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Internet Telephony and the Viewpoint of Telecommunication

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***Abstract*— In this paper, internet telephony technology is reviewed. The foremost distinctive quality amid VoIP and traditional telephony, packet switching versus circuit switching is re- examined. The TCP/IP and allied protocols that make internet telephony achievable are briefly described. The definite protocol principles, the hardware and software necessities for appropriate performance of the Internet telephony are outlined. Performance issues and strategies for upgrading of Internet telephony are discussed. From the study, it was observed that many countries are migrating their networks to internet telephony as a result of its much reward. Accordingly, internet telephony is suggested as the telecommunication technology for the future.**

***Index Terms*— Internet Telephony, Voice over Internet Protocol, Data Communication, Internetworking.**

I. INTRODUCTION

In less than decades, Voice over Internet Protocol (VoIP) has revolutionized the telecommunication industry[1]. While mobile phones have made the headlines as they gradually evolved from expensive bricks to packet supercomputers, in the background VoIP has helped to knock down barriers in international communication, providing a genuine alternative to telephone calls made using traditional telecoms infrastructure and providing near-universal access to cheap calls for anyone with a computer and internet connection.

Voice over Internet protocol (VoIP) is the process of transmitting voice information across an Internet protocol network, for an instance the Internet[2]. Thus, it is also called Internet Telephony; a tool that allows one to make telephone calls using a broadband connection as an alternative of a regular phone line. VoIP converts the voice signal from our telephone into a digital signal that travels over the Internet [3]. One of the most noteworthy rewards of VoIP over a traditional public switched telephone network (PSTN) is that one can make a long distance phone call and sidestep the toll charge. In VoIP, the voice signal is broken up into small pieces (packets) and sent though the network one-by-one. The process of packetization compresses the caller's voice signal, transfers it over the IP network and it is then decompressed at the other end. This integrated voice/data solution allows large organizations to carry voice applications over their existing data networks. Presently, Internet telephony is making communications cheaper all over the world. Some countries have already passed regulations on VoIP, forcing the incumbent national operators to slash their tariffs in reaction to the new competitive environment. Not only will this technological development have an impact on the large traditional telecommunications business, it has modified the pricing and cost structures of traditional telephony [4].VoIP wide acceptance is being driven by the advantage of convergence whereby one network is used for voice, video, and data. Some of the other advantages include: advanced call routing, unified messaging and then inherent flexibility whereby it is easy to add, change or remove phones [5][6]. However, some of the drawbacks are that some VoIP phones do not function during power outage and the location of Internet voice 911 callers cannot be easily determined. Some of the strategies for overcoming these shortcomings have also been studied.

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1.2 Components of an IP Telephony System

Below are components of IP system:

A. *IP Telephony Server(s):*

This is the heart of the IP Telephony systems which provides complete Call Control, Dial Plan control and all the basic voice applications (In case of smaller systems, all the functionalities of the below mentioned application servers can also be bundled with this)

B. *Application Servers:*

Sometimes applications like IVR (Interactive Voice Response – Auto Attendant), Call Recording, Voice Mail, Data Base Integration require to be hosted in separate servers – Especially for larger VOIP installations.

C. *IP Phones:*

These IP Phones connect directly to the IP Network (RJ-45 based UTP Cables) and provide all the voice functionalities hitherto provided by analog phones like caller ID display, speaker phones, speed dial keys, memory etc.

D. *Soft Phones:*

These are basically software utilities that have all the telephony functions but use the computer, head-set with microphone to make and receive calls.

E. *Wi-Fi Phones/ Dual Mode Cell Phones:*

Wi-Fi phones are based on IP Technology and connect to the wireless network and act as mobile extensions. Certain Cell phones come with Wi-Fi adaptors and can be used as a Wi-Fi Phone (if the manufacturer supports the same). Cell Phones can also connect to the IP Telephony server through 3G Networks/ CDMA networks for making a VOIP Call.

F. *Analog Telephony Adapters (ATA):*

These are specialized devices that connect to the LAN at one end and connect to FXO (Analog Trunks) or FXS (Analog Extensions) at the other end.

G. PRI Cards:

These are used to connect PRI/E1/T1 Trunk Lines to IP Telephony Servers – Usually they connect directly with the PCI/ PCI Express Slot in the server.

H. Computer IP Network:

An IP based Computer Network is used to carry the voice signals across the enterprise and sometimes even to remote locations.

Figure 1 shows a typical VOIP/ IP Telephony Architecture and Connectivity.

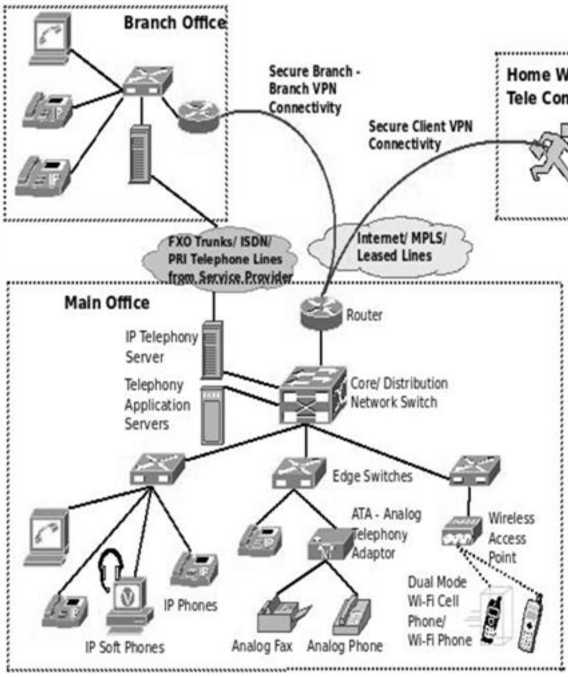

Figure1: VOIP/ IP Telephony - Architecture and Connectivity Diagram

II. BACKGROUND INFORMATION

There are two essential technologies that are obligatory for the reality of VoIP. The first, and most widely used, is the telephone. The second technology is the Internet. The telephone was invented by Alexander Gram Bell and Elisha Gray in the 1870s[VII]. The first regular telephone exchange was established in New Haven in 1878. Early telephones were leased in pairs to subscribers. In 1968 the Internet was first developed by ARPANET (Advanced Research Projects Agency Network), founded by the U.S. Department of Defense in 1957. ARPANET was developed to provide a decentralized communications network that would not be disrupted by a potential global war [VIII] . The first Internet phone software was released in February 1995 by Vocaltec, Inc a small company in Israel [4]. Their products were designed to run on a personal computer equipped with sound card, speakers, microphone, and modem. The software compresses voice signal and translates it to (Internet Protocol) IP packets. The voice packets are transported using IP in compliance with the International Telecommunications Union's H.32, the specification for transmitting multimedia (voice, video and data) across a network [9]. By 1998 some entrepreneurs started to market PC-to-phone and phone-to-phone VoIP solutions. The phone calls were marketed as "Free" nation-wide long distance calls. When the caller would start the call he/she had to listen to advertisements before the call was connected. Another development in 1998 was the hardware's foray into the market. There were three IP Switch manufactures that introduced VoIP switching software as a standard in their routing equipment. By the end of 1998 VoIP calls had yet to total 1% of all voice calls. By 2000, VoIP calls accounted for 3% and by 2003 that number had jumped up to 25%[10]. Presently, VoIP poses a competitive threat to the providers of traditional telephone services, and is considered as a "de facto" for future telecommunication.

III. TRADITIONAL CIRCUIT SWITHCHING VS PACKET SWITHCHING

A. Circuit Switching:

The fundamental problem with the existing Public Switched Telephone Network (PSTN) is their reliance on circuit switching. Circuit switching is the most familiar technique used to build a communication network. It is used for ordinary telephone calls. It allows communications equipment and circuits, to be shared among users. When a call is made between two parties, the connection is maintained for the entire duration of the call. In a typical phone conversation, much of this transmitted data is wasted because only half of the connection is in use at any given time. One party is talking and the other is listening. Circuit switching provides traffic isolation and traffic engineering, but at the expense of using bandwidth inefficiently and signaling overhead. Figure 2 shows a circuit switching.

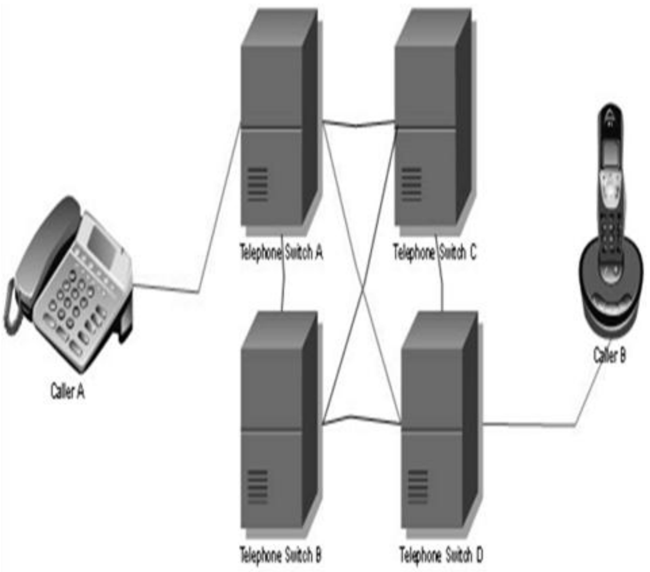

Figure 2. Circuit switching

B. Packet Switching

Packet switching is the basis for the Internet Protocol. While circuit switching keeps the connection open and constant, packet switching opens the connection just long enough to send a small chunk of the data called a packet from one system to another. These packets are sent, one by one, to the nearest router, which will look up the destination address, and then forward them to the corresponding next hop. This process is repeated until the packet reaches its destination. The routing of the information is thus done locally, hop-by-hop. Packet Switching is very efficient. It minimizes the time that a connection is maintained between two systems, which reduces the load on the network. It also frees up the two computers communication with each other so that they can accept information from other computers as well. VoIP technology uses packet switching to transmit voice signals using the Internet. Packet Switching allows several telephone calls to occupy by only one in a PSTN line; it also handles busty traffic (data) well hence it is used in computer network. However, bandwidth limitation leads to packet loss. Packet loss shows up in the form of gaps or periods of silence in the conversation. Figure3 shows a packet Switching.

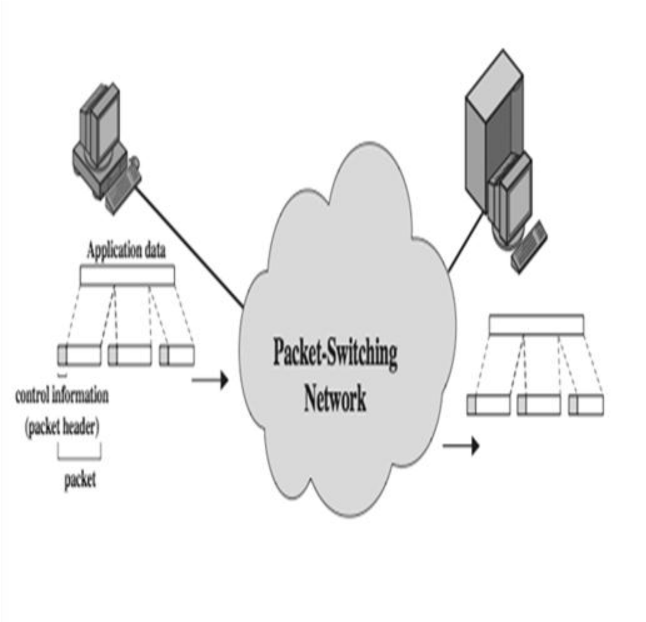

Figure 3: Packet Switching

C. *Transmission Control Protocol/Internet Protocol*

The TCP/IP is suite of protocols developed by United States Defense for computer network communications and the Internet [XI]. It is an industry standard suite of protocols that is designed for large networks consisting of network segments that are connected by routers. TCP/IP is the protocol that is used on the Internet, which is the collection of thousands of networks worldwide that connect research facilities, universities, libraries, government agencies, private companies, and individuals. IP is the core protocol for routing however, it is said to be unreliable and connectionless while TCP is a host-to-host, connection-oriented protocol that runs in the end-devices providing error checking and guaranteed-delivery. The main job of Routers is to forward IP packets using its address. TCP runs in the end-devices; the user's Pc and the server it is communicating with. It ensures that all of the data is delivered on the right order, and if there are any errors, TCP request for a re-transmission. The TCP accomplishes this by using a system of sequence and Acknowledgement numbers that are carried in the TCP header. When a TCP segment is received, the receiver tests it for errors and confirms that it is in sequence. If the message is correct and in sequence, the receiving TCP returns the received segment's Sequence number as an Acknowledgement number. Else, it will request for a re-transmission.

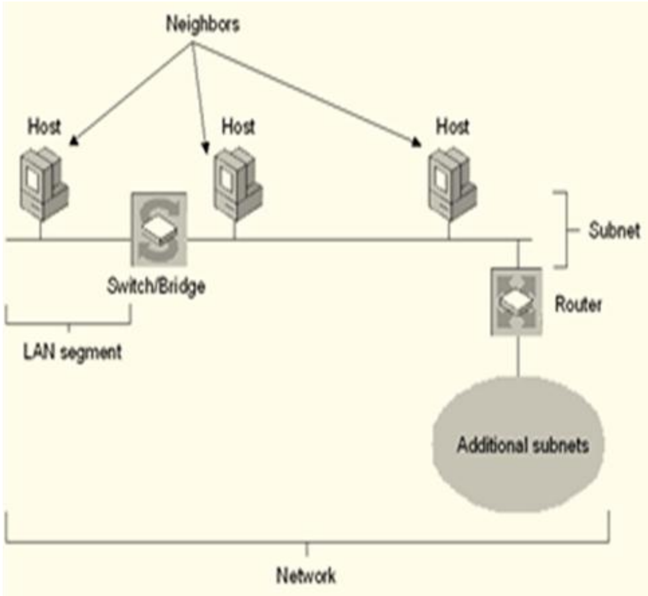

Figure 4: Transmission Control Protocol/Internet Protocol

IV. INTERNET TELEPHONY HARDWARE

VoIP supports Computer-To-Telephone Calls, Telephone-To-Computer Calls and Telephone-To-Telephone Calls. The Set of hardware for VoIP Include: User End-Devices, Call Processing Server/IP PABX, Media or VoIP Gateway and Internet. See figure 5.

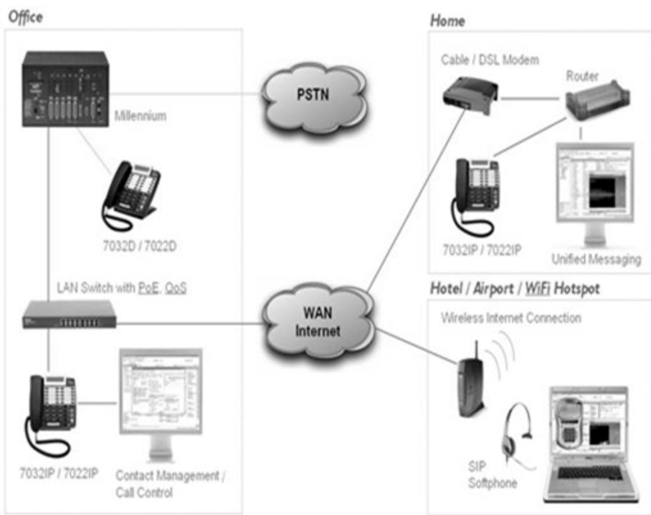

Figure 5: Internet Telephony Hardware

V. INTERNET TELEPHONY PERFORMANCE AND STRATEGIES FOR IMPROVEMENT

Subscribers want services analogous to traditional fixed line telephone. Some of the vital performance issues are propagation delay, jitter, bandwidth, reliability etc.

A. Propagation Delay

The most unrelenting criticism regarding packet telephony is the propagation or transit delay, this is because voice traffic is real time traffic; if there is too long a delay, the speech will be unrecognizable. Circuit switched services typically have shorter delays, domestic phone calls normally have a propagation delay less than 30 milliseconds, international phone calls have delays less than 50 milliseconds. Digital mobile phones have one-way delays in the order of 80 to 100 milliseconds. The worst-case scenario for delay is a satellite circuit that has a one-way delay of about 250 milliseconds. In Internet telephony, the delay is caused by the number of different factors and some of these include: processing delay, packet forwarding delay and network delay.

B. Timing Jitter

The lack of fixed path for IP communication sessions meant that IP packet for a given communication may arrive at different times or out of sequence, scrambling the voice. This variable delay between packets is called jitter. It can be removed by use of Real Time Protocol/ Adaptive Jitter Buffers whereby the packets are stamped with the time they were sent and adaptive buffers are used to hold them for a second or so at the receiving end, so that they can be assembled in the right order and with the right timing.

C. Lost Packets

IP networks may drop packets during congestion when the switching node buffers become full. The lower the voice bit rate, the more important each packet becomes. Most 8kbps voice systems can tolerate occasional packet loss. However, when packet loss reaches 3% to 5%, voice quality degrades significantly.

D. Bandwidth Problem

To optimally reduce delay, there is a need to increase bandwidth maximally. At LAN level, it is not a problem. It becomes a problem in the WAN or the Internet backbone. In 1991, Frame Relay a wide area packet network technology was introduced. Before many countries could migrate to frame relay, a hardware-based high speed packet switching technology called Asynchronous Transfer Mode (ATM) was developed with embedded quality of service for various categories of traffic [12]. With a further push for improved services, some other groups of researchers have enhanced the traditional Internet Protocol with a Multi-Protocol Label Switching (MPLS) technology. This new IP-based protocol makes use of the concept of virtual circuit (a label-switched path). This gives predictable performance and quality of service with regard to delay or packet loss.

VI. RESULTS AND DISCUSSION

From the survey, IP telephony has come to stay; the advantages are overwhelming. It is causing an cataclysm in the phone industry [13]. While the cataclysm is harsh for traditional companies, it is windfall for subscribers whose benefits are lower prices and new services. VoIP is available only at Cybercafés or big organizations where they use leased lines or VSAT. Better quality voice and video could be achieved by deployment of ATM, frame relay or MPLS since such technologies increase bandwidth for the Internet backbone. Also the choice of technology depends on what part of the globe one is and what the service provider prefer.

VII. CONCLUSION

This paper has reviewed VoIP technology formerly regarded as an innovation. The technology is attracting more and more users because it offers tremendous cost savings relative to the PSTN. Presently, it poses a competitive threat to the providers of traditional telephone services. Users can bypass long-distance carriers and their per minute usage rates and run their voice traffic over the Internet for a flat monthly Internet-access fee. VoIP could be applied to almost any voice communications requirements, ranging from a simple inter-office intercom to complex multi-point teleconferencing/shared screen environments.

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