VOICE OVER INTERNET PROTOCOL ADOPTION IN RUKARA COLLEGE OF EDUCATION, EASTERN PROVINCE, KAYONZA DISTRICT, RWANDA.

A RESEARCH THESIS

PRESENTED TO THE

COLLEGE OF HIGHER DEGREES AND RESEARCH

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IN PARTIAL FULFILLMENT OF THE REQUIREMENTS FOR THE DEGREE OF

MASTER OF SCIENCE IN INFORMATION SYSTEMS

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October, 2012

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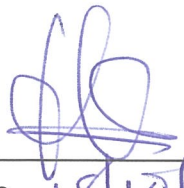

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DECLARATION B

I confirm that the work reported in this thesis was carried out by the candidate under our supervision.

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APPROVAL SHEET

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ABSTRACT

The purpose of this study was to evaluate the adoption of VoIP in Rukara College at a greater rate and at the same time find out the reasons for the slower adoption of VoIP. To determine the level of performance and efficiency of the existing technology in Rukara College, Rwanda, to determine the level of performance and efficiency of VoIP in Rukara college, Rwanda, to determine if there is significance difference between the existing technology and the proposed one. There study employed descriptive research design as well as quantitative and qualitative research designs were used to describe data collected from the field. Research Population of about 2140, 140 staff members and 2000 students. For staff members, this included both teaching and non teaching staff, The researcher Used the sloven's formula to compute the sample size, were by 42 staff members were selected as a sample from 70 staff members while 119 students as a sample were selected from 200 students, Sampling Procedure, Random sampling procedure was used since all members had equal chances and the Research Instrument and the tools used for collecting data from the respondents were personal questionnaires which were designed touching on every objective of the study. The conclusion was based on research objectives, analysis and interpretation of individual views on all problems associated with PSTN that are categorically summarized in percentages like, PSTN signals not enough where more than 70% said that PSTN signals were not good enough, on Inter-communication inefficient more that 80% said was so inefficient, also more than 80% said that the monthly charges make it expensive due to monthly revenues collected from the collage. Regulatory authorities since the authorities are state agencies incorporated VoIP with PSTN and regulated both of them as a way of giving a chance to institutions bit academic or business enterprises to asses, adopt and utilize the benefits a better technology since it is the state that gets revenue.

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LIST OF ACRONYMS

VOIP:	VOICE OVER INTERNET PROTOCOL
PSTN:	PUBLIC SWITCHED TELEPHONE NETWORKS.
PSDN:	PACKET SWITCHED DATA NETWORKS. .
IP:	INTERNET PROTOCOL.
ARPANET:	ADVANCED RESEARCH PROJECTS AGENCY NETWORK.
TDM:	TIME-DIVISION MULTIPLEXING.
SPSS:	STASTISTICAL PACKAGE FOR SOCIAL SCIENTISTS
KBPS:	KILOBITS PER SECOND
PBX:	PRIVATE BRANCH EXCHANGE
ATA:	ANALOGY TELEPHONE ADAPTOR
SIP:	SESSION INITIATION PROTOCOL
NAT:	NETWORK ADDRESS TRANSLATION
PCM:	PULSE CODE MODULATION
LAN:	LOCAL AREA NETWORK

CHAPTER ONE
PROBLEM AND ITS SCOPE

Background of the study

In simple terms, VoIP stands for voice over internet protocol and if you remove protocol from this abbreviation, it is very simple to understand, it says voice over internet which means this application provides verbal communication over internet just like common cell phones and landline phones but it is getting very popular because of low cost bills and free of cost calls (Giggs 2005).

Around thirty years ago, Internet did not exist and all communication was conducted through telephone systems via the PSTN (public-switched telephone network) but now technological advancement has given birth to new gadget and technologies which are quite useful to all of us and made us think to create new communication technologies for examples personal computers, mobile phones, and finally the Internet (Gideon, 2003).

Today, we are seeing a big revolution in the field of communication where everyone is using computers and internet at the workplace and after business hours when there is free time, people communicate with others to exchange personal and professional data and sometimes to talk with more people at a time using applications like IP phone (Internet Phone) or NetMeeting and Skype this is due to the advent of new communication technologies, we have started using real-time communication via PC and VoIP (Burn, 2005).

In the study of Voice over Internet Protocol, its global evolution and its widespread acceptance the momentum at which it is rising in the discrete world has in fact modeled and modified everyone's life. This is the reason why the interest and demand on different applications has been increased. The rise in demand has produced many new applications. Voice over Internet Protocol (VoIP) technology has become a

potential alternative to and supplement of the traditional telephone systems over the Public Switched telephone Network (PSTN), providing a versatile, flexible and cost-effective solution to speech communications. Internet telephone is a revolutionary technology that has the potential to completely rework the world's phone systems. And this has worked largely in the nest technological advancements (Sandvine, 2001).

The latest emergence of VoIP along with lowering of digital signal processors or DSP's which serve as key factors in the decompression and compression of voice are elements which have made possible the launch of this new technology. But in some countries like USA and Europe have already adopted and are fully enjoying the benefits of this technology. In summer 2005, Advanced Research Projects Agency Network (ARPANET), a provider of IT consulting and software solutions conducted a web based survey to assess the VoIP market and its growth potential in USA. The key findings were as follows:

The adoption of VoIP networks increased flexibility to (80%), simplified the network infrastructure to (55%), enabled new feature rich and integrated applications (52%), and lowering voice toll costs outside the enterprise (62%) (ARPANET, 2005).

The significant barriers to VoIP adoption is lack of experienced staff for high products and tools (37%). In developing countries, VoIP adoption is on small scale (less adopted) including east Africa, Rwanda in particular and therefore academic institutions and Rukara College of education inclusive. Regulators in different countries have different approaches to VoIP adoption VoIP is an important technology that has the potential to transform telephone line for communication. In countries like Rwanda the adoption of VoIP would facilitate and accelerate academic institutions and Rukara College of education inclusive pace for development due to VoIP service and features that save great deal money results from regional and international call bills (ARPANET, 2005).

VoIP is the transmission of voice signals from one party to another party digitally i.e. usage of packet switched data network (PSDN). The basic steps involved in VoIP are conversion of the analog voice signal to digital format and compression/translation of the signal into internet protocol (IP) packets for transmission over the internet, the process is reversed at the receiving end. Adoption is a process of changing to suit a new situation or assimilating to the new policy, in this case the study is finding out how Rukara College of education adapted to VoIP as a new situation or policy.

Statement of the Problem

The largest incentives to replacing your traditional phone line with a VoIP line is of course the reduction in price, signal strength and quality of service. A VoIP phone line can save you a great deal of money compared to a traditional phone service i.e. PSTN especially on long distance and international calls due to its additional new features. On the other hand PSTN which uses circuit switching (end-to-end) technology has dynamically allocated dedicated bandwidth. The bandwidth size and call distance determine the monthly bills. VoIP in spite of its advantages and features over PSTN has not been well embraced in developing countries and particularly so in academic institutions including Rukara college of education (Sharma, 2003).

Purpose of the study

The purpose of this study was:

1. To answer the research questions.
2. To validate the theory of innovation stated by Davis, 1993.
3. To bridge the gaps left by previous studies.
4. To generate new knowledge.

General objective

The general objective of this study was to evaluate the adoption of VoIP in academic institutions at a greater rate and at the same time find out the reasons for the

slower adoption of VoIP Technology is most important for consideration and how those reasons or problems can be rectified.

Specific objectives

1. To determine the demographic characteristics of the following categories of respondents:

   A. Academic Staff and administrators: age and highest academic qualifications
   B. Students: age and highest academic qualifications

2. To determine the level of performance and efficiency of the existing technology in Rukara College, Rwanda
3. To determine the level of performance and efficiency of VoIP in Rukara college, Rwanda.
4. To determine if there is significance difference between the existing technology and the proposed one.

Research questions

1. What are the demographic characteristics of the following categories of respondents?
   a) Academic Staff and administrators: age and highest academic qualifications
   b) Students: age and highest academic qualifications

2) Does the existing technology have problems with performance and efficiency?
3) Will the proposed technology provide optimal performance and efficiency?
4) How can I design a framework for VoIP adoption?

Null Hypothesis

There is no significant difference between the existing technology of PSTN and the proposed one of VoIP.

from the variables identified in the study and the last was theoretical scope which had a theory that guided this study.

Time Scope

The study covered seven month from January 2012 and ended in July 2012; the researcher accomplished and submitted the project as required by college of higher degrees and research of Kampala International University.

Geographical Scope

The research was limited to Rukara College of Education and data collected from the department of computing only, the college is in Rwanda its is located in eastern province, Kayonza District at one and half kilometers from the main road Kagitumba to Kigali high way.

Content Scope

This study concentrated on evaluation of VoIP adoption in academic institutions and Rukara College in particular, through assessing the effectiveness of the old communication technology from different respondents and the possibility for adoption of VoIP technology.

Theoretical scope

This study was guided by innovation Theory derived from Technology Acceptance Model; where the decision to adopt to a new technology was by the concept of primary and secondary adoption. Primary adoption is the organizational decision to adopt and acceptance the innovation, while secondary adoption is the individual's decision inside the organization adopt and acceptance the innovation (DAVIS, 1993).

Significance of the Study

This study is of benefit to the academic institutions, for future researchers and the researcher himself specifically in the field of VoIP communication technologies.

1) The study helped the academic institution to identify problems of the old communication technology and clearly showed the benefits of VoIP such as A VoIP phone line can save you a great deal of money compared to a traditional phone service especially on long distance and international calls and even fax, monthly bills due to dedicated bandwidth, i.e. PSTN. VoIP's feature of videos conferencing saves time and money wasted on travels for conferences from different parts of the world.
2) This study is of benefit to Readers or future researchers as secondary and related studies
3) The study has widened the researcher's knowledge on VoIP and other telecommunication technologies like PSTN and also enables the researcher to attain the Masters Degree in information systems.

Operational Definitions of Key Terms

VOIP: voice over internet protocol (VoIP is the transmission of voice signals from one party to another using internet as the media of communication

EVALUATION: Is a systematic determination of a subject's merit, worth and significance, using criteria governed by a set of standards. It can assist an organization to ascertain the degree of achievement or value in regards to the aim and objectives of an undertaken project.

ADOPTION: Is a process whereby a person assumes the parenting for another and, in so doing, permanently transfers all rights and responsibilities from the original parent or parents

ACADEMIC INSTITUTIONS: Higher education institution means a university, college, or campus, or any other autonomous institution that

offers a level of academic education and professional training that leads to full academic and or professional qualifications and competence.

CHAPTER TWO

REVIEW OF RELATED LITERATURE

Overview

This chapter focuses more detailed on VoIP technology, that is Internet connection and its bandwidth requirements for VoIP services, different types of VoIP in use today, different devices used and how they are used in overall working of VoIP technology and different protocols governing the VoIP technology will also be discussed. Finally present existing literature concerning the research area.

Concepts, Opinions, Ideas from Authors/ Experts

Internet telephone

Internet telephone refers to communication services, voice and voice message applications that are transported via the Internet rather than that public switched telephone network (PSTN). The basic steps involved in originating an internet telephone call are conversion of the analog voice signal to digital format and compression/translation of the signal into internet protocol (IP) packets for transmission over the internet, the process is reversed at the receiving end (Bradley, 2003).

Internet telephone is at times synonymous to VoIP. In other words Voice over Internet Protocol is the transfer of voice conversation as data over an IP network. Unlike traditional circuit switched calls on the PSTN. In VoIP, the telephone connection is packet-switched; with a packet-switched network multiple computer devices share a single data network and communicate by sending data packets to one another each packet containing addressing information that specifies the source and destination computer (Gideon, 2003).

Current trends

Organizations around the world seek to reduce rising communications costs. The consolidation of separate voice, fax, and data resources offers an opportunity for significant savings accordingly is done by VoIP technology. This is due to a number of factors including codec.

This stands for compressor/decompress or coder/decoder and is the hardware or software that samples analog sound convert it to digital bits which outputs at a predetermine data rate. The codec performs compression as well, to save bandwidth.
The challenge of integrating voice, fax, and data is becoming a rising priority for many network managers. Organizations are pursuing solutions that will enable them to take advantage of excess capacity on broadband networks for voice, fax, and data transmission, rather than PSTN.

There exists a unique perspective on the VoIP development in which the focus lies on the change in communication technology standards that organizations use and the development of education criteria based on new technology standards. In the past traditional telephone and standards/education based on this PSTN proved to be dominant. As internet and IP technology keep on evolving, the focus moved forward. This was also the case for the development of new communication standards and communication technology education criteria. Experts mention that currently we are in a transitional phase where traditional communication systems are being replaced by VoIP based ones. This transition did not happen overnight and there are still multiple hurdles to overcome. For the time being the old and the new communication technology will still have to function side by side until complete market dominance by IP based communication solutions occurs. (Bradley, 2003).

The situation where VoIP is rapidly penetrating the business market the projected PBX shipments for 2003 up to 2008 were depicted. A hybrid solution means that both traditional and VoIP systems are used (temporarily). TDM represents the traditional phone systems in the figure and pure IP represents a complete VoIP implementation where no legacy systems are active. The decreasing line for legacy systems and the increasing ones for hybrid and pure IP solutions show that VoIP adoption was increasing and that was likely to continue for some time (Sacker, 2006).

Difference between PSTN and VoIP

PSTN

- Typical business line rental $40
- Extra features available at extra cost from Telecommunication companies.
- Dedicated lines required from Telco Provider thus expensive.
- Excellent audio quality.
- Tele workers solutions are very costly and require additional dedicated lines
- To maintain a PSTN PBX can be very costly and require specific technical expertise
- Upgrading and extending a PSTN PBX is costly and a complex process, and requires specific expertise
- Because of a limited number of Telecommunication in every country, least cost routing is very limited
- No free calls possible
- Installation of PSTN PBX requires extra wiring, expensive proprietary hardware and telecom expertise
- Typically hard wired, remains on during power cuts
- Suitable for emergency calls and location can usually be traced
- Little or no integration at all with computer applications such as email clients
- Only audio can be transferred during a call

VoIP

- Typical business line rental $14
- Quality depends on internet bandwidth availability
- All calls can be transferred over the internet connection
- Extra features available for free or at a very low cost

• Remote extensions are standard features with VoIP PBX

• As long as you can use a computer, you can maintain a VoIP PBX network

• Upgrading and extending a VoIP PBX requires only an internet connection and software upgrades

• Thousands of VoIP providers in different countries to choose from, thus providing a cheap solution

• Possible to call other VoIP users using the same gateway for free

• VoIP PBX uses same computer networking cables, thus do not require additional hardware or specific skills

•Audio, video and data (such as presentations) can be transferred during a call • Not suitable for emergency calls. Location can be difficult to be traced

•Full integration with computer applications, such as email clients **(Bradley, 2003).**

VoIP and Competitive advantage

There are various factors influencing the adoption of VoIP by the business community. VoIP functions as a digitalization of communication data that can easily interact with the Internet and be transported over it. This allows the communication data to be stored, manipulated and combined with other data that can then be transported to any device that has a connection to the Internet. It is believed that VoIP will serve as a unifying platform for various business applications. In essence it is predicted that VoIP will support new communications functions that are not yet present today (Werbach, 2005).

There are aspects of VoIP that can incite organizations to move towards the new technology. VoIP provides some new features like Unified Communications which is the integration of real-time communications such as instant messaging, IP telephone and call conferencing with non real-time communications such as SMS messaging and e-mail. In the future organizations seek to broaden the functionality of the VoIP phone system by using all types of custom applications that can run on the IP phones and can be specifically designed to serve to the needs of the organization (Sacker, 2006).

According to the pilot program that was conducted by the Intel Corporation, there was a significant increase in the productivity and performance of employees that were using the VoIP solution in comparison to those that still utilized traditional phone solutions. For example, repetitive tasks, such as accessing voicemail, are completed faster when using a VoIP solution. The performance increase provided by the new technology results in a net productivity gain which translates back into cost savings for the organization (Sacker, 2006).

Toll bypass can be used by organizations in order to lower their communication costs, especially when multiple branch offices are involved. Normally or generations lease communication lines (Tie lines) between two offices in order to communicate directly. This option can be expensive, as these tie lines are required for every separate branch office in order for (secure) communications to occur. Another disadvantage of this option is that payment of the lease amount is required regardless of the level of usage of the communication line (Vetteth, 2009).

With toll bypass no extra communication costs are necessary as the organization's own data network is used for transportation which would allow the organization to acquire a certain level of (competitive) advantage over a competitor in terms of costs. Concretely VoIP provides a more cost effective alternative to traditional telephone and thus organizations that have implemented it enjoy cheaper communication and operational costs in the long run (Vetteth, 2009).

The extent that the previously mentioned benefits influence the core business of various organizations depends heavily on the nature of the core business processes. An organization in which the core business processes are heavily intertwined with communication technology, such as a call center or a technical support company suffers a greater impact from VoIP than for instance a construction company. An explanation could be that phone based communication is one of the principal business processes in a

call center and VoIP provides a cheaper and more efficient way to perform it. This enables the organization to gain an advantage over its competitors which still use traditional telephone. For a construction company the impact of VoIP can be of a lesser extent as phone based communication is not normally seen as a core business process but falls more into the supporting process category (Werbach, 2005).

Nevertheless VoIP can provide efficiency gains and added functionality that directly and indirectly add value to organization and its processes. Ultimately this allows it to acquire a competitive advantage over its rivals (Werbach, 2005).

Cost benefit analysis

As is the case with new projects that require a significant investment in terms of resources and manpower, there is always the question of how to accurately select an option that will lead to the most value for the organization. The term value is used loosely in this context as its definition can vary significantly depending in the individual or organization in question. Organizations can decide to generally see value as a greater level of payoff in terms of financial resources or as an improvement that affects their business processes in some way (Sacker, 2006).

Selecting the option (e.g. project) which will perceivably lead to the most value for the organization is not the only requirement in the selection process as the level of value gained needs to be relative with the amount of costs incurred. It would be a bad managerial decision to purely focus on the level of value gained. As an example, project one might have a high payoff with a larger amount of costs while project two has a slightly lower payoff with substantially lower costs.

In this case the choice should be made to go with project two despite its lower payoff because the gain for the organization is relatively larger when the costs are also

considered. The entire selection process for choosing the option with the most value for the organization while taking the costs into account is referred to as a cost benefit analysis (Michael, 2002).

Securing VoIP

Like many new technologies that are utilized in the transfer of possibly sensitive information, security remains an important issue. The security risks to the VoIP technology are therefore also frequently scrutinized. Despite the numerous benefits that can be achieved with VoIP, it appears that its security still remains a major deterrent for organizations to implement the technology (Michael, 2002).

The threats occur in the form of people gaining unauthorized access to the enterprise's PBX or the voice mail systems and thus acquiring free telephone calls that the organization has to pay for later (Jonathan, 2000).

This can even occur because the perpetrator has a strong desire for revenge, blackmail or greed. With the Internet, users have to constantly monitor which websites they visit, what is downloaded and how to secure their systems or else they could be the victim of a number of threats e.g. viruses, worms, malware etc. To a certain degree this is also true for VoIP since it makes use of the same protocol as the Internet, namely the IP protocol, making it susceptible to threats (Jonathan, 2000).

Such a threat can be Spam over Internet telephone, which disrupts the day to day workings of organization employees by making telephones ring as many times as one would receive a spam e-mail. This involves an employee receiving several unwanted phone calls that interrupt his tasks (Collier, 2007).

Internet Connections and its types

ADSL connections

In the recent times ADSL (Asymmetric Digital Subscribers Line) connections show

that they have heavily reached the market and can provide an excellent Internet connection. The connections work by splitting the phone line into two separate channels, one for data (internet) and one for voice (phone calls) (Vincent 2003).

Cable connections

Cable connections are considered one of the best types of internet connection available to the home use they offer very fast and reliable connections with a fixed monthly fee (Davidson, 2000).

Davidson (2000) again assert that 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. Most connections average about 400kb/second download and 128kb upload. Cable's largest advantage is its availability and ability to produce multiple up streams (when sending).

Satellite (HST)

This is the most expensive alternative for getting a high-speed connection to the Internet (Gideon, 2005), noted that if a 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 modem to connect for uploads. A 2-way connection, however, both sends and receives and is telephone-free. Average speeds for this type of connection are 600K and higher for download and 128k for the upload. Averages tend to be higher because there are far fewer users on the network to slow down connections.

Similarly Foster (1998) in their research noted 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, assume that user will need at least 88 kbps to use VOIP reliably. Note that VOIP service Providers will provide with their broadband requirements.

Gideon (2005) prettily states that users will need an Internet connection that can handle at least 128kbps 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. If users wish to use conference calls even more bandwidth will be required. While there is no set of rule for the amount of bandwidth that a household will need, count on 128kbps as a minimum.

If users have several computers and other devices connected to the Internet, consider going up to the maximum available. Note that these speed are for uploads the download speeds are usually quite a bit faster and present no problem for VOIP. The biggest attraction that VoIP technology offers is the cost benefit. Even some studies show that with VoIP call, a user has to pay 10 times less than a normal long distance call (Maxwell, 2000).

Voice over Internet Protocol (VoIP) is a technology that allows us to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. Some services using VoIP may only allow you to call other people using the same service, but others may allow you to call anyone who has a telephone number – including local, long distance, mobile, and international numbers. Also, while some services only work over your computer or a special VoIP phone, other services allow you to use a traditional phone through an adaptor (Sandvine, 2001)

Different forms of VoIP

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

Analog Telephone Adapter

As noted by Sharma, (2003), this is the simplest and most common form of VoIP. The analog telephone adaptor (ATA) allows a user to connect a standard phone to the computer or internet connection for use with VoIP. The ATA is an analog-to-digital converter which takes the analog signal from the traditional phone and converts it into digital data for transmission over the Internet.

Skype

Skype is a peer-to-peer VoIP client system developed by KaZaa that allows its users to make voice calls and send messages to other users of Skype clients. It encrypts calls end-to-end, and stores user information in a decentralized manner.
Skype is a proprietary peer-to-peer internet telephony (VoIP) network which competes against established open VoIP protocols like SIP or H.323. The system has a reputation for its broad range of features and its ability to use P2P technology to overcome common firewall and NAT problems. It does this by routing voice packets through firewalls or NAT's with no special configuration. Skype users can speak to other Skype users; use Skype Out to call landlines and mobile phones, use Skype In to receive calls from any phone, and also receive voicemail messages (Sandvine, 2001)

The main difference between Skype and other VoIP clients is that it operates on a peer-to-peer model rather than the traditional server-client model. The Skype user directory is entirely decentralized and distributed among the nodes in the network, which means the network, can scale very easily to large sizes (over forty million users) without a complex and costly centralized infrastructure (Bradley, 2003).

Skype is a pretty new application from which allows a computer user to make voice calls over the internet. It cuts costs in three ways: (1) bypassing the Public Switched Telephone Network that charges by time and distance in favor of ISP networks that charge flat monthly fees, (2) using resources only when conversation

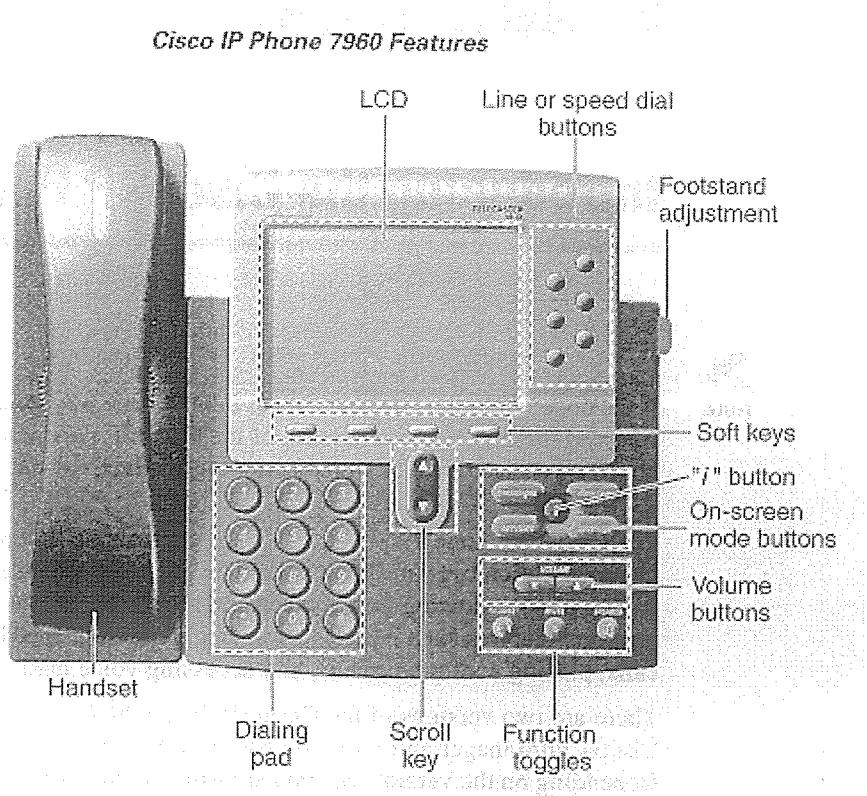
takes place (as opposed to keeping the line open even when the conversation falls silent) and even permitting more calls to take place simultaneously. (3) Enabling a single network to carry both data and voice, thus reducing equipment and management requirements.

It supports new types of communications by combining Internet applications and web-sites with human voices. That means people can collaborate online more easily, whether they are working together or remotely.

IP and conference Phones:

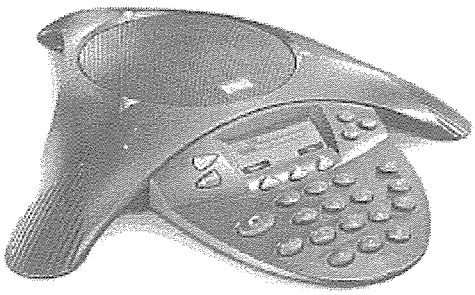
IP devices look just like normal phones with a handset, cradle and buttons. But instead of having the standard RJ-11 phone connectors, this type of phones have an RJ-45 Ethernet connector. They connect directly to the router and have all the hardware and software necessary right onboard to handle the IP call. Figure 2.1 illustrates how IP phone conference Phones look like (Vincent, 2003).

Figure 2. Illustrates IP Phones



Adapted by (Vincent, 2003).

Figure 3. Illustrates Cisco IP Conference Station



The Cisco IP Conference voice instrument is a full-featured, IP-based, full-duplex hands-free conference station for use on desktops and offices, and in small to medium-sized conference rooms. (Vincent, 2003).

The full-duplex design offers superior voice quality, eliminating echoes, clipped words, and reverberations for more natural conversation. It features superior sound quality with a digitally tuned speaker and three microphones, allowing conference participants to move around while speaking. In addition to the regular telephone keypad, the Cisco 7935 provides three soft keys and menu navigation keys that guide a user through call features and functions. The Cisco 7935 also features a pixel-based LCD display. The display provides features such as date and time, calling party name, calling party number, digits dialed, and feature and line status. The Cisco IP Phone is a full-featured, multiline telephone that provides voice communication over an IP network. This phone functions as a managerial-level office telephone, replacing traditional analog telephones in an IP telephone network. The Cisco IP Phone 7960 accommodates six available lines or speed dial buttons.

IP enabled PBX

This is a frequently selected option for adoption of VoIP as a solution in an organization. It involves making an upgrade of PBX in order to support the transport of digital data and it does not require any of the current cables and handsets used in PSTN to be replaced. The PBX needs to be updated in order to be able to function with the IP protocol and connect to the internet. This is the only piece of equipment that needs to be altered. Any telephone that is connected to the PBX will appear as an IP telephone to the outside world. Many IP telephone suppliers provide cost effective upgrade packages for PBX's allowing them to handle IP based traffic. There are also pure IP based PBX's in existence that do not support analog telephony but focus completely on digital telephony. Making an organization's PBX IP enabled provides an intermediate method for implementing VoIP. (Joseph, 2009).

Computer-to-computer:

The VoIP technology along this line is by far the easiest and the most basic to use

all you need is the software, a microphone, speakers, a sound card and an Internet connection, preferably a fast one like you would get through a cable or DSL modem.

Gideon, (2003) noted that the software enabled components exhibit a less complex architecture in VoIP transmissions. These three forms which have been discussed above provide an idea that these are the types which are prevailing in the market and are used by customers. Although there is the difference in capturing the data from the user but while sending this data the process is same. In all these types the captured data is packetized and sent over IP network. This was further emphasized by Leyland (2005) that, data in the packets exhibit a greater transmission ratio with relevance to standard environmental conditions.

Devices used in VoIP technology

The architecture of Internet telephone is similar to traditional telephone networks in many ways, but it also has some significant differences. Most fundamentally, Internet telephone is different from traditional telephone networks in that it naturally runs over the Internet, or more generally over IP networks.

The most significant consequence of having this underlying network is that it provides transparent connectivity between any two devices on the network. Whereas devices in traditional networks are restricted to communicating with those devices to which they are directly connected, and the telephone protocols themselves must handle all location and routing features, Internet telephone can rely on an underlying infrastructure which provides all these capabilities automatically.

The most common devices used in the Internet telephone are end systems, gateways and signaling servers.

End systems

In the preview concept, the electronic devices with which clients, these end systems respond and initiate to signaling, and receive and transmit media. These devices also maintain the track of calls and their status similar to gateways. Or users place and receive calls are clearly conceptualized by (Jolty, 2006).

Signaling servers

Holmes, (1997) noted that 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 (Bradley, 2001)

Gatekeeper

A gatekeeper is an H.323 entity on the network that provides services such as address translation and network access control for H.323 terminals, gateways, and MCUs. Also, they can provide other services such as bandwidth management, accounting, and dial plans can be centralized to provide salability. Gatekeepers are logically separated from H.323 (protocol) endpoints such as terminals and gateways. They are optional in a H.323 network, but if a gatekeeper is present endpoints must use the services it provides. Gatekeeper Zones and Subnets, a zone is the collection of H.323 nodes such as gateways and terminals registered with the gatekeeper. There can only be one active gatekeeper per zone. These zones can overlay subnets and one gatekeeper can manage gateways in one or more of these subnets.

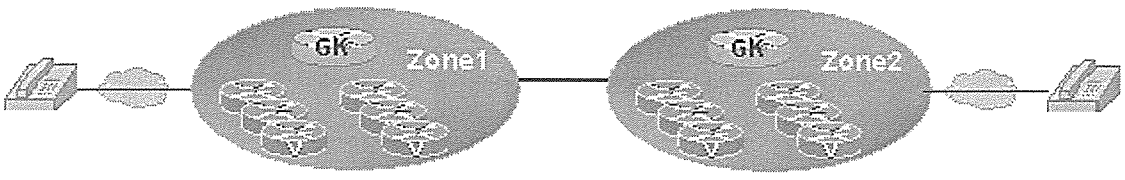


Figure 4. Illustrates gatekeeper

H.323

The H.323 standard provides a foundation for audio, video, and data communications across IP based networks. This protocol is an umbrella recommendation from the International Telecommunications Union (ITU) that sets standards for multimedia communications over Local Area Networks (LANs). It also addresses and specifies a common method for sharing video, data, and voice simultaneously using V.34 modem connections over a single analog Plain Old Telephone System (POTS) telephone line (ITUT Recommendation H.323, Packet-Based Multimedia Communications Systems. (Fred, 2005)

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 handled by establishing session parameters at both parties. And finally, the session management including termination, modification and invoking services are carried. (Fred, 2005)

How VoIP works.

VoIP works in line with the data technology transmissions, and over a network. The

above devices work in the following way in the overall working of VoIP. (Gideon, 2003)

Initially at the source the analog signals are converted into digital signals by ADC (analog to digital converter) techniques. Then there is speech compression. Traditional telephone networks use pulse code modulation (PCM) at 8K samples per second. 12-bit samples are compressed and expanded ('commanded') 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 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. After conversion of voice packets into data packets RTP (real time protocol) is used for time stamping and content identification of UDP voice packets, 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 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. (Gideon, 2003)

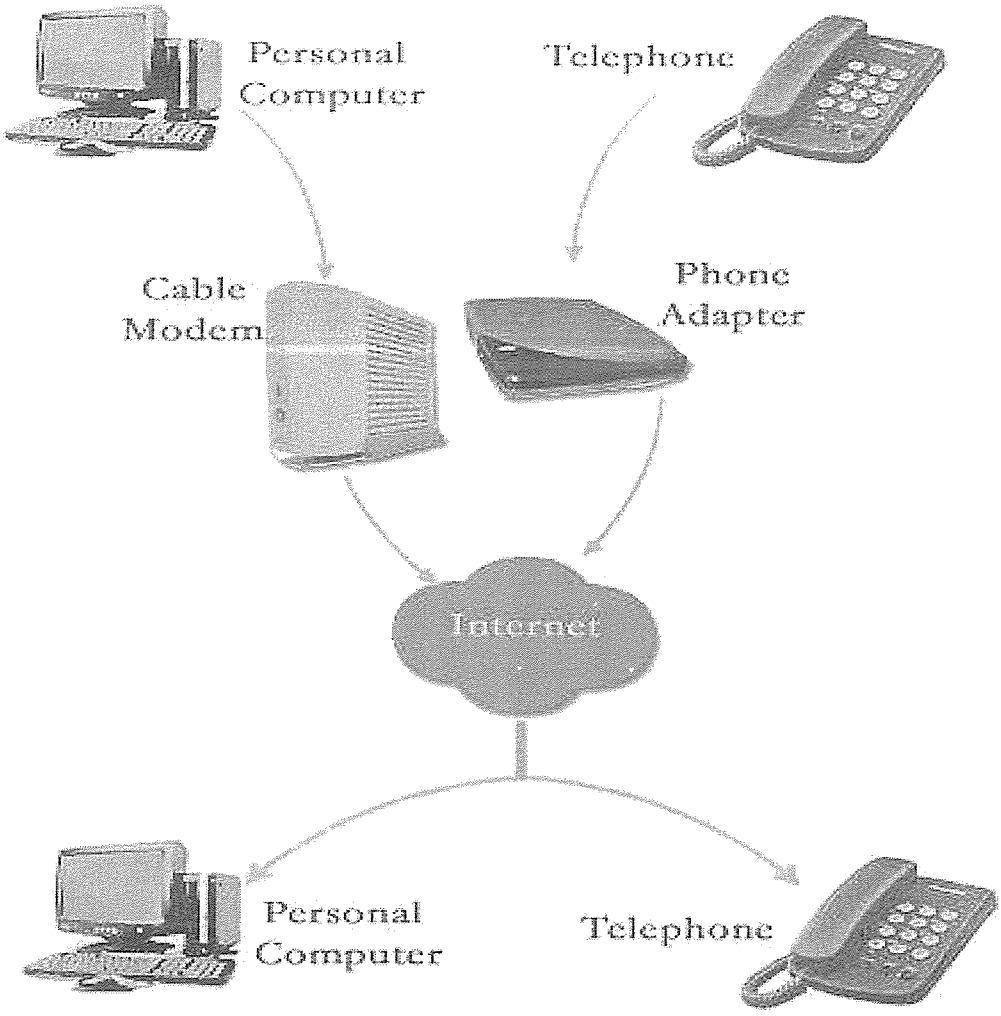
Gateways are devices that allow calls to be placed to and from other telephone networks, which are implemented between Internet and PSTN. Although gateway cannot support the same number of users as even the smallest local telephone exchange. In the case of outgoing calls VoIP phone captures the phone 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).

Finally at the receiving end, packets have to be disassembles 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 figure below shows how working of the VoIP technology, that in VoIP, calls could be of three types.

a) PC to PC: It is a call in which one PC communicates with other PC.
b) PC to Phone: In this type of communication a PC communicates with an analog phone.
c) Phone to Phone: In this type an IP phone communicates with an analog phone.

Figure 4. Illustrates Protocol Architecture of VoIP (Brawley, (2004)



As indicated by Brawley (2004) VoIP contains rules or procedures that are either way used by endpoints when they communicate on a network. For example, OSI (Open Systems Interconnection) has the seven different layers, for data interchange different protocols at each layer of this model interact. The same is the case of VoIP architecture. The telephone services of the Internet are built on a hierarchy of packet switching protocols, The function of signaling protocols include routing, reservation of resources, call acceptance, address translation, establishment of the call, as well as its management and billing (Brawley, 2004).

In Internet telephone, 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 (Giggs, 2005).

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 take some time so then TCP can take much longer time So TCP is highly unsatisfactory for fixed data transmission (Kings, 2001).

IP network can use an alternative higher layer protocol user datagram protocol UDP, which does not re-transmit the lost data and there are no acknowledgements also in case of UDP. In TCP if there are more number of hops, the acknowledgment takes longer time and in UDP no such acknowledgment so UDP competes more effectively than TCP in a congested IP network.

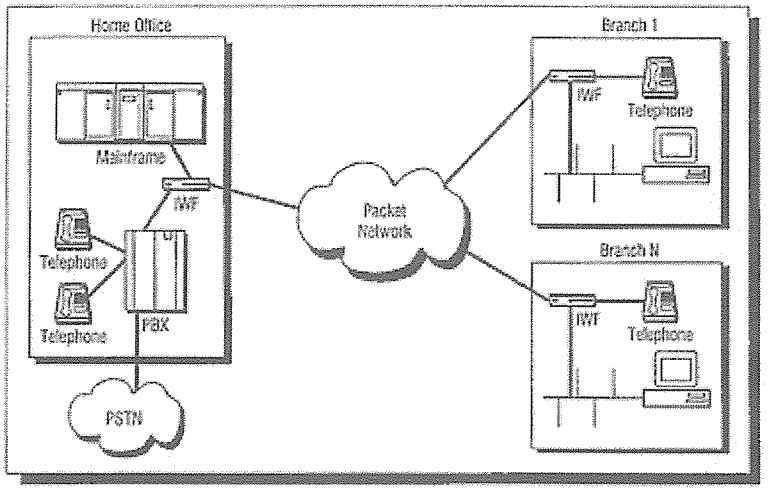
VoIP generally uses signaling protocols such as SIP (session initiation protocol) and H.323 has been developed for the translation of addresses of the application layer, and for establishing and controlling the calls (Bradley, 2003).

Organization with many branch offices (e.g. banks) that wants to reduce costs and combine traffic to provide voice and data access to the main office. This is accomplished by using a packet network to provide standard data transmission while at the same time enhancing it to carry voice traffic along with the data.
Typically, this network configuration benefits if the voice traffic is compressed as a result of the low bandwidth available for this access application. Voice over packet provides the interworking function (IWF), which is the physical implementation of the

hardware and software that allows the transmission of combined voice and data over the packet network.

The IWF must emulate the functions of both a private branch exchange (PBX) for the telephone terminals at the branches, as well as the functions of the telephone terminals for the PBX at the home office.

Figure 5 .Illustrates Branch Office Application, (Gideon, 2003)



Specifically, VoIP offers the following benefits:

Reduced Cost – VoIP offers efficient use of bandwidth and require fewer long-distance trunks between switches. The traditional circuit-switched networks, or the PSTN, have to dedicate a full-duplex 64Kbps channel for the duration of a single call. In contrast, the VoIP networks only require approximately 14Kbps, as voice compression is employed, and the bandwidth is used only when something has to be transmitted. Because the voice is delivered over the same data channels as any other data, it results in more network traffic over fewer leased lines. In addition, PSTN requires expensive tandem/toll switching capability, while using IP networks can bypass local and long-distance tolls. Therefore, VoIP is likely to be used first in places where cost savings are significant. An example might be a company with highly distributed sites worldwide connected through a private or public IP network. (Burns, 2003)

Unique Applications and Telephony Features Digitized data is easy to manipulate and analyze; the possibilities for new or enhanced applications and features are tremendous. For example, IP telephone can be integrated into interactive web pages, where a user clicks an icon to initiate a phone call. Businesses can utilize this feature to promote their products on the Internet (Valdes, 1998).

Utilizing Wireless LAN – Wireless LAN (WLAN), emerging as a mainstream technology in many organizations, are soon be upgraded to handle both voice and data traffic. While the cost of wireless long-distance services remains high, Wireless VoIP (VoIP) could provide users the option of connecting to IP networks for long-distance calls. Many leading handset providers, such as Motorola, Ericsson, Nokia, demonstrated combination wireless VoIP/cell phones. With such devices, users can make call via Wi-Fi (IEEE 802.11b) where WLAN is available, and make call via cellular in other places.

Bandwidth & Effective Bandwidth

In every network Bandwidth & Effective Bandwidth is required to be available if attempts are made to make use of it. VoIP networks have an additional issue that needs to be overcome as all traffic is conducted on a data network. Normally Bandwidth is only designed for a certain amount of data traffic. There with the addition of voice data these networks can become congested resulting in degradation of the service. This Congestion of data is also referred to as bandwidth congestion (Valdes, 1998).

The bandwidth of a VoIP network is negatively influenced by measures taken to secure the network. Furthermore the percentage of the bandwidth transporting data in comparison to the total bandwidth used is also adversely affected; the relation between bandwidth transporting data and the total bandwidth utilized is called effective bandwidth. (Valdes, 1998).

However VoIP has got a number of setbacks that have been identified by some previous

researchers.

Delay:

There are four types of delay in IP networks: propagation delay, network delay, accumulation delay and processing delay (Valdes, 1998).

Propagation delay: Is that which is caused b y the signal had to travel a distance.

Network delay:

This 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:

This depends on the type of voice coder used. It is caused because a finite amount of time (varying from a single sample time to several m/secs) is needed to collect a frame before the processing begins.

Processing delay:

Is caused by the actual encoding and collection of encoded samples into a packet for transmission.

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.

Latency

Refers to amount of delay between a voice packet being transmitted and received; jitter refers to variation in the amount of that delay. Packets often won't arrive at the destination according to the same route, or will be delayed at the router for different lengths of time, and therefore do not reach the target at a steady rate. Since

speech transmission requires that packets arrive in the proper order and with the proper timing, packet jitter results in erratic speech and a conversation dominated by unpredictable phrasings and pauses.

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. Quality problems are the most important barrier to the development of VoIP. Until a QoS management mechanism is available, this unreliability will remain a major restraint.

Unreliability:

Also Unreliability is due because of concerns in areas such as public safety. Public switched telephone networks (PSTN) are usually well engineered and very reliable. General services are rarely unavailable, and emergency services are extremely reliable for PSTN.

Scalability:

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, however sufficient business, service, network and element management tools and processes are not available, making scalability of general operations difficult. The problem of integrating VoIP gateways with existing PBXs (Public Branch exchanges), switches and routers often results in a complex configuration. This is due to a lack of experienced technicians and the closed architectures of most PSTN network elements.

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.

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. A higher drop-rate than this will result in an incomplete, and possibly indecipherable, message.

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.

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 point of stigmatization.

Regulatory Limitations

The technology provides almost similar functionalities like traditional (PSTN) then there has to be the different regulatory treatment for VoIP. However, there is an unsettled classification of VoIP as data service in telecommunication.

Theoretical Perspective

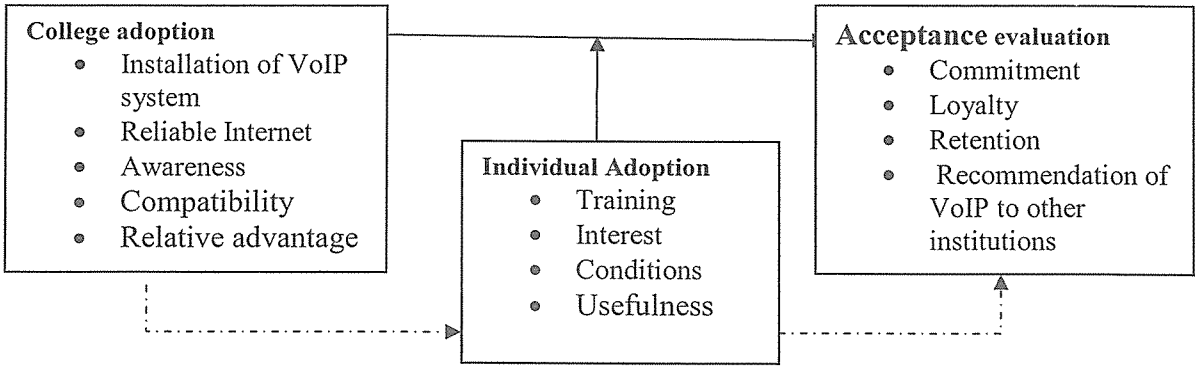
This study was guided by innovation Theoretical perspective; where the decision to adopt to a new technology was by specific concept for actors (individuals and organizations) some authors distinguish this concept into primary adoption and the

concept of secondary adoption. Primary adoption is the organizational decision to adopt and acceptance the innovation, while secondary adoption is the individual's decision inside the organization to adoption and acceptance the innovation (DAVIS, 1993).

The present study is mainly focused on primary adoption, even if we recognize that individual adoption may have an influence in shaping the organizational decision process for the formal adoption. This theory helped the researcher's study because the college has a different technology in place already with different features and different system, so once it adopts to the new technology there will be no need to start from zero buying gadgets and so on, instead the college will undergo a phased mode of renovation first primary and the secondary as stated above. I proposed a theoretical model that will integrate three different perspectives to predict intention to adopt: institutional, management fashion and individuals.

Rukara College benefited by adopting a phased innovation style model stating with institution adoption and its management fashion and then individuals which will ensure an efficient transition into VoIP and allow for necessary training and adopt to change effectively.

Conceptual Framework for VoIP adoption



Source: Modified from: Qureshi, Zafar & Khan, (2008),

This research adopted the theory of innovation model derived from the technology acceptance model (TAM). The model depicted on the conceptual framework above was used to

examine the factors influencing VoIP adoption acceptance. Such as, usefulness, relative advantage, compatibility, technology accessibility, Internet self efficacy, ease of use and reliability. Awareness is whereby the innovator creates awareness of the technology or service. Advertising and public relations are common tools for achieving awareness. In other words, communicate the availability of the new technology or service. Staff and students awareness

Related Studies

Adoption of VoIP in Sweden

This was a master's thesis for Tariq and Kranthi (2007) of Luleå University of Technology located in Sweden. Their research findings were based on some factors which contributed to the slower adoption of VoIP. They categorized these factors into technical and non technical.

With Technical factors they found out that unreliability of 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 had become expected on the PSTN, whereas VoIP was an immature technology, experiencing many problems in this area.

The Internet is a best effort network, where variable latencies and dropped packets occur. Because a voice service requires real-time transmission, VoIP often results in a heavily degraded QoS. They include delay, jitter, echo, unreliability, latency: packet loss, call dropping, calls blocking, presumable low quality of voice calls, evolving standards and interoperability. And with Non Technical Factors that affected the voice over internet protocol included Economic Reasons:

Among economic reasons they also considered the availability of budget for purchasing new equipment. If a company had recently invested heavily into its communications infrastructure, it was less likely that it would want (or be able) to spend a lot of money for the new technology without really good reasons, clear

objectives and high money savings. Though the researcher agreed to majority of the factors brought out by the above researchers, is another important factor among non technical which was left out. This is government revenue and interest. Most governments in developing countries Rwanda inclusive get a large portion of their revenue from telecommunications companies thus introduction and adoption of VoIP will greatly reduce their revenue.

Another Related Studies

Competitive advantage of VoIP adoption in Holland

This study was Master Thesis on competitive advantage of VoIP adoption of Economics & ICT at Erasmus University Rotterdam Holland by: X.R.C. Lahey was conducted on 3 August 2011. His study aimed at identifying the impact of adoption VoIP on the core business of Dutch organizations. An emphasis was put on exploring the possibility for organizations to acquire a competitive advantage by using VoIP.

This research concluded with such findings with the introduction of VoIP organization saved a lot of resources, for example, money and manpower. The introduction of VoIP changed the way of installing new phones in the organization, because the network lines/cables are already there. It is now a matter of requesting an internal number and coupling this number to the VoIP phone.

VoIP had the added benefit that the organization is less depended on the network (national phone network). The new services were beneficial to the organization in terms of costs since these had been significantly reduced in comparison to the previous analog telephone system. The IT department was by then being run by fewer employees. Before the introduction of VoIP, the organization employed more people in total. Thus VoIP ensures that certain processes require less time and manpower to be completed which in turn translates into cost savings. The telephone costs were reduced with the VoIP system, because for each employee the IT

department can control to which numbers an employee can dial (for example: only numbers in the organization, only numbers in the local area etc.

Gaps identified in the related Studies

The various scholarly sources reviewed registered numerous gaps this study intended to fill. Most studies used case study as contrary to quantitative analysis as was applied in this study. Most studies dealt with entities outside educational setting particularly the university/colleges (e.g. analyzing the characteristics of VoIP traffic done July/2007 from Canada it is obvious that having been carried out in 2007 and in Canada there is big variation in results obtained, combining internal and external perspectives for the adoption of VoIP in Italy also carried out in 2007, even some in different countries of which there was need of a similar study to be carried out in Rwanda, enterprise user adoption of VoIP, etc). Most studies seem to recommend rapid growth and reliable VoIP communication system.

In conclusion efficient and affective systems cannot exist in isolation instead need to be integrated. Their integration can be well managed by a critical consideration of the system adoption in regard to reliability, availability, maintainability and supportability.

CHAPTER THREE

METHODOLOGY

Research Design

There study employed descriptive research design as well as quantitative and qualitative research designs were used to describe data collected from the field. Both qualitative and quantitative approaches were taken but with more inclination towards quantitative. According to Yin (2003) quantitative approach results are based on numbers and statistics which are shown in figures. Qualitative approach focuses on describing an event with the use of words and opinions. It was easier to analysis quantitative data using computer based statistical tools as it involves use of numerals.

Research Population

As per records from director of administration and finance (DAF's) office, Rukara College of education has about 140 staff members and over 2000 students. For staff members, this included both teaching and non teaching staff giving a total of 2140 population.

However, for the purpose of this study the researcher mainly focused on staff members that are senior administrative staff (Deans and Head of Departments), academic staff, junior level administrators (technicians) and fourth year students under the department of computing only. The target population was 270 respondents it comprised of 70 staff members and 200 students.

Sample Size

To get the sample size to be used in this study, the researcher used the slovin's formula. The formula is as follows

$n = N / (1 + (N*e^2))$ Where:

n = number of samples and

N = total population

e = margin of error = 5% as a standard figures from social sciences research.

Table 2: Below shows population and sample sizes

Category	Population Size	Sample Size
Staff members	70	42
Students	200	118
Total	**270**	**160**

Using the sloven's formula above to compute the sample size, it's shown on the table above that 42 staff members returned questionnaires from a sample of 70 staff members while 118 students returned questionnaires from sample of 200 students

Sampling Procedure

Random sampling procedure was used for senior administrators and students where as biased sampling was used among the teaching staff and lower level administrators. Specific category of staff and lower level administrators were the ones targeted i.e. those from computing field mainly.

Research Instrument

The tools used in collecting data from the respondents, questionnaires were applied. Structured questionnaires were designed touching on every objective of the study. They were closed ended questions for easy analysis as they were particularly used to collected quantitative data. Open ended questions were minimally used but mainly to get qualitative data which included respondents' opinion.

Validity and Reliability of the Instrument

The test-test technique was used to determine the reliability (accuracy) of the researcher devised instruments 10 qualified respondents who weren't included in the sample size were given questionnaires twice before distribution. Since the questionnaires were answered by respondents the same way each time and the way the researcher expected, the instrument was reliable.

Using the test-test technique validity of the instrument was tested as follows

1) Is the questionnaire measuring what it what it intends to meet?
2) Does it represent the content?
3) Is it appropriate for the sample/ population?
4) Is the questionnaire comprehensive enough to collect all the information needed to address the purpose and the goals of the study?
5) Does the instrument look like questionnaire?

The researcher ensured A content validity index of at least 0.7 (Amin, 2003) rendered the researcher devised tool to be declared reasonably Reliability of the instruments after was tested using the Cronbach Alpha Method of at least 0.7 for the questionnaire to be declared reasonably reliable or consistent.

Data gathering procedures

Before data was collected, an introductory latter was obtained from college of higher degrees and research, addressed to the authorities of Rukara college of education seeking a permission to undertake the research from its area. Upon acceptance the researcher made an appointment to meet with respondents to individually get their consent and informed them that a research is being is being conducted for academic purposes and that their time is being requested to fill in the questionnaires.

During the administration of the questionnaires the respondents where requested to answer the questions and all the information gathered from these questionnaires would be totally confidential and the strictest confidentiality and anonymity shall be preserved.

Questionnaires were distributed among to all the 270 respondents and were left with them for 5 days then collected on collecting the researcher checked all returned questionnaires where checked to verify if all questions were answered. After researcher

gathered, all the data were encoded into the computer and statistically treated using the statistical package for social sciencetists (SPSS) software which contains tools of data analysis.

Data Analysis

The data collected through structured questionnaires were coded and entered into the computer and statistically treated using the statistical package for social sciencetists (SPSS) software which contains tools of that help to analyze data. Descriptive statistics was used to determine the distribution of respondents on personal information and the questions under variable in the research.

During the analysis Mean, frequency and percentages were used to determine the profile of the respondents and content from each variable used in the research. Data display is the major activity which the researcher went through, and this means taking the reduced data and displaying it in an organized, compressed way so that conclusions we more easily drawn. Miles and Huberman (1994) explain that humans are not powerful processors of large amounts of information and that extended text can overload human information-processing capabilities. It is further explained that good display are a major avenue to valid quantitative analysis.

Table 3: Interpretation for analyzed data

Mean range	Response mode	Interpretation
3.26-4.00	Strongly	Agree with No Doubt
2.51-3.25	Agree	with some doubt
1.76-2.50	Disagree	with some doubt
1.00-1.75	Strongly Disagree	with some doubt

Ethical Considerations

The researcher had permission from Rukara College where the study and data collection were conduct as the research was involving them. Some types of research done in a university require Institutional Board Approval and this was one of those. This means that the research had to be approved by an ethics review committee to make sure that the research does not violate any right.

Responsibility

The Researcher was responsible for maintaining the dignity and welfare of all participants. This obligation also entailed protecting them from harm, unnecessary risks, or mental and physical discomfort that may be inherent in the research procedure. Researcher was also responsible for conducting himself ethically and for treating the participants in an ethical manner at all times during data collection.

Reporting Results, the researcher did not fabricate or falsify data of his own wish during presentations. If discovered that the data reported are erroneous, it will be the researcher's responsibility to correct the error through retraction, an addendum, or other appropriate means. In addition, ethical researcher does not present the work of others as his own, or does not fail to give appropriate credit for the work of others through citations.

Limitations of the Study

During the study and data collections the researcher met difficulties; first and foremost Rukara College is far from kansanga its in Rwanda a bordering country to Uganda its 542 kilometers from Kampala to the site, therefore it was so costly for the researcherto get to the site.

It was real hard for the researcher to meet senior administrators in the offices or even get audience from them since they spent most of their time in meetings in and

outside the college. This affected the time scope and budget of the research. Because of their busy schedule they could return incomplete questionnaires this also would further affect data analysis, findings and study completion. Therefore the researcher returned to them questionnaires and wait until they complete.

Bias during data collections some respondents had bias that even if the correct view are gathered and taken the college can't change from existing technology of PSTN since the state owns up hand over it with big shares and collect big revenue from the same technology.

This researcher had problems like misplaced and unreturned questionnaires where by the researcher had to reprint and redistribute new copies of questionnaires. Based on researchers observation it was due to some respondents who were not cooperative.

CHAPTER FOUR
PRESENTATION ANALYSIS AND INTEPRETATION OF DATA

Introduction

This chapter deals with data presentation, analysis and interpretation and also discussion of findings under the objectives and research questions. Thus, collected data was intended to answer the research questions and enable the researcher draw a comprehensive conclusion of the study. It also covers all the processes involved in analyzing the requirements and how the data were interpreted after collection and analysis of the data

Analysis of Data Collection

The study was conducted through data collection methods that include; the use of questionnaires and observation to the specific respondents with knowledge on old communication system these included students, administrators and academic staff. A range of questions were set and distributed to the respondents with the intention of fetching the opinions and facts to the related research topic. The results obtained after the analysis were the baseline for designing a frame work for Voice over Internet Protocol (VoIP) adoption at Rukara college of education. SPSS software program was used to analyze the data obtained from the field.

This chapter also describes what adoption of VoIP should do to satisfy the Rukara College's needs. It also shows the analysis of the respondents that responded during the research study of this project.

Description of population which participated by type.

The Participants who participated where grouped into two groups the students and staff. The staff members who were given questioners were 70 and only 42 questioners were returned representing 75% of the questioners returned. The students

who were given questioners were 200 and 118 questioners were returned giving 65% of the respondents.

Data Presentation and Interpretations.

The table bellow answers the second objective of the study about problems associated with PSTN where by more than 80% of the 160 respondents accepted that making a call using call box is not cheap. This can be proved by analyzing the views on the individual problems like PSTN signals not enough where more than 70% agreed that the PSTN signals were not good enough, On Inter-communication inefficient more that 80 % agreed was so inefficient, also more than 80% said that the monthly charges make it expensive due to monthly revenues collected from the collage, more than 65% said that PSTN isn't Mobile hence making it inefficient, about PSTN network insecurity more than 65% of respondents agreed.

Table 4: Problems associated with PSTN

A call using call box is not cheap	**No of respondents**	**Percentages**
strongly disagree	1	0.6
disagree	25	15.6
agree	50	31.3
strongly agree	84	52.5
Total	160	100.0
Enough inter-office phone lines	**No of respondents**	**Percentages**
strongly disagree	4	2.5
disagree	21	13.1
agree	51	31.9
strongly agree	84	52.5
Total	**160**	**100.0**

PSTN signal not good enough	No of respondents	Percentages
strongly disagree	8	5.0
disagree	31	19.4
agree	63	39.4
strongly agree	58	36.3
Total	**160**	**100.0**

Inter-communication inefficient	No of respondents	Percentages
strongly disagree	4	2.5
disagree	19	11.9
agree	59	36.9
strongly agree	78	48.8
Total	**160**	**100.0**
Monthly charges makes it expensive	**No of respondents**	**Percentages**
strongly disagree	10	6.3
disagree	20	12.5
agree	36	22.5
strongly agree	94	58.8
Total	**160**	**100.0**
PSTN isn't mobile thus inconvenient	**No of respondents**	**Percentages**

strongly disagree disagree agree strongly agree	16 33 50 61	10.0 20.6 31.3 38.1
Total	**160**	**100.0**
PSTN network is insecure	**No of respondents**	**Percentages**
strongly disagree disagree agree strongly agree	17 37 23 83	10.6 23.1 14.4 51.9
Total	**160**	**100.0**

Using Interpretation Table bellow

Using Highest-Lowest/Highest

Whereby the Highest is=4 and Lowest is 3. Therefore 4-3/4=0.75.

This gives criteria for clear Interpretation. Criteria: 400-0.75= 3.2, 3.25-0.75= 2.50 and 2.50-0.75= 1.75 as indicated by David (1954).

Table 5: Interpretation for PSTN problems

Problems Associated with PSTN	Mean	Interpretation	Rank
A call using call box is not cheap	3.36	Very unsatisfied	1
Monthly charges makes it expensive	3.33	Very unsatisfied	2
Inter-communication inefficient	3.32	Very unsatisfied	3
PSTN signal not good enough	3.10	Unsatisfied	4
PSTN network is insecure	3,08	Unsatisfied	5
PSTN isn't mobile thus inconvenient	2.98	Unsatisfied	6
Average of Means	**3.20**	**Unsatisfied**	

Using the table above showing the interpretation of each problem associated with PSTN, the Average of the means clearly shows that the respondents agreed that there was a problem with PSTN communication System hence effective communication system is needed. Data Interpretations on Feasibility of VoIP Adoption

The table bellow answered the third objective of the study about feasibility of adoption of VoIP where by more than 80% of the 160 respondents accepted that there is no policy in place for VoIP adoption at the college. This can be proved by analyzing the individual views on the feasibility of VoIP adoption in the college where by more than 70% strongly disagreed that the Internet is not reliable to support VoIP adoption, On whether Sensitization is a factor for adoption of VoIP, more that 80 % strongly agreed that it's a big factor for the adoption since the college has reliable internet connection, also more than 85% said that the Teleconferencing can help on meetings held within and outside college premises this ensures saving both money and time hence service delivery is maximized, more than 65% of respondents said, Available free down loads make VoIP cheap since VoIP has some free features down loads its therefore a strong factor for its adoption by younger institutions, about VoIP technology are mobile thus convenient more than 80% of respondents accepted that once the technology is adopted would give free and friendly working environment since they would even do their work from home or whenever they would be, and more than 60% respondents agreed that academic institutions in Rwanda to be able to adopt to VoIP communication system regulatory authorities need to influence the adoption.

Table 6: Investigates feasibility of adoption of VoIP

Policy in place for VoIP adoption	No of respondents	Percent
strongly disagree	83	51.9
disagree	50	31.3
agree	27	16.9
Total	**160**	**100.0**
Internet isn't reliable to support VoIP	**No of respondents**	**Percent**
strongly disagree	75	46.9
disagree	47	29.4
agree	27	16.9
strongly agree	11	6.9
Total	**160**	**100.0**
Sensitization is a factor for adoption of VoIP	**No of respondents**	**Percentage**
strongly disagree	22	13.8
disagree	24	15.0
agree	28	17.5
strongly agree	86	53.8
Total	**160**	**100.0**

Teleconferencing can help on Meetings	No of respondents	Percentage
strongly disagree	27	16.9
disagree	8	5.0
agree	36	22.5
strongly agree	89	55.6
Total	**160**	**100.0**
Available free down loads make VoIP cheap	**No of respondents**	**Percentage**
strongly disagree	35	21.9
disagree	20	12.5
agree	37	23.1
strongly agree	68	42.5
Total	**160**	**100.0**

VoIP technology are mobile thus convenient	No of respondents	Percentages
strongly disagree	19	11.9
disagree	29	18.1
agree	30	18.8
strongly agree	82	51.3
Total	**160**	**100.0**

Authorities influence the adoption	No of respondents	Percentage
strongly disagree	28	17.5
disagree	33	20.6
agree	33	20.6
strongly agree	66	41.3
Total	**160**	**100.0**

Table 7: Interpretation for Feasibility of VoIP adoption

Feasibility for the adoption	Mean	Response mode	Interpretation	Rank
Teleconferencing can help on meetings	3.17	Agree	Satisfied	1
VoIP technology are mobile thus convenient	3,10	Agree	Satisfied	2
Sensitization is a factor for adoption of VoIP	3.11	Agree	Satisfied	3
Available free down loads make VoIP cheap	2.86	Agree	Satisfied	4
Authorities influence the adoption	2.86	Agree	Satisfied	4
Internet isn't reliable to support VoIP	1.84	Disagree	Unsatisfied	5
Policy for VoIP adoption	1.65	Strongly Disagree	Very unsatisfied	6
Average of Means	**2.67**	**Agree**	**Satisfied**	

Using the table above showing the interpretation of feasibility of VoIP adoption, the Average of the means clearly shows that the respondents agreed that a possibility for feasibility of VoIP adoption in the college is a prerequisite to make the college save a great deal of money from its advanced multiple features, Since it's been clearly identified that the old communication system does not meet the college's communication expectations.

Table 8: Interpretation of problems associated with VoIP

VoIP questions	Mean	Interpretation	Rank
Internet connection affects call quality	2.70	Satisfied	1
VoIP does not require technical skills	2.61	Satisfied	2
communication is only for online uses	2.58	Satisfied	3
Internet connectivity not reliable enough	2.51	satisfied	4
Internet intensively is unaffordable	2.41	Unsatisfied	5
VoIP network not secure enough	1.90	Unsatisfied	6
Signal quality not strong enough	1.71	Very Unsatisfied	7
Average mean	**2.35**	**Unsatisfied**	

From respondent's point of view the college experiences unreliable internet connection, the signal is not at all weak and VoIP network security is secure for those who have already used its services, the transmission quality of a call is affected by internet connectivity and users need training for them to be able to fully use VoIP features. Furthermore the college is a government institution that uses state funds therefore has capacity and ability to pay for even enough bandwidth.

Profile of the respondents

In this study targeted users were asked to indicate their age whereby above 80% respondents were below 25 years of age. The researcher wanted to know the age of each respondent to verify the validity of their responses in the sense of comparing their age and level of education for example if a respondent aged below 25 years said has PhD this would indefinitely show abnormality in matching the respondents age and qualifications especially from third world countries like Rwanda which had prolonged

tribal wars that affected education system. This is very important in assessing the validity of this crucial data that is beneficial to Rukara College, future readers and the researcher himself.

The study targeted users and were asked to indicate their gender whereby 101 of respondents out 160 were male and only 59 of them were female. The researcher wanted to know the appropriate number of male and female who participated in the study and whose responses helped to design a framework for the adoption of VoIP at Rukara College of education and at what degree does a particular sex participates in the study.

In the same study, targeted users were asked to indicate their level of education whereby more than 90% of respondents from the total of 60 respondents were staff members with at least a degree. Here the researcher wanted to know the level of education for each respondent to evaluate or assess the technicality of their responses in respect to adoption of VoIP in the college, this gives a feeling and confidence that people responded are educated and well knowledgeable with old communication system, hence helping to design a framework.

Targeted users were asked to indicate their level of position whereby 90% of respondents from the total of 60 were academic staff this was as a result of being the majority in the sampled staff members. Here the researcher wanted to know the level of position for each respondent; this helped the researcher to know how deep the respondents are in respect to the old communication system and their views on possibilities for the new communication system, therefore giving the exact image of the existing technology at the college.

The study also targeted users and were asked to indicate their courses taken that is on behave of students whereby 46 respondents (students) from the total of 100 questionnaires returned were taking IT/Education which contributed 46%, 42 were

taking Computer Science/Education that made 42% and finally 12 were taking MIS/Education that made 12%. Here the researcher wanted respondent with deep background of the telecommunication technology in place and since these are 4th year students it's understood to be fully equipped with knowledge; this helped the researcher using respondent's views in indentifying problems of the old communication system and getting the exact image of the existing technology used at the college.

Table 9: Shows the profile of respondents.

Age group	**No of respondents**	**Percentage**
below 25	84	52.5
26-39	63	39.4
40-54	12	7.5
55 above	1	.6
Total	**160**	**100.0**
Respondent's sex	**No of respondents**	**Percentage**
Male	101	63.1
female	59	36.9
Total	**160**	**100.0**
Respondent's position	**No of respondents**	**Percentage**

senior admin	8	5.0
junior admin	18	11.3
academic staff	34	21.3
Total	60	37.5
Missing	100	62.5
Total	**160**	**100.0**
Level of study	**No of respondents**	**Percentage**
Diploma	8	5.0
bachelors	22	13.8
masters	27	16.9
PhD	3	1.9
Total	60	37.5
Missing	100	62.5
Total	**160**	**100.0**
Course taken by a student	**No of respondents**	**Percentage**
CS/ED	42	26.3
IT/ED	46	28.8
MIS/ED	12	7.5
Total	100	62.5
Missing	60	37.5
Total	**160**	**100.0**

Qualitative analysis study

The last part of the questionnaire has structured questions that are studied and analyzed qualitatively; this kind of study focuses on qualitative information for inspecting

the phenomenon. Such information is usually in the form of words and narratives. It provides a different approach to studying complex phenomena and for some cases it is viewed as the most adequate method to be used. Qualitative research involves the utilization of data that is qualitative in nature, such as the answers provided from structured questions, documentation and participant observation.

The reason for the selection of a qualitative research method is founded in its characteristics. Qualitative research is a method that better suites this study since quantifiable data is not available or is harder to acquire using such questions. In this study the "phenomenon" to be studied is VoIP and its adoption in the academic institutions in Rwanda. Although VoIP is not so much a direct observable occurrence (phenomenon), its development and market progress can be defined in terms of quantitative and qualitative measurements.
These are structured questions that are earlier alone mentioned in this study and require qualitative study because of their nature. What are the major advantages of using VoIP? What are the major challenges of VoIP Technologies?

The above kind of questions were distributed along with others questions since they were contained on one questionnaire but more than 85% of respondent from 160 respondents on the first question which was the advantages of using VoIP, they answered it's a cheap communication technology. And despite of other factors, but being cheap gives a big alarm and a sign for the college to adoption to this technology. These were personal views due to individual understanding based on the given experience achieved without any vital influence from Rukara College.

Furthermore 75% of respondent from 160 respondents on the second question which was the challenges of VoIP Technologies, they answered unreliability of internet due to low bandwidth of 60mbs available in the college which can't support VoIP services. They also emphasized on unawareness of VoIP in place, that besides unreliability of internet connectivity but people around the college and Rukara College in

particular are not well informed about the technology, however some staffs and students have some knowledge due to individual merits. This indicated that some VoIP trainings and increase on band width will ensure VoIP adoption and success at the Rukara College.–

Pearson's Linear Correlation Coefficient for the existing system of PSTN and the proposed system of VoIP in Rukara College in Rwanda.

		The existing system	The proposed system	Interpretation	Decision on Ho
The existing system	Pearson's Correlation	1	.980	Positive and significant	Rejected
	Sig.		.000		
	N	7	7		
The proposed system	Pearson Correlation	.980	1		
	Sig.	.000			
	N	7	15		

The results in table above show that there is a positive significant difference between the level of performance and efficiency of the existing system and the proposed system for VoIP in Rukara College. For instance the level of performance of the existing system is positively and significantly correlated with the proposed system performance (r=0.980, sig. =.000), The researcher rejected the null hypothesis and affirmed that there is a significant difference between the existing system of PSTN technology and the proposed one of VoIP technology in Rukara College in Rwanda.

CHAPTER FIVE

FINDINGS, CONCLUSIONS AND RECOMMENDATION

This Chapter presents the outcome from the study in the form of discussions, conclusions and recommendations. Discussions and then general conclusions will be presented with implications for future research. The evaluation of VoIP adopted, can be recalled at this early stage, and consequently, how well the future can be foreseen with the emergence of this new technology. In this regard the researcher focused on the major causes or the problems which hindered the adoption of VoIP technology by academic institutions in Rwanda.

Therefore designed a framework for its adoption and then advice on how to improve those factors or to rectify those problems and in order to achieve the study purpose. The convergence between voice and data transmission networks is inevitable. While VoIP is still fairly new especially in developing countries including east Africa and Rwanda in particular, it has great potential because it is capable of supporting both data and voice since its internet based. According to the results from the respondents demonstrate that 80% of the 160 respondents accepted that making a call using call box is not cheap. Therefore since using call box is PSTN technology it's the right time to change and go against this costly technology.

FINDINGS

It was confirmed by the administrators and other senior officers of Rukara College that the existing technology for communication does not deliver quality of service based on today's institutional development needs. This research was therefore aimed at evaluating the possibility for Rukara College of education to a adopt to VoIP technology which has new features that would give additional advantages that enhance the college to save great deal of money spent due to old communication technology in place, however the same technology was partially still used country wide.

This will greatly enabled development to the Rukara College of education due to a number of factors including time saved for meeting done in and outside the college, efficiency at work for employees since they will use portable and mobile technology.

Future Research Suggestions

This study had some limitations that needed to be taken into account. First this research did not seek to research or make any assumption about the more advanced technical aspects of the VoIP technology, nor did it focus on the general technical aspects of communication technology. Furthermore during data collections some respondents had bias that even if the correct views were gathered and taken the college couldn't change from existing technology of PSTN since the state owned an up hand over the communication technology with big revenue.

The researcher utilized a managerial approach and was meant to provide an insight into the possibility for institutions to acquire the advantage of using VoIP technology. In order to retain the scope and due to the time constraints of this research only a relatively one academic institution was interviewed and Also in case of VoIP, there is the transmission of voice, data and even the videos, but due to lack of time the researcher mainly focused on voice transmission, so in future, researcher(s) can also do the research about the transmission of Videos over the Internet protocols. Future research could include performing an exploratory case study in order to assess the relative added benefits for organizations that introduce VoIP Communications. Further quantitative study could be conducted by interviewing a wider range of organizations/institutions on VoIP.

It is also important to ensure that phone-line is minimal so that it is inexpensive to the business. If this system costs too much, the benefits of VoIP convergence are lessened. For businesses unable to undergo a project of this scope at this time I would

recommend staying educated about current events in VoIP advances and planning to implement in upcoming years as it becomes more practical for them to do so.

Conclusion

The conclusion was based on research objectives, analysis and interpretation of individual views on all problems associated with PSTN that are categorically summarized in percentages like, PSTN signals not enough where more than 70% said that PSTN signals were not good enough, on Inter-communication inefficient more that 80% said was so inefficient, also more than 80% said that the monthly charges make it expensive due to monthly revenues collected from the collage, more than 65% said that PSTN isn't Mobile hence making it inefficient, about PSTN network insecurity more than 65% said that PSTN isn't Mobile hence making it inefficient, about PSTN network insecurity more than 65% of respondents said its insecure. Therefore the mean of 72% from the above all answers indicate that there are strong problems with existing technology that altered college development.

From the feasibility study carried out for the adoption of VoIP many questions were asked in respect to VoIP adoption with an aim for overview of whether there are possibilities for Rukara college to adopt to the new technology, using mean to summarize all the answers that were answered over 70% that the college has high possibilities to adopt to the new technology.

Recommendation

Rukara college is small an academic institution with vision of development it is therefore recommended that rukara college should adopted to VoIP technology due the fact that VoIP gives a great deal of saving huge amount of money and enable efficiency and effectiveness at work to employees.

This recommendation is based on interpretations of data analyzed from respondent's views in respect to research questions. A policy in place for VoIP adoption will indefinitely benefit the college in terms of adoption, this should involve in sensitization and awareness of VoIP technologies and its benefits, the college should have reliable internet connectivity with enough band width for consistency and faster communication.

Training users, from the respondents point of view, some users have acquired skills about VoIP on an individual merit without any effort from the college, this has completely delayed the adoption, and therefore the college should put a system for capacity building in place and get ready to adopt to VoIP technology since its benefits out way that of PSTN as an existing technology.

Regulatory authorities since the authorities are state agencies they should incorporate VoIP with PSTN and regulate both of them as a way of giving a chance to institutions bit academic or business enterprises to asses, adopt and utilize the benefits a better technology since it's the state that gets revenue from whatever technology in place, better if the technology gives quality of service to match the expected demands based on instruction's goals and vision.

REFERENCES

Amin (2003), (2005) reliability of instrument.

Bradly Mitchell, (2003) (VoIP) Voice over internet protocol.

Brawley (2004) fundamentals of telecommunication.

Davidson (1954). Mathematical Statistics Universal Bounds for Mean Range and extreme Observation.

Davis Rogers (1993) on theoretical perspective

Endler & collier (2007) telecommunication second edition.

Foster (1998). Industrial Marketing Communication: An empirical Investigation on the use of marketing communication tools, Department of business administration and social science.

Fred Davis (2005) telecommunication engineering second edition.

Gideon Burn, (2003), (2005) Networking fundamentals.

Giggs (2005) VoIP Call Features and internet protocol CCNA.

Holmes et la, (1997), internet security Information (November2000).

Jonathan Davidson, James Peter (2000), Voice Over IP Fundamentals, Cisco Press.

Jonathan Lennox, Henning (2000), 'Feature Interaction in Internet Telephony', Columbia University, May 2000.

Jonty, 2006. "VoIP in Asia to Explode by 2009"

King (2001) internet protocol CCNA.

Leyland (2005) Computer Networking: A top-down approach featuring the Internet

Third edition, Addison Wesley.
Maxwell (2000) 'Challenges for VoIP Technologies in Corporate Environments'
Micheal L. Zubey, William Wagner, Fames R. Otto (2002), A conjoint Analysis of voice.
over IP attributes, Electronic Networking Applications and policies Vol12.
Miles and Huberman (1994), 'Qualitative Data Analysis'. Second Edition. London:
Sage publications.
Sacker et al (2006) (VoIP) Voice over internet protocol.
Sandvine (2001), Using Network Intelligence to Provide Carrier-grade.
Sharma (2003): Impact of VoIP and QoS on Open and Distance Learning.
Tariq and Kranthi (2007) Luleå University of Technology.
Valdes, R. How VoIP Works.
Vetteth & Joseph (2009), Voice over IP Vol. 45, No. 1.
Vincent, R, 2003. Why Voice over IP Is on Hold.
Webach (2005) fundamentals of networking Second edition.

APPENDIX 1

TRANSMITTAL LETTER FROM SPGSR

KAMPALA
INTERNATIONAL
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Mobile: +256782982360/
E mail: mpholixx@gmail.com

COLLEGE OF HIGHER DEGREES AND RESEARCH (CHDR)
OFFICE OF THE HEAD OF DEPARTMENT, APPLIED SCIENCE AND TECHNOLOGY

Date: 16th April 2012

Dear Sir/Madam,

RE: REQUEST FOR KALAMUZI JAMES MIS/1806/72/DU TO CONDUCT RESEARCH IN YOUR INSTITUTION

The above mentioned is a bonafide student of Kampala International University pursuing **Masters of Science in Information System**.

He is currently conducting a research entitled **"Adoption of Voice Over Internet Protocol in Academic Institutions; A case of Rukara College of Education"**

Your Institution has been identified as a valuable source of information pertaining to his research project. The purpose of this letter is to request you to avail him with the pertinent information he may need.

Any information shared with him from your Institution shall be treated with utmost confidentiality.

Any assistance rendered to him will be highly appreciated.

Yours truly,

KAMPALA INTERNATIONAL UNIVERSITY CHDR P.O. BOX 20000, KAMPALA

Businge Phelix Mbabazi
Head of Department, Applied Science and Technology

NOTED BY:
Dr. Sofia Sol T. Gaite
Principal-CHDR

RUKARA COLLEGE
OF EDUCATION

Education is our Shield

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E-mail: rukarac.education@gmail.com

01/01/2012

RE: LETTER OF ACCEPTANCE

The management of Rukara College of education has accepted Kalamuzi James from Kampala international university to carry out an academic researcher on VoIP (voice over internet protocol) with a case study of Rukara College.

Based on his study objectives and purpose the entire Rukara college is privileged to be a monger the beneficiaries of this study since it entirely tackles on communication problems of this College, we therefore request all the targeted respondents to give him all the required information in relation to his research study as indicated on the questionnaires.

Muhire Cassia

Personnel manager

01/01/2012

APPENDIX III

INFORMED CONSENT

Dear Participant,

I am a candidate for masters of management information system at Kampala International University, currently carrying out research on adoption of Voice over internet protocol. You are kindly requested to participate in this research to develop a framework for Adoption of Voice over internet protocol in academic institutions in Rwanda. The questionnaire seeks to gather information about how VoIP as new emerged technology in communication industry with pressure to solve problems associated with PSTN technology. Please ***note*** that: There are no ***correct*** or ***incorrect*** responses, so please kindly provide the most appropriate information as indicated in the questionnaires and please do not leave any item unanswered. All the information gathered from this questionnaire will be **totally confidential** and the strictest confidentiality and anonymity shall be preserved. The results of a statistical analysis of the data from this study will be used to develop a framework for the adoption of VoIP in academic institutions in Rwanda.

Yours faithfully

Kalamuzi James

Master's Degree Candidate

INFORMED CONSENT

In signing this document, I am giving my consent to be part of the research study that will focus on Adoption of Voice over internet protocol in academic institutions in Rwanda. I shall be assured of privacy, anonymity and confidentiality and that my name shall not be publicized in the final report nor will there be any cross-references made that can link the results of the questionnaire to me.

Code#.....respondent..... Date received.....

APPENDIX IV

RESEARCH INSTRUMENTS

Questionnaire for collecting data on evaluation for adoption of Voice over Internet Protocol (VoIP) at Rukara College

SECTION A: (RESPONDENT'S BACKGROUND)

1.0 How old are you: (Please cycle correct answer)

(1) Below 25 (2) 26 – 39 (3) 40 – 54 (4)55 above

1.1 What's your gender? (1) Male (2) Female

1.2 What position/s do you hold in the college if a staff?

(1) Senior admin (2) junior admin (3) academic staff

1.3 What are your academic qualifications?

(1) Diploma (2) Bachelors (3) Masters (4) PhD

Direction: On the space provided before each option, indicate your best choice by using the rating system below:

Response Mode	Rating	Description
Strongly Agree	(4)	You agree with no doubt at all
Agree	(3)	You agree with some doubt
Disagree	(2)	You disagree with some doubt
Strongly disagree	(1)	you disagree with no doubt at all

SECTION B: PROBLEMS ASSOCIATED WITH PUBLIC SWITCHED TELEPHONE NETWORKS (PSTN)

Public Switched Telephone Network refers to the normal international telephone phone network, based on copper wires or land line carrying voice data. It has dynamically allocated dedicated bandwidth.

PROBLEMS ASSOCIATED WITH PSTN TECHNOLOGY

_____2.0. Making a call using call box is not cheap.

_____2.1.The College has enough inter-office fixed phone lines public branch exchange (PBX)

_____2.2. PSTN network signal is not good enough

_____2.3. Have insufficient skills in PSTN use

_____2.4.The inter -office communication through PBX Is not efficient enough

_____2.5. Monthly charges for phone line connection, plus per-minute charge for long distance makes it expensive

_____2.6. PSTN technology uses a fixed land line thus makes it more inconvenient for a college because they are not mobile

_____2.7. PSTN network is insecure

SECTION C: PROBLEMS ASSOCIATED WITH VOICE OVERINTERNET PROTOCOL (VOIP)

VoIP refers to the movement of voice traffic over Internet Protocol (IP) based networks. Example **Skype, teleconferencing and others**

Problems associated with VoIP technology

_____3.0. VoIP requires more ICT technical skills.

_____3.1. VoIP quality signal is not strong enough

_____3.2 VoIP services are internet based and communication is possible only if both parties (users) are online

_____3.3. Call quality is affected by the quality of

internet connection

_____3.4 have insufficient skills in VoIP use

_____3.5 VoIP is Internet intensive and yet daily reliable Internet is unaffordable at the college

_____3.6. VoIP network is not secure enough

_____3.7.The internet connectivity is not reliable enough to support VoIP in the college

SECTION D: FEASIBILITY FOR VOIP ADOPTION

Feasibility for VoIP adoption

_____4.0. The college has policy in place for VoIP adoption.

_____4.1. Internet connectivity is reliable enough to support VoIP

_____4.2 VoIP awareness and sensitization are factors which contribute to better adoption of VoIP in the college.

_____4.3. Teleconferencing can help me on meetings held within and outside the college premises

_____4.4. Most of VoIP features are available on Internet therefore can be down loaded freely e.g. Skype hence cheap

_____4.5. Some VoIP technologies are Mobile and portable therefore can use VoIP services anywhere once internet connection is in place

_____4.6. Regulatory authorities influence the adoption

5.0 What are the major advantages of using VoIP?

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6.0 What are the major challenges of VoIP Technologies?

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1999- 2002: **Station Manager,**

Employer: Investco (Investment Company).

Major achievements

- Worked with INVESTCO as station manager, established business plans, project planning and management, controlled all the funds and managed the human resource of the station.
- Involved on procurement committee.

EDUCATIONAL BACKGROUND

2010	CCNA (Cisco certified network associate) from Monaco labs Pursuing a Master of Science in management information system degree (KIU), Kampala International University.
2002-2005:	A Bachelor Degree in computer science and information technology Kampala I University (KU), Kampala Uganda.

Important Skills:

Programming in C, PhP, VB 6.0.

- Developing applications in Oracle, Informix dynamic Server, my SQL and SQL server 2000/2005.
- Implementing and managing computer Networks
- IT project management and planning.

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