ISSN22773061

Performance Evaluation for VoIP on Campus

Rendy Munadi[1], Iman Hedi Santoso[2], Asep Mulyana[3]
Telkom School of Engineering, Telkom University, Indonesia
rnd@ittelkom.ac.id[1]
Telkom School of Engineering, Telkom University, Indonesia
ihs@ittelkom.ac.id[2]
Telkom School of Engineering, Telkom University, Indonesia
asm@ittelkom.ac.id[3]

ABSTRACT

The VoIP Campus implementation is to make the existing VoIP technology become more beneficial for campus stake holder. This VoIP on Campus (VoC) technology make use of a web server, facilitating users to carry out VoIP registration, get and changing account, and also to see others who have register and active in this VoIP network. Basically, this VoC infrastructure uses asterisk as VoIP server and playVoIP as web server interface, those programs included in a server computer. Furthermore, the server interconnected with several servers, such as, PBX, SMS gateway, ENUM server, softphone and smartphone. At this moment, VoC network serve locally, but next time it will be developed so that it could be served in public network, and further VoC network could be connected to VoIP Rakyat, the biggest VoIP network in Indonesia. In this research, VoC network have been tested for several QoS parameters, such as throughput, delay, jitter, packet loss, and MOS. Average value for each parameter, are : 27 kbps throughput, 20.08 ms delay, 3.54 ms jitter, 0.08% packet loss, and 3.3 MOS. Those results indicates that VoC network have a good performance.

Keywords: VoIP, QoS, MOS, Web-Server, Campus

INTRODUCTION

*VoIP (*Voice over IP) is a telephony technology and group of technology for the delivery of voice communications over Internet Protocol (IP) networks, such as the Internet, so that the expense become lower than that implemented on conventional TDM network [5]. Establishment cost for the VoIP network infrastructure quite inexpensive, moreover the expense could be set to minimum if during development using open source software.

The VoIP technology that have been commonly known by campus students are not applicable because there is a trouble to operate it or the VoIP interfaces are not user-friendly. This fact encouraging the development of a VoIP on Campus Service using web interface as a mean to register and get account. And also, in order to facilitating users in accessing service from VoC network, the web interface have set aside various softphone which could be install in personal computer, laptop, notebook, or smartphone.

The special goal in this research, ie. developing VoIP network on Campus which are composite of VoIP server, PlayVoIp, PBX, SMS gateway, ENUM server, softphone, and smartphone, in order for all campus student could use VoIP network as a facility to make everyday communication in campus.

Several aspects related with this research are:

a. To develop VoIP network as a mean to make low cost communication become available in everyday communication in campus.
b. As a beginning in realizing NGN communication in Indonesia.
c. To expand VoIP network in Telkom Education Region and the surroundings.
d. To expand VoIP network in Indonesia.

RELATED WORKS

VoIP applications in the Internet have attracted research on QoS for IP voice services [1]. In [2], Shen evaluated the performance of VoIP codecs on GPRS networks and showed that the VoIP approach may create some capacity gain over traditional circuit switching, with acceptable guarantees in quality of service. Furuya [3] evaluates the relationship between network parameters (e.g., capacity and delay) and the quality of VoIP services.

Although the objectives and the test environments are similar to ours, this work evaluates performance of VoIP on Campus Implemetation, same with Furuya's experiments were conducted specifically with the G.711 codec. James et al. [4] evaluate the effect of loss, delay and error recovery, among other factors, in the perceived voice quality using many codecs (e.g., G.711, G.728 and G.729).

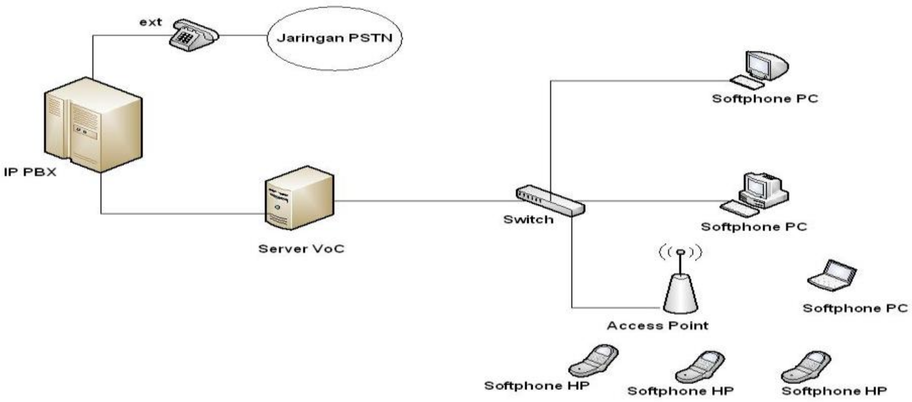
The work of Chen et al [5] correlates the duration os Skype call with QoS factors : transmitted rate, delay, jitter and packet loss. Assuming that call duration may affect quality, the work defines and validates the Mean Opinion Score (MOS), is a subjective evaluation of user satisfaction based on QoS factors.

The experiments with VoIP on Campus applications conducted in this paper demand additional efforts in understanding the control policies used for application adaptation to change in the state of the network Campus, while the traffic load is low or high. To the best of our knowledge, this is the first research study that provides a reasonable to the audio quality under several network conditions.

EVALUATION AND METHODOLOGY

Experiment Environment

Basically, the role of VoC network is a campus access network before entering public network, such as PSTN or PLMN. Hence, the equipments that arrange VoC are comprised: VoC server, IP PBX, and 16 Port Switch (that can be connected to softphone and access point), as can be seen on figure 1 below. On VoC Server have been installed linux operating system, which equipped with Asterisk and PlayVoIP as a web-based interface [6],[7],[8]. The VoC network can be reached by users via LAN or WLAN, where the access points used in this network constitute high power access point with range for about 2.4 km.



ISSN22773061

The wifi hotspot network used in this

Fig 1: Network Topology VoIP on Campus

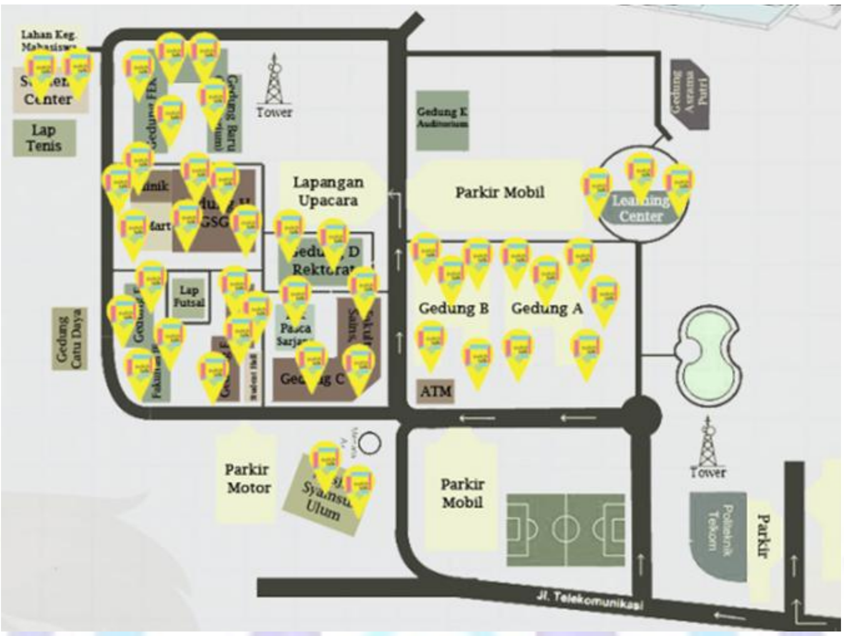


Fig 2: WiFi Access Point Network

Before connecting VoC network with PSTN, we integrate the VoC network with IP-PBX, with the result that the user can make a phone call to analog or IP phone. With softphone that have been installed in each user's smartphone, mobile call can be served by VoC network as long as those users are in access point service area.

The web-based interface of VoC network can be seen on figure 3. The functions of web-based interface are for registration and account distribution [9],[10].

Fig 3: Web-Based User Interface

Metrics of Interest.

The performance parameters of VoIP on Campus evaluated in this reserach are: a) the througphut of the audio received; b) the delay from the sender to the receiver; c) the jitter and d) the MOS. Although the MOS result is significant, the difficulties in performing such a large scale evaluation motivated the development of objective techniques for MOS calculation, which serves directly as a measurement for adaptability.

Although the experiments were conducted in a controlled environment, in this case it is not always possible to control all network variables. For example, when configuring a network path with a lower capacity than the required by applications, some packets are buffered and, consequently, some jitter occurs. Besides, some packets may be lost, and the received rate may be different from the transmitted rate. Knowing that packet loss and jitter do affect the received voice quality [1].

Experiments Description

The experiment was executed for QoS measurement with VoC server using Asterisk server that connected by several clients. This figure below (Fig.4) shows the flow of calling VoIP under VoIP server from one client to the other clients under time based of process.

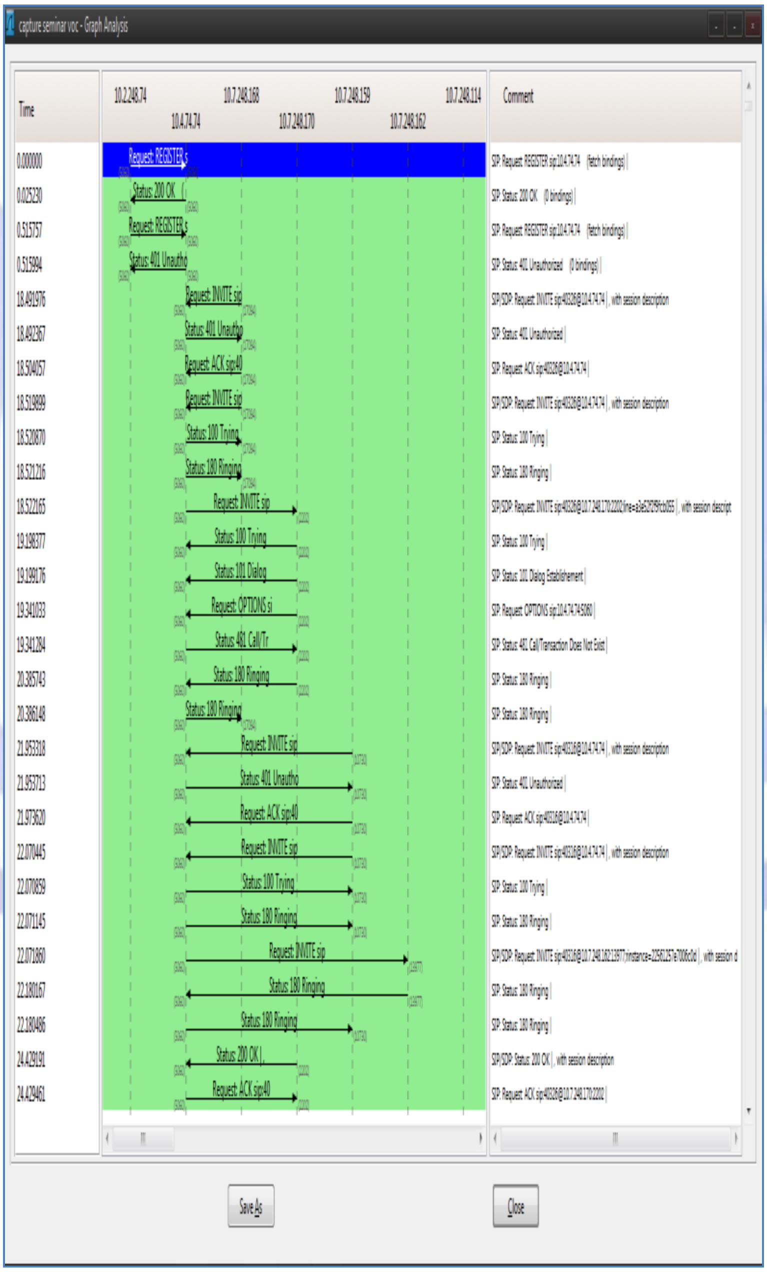


Fig 4: Capture of calling flow

ISSN22773061

Then the next figure (Fig.5) is SIP Capture from source to destination during call flow phase; the length and information each protocol can be showed.

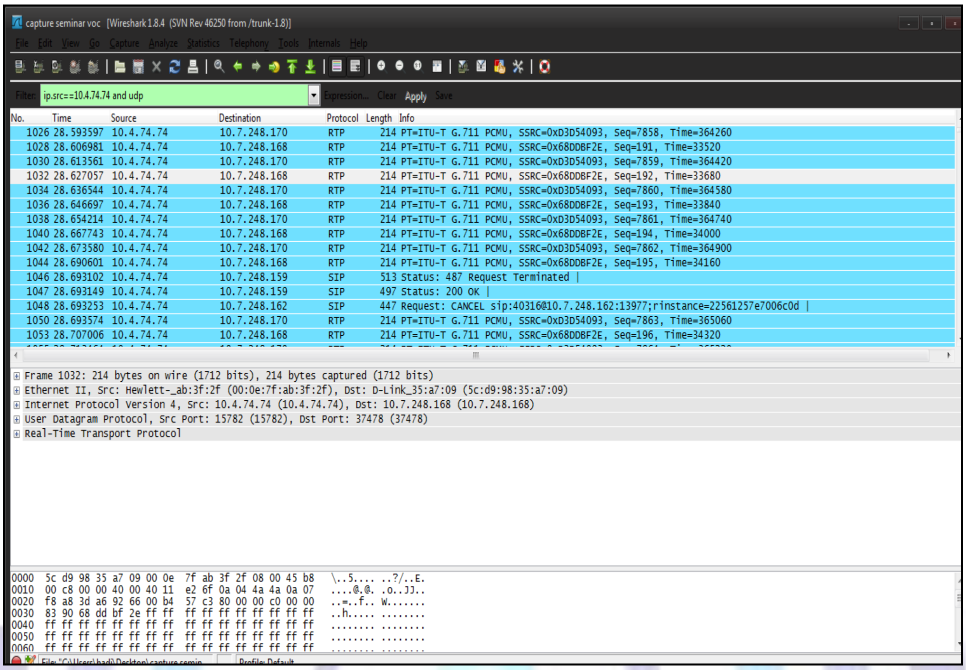


Fig 5: Capture of SIP

From the results (as mention in the figures), the codec G.711u (PCMu) is used as establishment codec for all process communication between source and destination. With this codec, the transport protocol using UDP, the real time transport protocol is RTP and for signalling protocol using SIP.

On this experiment, we use only G.711u codec, without using other codec, in order to get optimal results for all performance parameters.

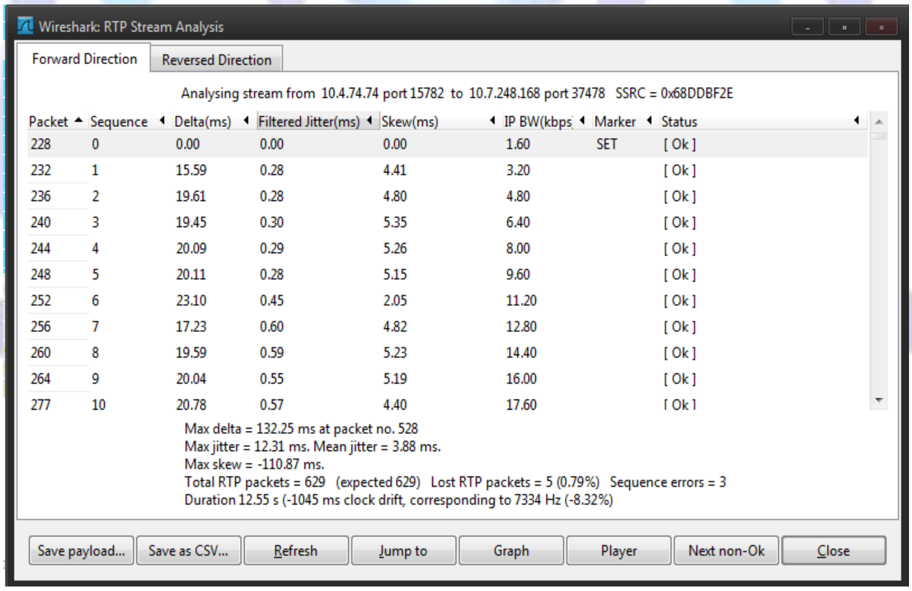


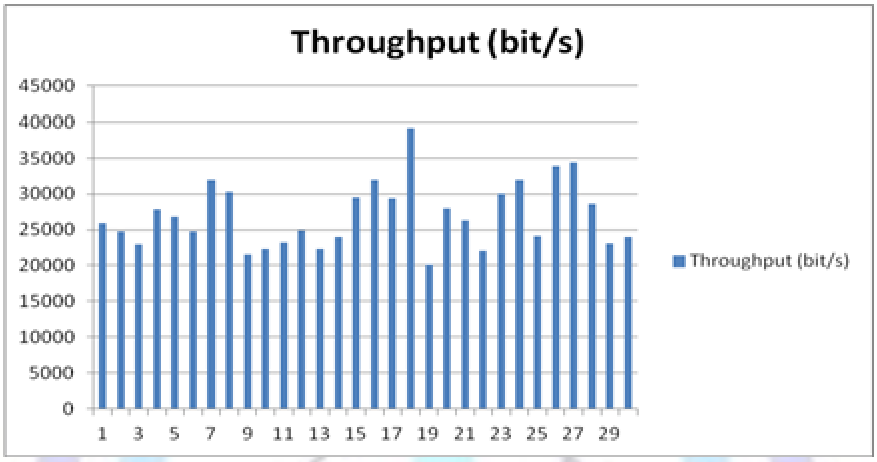
Fig 6: Streaming Analysis

The final process of experiment description is streaming analysis (Fig.6 above), to obtain all performance parameters, such as delay, throughput etc.

PERFORMANCE EVALUATION

Throughput

The definition of throughput is the ratio between the packets accepted with the packets send on observation time. Unit used is bps (bits per second). Throughput measurement objective is to determine the reliability of the network in forwarding incoming packets to reach the destination. The experimental result is in graph 1.

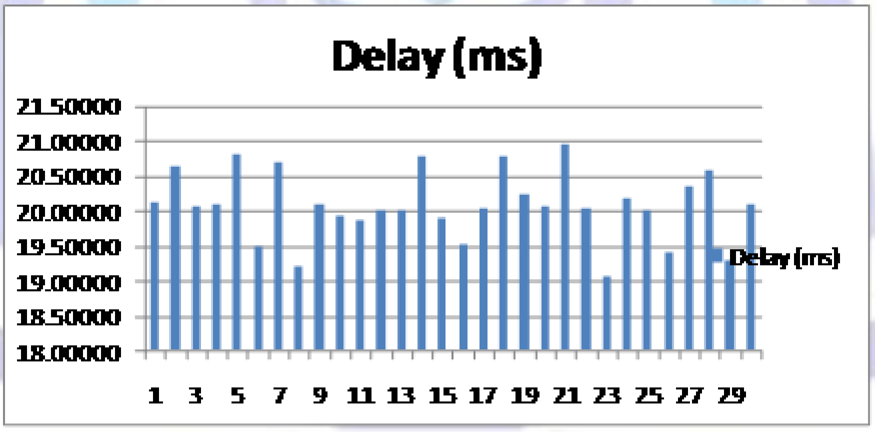


Graph 1: Graphic of throughput

From this graph the throughput average is 27.009 bit/s. This value can be representatived that the network provide delivery packet on the best results.

Delay

The other parameter performance is delay, it's the time required for a packet to move from sender to receiver. Unit used is milliseconds (ms). Delay is used to find out how fast the network is used in forwarding packets from sender to receiver. The experimental result is in graph 2.

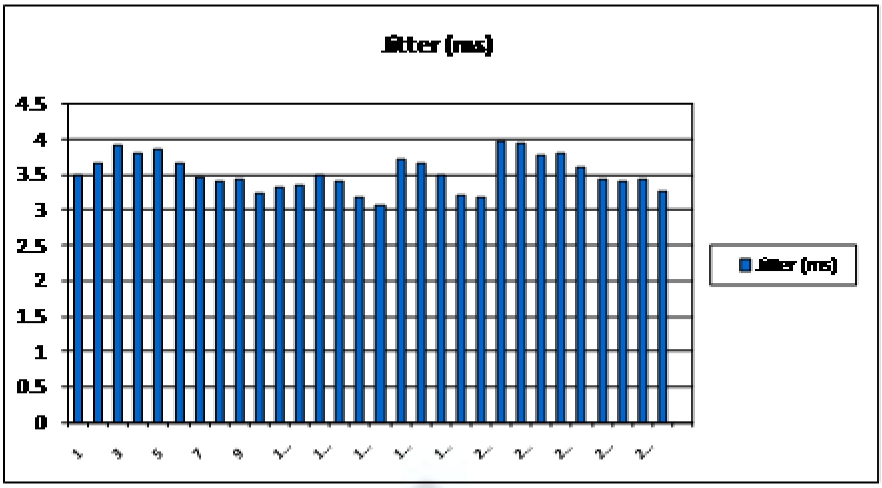


Graph 2: Graphic of delay

From this graph the delay average is 20.07673 ms. This value can be representatived that the network provide delivery packet with very past from end to end.

Jitter

The received voice quality can be affected by jitter, it's the variation delay that occurs due to unstable network, or any packet in buffer, so time of each packet received is different. Jitter analysis is used to know jitter size obtained from each of the network used. The result is in graph 3.

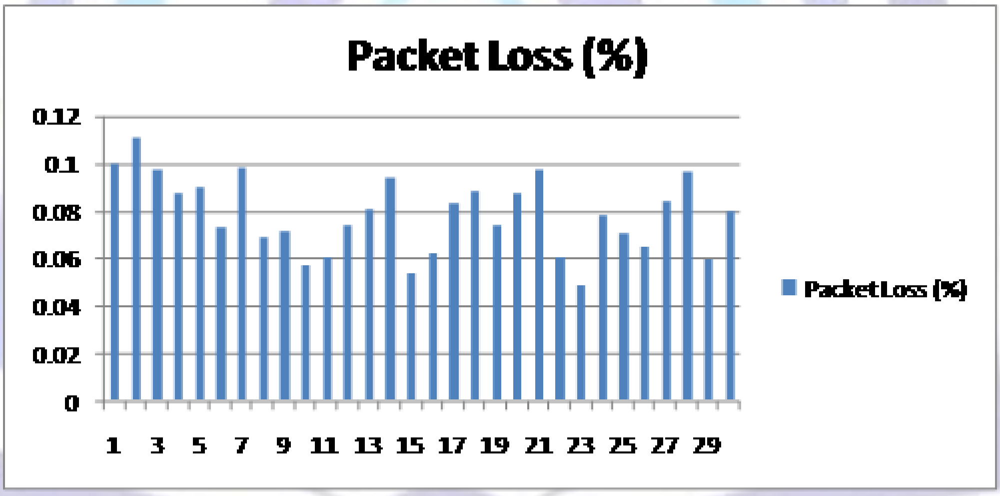


Graph 3: Graphic of jitter

From that graph the jitter average is 3.542977 ms. This value can be representatived that the network provide delivery packet with very past from end to end, and the buffer is in empty condition.

Packet Loss

The last stage in this experiment results is packet loss. Packet Loss is the number of packets lost during the delivering process takes place compared to the number of packages that survived. The unit used is the percent (%). Packet Loss measurement objective is to determine how reliable the technology is, in order to maintain the packet to be forwarded. The experimental result is in graph 4.

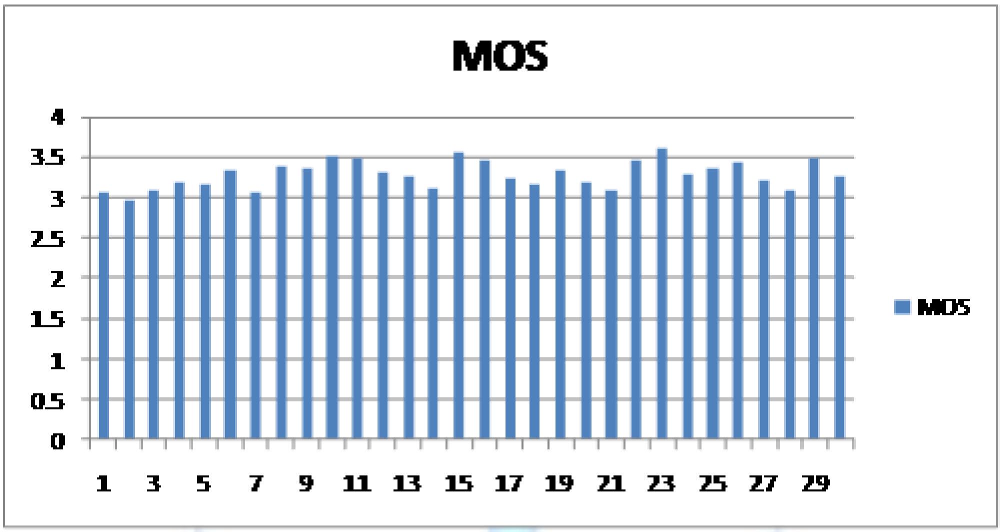


Graph 4: Graphic of packet loss

From graph 4, it can be seen that the packet loss average is 0.078346667%. This value can be representatived that the network provide delivery packet is very good, so the throughput can acheives maximum value.

Mean Opinion Score (MOS)

In this experiment, the study uses MOS approach to evaluate our VoIP Campus performance using only one type of codec. MOS is a subjective evaluation, calculated by averaging the grades given by a large sum of people that listen to an audio sample that went through a coding/decoding process. The grade is in the range from 1 (bad) to 5 (excellent). The result is in graph 5.



Graph 5. Graphic of MOS

From this graph the MOS average is 3.28104. This value can be showed that the received voice quallity is not excellent but good enough, so the network should be improved by using other codec or other scenario.

CONCLUSION and FUTURE WORK.

This research in focused to develop VoIP network on Campus which can be extended to reach another campus in the next time; for the next goal, this VoC can be connected to another existing VoIP network that have been operationally matured. The results of some experiments and tests that have been carried out, exhibit that our VoIP network have good performance. In future work, it should consider implementing techniques to improve quality of VoIP network on Campus, especially in access network or other software based network.

ACKNOWLEDGMENTS

The authors would like to thanks the Directorate General of Higher Education of Indonesia and Telkom University in Bandung for their financial support. They also express their sincere thanks and appreciation to switching laboratory student and all colleagues at Telkom School of Engineering for their kind help and usefull suggestions.

REFERENCES

[1] Rodrigo, B.; Arthur, C.; Carlos, K.; Stenio, F.; Denio, M.; Djamel, S." Performance evaluation of P2P VoIP applications", ACM 978-1-5959-3-746-9, June, 2007.

[2] Shen. Q.,"Performance of VoIP over GPRS", 17th International Conference of Advanced Information Networking and Applications (AINA'03), 2003.

[3] Furuya, H.; Nomoto, S.; Yamada, H; Fukumoto, N; Sugaya, F.,"Experimental investigation of the relationship between IP network performences and speech quality of VoIP", 10th International Conference on Telecommunication(ICT 2003), March 2003.

[4] J. H. James, Bing Chen, and Laurie Garrison, "Implementing VoIP: A Voice Transmission Performance Progress Report",IEEE Communication Magazine, July 2004.

[5] Kuan-Ta Chen, Chun-Ying Huang, Polly Huang, Chin-Laung Lei, "Quantifying Skype User Satisfaction," ACM SIGCOMM 2006, September 2006.

[6] Cisco, *Virtual Router Redundancy Protocol, Manual book,* Cisco System Inc.

[7] Cisco, *Virtual Local Area Network, Manual Book,* Cisco System Inc.

[8] Cisco, 2007, *Catalyst 5500 Series Switch, Manual book,* Cisco System Inc.

[9] Virtual Router Redundancy Protocol, *White Papper.* CaseXcommunication.

[10] Virtual Router Redundancy Protocol. RFC 2338.

ISSN22773061

Author' biography with Photo

Rendy Munadi received his Doctor in Telecommunication Engineering from Indonesia University. He is senior lecturer of Telkom University Bandung-INDONESIA and he is presently as Head of the Expertise in Networks and Multimedia. He has served on the program committee of several conference and as reviewer of papers. He is current reserach in the area of Next Generation Network and New Generation Network, Wireless Network, Wireless Sensor Network, IMS, IP/MPLS Network, Routing Management and protocols/interface Next Generation Network.

Iman Hedi Santoso received his Master Degree in Telecommunication Engineering from the Bandung Institute of Technology in 2007. He has been involved with joint projects between Telkom University and Telecommunication Ministry of Indonesia, related to major issue in Indonesian telecommunication and maritime, those project are: Survey Analysis the Performance of Services in the Indonesian Mobile Operator, Analysis of the Impact of new generation network application in Indonesia, Maritime Call Sign Regulation in Indonesia, Operational Standard in Maritime Telecommunication in Indonesia, and Digital Maritime. He also participated in researches funding by Directorate General of Higher Education of Indonesia. His research and teaching interests include traffic engineering, switching technique, information and telecommunication network, computer network, and wireless network.

Asep Mulyana received his Master Degree in Computer Engineering from Bandung Institute of Technology. He has experienced in the field on switching system (exchange), experienced as trainer on various technology of switching system, signalling system and telecommunication networks in the Training Centre PT. TELKOM Bandung. Currently he is a lecturer on Switching.Technique, Telecommunication Networks, Traffic Engineering, Access Networks, Signalling System and Next Generation Networks (NGN). At this moment he is interest and concern to research on NGN and IMS.