

# Impact of VoIP and QoS on Open and Distance Learning

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## ABSTRACT

Voice over Internet Protocol (VoIP) is becoming a reality in many organizations. The potential for mobility in voice over wi-fi networks will derive demand for the technology. Wireless VoIP is poised to rival VoIP as an alternative telephony tool. Internet has been used to transport data in the form of packet. In the past, Internet did not support any kind of sophisticated quality of service (QoS) mechanism. Although the type of service (TOS) field in the Internet protocol (IP) header has been existing and has been allowing the differentiated treatment of packets, it was never really used on a large scale. The voice is sensitive to delay and jitter so bandwidth must be guaranteed while transporting it. With the extensive use of Internet for carrying voice, there is a need to add QoS functionality in it. QoS with reference to VoIP has been discussed in the paper. Limited bandwidth and network latency are the issues which need to be considered while using wireless LAN for packetized voice data. Efforts of standards like 802.11e which will take care of these issues, have also been explored. The impact of these technologies on distance education has also been explored in the paper.

**Keywords:** Computer Networks; Voice over Internet Protocol (VoIP); TCP/IP; Wireless LAN; Quality of Service (QoS); Open and Distance Education; Learner Support.

## INTRODUCTION

Social changes always follow technological changes; indeed each new invention brings about advances in our lifestyle. Not long ago letter writing was the only means of keeping in touch with relatives and friends who did not live nearby. With Alexander Graham Bell's invention of the telephone, conversations could be held at a distance without the requirement of displaying oneself. Marconi's wireless brought the news of the world into the living room of the nations. The communication technologies kept evolving till date. Radio, television, satellites, mobile phones and other developments followed. A few years ago, the telephone companies used to have a monopoly on voice communications. This situation is changing rapidly. Media and communications are expanding in directions never dreamed of several years ago. Co-axial cables are competing with telephone wires. Satellites and mobile technologies have made cellular phones proliferate everywhere. Desktop and portable computers equipped with the required interfaces and connected to the Internet, are becoming almost ubiquitous. Miniaturization and portability of electronic devices, as well as effective means of communicating are omnipresent in

the digitalized world around us. IP has become an accepted standard for communications over data networks worldwide, and enterprises have successfully implemented it. Recently IP was used to transport voice between networks and devices. Voice can now be packetized and sent over a data network using VoIP technology (Handley, Schulzrinne, Schooler, & Rosenberg 2001). This revolutionary technology is now setting off a new trend i.e. the convergence of voice and data networks of new applications. Current landline telephone systems are based on circuit switching. A closer look at circuit switching will ensure the better understanding of VoIP and its packet switching nature.

### **Circuit Switching**

In PSTN telephone architecture, when a phone call is made to a particular number, a dedicated channel is assigned for the connection. This channel circuit typically uses 64 Kbps (kilo bits per second) bandwidth in each direction, totaling the transmission rate to 128 Kbps. The voice traffic passes through the carrier switch at the callers and receivers end, and all the switch ports that support the connection are used throughout the duration of the call. TDM (Time division multiplexing technology, which transmits multiple signals simultaneously over a single transmission path) is used to accommodate more connection within the limited capacity. Circuit switching has many disadvantages. When someone talks for 10 minutes, bandwidth used is 128 Kbps (or 16 kilo bytes per second) that the circuit remains open. The total transmission for the length of the conversation is 9600 KB or roughly 9.4 MB. During this time, the circuit is continuously open between the two phones. In a typical voice conversation while someone talks, the other party listens. This means that only half of the connections are in use at any given time. There are lots of gaps or silence period when no party is talking. Since the circuit is dedicated the bandwidth during the silence period is unutilized and wasted. So, In effect the file can be cut in half down to about 4.7 MB. Subsequently, the switch ports at the local exchanges are also dedicated for the entire duration of the call regardless of the silence. This makes bandwidth provisioning difficult and time consuming.

The PSTN switching infrastructure may not be able to handle higher traffic easily and may buckle. The carrier is not able to cut down on operation costs and consumers do not get any cost benefits. The billing is based on time and distance, not on amount of traffic. It is not easy to initiate new services and/or incorporate any changes in a PSTN network. Capital costs are high partly due to the over-engineering required to support peak time traffic. The network cannot be integrated well with multimedia applications and existing voice margins are very flat. Different networks are required for different services.

### **Packet Switching**

VoIP uses packet switching, a technology commonly used by data networks. The message is divided into small pieces called packets. While circuit switching keeps the connection open and constant, packet switching opens the connection just long enough to send a small data packet from one system to another. Each packet in a packet switched network contains a destination address. This allows all packets in a single message to be dynamically routed in different paths over the network depending on availability. The destination computer reassembles the packets back into their proper sequence. A technique called statistical multiplexing is used to accommodate more signals in a channel. This technique analyzes the traffic and dynamically changes its pattern of interleaving to use all the available capacity of an outgoing channel. Packet switching minimizes the connection time between two systems and reduces the load on the network. It frees up the two systems

communicating with each other so that they can accept information from other systems simultaneously. Several telephone calls can occupy the amount of space used by only one call in a circuit switched network. The size of each call can be further reduced with the use of data compression.

### TYPES OF INTERNET TELEPHONY

When Internet is used to transport voice, four types of communication according to the terminal used by each of the two correspondents are possible (Bruno, 2001). These are PC to PC Communication, PC to phone connection, phone to PC communication and phone to phone communication. These communications modes are explained below.

#### PC to PC Communication

Computers can be a part of Intranet (LAN) or they are connected through dialup lines or Internet. To communicate via voice over packet switched networks such as Internet the ITU (International Telecommunication Union) standard H.323 "Packet based multimedia communication systems" is used. When Computers are part of intranet they can communicate directly i.e. from H.323 terminal (a) to H.323 terminal (b2) using real time protocol (RTP). Real time control protocol (RTCP) is used when H.323 terminal (c) is communicating with an H.324 terminal (d). (access and terminating via telephone network).

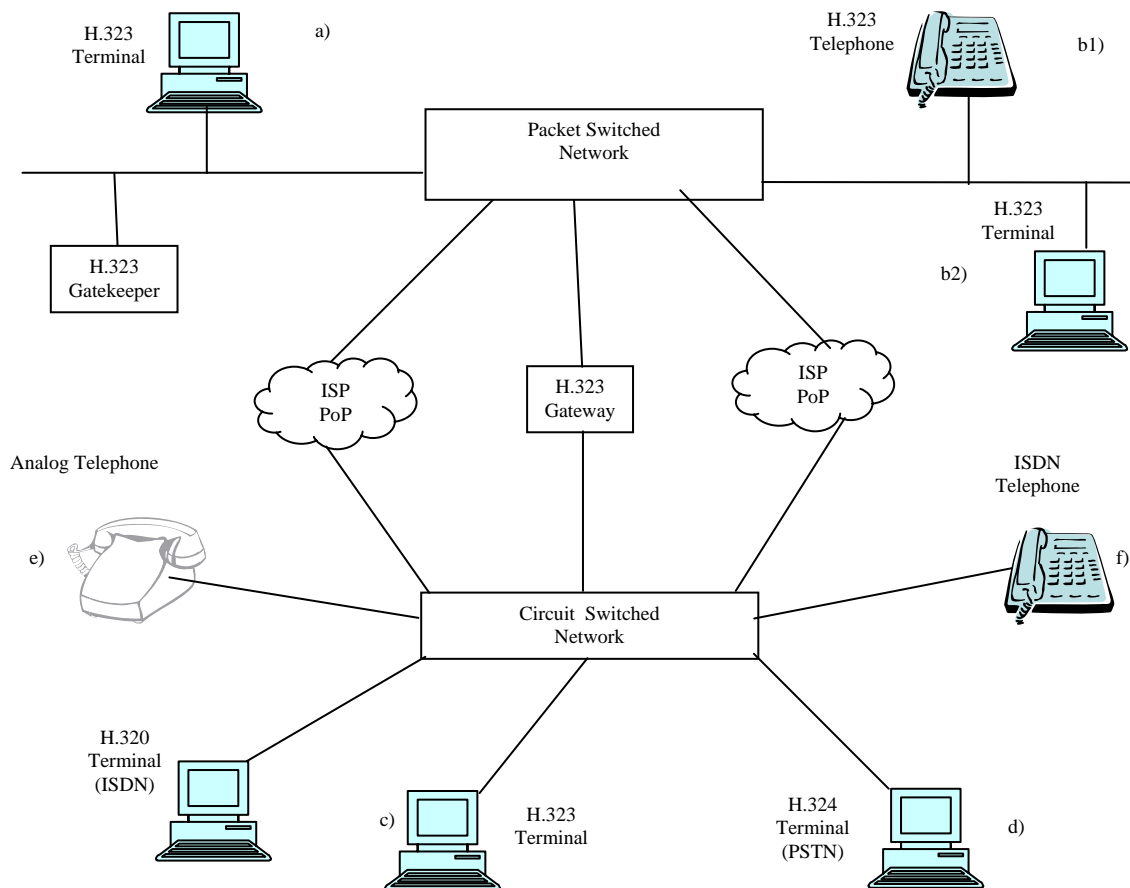


Figure: 1 VoIP Configurations

### PC to Phone Communication

In this mode of communication, a H.323 terminal (a) communicates with an analog telephone (e). For this conversation to take place, a gatekeeper is required for connection control between terminal and telephone. A gateway for the conversion of the language segments arriving in packages in continuous digitalized language (and vice versa) is also necessary.

### Phone to PC Communication

This type of communication requires an Internet telephony service provider (ITSP) which serves as gateways and makes procedures available to the selected PCs (i.e. via its IP address) at the telephone. Such procedures are mostly offered as a subscription services.

### Phone to Phone Communication

This type of communication takes place between an analog phone and ISDN phone. The analog phone (e) dials the gateway number of ITSP and then the number of the destination phone (f). The packaging and transfer of the voice data goes via the Internet to other gateway of ITSP near the participant (f), from there comes the call to (f) via ISDN.

## THE ARCHITECTURE OF INTERNET TELEPHONY

The telephony services of the Internet are built on a hierarchy of packet switching protocols, as illustrated in figure 2. The function of

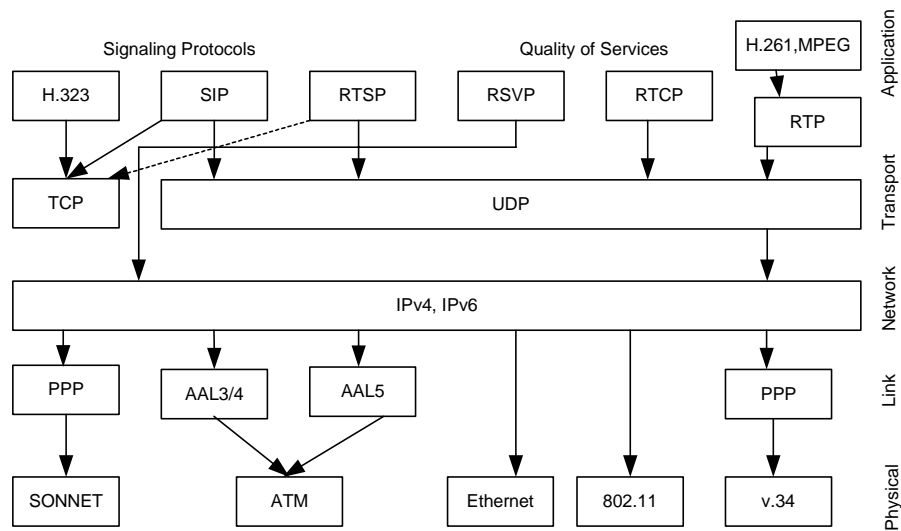


Figure: 2 Protocol Architecture of Internet Telephony

signaling protocols (Mortada & Probst 2001) include routing, reservation of resources, call acceptance, address translation, establishment of the call, as well as its management and billing. In an Internet environment, the routing is controlled by protocols such as BGP (border gateway protocol), the reservation of resources by

RSVP (resource reservation protocol) (Barden, Zhang, Berson, Herzog, & Jamin, 1997) or other such protocols. At the same time signaling protocols such as SIP (session initiation protocol) and H.323 have been developed for the translation of addresses of the application layer, and for establishing and controlling the calls. On the other hand, at present no billing protocol exists for Internet telephony. The Internet telephony requires a set of controls for the establishment of connections, exchange of capacities and conference control. Presently, H.323 of ITU and SIP of IETF (Internet Engineering Task Force) are the two protocols, which have been designed to meet these needs.

### H.323

The H.323 specification was ratified by the ITU (ITU-T Recommendation H.323). It defines how voice, data, and video traffic will be transported over IP based LANs (local area networks) and WANs (wide area networks). H.323 is a protocol family comprising of various protocols for coding voice data (audio e.g. G.711, G.722), video signals (e.g. H.261, H.263) and their transfer as well as for data communication (e.g. T.124). In addition, H.323 defines protocols for the control of access building and terminating and for communication control and switching (Minoli, & Minoli, 1998). In figure 3, the most important protocols belonging to H.323 family are shown. The RTCP controls the quality of the voice data and video data transfers via RTP. H.225 defines the protocols for the connection building and disconnecting, the packaging of the bit stream control from the voice data transmission status, etc. H.245 defines the procedures for the exchange of communication parameters between the terminals (supported services, codecs, flow control, data conversion, etc).

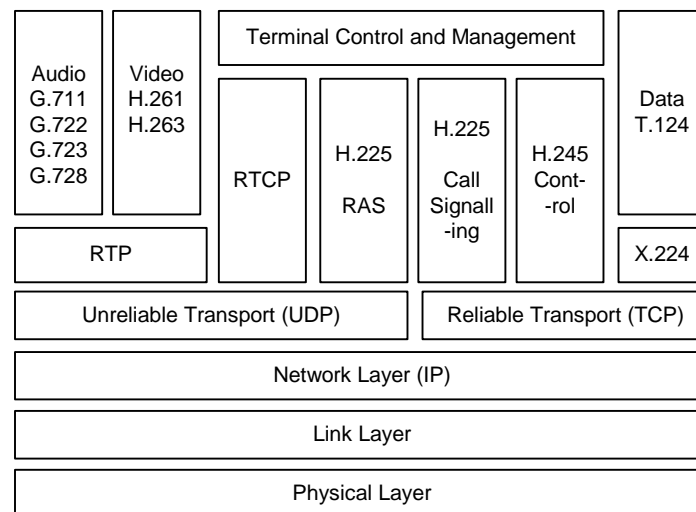


Figure: 3 H.323 Protocol stack

For the transfer of audio and video packet, UDP (user datagram protocol) is used and not TCP (transmission control protocol), as a correction of a mistake and the confirmation of the arrival of the voice data packages with TCP would lead to unsupportable delays. Packet with signaling and control information is transferred with TCP.

A codec, which stands for coder-decoder, converts an audio signal into a compressed digital form for transmission and back into an uncompressed audio signal for replay.

### Session Initiation Protocol

SIP is a signaling protocol for establishing real-time calls and conferences over IP network standardized by IETF. Each session may include different types of data like audio and video although currently most of the SIP extensions address audio communication. As traditional text based IP, it resembles the HTTP (hyper text transfer protocol) and SMTP (simple mail transfer protocol). SIP uses SDP (session description protocol) for media description. The basic architecture of SIP is client and server in nature. The main entities in SIP are the user agent, the SIP proxy server, the SIP redirect server and the registrar (Handley, Schulzrinne, Schooler, & Rosenberg 2001; and Donovan, & Cannon, 1998).

SIP is independent of the packet layer. The protocol is an open standard and is scalable. It has been designed to be a general purpose protocol. However, extensions to SIP are needed to make the protocol truly functional in terms of interoperability. The protocol also enables personal mobility by providing the capability to reach a called party at a single, location independent address.

### SIP Versus H.323

The ITU supporters have the objective of gaining the support of several manufacturers, while those of SIP doubt the interoperability of the H.323 products and demonstrate the technical advantage of telephony with SIP. Both SIP and H.323 define mechanisms for call routing, call signaling, capabilities exchange, media control, and supplementary services. SIP is a new protocol that promises scalability, flexibility and ease of implementation when building complex systems. H.323 is an established protocol that has been widely used because of its manageability, reliability and interoperability with PSTN. In term of functionality and services supported, version 2 of H.323 and SIP are very similar. The advantages of SIP include flexibility in adding new parameters; it is easy to implement and to debug and its uses in supporting a textual description session. On the other hand, H.323 has certain advantage over SIP, which consists of its strong interoperability with traditional telephony, its backward compatibility, and its power to change capacities between entities on a H.323 network. It can be noted finally that the competition between these two protocols serves to diminish their difference in each successive version, as their designer closely scrutinize each other's work.

### QUALITY OF SERVICE

QoS is a major concern for the converging network. It is simple economics. If better quality in any dimension like better sound quality and better application availability is provided to the users, they will spend more time using applications and services. The notion of QoS as applied to communications provides for the establishment of two lists submitted to the network during a request for connection. The first contains the desired quality parameters to be reached and maintained. The second specifies the minimal acceptable value for this quality of service. None of the criteria generally defined for a telephone conversation on the Internet can screen the quality of service because of delay and reliability. The delay of transport is precarious by nature, the volume is not guaranteed and the delivery of the packets is not guaranteed (ref. [http://qos.itc.ukans.edu/ipqos/ip\\_qos.htm](http://qos.itc.ukans.edu/ipqos/ip_qos.htm))

Although QoS usually refers to the fidelity of the transmitted voice and facsimile documents, it can also be applied to network availability, use of value added features like conferencing and calling number display, and scalability.

### **Factors affecting QoS**

The Internet is not currently adapted to a reliable transport of voice data. This is due to the way network operates, the slowness of message transmission (and eventually re-transmission) and of sending and receiving of packets. The network is sometime too slow in routing the packets and respects neither the order nor the time interval between them. Within the network, packets are treated equally i.e. they take their place in the queues of the routers. It is without regard to the information they carry or which services they provide. It is known that the router is sometime slow to deliver the accumulated messages. It then follows that the telephony, which requires very short routing delays, is not well suited to this methods of operation and that the routers are not adapted to the constraints of real time. However, the loss of voice packets in data networks is preferable to their re-transmission. Even if some packets are lost in transit, the loss will be imperceptible to the human ear, thus preserving as an acceptable listening quality (Gupta, 2003). There are three factors that can profoundly impact QoS. These are delay, jitter and packet loss.

#### **Delay**

High end-to-end delay in a voice network gives rise to echo and talker overlap. Echo becomes a problem when a round trip delay is more than 50 milliseconds. Since echo is received as a significant quality problem, VoIP system must address the need for echo control and implement some mean of echo cancellation. Talker overlap (the problem of one caller stepping on the speech of other caller) becomes significant if the one way delay become greater than 250 milliseconds. The end-to-end delay budget is therefore the major constraints and drives the requirement for reducing delay through a packet network. A technique called silence suppression, detects whenever there is a gap in the speech. It suppresses the transfer of pauses, breaths, and other periods of silence. This can amount to 50-60 percent of the time of a call, resulting in considerable bandwidth conservation.

#### **Jitter**

Jitter is the variations in inter packet arrival time due to variable transmission delay over the network. Removing jitter requires collecting packets and holding them long enough to allow the slowest packet to arrive in time to be played in the correct sequence. This causes additional delays. The jitter buffers add delay, which is used to remove the packet delay variation that each packet is subjected to as it transits the packet network.

#### **Packet Loss**

IP networks cannot provide a guarantee that there will be no packet loss and the packet will certainly be delivered in the order. Packets may be dropped under peak loads and during period of congestion caused by link failures or inadequate capacity. Due to the time sensitivity of voice transmission, the normal TCP based re-transmission schemes are not suitable. Packet losses greater than 10 percent are generally not acceptable.

#### **QoS Solutions**

There are three techniques that can be used separately or in combination to improve the QoS in the network.



### **Controlling Network Environment**

It is necessary to provide a controlled networking environment in which the capacity can be pre-planned and adequate performance can be assured. This would generally be the case with a private IP network or an Intranet that is owned and operated by a single organization.

### **Using Management Tools**

Management tools can be used to configure the network nodes for VoIP (including upgrading RMON probe and protocol analyzers to recognize and decode VoIP), monitor performance (by determining existing call traffic statistics and predict future statistics including cost, average simultaneous calls, average duration, and source/destination pairs), and manage capacity and flow on a dynamic basis. Identify the type of traffic on the network and prioritize them. Traffic can be prioritized by location, by protocol, or by application type. This allows the real type of traffic to be given precedence over non-critical traffic. Voice may actually not be the most important. Queuing mechanism can also be manipulated to minimize delays for real time data flow.

### **Adding Control Protocols and Mechanism**

Control protocols and mechanisms should be added that help avoid or alleviate the problem inherent in IP network. Protocol like RTP and RSVP can also be used to provide greater assurances of controlled QoS with in the network. RSVP operates by reserving bandwidth and router/switch buffer space for certain high priority IP packet like those carrying voice traffic. In effect, RSVP enable a packet switched network to mimic some of the characteristics of circuit switched multiplexer network. RSVP is still only able to set a path to high priority traffic on a best effort basis (Barden, Zhang, Berson, Herzog, & Jamin, 1997). Other mechanism like admission controls and traffic shaping may also be used to avoid overloading of a network.

## **IMPLEMENTING VOIP**

Implementing VoIP doesn't mean throwing out the existing investments made to telecom setup of the organization. The beauty of the system is its modularity. It can be easily integrated with existing setup. To implement VoIP, analysis of the existing network topology and graphical distribution of the office is required to be done. Other deciding factors include internal investment required, running costs, etc. There are three types of components required for deploying VoIP.

### **Front Hand Equipment**

The front end of VoIP setup consists of IP terminal. This can be a IP phone, a voice box or a standard desktop PC running a soft form terminal application. Headphone can be used for PC based IP telephony and would require a full duplex sound card. IP phone are different in the technology used. While a regular phone uses electrical signals for carrying voice, IP phone uses digital voice and carry it over IP like a regular data packets.

A voice box is similar to an IP telephone, except that it houses the circuitry to accept input from a standard analog phone. In this case, existing analog telephone instrument is connected to a voice box, which in turn connects to existing data network to send voice over it. Voice boxes are readily available off-the-shelf from various vendors, and come with different numbers of input ports (two lines, four lines or eight lines).



There are terminal applications on the desktop that can be used. These would have dial pads, and require a full duplex sound card for operation. Microphone and headphone should be used with PC, and telephone numbers or IP address of any other IP telephony device can be dialed.

### **The Back End Equipment**

The device at the backend in a VoIP setup includes an IP gateway, gatekeeper, voice router and software to manage and maintain the voice quality. Software is deployed to provide additional services like call waiting, forwarding, answering machine, directory services etc. The gateway interfaces data network to the front hand device and provides simple services like IP address and telephone number mapping. A gatekeeper store information about the IP calling device on network, their authentication and data on route to be taken for a voice call.

They do address translation (A phone number to IP) and may have access control rules. Gatekeepers can be roughly equated to a company PBX, but working on IP. There are some solutions that allow to use existing PBX and implement VoIP technology on them. Voice router is needed to route voice traffic.

It will take VoIP data as input and route it over Internet through existing gateway or over a private network. This can be hardware or software based wherein hardware is used for small call volumes while software is used for larger once. This is because software is scalable and capacity can be added to it easily. Schemes like QoS are implemented on calls. Priority rules are also defined here.

### **The Carrier**

Once a call is originated, a carrier for voice traffic is required for voice bandwidth. As the QoS is very important, ISPs with their own gateways for the data traffic should be preferred over the regular ISP. QoS can be ensured with appropriate service level agreement (SLA) laid out to ensure optimum end-to-end voice quality. Factors determining voice quality are the amount of bandwidth allowed to voice, good mean opinion scores (MOS) and good round-trip times. Checks can be incorporated to ensure better 'handshakes'. The carrier is responsible for terminating voice call to PSTN phones abroad. For this termination partners who talk over traffic and route it to PSTN are required. They need gateway to translate network protocol to PSTN signal. However, if a voice call is to go on to public Internet in any manner (to a dial up PC user as a receiving party), the quality of service cannot be assured.

### **VOICE OVER WIRELESS LAN**

When most people think of wireless LANs (WLANs), they generally only consider transferring data while using applications such as a web browser, e-mail client, for file transfer, etc. It is possible to use a WLAN as the transport system for carrying telephone traffic from mobile users as well. A significant benefit of mixing telephone traffic with data on a WLAN is to provide mobility and make use of a common infrastructure (Lipset, 2004). The support of a common system for both data and voice traffic is generally simpler and less expensive than two separate entities.

The use of WLAN phones within the offices of an enterprise is compelling as well. In this case, an organization company can avoid the need to wire and rewire telephone outlets as the organization size shifts. In fact, users could utilize an 802.11 phone to make very inexpensive long distant phone calls through the Internet.

### **Key Components**

In addition to a WLAN backbone consisting of access points and a distribution system, a key component for implementing voice over WLANs is a telephone equipped with an 802.11 radio. Software is another component required so that it can turn a pocket PC-based personal digital assistant (PDA) into a telephone. This can be a less expensive alternative if PDAs are already there on site (Geier, 2004).

Most voice over WLAN systems also require a gateway or access point enhanced to handle special bandwidth control requirements of voice traffic. For example, gateway allows the phones to work with call manager IP phone. The gateways will add cost to the system if plan to effectively support voice traffic is required. Despite these costs, a 802.11 based phone system is generally less expensive to install and support than a wired system. Of course one needs to pay specific attention to the potential problems from WLANs, such as radio frequency (RF) interference and denial of service attacks.

### **Performance Requirements**

Performance should be the primary consideration when installing a WLAN system that supports voice. 802.11b is only capable of running three uncompressed audio streams smoothly. As a result, it is important that the system compress the audio signals before transmission in order to increase the number of supported audio streams. Another way to increase the number of audio streams is to utilize a higher performing standard, such as 802.11a for the WLAN backbone. 802.11a has the capacity to handle approximately four times as much voice traffic as 802.11b. The problem with 802.11a is that other non-voice users of the network may only be equipped with 802.11b. As a result, consider installing access points that include both 802.11a (for voice users) and 802.11b (for data users).

A WLAN can support voice implemented with high performance and QoS in mind. 802.11e adds QoS features and multimedia support to the existing 802.11b and 802.11a wireless standards, while maintaining full backward compatibility with these standards. It will prioritize traffic on the network, making data give way to voice packets ([grouper.ieee.org](http://grouper.ieee.org)). Until 802.11e is available, WLANs need to deploy proprietary QoS mechanisms to enable effective blending of voice and data. It is a problem when trying to support voice traffic over a public WLAN. It is not practical for public hotspot operators to mandate the use of a particular QoS method because of the corresponding requirement of a common vendor radio card or specific software for each client.

### **VoIP FOR OPEN & DISTANCE LEARNING INSTITUTIONS/LEARNERS**

Originally regarded as a novelty, VoIP technology is attracting more and more users worldwide because of the benefits it offer to the enterprise, service provider (Learner support Centres), and ultimately, the customer/Learners. VoIP has been proved to be an innovative mode of communications for students, for they are now enabled to join a lecture via cell phone because the IP network allows links to many video conferencing sites and audio conferencing sites. Since the voice, video and data network has its footprint covering a vast area, distance learning institutions are a major beneficiaries of this technology. It has shown to have an increase in enrolment and retention as the remote campuses can be linked to main campus via videoconferencing and thus learners can have access to instructions and tutors.

The developments in the field of Internet and web based technologies have prompted educational institutions to establish virtual classrooms and adopt e-learning

strategies for instructions and effective learning in an integrated mode for traditional teaching. It has its implications in the form of pure e-learning, fully online mode or mixed mode (blended e-learning). The virtual learning has been reported to have many benefits in terms of offering easy access to global educational resources, greater interaction through email and discussion forums, time and place independent delivery of course material and multimedia learning materials, interaction with other students, faculty and guest experts from all over the world (Starr, 1997; Mason, 1998; Mitra, 1999; Berg, Collins, and Dougherty, 2000; Mayes, 2000; Weller, 2000; Bates, 2001; Rossen, and Hartley, 2001; Laurillard, 2002).

Studies also report a positive response from students in finding virtual environment as very intuitive for working and learning. Coffman (2001) reported the benefits of VoIP for library services and reported a simultaneous talk-back and forth between the librarian and user. The provision of two-way voice (VoIP); integrated instant text messaging; and Interactive Whiteboard offer greater potentiality of enhanced learning in such synchronous teaching settings. Institutions such as Mount Saint Vincent University, Canada, University of Massachusetts, United States Military Academy at West Point (NY), University of Arkansas, Seton Hall University (NJ), Southeast Kentucky Community and Technical College, New York University, Oregon Health & Science University, and Ohio State University have adopted variations of VoIP. Some of the benefits of VoIP for Open and Distance Learning institutions/Learners are:

#### **Cost saving**

Learners can bypass long distance carriers and their per minute uses rates and run their voice traffic over the Internet for a flag monthly Internet access fee. IP networks can be significantly less expensive to operate and maintain. The simplified network infrastructure of an Internet telephony solution cuts cost by connecting IP phones over the LAN wiring system and eliminates the need for dual cabling. By using the extra bandwidth in WAN for IP telephony leverage the untapped capability of existing data infrastructure to maximize the return on Open Distance Learners Institution's current network investment. Shaw University, USA, has linked its remote campus sites through Internet Protocol (IP) Telephony communications and has observed reduction in expenses and improvement in communications.

#### **Decrease in travel time**

Since the remote sites can be connected to main site easily, students and administrative staff does not need to travel long distances. This has its impact on reduction in travel time, travel costs, lesser staff requirements, and increase in productivity.

#### **Portability and flexibility**

Learners are no longer confining by geographic location. IP telephones work anywhere on the network. Even over a remote connection, services can be extended to remote side over cost effective IP link.

#### **Simplicity and consistency**

A common approach to deployment can allow cost saving with the use of common management tools resource directories and a consistent approach to network security.

The ability to network existing PBX using IP can bring new values to the enterprise for experiencing the ability to consolidate voice mail on to single system, or a fewer system make it easier for voice mail users to network.

### **Ubiquity**

Internet telephony is supported over a wide variety of transport technologies. A Learner/Mentor can gain just about any business system, whether it is through an analog line, a DSL line, a LAN, Frame relay, ATM, (a synchronous transport), sonnet or wireless.

### **Operational agility**

Open and Distance Learning Institutions can add new services and learners to the network with fewer burdens on existing system. This can pave path for more revenue earning possibilities by providing video and audio conferencing services to the interested parties for training classes or meetings.

The ODL Institutions that have already established a MAN and WAN for data transport, adding voice service does not require to have additional expenditure. Packet infrastructure is cheaper than switching infrastructure and offer better granularity and flexibility than circuit switches. Moreover, a distributed switching infrastructure can provide modularity.

### **Transport cost**

Since ODL Institutions can add voice services over the same network, the cost of data transport can increase in the beginning. But with economic of scale due to business volumes, the cost will reduce substantially. ODL Institutions can use many ways to control bandwidth cost. It can use compression technique to increase capacity, billing can be based on uses and the flexibility of IP can be used to get efficient bandwidth prioritization

Operation cost will not increase very much because skill sets required to manage VoIP infrastructure are common and specialized VoIP personal in the organization are not required.

## **CONCLUSION**

The developments in the field of educational technology over the past two decades have changed the very face of teaching and learning in the classrooms. Radical improvements in the tools and technology of ICT has also transformed distance education starting with traditional correspondence print based to now e-learning and m-learning based (Taylor, 2001 and Garrison, 2004). The IP based networks allow greater bandwidth at lower costs and thus enables learners to access their teacher or class sitting at a remote place, even at a distance of miles away. VoIP empowers a single network to provide services of phone, fax, internet, data and video. The distance learner can listen to lectures live and in real time. They can also participate in live or through threaded email discussions taking advantage of built-in web-courseware tools. Even if there is no access to web, a learner can attend to a class via telephone or a pocket radio. VoIP can significantly affect learner support services through interoperability, scalability and reliability. VoIP can also deliver news, information, research material, music or movies on demand to a location. It is becoming very popular in university sector for mass delivery of course material. Recent developments are in the field of broadband is the emergence of Video-over-IP, which enables institutions to deliver voice, data and video signal together over

network. It is the technology of the future to serve distance education; still this technology is not free from flaws and present bandwidth challenges.

## REFERENCES

Barden, B.; Zhang, L.; Berson, S.; Herzog, S.; & Jamin, S. (1997). *Resource reservation Protocol (RSVP) version 1: Functional Specification, RFC 2205*, Internet Engineering Task Force, October, 1997.

Bates, T. (2001). *National strategies for e-learning in post-secondary education and training*, Paris: UNESCO, IIEP

Berg, Z., Collins, M. and Dougherty, K. (2000). Design guidelines for web-based courses, in Abby, B. (ed) *Instructional and cognitive impacts of web-based education*, (pp. 32-40), Hershey: IDEA Group

Bruno, S. (2001). Fixed mobile internet convergence (FMIC). *Telematics and Informatics IS (2001)*, pp. 135-141.

Coffman, S. (2001). Distance education and virtual reference: Where are we headed, *Computers in Libraries*, 21 (4), April 2001, pp. 20-25.

Donovan, S. & Cannon, M. (1998). *A Functional Description of SIP-PSTN Gateway*, Internet Draft, IETF, November.

Garrison D. (2004). "E-Learning in the 21<sup>st</sup> Century", London: RoutledgeFalmer

Geier J. (2004). *Voice over wireless LAN*, wi-fi Planet.

Gupta, S. D. (2003). VoIP: Packaged voice and more, *Network magazine*, pp. 54-58.

Hamdi, M.; Verscheure, O.; and Hubaux, J.-P. (2003). Voice service interworking for PSTN and IP networks. *IEEE Communication Magazine* 5, pp. 104-111.

Handley, M.; Schulzrinne, H.; Schooler, E.; & Rosenberg, J. (2001). *SIP: Session Initiation Protocol*, Internet Draft, IET F, January.

[http://grouper.ieee.org/groups/802/11/Reports/tge\\_update.htm](http://grouper.ieee.org/groups/802/11/Reports/tge_update.htm)

[http://qos.ittc.ukans.edu/ipqos/ip\\_qos.htm](http://qos.ittc.ukans.edu/ipqos/ip_qos.htm)

*ITU-T Recommendation H.323, Systems for multimedia communication in packet mode*, 1998.

Laurillard, D. (2002). *Rethinking University Teaching: a conversational framework for the effective use of learning technologies*, London: Routledge

Lipset V. (2004). *VoIP Synergy: Voice to drive wireless adoption*, wi-fi Planet.

Mason, R. (1998). Models of on-line courses. *ALN Magazine*, 2 (2) Retrieved March 24, 2006, from <http://www.aln.org/publications/magazine/v2n2/mason.asp>

Mayes, T. (2000). Pedagogy, Lifelong Learning and ICT, Retrieved on March 24, 2006 from <http://www.ipm.ucl.ac.be/ChaireIBM/Mayes.pdf>

Minoli, D.; & Minoli, E. (1998). *Delivering Voice over IP Networks*. New York: Wiley.

Mitra, S. (1999). Virtual institutions in the Indian subcontinent, (pp. 125-134), in Farrell, G. (ed) *The Development of Virtual Education: A Global perspective*, Vancouver: The Commonwealth of Learning

Mortada, I. & Probst, W. (2001). Internet telephony signaling. *Telematics and Informatics 18 (2001)*, pp. 159-194.

Rossen, E. and Hartley, D. (2001). Basics of E-Learning, Info-Line, 109, September 2001.

Starr, R. M. (1997). Delivery instruction on the World Wide Web: Overview and basic design principles, *Educational Technology*, 37 (3), 7-15

Weller, M.J. (2000). Creating a large-scale, third generation distance education course, *Open Learning*, 15 (3), 243-251.

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