High Quality of Service Video Conferencing over IMS

Khoem Sambath, Maman Abdurahman, and Vera Suryani

Abstract—**IMS is an evolving definition of an architecture that solves the continuing demands and frustrations of users and enterprises. IMS is a whole new way to deliver multimedia services (voice, video, data, etc.) regardless of the device (mobile phone, landline phone, cable, Internet, etc.) and will change the way all of us relate to our increasingly digital world. Real-time applications such as voice and video have a great deal of demand over IMS network. The QoS is the major concern for real-time application such as voice and video. In order to fulfill the users demand, it is necessary to improve the QoS.**

In this work, we implement IntServ and DiffServ with MPLS which can have great potential in improving the QoS scheme in IMS for video conferencing. We evaluate the performance of QoS for video conferencing over that mechanism perform with IMS. The performance is evaluated based on some QoS parameters such as end-to-end delay, packet loss and jitter. In order to investigate the performance of QoS scheme in IMS, we analyze the simulation results. The investigation shows that proper adaptation of QoS provides qualitative transmission of video conferencing in wireline.

Finally on the basis of the simulation results, we study the QoS performance and propose difference schemes to enhance the QoS performance in IMS based on the QoS parameters.

Index Terms—**IP multimedia subsystem (IMS), multi-protocol label switching (MPLS), video conferencing (VC), quality of service (QoS).**

I. Introduction

Communications companies have been dabbling in this technology essentially since the invention of television. It was mostly impractical or limited in its use however before the advent of broadband internet.

We all communicate, every day during work or leisure time with friends, college, company, organization adversaries and often with ourselves. Until quite recently we had to see or hear the others if we wanted to communicate with him or her personally. This is certainly not true anymore telephones, mobile phones, internet based communication.

We have seen a very fast and large scale development in the last decade. During the past decades the evolution of these network technologies has enabled the development to better achieve user's satisfaction. New innovations in next generation networks are also emerging subject to the demand of users. An IMS is an evolving definition of an architecture that solves the continuing demands and frustrations of users and enterprises [1]. It is a whole new way to deliver multimedia services regardless of the device and will change the way all of us relate to our increasingly digital world. At present, with increasing number of demand driven applications, different types of services are available to fulfill the expectation of users. Against this backdrop, efficient high quality of service is required to sustain and performance the quality of the multifarious services aspiring to satisfy user's satisfaction. The real-time applications such as voice and video conferencing have a great deal of demand over IMS network. A real-time, two-way exchange of information between two or more geographically disperses locations using video, audio and sometimes data. This need is much more expedient in real-time application such as voice and video conferencing, QoS has enormous importance in providing efficient services in order to fulfill the user's expectation.

II. IP Multimedia Subsystem

The mobile and IP networks have enormously changed and development within the last 20 years [1]. From the very first mobile network 1G (first generation) which provide to the users with only some basic services having a high cost and the cell phones were difficult to be used or be carried by the subscribers. During the 90's was the appearance of the 2G (second generation) of mobile networks. The cellular network companies provides the user not only voice, but also some other alternative services. The 3rd generation has great advantages in comparison to the predecessors. The users have the opportunity to exploit a larger bandwidth, a variety of voice services and the prices have become accessible to the average person who can easily afford them [2].

The IMS is the coexistence of mobile and fixed networks. It is an architecture framework for delivering IP multimedia services to the cell phones users. The aim of IMS is not only to provide new services but also to make the Internet services available everywhere which is an additional advantage of QoS [3]. Another advantage is that the IMS users can work and exploit the internet services while roaming. Nowadays the people have to work round the clock and be online with work colleagues for video conference sharing ideas with e-mails and so on. Moreover the IMS is implemented with the same protocols as the Internet and so this fact helps developers to create applications in an easy way using the already existing Internet protocols [3].

III. IMS Architecture

A. IMS Components

The IP multimedia subsystem network is a combination of different functions linked by standardized interfaces [4]. The developers are able to include two or more functions in one node and divide one function in two or more nodes. The main

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The authors are with School of Computing, Telkom University, Bandung, Indonesia (e-mail: khoemsambath@yahoo.com).

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system of the IMS architecture is the call session control function also referred as CSCF. The CSCF have a lot of duties since it proxies SIP signaling traffic and provides services to the subscribers. The CSCF provides the quality of service control such assession control services routing, roaming registration and subscription, the charging and billing main service, the authentication services.

Furthermore, the CSCF is divided into three subsystems Proxy-CSCF, Interrogating-CSCF and Servicing-CSCF. Each of the three control functions has a different role in the IMS core.

1) *The Proxy Call Session Control Function*: The P-CSCF is the first contact point for the IMS user equipment to the IMS network. The SIP signaling traffic is transferred from the user IMS terminal to the network and all SIP signaling is sent from the IMS network to the UE (user equipment) that means the requests responses initially go through P-CSCF.

The incoming calls are forwarded through the P-CSCF on the S-CSCF and Mw interface which is also compatible [5].

2) *The Serving Call Session Control Function*: The S-CSCF performs as a SIP server that makes session control. Its duty is to perform session inspection so it is more than a SIP server. The S-CSCF is located within the subscribers home network and call signaling is routed through the user's home network which is not considered ideal, however this strategy is useful only for signaling traffic. Standard IP routing is used to forward call traffic within the IP networks as the IMS [6]. It can be referred to as the core of the CSCF, because it used to manage all the aspects of user's services and the IMS session. The S-CSCF controls the registration subscriber's status for the applications and has the same operation as the UE remains registered over these services. The S-CSCF performs SIP routing services since the entire incoming and outgoing SIP signaling on an IMS terminal passes through the S-CSCF. It controls the SIP messages in order to identify which services will be used before reaching the final specified destination [6].

3) *The Interrogating Call Session Control Function*: The I-CSCF is responsible for retrieving the path within the network so that the subscriber can communicate with the parts of the network and is usually located in the home network. It routes the SIP signaling requests to the necessary destination and is the first communication point for other networks. The incoming registration requests first pass through the I-CSCF which checks the SIP signaling for unauthorized requests so this way it protects the HSS and the S-CSCF. The I-CSCF determines the location of the requests that will be forwarded to the network components as hiding identity information from other operators [6].

The SIP server defines the next hop for the SIP messages and it checks for the address of I-CSCF which it optionally encrypts in the SIP messages secure information like the DNS and the number of the servers in the domain. Also its duty is to define the name of the next hop from the Home Subscriber Server known as HSS [5], [6].

4) *SIP Protocol*: Is the basic signaling protocol of the IMS standardized by the IETF (Internet Engineering Task Force). The SIP provides the user with mobility through registration and session establishment. It used to signal and control telecommunications sessions. Other words it is a multimedia signaling protocol [7]. The SIP protocol improved the telephony system which gained flexibility and scalability for new services because of the IP network. The SIP protocol can transfer media such as video and voice even file sharing without the need to know the user's location which is the basic reason behind it being the backbone of IMS network [5]. The SIP is a text based protocol and in SIP the user's identities are similar to the email format.

IV. Quality of Service

In the many type network connection has different types of applications are available from different type of networks and different of traffic flows are available from different networks. In order to manage the various traffic flows appropriate QoS is required for proper resource management. In order to manage the resources QoS manager is necessary because it can allocate the resources dynamically for appropriate application and it is able to manage the network resources for different application that comes from other network as well as for its own network application [8]. QoS manager plays an important role in different types of handovers. QoS guaranty is required due to mobility of the users. QoS manager reduce packet losses and delay by provisioning the appropriate resources during handover. Every network and domain contains QoS manger which is defined as Domain QoS (DQoS) manager [2]. IP core network also hold a QoS manager which interact and share knowledge with Domain QoS manager [6].

IP based networks will become the public network infrastructure for time sensitive services such as video conferencing, voice and multimedia service and value added applications such as financial transactions and just in time inventory tracking. The successful delivery of these services is dependent on the ability to provide reliable, predictable and class aware IP transport [5], [9].

In order to attract and retain customers in nowadays highly competitive environment service providers must offer new IP based services with an associated quality of service (QoS). The ability to differentiate and guarantee service offerings enables service providers to charge customers according to the quality of service (QoS) that is delivered. By offering unique, value added and customized services; service providers are better able to differentiate themselves from competitors and leverage their networks to increase revenues [10]. To deliver IP QoS, the Internet Engineering Task Force (IETF) has a number of initiatives that augment the IP protocol to provide reliable, classful delivery of IP traffic. Network equipment manufacturers are also implementing architectural improvements and support for new technologies within core network routers that enable these devices to deliver quality of service guarantees.

A. Integrated Service (IntServ)

The essence of IntServ is to reserve resources for each individual flow so that the service quality can be guaranteed.

Before starting the session an application must specify its requirements and an admission control routine decides whether the request for resources can be granted [11].

This approach is similar to the traditional telephone switching infrastructure where resources are reserved for each call. One problem is the increased burden on the routers, which need to know and store the state of many flows at the same time. Moreover all of them along the connection between the two end-points must be enabled to support protocols and procedures like RSVP admission control and packet scheduling [12].

B. Differentiated Service (DiffServ)

DiffServ divides traffic into different classes and gives them differentiated treatment and to distinguish the classes, the Type of Services (ToS) field is used in the IPv4 header. Using different classification policing shaping and scheduling rules several classes of services can be provided: this decision is in charge of the Internet Service Provider (ISP). There are only a limited number of service classes which means that the routers have to store data proportional to the number of classes and not depending of the flow. DiffServ is therefore more scalable than IntServ and easier to implement. There is still an unsolved problem when high priority traffic concentrates on one router: the bandwidth can be so saturated that it adversely affects performance [12].

DiffServ approaches the problem of QoS by dividing traffic into a small number of classes and allocating network resources on a per-class basis. The class is marked directly on the packet, in the 6-bit DiffServ Code Point (DSCP) field. The DSCP determines the QoS behavior of a packet at a particular node in the network. This is called the per-hop behavior (PHB) and is expressed in terms of the scheduling and drop preference that a packet experiences. From an implementation point of view, the PHB will be translated to the packet queue used for forwarding, the drop probability in case the queue exceeds a certain limit, the resources (buffers and bandwidth) allocated to each queue, and the frequency at which a queue is serviced [13].

The differentiated services architecture is based on a simple model where traffic entering a network is classified and possibly conditioned at the boundaries of the network, and assigned to different behavior aggregates. Each behavior aggregate is identified by a single DS code point. Within the core of the network, packets are forwarded according to the per-hop-behavior (PHB) associated with the DiffServ code point [13].

The IETF defined a set of 14 standard PHBs as [13]:

- Best effort. Traffic receives no special treatment.
- Expedited Forwarding (EF) PHB is the key ingredient in DiffServ for providing a low-loss, low-latency, low jitter, and assured bandwidth service.
- Assured forwarding (AF) are defined to provide different forwarding assurances. The AFxy PHB defines four AFx classes such as AF1, AF2, AF3 and AF4. Each class is assigned a certain amount of buffer space and interface bandwidth to guarantee certain QoS.

V. Implementation of DiffServ with MPLS

The challenge with support DiffServ with MPLS network is label switching routers (LSRs) make their forwarding decisions based on the MPLS shim header alone, so the PHB needs to be inferred from it. In network can support more than eight PHBs, the EXP bits alone cannot carry all the necessary information to distinguish between PHBs. Thus, the PHB is determining from both the label and EXP bits. LSPs which use the label to convey information about the desired PHB are called L-LSPs. L-LSPs can carry packets from a single PHB or from several PHBs that have the same scheduling regimen but differ in their drop priorities (such as AFxy where x is constant and y is not constant) [12], [13].

MPLS and DiffServ are complementary techniques that can be implemented in an IP QoS network to implement an endto-end QoS solution. When used both, DiffServ is provide the standardized QoS mechanisms and MPLS are provides routing techniques increasing the network resource optimization and providing traffic engineering. MPLS domain uses MPLS signaling protocols to establish a label switched path to forward data through a common path. The ingress LSR labels the packets, and the LSRs along the LSP forward the packets to the next hop. In Diffserv the ingress router classifies the packets and then marks them with the corresponding DSCP. The intermediate routers use PHB to determine the scheduling treatment and drop probability for each packet [13].

MPLS makes the DiffServ more reliable and faster due to its path oriented feature. The MLS with Diffserv techniques separate classes of services supported with separate LSPs are routed separately and all classes of service supported on the same LSP are routed together.

VI. Simulator Model (OPNET Modelor)

The requirements of the QoS in the IMS a simulation model of IMS sub network was designed and implemented. Since the IMS is the NGN technology the network characteristics investigation through a simulation tool is required. Gained network characteristics should follow expected results and help to think of the IMS as simulation capable architecture. The OPNET Modeler implements some important elements for IMS simulations. From the Internet protocol suite layer architecture point of view the Internet protocol is supported on the Internet layer. The most well-known transport protocols TCP and UDP are implemented in the OPNET Modeler solution. Also SIP RTP and RTCP protocols are fully supported [13].

The diameter protocol is available in a contributed model. In order to solve QoS challenges in the IMS the contributed SIP IMS simulation model for the OPNET Modeler simulator was developed. The SIP IMS Model for OPNET Modeler is an enhanced model of the original SIP model provided by the standard library. It enables full implementation of the IMS session establishment mechanism. The S-CSCF (Serving Call Session Control Function) P-CSCF (Proxy Call Session Control Function) and I-CSCF (Interrogating Call Session Control Function) servers are supported. Two basic elements are used for sub network application selection and configuration in OPNET Modeler tool: Application Configure and Profile Configure. These elements are independent simulation components which predefine the applications and profiles that will be used by a sub network [13].

The Application Configure element defines an applications namely VoIP video conferencing e-mail FTP (File Transfer Protocol) web browsing database access and others. The Profile Configure element assigns profiles to particular applications. The setup of start time offset, duration or repeat ability is able in Profile Configure table. In order to configure the QoS support the QoS Attribute Configure model is implemented to the simulation tool. The configuration of QoS global parameters is involved [13].

There are many different end-to-end scenarios that may occur in the network. The following scenarios are considered to be significant. Two scenarios were configured in the OPNET Modeler simulator for testing. These scenarios give examples of different QoS mechanisms in different parts of the network which should deliver end-to-end QoS. The first scenario is based on the IntServ mechanism the second scenario is based on DiffServ with MPLS network [13].

VII. The Simulation Scenarios

OPNET Modeler 14.5 has been used for the simulation analysis. This part of this paper describes the network model used in this research. Two network scenarios have been considered to implement. The first scenario 1: IntServ for video conferencing network with QoS implementation. Second scenario 2: DiffServ with MPLS for video conferencing network.

A. *Scenaro 1: IntServ for Video Conferencing*

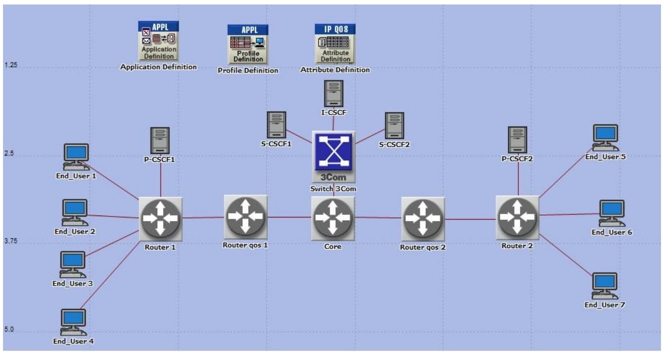

Fig. 1. Low quality for video conferencing.

In the both of scenarios, different traffic classes have been considered for different users. For video and voice users, conversational class has been used (see Fig. 1).

In this scenario with QoS, we used different Type of Services (ToS) to forward the packets such as: Background, Excellent Effort, Streaming multimedia, Interactive multimedia, Best effort, Standard class, Interactive voice and video are respectively. In this Scenario, have SIP IMS configuration, profile definition configuration, application definition configuration is used in this network which are described the following:

The seven nodes wkstn-adv they are namely End-User1, End- User2, End-User3, End-User4, End-User5, End-User6 and End-User7 used in this scenario. In these nodes is also called base work station. The interfaces between End-User1, End- User2, End-User3, End-User4, End-User4, End-User5, End- User6 and End-User7 are covers connected by the routers networks. The end user station before call or conversation to participants they need register with P-CSCF admission control which is distinct by a position to determine for authentication where are has the users transit the network.

B. *Scenario 2: DiffServ with MPLS for Video Conferencing (see Fig. 2)*

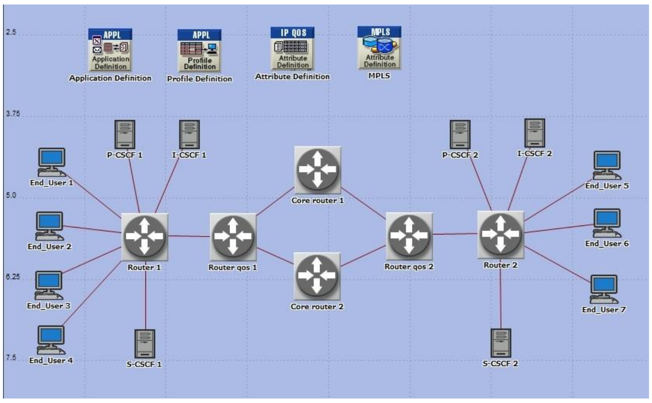

Fig. 2. High quality for video conferencing.

The main objective of these networks is to examine the effect on video conferencing over the traffic of MPLS network. In this scenario, traffic is introduced video conferencing. To evaluate the performance of the QoS, we import IP-QoS attribute to implement QoS in the network. Under the QoS, different types of queuing mechanisms are available [14], [15].

For our simulation, we test different queuing schemes such as MWRR, FIFO and WFQ [14]. We got better performance using MWRR as a result for video conferencing [14]. MWRR is reliable, easy to implement and efficient to define the priority of the traffic in the network. In order to optimize the performance, DiffServ and MPLS have been used [16], [17].

The main advantages are that DiffServ has the administrative control of delay, bandwidth and packet dropping preferences. In order to enable DiffServ in the network, we used Differentiated Services Code Point (DSCP) for the different types of applications such as voice and video transmission [18]. DSCP is used to mark the packets for classifying and forward these packets on the basis of Per-Hop-Behavior (PHB) that are associated with different traffic classes [19]. PHB supports two types of forwarding scheme which are Expedited Forwarding (EF) and Assured Forwarding (AF); these are called Differentiated Service Code Point (DSCP) [15].

In our case, in scenarios with QoS, we used DSCP rather than type of service (ToS) to utilize the DiffServ in order to forward the packets more efficiently in the network.

VIII. Simulation Results

A. *End-to-End Delay*

Time needed for a packet to traverse from the user equipment (Source) to user equipment (Destination) in the network is called end-to-end delay and measured in seconds [10], [11].

End-to-End Delay Scenario 1: In this scenario, frame inters arrival time and frame size for video application is set to 10 frame/second and 9000 bytes/second (see Fig. 3).

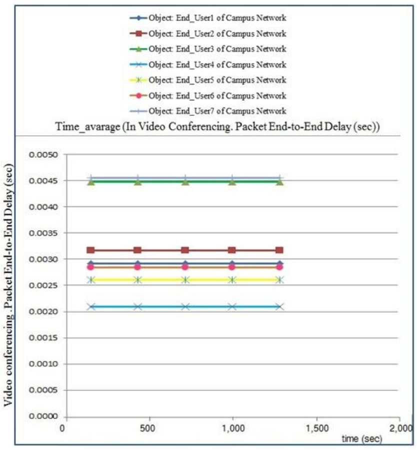

Fig. 3. End-to-end delay for scenario 1.

End-to-End Scenario 2: In this scenario, frame inter-arrival time and frame size for video application are set to 30 frame/second and 26000 bytes/second (see Fig. 4).

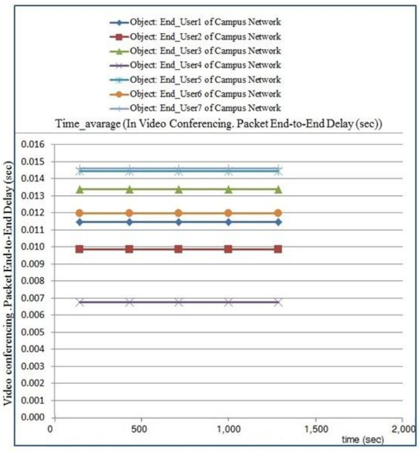

Fig. 4. End-to-end delay for scenario 2.

B. Packet Loss

In the network, when packets of data travelling from source to destination, there might be a chance to lose packets. The ratio of dropped packets to the total number of packets [10] [11].

Packet Loss Scenario 1: Packet sent and received (see Fig. 5).

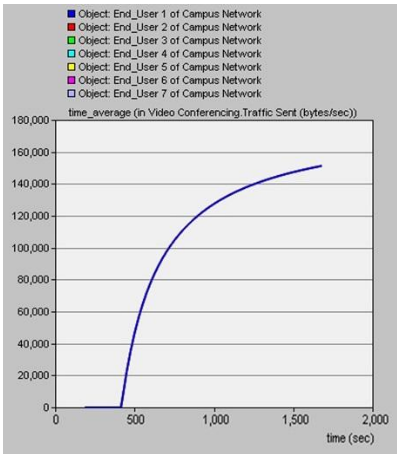

Fig. 5. Packet traffic sent (bytes/sec).

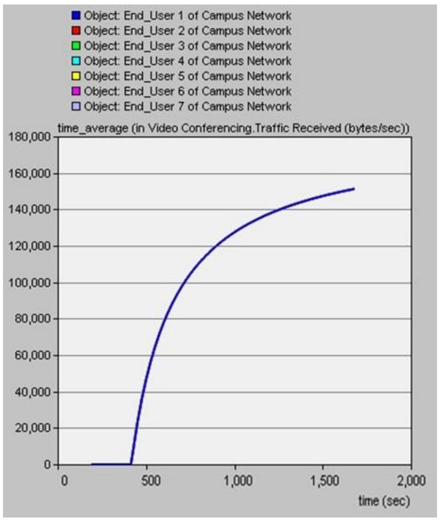

Fig. 6. Packet traffic received (bytes/sec).

Packet Loss Scenario 2: Packet sent and received (see Fig. 6 to Fig. 8).

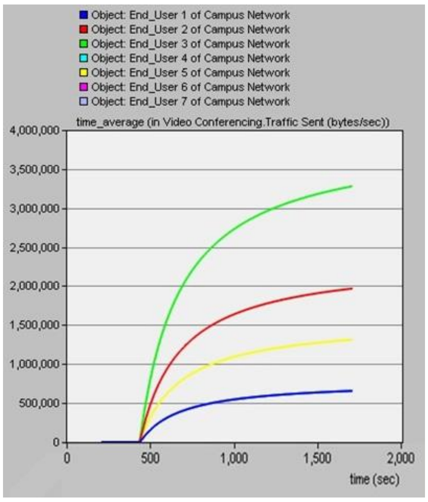

Fig. 7. Packet traffic sent (bytes/sec).

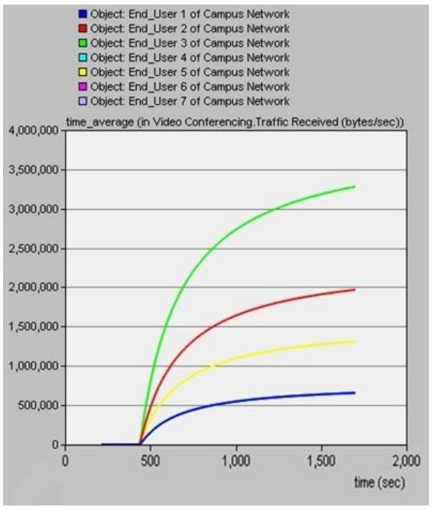

Fig. 8. Packet traffic received (bytes /sec).

C. Jitter (Delay Variation)

Delay variation is important to size of play out buffers for applications requiring the regular delivery of packets like voice or video play out. Jitter is the variation in the delay of the packets sent from the same source towards the same destination [10], [11].

Jitter Scenario 1: IntServ for video conferencing (see Fig. 9).

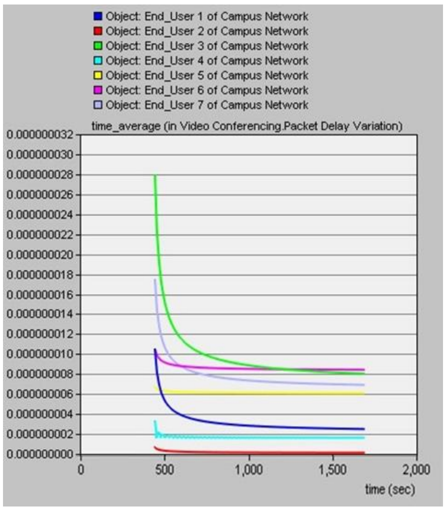

Fig. 9. Jitter for scenario 1.

Jitter Scenario 2: DiffServ with MPLS for video conferencing (see Fig. 10).

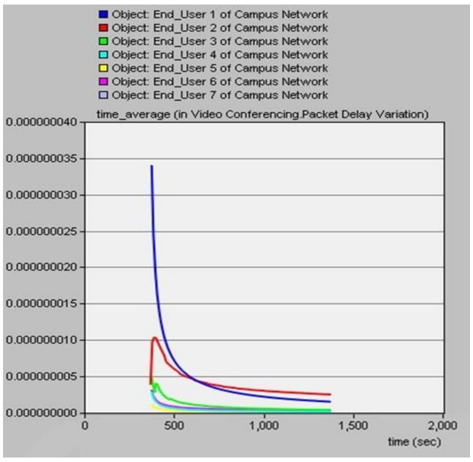

Fig. 10. Jitter for scenario 2.

IX. Conclusion

We implement IntServ and DiffServ with MPLS which can have great potential in improving the QoS scheme in IMS network. Our evaluation is made by simulation in OPNET Modeler based on some performance metrics such as end-to-end delay, packet loss, and jitter.

The performance of the QoS parameters depends on the adaptability of the QoS parameters with the applications. From simulation result and analysis, it turns out that in we proposed QoS schemes with the QoS parameters of in IMS, is well supported due to proper adaptation of QoS for video conferencing transmission. The QoS scheme fulfills the requirement for user's video conferencing transmission in IMS [20].

The appropriate resource allocation and proper adaptation of QoS can be one of the ways to improve the quality of video and audio transmission in IMS network. For proper allocation and adaptation of the network resources, QoS manager and adaptive resource management should be considered for each network. The performance of IntServ, DiffServ with MPLS can have great potential in improving the QoS scheme in IP multimedia subsystem session (IMS) network [21]. The RSVP of IntServ enables the end user to reserve the resources for utilization in the network whereas DiffServ can provide better support for scalable network. With the support of DiffServ and MPLS can make provision for constraint base routing and fast forwarding decision of the packet in order to improve the overall quality of the video and audio transmission on IMS networks

The IMS network will be heterogeneous and will be able to serve a large number of subscribers [22]. It will also be able to handle different technologies in a common platform.

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Khoem Sambath was born on February 25, 1987 in Phnom Penh, Cambodia. He earned his master degree in media infomatics from Telkom University, Bandung, Indonesia, in 2014. He is a graduate student from Telkom University. He received the bachelor degree of computer science from National Polytechnic Institute of Cambodia (NPIC), Phnom Penh, Cambodia, in 2010. He became a lecture of computer science and networking in 2010 - 2012 in National Polytechnic Institute of Cambodia (NPIC). His research interests include switching, routing, distributed and core system.

Maman Abdurahman received his doctor degree in planning embedded System from Insitut Teknologi Bandung, Indonesia in 2010. He received the master degree in information technology from Insitut Teknologi Bandung, Indonesia in 2004. And he received his bachelor degree in informatics engineering from Institut Teknologi Telkom in 1998. Currently, he is the dean of the School of Computing, Telkom University, Bandung, Indonesia. And he is the head of Laboratorium Unified Communication (Lab UC) in Telkom University, and his research interests include the machine to machine (M2M) and IP multimedia subsystem (IMS).

Vera Suryani received her master degree in information techology from Insitut Teknologi Bandung (ITB), Indonesia in 2009. She joined as a lecturer in the Department of Informarics of Telkom University, in 2003. She is currently one member of Laboratorium Unified Communication (Lab UC) in Telkom University. Her research interests include wireless sensor network, distributed system, and internet of things.