Voice over Internet Protocol (VOIP): Overview, Direction And Challenges

[1] U. R. ALO   and   [2] NWEKE HENRY FIRDAY
Department of Computer Science
Ebonyi State University Abakaliki, Nigeria
[1]Email:- auzomarita@yahoo.com
[2]Email: ufoo2010@yahoo.com

ABSTRACT
Voice will remain a fundamental communication media that cuts across people of all walks of life. It is therefore important to make it cheap and affordable. To be reliable and affordable over the common Public Switched Telephone Network, change is therefore inevitable to keep abreast with the global technological change. It is on this basis that this paper tends to critically review this new technology VoIP, x-raying the different types. It further more discusses in detail the VoIP system, VoIP protocols, and a comparison of different VoIP protocols. The compression algorithm used to save network bandwidth in VoIP, advantages of VoIP and problems associated with VoIP implementation were also critically examined. It equally discussed the trend in VoIP security and Quality of Service challenges. It concludes by reiterating the need for a cheap, reliable and affordable means of communication that would not only maximize cost but keep abreast with the global technological change.
Keywords: Voice over Internet Protocol (VoIP), Public Switched Telephone Network (PSTN), Session Initiation Protocol (SIP),   multipoint control unit

1. Introduction

Voice over Internet Protocol (VoIP) is a technology that makes it possible for users to make telephone calls over the internet or intranet networks. The technology does not use the traditional Public Switched Telephone Network (PSTN); instead calls are made over an internet protocol data network. VoIP has great benefits of increased saving, high quality voice and video streaming and several other added value services. Examples of VoIP software are: Skype, Google talks and windows live messenger (Di Wu, 2002).

2. Overview of VoIP

VoIP stand for Voice over Internet Protocol. VoIP enables us to compress and convert voice signal to digital signal and transmit it through Internet Protocol (IP)-enabled network like Internet, Ethernet and Wireless LAN. VoIP uses Internet protocol to manage voice packet over Internet protocol (IP) network.

Types of VoIP Service

There are different types of VoIP based on the infrastructures employed by the owner of the network. Listed are some popular services used in VoIP.

Computer to Computer

Computer to Computer service provides Internet telephony free using the same softphone software such as Skype, Instant Messaging, AOL etc. It is a software-based VoIP service and both the caller and the receiver must be using their computer in order to place calls. The following requirement must be met to use computer to computer VoIP service: softphone software, a sound card and good Internet service. With computer to computer VoIP service, the user may not be able to call either landline or mobile phones, and also the recipient must be online in order to call him/her.

Computer to phone and vice versa

This is a software-based and hardware-based service. Softphone software is used to route the call to an Internet protocol and hands off to a conventional telephone network . To use the service, one needs to subscribe and be charged at a low rate. Examples include Skype, MSN and Google Talk that provides the service to enable their customers to call landline from their computer. Computer to phone requirements are Internet-enabled phone and computer, VoIP service subscription, modem and Analog Terminal Adapter to convert the call signal to digital signal and also back to analog signal. Computer to phone does not allow emergency call users and needs to have a computer connected to the internet.

Phone to Phone

This is a hardware-based service that allows the caller and receiver to call each other using the Internet. Many telephone companies use this to handle long distance calls. VoIP convert the audio sound into data packets and transfer these packets over the Internet. It allows emergence calls and does not need PSTN for initiation and termination of calls.

2.1 VoIP system

Voice over Internet Protocol (VoIP) system is made up of End User Equipments, Network component, call processor, Gateway and Server. The End User Equipment is the part mainly used to access the VoIP system and communicates with other End point (Greg Tucker , 2004). End User equipments initiate and maintain the signals needed to establish calls over the IP network. It also helps to convert voice to data packet, voice and video communication, monitoring and network surveillance capability. They are connected to the network with cable and maybe wireless devices. Examples of end user equipment are IP-enabled phones, multimedia computers and VoIP enabled workstations installed with softphone . The Network Component includes routers, switches, cables, firewalls and wireless access point. There are many existing institution's Internet Protocol networks that will enable the installation of our VoIP system. The network must be strong enough to support the VoIP system because the impact of IP network is higher than the data network. Gateway is an optional component of VoIP network that helps to route calls, handle call connection, and manage terminal and Multipoint control unit (MCU) (Mona & Nirmala, 2002). It is responsible for providing address translation, bandwidth control, and access control and helps to improve the quality of service and security of VoIP network. Gateway serves as an interface between public switched telephone networks (PSTN) and VoIP network. Gateway is made up of Signalling Gateway (GS), Media Gateway (MG), Media Controller and multipoint control unit (MCU). Signalling Gateway helps to manage signal traffic while the media gateway manages media signal. The media gateway controller manages traffic between signalling gateway and media gateway while multipoint control unit is the part that establishes conference calls between three or more people for converged voice, video and data conference. It provides audio or combination of audio, video and data conference depending on the strength of participating endpoint (Greg, 2004).

Call processor is software running on end user equipment operating system that helps in call set up, call monitoring, user authorisation, signal coordination and bandwidth control. It translates a phone number to IP address. Figure 1 shows the components of VoIP system.
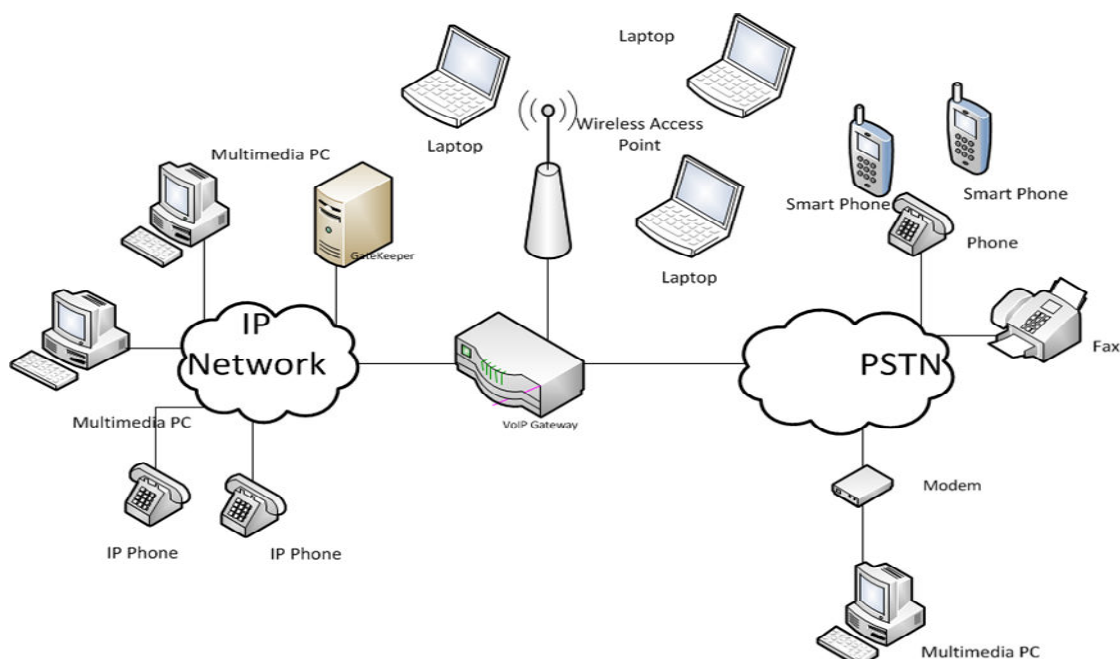

Figure 1   Typical topology of VoIP network

The server enables the establishment of call and support for other features in the system. The Session initiation server allows the user to forward calls to different location in the VoIP network.

2.2 VoIP Protocols and Standard
There are many protocols and standards for VoIP network (H.323, SIP, MGCP, RTSP, RSVP, SDP etc) but only the most commonly used ones are discussed.

H.323 protocol
H.323 protocol was developed by International Telecommunication Union and Telephony (ITU-T) in 1997 based on Real-time Transport Control protocol (RTCP) for sending voice, video and data over IP-based network (Rakesh, 2000). H.323 provides multimedia conferencing on Local Area Network (LAN) and brings together point to point communication and multipoint conferences. It was widely adopted because it is reliable and easy to maintain. The components include terminals, Gateway, Gatekeeper and Multipoint control unit (MCU). Terminal is the end point that provides real-time communication for the VoIP network. Gateway is the interface between IP network and the PSTN to another H.323 gateway. It provides Internet translation between the different terminals. Gatekeeper is the most vital component of the H.323 protocol. It acts as the central point for all calls and provides services for end point registrations. The functions of Gatekeeper are:

- Translate alias address to transport address
- Deny or grant access based on call authorisation, source and destination address.
- Call signalling with end point terminals
- Control the number of terminals permitted at a time in H.323 protocol (bandwidth management)
- Maintain the list of ongoing call H.323 calls to determine the busy terminals for bandwidth management.
- Rejection of call from terminal due to authorisation failure in the use of H.225 signalling (Call authorisation).

Another important component of H.323 protocol is the multipoint control unit (MCU). It acts as a bridge that enables two or more terminals and Gateway to participate in a multipoint conferencing. MCU is made up of Multipoint controller (MC) and Multipoint processor. Multipoint controller determines the capabilities of the network terminal using H.245 protocol stack but does not perform multiplexing of audio, video and data. Multipoint processor is responsible for multiplexing of media stream . H.323 consists of a number of protocol suites. The protocol suites and functions are listed:

- H.245 provides capabilities for channel usage, advertisement, establishment and conference control.
- H.255 for call control
- Q.931 for all signalling, call control and setup.
- Registration Admission status (RAS) is used for communicating with H.323, endpoint and gatekeeper. It provides interaction between H.323 and the gatekeeper.

The table 1 lists the use of protocol stacks in audio, video and data packet, and their transport protocols.

Data	Control and Signaling		Audio/Video	Registration
T.120	H.225.0 Call Signaling	H.245 Conference Control	RTP/RTCP	H.225.0 RAS
TCP			UDP	
Network Layer				
Data link Layer				
Physical Layer				

Table 1: Protocol stack and transport protocols (Greg , 2004).

H.323 Call Flow

The stages in audio connection between terminal, gateway and gatekeeper in H.323 protocol are:

- Registration of terminal and Gatekeeper registration (H255-RAS).
- Call routing between the terminal and gatekeeper (H.225-RAS and H.225-Q.931).
- Initial communication and capability exchange
- Establishing audio communication (open logical channel)(H.245)
- Audio transmission (RTP/RTCP).

H.323 follows the following steps to establish connections between clients (Mona &, Nirmala, 2002). The two endpoints will register with the gatekeeper.

- Terminal A will start Registration admission status message exchanged by initiating call to the gatekeeper.
- The information provided by gatekeeper is used by Terminal A to contact B.
- Terminal A sends a SETUP message to Terminal B
- Call proceeding message is sent by terminal B and request permission from gatekeeper.
- Terminal B sends an Alerting and connect message.
- Determination of master/slave terminal and open logical channels by Terminal A and B using H.245 message exchange.
- Establishment of Real-Time protocol (RTP) paths.

The problems of H.323 protocol are lack of flexibility, high connection setup latencies implementation difficulties. Figure 2 and 3 shows the architecture of H.323 and the connection procedures.
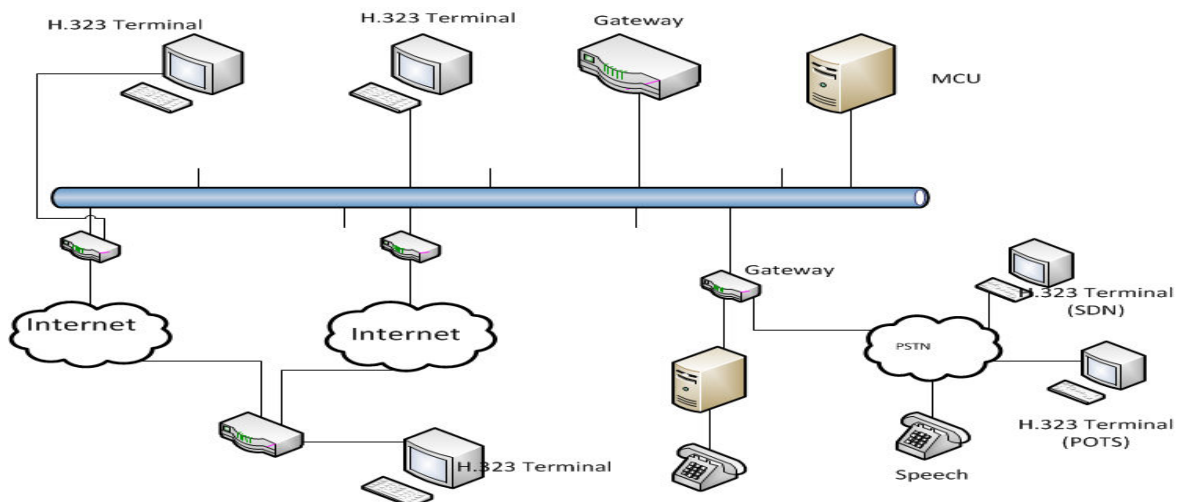

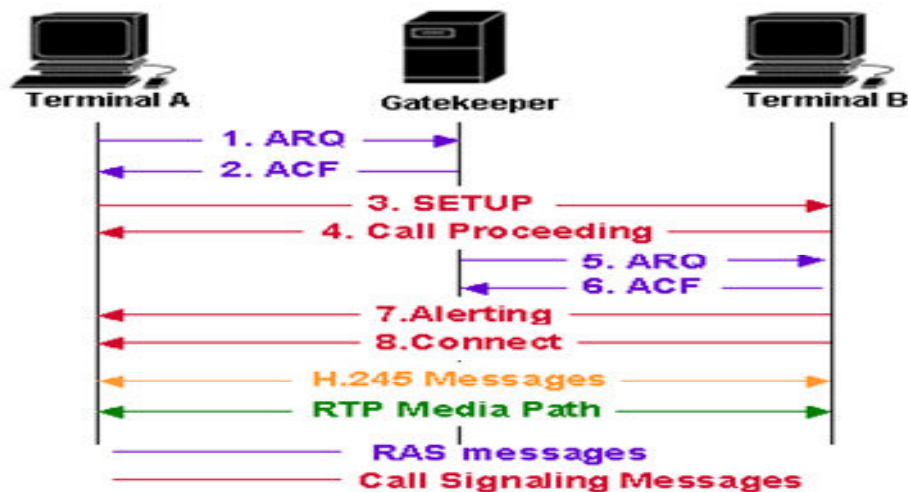Figure 2   Typical H.323 Architecture

Figure 3:   H.323 connection and call flows (Mona &, Nirmala, 2002)

Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) was developed by Internet Engineering Task Force (IETF) designed to initiate and terminate VoIP session with one or more participant (Rakesh, 2000), (Mona &, Nirmala, 2002). It is an ASCII-based peer to peer application protocol that initiates, modifies, creates and terminates interactive multimedia communication session between users. Because of the flexibility of SIP, it is used for audio, video and data packet transmission and communication. Session Initiation Protocol is similar to hypertext transfer protocol (http) made up of client-server. The client sends a request to the server and the server process the request and sends back to the client in a process called transaction. SIP is used in applications such as instant messaging, Apple chart, MSN messenger. The use of User description protocol (SDP) for carrying out negotiation for codec identification enables the support for user mobility by proxy and redirect server to the user's current location.

Components of SIP

SIP consists of User Agent and Network server. User Agent is the endpoint that acts on behalf of the user and maybe client or server. The client is called the user agent client and helps to initiate SIP request while the server known as user agent server receives the request, process it and returns the responses on behalf of the user. Network servers include registration server, proxy server and redirect server. Registration server is used for uploading current location of user, proxy server receives the request forward it to the next hop while redirect server on receiving request, determines the next hop and returns the address of the next hop server to the client instead of forwarding the request.

SIP call Flow

The six steps for establishing communication in SIP protocol are:
- User registration, initiation and location.
- Media determination.
- Determining the willingness to accept or reject calls.
- Call setup.
- Call modification and handling.
- Terminating the call (end of call).

SIP Messages

SIP defines several messages for communicating with the client and SIP server (Rakesh, 2000). Some of the messages are listed
- INVITE- used to initiate a call by inviting user SIP session call.

- ACK -for reliable exchange of invitation messages.
- CANCEL -to cancel a pending request by client.
- BYE- indicates termination of initiated call.
- REGISTER-provide information about user location to the SIP server.
- OPTION-used to get information about call capabilities.
- INFO-process out-of-bound information.

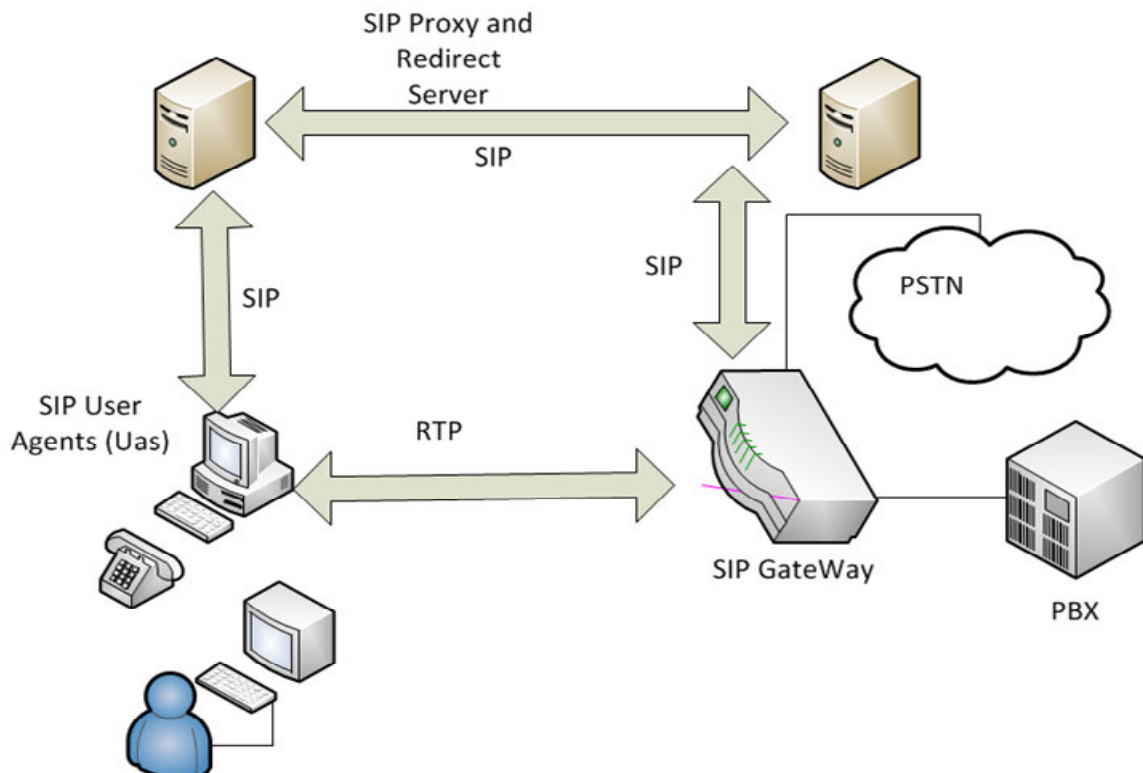Figure 4 and 5 shows the architecture of SIP standard and call flow in VoIP systems.

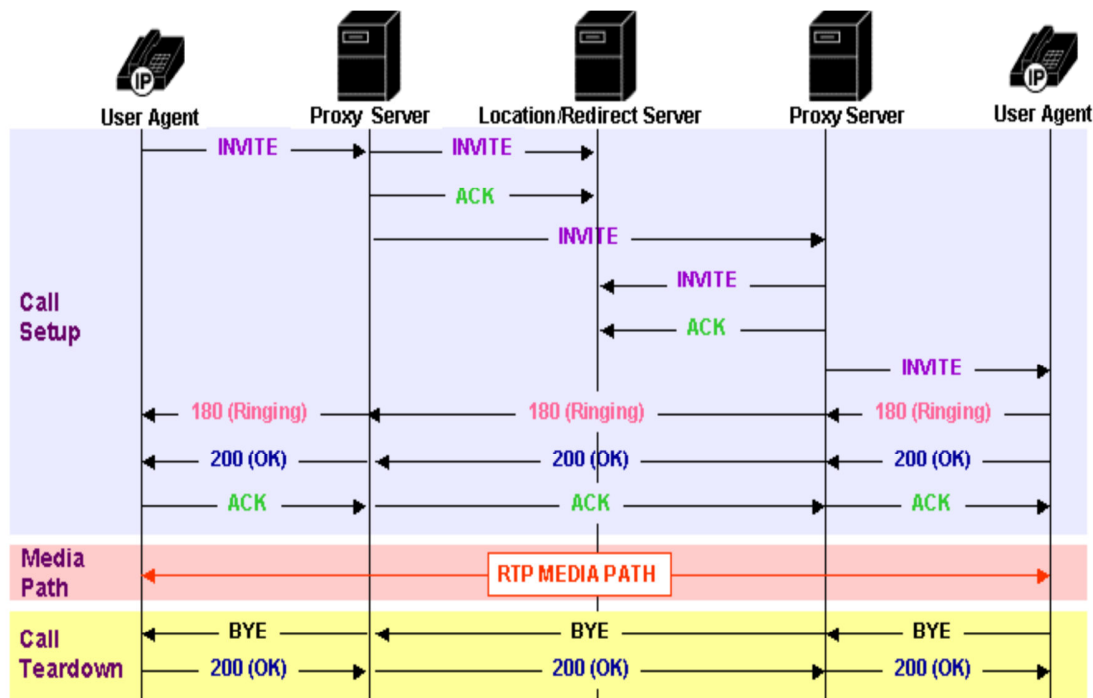Figure 4:   Typical SIP Architecture

Figure 5   : SIP call flows (Mona& Nirmala, 2002)

Comparison of H.323 and SIP protocols

SIP and H.323 provides similar functions but exhibit different strength depending on their applications. Both SIP and H.323 perform such functions as call set up and tear down, call control, waiting, holding and transfer, forwarding, return, identification, Call Park and capabilities exchange. H.323 is mainly used for multimedia conferencing which can support voice and video conferencing, white boarding, and data collaboration. Most people who use H.323 protocol claim it is complex and has overhead hence it is inefficient for VoIP system. It also lacks extensibilities and scalability required of VoIP signalling protocol. But SIP reuses most of its header field, encoding rules, error code and authentication mechanism of hypertext transfer protocol (HTTP). Table 2 summarises the differences between SIP and H.323 protocols

SIP	H.323
Simple to implement	Very complex protocol
Use binary representation for its message	Use textual representation
Not very modular	Very modular
Not scalable	Highly scalable
Need full backward compatibility	Does not need backward compatibility
Use complex signalling	Use simple signalling
Has a lot of elements	Has only 37 elements
Loop detection is difficult	Loop detection is easy
Large share of market	Backed by IETF

Table 2   Comparison SIP of and H.323

2.3 VoIP compression Algorithms

Codec (compression –decompression) is used to convert analog signal (video, voice) to digital data or convert digital data stream to analog signal. This helps to provide compression capability to save network bandwidth. There are different audio codec available for voice application (Hersent, 2011). They include:

G.711: approved in 1965 and is the simplest way of digitizing analog signal. The algorithm uses Pulse Code Modulation with less than 1% acceptable packet loss factor. The encoded audio stream of G.711 is 64kbit/s, so is worst in terms of bandwidth but the best in quality among the entire scheme.

G.722 was approved in 1988 and provides higher quality digital coding at 7 KHz of audio spectrum at only 48, 56, or 64bits. It is mainly used for all professional conversation voice application such as video conference and IP phone applications

G.722.1 is a wideband coder designed by Picturetel, which operate at 24bit/s or 32bit/s. It encodes a frames of 20ms with workload of 20ms, 16kbit/s version of G.722.1 supports windows messenger.

G.723.1 was approved in 1995 for use in H.323 communication and UMTS 99 video cell phones. It uses frame length of 30ms and needs a workload of 7.5ms in 64kbt/s or 5.3kbit/s operation modes. The algorithm is not designed for music and its difficult to be used in a fax and modem signal transmission. The International Telecommunication Union and Telephony recommends it for use in narrow band video conferencing and 3G wireless multimedia devices.

G.726 was approved in 1990 and uses Adaptive differential Code modulation (ADPCM) techniques to encode G.711 bit stream in words of 2, 3, or 4 bits resulting in bit rate of 16, 24, 32 or 64kbits/s .

G.728 uses low delay, codec executed linear prediction (LD CELP) coding techniques with a mean opinion score (MOS) similar to G.726. The algorithm is used for Fax and modem transmission, and also for H.323 video conference.

G.729 is conjugate-structure, Algebraic Code Excited Linear Prediction (CS-ACELP) speech compression algorithm approved by ITU-T for use in voice over frame relay application. It produce 80-bits frame encoding 10ms of speech at a bit rate of 8kbit/s. The scheme is not designed for music and does not support Dual-Tone Multi-Frequency (DTMF) signalling tones reliably. Listed in table 3 are properties of common voice codec schemes.

Codec	Bit Rate	Payload	Packets per Seconds (pps)	Quality	bandwidth	Sample period	algorithm
G.711	64kbit/s	160bytes	50pps	Excellent	95.2kbps	20ms	PCM
G.729	8kbit/s	20bytes	50pps	Good	39.2kbps	10ms	CS-ACELP
G.723.1	6.3kbit/s	24bytes	34pps	Good	27.2kbps	30ms	MPC-MLQ
G.723.1	5.3kbit/s	20bytes	34pps	Good	26.1kbps	30ms	ACEP

Table 3   Different codec scheme

2.4 Advantages of VoIP

There are many advantages of deploying VoIP network by companies, individuals and especially educational institutions. Some of these advantages are:

- **To save cost:** VoIP system is mostly deployed in existing network thereby reducing the cost of buying network equipments and wiring. IP network equipments are very cheap, interoperable and can be sourced from multiple vendors compare to traditional PSTN landline equipments. Another significant reason for deploying VoIP is because it can reduce the cost of making long distance calls. Softwares such as Skype, Windows live messenger e.t.c allow free PC to PC calls. Calls to mobile and landline are at reduced price.
- **Advanced Multimedia application:** VoIP network enables multimedia applications, voice and data such as voice/video conferencing, white boarding, file transfer to increase productivity and creates a more flexible

communication system learning environment.

- **VoIP network integrates voice and data:** VoIP integrate both Voice, video and data together to save management and operational manpower and effective use of communication links between network sites.

- **Widespread availability of Internet protocol (IP):** IP network is readily available all over the world, with people having access to PC linked to internet. Furthermore, availability of gateways to/from PSTN allows calls to use VoIP for voice and video calls (Mona & Nirmala, 2002).
- **Reduce the cost of Ownership:** VoIP integrates data and voice communication traffic into a single network thereby reducing the cost of infrastructural ownership and maintenance redundancies. It brings different network elements together such as call server, application server and client server (Bhogal et al, 2004).
- **Efficient utilisation of network resource**: VoIP network improves the network bandwidth efficiency and quality of service by eliminating silence during conversation, reduce repetitive pattern in human speech and increases inefficient data throughput.
- **Greater operational flexibility**: IP-based network is made up of different layers of separate components that can be integrated to form a whole system. This allows the system, application, and services to be dynamically managed resulting in a customised, flexible and extensible system.

2.5 Challenges Associated with VoIP Implementation

There are number of challenges hampering the implementation of VoIP network for everyday use . Because the Internet was not designed to transfer real time data, network equipment drop most packets leading to queues that cause jitter in packet transfer. Routing in IP network is time consuming compared to switching network. Delay and packet loss in network affect quality of service (QoS). Efforts are on the way to reduce these challenges which include the use of Reservation protocol (RSVP), design of high priority queues for real time traffic and the use of multiprotocol label switching (MPLS) to speed up packets through routing points. Another problem of the Internet is that it is not managed by a single centralised operator that can coordinate the flow of calls, but made up of different networks and service providers which makes it difficult for providing real time traffics.

For efficient and effective real-time communication, network bandwidth is of paramount importance. A network is made up of individual nodes and links that generate huge amount of traffic. If these nodes send more packets greater than the network can accommodate, it can lead to network congestion which can cause QoS problems. When a network congestion occurs, packets need to be queued leading to jitter and latency in the network . The network should be designed to support data and real-time transmission of voice and video for better service quality.

When discussing problems of VoIP implementation, the issue of power failure and backup system are mentioned. Traditional telephones operate on internal 48 volt internal power supply. Therefore it can still continue to work even if there is power outage. So, they need to have backup power system using Uninterruptible power supply (UPS) and proper power back up assessment done regularly to ensure there is sufficient power for the VoIP system.

The use of softphones software is another challenge of VoIP implementation. These softphones are vulnerable to worms, virus, Trojan horses and spyware attack and can compromise the security of the network. Computers connected to the Internet can be attacked if a user visits a compromised website even if he/she did not open the website. Traditional PSTN telephone can provide emergency calls because it is tied to physical location and emergency service provider can easily track caller's location to the emergency dispatch office. But an individual using VoIP network can be from anywhere around the globe, thereby making emergency call location difficult because the caller's location and identity can't be traced easily. Even if the VoIP service provider can provide for emergency calls, there is lack of standard in VoIP environment.

Most of the Wireless devices in the market today come with Wired Equivalent Privacy security algorithm. WEP has been found to be very weak and can be cracked with free software available online. With the recent introduction of Wi-Fi Alliance's Wi-Fi Protected Access (WPA/WPA2), there is an improvement which can be integrated into VoIP system. VoIP compression-decompression algorithm (codec) and gateway can be encrypted to provide improved security to avoid the latency problem of link or network-layer encryption.

2.6 VoIP Quality of Service (QOS)

Quality of service is the ability to control and measure data transmission rate or throughput and error rate and provides better, predictable result to selected network traffic over different underlying technologies. Voice and video

packet is intolerant to packet loss, jitter and delay unlike traditional data packet that has minimal delay in delivery of their packets; there is no need to address Quality of Service (QoS) issues. To convey voice traffic over IP network, there is need to ensure reliable arrival of the packets. Therefore, quality of service provides dedicated bandwidth, controlled jitter and latency, and also improved loss characteristics. This section will discuss these Quality of Service (QoS) parameters as they affect VoIP.

Delay: delay is the amount of time it takes to transmit data packet from source to destination. It is the end to end delay or time delay incurred in speech by VoIP system. To ensure high quality, delay should be controlled so that communication delay should be less than 150ms (Di Wu, 2002), (Jeomgoo, Inyong & Suh, 2010). Delay is caused by three major factors such as codec algorithm, queue algorithm of communicating equipment and variable delay caused by network condition at the time of transmission. It is important to minimise delay to an acceptable level of 150ms to ensure better quality of service. Codec (compression-decompression) introduces three kinds of delay:
- Processing or algorithmic delay which is the time required for codec to encode one voice frame.
- Look ahead delay, the time required for a codec to examine part of the frame
- Frame delay is the time required for sending system to transmit a single frame.

Compression algorithm affects delay, the higher the level of compression the higher the delay the system.

Packet Loss: Packet Loss is caused by hybrid circuits where it changes from 4-wire to 2-wire. It occurs when there is packet drop in the network leading to loss. And VoIP packet is very sensitive, packet loss can greatly affect the Quality of Service (QoS) of VoIP system. The acceptable packet loss in VoIP system is below 1%, and anything beyond this limit is unacceptable. The major causes of packet drop are congestion in the network and buffer size, every effort should be made to ensure the network is design to counter network congestion.

Jitter: Jitter is the variation in inter-packet arrival rate which introduce variable transmission delay over the network. Because VoIP use User Datagram Protocol (UDP), IP network cannot guarantee the delivery time to the packets leading to inconsistent rate of arrival. Jitter can be removed using jitter buffer, allowing an equal stream to collect a packet and store them long enough to permit slowest arrival in correct sequence. Jitter buffer adds to the overall delay. To support VoIP traffic reliability, the network should guarantee the following:
- Packet-forwarding latency that should be within maximum tolerable for VoIP conversation.
- Packet forwarding jitters within tolerable level to sustain a VoIP session.
- And guarantee bandwidth and capacity for VoIP session in case of network congestion.

The network should provide low latency and jitter to maintain high quality. We need to control all the mentioned parameters to ensure high quality of VoIP service for students and staff. Sometimes we also need to prioritised network application and limited shared network resources.

2.7 Security of VoIP

The Internet protocol (IP) used to carry traffic over packet-switched data network is vulnerable to different types of attacks by hackers. A hacker can use sniffer or other publicly available hacking tools to modify, identify, store and play back voice traffic transmitted over IP network. The hacker can have access to users' call and determine the identity of the caller. Another security breach of VoIP is eavesdropping, where a hacker can listen to the communicating parties. Despite the facts that the end user equipment is deployed on an internal protected network, the endpoints are not individually protected by the network firewall and maybe threatened.

Softphone software installed on the endpoint may have vulnerabilities in the operating system (OS) it is running on. The operating system should be installed with an anti-virus software, virus detection host-based firewall or host-based intrusion detection (Di wu, 2002). The PSTN media gateway is also vulnerable to attacks; IPSec can be used to prevent interference with user call and to prevent unauthorised call from being set up.

Many security requirements are needed to secure VoIP service and application. There include:
- Protection of privacy of call conversation.
- Authentication of calls and end points protection from insecure of network resources.
- Ensuring of correct billing by the service provider and protection of billing information form authorised access.
- Protections of caller behaviour or statistical information from authorised access.

- Protection of network servers and endpoints from well known threat and man in the middle attacks.

3 CONCLUSION

To keep abreast with the global technological change and maximizing cost, a reliable and cheap means of communication is inevitable. This paper has been given a critical and succinct digest on voice over Internet protocol (VoIP). It systematically educates one on the VoIP system, its standards, protocols and security challenges. VoIP however is presented as a sure alternative over the Public Service Telephone Network (PSTN).

REFERENCES

1   Bhogal Amit; Hamza Idrissi; Thai-son Nguyen; Michael Wakahe (2004) "*Voice over Internet Protocol*". Online at www3.sympatico.ca/albert_nguyen/project/VoIP.pdf

2   Di Wu (2002), " Performance studies of VoIP over Ethernet LANs", Online at http://www.autoresearchgateway.ac.uz/bitstream/10292/677/5/Diw/pdf. (Accessed : 28/06/2011)

3   Greg S Tucker (2004) "Voice over Internet Protocol (VoIP) and security", Online at www.sans.org/reading_room/whitepapers/voip/voice-internet-protocol-voip-security-1513.   (Accessed: 30/08/2011)

4   Hersent Oliver (2011) *"IP telephony: deploying VoIP protocol and IMS Infrastructure"*, Second Edition, John Wiley and sons, UK

5   Jeomgoo Kim, Inyong Lee, Suh ron Noh (2010) "*VoIP Quality of Service Design of measurement management process model"*, International Conference on information science and applications (ICISA) Pp 1-6.

6   Mona Habib, Nirmala Bulusu (2002), "Improving QOS of VoIP over WLAN (IQ-VW)". Online at http://www.cs.iccs.edu (Accessed: 15/05/2011)

7   Rakesh Arora (2000) *"Voice over IP: Protocol and Standards"*, Online at http://www.cse.wustl.edu/~jain/cis788-99/ftp/voip_protocols.pdf (Accessed; 29/05/2011)

CALL FOR PAPERS

The IISTE is currently hosting more than 30 peer-reviewed academic journals and collaborating with academic institutions around the world. There's no deadline for submission. **Prospective authors of IISTE journals can find the submission instruction on the following page:** http://www.iiste.org/Journals/

The IISTE editorial team promises to the review and publish all the qualified submissions in a **fast** manner. All the journals articles are available online to the readers all over the world without financial, legal, or technical barriers other than those inseparable from gaining access to the internet itself. Printed version of the journals is also available upon request of readers and authors.

IISTE Knowledge Sharing Partners

EBSCO, Index Copernicus, Ulrich's Periodicals Directory, JournalTOCS, PKP Open Archives Harvester, Bielefeld Academic Search Engine, Elektronische Zeitschriftenbibliothek EZB, Open J-Gate, OCLC WorldCat, Universe Digtial Library , NewJour, Google Scholar