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In the Name of Allah, the Beneficent, the Merciful

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LIST OF ABBREVIATIONS

MFSP	Multiple Frames into a Single Packet
IP	Internet Protocol
3GPP	Third Generation Partnership Project
3G	Third Generations
NS2	Network Simulator second version
OTcl	Object-Oriented Tool Command
PSTN	Public Switched Telephone Network
CBR	constant Bit Rate
VBR	Variable Bit Rate
VoIP	Voice over IP
UDP	User Datagram Protocol
VSAT	Very Small Aperture Terminal
TCP	Transmission Control Protocol
RTP	Real Time Protocol
QoS	Quality of Service
LAN	Local Area Network
WAN	Wide Area Network
PC	Personal Computer
ISP	Internet Service Provider
RSVP	Resource Reservation Protocol
RTCP	Real Time Control Protocol
MAC	Medium Access Control
FDDI	Fiber Distributed Data Interface
PCM	Pulse Code Modulation
ITU	International Telecommunication Union
ACELP	Algebraic Code excited Linear Predication
DSP	Digital Signal Processor
KL	Kuala Lumpur
PEN	Penang
PBX	Private Branch Exchange

PENYIASATAN KEADAH CEKAP DEMI MEMPERBAIKI KUALITI PERKHIDMATAN UNTUK PROTOKOL SUARA MELALUI INTERNET (VOIP) UNTUK SATELIT

ABSTRAK

Kepentingan Protokol Suara Melalui Internet (VoIP) semakin diperakui oleh industri telekomunikasi. Sejak dekad yang lalu banyak perisian VoIP telah dibangunkan yang menawarkan banyak faedah kepada kedua-dua pembekal perkhidmatan rangkaian dan telekomunikasi. Bagaimanapun kualiti perkhidmatan yang wujud dalam rangkaian VoIP kini tidak setanding dengan kualiti suara dalam system telefon tradisional. Kelemahan VoIP kini termasuklah penggunaan lebarjalur yang tinggi, kehilangan paket, kesesakan trafik, dan masa lengah penghantaran lampau.

Objektif kajian ini ialah untuk memperbaiki prestasi VoIP malelui sambungan satelite. Ia dicapai secara menggunakan gateway Protokol VoIP satelit yang mengurangkan bilangan paket VoIP yang dihantar serta mengurangkan penggunaan lebarjalur secara meningkatkan bilangan bingkai VoIP yang dihantar dalam setiap paket. Protokol yang dicadankan adalah Protokol Bingkai Berganda Dalam Satu Paket (Multiple Frame Single Packet-MFSP). Perisian simulasi rangkaian (NS2) versi 2.28 telah digunakan untuk menguji pelbagai senario yang berbeza untuk mengenalpasti keberkesanan MFSP.

Senario yang dikaji termasuk berbagai pengekod-penyahkod yang menggunakan saiz kandungan berbeza, kesan kegunaan Kadar Bit Berubah VBR (Variation Bit Rate) berbanding dengan Kadar Bit Tetap CBR (Constan Bit

Rate) ke atas penggunaan jalurlebar, serta kesan protokol MFSP ke atas masa penghantaran dan perbezan masa penghantaran yang dialami.

Kajian ini mendapati protokol MFSP bukan saja mengurangkan bilangan paket VoIP yang dihantar melalui rangkaian malah ia juga mengurangkan penggunaan jalurlebar. Tambahan MFSP megatasi masalah paket kecil yang dihantar melalui sambungan satelit yang bermasa lengah tinggi. Berdasarkan keputusan yang diperolehi, MFSP adalah suatu mekanisma baru yang dapat memperbaiki kualiti perkhidmatan keseluruhan VoIP secara berkesan.

AN INVESTIGATION ON AN EFFICIENT APPROACH OF IMPROVING QUALITY OF SERVICE OF VOIP OVER SATELLITE

ABSTRACT

The importance of Voice over Internet Protocol (VoIP) is slowly being recognized by the telecommunications industry. In the past decade, many VoIP applications have been developed, offering a wide range of benefits to both telecommunications and network service providers. However, the Quality of Service currently available in VoIP networks is not comparable to traditional telephone voice quality. The current limitations of VoIP include relatively high bandwidth consumption, packet loss, traffic congestion and excessive delay.

The objective of this study is to improve the performance of VoIP over satellite communication links. This is achieved by implementing satellite VoIP protocol gateways that would reduce the number of transmitted VoIP packets and reduce bandwidth usage by increasing the number of VoIP frames sent per packet. The proposed protocol is called Multiple Frames Single Packet (MFSP). Network simulator (NS2) version 2.28 was utilized to evaluate different scenarios to determine the effectiveness of MFSP. Scenarios including different codecs with different payload sizes, the effect of Variable Bit Rate (VBR) vs. Constant Bit Rate (CBR) vis-à-vis bandwidth consumption, as well as the effect of MFSP on packet delay and jitter were studied.

The study found that the MFSP protocol not only reduces the number of VoIP packets sent through the network but it also reduces the overall bandwidth usage. Moreover, MFSP addresses the problem of smaller packets transmitted over satellite links with high propagation delays. Based on the results, it could be seen that MFSP is an effective mechanism through which the overall QoS of VoIP over satellite links can be enhanced.

CHAPTER 1 INTRODUCTION

1.0 Introduction

This chapter presents a brief introduction of the thesis. It gives general background information on Voice over Internet protocol (henceforth abbreviated as VoIP) technology, different components needed for the VoIP over Satellite Communication network as well as how existing VoIP packets are organized. The chapter then covers the problem statement and objectives of the study. The chapter also presents how the study will be implemented and the application used.

1.1 Background information

VoIP is a rapidly evolving technology that has revolutionized the telecommunication industry (Samrat, et al, 2006). During the last few years, the VoIP technology has gained increased popularity (Samrat, et al, 2006). The growth of the Internet has directed a lot of interest towards VoIP. This technology has considerably reduced the telecommunication costs for the end-user and thus it has been used to some extent to replace the traditional long-distance communication. However, in order to make the VoIP commercially viable, the physical infrastructure and quality of service (henceforth abbreviated as QoS) needs to be at least close to the one provided by the Public Switched Telephone Network (henceforth abbreviated as PSTN).

VoIP associated technology has been able to bring to the end user value added services that are currently not available in the PSTN.

In fact, VoIP has recently emerged as the latest and greatest Internet application. Furthermore, the VoIP application extends to incorporate the Satellite and Wireless environments as well as the cellular networks. All those combinations points VoIP being a growing industry and it indicates that it will dominate telecommunication sector in the future. In fact, many cellular network operators applying to transport are intent on using VoIP direct to the cellular phones through WiFi (Wireless Fidelity) or WiMAX (World Interoperability for Microwave Access) (Grech, 2000)

The above-described statement suggests where the future of the Telecommunication technology is heading. This has been a source of encouragement for carrying out this study. Certainly, for full end-user acceptance and deployment, the VoIP service should provide a reasonable voice quality. There are many challenges for achieving reasonable voice quality. Among of them, packet loss, delay, and delay jitter as well as additional overhead brought by the VoIP protocol stack. For instance, it is likely that the packet switched technology generates more delay jitter than the circuit switched technology initially used for data applications such as email and file transfers (Worster, 1998).

For a successful VoIP deployment, voice quality must match up to well with the voice quality which users have come to expect from PSTN. To achieve comparable voice quality over the Internet, packet loss must be prevented and packet delay minimized for all packets carrying digitized voice (Bates, 2000).

The focus of this study is on enhancing the QoS of VoIP by minimizing the traffic throughput of the link and at the same time maximizing the bandwidth utilization for long propagation delay such as satellite links. Satellite bandwidth is costly. Therefore, internet service providers are keen to find a solution, which reduces bandwidth usage. The study also strives to find out ways to decrease the VoIP traffic burst size (i.e. the number of packets traveling on the network) as well as to address the primary factors that contribute to poor QoS: packet loss and delay jitter.

1.2 VoIP over Satellite Communication

For multi-location businesses, connecting remote offices can be challenging especially when DSL and cable are not available to all areas of a country, especially in the rural areas. Satellite communications has made it possible to reach such areas. Although communications over satellite can be an expensive transmission medium, it still provides an effective high-speed IP data link. Combined with VoIP, satellite communications proves to be an even more efficient connectivity option as it virtually eliminates traditional telephony for long distance costs.

For example, MultiVoIP gateway[1] is an ideal for businesses with multiple locations connected via satellite. MultiVoIP connects directly to phones, fax machines, key systems, or a PBX and plugs into the IP data network to provide real-time, toll-quality voice connections to any remote office over a satellite link. It is a point-to-multipoint solution (one gateway is required at each location). It simply merges voice from the telephone onto the IP network and then utilizes another MultiVoIP gateway, at the remote end to separate the voice from the data network and send it back to the PBX, telephone, or fax machine. This function provides significant savings on inter-office long distance charges (Richharia, 1999).

1.3 Problem Statement

VoIP is a relatively new concept in telecommunication industry and it has been faced many problems for it to be accepted as an alternative medium compared to existing telephone systems. Since this is a new technology and is in its infancy, it is important to address those factors that have an affect on its use as a viable medium. Among those factors are.

Bandwidth mismanagement

Congestion

Packet loss

Delay

Delay jitter

The way that the existing VoIP packets are organized is a single frame pattern results, unnecessary packets overhead and high bandwidth utilization.

[1] MultiVoIP gateway is a hardware product, which provides voice communications over the Internet or Intranet.

It creates also the network congestion, which results packet loss as well as increasing overall traffic throughput of the link. However, the VoIP industry needs more research to find a solution, which reduces the above-mentioned deficiencies.

This study is designed to come up with a packet assembly technique that line up several voice frames into larger packets, without causing intolerable delays in packet processing. Figure 1.1 graphically illustrates how multiple small VoIP packets are sent through network resulting in congestion, high bandwidth consumption and increased delay.

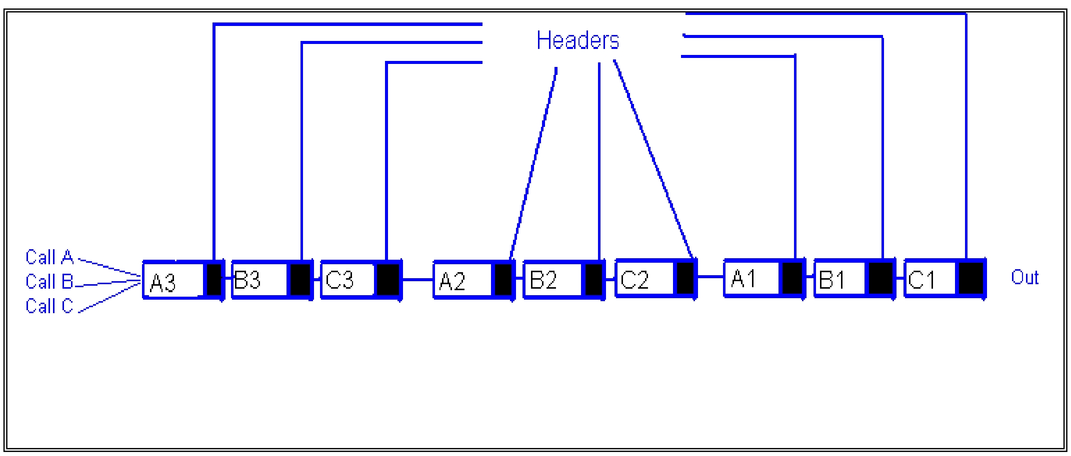

Figure 1.1: Many smaller VoIP packets through the network
Source: Author

1.4 Study Objectives

- The main objective of the study is to improve the bandwidth utilization of expensive satellite links for VoIP traffic:
- Determine the impact of high propagation delays through geostationary satellite links on VoIP.

- Develop a scheme to increase the throughput of VoIP traffic while reducing the overall bandwidth utilization of the satellite link.
- Analyze the performance of the proposed scheme via simulation experiments.

1.5 Scope of research

The scope of this thesis is to investigate issues relating to QoS in VoIP over Satellite links. It examines the existing VOIP packet payloads contents and proposes a technique, namely the (M.F.S.P) protocol to mitigate the particular challenges of transporting VoIP over Satellite links by reducing the total bandwidth utilization in order to improve network performance necessary for QoS support. In other words, the MFSP protocol increases the amount of payload per packet in order to avoid network congestion and maximize bandwidth utilization. Other factors such as call admission control and subjective via quality measurement will not be addressed in this thesis.

1.6 Implementation Method

For implementation purpose, a network simulator (henceforth abbreviated as NS2[2]) application version 2.28 was utilized. NS2 is the preferred software because it gives simulation both satellite and non-satellite networks.

[2] NS2 is a discrete event simulator meant for networking research. NS2 is written in OTcl (Object-Oriented Tool Command language) and C++. OTcl is its primary Command and Configuration Language. It implements network protocols such as Transmission control Protocol (henceforth abbreviated as TCP) and User Datagram Protocol (henceforth abbreviated as UDP) over wired and wireless (both local and satellite) networks, and also traffic source behavior such as FTP, Telnet, Web, CBR and VBR ((Peterson, 2000, Kevin, 2003).

1.7 Thesis Organization

The thesis is organized into six chapters. Chapter one outlines the problem statement and objectives of the study as well. The chapter starts with an overview of VoIP technology, and different components needed for VoIP over a Satellite Communications network. The limitations of VoIP and how, the existing VoIP packets are organized is highlighted. The chapter then outlines the problem statement and objectives of the study as well as the scope of the research. The chapter finally presents the implementation method for the study.

Chapter two presents a review of the literature pertinent to the topic of the study. The chapter starts with providing brief background information on the development of telecommunication in general and today's last mile technology. In addition, the chapter outlines the VoIP problems among of convergence network and network design as well as delay issues and jitter delay. Different types of VoIP protocols will be discussed. It pointed up the QoS for VoIP. It presented solutions for improving the QoS such as RSVP, DiffServ and Int-Serv. In addition, it presented an overview of VoIP over satellite as well as QoS of VoIP over satellite.

Chapter three introduces the proposed protocol namely Multiple Frame Single Protocol. Furthermore, the chapter discusses how the existing VoIP problems can be overcome through the MFSP protocol. . It illustrated relation of the MFSP design and Codecs behaviors as well as MFSP and data sources. Finally, the chapter presents a number of limitations for the MFSP protocol.

Chapter four covers implementation issues for the proposed protocol starting with the topology used in the NS-2 implementation and its main components, network equipments and its main accessory. Furthermore, the chapter discusses the essential physical equipments, Variable Bit rate (VBR) and Constant Bit Rate (CBR) packet source as well as system capacity. The chapter finally deals with the software tool used for setting up VoIP Gateway Modeling in NS2, and MFSP protocol employed in different scenarios and settings.

Chapter five introduces findings of the study. The chapter gives an analysis on the simulated data as well as discussion based on the findings of the study. In addition, the chapter discusses MFSP protocol experiments carried out to evaluate VBR and CBR traffic types. In addition, a comparison between the MFSP and existing single packet VoIP implementation is done. It outlines testing of three different codecs. The bandwidth utilization, network congestion, delay jitter and propagation delay results obtained through simulation is then discussed.

Chapter six contains the conclusion and recommendations based on findings of the study.

1.8 Summary

This chapter discussed the background information of VoIP and how the VoIP has revolutionized the telecommunication industry. It presented the importance of the combination of VoIP and satellite, in particularly the rural or remote area where the regular connection is not available. The chapter presented problem statement of the thesis, the research objectives as well as the scope of the research. Finally, it illustrated the implementation of thesis as well as how it organized.

CHAPTER 2 LITERATURE REVIEW

2.0 Introduction

This chapter presents a review of literature on telecommunications technologies with emphasis on the developments of VoIP communication technology and QoS. Various types of VoIP protocols, IP networks, network design, causes of VoIP limitations such as voice delay, packet loss, jitter and voice circuit compression over IP from end to end process are discussed presented additionally VOIP weaknesses and an overview of satellite communications.

2.1 Telecommunication Development

Telephony was initially analogue meaning that a telephone conversation required a pair of copper wires to carry the speech path. Both sides of the conversation were carried on a single pair. This was sufficient for short distances. However, the longer the distances, the louder one had to speak. Eventually, when the distances become too long for shouting, an electronic amplification had to be added. This required two pairs of wires since the two speech paths require separate amplification, one for the mouthpiece and one for the earpiece. The second revolution was the emergency of Digital Telephony Systems (DTS). DTS made it possible to convert the speech from an analogue signal to a binary signal and once the speech is digitized, transmission and switching become simpler. Binary systems are relatively simple, reliable with enormous capacity and above all cheap. DTS considerably improved the QoS.

In addition, DTS has made easier to transmit a noise free binary signal as opposed to analogue system whereby voice is amplified along with the voice signal and after a number of amplification stages, the voice signal disappears into the noise (could not distinguish between voice and noise), making the speech unintelligible. In a DTS, it is very easy to remove any noise that has been added to a binary signal along the transmission path. In addition, a digital filter can regenerate an error free binary signal by removing the noise (Keiser, et al, 1985).

2.2 An Overview of Voice over Internet Protocol

The significance of VoIP has been recognized by the telecommunication industry in recent years. VoIP became very important for telecommunication issue both in terms of economic and infrastructure. Consequently, many VoIP applications have been developed. VoIP is a technology that allows making a call utilizing broadband or even dial-up Internet connection instead of a regular phone line. Some services using VoIP allow calling other people using the same service, while others may allow calling anyone who have a telephone number including local, long distance, mobile, and international numbers. VoIP has been envisioned as the next likely revolution in the telecommunications and computer network industry (Perini, 2001).

IP networks are normally best effort networks where the transmitted data is not guaranteed to arrive at its destination, with time being varied. This is not acceptable for VoIP and any reliable application has to compensate for these impairments. Speech compression is an essential part of any VOIP systems.

It determines how one encodes the speech for transmission and has an enormous affect on the speech quality (Sundstrom, et al, 1997)

According to Rappaport (2002), VoIP involves digitization of voice streams and transmitting the digital voice as packets over conventional IP data networks such as the Internet. The use of VoIP will eventually result in the convergence of existing data and telephony networks. The existing Public Switched Telephony Network (PSTN), the cellular network, the wired Internet and the WLANs could possibly converge forming a single network offering multiple services.

VoIP technology has been known for long time, but it made considerable progress in the 1990's. Business oriented organizations began to set up gateways to first allow PC-to-Phone and later Phone-to-Phone connections. Some of the companies saw this as break through and started providing users a way to make free phone calls using a regular phone (Sanneck, 2001, Goode, 2002).

At present, there are two standards which are in use for the VoIP switching and gateways namely (a) SIP and (b) H.323. SIP mainly relates to end-user IP Telephony applications whereas H.323 is a new The International Telecommunication Union (ITU) standard for routing between the circuit-switched and packet-switched worlds. H.323 is used for termination of an IP originated call on the PSTN, but the converse is also becoming common at a very fast rate.

Bates (2000) notes that today an integration of voice, data and video is needed and thus, local area network (LAN) is a solution to integrate all these services into a single infrastructure. The VoIP can be moved to any other location within the building quickly and easily, the only limitation being there has to be a LAN connection. Figure 2.1 illustrates Local area network sharing with both data and real-time voice traffic through Internet.

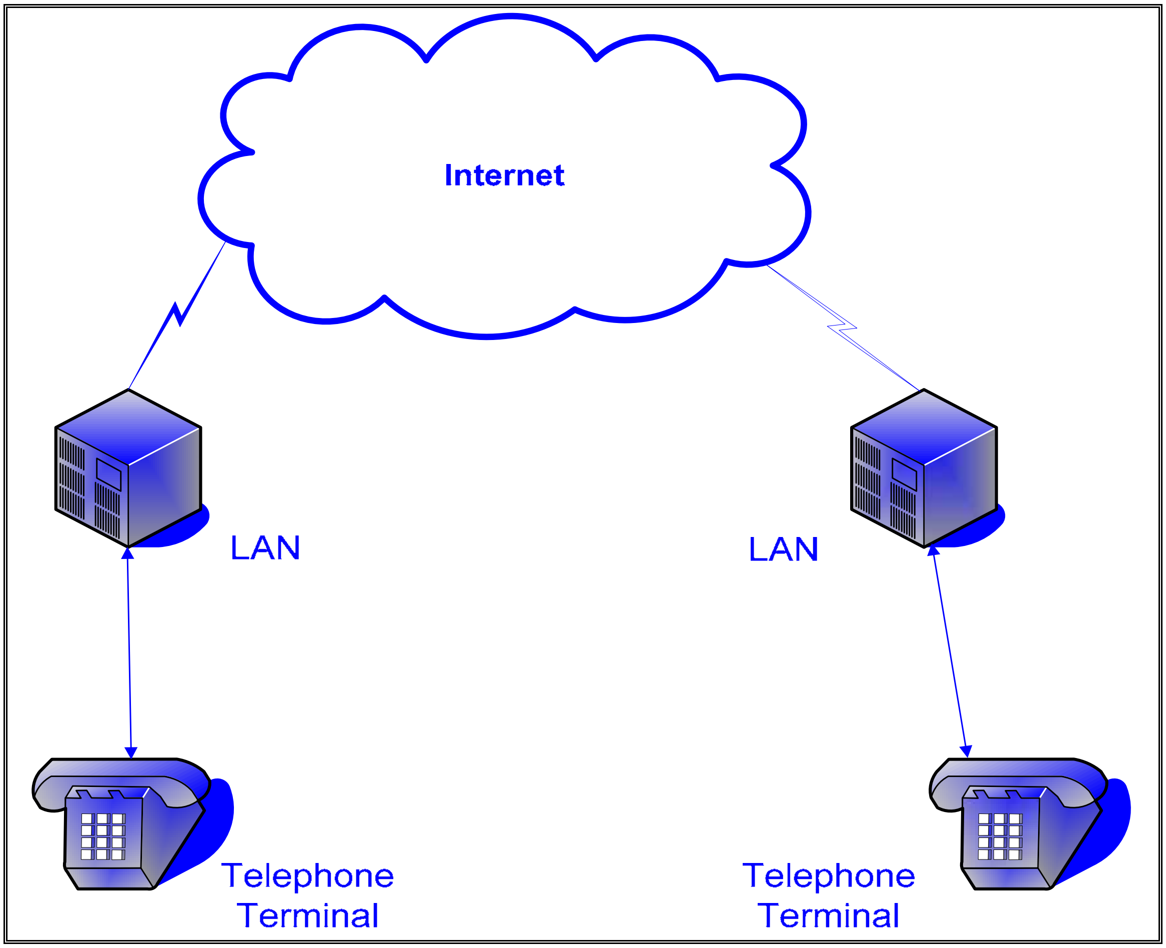

Figure 2.1: Converged network sharing with both data and real-time voice traffic
Source: Author

2.3 Convergence Networks

It is not a new concept that the data and voice applications share the same network. However, since early in the 1990s, the concept has become prevalent in the network technology industry. The network traffic is made up of accumulate packets, of which voice packets are a very small percentage out of the total traffic through the network. Even taking into consideration the header size of the voice packets, the overall voice packet size is still small compared to data applications packet size.

Comparing the growth of data packets to the growth of the voice packets, the growth of data packets outweighed than the voice packets over Internet. As a result, the bandwidth concept as long as networks are assigned enough bandwidth for data, there will be also enough bandwidth for voice too (Chong and Matthews, 2004).

The data networks are traditionally classified as being either LAN or a wide-area network (WAN). A LAN usually covers a small region while a WAN covers a larger geographical region. The VoIP technology has the capability to work with both these networks. Data network and voice network have different characteristics. Traditionally, voice networking infrastructures are connection oriented. On the contrary, the data networking infrastructures are more or less connectionless. Data network is defined as “a collection of terminals connected together for the purpose of exchanging information”. It is also often called “packet-switched networks” meaning that the information exchanged between the different terminals is done through a small entity called the packet. Whenever a message is to be exchanged between two or more terminals, the message must first be broken into packets before it can be transmitted. The terminals involved are most commonly computers, but it could be any device, which is capable of communicating with the other terminals (Stallings, 1997).

By definition, VoIP is simply a part of data network that carry a voice message through the existing data network. Voice and data still share the same path, some of the protocols like IP address, UDP, TCP and many other protocols as well hardware like hubs, switch, gateways routes and many other devices. Despite IP protocol being the most widely used data networking protocol, IP is also defined as any data network that recognizes the IP protocol (Minoli, 1998).

2.4 VoIP Protocols

The most important protocols related to the Internet and VoIP are SIP, H.323, IP, UDP, TCP, and RTP. Given that VoIP is the transmission of voice over an internet protocol network, it could therefore be transmitted over other data networks such as Frame Relay instead of IP. However, IP is the most widely used network today, and therefore it is the preferred medium for transmission of voice over data networks (Soulhi, 1999). SIP protocol can be used for VoIP signaling and call control, as well as Billing mechanisms for messaging of VoIP (Sok-Ian Sou et al. 2005) whereas H.323 protocol suites video telephony standard. The H.323 is recommended by the VoIP Activity Group of the International Multimedia Telecommunications Consortium (IMTC) and is suitable for multimedia communications over packet data networks. IP protocol makes it possible that voice, data, and video can use a single physical network and this protocol is adaptable over many technologies.

The Transmission Control Protocol /Internet Protocol (TCP/IP) protocol uses Internet applications. Since the voice is delay sensitive, the TCP protocol is not suitable protocol to VoIP, because the acknowledgment characteristic would lead to excessive delays. Moreover, this protocol is connection oriented and it uses acknowledgments and retransmission to ensure that the data is received.

The User Datagram Protocol (UDP) is connectionless, with delivery service using IP to transport messages over the Internet. The UDP is combined with Real Time Protocol (RTP) giving end-to-end network transport functions for applications transmitting real-time data (Goode, 2002).

2.5 Voice Digitizing and Compressing

The voice circuit is compressed over IP from the start to the end. The coder's function is to convert the analogue voice into binary form. In other words, the analog signal from the telephone is digitized into pulse code modulation (PCM) signals by the voice coder-decoder (CODEC). The PCM samples are then passed to the compression algorithm, which compresses the voice into a packet format for transmission across the WAN (Wallace, 2006).

On the receiver side, the very same functions in reverse order are performed. However, where the signal digitizing and compression to be done is depend on the type of the network as well as its functionality. The router or the gateway can perform either the codec and compression functions or only one of them. For instance, if an analog voice system is used, then the router or the gateway performs the CODEC and the compression functions. However, if the voice system used is digital then Private Branch Exchange (PBX) is used and it performs the codec function. Therefore, the Router just processes the PCM samples passed to it by the PBX (Baher, 1990).

In order to transport the analog voice signal over a digital transmission system, as is described in the beginning of this section the signal digitizing as well as compressing take several steps to be processed, thus this processing utilized bandwidth resources. This consumption of bandwidth is different form codec to codec. However, in terms of bandwidth utilization it depends on the codec is used or implemented. For instance, the codec G.723.1 operates at 5.3/6.3 kbps and uses Algebraic Code Excited Linear Prediction (ACELP) and this is the one of the lowest rate codec. The sample delay of this algorithm is 30 milliseconds of speech and maximum delay tolerance is 37.5 milliseconds (Wallace, 2006).

The G.729 codec is for real time network. It is an important codec in terms of packet loss as well bandwidth utilization. This codec algorithm operates at 8 Bps. It uses Conjugate structure (ACELP). Its sample delay is10 milliseconds of speech and maximum delay tolerance are 15 milliseconds. This codec originally was designed for wireless environment (Wallace, 2006).

Figure 2.2 shows the analog signal originated from analogue devices converted into binary form to transmit it through a data network and at the end, the signal converts back as its original.

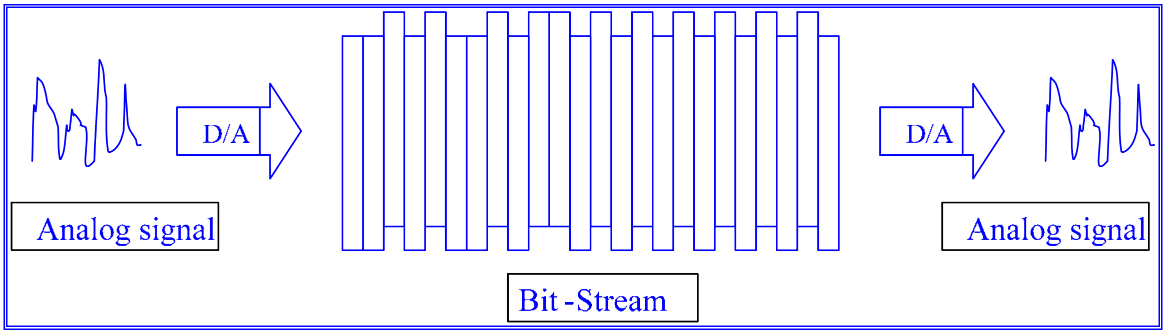

Figure 2.2: Shows the analog signal converted into binary form
Source: Author

2.6 Network Design

When designing networks that transport voice over packet, frame, or cell infrastructures, it is important to understand the delay components in the network. It is equally important to correct count all potential delays in order to ensure that overall network performance is acceptable. Overall voice quality is a function of many factors including the compression algorithm, errors and frame loss, echo cancellation, and delay. In order to avoid unacceptable voice delay and packet loss, the design of the network has to be taken into account (Bates, et al. 2000).

2.7 Delay of VoIP

There can be a significant voice delay in a VoIP compared to the conventional PSTN. This is mainly because of the components involved in a VoIP system and the fact that the data network is not designed for real-time applications. VoIP delay can be as high as 556 ms in some implementations. Since user studies indicate that telephone user's find round-trip delays of greater than 300 ms more like a half-duplex connection than a conversation, such delays are an excessive.

It is however, worth noting that human tolerance for delay in a telephone conversation varies from one user to another and some more tolerant users are satisfied with delays of 300-800 ms (Goodman, 1999).

Yensen T. et al. (1998) describe that the work of the echo canceller causes high delay. Longer delays for the same echo are more disturbing, thus, the echo canceller must remove more of the echo to keep the quality acceptable. However, the VoIP applications require better echo cancellors compared to the PSTN where the delay is low and the echo cancellor does not need to work so hard.

It is also important to determine the amount of delay in the system accurately. In addition to implementing a VOIP system, one should be careful of limiting the amount of delay introduced as much as possible for the two reasons described previously.

Delay is a deterrent to proper two-way conversation and delay complicates the work of the echo canceller. This means that for all components over which we have control, the delay introduced by the components should be kept as low as possible. Figure 2.3 specifies delays between components in a VoIP system while Table 2.1 provides the corresponding typical numerical values of these delays.

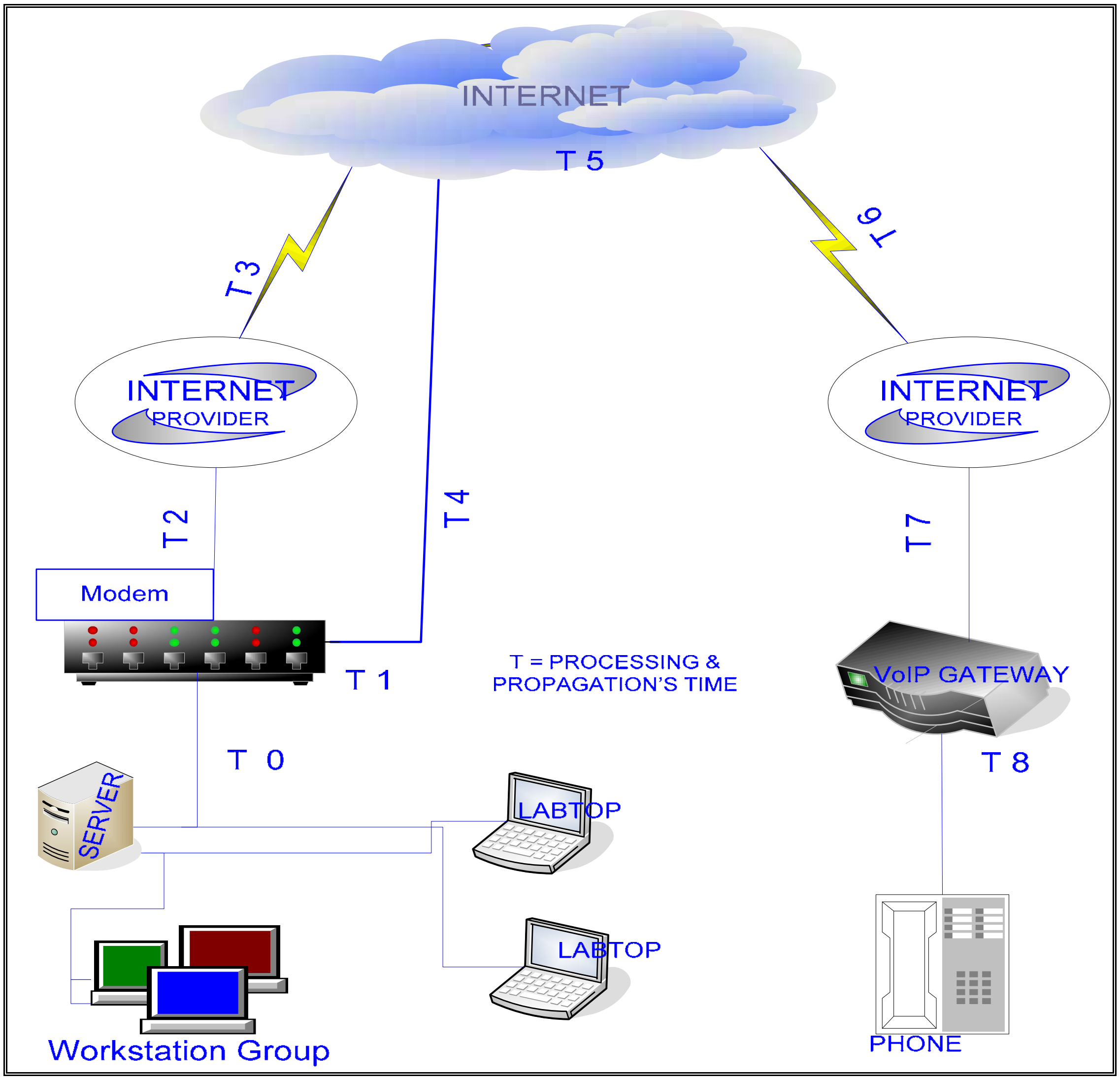

Figure 2.3: Block Diagram of VoIP system: Source (Dive, et al. 2000)

Modified Author

When the access method is a PC, the main sources of delay in a VoIP system is (a) the PC itself (b) the modems involved (if we are dealing with an ISP) and (c) the IP network. If the access method is a telephone, the main sources of delay are (i) the VoIP gateway and (ii) the IP network.

The delay associated with an IP network heavily depends on the amount of traffic in the network and the number of router hops required in order for reaching the intended destination. If the transmission is multi-hop delay on the public Internet, it can be as high as 500 ms or even more. For satellite links, it will increase even more due to the high propagation delay. The value, which is being given in Table 2.1 below, is only for a particular Internet connection.

Table 2.1: Delay associated with Symbol and Explanation

Symbol	Delay (ms)	Explanation
T0	150	Delay associated with PC processing in sound card, speech compression algorithm and play out buffering
T1,T2, T3	150	Delay associated with modems at PC and Internet Service Provider
T4	negligible	Delay introduced by network interface card
T5	96	Delay associated with IP network (value given is for a particular. Internet connection, value for corporate network would be lower)
T6	negligible	Delay for gaining access to IP network from VoIP gateway
T7	160	Delay introduced by VoIP gateway because of speech compression, play out buffering and other VoIP tasks as well as buffering delay (a VoIP gateway serves many sources)
T8	negligible	Delay for telephone line
T9	negligible	Delay introduced by telephone

Source: (Dive, et al. 2000)

2.7.1 VoIP Delay Processing

Delay is defined as “the time from when a sound is digitized by a sending phone until the receiving phone user hears” (Chen, et al. 2005). When a telephone sends voice to someone over a VoIP network, the voice traffic splits and groups it into packets. Packets that travel a long way or over a slow network link can cause noticeable delay.

On a VoIP call, a packet containing the previous 50ms of voice audio is sent to the receiving telephone every 50ms. If the network is functioning perfectly, then every 50ms the person listening to a call will hear 50ms of audio. It will not be heard at exactly the same time that it is being said, as the packets take time to travel to the other telephone. In addition, the originator telephone has to wait 50ms before it can send 50ms of voice to the receiving telephone.

Delay includes the time it takes to place a full 50ms of voice traffic in the packet, the time for the network to send the packet to the receiving phone, and the time for that phone to decode the packet and convert it into audio again. There are two main parts of delay namely fixed and variable. Fixed delay components add directly to the overall delay on the connection. Variable delays arise from queuing delays in the egress trunk buffers on the serial port connected to the WAN. These buffers create variable delays, called jitter, across the network. Variable delays are handled via the de-jitter buffer at the receiving router or gateway (Chen, et al. 2005). Shown below are the most common delay sources:

I. Fixed Codec delay
II. Processing (encoding –decoding)
III. Packetization
IV. Serialization

V. Propagation delay which depends on the network design and the number of hops
VI. Switching delay
VII. De-jitter buffer delay

2.8 Packet Loss

Packet loss or loss is defined as “the network drops of VoIP packet” (Paxson, 1997). Networks often drop packets when equipment is congested or time exceeds proper queue time. The human ear can tolerate small gaps in voice audio because it is possible for people to understand words even when part of the word is missing. To assist listener comprehension, most VoIP telephones implement packet loss concealment. A telephone that detects a missing packet might rely on the audio in the lost packet sounding roughly the same as the previous packet. Using this assumption, the receiving telephone plays the previous packet again at a lower level and attempts to trick the human ear into hearing uninterrupted audio (Yajnik et al. 1990).

2.9 Jitter

The concept of Jitter is a difficult to understand. As described elsewhere in the chapter, in a perfect network, packets containing 50ms of voice will arrive exactly 50ms apart. In this situation, the receiving telephone can play the packets as they arrive without gap and without having to skip any of the sound when playing it back.