International Journal of Computing and Digital Systems

http://dx.doi.org/10.12785/ijcds/020205

Simulation of SIP-Based VoIP for Mosul University Communication Network

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Received 15 Oct. 2012, Revised 21 Nov. 2012, Accepted 20 Dec. 2012

Abstract: The opportunity to use data networks for telephone conversations is appealing. The technology to do this is commonly known as Voice over Internet Protocol (VoIP) or IP telephony, and has become widely available during the past few years. As part of our work concerning the deploying of Unified Communications (UC) for Mosul University network, this paper presents simulation for SIP-Based VoIP as one technology of UC technologies. Voice telephony is considered as real-time communication service and time sensitive service. To ensure if the University IP network is ready and capable for this new type of traffic before adding any new components, Mosul University IP network will be simulated using OPNET network simulation software then the new VoIP service will be added to the University network. Network performance and voice quality are tested after adding the VoIP for different codec types. The results show that Mosul University IP network is capable to implement the VoIP.

Keywords: VoIP, Session Initiation Protocol (SIP), IP Telephony, Simulation, OPNET, Analysis.

I. Introduction

VoIP stands for voice over internet protocol and is also known as IP telephony. VoIP is a recent technology that allows us to make voice calls using IP networks. Many network managers are finding it very attractive and cost effective to merge and unify voice and data networks into one. It is easier to run, manage, and maintain. However, one has to keep in mind that IP networks are best-effort networks that were designed for non-real time applications. On the other hand, VoIP requires timely packet delivery with low latency, jitter, packet loss, and sufficient bandwidth. To achieve this goal, an efficient deployment of VoIP must ensure these real-time traffic requirements can be guaranteed over new or existing IP network [1]. Most telecom operators have deployed IP technologies into their transportation network. The development of broadband digital access technologies, such as optical fiber, allows subscribers to use VoIP technologies for their voice communications. VoIP employs a variety of protocols, including RTP (Real-time Transport Protocol) for transport of multimedia data and SIP (Session Initiation Protocol [2]) or H.323 [3] for signaling, i.e., establishing and controlling sessions.

SIP is designed to integrate with other Internet services, such as email, web, voice mails, instant messaging, multi-party conferencing and multimedia collaboration. In our simulation we have designed a SIP-based VoIP system for Internet telephony. We choose the SIP as a VoIP protocol because it was designed to be very simple and with a limited set of commands. It is also text-based, so human can read the SIP messages passed between endpoints in a SIP session. The setup allows us to extend our PBX capacity and eventually replace it, while keeping our existing phone numbers. The environment provides inter-operability with the PSTN, programmable Internet telephony servers, IP-based voice mail, integration with web and email for unified messaging, multi-party multimedia conferencing, and inter-operability with existing multimedia tools.

The rest of the paper is organized as follows. Section 2 gives an overview of SIP. Section 3 the methodology of our work. The proposed VoIP network is described in Section 4. Simulation design is illustrated in Section 5.

We analyze some voice test factors in Section 6. Finally, conclusion is discussed in Section 7.

II. Overview of SIP

SIP stands for Session Initiation Protocol. SIP is an IETF (Internet Engineering Task Force) signaling protocol for session management for text and multimedia exchanges, like VoIP, instant messaging, video, on-line games and other services. SIP is designed for real-time transmission, uses fewer resources and is considerably less complex than H.323, the signaling protocol defined by the ITU for establishing voice and video conference calls across the internet.

SIP is a signaling protocol, which means that it is not actually responsible for transmitting the voice data, rather its purpose is to initiate (hence the name), coordinate and tear down a communication session between two endpoints - peers. Compared to a traditional telephone, the ringing of a phone, the busy tone and the ending of a call are all functions the SIP protocol is responsible for.

SIP has the ability to redirect calls through User Datagram Protocol (UDP) and Transmission Control Protocol (TCP); however, other VoIP protocols are not capable of supporting TCP and only support UDP. It has various features for example it allows media to be added or removed during the sessions and it also permits multicast conferences in the existing sessions [4].

SIP uses a Universal Resource Locator (URI) [5] to identify a logical destination, not an IP address. The address could be a nickname, an email address (e.g., sip:user1@uom.edu), or a telephone number. In addition to setting up a phone call, SIP can notify users of events, such as “I am online,” “a person entered the room,” or “e-mail has arrived.” SIP can also be used to send instant text messages.

SIP has two components: User Agents (UA) and SIP servers. User Agents are endpoints of SIP system, typically an IP Phone or IP Softphone. User agents could be either an agent client or an agent server. A user agent client initiates by sending a SIP request. A user agent server can accept, terminate or redirect the request as responses to this SIP request. There are three types of SIP servers include SIP proxy servers, SIP registrar servers, and SIP redirect servers. A SIP server functions as a server that handles these requests, e.g. requests transferring, security, authentication, and call routing.

Another interesting aspect of SIP is its ability to connect IP networks to the PSTN or other IP networks. Carriers are offering SIP trunking packages to enable an enterprise to connect by means of a session border controller (SBC) device to the PSTN or to another IP network.

III. Methodology

Currently Mosul University IP communication network provides the Internet service for university staff. To enhance the communication between staff in the campus, additional services can be added such as voice over IP (VoIP). We have simulating a real campus network to analyze and measure network capability of implementing VoIP service at Mosul University IP local area network. Fig. 1 shows the framework of the VoIP service in campus environment. There are four steps development process such as: 1) planning and research; 2) development; 3) simulation and 4) testing.

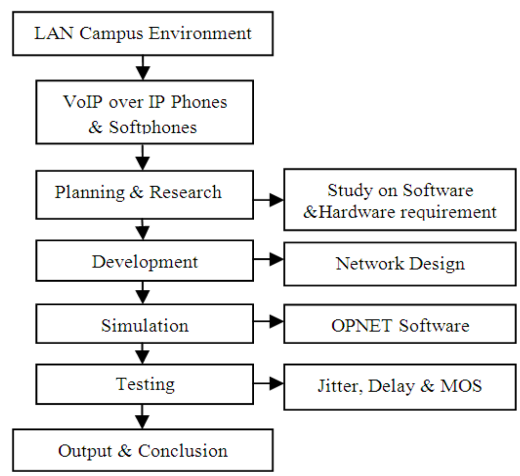


Figure 1. Framework of VoIP service in Campus Environment.

IV. Proposed VoIP Network for Mosul Campus Environment

The University of Mosul has implemented a traditional analog telephone system (Panasonic phone system) in the campus environment. The supported service is only calls between instructors in the campus. Now fast growing technological developments including multimedia communications requires fast paced and integrated services.

The Internet protocol provides a healthy environment for integrated services combing voice, video and data services. There are two facilities motivate us for deploying the VoIP in the campus such as: 1) the University has fiber optic backbone cables covering the campus; 2) Switches (Cisco Switch 2950) distributed in the campus support the VLAN technique and QoS which can improve the voice quality in the future.

Fig. 2 shows the proposed VoIP architecture for Mosul University Campus environment which consists of the following components:

A. User Agents (UA)

SIP user agents allow users to interact with the system over an IP connection. They can be either hardware (IP Phone) or software braced (IP Softphone).

B. SIP Server

SIP proxy is the redirect and registration server enabling the UAs to communicate with others.

C. SIP/TDM Gateway

Panasonic PBX (KX-TDE600) has virtual SIP/TDM Gateway card (V-SIPGW16) for connecting the PBX to the campus LAN.

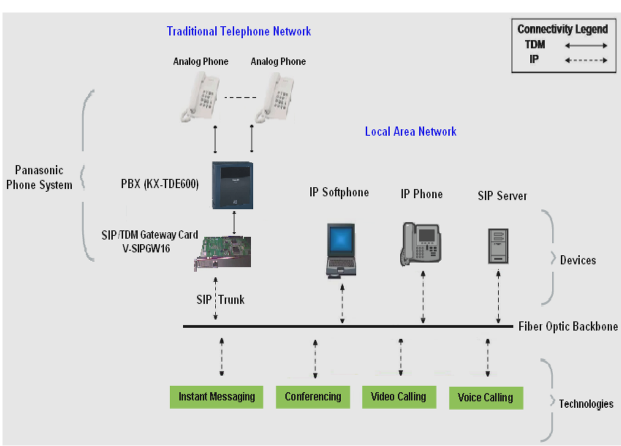


Figure 2. VoIP Architecture for Campus Environment.

V. SIMULATION

Mosul University IP-based network which covers most of its departments and colleges has been implemented in a star topology. Fiber optic cables are distributed around the campus of the university connecting university buildings. By using Google Earth software, Mosul University campus picture is captured and used as background for our simulation as shown in Fig. 3.

All University buildings are connected by fiber optic cables in a star topology with the core switch (Cisco Switch 6509) as of the real network.

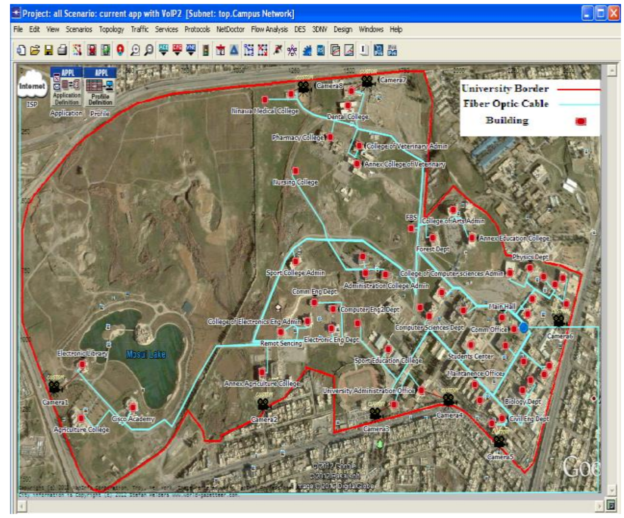


Figure 3. The whole network.

OPNET modeler 14.0 [6] is used as simulation software for the university campus network. In the whole network, every node represents a building having unique VLAN and subnet. Within each subnet, there is Cisco switch (Cisco 2950) as a main building switch, number of branch switches, hosts (PCs) and links (100 BASE-T).

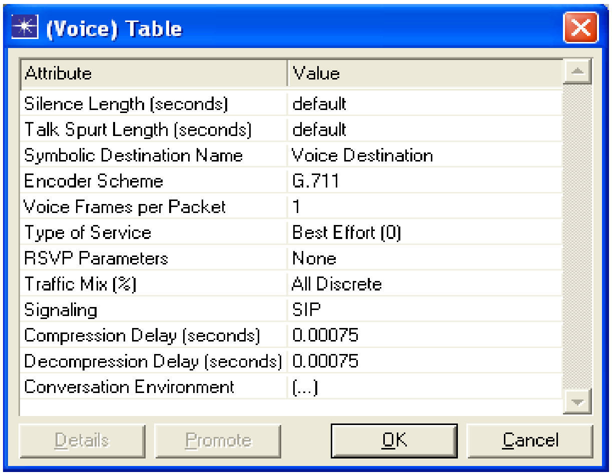
The type of service provided to the subnet should be set-up within the simulation configuration by using the Application Definition and Profile Definition shown in Fig. 3. In the application configuration the VoIP service is supported. Some main factors must be specified such as codec type, compression/decompression delay and the signaling VoIP protocol. A codec is software that converts audio signals into digital frames and vice versa. Codecs are characterized with different sampling rates and resolutions. Different codecs employ different compression methods, using different bandwidth and computational requirements.

The common codec types used in VoIP networks are G.711, G729 and G.723, table I presents the bit rate and compression delay for each codec type.

TABLE I. CODEC FACTOERS

Codec Type	Bit Rate (kbps)	Compression Delay (ms)
G.711	64	0.75
G.729	8	10
G.723	5.3	30

To choose the best codec for our network, three VoIP scenarios are simulated with different codec types. In each scenario, voice factors such as codec type and compression delay must be entered in the voice application table. Fig. 4 shows the voice application factors for the G.711 scenario.



Attribute	Value
Silence Length (seconds)	default
Talk Spurt Length (seconds)	default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.711
Voice Frames per Packet	1
Type of Service	Best Effort (0)
RSVP Parameters	None
Traffic Mix (%)	All Discrete
Signaling	SIP
Compression Delay (seconds)	0.00075
Decompression Delay (seconds)	0.00075
Conversation Environment	(...)

Figure 4. Voice application factors.

To test the network performance with the VoIP service, VoIP traffic will be added to the current network traffic. Then the VoIP traffic should be increased concurrently in the simulation to test the network performance. To generate VoIP traffic, IP Phones added to the university subnets in addition to the current connected PCs which are used for Internet browsing and FTP download. Fig. 5 shows one of university subnets after adding the IP Phones.

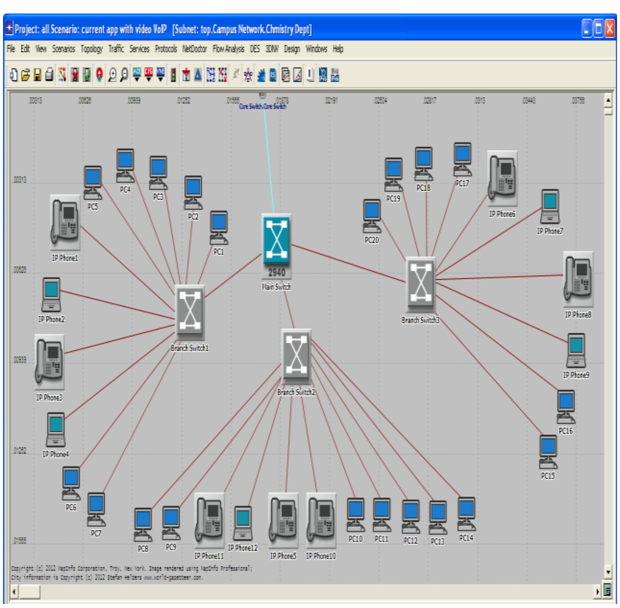


Figure 5. Chemistry dept. subnet.

SIP server must be added as a proxy server for supporting the SIP signaling; in our simulation we choose the computer center building where to place the SIP server. Fig. 6 shows the computer center LAN layout.

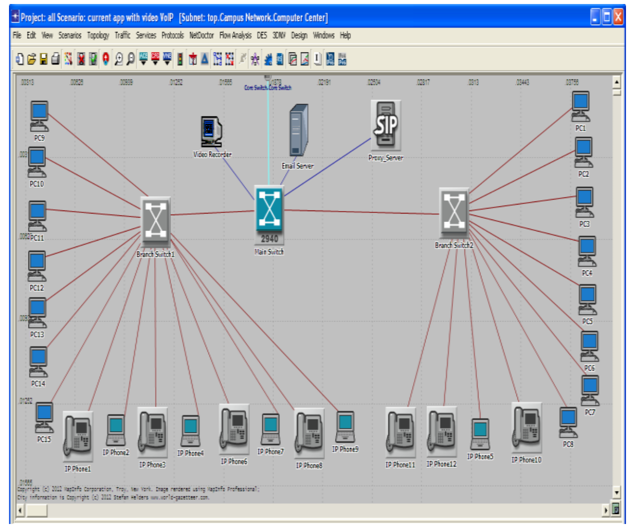


Figure 6. Computer center subnet.

After providing the IP Phones to all university buildings and defining the VoIP service, now the network is ready for running the simulation.

VI. Results and Discussion

After completing the VoIP network, simulation statistics must be specified before running the simulation. One of the useful statistics is the number of active SIP calls. In our simulation we need to increase the number of simultaneous SIP calls to test the network performance, as shown in Fig. 7 the number of SIP calls increases with simulation time. The duration for each call is set to 30 sec.

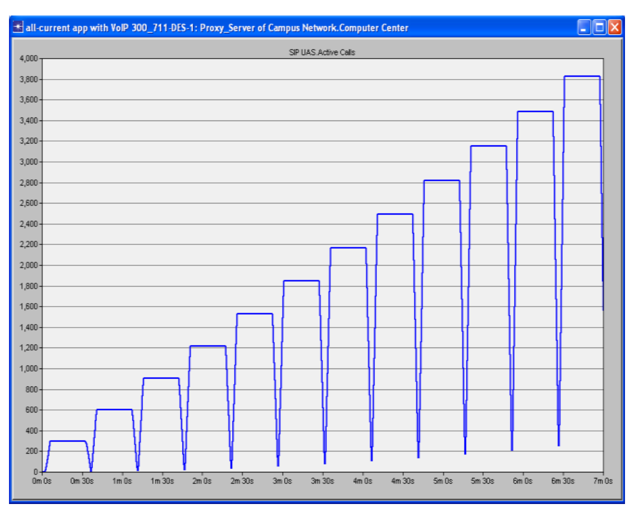


Figure 7. Active SIP calls.

The performance analysis will focus on delay, voice quality, bandwidth and Jitter over campus LAN. The simulation results are presented as follows:

A. Voice Delay

Delay is the amount of time it takes a packet to get from one end of the network to the other. Unlike many data applications, voice cannot tolerate high levels of latency or delay. Toll-quality voice requires that sounds take 100 ms or less to travel from the speaker's lips to the receiver's ear. Delays exceeding 150 ms can start to irritate callers. When delays approach 500 ms, voice communication becomes impossible without reverting to the old practice of saying "over" to signal the end of a transmission [7]. As shown in Fig. 8 the max voice end to end delay is 120 milliseconds for G.723 codec type. The min delay is 22 milliseconds for G.711 codec. 120 milliseconds voice delay is accepted value in ITU standard as shown in table II [8].

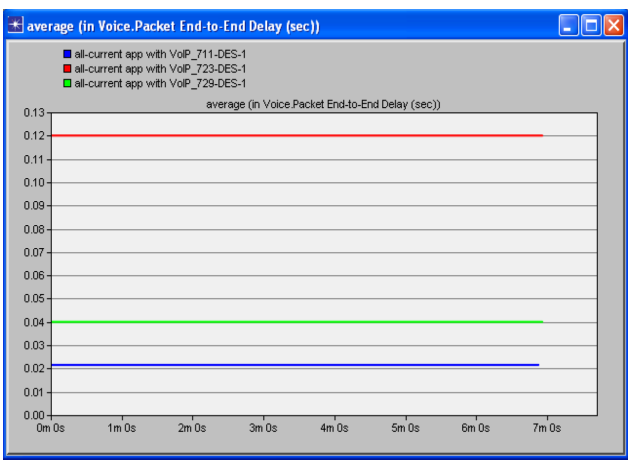


Figure 8. Voice End-to-End Delay Comparison.

TABLE II. ITU STANDARD RECOMMENDATION FOR DELAY.

Range in Milliseconds	Description
0 - 150	Acceptable for most user applications.
150 - 400	Acceptable provided that administrators are aware of the transmission time and its impact on the transmission quality of user applications.
Above 400	Unacceptable for general network planning purposes.

B. Voice Quality

In the VoIP, voice quality is measured by the Mean Opinion Score (MOS), which is a number between 1 and 5 used to quantitatively express the subjective quality of speech in communications systems, especially digital networks that carry VoIP traffic. Anything above a 4.0 is considered toll grade (see table III) [9].

TABLE III. MOS RATING MEASUREMENT

MOS	QUALITY	IMPAIRMENT
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very Annoying

Fig. 9 shows the MOS factor of our network for three codic types. We notice that the MOS factor for G.711 is greater than 4.0 which indicates good voice quality for our proposed service.

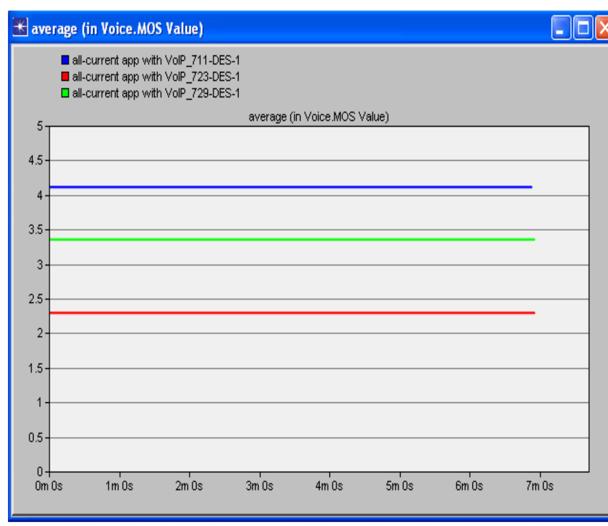


Figure 9. MOS factor.

C. Jitter (Delay Variation)

Another network quality issue to examine is jitter. Each packet of voice information takes a different amount of time to go from one end of the network to the other. This variation is called "jitter". Jitter is vital in voice communication because if transmission delay varies too widely during a VoIP conversation, the quality of voice will be degraded [10]. For successful VoIP implementation, the jitter value should not exceed (20–50) milliseconds. Fig. 10 shows the Jitter value for our proposed VoIP system.

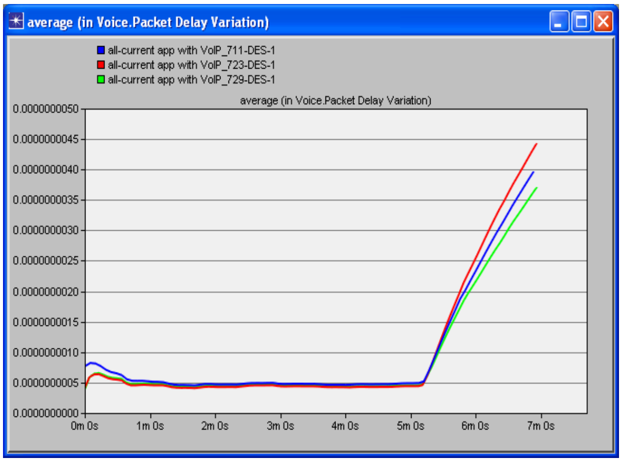


Figure 10. Voice Jitter.

As shown in Fig. 10, the jitter value is 0.5 ns for the duration of 5min simulation time then the value increased sharply. At 5min the number of simultaneous calls raised to 2800 VoIP simultaneous calls, which cover the university need.

D. Voice Bandwidth

One of the important factors for testing the codec performance is the required bandwidth for voice transmission. As shown in Fig. 11, G.711 consumes larger bandwidth with respect to G.723 and G.729 because no data compression is done.

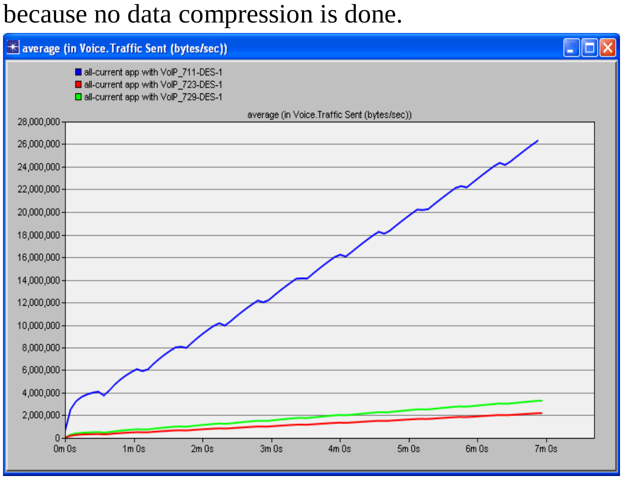


Figure 11. Average VoIP traffic.

VII. Conclusion

Refer to our previous work [11], simulation for the complete real communication network of the Mosul University is carried out using OPNET software. As one of UC services, VoIP is added to the university network. Three codec types (G.711, G.723 and G.729) are used and investigated considering their impact on delay, voice quality, bandwidth and jitter. Results show that G.711 gives the better value of Jitter, delay and MOS factors at the expense of bandwidth. G.723 and G.729 codecs give better performance from bandwidth point of view.

It is recommended to add the VoIP service to the university network using G.711 type since it has optical fiber infrastructure.

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