services (GPRS), along with Dedicated Short-Range

Communications (DSRC) to interconnect with surrounding vehicles known as vehicle-to-vehicle network (V2V) or to communicate with a roadside infrastructure, such as a vehicle with neighbouring infrastructure or stationary equipment known as vehicle-to-infrastructure (V2I) [2]. Based on VANETs technology, a large number of safeties related applications and non safety application have been implemented. Applications such as cooperate collision warning, traffic signal violation warning and many more [3–5]. Another kind of application that is also important for VANET successful deployment is infotainment [6]. Infotainment services offer information and entertainment for both drivers and passengers, e.g., Internet Protocol Television (IPTV), internet access, multiplayer games, multimedia applications, etc. Variety of relevant data such as weather information, tourist information, gas prices and parking space information can also be spread using this same procedure. These services can provide limitless opportunities for vehicle internet applications to make the driver and passengers experience more pleasurable.

One major challenges face by VANET now is the issue of quality of service (QoS) and its operation, unlike other mobile network VANET has certain unique characteristic that presents several difficulties in providing an effective QoS. Consequently, IPTV requires a constant stream for QoS which at the moment is quite difficult due to the inherent VANET characteristics. QoS is the measure of service availability of a network and its transmission quality [7]. The availability of service in a network environment is a crucial fundamental element of QoS. In order to provide a minimal level of IPTV quality to the final user, VANETs must satisfy the compelling QoS requirement of real-time traffic streaming. The transmission quality of IPTV network is defined by four major factors namely minimum bandwidth, packet loss, jitter and delay. For VANET to deliver better quality IPTV services, the network has to conform to certain demands for bandwidth, delay and jitter [8]. Unlike none real-time-traffic applications, IPTV due to its real-time delivery nature, requires huge bandwidth and strict performance service quality, any variation in the packet arrival time can generate great numbers of unexpected error which can conduct to miserable or no service delivery. IPTV application can not tolerate bandwidth fluctuations; therefore guarantee QoS is indispensable required. There are numerous metrics defined in the literature for guaranteeing QoS to multimedia traffic streaming. Some good examples are the Mean Opinion Scores (MOS), V-Factor, where the focus is either on the subjective observation of video quality based on human perception or on buffering techniques to suit the network jitter. However, in this project we adopted an objective method Media Delivery Index (MDI) as our QoS parameter measurement metric in place of the commonly used subjective method. So, the aim of this research is to experiment IPTV services over VANETs by evaluating these QoS parameters through simulation, to assure a priority traffic handling for bandwidth allocation, jitter, and delay such that will reduce the percentage of packet loss that will assure a quality IPTV deployment over VANETs.

### 1.1 Novelty of this paper

Our work in this paper is limited to a simulation based study. In contrast to previous studies, our objective in this study does not include proposing a new protocol. But rather, we intend to demonstrate through discrete event simulation how QoS can be undertaken to support real-time multimedia traffic such as IPTV over an arbitrarily network such as VANETs, by efficient utilization of the existing protocols. In this context our work could be seen as an extension of the work done by Zhou et al. [9]. While their work is based on analytical modelling of VANETs connectivity to support media service distribution, our study is based on a discrete simulation study of the QoS parameter that will support the efficient distribution of such media services. Thus, this paper could be viewed as a complement to their work with the following contributions:

- By applying the delay factor (DF) component of the MDI scheme, we demonstrated that MDI:DF is an effective metric that could be utilized to adequately monitor and fix VANETs intrinsic QoS problems. Consequently, adopting the MDI scheme, we formulate a method and heuristic for adequate buffer memory size configuration. Considering VANETs characteristics as regard to its frequent link instability. In order to reconnect the broken links, vehicle need to catch up with another vehicle ahead, and this takes time. During the catch up period, the routing protocols employ a strategy known as Cary-and-forwarding, a situation where node forwarding packets hold the packet for a next optimum hop yet to be available [10], by buffering the packets and transmit it in a later available opportunity. However, the buffer memory needs to be sized accordingly. For example; a small sized buffer may result in frequent loss of packets, when there is network congestion. Consequently, a relay node may experience frequent buffer overflow when there is no reachable node to convey packets being sent to it from the uplink node. Furthermore, large buffer could also affect the quality of the IPTV traffic, as packets may miss the delivery deadline due to large buffer size. As such, the network buffers need to be sized accordingly to be able to accommodate these inconsistent VANETs situations. Using the DF component of the MDI scheme one could deduce the length of time that will be required to buffer a stream packet to avoid packet loss.

- Through extensive simulation in NS-2, we evaluate the feasibility of the underlying VANETs in terms of the network's ability to deliver qualitative IPTV traffic. Where we characterized the connection between throughput, delay, loss and the IPTV traffic degradation as a function of node density and node speed. Most VANETs simulation results show that the time by which VANETs communication channels stay connected is limited, this means that nodes connectivity experience frequent interruption. In spite of these evidences (i.e., the uncertain events that randomly interrupt the node connectivity and alter their distribution in VANETs); most previous works in VANETs real-time multimedia traffic QoS assessment ignored these phenomena. However, this phenomenon (i.e., VANETs characteristics in terms of its link instability due to node density and node speed) has a significant effect on the quality of multimedia traffic across the network.

## 2 Background and related literature

The importance of getting an understanding of IPTV QoS over wireless network has been highlighted in various recent works. Previous work in [11] has shown that wireless error rate is higher when compared to its wired counterpart due to fluctuation channel conditions, for this known fact, the use of wireless streaming technic is usually not straightforward as in used in wired networks. Different methods of streaming have been recommended to solve the wireless streaming condition. Meanwhile, another work in [12] shows in their observation that when streaming multimedia content, the wireless network is usually underutilized. And so they proposed a transmission arrangement that explores using available wireless bandwidth efficient when conveying non-scalable multi-media data streaming. They observed that the use of any TCP or TCP-friendly rate control (TFRC) connection fail to fully utilize the available bandwidth and so they suggested the use of multiple TFRC connection to a known wireless application streaming. Their result shows that bandwidth is efficiently utilized if more than one connection is used. However, the changed dynamically in the number of connections may alter the quality of video streaming. A seamless cross-layer interworking between broadcasting network and 802.11 WLAN network for delivery adaptive and interactive mobile TV was proposed in [13]. They evaluate the packet losses and the TV quality perceived services using an experimental test-bed, the test-bed result, well, validated their proposed algorithm, but however, the test-bed base performance realization consumed allot of time and the setup is too complex. Study in [14] investigates the performance of IPTV loss packet that can be caused by overflow buffer in various scenarios of home networks which include constant rate data wired link, multi-hop wireless path and variable data rate for single-hop wireless link. An on–off traffic sources was used for the arrival process to characterize the video source and for the service process, an exponential distribution was presumed for service time distribution for the examined data link. However, using on–off arrival model for exponential services time and video source distribution for the protocol behaviour of complex network like IEE 802.11 MAC protocols considerably limits the feasibility of the analytical results. Study in [15] proposed an effective model to study numerical QoS assurances, based on efficient bandwidth and delay-bound violation possibility for multi-layer audiovisual conveyance across a diminishing wireless channels, however only the separate queues manages the multi-layer video arrival process in their evaluation and so fail to justification the statistical features of the video traffic. Study in [16] proposed an extended two-dimensional Markov-chain model for IEEE 802 multi-hop wireless network, throughput analyzing and taking into consideration the error-prone channels, non-persistent traffic, post-back off stage and finite retry limit. The analysis of the throughput, delivers the upper-bound throughput achievement of the streaming audiovisual. However, traffic that was measured lacks a description of the characteristics of video because it is generally non-persistent traffic. The process of IPTV streaming over VANET has two phases as studied in [17]. In the first state, a video trigger is sent to the interested area or a vehicle to send through another vehicle, and in phase two video streaming data is transmitted back from the source, so they proposed a V3 architecture for live traffic video streaming over VANET where they evaluate the video forwarding trigger and the streaming video. But however due to their inability to use in their simulation a real video streaming data and the fact that only the delay aspect was reported in their experiment, end-to-end delay alone cannot reflect the achievement of the streaming video directly, because in numerous situations, delay variation could be consumed by the receiver buffer. QoS IPTV guaranteed service mechanism was proposed in [18], where they proposed a connection admission controlled according to bandwidth remains. A connection will be provided if the available bandwidth is enough to allocate a new flow and once the connection is established, the service might be guaranteed certainly. Both fall short because the policy is not suitable in a situation where there are several classes of traffics with different level of QoS such as IPTV because it guaranteed only the total bandwidth related throughput and ignored the statistical characteristics of the video traffic. Although there are many metrics defined for providing QoS for audio video traffic streaming over VANET such as the one proposed in, [19] where they concentrate on monitoring video quality based on human perception (human judgment) subjective, however the scheme lack monitoring as well as solution to fixed jitter, delay nor loss problem in a network and these are the major parameter that call the shots as to whether a network can convey good quality IPTV services or not. However MDI measurement metric as proposed in [20] is different, MDI tell what is wrong with a streaming multimedia, where the problem occurred or is occurring and also provide detailed information on how to go about fixing such problem.

### 2.1 Overview of vehicular ad hoc networks and its characteristic

VANETs is a network technology that makes use of moving cars as nodes connected such that the vehicles involve can receive and convey messages to other vehicles in the network via radio frequency. However, this kind of network is temporary in the sense that vehicles that are involved in the network formation are only connected for a short period of time, this is due to vehicle movement which is characterized by high node mobility that result in frequent changes in the network topology. There are two most prominent communication in VANETs, Vehicle to Vehicle communication (V2V) and Vehicle to Infrastructure (V2I). In the V2V communication, vehicles interact with each other directly or through other vehicles in a multi hop fashion. Similarly, communication in the V2I is done through the support of roadside unit (RSU) infrastructure, the combination of these two modes of VANETs communication is regarded as the hybrid VANETs communication. A typical hybrid VANETs topology is shown in Fig. 1. VANETs differs from other Mobile Ad Hoc Networks based on some unique characteristics, these characteristics pose a real challenge in providing an effective QoS to meet application service requirements. Some of these characteristics are stated as follows [21, 22]:

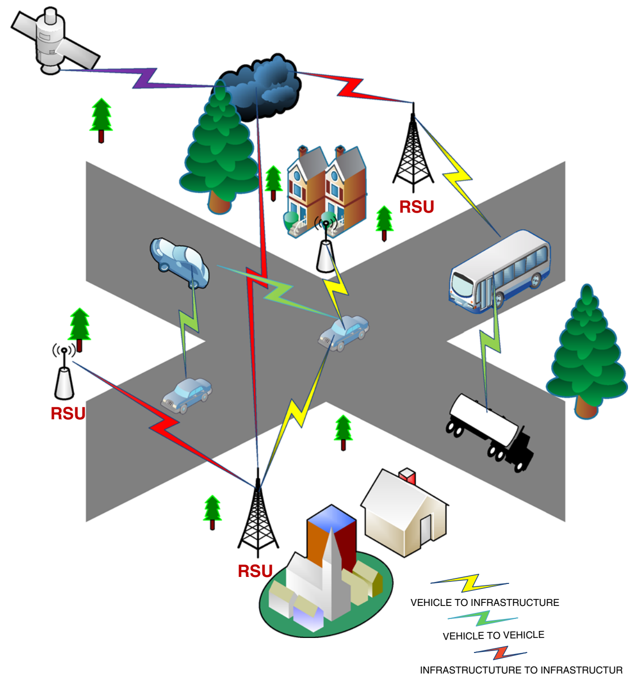

**Fig. 1** VANET scenario

- *High dynamic topology:* VANETs is characterized with a high dynamic topology change as a result of the vehicle (node) high speed.
- *Frequent disconnecting network:* The dynamic nature of the vehicular network results to continuing topology change, and as such leads to frequent network disconnection. Especially in situations of low vehicle density the probability of network disconnection is high.
- *Mobility modelling and predictions:* Mobility model plays a vital role in VANETs network protocol design, considering the high node mobility and the dynamic changes in the network topology.
- *Energy and storage capacity:* Unlike mobile ad hoc network that is constrained by limited power, VANETs is characterized with abundant energy which usually comes from the vehicle battery and also possessed high storage and processing capability.
- *Strong delay constraints:* The safety conditions like accidents, brake event of vehicular application calls for on time message delivery to appropriate nodes. Thus, high data rates issue in VANET is less relevant compared to strong delay constraint issues.
- *Interaction with on-board sensor:* It is usually presumed that nodes in VANETs are equipped with on-board sensor that can provide useful information to form a communication link and for use in the routing determination.
- *Numerous communication environments:* There are usually two modes of communication environment in VANETs: the highway traffic scenario and the city traffic scenario. The later (city scenario) is said to be much more complex, while the highway scenario is considered to be much more straight forward and less complex as it is constrained with one-dimensional movement.

### 2.2 IPTV technology

IPTV technology is one of the rapidly growing technologies aimed at facilitating access to audio video entertainment. It provides access for delivering broadcast digital TV and other interactive multimedia services over a secure internet protocol medium of transport from head-end device to the user's computer or TV set. IPTV uses an internet protocol over broadband connection to provide services along side with the internet connection of the user or subscriber. This is made possible by using the same network root but over a dedicated bandwidth allocation. Thus IPTV can be define as a system in which digital television services are rendered to different subscriber across a broadband connection adopting the internet protocol [23]. The term IPTV has a broad meaning and can be split into two parts, one part is the IP and the second part is TV.

- IP is an acronym for Internet protocol; it specifies packet formats and addressing scheme for a network. IP is a common protocol that provides data transmitting mechanism for managing flows of packet between end-to-end devices connected to the Internet. In IPTV system an audio video signal is divided into multiple IP packets in order to be successfully transmitted across the IP network [23].
- TV (television) this specifies the communication medium that operates through the transmission of audio visual content [23].

The combination of these two components specifies the medium of communication of video and sound through an IP network.

#### 2.2.1 IPTV packetization

Generally speaking, in order to transmit any multimedia data over any IP network, the data need to be encoded, compressed and packetized. The process of transforming audio–visual data into packet according to a specific protocol is known as Packetization. The Packetization specifies the format and method by which audio and video content can be sent across an IP network. IPTV signal is a series of images made up of frames, which are divided into multiple IP packet fragments and transmitted at a rate of 15–60 frames per seconds over an IP network. In order to transmit any frame over an IP network, the frame needs to be compressed, break into small segment known as transport units, each transport unit is encapsulated in a transport packet (RTP) and then into UDP and finally into an IP packet before being transmitted across an IP network. Figure 2 is a typical illustration of an IP video packet sent over an IP network. However, since our interest in this paper lies on what happens to the IPTV packet as they travel across the network, we hereby refer the readers to [24], for more in-depth information on IPTV encoding and packetization.

#### 2.2.2 Iptv streaming over wireless network

Wireless IPTV is aimed at making an extension to IPTV and other related services available to the user anywhere, on any device and at any time. This objective requires an advanced wireless network technology, which will support the adaptive nature of networks, services and audio visual content, in order to be able to respond to the needs of users irrespective of their location and their preferences. This task is very challenging due to the fact that wireless network signal by nature are unreliable due to their vulnerability to interference, their limited throughput, high rate of packet loss and end-to-end delay varying [25, 26]. Supporting an application like IPTV makes it even more challenging due to the fact that IPTV is a real time application which requires constant steady flow of information and has a time limit for packet delivery. Figure 3, is a typical illustration of an IPTV system architecture that will support applications such as VoD (Video on Demand) and digital television broadcast [27]. The major components are:

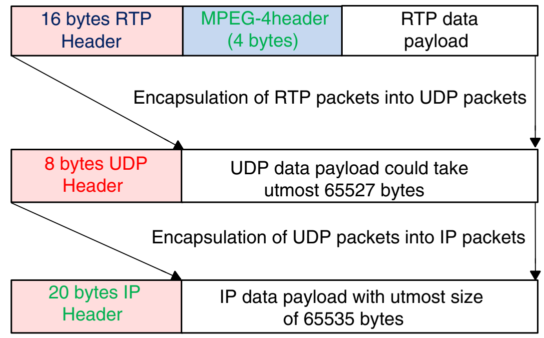


**Fig. 2** Format of encapsulation of RTP packets into UDP/IP packets

- *Super head end* this is where most of the IPTV channels from the national broadcast enter the network. This end receives broadcast TV and on-demand content from different sources and transforms it into forms that can be transmitted over the IP network, this end comprised of the satellites, the encoders, broadcast TV server, subscriber manager, middleware, VOD server etc.
- *Core network* is usually used for transporting the encoded groups of channels to the access network.
- *Access network* is the basic link from IPTV point of distribution (core network) to the user (individual subscriber) Head end
- *User's premises* this is where the broadband network functionality terminates.

#### 2.2.3 IPTV over VANETs architectural design

IPTV technology is one of the rapidly growing technologies aimed at facilitating access to audio video entertainment. It enabled the delivering of broadcast digital Television and other interactive multimedia services across a secure internet protocol medium of conveyance from the

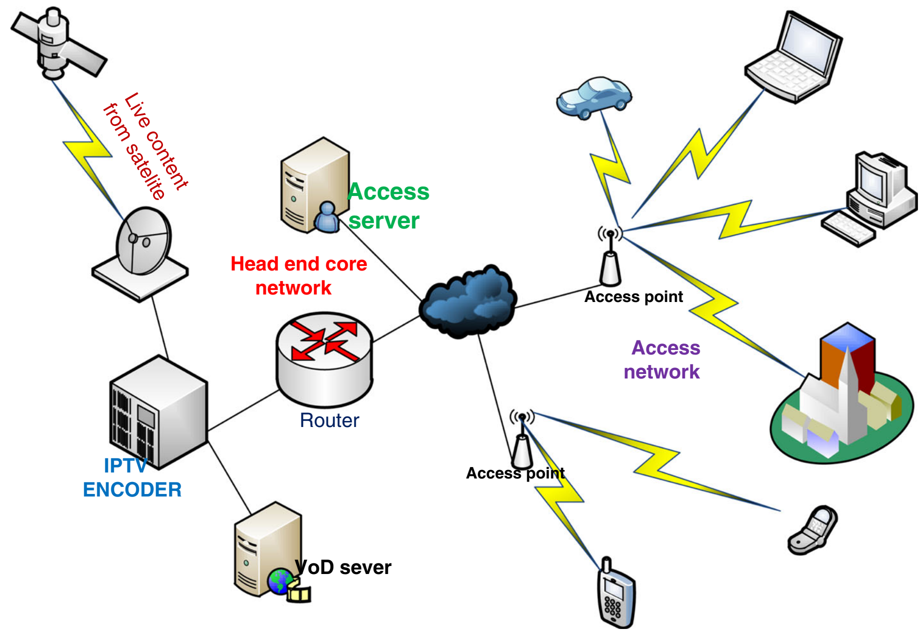


**Fig. 3** Architecture of IPTV over wireless

point of transmission to the user's computer or Television set. IPTV uses an internet protocol over a broadband connection to provide services along side with the internet connection of the user or subscriber. This is made possible by utilizing the same network root, but over a dedicated bandwidth apportionment. Thus IPTV can be defined as a system in which digital television services are rendered to different subscriber across a broadband connection adopting the internet protocol [23].

The architectural design of IPTV services in VANET technology (Fig. 4), is an infrastructure scenario where base stations are deployed over the highway such that their signals overlap to avoid coverage blackout on the road. All the base stations are connected to the Internet Fig. 4. The internet connects the IPTV generator server and a channel changing video on demand server (VOD) which in turn is also connected to the IPTV stream generator server. The original IPTV video as broadcast by the television station or satellite receiver are formatted, encoded and compressed using some sets of H.264 encoder. The encoded, compressed data, then send to the IPTV generator stream server, the VoD server uses a special file format, for example AVI, QuickTime, Real media, etc. to store the data which can later be streamed across the network to a destination vehicle based on demand.

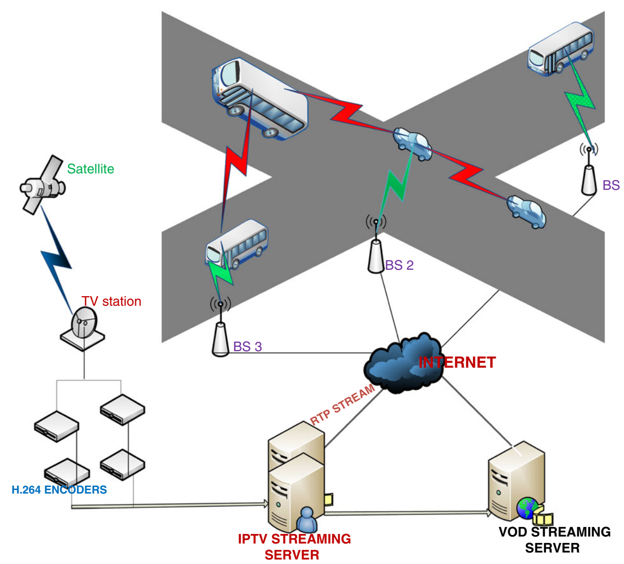


**Fig. 4** Network architecture for IPTV over VANET

## 2.3 IPTV system quality of service

In real-time based streaming applications, such as IPTV, QoS is a very important factor to consider during the deployment of multimedia application. QoS can be described as the measure of service availability of a network and its transmission quality. For a real-time traffic such as IPTV, the ultimate QoS objective is to offer a minimum degree of control over the existing "best-effort" service supported by a common IP network. However, QoS management for digital TV transmission over the Internet is an overwhelming task, due to the lack of data transfer speeds and delivery time guarantees. More also, as more users connect to a service, degradation of service quality is seen in direct result of diminished network capacity. QoS is not only limited to the physical aspects of transmission, as the overall quality of the session is greatly influenced by the encoding parameters used and the error and congestion control. The varieties of user equipment also bring about particular QoS requirements in terms of latency, encoding parameters, processing power and bandwidth. The transmission quality of IPTV service in VANET is determined by three major factors namely (1) minimum packet loss (2) jitter and (3) delay. Packet loss, jitter and delay can affect greatly the quality of streaming video at the receiver's end. Many things can cause any of these three problems, congestion, link error, propagation delay are a few examples of the many causes of video quality downgrading.

#### 2.3.1 Packet loss

As packet travel from source to destination, some packet may not get to their destination successfully. This may be caused by a lot of reasons, ranging from network congestion to packet corruption. Unlike connection-oriented network where loss packet can be resent using Transport Control Protocol (TCP), IPTV like any other real time traffic, is a connectionless application, and as a connectionless-oriented application video is transferred via user datagram protocol (UDP). There is no such guarantee that packet sent from one destination can get to the other end intact and so packet loss is inevitable. For real-time multimedia service such as IPTV, packet can also be loss due to excessive delays and jitter, since any packet that fails to reach the destination on or before the time limit can no longer be played out, as it is considered a loss packet which must be dropped at the destination. Packets may also be lost during transportation, over the carrier network due to corruption or as a result of buffer overflows.

Assuming $p_s$ is the possibility that a packet is successfully transmitted from a sender to a receiver. A perfect protocol would transport every packet successfully ($p_s = 1$), until the traffic load $T_L$ exceeds the capacity $N_c$ of the network channel. The probability of packet loss as a result of the network traffic load $pl(T_L)$, is complementary to the success probability $p_s(T_L)$:

$$pl(T_L) = 1 - p_s(T_L) = 1 - \frac{B_w}{T_L} \tag{1}$$

where $B_w$ is the capacity per flow of vehicular network (i.e., the throughput or the available network bandwidth).

And to determine the packet loss rate of the network, the modest and simplest method to determine packet loss rate is to calculate the total, loss rate which is the ratio of the total number of loss packet to the number of packet drop and it is expressed in Eq. 2 [28].

$$Packet\ loss\ rate = \frac{total\ number\ of\ packets\ drop}{total\ number\ of\ packet\ transmitted} \tag{2}$$

#### 2.3.2 Packet delay

Delay is the duration it takes a packet, sent from a source, to arrive at the receiver's end (destination). In a packet-based network, the path from a point of transmission to the destination point may not be the same as from destination back to the source because as packet are being forwarded from a one end to the other, packet may choose different paths as they travel from source to destination and destination back to source thereby arriving at different times and in many cases out of order. The packet needs to be reordered at the destination end, but the problem now is for a real time traffic like IPTV there is a specific needed time (threshold time) by which these packets must arrive at the destination and any packet that fail to get to the destination at that time is considered a loss packet, and such packet is dropped. If too many packets are dropped as a result of delay, this can strongly affect the quality of IPTV signal because these packets need to be reconstructed back to its original form. If a lot of packets are lost on the way, there would not be enough packets left to be re-constructed and therefore it becomes impossible to regenerate the same video as the one sent from the source. The delay a transmitting packet experiences is determined by the amount of time that the packet has to queue before the channel is accessed, and the number of necessary retransmission until the packet is successfully delivered (every retry will add to the delay that the packet experiences). Let us denote the average time until the channel is accessed as $AC_t$. Then, we can determine the delay experience by a packet at time t, until the transmission is successful (i.e., the transmission delay $T_h$) as:

$$\begin{aligned} T_h &= p_s(\tau + (1 - p_s)) \cdot 2AC_t \cdot + p_s(1 - p_s)^2 \cdot 3AC_t \cdot \\ &\quad + p_s(1 - p_s)^3 \cdot 4AC_t \cdot + \cdots \\ &= AC_t \cdot p_s \cdot \sum_{n=0}^{\infty} (1 - p_s)^n (n + 1) \end{aligned}$$

$$T_h = \frac{AC_t}{p_s} \tag{3}$$

where $p_s$ is the probability that a packet is successfully transmitted and $AC_t$ is the average time taken before the channel is successfully accessed.

However, end to end network delay can be grouped into three: (1) propagation delay, which has to do with the distance between the source and the receiving vehicle. (2) Queuing delay which is a variable delay that depends on the speed of trunk and the condition of the queue. And (3) transmission delay which is the time that is required for all bits of data blocks to be transmitted [29]. This depends on the network size, its bit rate, the rapid changing topology of VANET as a result of it high moving speed can also result to packet delay as links breaks frequently due to hand off. Therefore, the total network delay $D$ is given by Eq. 4 [30].

$$D = T_f + l + \sum (T_h + Q_h + P_h) \tag{4}$$

Therefore, substituting Eq. 3 into Eq. 4, the total network delay will now be:

$$D = T_f + l + \sum \left( \frac{AC_t}{p_s} + Q_h + P_h \right) \tag{5}$$

where D total delay, $T_f$ packet formation time, $l$ look-head delay, $\frac{AC_t}{p_s}$ transmission delay, $Q_h$ queuing delay, $P_h$ propagation delay, h number of hops.

#### 2.3.3 Delay variation

Delay variation or jitter: As packet sent from source to destination can follow different routes to get to their destination, so also is the time they take to get to the destination differ. Packets Delay differs and so the disparity in packet arriving time at the destination is known as packet delay variation. The inter packet Jitter is determined using the formula below Eq. 6 [28].

$$Jitter = \frac{Jitter_1 + Jitter_2 Jitter_3 + \cdots + Jitter_N}{N} (s) \tag{6}$$

### 2.4 IPTV QoS measurement metrics

There are many standard used to asses IPTV quality delivery, such standard like the V-factor, MOS, video quality metric (VQM), peak-signal-to-noise-ratio (PSNR), MDI, moving picture quality metric (MPQM), structural similarity index (SSIM) [31], to mention but a few. No matter the optimization metric used, the touchstone for assessing streaming media quality can be grouped into two methods, subjective method and objective method.

- *Subjective method:* This assessment method is used to evaluate the performance of IPTV base on user perspective. This involves a group of people to watch the video clips and thereafter provide a quality score, which is graded on a scale of 1–5 as recommended by ITU-T in [32] where 5 is rated as the most excellent possible score. Example of this method is MOS. However, in wireless environment, performing a subjective test to try to asses' video quality is expensive in term of time and resources.
- *Objective method:* Objective method present mathematic technique or model that is based on metrics that can be measured objectively and evaluate automatic using a computer program, some good examples of objective approached are PSNR (peak-signal-to-noise-ratio), Structure Similarity Index (SSIM), MDI. In this research, we used the MDI metric because of the detail information it can provide on how well a video is being delivered.

#### 2.4.1 Media Delivery Index (MDI)

The demand for internet to support real time video and television service like IPTV has posed a great challenge for test engineers who major concern is to evaluate network performance and equipment in order to provide a deep understanding of the ability of the internet to handle real time video streaming. The desire to get a measurement metric that will provide information about good delivery, that is, how well a video is being delivered rather than trying to analyze the score and video, we need something that will tell us what to do with the poor score if we are faced with one, something that can provide us information about loss, jitter and delay in a way that they can easily be determined and measure. If video is encoded properly the most important thing to consider is how well is it being delivered across the network to the user, this is the primary concert of QoS in the first place. MDI [33] is a metric that will let us know immediately at any point in the network if there is a problem with delivery and provide detailed information on how different streaming can affect the network, as it values are base on the bit rates of the streaming video. MDI is believed to be an appropriate metric that is capable of given all the necessary information needed to detect all problem that can lead to network delivering real time video streaming applications, because it provide us with the information about the way jitter is affecting the IP delivery, in order for us to take necessary measure that will provide solution to solving such problem. MDI is made up of two parts namely: the Media Loss Rate (MLR) and DF [33], collectively they provide vital insight into what might have happen to any video traffic streaming as it propagate through the network. MDI provide information regarding network nodes and link that may be congested at any time in any part of the network, these result are very vital because it enabled one to easily determined whether the buffer settings on the different devices is capable of providing the required bit rate for successful MPEG transport stream.

#### 2.4.2 Media loss rate (MLR)

MLR is a measurement metric that is used to measure packet loss rate or in some cases packets that are out of sequence because out of packet sequence is also consider as packet loss if the decoder is unable to provide resequencing of such packet. A sign of buffer overflow or packet corruption is denote by the change in MLR from one node to the other [34], loss which might be as a result of network device miss-configuration or network congestion are periodical, and always result to many positive interval of similar MLR magnitude, so therefore, displaying and capturing the MLR maximum value over a measurement period of time will show over the period of such measurement the largest magnitude of upset in an interval, this can give an early tipoff of imminent packet loss. MLR is define in [34] as the number of media packet that are loss or media packet that are out of order per unit time taken the equation for the calculation is shown below in Eq. 7.

$$MLR = \frac{(Expected\ packets - Received\ packets)}{Time\ interval\ in\ seconds} \tag{7}$$

*2.4.2.1 Delay factor (DF)* DF measures the aberrations that exist amidst the rate at which audio visual data arrived at a node and rate at which it is being drained over a specified interval. As each packet arrived DF calculates the depth of the virtual buffer being used, the ratio of the different between the maximum and the minimum virtual buffer depth to the drain rate is taken after a specified measurement time interval and the result is the indication of how long it will take in millisecond to buffer the data at the node that will avoid loss of packets. DF also provides us with enough information on the lowest buffer capacity that will be required at the node downstream. DF is expressed in Eq. 8 [34].

$$DF = \frac{\max\ vb - \min\ vb}{media\ rate} \tag{8}$$

where vb = received bytes–sent bytes

## 3 Simulation scenario

We simulate our model using the network simulator NS2 [35]. In order to generate IPTV traffic similar to the one transmitted over IP network, we use the MPEG-4 video traces data which is freely available on the Internet [36]. Similarly, to simulate IPTV scenario as realistic as possible, we concurrently stream a combination of 3 different video traces (Jurassic Part 1, ARD News, Soccer) of YUV:MP4, to generate an aggregated IPTV traffic. Each video clip starts at a random selected time of 0.3 s. All other appropriate data as regards to the three video traces are detailed in Table 1.

The architecture proposed in the Fig. 4, is simulated through a simple scenario, where base stations were deployed over a specific section of the highway within the coverage area of 1,000 m by 1,000 m, to ensure that their signals are overlapping. The base stations were connected to a central router which in turn is linked to an IPTV generator server and a channel changing VOD server that is related to the IPTV stream generator server. The IPTV generator server and the VOD server are the source generating traffic, feeding an intermediate buffered node, which connects to the base stations (BS). The BS in turn is connected to the traffic destination nodes (i.e., vehicles).

**Table 1** Frame statistic of MPEG-4 traces [36]

| Trace | Compress. ratio YU:MP4 | Mean frame size (Kbyte) | Mean bit rate (Mbps) | Peak bit rate (Mbps) |
|---|---|---|---|---|
| Jurassic part 1 | 9.92 | 3.8 | 0.77 | 3.30 |
| ARD news | 10.52 | 3.6 | 0.72 | 3.4 |
| Soccer | 6.87 | 5.5 | 1.10 | 3.6 |

The traffic source uses UDP for transferring Constant Bite Rate (CBR) flow application. In this research, the interest is on forwarding IPTV traffic on vehicles on highway, the density of vehicles on the highway is more acceptable for CBR application flow. The simulation considered a two ways heterogeneous lane highway. In addition, varying speed starting from 0 m per seconds (m/s) when the vehicles were stationary to 20 m per seconds (m/s) was considered. Three traffic scenario considered include: scenario, one with 2 vehicles, scenario two with 3 vehicles and scenario three with 6 vehicles. 1,024 bytes of constant packet size were used, the simulation run for 200 min. The wireless access standard used is the IEEE 802.11p. DSRC PHY and MAC layer were implemented using standard set by ASTM. The data rate set to 1 Mbps. All other parameters as used in the simulation are detailed in Table 2.

In this simulation scenario, the considered assumptions were as stated below:

- All necessary encoding and decoding have been properly executed,
- No packet is lost in the area of the wired segment of the network connection path
- Vehicles on the road are mobile nodes with wireless capability which include on-board computing and wireless communication devices and that each vehicle was capable of supporting large computational and storage recourse.
- The power supply was considered not to be a problem because the engine of the vehicle should be able to provide enough power for computation and whatever should need power to run effectively.

Advance on demand vector routing protocol (AODV) was utilized for the routing between nodes and the MAC used for the node is IEE 802.11Ext. The simulation results of the throughput, delay and loss rate were recorded as shown in Tables 3, 4 and 5.

### 3.1 Test, results and discussion

We validate the proposed results using three metrics; throughput, delay and loss, which due to network mobility

**Table 2** Simulation parameters

| Parameter | Value |
|---|---|
| Wireless access | IEEE 802.11E |
| Radio propagation | Two ray ground |
| Simulation time | 200 s |
| Ad hoc routing protocol | AODV |
| Protocol | UDP |
| Packet size | 1024 |
| Traffic | CBR |
| Traffic density | 2, 3 and 6 |
| Node speed (m/s) | 0, 5, 10, 15 and 20 |

**Table 3** Average delay, loss and throughput for 2 nodes

| | Speed | | | | |
|---|---|---|---|---|---|
| | 0 m/s | 5 m/s | 10 m/s | 15 m/s | 20 m/s |
| Throughput | 0.7134 | 0.2996 | 0.2393 | 0.2188 | 0.2090 |
| Delay | 0.01106 | 0.00465 | 0.00367 | 0.00381 | 0.0031 |
| Loss | 6.1500 | 7.3600 | 10.6700 | 16.7100 | 60.7900 |

**Table 4** Average delay, loss and throughput for 3 nodes

| | Speed | | | | |
|---|---|---|---|---|---|
| | 0 m/s | 5 m/s | 10 m/s | 15 m/s | 20 m/s |
| Throughput | 0.5794 | 0.2466 | 0.1982 | 0.1815 | 0.1736 |
| Delay | 0.0136 | 0.0057 | 0.0045 | 0.0047 | 0.0038 |
| Loss | 8.5100 | 9.5000 | 11.8100 | 19.3000 | 69.5300 |

**Table 5** average delay, loss and throughput for 6 nodes

| | Speed | | | | |
|---|---|---|---|---|---|
| | 0 m/s | 5 m/s | 10 m/s | 15 m/s | 20 m/s |
| Throughput | 0.37967 | 0.1713 | 0.13724 | 0.12719 | 0.12177 |
| Delay | 0.0212 | 0.0192 | 0.0067 | 0.0061 | 0.0058 |
| Loss | 10.5500 | 13.8000 | 16.7000 | 45.0000 | 82.1500 |

limit IPTV service transmission during handoff. To successfully measure QoS in a clearer term, and to minimize losses in order to deliver a better quality IPTV package we use MDI:DF measuring metric.

### *3.1.1 Throughput*

Figure 5, shows the average throughput with respect to speed for the three different scenarios. It could be observed that throughput is higher in the scenario with the lowest

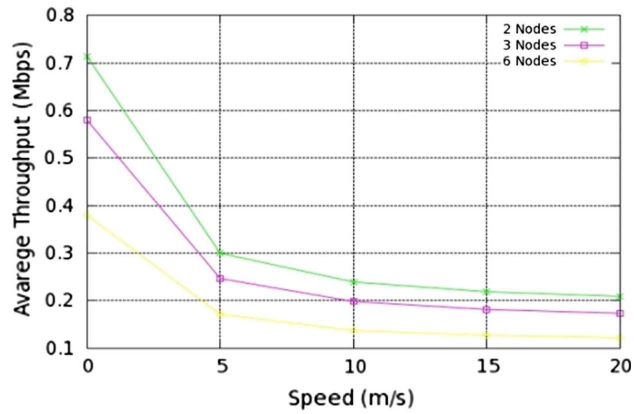


**Fig. 5** Average throughputs for 3 different vehicle density scenarios versus speed

vehicle density (that is scenario 2 and 3) in all the speed level. This means that the average rate by which IPTV traffic can be delivered is greater with low vehicle density than high density. Which imp-plies that the higher the number of vehicles on the road the lower the rate of IPTV traffic delivery (higher density leads to poor throughput). Also in Fig. 5. It can be observed that the throughput drops (reduces) as the vehicle speed increases and the rate of drop is higher in denser vehicle scenario (scenario 3 with 6 nodes) than the less dense vehicle scenarios, this also implies that the higher the vehicle density mode the greater the throughput drops with respect to vehicle speed.

In conclusion, throughput is greatly affected by speed and vehicle density, with a lower vehicle density leading to higher data rate, which is greatly reduced by vehicle speed.

### *3.1.2 Average packet loss*

Figure 6 indicates the average packet loss and the effect both vehicles speed and density have on the quality of IPTV streaming traffic over VANETs. From the graph it could be observed that the packet loss rate increases as both the velocity and the density of the vehicle increases. Only, from the speed of 0 m/s when the vehicles are stationary to the speed of 10 m/s, the average loss rate of the three density scenarios can be observed to be consistent (i.e., the percentage loss of all the three vehicle density scenarios is uniform and minimal within the speed limit $\leq$10 m/s. This could be attributed to fewer link disconnection as a result of even nodes distribution within such speed limit, and so buffer overflow or buffer underflow is minimal, and thus it will be possible within this stage to reproduce a video with full tone. Nevertheless, from speed above 10 m/s, the packet loss of IPTV traffic can be noted to be higher and the pace at which packet is loss increases with vehicle

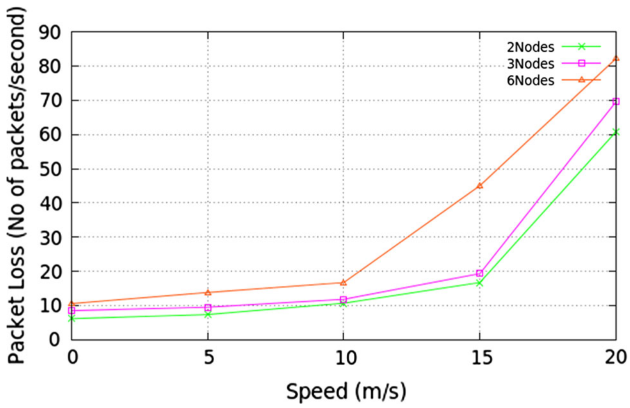


**Fig. 6** Average packet losses for 3 different vehicle density scenarios versus speed

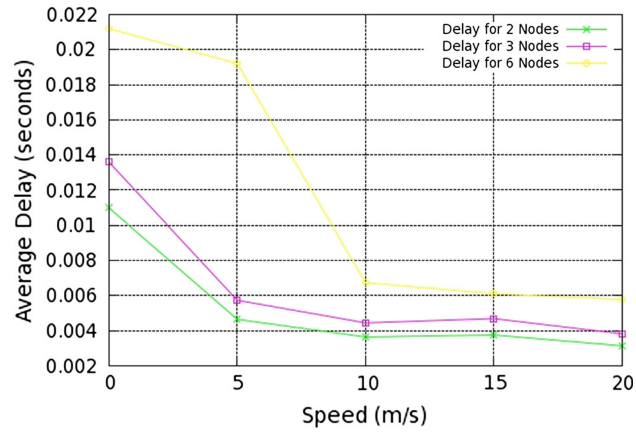


**Fig. 7** Average delays for 3 different vehicle density scenarios versus speed

speed and vehicle density. This is imputable to the fact that the rise in vehicle speed increases the pace at which the nodes moves away from the transmission range. And as the vehicles seed increases, the signal strength diminishes, thereby, increasing the possibility of link disconnections. The increase in link disconnection will result in a rapid topology change, as a result increases the rate of packet loss, thereby, decreases the possibility of reproducing IPTV traffic streaming with reliable quality. In conclusion, as vehicle speed increase above 10 m/s it will be difficult to reproducing an IPTV stream with good quality without any special technique (special technique such as: buffering, error concealment etc.).

### *3.1.3 Packet delay*

Figure 7 indicate the average delay for three different vehicle scenarios with respect to speed. It could be taken note that packet delay increases as the vehicle speed increases and in some cases the packet delay decreases even when the speed increased. This can be attributed to the number of handoff. Packet delay depends on the number of handoff, this can be observed from the graph in Fig. 7. The scenario with six nodes was affected more, this is due to the fact that scenario with six vehicles contained more nodes, as such, more handover were expected compared to the other scenarios of lesser density. Depending on the number of handover, the packet delay can decrease as the velocity of the vehicles increases. The increase in end to end delay as the vehicle speed increases can be attributed to high propagation delay. As the vehicle speed increases, so also is their distance apart. And as the inter vehicle gaps widen, the stability of the link is affected (as the higher the gap between vehicles the lesser the transmission range and the higher the probability of link disconnection). In order to reconnect a broken link that might be caused as a result of the enlarge inter vehicle gap, vehicles will need to catch up with other vehicles ahead to reconnect, and this takes time. During the catch up period in VANETs, the routing protocols employ a strategy known as Cary-and-forwarding, a situation where node forwarding packets hold the packet for a next optimum hop yet to be available [10], by buffering the packets and transmit it in a later available opportunity. However, for real-time traffic such as IPTV, there is a specific needed time (threshold time) by which these packets must be buffered. And so, depending on the stability of the link, the threshold time can decrease or increase. With an increase in threshold time, resulting in the decrease in the delay probability and decrease in the threshold time given rise to high delay probability. This result conformed with proposed distribution media service model in [9]. Consequently, in Fig. 7, as the vehicle density increases, the delay can be observed to have intensified. The reason could be attributed to the increasing number of transmissions competing to access the limited available network link (as higher vehicle density implies increase in the number of nodes relaying messages). Thereby increasing the amount of time required for packets to queue before the Media Access Control (MAC) to literally get to access the channel. Thus, increasing the likelihood of a delay exceeding the maximum delay constraint.

### *3.1.4 Media Delivery Index:delay factor*

With the simulation result obtained, a series of calculation was taken employing the interval of one second each and using the three different density scenarios of the vehicle nodes going with the speed of 20 m/s. To obtain MDI:DF values, eight different measurements each, of the three density vehicle scenarios were considered. The value was

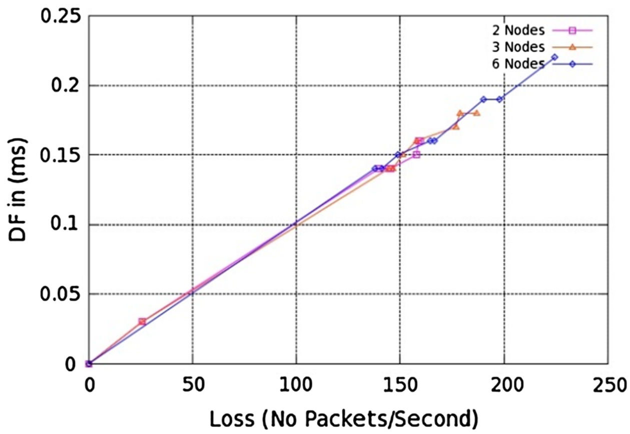


**Fig. 8** MDI:DF value of 2, 3 and 6 nodes at speed of 20 ms versus aggregate loss

applied to ascertain the loss or out-of-order packet over the time interval of 1 s and to obtain the relationship between MDI:DF and loss, as shown in Fig. 8. It could be observed that as the packet loss increases, so also was the DF value. It could equally well be observed that for a small loss of 25.56 packets per second the increment of the DF was moderate for all density scenarios, but when loss exceeded 25.56 packets per second the DF value could be seen to be really high. This means that larger buffer will be needed at the stage where the DF value is high, in order to deliver an acceptable quality IPTV service. However, for given equipment, the size of its buffer is fixed, therefore the qualities of video become worse when DF is higher than the size of the buffer. For instance, in the example shown in Fig. 8, the stage where the loss was 25.56 MB and the DF value was 0.025. (To obtain these values, the initial packet sent was 1,024 Mb and at the close of one second, 998.44 Mb was delivered at the receiving end, this means that 25.56 MB was lost as shown in Fig. 8). To obtain the DF value, the 25.56 Mb loss packets were divided by 1,024 Mb/s to obtain the result 0.025 s. This demonstrated that in order to avoid packet loss of the jitter observed, the receiver virtual buffer would have to be about 25.56 MB or 25,560 kb which will inject 0.025 Mb/s or 25 ms of acceptable delay that will guarantee better IPTV service quality. The DF component of the MDI as demonstrated in this experiment, indicates the length of time a streaming media packet needs to be buffered, to avoid situations that might lead to a buffer overflow or under flow at the receiver end.

## 4 Conclusion

Along with the wonderful promising new infotainments benefits in VANETs come new challenges. For VANET to deliver better quality IPTV, the network has to satisfy certain requirement for bandwidth, delay and jitter, because unlike data application (non real-time traffic applications), IPTV due to its real time delivery nature, any variation in the packet arrival time can generate large numbers of unacceptable error that can lead to poor or no service delivery. To clearly measure QoS, a specific real time video quality measurement metrics that can guarantee QoS service provisioning is required. And to successfully measure QoS in a much clearer term, the knowledge of how to objectively measure the IPTV video qualities is required. And so in this paper, we describe some QoS that can be employed to characterize the operation of vehicular network, and how these metric particularly apply to IPTV quality service provisioning. We also outlined how MDI as an objective QoS metric can be to use to obtain information on network conditions, and how, using these information's can help in shaping network parameters to optimize service flows. Although many other video measuring metrics have been used in literatures, such as VQM, SNR, PSNR etc. However, these metrics only focus on the subjective reflection of human rather than supplying the accurate measurement of those parameters that affect the quality of streaming television (i.e., jitter, delay and loss). MDI does not analyse the video signal decoding, but it detailed how well a video is being delivered, which is the most important outcome, if the video was properly encoded in the first place. And since packet loss, delay and variation in packet delay are the major feature that calls the shots as to whether a network can convey good quality IPTV services or not, the MDI as presented in Sect. 3.1.4 prove to be an effective way to ascertain how well a network can convey video and how well it can fix network problems whose performance has become worsen as a result of traffic load changing or reconfiguration. As our simulation results demonstrate how delay, jitter and packet loss can be minimized to a minimum such that can guarantee good quality IPTV delivery over VANETs.

Further study of the effective ways for delivering IPTV packet over VANETs will require modification in the architecture of all protocol layers of the network. For instance, at the application layer, there is an important need for new object base video coding, error adaptability and error concealment to be specifically designed to be utilised in the coding. Similarly, a new transport layer design that will provide a reliable real time packet delivery along with a network forwarding layer that will be able to differentiate real time traffic packet from non real time traffic (best effort) packet, should be encouraged to be researched upon. And finally, there is a need for research in refining traffic streaming media performance metrics.

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