ISSN: 0976-3031

Available Online at http://www.recentscientific.com

International Journal of Recent Scientific Research
Vol. 6, Issue, 6, pp.4389-4394, June, 2015

International Journal of Recent Scientific Research

RESEARCH ARTICLE

INVESTIGATING THE PERFORMANCE OF VOIP OVER ETHERNET LAN IN CAMPUS NETWORK

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ARTICLE INFO

Article History:

Received 5th, May, 2015
Received in revised form 12th, May, 2015
Accepted 6th, June, 2015
Published online 28th, June, 2015

Key words:

VOIP, Internet, QoS, Ethernet LAN, Telecommunication, MSN.

ABSTRACT

VoIP (Voice Over Internet Protocol) is an advanced technology that has a great potential to develop new telecommunication with much lower cost and better QoS. VoIP, a new technology has been increasingly popular in recent times due to its affordability and reduced cost in making calls over broadband Internet. In this Paper, we shall analyze the performance of VoIP over digital communication on popular application such as Skype and MSN. Voice over Internet Protocol (VoIP) is a rapidly growing technology that enables transport of voice over data networks such as Ethernet local area networks (LANs). This growth is due to the integration of voice and data traffic (telecommunication convergence) over the existing networking infrastructure, low cost, and improved network management offered by the technology. In the investigation, the impact of increasing the number of VoIP clients, voice codec schemes, and traffic distribution on system performance is considered. This study provides an insight into the VoIP performance over Ethernet LANs.

INTRODUCTION

In recent years, there is a growing trend in real-time voice communication using Internet protocol (IP). Voice over Internet Protocol (VoIP) is a technology that allows users to make telephone calls over an IP data network (Internet or Intranet) instead of traditional Public Switched Telephone Network (PSTN). Therefore, VoIP provides a solution that merges both data and voice which gains benefits include cost savings, high quality and value added services. Today, VoIP is becoming one of the most widely used technologies, more and more people and organizations are using VoIP systems worldwide. There are various VoIP communication software products are already available on the internet: Skype, Google Talk, and Windows live messenger. All of them can provide good quality, cheap, and even free phone calls.

VoIP is not only popular through the internet; it is also a rapidly growing technology through data networks such as Ethernet LANs. Ethernet is considered a good platform for VoIP as Ethernet based LANs is very common in enterprises and other organizations for data networking. Therefore, there is a tremendous growth of VoIP. This growth is due to the integration of voice and data over the existing networking infrastructure, low cost, and improved network management offered by the technology. In addition, wireless Ethernet networks (IEEE 802.11) allow mobile users to connect to the network from the location where network cables may not available or may not be the best choice, such as old buildings, Hospitals, and conference rooms. Therefore, WLANs are other important segments for VoIP deployments. The performance of VoIP over WLANs is also investigated in this dissertation.

Overview of VoIP technology and comparison between PSTN and VoIP

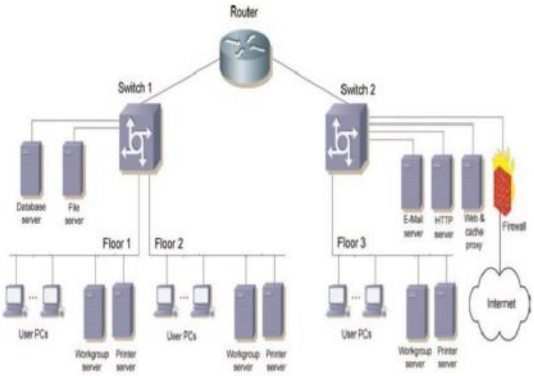

Figure 1 Typical VoIP network

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A tremendous increase in popularity over the real-time voice communication over Internet protocol (IP) is observed in recent years. Voice over Internet Protocol (VoIP) is a high modern technology that provides the capability of users to generate telephone calls over an IP data network (Internet) using packet-switching technology instead of the traditional Public Switched Telephone Network (PSTN). With the creation of VoIP, nowadays digital communication has greatly reduced in cost while preserving high quality service, as VoIP's capabilities to merges both data and voice in a single channel.

PSTN versus VoIP: A Feature Comparison

PSTN	VoIP
• Dedicated Lines • Each line is 64kbps (in each direction) • Features such as call waiting, Caller ID and so on are usually available at an extra cost • Can be upgraded or expanded with new equipment and line provisioning • Long distance is usually per minute or bundled minute subscription • Hardwired landline phones (those without an adapter) usually remain active during power outage • When placing a 911 call it can be traced to your location.	• All channels carried over one Internet connection • Compression can result in 10kbps (in each direction) • Features such as call waiting, Caller ID and so on are usually included free with service • Upgrades usually requires only bandwidth and software upgrades • Long distance is often included in regular monthly price • Lose power, lose phone service without power backup in place • 911 emergency calls cannot always be traced to a specific geographic location

Not only limit to internet, VoIP has also shown its popularity in data networks such as Ethernet LANs. The reason is that Ethernet would be an ideal data networking platform for enterprises and other organizations to establish LAN communication, as it provides its user with a high level quality of services while greatly reducing cost comparing to the traditional PSTN. In addition to that, wireless Ethernet networks (IEEE 802.11) allow distance digital communication for users to connect to the network, which is ideal for locations that are difficult to setup tool such as Hospitals, two offices located far apart and conference rooms.

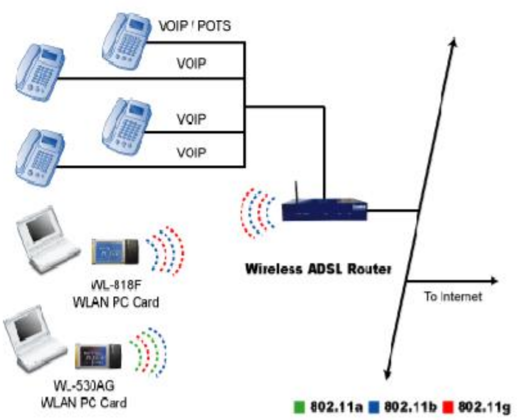


Figure 2 Ethernet of VoIP

Objectives of this study

Despite the potential benefits of VoIP over Ethernet LANs, one of the significant challenges faced by designer of VoIP is to provide a quality of service (QoS) to all users on the network, especially under medium-to-high traffic loads. However, IP networks were originally designed for data networking, not for voice, and additionally, an IP network is shared and utilities by many different devices and services. Unlike the classical applications such as file transfer or mail, VoIP is a real time service, the access competition can result in delays or packets lost which is detrimental to real-time applications.

The purpose of the project is to conduct several of test cases in VoIP by constructing different simulation scenarios under OPNET 14 Software. The reason behind it is that the successful implement of the project would reflect the advantage of VoIP over the traditional PSTN and thus proving that VoIP would be ideal candidate for the modern technology in network communication.

The aim of this research was to investigate the effect of the following factors on system performance:

- Increasing the number of VoIP clients
- Traffic arrival distributions
- Voice codec schemes

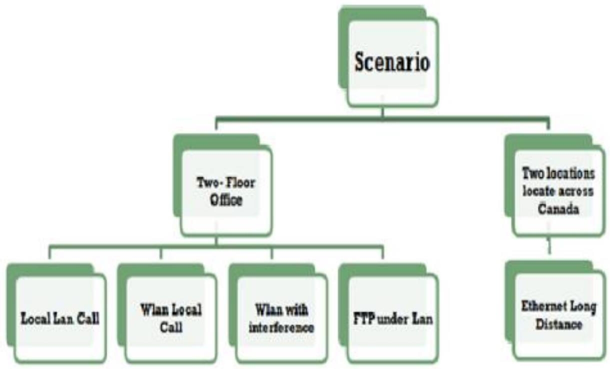


Figure 3 OPENT simulation scenarios break down

Important Concepts

There are three important concepts available in VOIP.

1. End to end Delay
2. Jitter
3. Packet Loss
4. Delay

End to End Delay

End to end Delay is the total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system. Delay time is one of the most important factors in determining the quality of a call. Echoing has been a major problem that is caused by end to end delay. However, delay is able to be kept as small as possible by utilizing the project model/ topology. Typically, for optimum VoIP call quality, end to end delay must be less than 150ms.

Jitter

Jitter is one the most common VoIP problems. Jitter is the undesired time delay from the packets sending end to receiving end in VoIP or other video communication network. The jitter can be affected by computer usage, the length and quality of the Ethernet cables and some other issues. The delay is inevitable and high levels of jitter leads to large numbers of packets to be discarded 8 / 29 by the jitter buffer in the receiving IP phone or gateway. This will result in severe distortion in call quality or large increases in delay.

Packet Loss

Failure or one of more packets to reach their destination across the network is recognized as packet loss. The occurrence of lost and dropped packets are extremely noticeable with real time streaming technology such as Skype and online gaming. On another hand, there is always a degree of packet loss allowance in almost every network. There are possible causes that lead to packet loss, such as channel congestion, corrupted packets rejected in-transit and poor networking hardware. To properly recover the loss packets, reliable network transport protocol such as TCP are used insure an acceptable and stable transmission. Using the Acknowledge technology, the network can reassure that the packets have been successfully delivered. In our simulation, we would try to maintain a maximum of 10% packet loss threshold.

Delay

High QoS should be assured by control delay so that one-way communication delay should be less than 150 ms. (ITU states that one-way, end-to-end telephony applications should have less than150 ms delay in echo-free environments to ensure user satisfaction). Delay mainly comes from three components: (1) delay caused by voice codec algorithms (2) delay caused by queuing algorithms of communications equipment (3) variable delay caused by various factors (i.e. network conditions, VoIP equipments, weathers etc). It is very important to minimize the voice traffic delay. Thus, a codec algorithm and queuing algorithm needs to be carefully considered. Although traditionally think the end-to-end delay of 150 ms was considered as acceptable for most applications. In this study, 200ms will be considered as the maximum acceptable one way end-to-end delay, high end-to-end delay can cause bad voice quality perceived by the end user.

Types Of Voip

There are several different types of VoIP service depending on the infrastructure used for the communication:

- Computer-to-computer based VoIP (VoIP device to another VoIP device);
- Computer-to-Phone based VoIP (VoIP device to a PSTN device); and
- Phone-to-Phone based VoIP (PSTN device to another PSTN device).

Computer to Computer Internet telephony services via computers are totally free VoIP services. This type of VoIP services via specialized software applications (soft phone software) such as Skype, AOL Instant Messenger, and MSN Messenger etc. These services require users to download their software and get them installed on PC, Caller and receiver need to use same VoIP software application (For instance, Skype to Skype, MSN to MSN etc), caller and receiver are communicated based on peer-to-peer approach through the Internet.

The requirements for computer to computer Internet telephony include:

- Soft phone software,
- A sound card
- Internet access

Computer to Phone Because the Internet and conventional circuit switched telephone systems use different systems. Thus, soft phone software need to routes the call through internet protocol and hands it off to a conventional telephone network. Skype, MSN, and Google Talk also provide services to users make phone calls from computers to typical landline phones.
Equipment requirements:

- VoIP service subscription
- Internet access
- A modem
- An Analog Terminal Adapter (ATA) that converts the analog call signal to digital signal (and vice versa).

Phone to Computer Users can make phone calls from traditional landline phones to computers with this service. A phone number will be assigned to a computer’s IP address. A user can dial this number just like making normal phone calls. Therefore, wherever you are, you can receive phone calls on your computer from landline phones via the number assigned. Skype now allows users to purchase phone to computer VoIP services.

Phone to Phone This is the ultimate step of VoIP services. Currently, many telephone companies already use this service to handle long distance calls. In the future, telephone companies are able to use the internet to handle all the telephone calls. Therefore, VoIP services completely do not need the traditional PSTN for both call origination and termination.

Voip System

Figure 4 shows a typical VoIP network topology that includes following equipments:

- Gatekeeper
- VoIP Gateway
- VoIP Clients

Gatekeeper

A gatekeeper or call manager node is optional for a VoIP network. In an H.323 IP telephony environment, a gatekeeper works as a routing manager and central manager that manage all the end nodes in a zone. A gatekeeper is useful for handling VoIP call connections includes managing terminals, gateways and MCU's (multipoint control units). A VoIP gatekeeper also provides address translation, bandwidth control, access control. Therefore, A VoIP gatekeeper can improve security and Quality of Service (QoS).

Figure 4 A typical VoIP Network Topology

VoIP Gateway

A VoIP gateway is also required to handle external calls. A VoIP gateway functions as a converter that converting VoIP calls to/from the traditional PSTN lines, it also provides connection between a traditional PBX (Private Branch Exchange) / Phone system and an IP network.

VoIP Clients

Other required VoIP hardware includes a VoIP client terminal, a VoIP device could be an IP Phone, or a multimedia PC or a VoIP-enabled workstation runs VoIP software.

Voip Protocols

There are two standard protocols used in VoIP network:

- H.323
- Session Initiation Protocol (SIP)

H.323

H.323 is ITU (International Telecommunication Union) standard based on Real-time Protocol (RP) and Real-Time Control Protocol (RTCP); H.323 is a set of protocols for sending voice, video and data over IP network to provide real-time multimedia communications. H.323 is reliable and easy to maintain technology and also is the recommendation standard by ITU for multimedia communications over LANs.

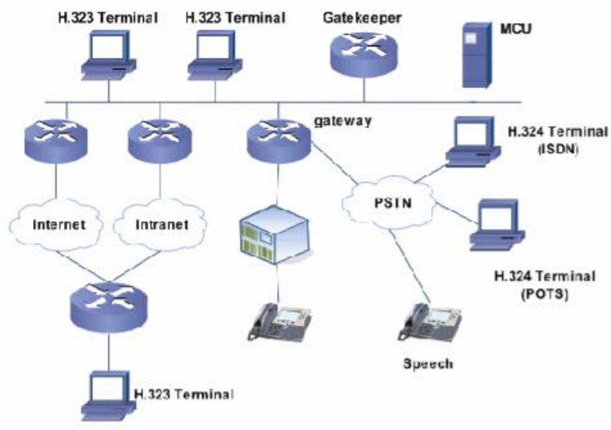


Figure 5 H.323 Architecture

There are four basic entities in a default H.323 network: terminal, gateways (GW), gatekeepers (GK) and multipoint control units (MCU): H.323 terminal also called H.323 client is the end-user device. It could be IP telephone or a multimedia PC with another H.323 client.

That provides real-time two-way media communication. A Gateway (GW) is an optional component that provides inter-network translation between terminals. A Gatekeeper (GK) is an optional component provides address translations and access control services. A Multipoint Control Unit (MCU) functions as a bridge or switch that enables three or more terminals and gateways in a multipoint conference.

Session Initiation Protocol (SIP)

H.323 has some limitations such as lack of flexibility, thus another protocol SIP is getting popular in VoIP. SIP (stands for Session Initiation Protocol) was developed by the Internet Engineering Task Force (IETF) and published as RFC 3261. SIP is a signaling control protocol which is similar to http, it's designed to initial and terminate VoIP sessions with one or more participants. It is less weight and more flexible than H.323 that also can be used for multimedia sessions such as audio, video and data.

SIP has two components: User Agents and SIP servers. User agents are peers in a SIP. 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.SIP is not only popular in VoIP applications but also widely used in applications include instant messaging and some other commercial applications, e.g. Microsoft MSN Messenger, Apple iChat.

Reasons For Voip Deployment

Cost Saving

This can be achieved by reusing the devices and wiring for the existing data network as most of the organizations already have their own networks. However, the most attractive reason to adopt VoIP maybe is dramatically reduced phone call cost. Soft phones such as Skype [5] enable PC-to-PC users can bypass traditional long-distance toll calls charge as voice traffic over the Internet, they only need to pay flat monthly Internet-access fee. Soft phones also allow a PC as a VoIP phone to call a mobile phone or a home line phone at a lower rate.

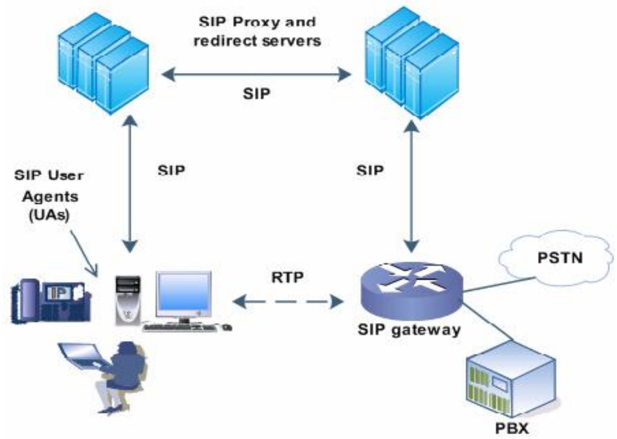


Figure 6 SIP Architecture

Advanced multimedia applications

Cost effective is only one of the good reasons to use VoIP. VoIP also enables multimedia and multi-service applications that increase productivity and create a more flexible work environment, e.g. real time voice-enabled conferencing systems that may include white boarding, file transferring, etc. which combine both voice and data features.

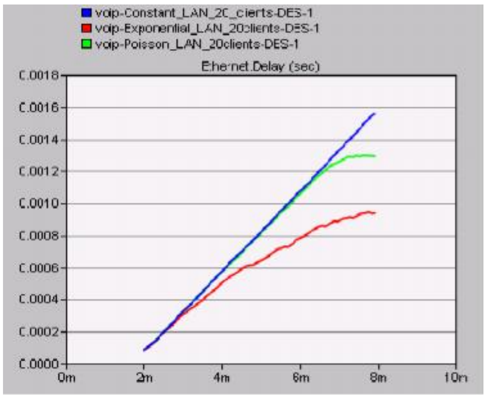
RESEARCH METHODOLOGY

Simulation methodology was adopted in this dissertation for the performance modeling and analysis of VoIP over Ethernet LANs. Simulation has become a popular approach for network studies and performance modeling. Some simulation-based studies for VoIP network system recently, these studies deal with performance and perceptual quality of VoIP network system. Another important reason of using simulation is that it can be easily control the scale of the network.

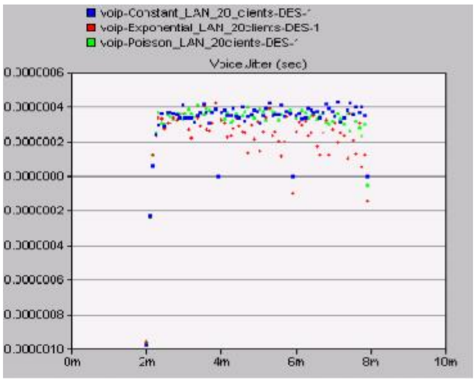
This dissertation is going to investigate the performance of VoIP using OPNET based on planned and designed network scenarios. In, the authors present a survey study that investigated VoIP performance over wireless networks and their study shows VoIP performance of wireless networks were much worse than wire line networks. They identified the maximum number of simultaneous voice connections that could be supported for a reliable wireless voice communication, and they suggest MAC (Medium access control) protocol, queue management schemes, voice codec choice and payout buffer algorithms as effective way to improve the VoIP performance over WLANs.

Simulation is widely used research methodology for network performance evaluation. This is because sometimes a networks may contains a large number of network nodes and services, it will be too much time consuming and costly to establish physical networks. Thus, network planners and engineers are often use simulation before deploying real networks.

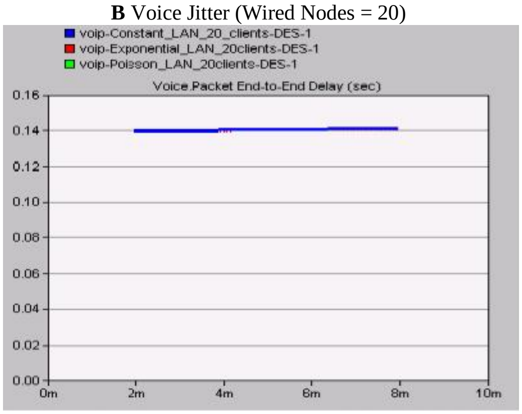
Their reported results include voice end-to-end delay, delay variation for each call, and Ethernet performance parameters. However, all these studies only considered wired network. That is all the previous studies in literatures based on either wired or wireless network. Therefore, it is necessary to do investigation on both wired and wireless network from different aspects. This dissertation investigates performance VoIP over Ethernet LANs include both wired and wireless network components. The findings of this dissertation can help organizations make decisions for adopting of VoIP system and expansion plans for VoIP services.



A Ethernet Delay (Wired Nodes = 20)



B Voice Jitter (Wired Nodes = 20)



C Packet End-to-End Delay (Wired Nodes = 20)

Graph 1 Performance of the VoIP Traffic Distributions

CONCLUSION

This study evaluated the performance of VoIP over Ethernet LANs. This study also measured different real time communication parameters, such as packet end-to-end delay, jitter, and tried to determine the maximum number of VoIP calls which the network can support. The number of VoIP clients has significant impact on VoIP performance for both wired and wireless LAN. Especially for wireless LAN, the impact of increasing the number of wireless nodes will be more than impact of the increasing the number of wired nodes. The major factors that affect VoIP quality such as delay, jitter and packet loss, are measured by simulation. Various issues related to the deployment of VoIP are also discussed. These issues include VoIP security and traffic characteristics and QoS requirements. Through the results we got found out that Ethernet has a more stable and less delay connection than wireless connection. Interference near wireless router greatly reduces QoS.

Future Work

This study only considered peer-to-peer voice calls. VoIP conferencing and messaging options are suggested as future research. This study considered VoIP traffic only. In future studies, more realistic traffic applications such as background traffic, FTP, and Email can be considered. As a result, although VOIP has some negative sides, we consider that VOIP is a great alternative way to replace the traditional circuit switch phone network in the future.

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How to cite this article:

S.Dhanalakshmi *et al*, Investigating The Performance Of Voip Over Ethernet Lan In Campus Network. *International Journal of Recent Scientific Research Vol. 6, Issue, 6, pp.4389-4394, June, 2015*

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