The Technologies of Voice over Internet Protocol (VoIP) Based Telephony System: A Review

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Abstract *-Voice over Internet protocol (VoIP) technology of a high degree reliability and resilience that allows users to make telephone calls over an internet protocol (IP) network has been developed in this research to tackle the various problems of Public Switched Transfer Network (PSTN) such ashigher costs per call, especially for long-distance calls ,higher infrastructure costs among others.Prototyping was deployed to build a prototype VoIP-based telephony systemwhile Cisco packet tracer was configured to run several simulation sessions of the developed specifications. The simulation results showed that VoIP can be successfully deployed to provide in a flexible manner, additional data-driven services that requires cost saving, phone portability, service mobility, integration and collaboration with other applications and there is no geographical boundary for the technology.*

Keywords-- *VOIP, SIP, IP Telephony, PSTN, Packet Tracer*

I.INTRODUCTION

The Public Switched Transfer Network (PSTN) is a connection-oriented, circuit-switched network in which a dedicated channel (or circuit) is established for the duration of a transmission. In 1960's when digital voice communication first emerged, the Public Switched Transfer Network (PSTN) has been supported worldwide as the primary means of voice communication. Originally transmitting only analog signals, the PSTN ultimately switched to using digital communication, which offered solutions to the attenuation, noise and interference problems inherent in the analog system. The modern PSTN uses Pulse Code Modulation (PCM) to convert all analog signals into digital transmissions at the calling end that initiated communication and reverses the processes at the receiving end.

Internet Protocol (IP) Telephony is the transmission of voice, fax and related services over packet switched IP-based networks. In the longer term, as more and more voice traffic becomes IP data traffic, there will be little to distinguish between IP telephony and circuit-switched telephony. Internet Protocol (IP)-based networks are the latest innovation to offer solutions to such problems and are increasingly being used as alternatives to the traditional circuits switched telephone service. Voice over Internet Protocol (VoIP) is an upcoming technology and has great potential for future voice communication services. There are many technical aspects of VoIP that make it cutting edge technology in this communication era. One aspect of VoIP from a corporate standpoint is the cost reduction with operations by implementing a VoIP network. A key feature that makes VoIP an attractive technology to adopt is the idea of a converged network - One single network infrastructure that carries voice, data and video over the Internet Protocol using existing network equipment.

II.LITERATURE REVIEW

A. Types of Network Connections

1. *Dial-up connections*

Dial-up connections are the most common type of Network connection available from internet service provider (ISP's); they are also the slowest and usually the most inexpensive. A dial-up connection allows you to connect to the Internet via a local server using a standard 56k modem; user personal computer (PC) literally dials (hence the name) a phone number (provided by ISP provider) and connects to the server and therefore the Internet [4].Once connected, the user can utilize all aspects of the internet, the drawback with a dial- up connection is the speed, a standard 56k modem can theoretically transfer 56 kilobits of data a second, this means that user can (again theoretically) transfer up to 7 kilobytes a second although to get a full 7kbps is near impossible due to the compression overhead.

2.ADSL connections

Asymmetric Digital Subscribers Line (ADSL) connections are becoming more and more widely available and can provide an excellent Network connection. The connections work by splitting the phone line into two separate channels, one for data (internet) and one for voice (phone calls) which means one can talk on the phone and be connected to the Internet at the same time. One can often see ADSL connection services advertised as having different speed specifications, below are some common configurations:

(i).256kbps/128kbps (ii).512kbps/128kbps (iii).1mbps/256kbps

Notice there are two values to each configuration, the first figure states the download speed and the second figure is the maximum upload speed. As an example let's take the second configuration 512kbps/128kbps, this means that one can potentially download data at a speed of 512kbps and upload data at 128kbps.

3. Cable connections

Cable connections are considered one of the best types of internet connection available to the home user; they offer very fast and reliable connections with a fixed monthly fee [5]. Cable Modem Broadband connection through an ordinary coax cable through your digital cable provider is the easiest and most common way to connect to the Internet at high speeds [2]. Most connections average about 400K/second download and 128K upload. Cable's largest advantage is its availability and ability to produce multiple up streams (when sending). The biggest downside to a cable connection is the slow- downs you'll experience during gluts of service when several people are sharing the network.

4. Satellite (HST)

This is the most expensive alternative for getting a high-speed connection to the Internet. If user lives in a rural area or a spot where other options are not available, then this is probably users only hope for a high-speed connection. These come in two varieties, 1- way and 2-way. One-way satellites are like television receivers: they only accept signals. User will still have to use his/her modem to connect for uploads. A 2-way connection, however, both sends and receives and is telephone-free. [1] Emphasized that average speeds for this type of connection are 600K and higher for download and 128K for the upload.

Bandwidth requirement for VoIP calls depends on the codec (enCOder/DECoder) used to compress the data anywhere from 16 to 64 kbps is normal. Add onto that the extra overhead of about 10 to 24 kbps. In total, VoIP uses 26 to 88 kbps. As a general rule, the user will need at least 88 kbps to use VoIP reliably [3]. Note that VoIP service providers will provide with their broadband requirements. Users will need an Internet connection that can handle at least 128 kbps on the upload side. Extra bandwidth will allow others in the household to surf the net or download files while you are talking on the phone.

5. Wireless Technology

With far reaching access of electronic data systems, it becomes increasingly bothersome to be tethered to them by wires. Thus, wireless communication brings about these merits:

(a) Low-cost deployment, (b) Broadcast same data too many locations simultaneously, (c) Deployment ease in hostile environs, (d) Mobile communication.

Its demerits: (a) lesser data rates, (b)lesser reusable frequencies than in guided wired media, and (c) more susceptible to interference.

B. Different Forms of VoIP

In case of VoIP the interesting thing about VoIP is that there is not just one way to place a call. There are three different "flavors" of VoIP service in common use today:

1. ***ATA:***This is the simplest and most common way which can be done with the use of a device called an ATA (analog telephone adaptor). The ATA allows you to connect a standard phone to your computer or your Internet connection for use with VoIP. The ATA is an analog-to-digital converter; it takes the analog signal from your traditional phone and converts it into digital data for transmission over the Network [2]. We simply crack the ATA out of the box, plug the cable from the phone that would normally go in the wall socket into the ATA, and can be ready to make VoIP calls. Some ATAs may ship with additional software that is loaded onto the host computer to configure it; but in any case, it is a very straightforward setup.

2. ***IP Phones*****:** These specialized phones look just like normal phones with a handset, cradle and buttons. But instead of having the standard RJ-11 phone connectors, IP phones have an RJ-45 Ethernet connector. IP phones connect directly to the router and have all the hardware and software necessary right onboard to handle the IP call. Wi-Fi IP phones are available, allowing subscribing callers to make VoIP calls from any Wi-Fi hot spot.

3. ***Computer-to-computer:***This is perhaps the simplest way to use VoIP. We don't even have to pay for long-distance calls. There are several companies offering free or very low-cost software that we can use for this type of VoIP. All you need is the software, a microphone, speakers, web camera, a sound card and a Network connection; preferably a fast one like you would get through a cable or DSL modem. Except for your normal monthly ISP fee, there is usually no charge for computer-to- computer calls, no matter the distance.

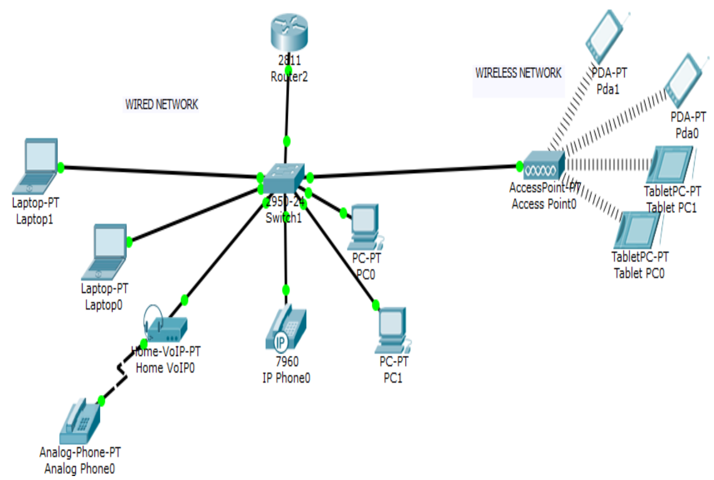

Figure1. A simulation diagram showing that wired and wireless network can be combined together while performing IP Telephony

C. *Devices used in VoIP Technology*

The architecture of Internet telephony is similar to traditional telephone networks in many ways, of course, but it also has some significant differences. Most fundamentally, Internet telephony is different from traditional telephone networks in that it, naturally, runs over the Network, or more generally over IP networks. The most common devices used in the Internet telephony are end systems, gateways and signaling servers.

1. End systems are electronic devices with which clients or users place and receive calls. These end systems responds and initiate to signaling, and receive and transmit media. These devices also maintain the track of calls and their status.

2. Gateways are devices that allow calls to be placed to and from other telephone networks.

3. ***Signaling servers*** handle the application-level control of the routing of signaling messages. They are typically used to perform user location services; a signaling server can maintain information about where a user can currently be found and forward or redirect call setup requests to the appropriate current location. Signaling servers are the devices which, from the point of view of feature- creation, are most similar in functionality to service control or switching points in the circuit-switched network; they can programmatically direct, block, or alter call signaling messages based on their own internal logic.

D. *Protocols Architecture of VoIP*

Protocols are set of rules or procedures that either way used to endpoints when they communicate in a network. For example, OSI (open systems interconnection) has the seven different layers, for data interchange different protocols at each layer of this model interacts. In Internet Protocol Telephony data is transmitted in the form of datagram. Every datagram has a source address, destination address and sequence number. Each datagram of file/message is independently routed across the network and datagram are reassembled at the receiving end. The internet was designed to deliver the datagram reliably without considering delays. The Internet Protocol (IP) is provided for routing datagram between any two nodes with checking for corruption and loss. A higher layer protocol TCP is provided for re-transmission of lost data and acknowledgments have also been sent back. Retries for re-transmission of data will be taken some time so then TCP can take much longer time. So TCP is highly unsatisfactory for fixed data transmission.

E. *Session Initiation Protocol*

The Session Initiation Protocol (SIP) is an application layer control protocol standardized by the Internet Engineering Task Force (IETF) in March 1999 that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, internet telephony and similar applications. There are five consecutive communication operations of setting up and ending session that are supported by the SIP. First, the user location is determined to create communication. Later, the willingness of the called party to participate in session is verified. Thirdly, the media capabilities of the both parties (calling and called) are resolved. Fourthly, the session setup is handheld by establishing session parameters at both parties. And finally, the session management including termination, modification and invoking services are carried by SIP.

III.REASONS FOR SLOWER ADOPTION

(i) Technical factors: The unreliability of the internet is the main market restraint of VoIP. There are many Quality of Service (QoS) issues experienced by packet-switched networks that do not affect circuit-switched networks. Acceptable sound quality has become expected on the PSTN, whereas VoIP is an immature technology, experiencing many problems in this area.

(ii) Delay: There are four types of delay namely, *propagation delay* "that which is caused by the signal before travelling a distance". *Network delay* "it is a function of the capacity of the pipes in the network and the processing of the packets as they transmit the network". *Accumulation delay*" depends on the type of voice coder used. It is caused because a finite amount of time is needed to collect a frame before the processing begins". *Processing delay*" it is caused by the actual encoding and collection of encoded samples into a packet for transmission.

(iii) Jitter: Jitter is a variable inter-packet timing caused by the network a packet traverses. This is removed by buffering fast packets in order that the slowest packets arrive in time to be sequenced correctly. This causes additional delay

(iv) Echo: Echo is caused by signal reflections in a hybrid circuit that is converting between a four-wire circuit and a two-wire circuit. It is present in all telephone networks, but is acceptable in a circuit switched network because round-trip delays are usually short enough to go unnoticed.

(v) Unreliability: This will also be cause for concerns in areas such as public safety. PSTN are usually well engineered and very reliable.

(vi) Scalability: It is also a problem because there are many needs in this area. The port density of each gateway can be easily increased, as can the overall number of users due to the distributed nature of the internet.

(vii) Latency: Latency occurs when voice data is queued at a router, or other network element, and delayed from reaching its destination. This is the direct result of network congestion.

(viii) Packet Loss: It is important that enough packets reach the destination for the speech to be recognizable, however it is not a given that 100% of all packets must reach the intended target. In actuality, a voice stream can lose up to 5% of its packets and still be recognizable.

(ix) Call Dropping: The dropping of a call refers to the unexpected termination of a VoIP connection. This could result from an equipment failure at either end or at a midpoint element, or from excessive network congestion and the subsequent dropping of a large number of packets.

(x) Call Blocking: Similar to call dropping, call blocking occurs due to congestion at the network elements. When a socket attempts to initiate a VoIP call but insufficient resources exist on the network, the call will not reach the intended recipient at all. Call blocking can also arise from control parameters set by a service provider, although this is becoming unpopular to the pint of stigmatization.

A. *Significance of VoIP:*

The reason for the prevalence of VoIP is that it gives significant benefits compared to legacy phone systems. The key benefits are as follows:

(i) ***Cost savings***: The most attractive feature of VoIP is its cost-saving potential. When we move away from public switched telephone networks, long-distance phone calls become inexpensive. Instead of being processed across conventional commercial telecommunications line configurations, voice traffic travels on the Internet or over private data network lines. For the enterprise, VoIP reduces cost for equipment, lines, manpower, and maintenance.

(ii) ***Phone portability*****:** The legacy phone system assigns a phone number with a dedicated line, so one generally cannot move a home phone to another place if one wants to use the same phone number. It is a common hassle to call the phone company and ask for a phone number update when moving to a new house. However, VoIP provides number mobility: The phone device can use the same number virtually everywhere as long as it has proper IP connectivity.

(iii) ***Service mobility***: The context of mobility here includes service mobility. Wherever the phone goes, the same services could be available, such as call features, voicemail access, call logs, security features, service policy, and so on.

(iv) ***Integration and collaboration with other applications*****:** VoIP protocols (such as Session Initiation Protocol) run on the application layer and are able to integrate or collaborate with other applications such as email, web browser, instant messenger, social-networking applications, and so on. The integration and collaboration create synergy and provide valuable services to the users. Typical examples are voicemail delivery via email, click-to-call service on a website, voice call button on an email, presence information on a contact list, and so on.

(v) ***User control interface***: Most VoIP service providers provide a user control interface, typically a web GUI, to their customers so that they can change features, options, and services dynamically. For example, the users log in to the web GUI and change call forwarding number, speed dial, presence information (online, offline), black/white list, music-on-hold option, anonymous call block, and so on.

(vi) **No geographical boundary**: The VoIP service area becomes virtualized without geographical limit. That is, the area code or country code is not bound to a specific location.

B. *Why VoIP Is Preferred Over Public Switched Transfer Network (PSTN)*..

In PSTN, communication between individuals and inter personal affairs were conveyed through physical means, which means that for information to be disseminated one has to wait till the intended recipient of the message arrives or maybe by dialing the persons mobile GSM phone line to reach the person which is usually charged by Internet Service Providers.

With VoIP technology, information and communication will be conveyed in an easy way. This will require each user or client on the network to have either an IP phone or an analogue telephone with ATA (Analogue Telephone Adaptor), mobile phone, tablets and personal computers with built in wireless functionality (or wired as the case may be) before participating in the network. For effective deployment of this system, both software and hardware network devices must be readily available before calls can be signaled and routed over the network. As a result of this, VoIP networks implementation cannot work effectively if necessary hardware or software component is missing and not fully installed or configured. Devices such as router, network switches, access points for wireless transmission, IP phones, cables and many more others must be present, crimped and configured properly too before call signaling can be enhanced.By visualizing this data and assessing voice quality, this tool helps boost productivity in debugging VoIP networks, software, and hardware. The architecture of VoIP is shown in Figure 2 below:

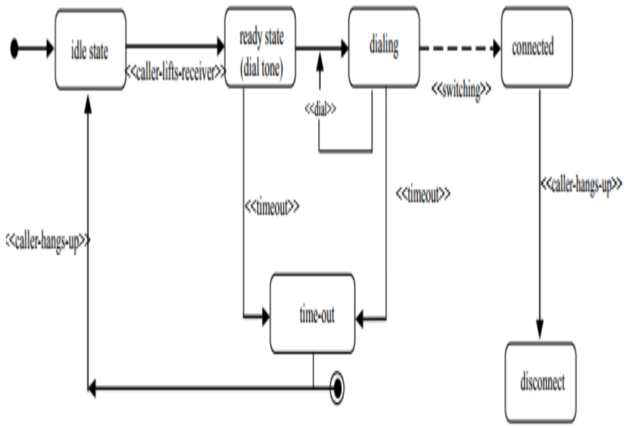

Figure 2. Architecture of VoIP System

C. *How VoIP Works*

The following is the overall working of VoIP.

(i) Initially at the source the analog signals are converted into digital signals by ADC (analog to digital converter) technique.

(ii) Then there is speech compression. According to [2], Traditional telephone networks use Pulse Code Modulation (PCM) at 8K samples per second. 12-bit samples are compressed and expanded by a nonlinear look-up table into 8-bit words giving a transmitted rate of 8kbit/s. The compression typically used by an Internet phone today is of the order of 16 to 1 (128kbit/s to 8kbit/s). Such compression is beyond PCM, ADPCM (32kbit/s, used in CT-2 cordless phones), or sub-band coding (down to 16kbit/s for speech bandwidths, normally used for music at higher bit rates). In case of a LAN (local area network) where there is a sufficient bandwidth there is no need of compression.

(iii) After conversion of voice packets into data packets, RTP (real time protocol) is used for time stamping and content identification of UDP voice packets.

(iv) Then signaling system has to perform its work and it does the following tasks:
   (a) Try to find out the destination IP address
   (b) After finding destination IP address and it establishes communication with that party.
   (c) After negotiating, the Internet protocol performs voice compression, buffer length and time stamping of packets and starts communication. However situation becomes

more complex if signaling system has to communicate with gateway between the Internet and PSTN. Gateways are devices that allow calls to be placed to and from other telephone networks, which are implemented between Internet and PSTN. Although gateway cannot support the same number of users as even the smallest local telephone exchange. In the case of outgoing calls VoIP phone captures the phone number and the IP address of gateway. But in the case of reverse direction that is from PSTN to internet it is rather impractical for the PSTN user to enter the telephone number of the gateway and then the numeric IP address of the desired party (who may be offline in case of VoIP application).

(d) Finally at the receiving end packets have to be disassembled for data extraction and for converting the data into analog voice signal and send those signals to the soundcard of the respective device.

It is evident from the above; working with VoIP technology calls could be of three types:

(a) Pc to Pc: it is a call in which one PC communicates with the other PC

(b) Pc to phone: in this type of communication a Pc communicates with an analog or digital phone

(c) Phone to phone: in this type an IP enabled phone communicates with an analog or digital phone.

IV. CONCLUSION

The researchers in this work had examined the underlining technologies of Voice over Internet Protocols (VoIP) and the possibilities of implementing a campus-wide telephony system using the technology. A prototype design was specified and a simulation was run using the Cisco Packet Tracer to demonstrate that VoIP can be successfully deployed to provide in a flexible manner, additional data-driven services in campus-wide telephony through a merger of telephone and information technology facilities.

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