Traffic Analysis of Quality of Service (QoS) for Video Conferencing between Main Campus and Sub Campus in Laboratory Scale

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Abstract

Recently, in the distance learning system, video conferencing becomes one of expected course material delivery systems for creating a virtual class such that lecturer and student which are separated at long distance can engage a learning activity as well as face to face learning system. For this reason, the service availability and quality should be able to guaranteed and fulfilled. In this research, we analyze QoS of video conferencing between main campus and sub campus as the implementation of distance learning system in laboratory scale. Our experimental results show that the channel capacity or bandwidth of WAN connection between main campus and sub campus at 128 kbps is able to generate the throughput of video transmission and reception at 281 kbps and 24 kbps, respectively. Meanwhile, throughput of audio transmission and reception is 64 kbps and 26 kbps with the number of total packet loss for video and audio transmission is 84.3% and 29.2%, respectively. In this setting, the total jitter for video and audio transmission is 125 ms and 21 ms, respectively. In this case, there is no packet loss for traffic transmitting and receiving with jitter is not more than 5 ms. We also implemented QoS using Trust CoS model dan Trust DSCP for improving the quality of service in term of jitter up to 12.3% and 22.41%, respectively.

Keywords: quality of service, throughput, delay, jitter, packet loss, Trust CoS, Trust DSCP

1. INTRODUCTION

Based on Agosti et. al. [1], distance learning system can be defined as a learning system or process whereas the instructor and student are separated at long distance. By enabling information and communication technology (ICT), course materials can be distributed to the student properly. The used

of ICT involves network infrastructure such as Internet and equipments to deliver and fetch course materials such as personal computer and laptop PC.

Video conferencing is one of often applied and important distance learning models. In addition, other models such as e-learning, interactive media, e-library, etc are also famous methods for distributing course materials in the distance learning system. Throughout video conferencing, learning process can be conducted as well as face to face learning system if it is supported by a good ICT. In this case, the long distance location between lecturer and student is no longer problem any more in the distance learning system.

Video conferencing technology based on IP is one of very important things for universities in guaranting and ensuring course material distribution as a good university's service to the student. This mechanism will be very important when it is used for distance learning system. However, the operational of video conferencing in two or more different locations and far away one and another will generate a good result if and only if the availability of network resources is guaranteed with low level of latency/delay, packet loss and jitter. This enables a big challenge in the distance learning system which adopts video conferencing system.

Aditya Arie Nugraha et. al. [2][3][4] developed a concept of improving education quality in the elementary school by using ICT to create a virtual class between lecturer and student instead of face to face learning system. The testbed was implemented in the community of elementary school at Keerom-Papua and Subang, Jawa Barat. Authors also analyzed the QoS based on the configuration of audio/video codecs with respect to codec's bit rate. Analyzation was performed by designing and implementing a virtual class service in the laboratory scale and test bed was realized for the distance learning system in Serang Panjang, Subang Jawa Barat [4]. From the investigation result was obtained that a combination between video bit rate at 512 kbps and audio bit rate at 128 kbps causes network load is only 9.6% with the network throughput 9 Mbps in the laboratory scale. This experimental was done in the ideal topology using network equipments in the laboratory through audio quality measurement with Perceptual Evaluation of Speech Quality (PESQ).

Slamet Sudiarjo et. al. [5] measured QoS of video streaming traffic through Switch layer 2 in the IP based local area network which connects End-to-End codec between two switches. The experimental result was delay time measuerement for audio and video protocol, jitter for each audio and video traffic and packet loss occurs in the video streaming.

Cassidy D. Smith [10] introduced an implementation of video conferencing through IP based network with QoS configuration both in the core network, distribution network and access network using Cisco equipments to connect End-to-End codec over WAN connection. In addition, Cisco Corp. introduced a solution regarding communication and

collaboration throughout Cisco Unified Communication [8] and its application for video conferencing [9].

In this research, we use network infrastructure in the Network Administration and Security Laboratory at EEPIS campus. Developing a video conferencing system as a part of Unified Call Manager System [8][9] for real conferencing between main campus and sub campus is our main target. This system is inspired from the concept of video conferencing in the company when they have online meeting between End-point in headquarter office and End-point in the branch office instead of face to face meeting. This mechanism is performed to reduce the cost of business travelling and phone call usage. Comprehensively analysis of video conferencing which consist of all configuration of various audio/video codec with its bit rate and other QoS parameters such as throughput, delay, jitter and packet loss is conducted in this research.

2. IP BASED VIDEO CONFERENCING TECHNOLOGY

IP based video conferencing technology is one of very important for universities in guaranting a course materials distribution as an offering service to student. This mechanism will be realized for the appropriate goal such as distance learning system. In this section, we describe the fundamental technology related to this topic of interest.

2.1 End-to-End Point Quality of Service in Campus Video Conferencing

Network connects two campuses through WAN connection. Usually, WAN coonection is provided by city network provider such as ISDN/ATM, DSL, SONET, Wireless WAN, Metro Ethernet, and other public connections including Internet. Edge routers forward audio and video traffic from Gatekeeper or WAN and keep the quality of service based on queue priority. Network devices such as Layer 2 Switch maps Class of Service (CoS) value into Differentiated Service (Diffserv) value to network devices Layer 3. In these Layer 3 devices, the route of traffic is determined based on priority. Furthermore, Layer 2 Switch classifies and prioritizes bits of CoS. Network devices used for video conferencing mark the traffic by assigning CoS values or Differentiated Services Code Point (DSCP) values, In this situation, Switch only needs a trust in CoS values of port where the codec connected.

2.2 QoS Technology

The important issues in video conferencing are the availability of bandwidth, packet loss, delay and jitter or delay variance. Bandwidth availability means the available data during transmission and it is expressed for deciding channel capacity in megabits per seconds (Mbps). Packet loss denotes a prosentage of failed received data by codec from the transmission process. Delay or latency can be represented by the average time between data transmission and data reception and it is usually expressed by milliseconds (ms). Meanwhile jitter is a latency variance in the part of

transmission process. Jitter can be measured through various ways such as the different between the highest latency and the lowest latency, standard deviation of latency, etc. On the other word, jitter is always given in qualitative or delay variation of data traffic during transmission process.

QoS technology could be enabled in the Layer 2 and Layer 3 network devices. To guarantee the quality of End-to-End communication, it is required a mapping process from Layer 2 to Layer 3. In this case, QoS values are translated between CoS values in the Layer 2 and DSCP values in the Layer 3. Table 1 shows the example of mapping values from Layer 2 to Layer 3. There is no standard for priority of various traffic types should be obtained. This is just as a recommendation when working on video conferencing.

Table 1. Mapping CoS values in Layer 2 into DSCP values in Layer 3. [10]

Type of traffic	CoS values in Layer 2	IP precendence in Layer 3	DSCP values in Layer 3	
Reserved	7	7	-	(56)
Reserved	6	6	-	(48)
Voice	5	5	EF	(46)
Video Conference	**4**	**4**	**AF41**	**(34)**
Call Control	3	3	AF31	(26)
High Priority Data/Streaming Video	2	2	AF21	(18)
Medium Priority Data	1	1	AF11	(10)
Best Effort Data	0	0	BE	(0)

2.3 Non-QoS Parameter Impact in Video Conferencing

In addition of above QoS parameters such as throughput, delay, packet loss, and jitter, regarding [10] the following are non-QoS parameters influence quality of service in video conferencing:

- Call Manager Software: this software should be installed in backbone network and edge routers. Ideally, this software should be directly connected to the core switch and it is recommended to have not more than one hop from edge router.
- Ethernet speed and duplexing: it is recommended to configure ethernet connection manually to 100 Mbps and full-duplex. Normally, video transmission requires at least 384 kbps for 10 Mbps half-duplex Ethernet connection to get a good performance. However, full-duplex setting is not enough if there is no matching duplex among network devices. As the result, video conferencing may have a problem. Therefore, it is recommended to use manageable devices such that Ethernet speed and duplex can be configured manually.
- Spanning tree is a mechanism to prevent routing-loop, due to routing-loop affects longer convergence time. By controlling spanning tree, the convergence time of network topology can be achieved as fast as possible. In video conferencing implementation, convergence time is very important to improve the quality of video conferencing.

- Codec: several codes are not easy to work in high speed and duplex Ethernet ports. Some other else may have a feature to mark incoming packet and perform priority mechanism based on CoS and DSCP values. Higher end codecs usually have both priority mechanism and packet marking features to eliminate packet loss.
- Other equipments such as camera, microphone and speaker that are not working properly cause lower quality of service in video conferencing system.
- Firewall: this feature also introduces delay time. Applying firewall usually used for enabling security feature, but it can be decreased the quality of service. To get the best performance in video conferencing, it is recommended to disable firewall feature.

3. VIDEO CONFERENCING SYSTEM BETWEEN MAIN CAMPUS AND SUB CAMPUS

This research explains about quality of service for video conferencing system practically by using network devices in the Network Management and Security Laboratory at EEPIS campus. This section will describe method, network topology, implementation, and quality of service measurement and its investigation.

Video transmission over IP based network is very sensitive from delay and jitter may occur in the network. This situation is very important for campus to ensure and guarantee the quality of video transmission as a good service to the student when such mechanism is used for delivering course materials. This research illustrates network configuration with fulfillment all requirements in implementing quality of service of End-to-End devices involve in the video conferencing system between main campus and sub campus. Main campus network and sub campus network are connected over WAN connection. In this case, quality of service of WAN connection also determines the quality of video transmission in the video conferencing system.

Previous sections have mentioned that the main goal of this research is to develop a Unified Call Manager system to establish a video conferencing service for distance learning system and analyze its quality of service comprehensively with respect to affected factors such that the suitable quality of service can be achieved for a good result of distance learning system.

Topology model of Unified Call Manager System is used for providing video conferencing service in the laboratory scale. In this research, the network topology of the system is described in Figure 1. We use Cisco devices to develop the system infrastructure.
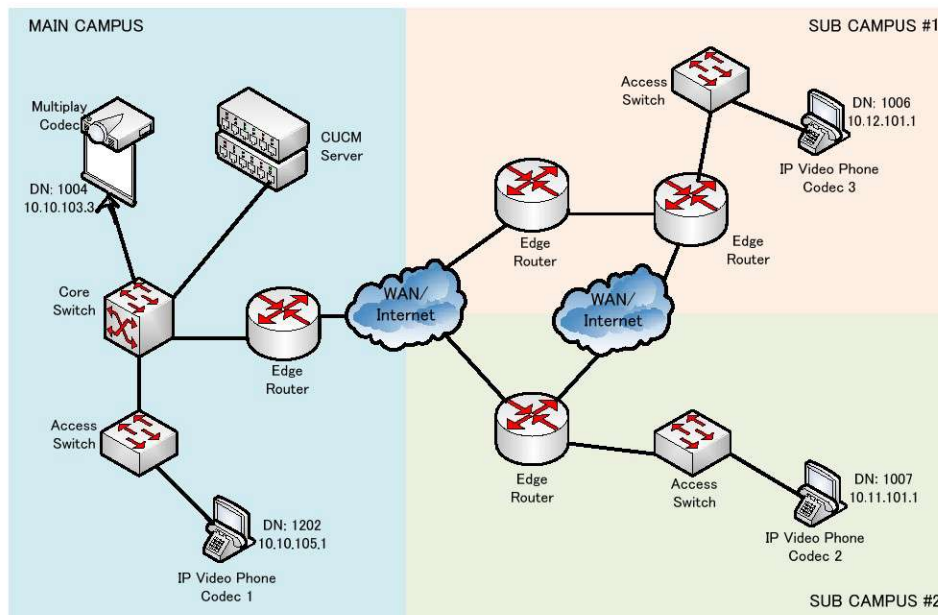
Figure 1. Topology model of video conferencing system between main campus and sub campus in laboratory scale.

In this system model, we collaborate the functions of IP communications, telepresence, conferencing (i.e., including video conferencing), messaging, and mobile applications [8]. Telephony function is handled by equipments such as: (1) Call Manager which has a function as IP PBX system for registration process, dial-plan, voice bandwidth management and voice gateway management; (2) Voice Gateway which has a function as a gateway in the network devices to convert realtime voice call between circuit-switched telephone network and IP network; (3) Endpoint IP Phone which has a function as an End-point of user to perform voice call. In this research, QoS analyzing will be executed based on four issues needed for fulfilling quality of service such as packet delay, jitter, packet loss and throughput.

Cisco Unified Call Manager (CUCM) [8] provides telephony feature and forwarding feature to route telephony packets passthrough network devices. These packets comprise IP phone, media processing device, VoIP gateway, and multimedia applications. Meanwhile, data, voice, and video service such as convergence message, conferencing, collaborative contact center and interactive voice response system (IVR) interact with IP telephony through CUCM application. Based on [8], the functions of CUCM are as follows:
- Call processing means a fuction for originating, routing and call terminating, including billing system.
- Signaling and device control: CUCM sets signaling connection between endpoint call and devices such as gateway and conferencing-bridge to establish streaming connection. This function also is used for calling control and calling setup.

- Dial plan administration: this function includes setting list of configuration for call routing. CUCM responsibles to analyze digit number for all call and allows user to make dial plan.
- Phone feature administration: this function covers additional features of CUCM such as hold, transfer, forward, conference, speed dial, redial, call park, and other features for IP phone and gateway.
- Directory service: CUCM uses its own database to store user information. User authentication is done locally or using external directory.
- Programming interface to external applications: CUCM provides interface programming to external applications such as IP softphone, IP communicator, Unified IP interactive voice response (IP IVR), personal assistant, unified personal assistant and console.

4. IMPLEMENTATION OF TRUST CoS and TRUST DSCP

In this work, we implemented Trust CoS and Trust DSCP on the interfaces of the teleconferencing devices in our video conferencing system between main campus and sub campus as shown in Figure 1.

4.1 Trust CoS

Basically, this model is able to improve the quality of service by increasing the performance in the Layer 2 devices such as switches. The algorithm of Trust CoS is described in Algorithm 1.

Algorithm 1. Trust CoS generation algorithm.

```
INPUT: Data packet
OUTPUT: Data packet + Trust Cost header (3-bit)
1. class-map generation
     packet classification based on Class of Service (CoS)
     match presedence 5 (critical);
2. Policy-map generation
     set precedence 5 (critical);
3. Packet marking
     packet header generation;
     packet header labeling based on policy-map into marked-
packet;
4. marked-packet and scheduling queue
5. policy-map on the interface
     service-policy input (name_policy-map);
```

Trust CoS implementation on the devices, especially switches is started by class-map generation which has a function to group the packets based on the class of service type, such as IP address or source port. Then, it generates policy-map to provide the priority of generated class. We set precendence 5, due to the priority of the packet is critical. After determining the priority of policy, such policy is applied on the interfaces.

4.2 Trust DSCP

Different from Trust CoS, this model is able to improve the quality of service by applying the performance of Layer 2 and Layer 3 devices such as routers. The mechanism of Trust DSCP is described in Algorithm 2.

Algorithm 2. Trust DSCP generation algorithm

```
INPUT: Data packet
OUTPUT: Data packet + Trust CoS header (3-bit) + Trust DSCP
header (1-byte)
1. class-map generation
     packet classification based on Class of Service (CoS)
     match precedence 5 (critical);
     match dscp af41 (video-conferencing);
2. polic-map generation
     set precedence 5 (critical);
     set dscp af41 (video-conferencing);
3. packet marking
     packet header generation;
     packet header labeling based on policy-map into marked-
packet;
4. marked-packet and scheduling queue
     service-policy input (name_policy-map);
```

Similar to Trust CoS, Trust DSCP implementation on the devices is started by class-map generation which has a function to group the packets based on Class of Service type such as IP address and source port.
We selected af41 for grouping inbound traffic of video conferencing. Then, policy-map is generated to provide a priority of the class. Set precedence 5 is selected, due to this set is critical. In this priority decision, it is added teleconference packets so that a set of priority is also af41. And finally, the policy is applied on the interfaces.

5. EXPERIMENTAL RESULTS AND ANALYSIS

In this section, we describe our experimental results and the analysis of video conferencing performance between End-point in the main campus and End-point in the sub campus. Measurement and analysis comprise the investigation of packet delay, jitter, throughput and packet loss from the quality of audio and video.

5.1 Network Topology in the Testbed

Figure 2 illustrates the network topology in our experiments to investigate the quality of video conferencing system for distance learning between codec in main campus and codecs in sub campus #1 and sub campus #2.
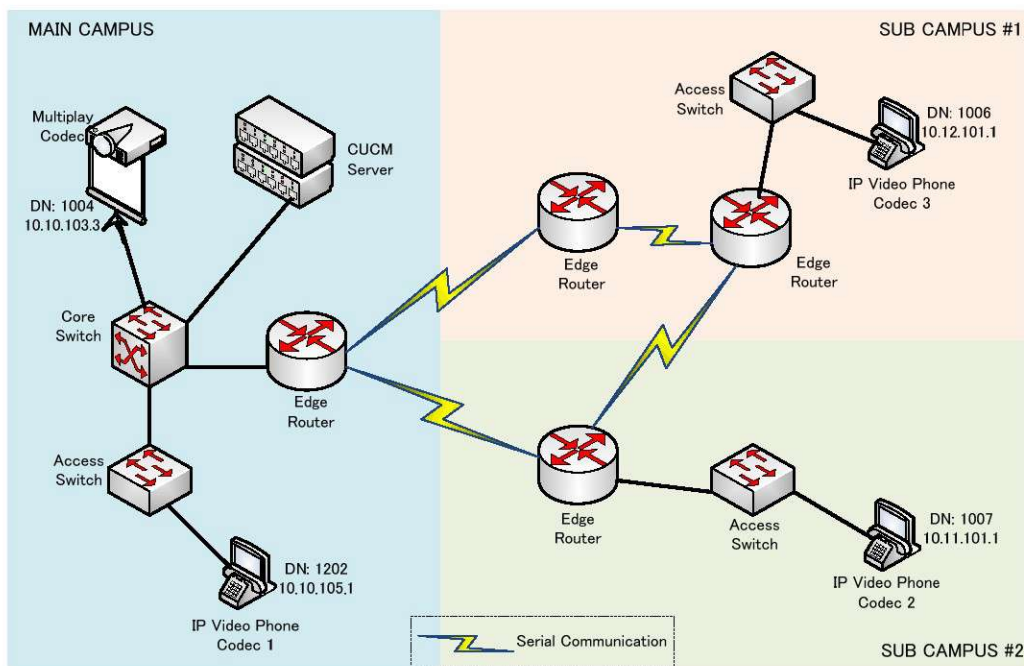
Figure 2. Network topology in the testbed.

We use Cisco EX90 Multiplay codec and IP video phone codec as the End-points in main campus. Cisco EX90 codec has a feature to conduct a room video conferencing with maximum 4 connected codecs simultaneously and it is assigned into a dial number (DN) 1004 with IP address 10.10.103.3. Whilst, IP video phone is set to DN 1202 with IP address 10.10.105.1 over main campus network in Gbps capacity. Meanwhile, End-point codec in sub campus #1 and sub campus #2 use IP video phone over sub campus network in Gbps capacity. They are assigned to DN 1006 with IP address 10.12.101.1 in sub campus #1 and DN 1007 with IP address 10.11.101.1 in sub campus #2, respectively.

5.2 Quality of Service Measurement and Analysis

Measurement is carried out by seting channel capacity of WAN connection 128 kbps, 512 kbps, 1 Mbps and 2 Mbps. To do so, we operated serial connection to acquire the expected channel capacity instead of WAN connection (see our testbed network topology in Figure 2). We utilized H264 format of video traffic with transmission frame resolution 640x368 and reception frame resolution 352x288. Meanwhile, G722-Mono is utilized for audio traffic transmission. Table 2 shows the quality of service measurement for channel capacity of WAN connection 128 kbps. This measurement is taken from End-point Multiplay codec.

Table 2. Quality of service for channel capacity of WAN connection 128 kbps.

Traffic Type	QoS Parameter	DN: 1006		DN: 1007		DN: 1202	
		Transmit	Receive	Transmit	Receive	Transmit	Receive
Video	Protocol	H264	H264	H264	H264	H264	H264
	Resolution	640x368	352x288	640x368	352x288	640x368	352x288
	Frame Rate	30	3	30	3	30	30
	Channel Rate	307 kbps	21 kbps	248 kbps	21 kbps	49 kbps	317 kbps
	Total Packet Loss (%)	0.0 %	91.0 %	0.0 %	90.0 %	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	92.5 %	0.0 %	93.3 %	0.0 %	0.0 %
	Jitter	0 ms	143 ms	0 ms	143 ms	0 ms	1 ms
Audio	Protocol	G722-Mono	G722-Mono	G722-Mono	G722-Mono	G722-Mono	G722-Mono
	Channel Rate	64 kbps	23 kbps	64 kbps	23 kbps	63 kbps	63 kbps
	Total Packet Loss (%)	0.0 %	62.4 %	0.0 %	52.2 %	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	64.9 %	0.0 %	62.3%	0.0 %	0.0%
	Jitter	0 ms	43 ms	0 ms	28 ms	0 ms	5 ms

In the setting of 128 kbps WAN connection of video conferencing between End-point in main campus (DN: 1004) and End-points in sub campus #1 (DN: 1006) and sub campus #2 (DN: 1007), throughput of video transmission is 307 kbps and 248 kbps, respectively, whereas video reception is 21 kbps. In this situation, total packet loss of video reception is about 90% and jitter is 143 ms. On the other hand, the throughput of audio transmission is 64 kbps and 23 kbps for the reception with total packet loss of audio reception is 62% and 42%., respectively, whereas jitter is 43 ms and 28 ms, respectively. This shows that quality of service is very bad for channel capacity of WAN connection 128 kbps. Meanwhile, the quality of video conferencing between End-points in main campus (i.e., between DN: 1004 and DN: 1202) which has bandwidth in Gbps orde yields the throughput of video reception 317 kbps with jitter is only 1 ms and no packet loss, whereas the throughput of audio reception is 64 kbps with jitter is only 5 ms. Furthermore, by applying QoS in the Access Switch in sub campus with Trust CoS model, we can improve the quality of service as shown in Table 3. Again, this quality of service improvement is measured from End-point in main campus (i.e., DN: 1004).

Table 3. Quality of service for channel capacity of WAN connection 128 kbps with QoS.

Traffic Type	QoS Parameter	DN: 1007	
		Transmit	Receive
Video	Protocol	H264	H264
	Resolution	640x368	352x288
	Frame Rate	30	4
	Channel Rate	281 kbps	24 kbps
	Total Packet Loss (%)	0.0 %	84.3 %

	Current Packet Loss (%)	0.0 %	90.9 %
	Jitter	0 ms	125 ms
Audio	Protocol	G722-Mono	G722-Mono
	Channel Rate	64 kbps	26 kbps
	Total Packet Loss (%)	0.0 %	29.2 %
	Current Packet Loss (%)	0.0 %	57.2%
	Jitter	0 ms	21 ms

The throughput improvement after applying QoS of video traffic transmission increases 40 kbps and 3 kbps for the reception, respectively with packet loss decreases into 84% and jitter becomes 125 ms.

Table 4 and Table 5 show the result of quality of service measurement before applying QoS and after applying QoS with Trust CoS model for the channel capacity of WAN connection 512 kbps. Theoretically, the minimum standard of channel capacity of WAN connection to conduct a good quality of video conferencing is 384 kbps. Our experimental result shows that the throughput of video traffic transmission and reception is 135 kbps and 315 kbps, respectively without packet loss and jitter is only 6 ms for transmission and 8 ms for reception. Meanwhile, the throughput of audio traffic transmission and reception is 64 kbps without packet loss and jitter for transmission is 7 ms and 8 ms for reception.

Table 4. Quality of service for channel capacity of WAN connection 512 kbps.

Traffic Type	QoS Parameter	DN: 1006		DN: 1007		DN: 1202	
		Transmit	Receive	Transmit	Receive	Transmit	Receive
Video	Protocol	H264	H264	H264	H264	H264	H264
	Resolution	640x368	352x288	640x368	352x288	640x368	352x288
	Frame Rate	30	30	30	30	30	30
	Channel Rate	135 kbps	312 kbps	134 kbps	315 kbps	155 kbps	317 kbps
	Total Packet Loss (%)	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %
	Jitter	0 ms	6 ms	0 ms	8 ms	0 ms	1 ms
Audio	Protocol	G722-Mono	G722-Mono	G722-Mono	G722-Mono	G722-Mono	G722-Mono
	Channel Rate	64 kbps	63 kbps	64 kbps	64 kbps	63 kbps	63 kbps
	Total Packet Loss (%)	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	0.0 %	0.0 %	0.0%	0.0 %	0.0%
	Jitter	0 ms	8 ms	0 ms	7 ms	0 ms	0 ms

Table 5 shows that applying QoS for channel capacity of WAN connection above 384 kbps (i.e., 512 kbps) is not significant impact on changes in quality of service for video conferencing.

Table 5. Quality of service for channel capacity of WAN connection 512 kbps with QoS.

Traffic Type	QoS Parameter	DN: 1007	
		Transmit	**Receive**
Video	Protocol	H264	H264
	Resolution	640x368	352x288
	Frame Rate	30	30
	Channel Rate	155 kbps	317 kbps
	Total Packet Loss (%)	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	0.0 %
	Jitter	0 ms	8 ms
Audio	Protocol	G722-Mono	G722-Mono
	Channel Rate	64 kbps	64 kbps
	Total Packet Loss (%)	0.0 %	0.0 %
	Current Packet Loss (%)	0.0 %	0.0%
	Jitter	0 ms	7 ms

Table 6 shows the comparison of quality of service for channel capacity of WAN connection 128 kbps, 512 kbps, 1 Mbps, and 2 Mbps both video and audio traffic transmission and reception.

Table 6. Quality of service for various setting of channel capacity of WAN connection.

Traffic Type	QoS Parameter	128 kbps		512 kbps		1 Mbps		2 Mbps	
		Transmit	Receive	Transmit	Receive	Transmit	Receive	Transmit	Receive
Video	Throughput	281 kbps	24 kbps	155 kbps	317 kbps	279 kbps	313 kbps	321 kbps	346 kbps
	Packet loss	0.0 %	84.3 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %
	Jitter	0 ms	125 ms	0 ms	8 ms	0.0 %	3 ms	0.0 %	1 ms
Audio	Throughput	64 kbps	26 kbps	64 kbps	64 kbps	64 kbps	64 kbps	64 kbps	64 kbps
	Packet loss	0.0 %	29.2 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %	0.0 %
	Jitter	0 ms	21 ms	0 ms	7 ms	0 ms	2 ms	0 ms	1 ms

5.3 Impact of Trust CoS and Trust DSCP QoS Model

We measured Trust CoS and Trust DSCP from main campus to sub campus #2 to show the impact of enabling Trust CoS and Trust DSCP. We enabled Trust CoS for all interfaces of Layer 2 devices (i.e., switches), whereas teleconferencing data passing-through. Meanwhile, Trust DSCP is configured on all interfaces of Layer 2 devices and Layer 3 devices (i.e., routers).
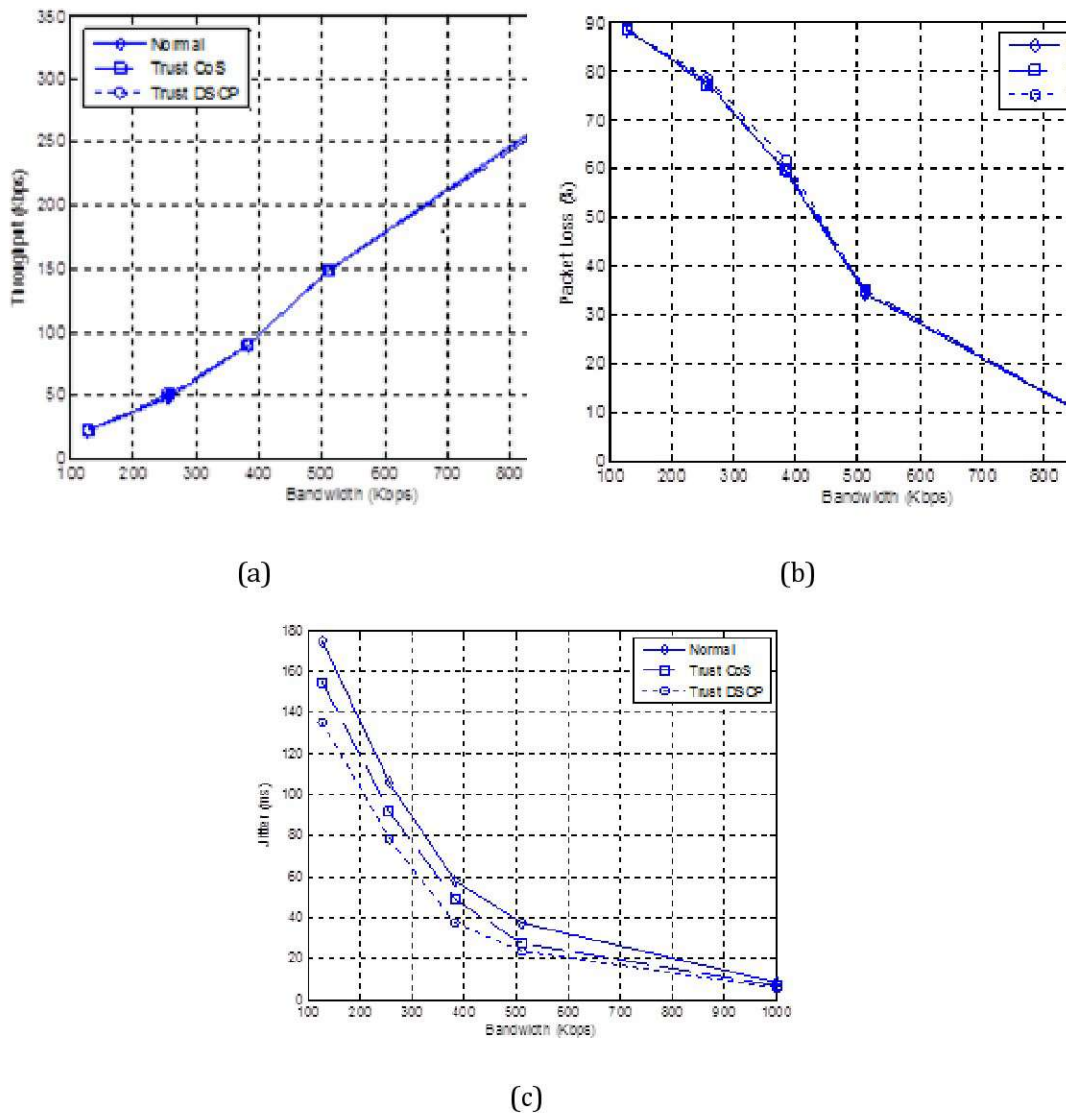
(a)

(b)

(c)

Figure. 3. Quality of service in our conferencing system comparison in term of (a) Throughput (b) Packet Loss, and (c) Jitter.

Figure 3 shows the quality of service comparison between 3 measurements, the first measurement is in the normal conferencing system which means no any QoS model inserted. The second and third measurement are inserted Trust CoS and Trust DSCP, respectively. Here, the impact of QoS model both Trust CoS and Trust DSCP is the priority of packet delivery which has critical state on the class-map, so that jitter becomes smaller whereas throughput and packet loss are relatively constant. Jitter decreases from 174.67 ms to 154.33 ms and in the second measurement. Meanwhile, it becomes 135 ms in the third measurement. In this case, jitter decreases 12.33% when the used of Trust CoS, and 22.41% when the used of Trust DSCP.

The second scenario, we flooded the path of conferencing traffic data with file transfer packet traffic. Figure 4 shows the impact of this additional traffic data on the conferencing system. Here the role of Trust CoS and Trust

DSCP is still in decreasing jitter. It is 11.5% and 20.99% decreasing when implementing Trust CoS and Trust DSCP, respectively.
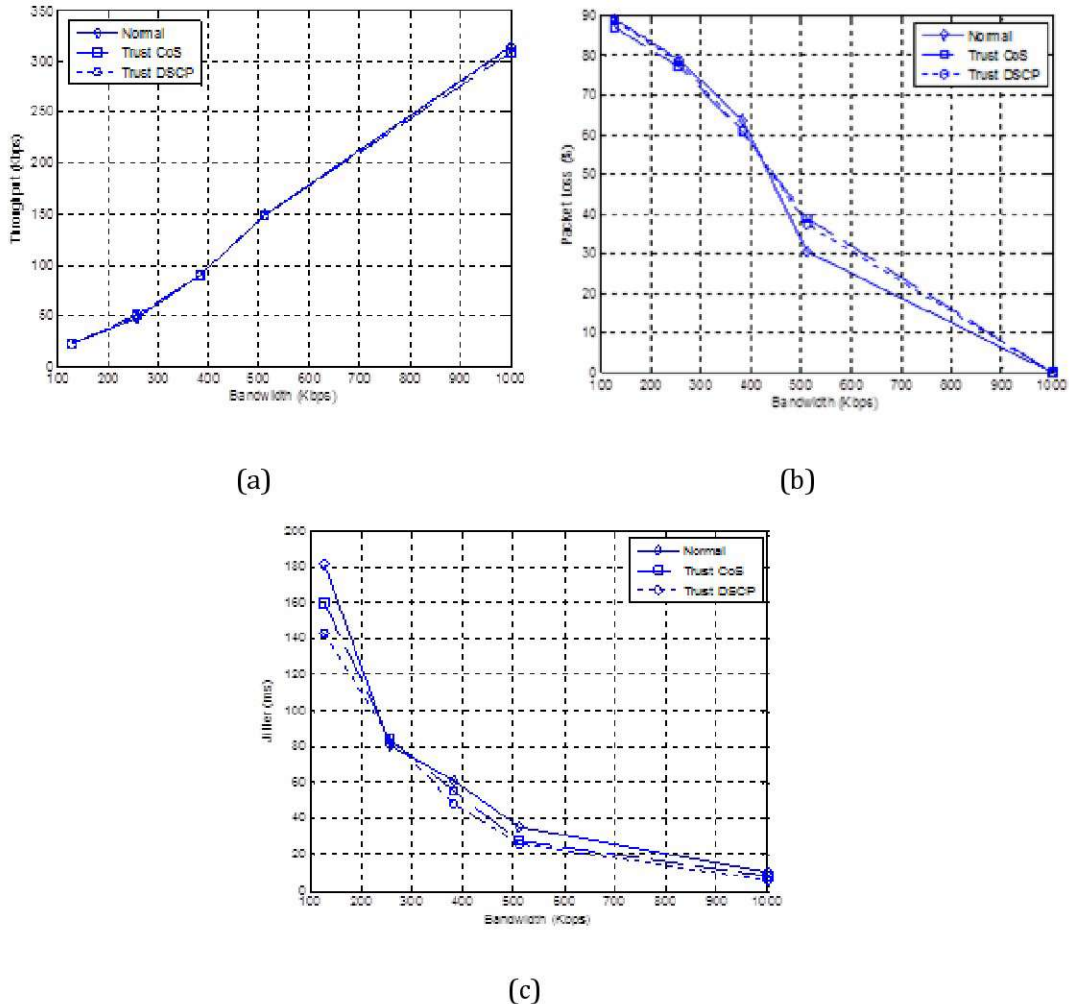

(a)   (b)

(c)

Figure. 4. Quality of service in our conferencing system comparison with inserting file transfer packet traffic in term of (a) Throughput (b) Packet Loss, and (c) Jitter.

The third scenario, we flooded WAN connection between main campus and sub campus #2 with UDP traffic data. Here, we utilized UDPFlooder as a tool for flooding UDP data into WAN connection. Figure 5 shows the impact of inserting UDP traffic data to the performance of conferencing system. Again, additional traffic data decreases jitter when packet delivered. The number of packet is varied from 0 to 10,000,000. Meanwhile, throughput and packet loss are relatively constant.
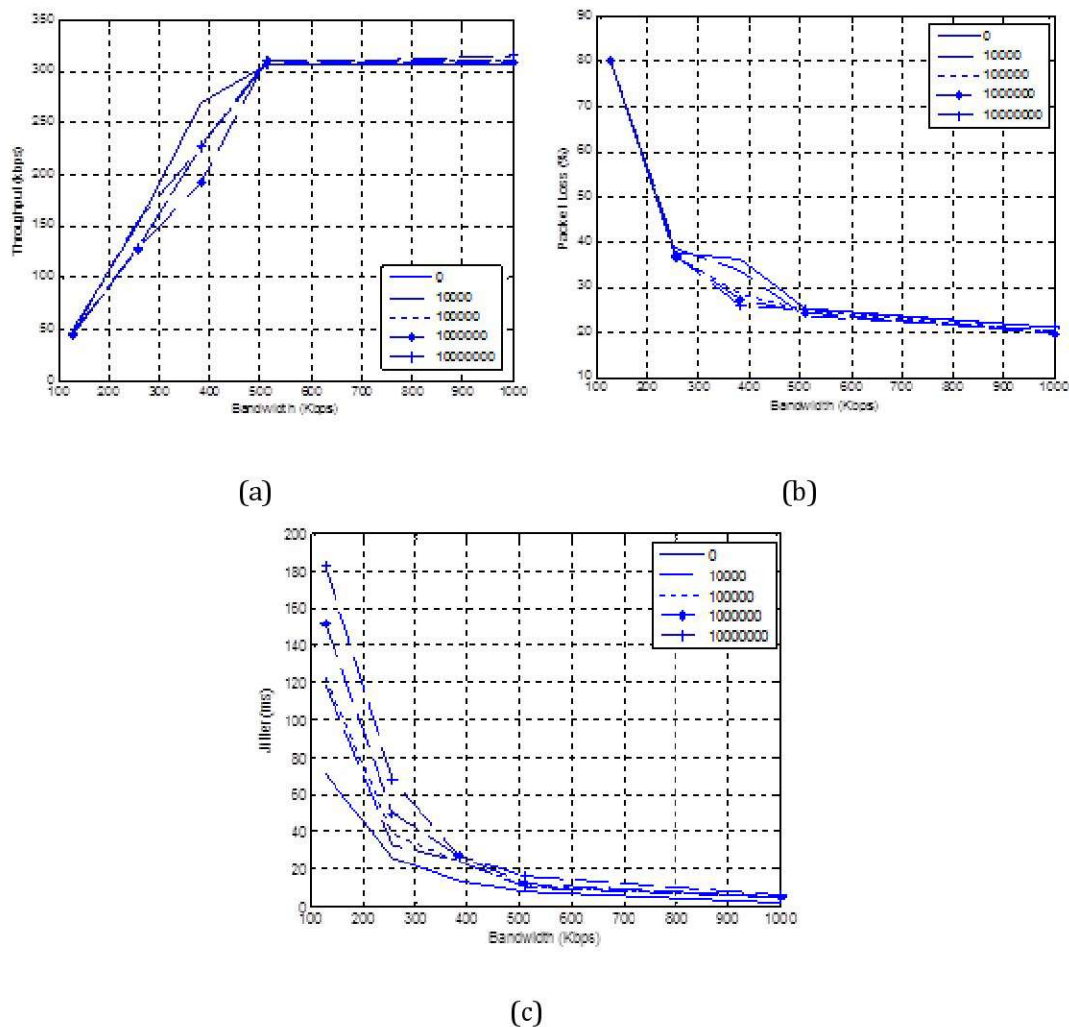
(a)   (b)

(c)

Figure. 5. Quality of service in our conferencing system comparison with inserting UDP data traffic in term of (a) Throughput (b) Packet Loss, and (c) Jitter.

6. CONCLUSION AND FUTURE WORKS

We have presented a testbed of video conferencing system for distance learning between End-point in main campus and End-points in sub campus for laboratory scale. Our experimental results show that the quality of service:

- for channel capacity (bandwidth) of WAN connection 128 kbps produces the throughput of video traffic transmission and reception is 281 kbps and 24 kbps, respectively with total packet loss is 84.3% and jitter is 125 ms, whereas the throughput of audio traffic transmission and reception is 64 kbps and 26 kbps, respectively with total packet loss is 29.2% and jitter is 21 ms;
- for channel capacity of WAN connection 512 kbps produces the throughput of video traffic transmission and reception is 290 kbps and 130 kbps, respectively with total packet loss is 35.4% and jitter is 25 ms,

whereas the throughput of audio traffic transmission and reception is 64 kbps with no packet loss and jitter is 14 ms;
- for channel capacity of WAN connection 1 Mbps or more (i.e., 2 Mbps), the throughput of video traffic transmission and reception is 279 kbps and 313 kbps, respectively, whereas the throughput of audio traffic transmission and reception is 64 kbps without packet loss and jitter is not more than 5 ms.
- applying QoS model in the conferencing system decreases jitter. Trust CoS and Trust DSCP model decrease jitter up to 12.3% and 22.41%, respectively.

Our future works include the application of QoS for all nework devices in access, distribution, and core levels where video/audio traffic passthrough. In addition, we also would like to optimize the quality of service by collaborating non-QoS parameters and QoS application to achieve better performance of video conferencing, and implement video conferencing system for real distance learning.

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