Voice over Wired and Wireless Communication for Academic Institutions

John Benedict L. Bernardo, MIT and Consorcio S. Namoco, Jr., D.Eng.

Abstract—**An effective communication system is very important requirement for an efficient academic institution. Most academic institution in the Philippines relies only on a phone-based communication to establish contacts with and any of its continuants. Phone-based communication, however limit communications between offices and hence, contact can no longer be established in a place that has no telephone lines. In this study, a Voice over Internet Protocol (VoIP) system over wired and wireless communications for the academic institution has been proposed and developed. This system can be used to make and receive calls by simply accessing the web browser and dialing registered contacts. The main advantages of this proposed system is that it can be accessed using smartphones, laptops or personal computers.**

Index Terms—**VoIP, Mobile, Wired, Wireless Technologies, Communications, Academic Institutions.**

I. Introduction

Communication is essential within the Academic Institutions. Nowadays, a digital network is mainly use in the entire information that provides a new way of communication through the combination of telecommunication and computer technology. It allows voice communication to be deployed on campus networks. These networks are being explored and studied so that existing data networks in a campus can be used optimally. The use of telecommunication services will incur additional expense in the operation of the organizations; thus, there is a need of finding new alternatives and solutions to make the telecommunication cheaper, more efficient and cost effective [1].

During the past 10 years, cellular telephones (wireless devices that can access networks and inter\intra nets) have evolved from a rarity to an everyday accessory. These and other wireless devices prompted the International Telecommunications Union, the Institute of Electrical and Electronics Engineers, and the European Telecommunications Standards Institutes to develop standards for wireless communications. Users of such devices have the flexibility, convenience, and can access information from any location [2]. The cellular world, as a means of telecommunication in general, and that of smartphones in particular, is an ever moving and changing world, a reality that has been highlighted in recent studies of worldwide trends. A smartphone is a mobile phone that offers more advanced computing ability and connectivity than a contemporary featured phone. Smartphones and featured phones may be thought of as handheld computers integrated with a mobile telephone, but while most featured phones are able to run applications based on platforms such as Java ME [3], a smartphone allows the user to run and multitask applications that are native to the underlying hardware. Smartphones run complete operating system software providing a platform for application developers.

With the immense growth of digital networks, the combination of telecommunication technology and computer technology provides a new way of communication. Networks such as Wired local area network (LAN) and Wireless LAN are being explored and studied so that existing data networks in a campus can be used optimally. Voice over wired LAN was first deployed around 2003 and was mostly used in establishments with many employees like hospital environments, manufacturing and warehousing environments. Employees can get in touch with each other through the VoWLAN services and yet incurring no extra cost to management [4]. The implementation of VoIP networks in institutions is expected to increase the network flexibility, simplify the network infrastructure, enable new feature rich and integrated applications and reduce voice toll costs for out-of-campus calls. In fact, circuit-switched technologies that have worked well for voice communication in the past are now starting to be replaced by VoIP [5].

Considering that communication is a vital tool in the social and economic development of a country, the availability of quality and affordable basic communication services to everyone is very essential. The existence of free or affordable communication services improves access to market information, reduces the cost of general ownership and promotes wide usage. Hence, it is imperative that communication tools and infrastructures be provided to the various stakeholders of the academic institutions to effectively interact among each other, and thus carry out their assigned duties and tasks more efficiently. However, the requirement for communication devices and facilities oftentimes are not met due to budgetary constraints and limited resources. This is primarily due to the fact that most academic institutions have other priorities in spending their meager resources, such as scholarship, personnel services, libraries, laboratory and building improvements, and many others.

In order to address this concern, the establishment of a wired and wireless VoIP communication in the academic setting is proposed. Wireless VoIP is a considerable influence over the communication networks of the educational system. Since majority of academic institutions have computer facilities with high-speed Internet connection, the establishment of wireless access points throughout the campus is all that is required so that a cost-effective

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J. L. Bernardo and C. S. Namoco Jr. are faculty members of the College of Industrial and Information Technology, Mindanao University of Science and Technology, Cagayan de Oro City, Philippines
(e-mails: bernardo_jbl@yahoo.com and csnamocojr@yahoo.com).

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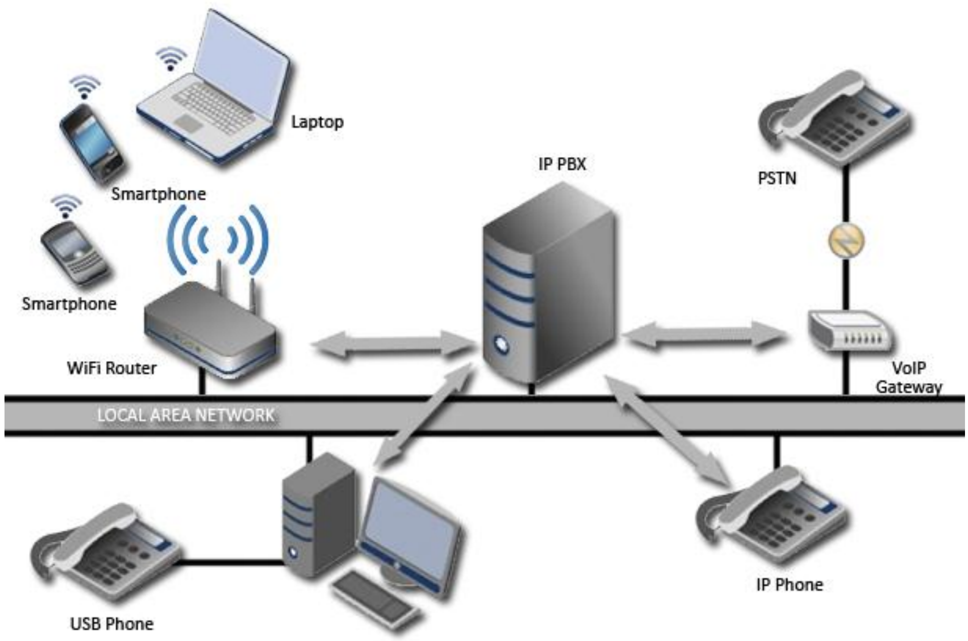

Fig. 1. Hardware Network Connectivity

communication system can be provided. Communication can be made through the use of personal computers, laptops, tablet PC and smartphones. The number of different devices using wireless communications is rising rapidly. Mobile computers are regularly connected to e-mail and Internet services through wireless communications, and wireless local area networks. Hence, this study considered a wireless communication system for the proposed MUST VoIP system.

This study designed wired and wireless LAN Framework and implemented a voice communication system in a University setting to provide a sustainable and cost effective access to telecommunication services. The system was implemented in Mindanao University of Science and Technology (MUST), Cagayan de Oro City, Philippines utilizing the existing network of the university.

II. METHODOLOGY

System Implementation

A program was designed and developed using appropriate languages, techniques and codes that met the user's needs. The VoIP module in the project was implemented using a SIP/WebRTC based HTML5 client entirely written in JavaScript. The SIP protocol was used to establish, modify, and terminate voice and video sessions. The actual data, comprising of voice packets would be communicated by transferring and receiving RTP based voice packets encapsulated in UDP packets (using the UDP Runtime Library).

Fig. 1 illustrates the point-to-point connection of hardware components through the networks. The VoIP server is connected to all wired and wireless devices such as smartphone, laptop and IP Phone to establish communication.

The system uploaded in VoIP server was designed to be ready for use by students, faculty and staff of any academic institution. To test the functionality of the proposed study, the proposed VoIP system was installed in the local area network of the MUST. The system can easily be accessed by logging into a specified IP address. However, each user is required to obtain an account from the system administrator. This is important because the personal user account is used in place of phone numbers to establish the connection between peers.

The MUST- Information and Communication Technology (ICT) Team implemented the MUST's network topology in 1997. The local area network is capable to handle data traffic for an average of 100(Mbps) Megabits per second. Fig. 2 shows the connections in all offices and computer laboratories connected in ITB Building where the Server is maintained, which suggest that all network communication activities pass through this server. The VoIP system is implemented in this network topology setting. Additional VoIP server was added to be able to apply the voice communication system.

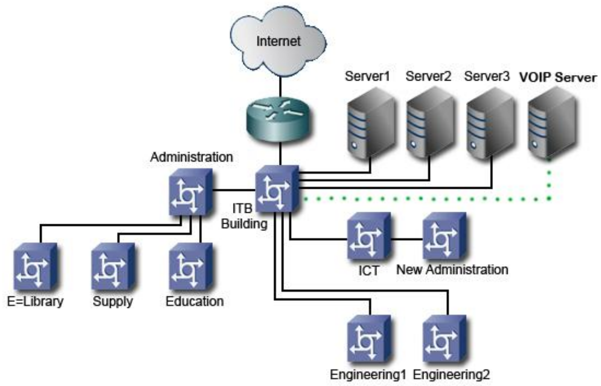

Fig.2. Architectural Design of Networks in MUST with Additional VoIP Server

VoIP Server was added to establish a peer-to-peer connection on a university local area network between the sender and receiver. The local SIP server was then run and

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configured. SIP is an application-layer control protocol for creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls and multimedia distribution. SIP invitations were used to create sessions and carry session descriptions that allow participants to agree on a set of compatible media types. SIP then made use of proxy servers to help route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users. Asterisk, an open source software, was used to establish SIP communication services. In this study, the Asterisk was customized by implementing Secure Real-time Transport Protocol (or SRTP), a profile of RTP (Real-time Transport Protocol), intended to provide encryption, message authentication and integrity, and replay protection to the RTP data in both unicast and multicast applications as WebRTC only allows communications using secure / encrypted connections.

In order for the users to communicate with each other, account for each user in the channel driver that corresponds to the protocol were configured since MUST-VoIP is using the SIP protocol. The accounts were configured in the SIP channel driver configuration file, called sip.conf. (This file resides in the Asterisk configuration directory, which is typically /etc/asterisk.) For example, user names for a certain faculty and a certain student were configured as Student1 and Faculty 1, respectively, to easily differentiate between the two. These accounts were then assigned their respective user ID. Fig. 3 shows the configured users account.

```
;created Student1

[Student1]
type=peer
username=00001 ; userid for student
host=dynamic
secret=test00001 ; put a strong, unique password here
context=default    hasiax = no
hassip = yes
transport=udp,ws,wss
encryption = yes
avpf = yes
icesupport = yes
videosupport=yes
directmedia=no

;created Faculty1

[Faculty1]
type=peer
username=11111 ; userid for student
host=dynamic
secret=test11111 ; put a strong, unique password here
context=default
hasiax = no
hassip = yes
transport=udp,ws,wss
encryption = yes
avpf = yes
icesupport = yes
videosupport=yes
directmedia=no
```

Fig. 3. Configured account in sip.conf

The client can use different operating systems to access the system connected via SIP network to asterisk server using a web browser to make and receive audio and video calls. No extension, plugin or gateway is needed for the proposed system to work.

Evaluation Performance

Performance evaluation was conducted to evaluate if the proposed VoIP system works efficiently. A survey questionnaire was made for this purpose. The respondents of the survey include IT/non-IT faculty, office staff, ICT technical staff and students, who are the expected users of the proposed VoIP system.

In the conduct of the evaluation, it was ensured that all respondents have the necessary communication device, such as laptop, smartphone, or PC with headset, and that Google Chrome web browser has been installed in their device. The respondents were then asked to use the proposed system to make and receive calls. They were then asked to evaluate the system based on specified criteria in the questionnaire. This includes ease of use, functionality, expandability, quality of service and reliability. The respondents were asked to evaluate proposed system using both the wired and wireless connections. Respondents were asked to evaluate the system using a 5-point scale ranging from 1 (Strongly Disagree) to 5 (Strongly Agree).

III. RESULTS AND DISCUSSION

The main purpose of this study is to come up with a VoIP system over Wired and Wireless Communication for academic institutions. The MUST VoIP system is an alternative tool of the university to make the telecommunication cheaper, more efficient and cost effective.

The performance of the MUST-VoIP system for wired and wireless communication are shown in Table 1. For wired communication, all of the respondents agree that the system is user friendly, has the ability to support network users, capable to handle more than two simultaneous users on the network, has clear voice quality and is a good communication tool within the University. However, for wireless communication, the users are not satisfied in terms of expandability, functionality, and quality of service since it can only accommodate of 2 callers simultaneously since the MUST wireless connection is running a typical SOHO (Small Office/Home Office) Wi-Fi router designed to be used by a very small business. The existing network devices in the MUST Network infrastructure needs to be upgraded or replaced in order to support the extra traffic that will be generated from the deployment of the proposed MUST VoIP system and allow more simultaneous calls for both wired and wireless connection. Changing the network design for greater efficiency, reconfiguring or tuning the network for QoS, or a combination of these may also be done in order to enhance the communication in the academic campus.

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TABLE 1: PERFORMANCE OF THE MUST-VOIP SYSTEM FOR WIRED AND WIRELESS COMMUNICATION

Parameters	Wired			Wireless		
	n = Simultaneous Call	**Mean**	**Adjectival Rating**	**n = Simultaneous Call**	**Mean**	**Adjectival Rating**
A. Ease of Use	n=1	3.9	Agree	n=1	4.8	Strongly Agree
	n=2	4.3	Agree	n=2	4.8	Strongly Agree
	n=3	4.3	Agree	n=3	1.0	Strongly Disagree
	n=5	4.3	Agree	n=5	1.0	Strongly Disagree
	n=7	4.4	Agree			
B. Expandability	n=1	3.6	Agree	n=1	5.0	Strongly Agree
	n=2	3.9	Agree	n=2	4.0	Agree
	n=3	4.0	Agree	n=3	1.0	Strongly Disagree
	n=5	4.0	Agree	n=5	1.0	Strongly Disagree
	n=7	4.1	Agree			
C. Functionality	n=1	4.3	Agree	n=1	5.0	Strongly Agree
	n=2	4.2	Agree	n=2	4.3	Agree
	n=3	4.2	Agree	n=3	1.0	Strongly Disagree
	n=5	4.2	Agree	n=5	1.0	Strongly Disagree
	n=7	4.3	Agree			
D. Quality of Service	n=1	3.3	Neutral	n=1	3.1	Neutral
	n=2	3.4	Neutral	n=2	2.8	Neutral
	n=3	3.4	Neutral	n=3	1.0	Strongly Disagree
	n=5	3.4	Neutral	n=5	1.0	Strongly Disagree
	n=7	3.5	Agree			
E. Reliability	n=1	4.1	Agree	n=1	5.0	Strongly Agree
	n=2	4.2	Agree	n=2	4.2	Agree
	n=3	4.3	Agree	n=3	1.0	Strongly Disagree
	n=5	4.3	Agree	n=5	1.0	Strongly Disagree
	n=7	4.3	Agree			

IV. CONCLUSION

The VoIP system for academic institution has been developed study and tested at the MUST. Evaluation of the proposed system shows the MUST-VoIP system is easy to use for making and receiving calls for up to 7 simultaneous users over wired communication, however, it can be used efficiently up to 2 simultaneous users over wireless communication.

There is an urgent need to upgrade the existing network devices in the MUST Network infrastructure to support the additional traffic generated in the deployment of the system and to make it capable in allowing more simultaneous calls for both wired and wireless connections.

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Authors Profile:

John Benedict L. Bernardo was born in Hagonoy Bulacan, Philippines, on the 2nd of May 1987. He is a Master of Information Technology graduate from Mindanao University of Science and Technology, Cagayan de Oro City Philippines. He presently works as an Instructor in the said university. He specializes in the field of computer network and programming.

Consorcio S. Namoco, Jr. was born in Bohol, Philippines on April 16, 1975. He graduated his Doctor Engineering from Kyoto Institute of Technology, Kyoto City, Japan last 2012. His research interests are in field of metal forming, computer simulation, materials processing, industrial technology and information technology education. Presently, he is a full professor and the dean of the College of Industrial and Information Technology, Mindanao University of Science and Technology, Cagayan de Oro City, Philippines. He also serves as editor board member and peer reviewer to various local and international research journals.

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