A FEASIBILITY STUDY ON VoIP-PERSPECTIVE BANGLADESH

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This Report Presented in Partial Fulfillment of the Requirements for the Degree of Bachelor of Science in Electronics and Telecommunication Engineering

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APPROVAL

This Project titled "**A Feasibility study on VoIP-Perspective Bangladesh"** submitted by Md. Billal Hossain, and Md. Mizanur Rahman Bhuiyan to the Department of Electronics and Telecommunication Engineering, Daffodil International University, has been accepted as satisfactory for the partial fulfillment of the requirements for the degree of B.Sc. in Electronics and Telecommunication Engineering and approved as to its style and contents. The presentation has been held on 26 February 2011

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DECLARATION

We hereby declare that, this project has been done by us under the supervision of Md. Mirza Golam Rashed, Assistant Professor, Department of ETE Daffodil International University. We also declare that neither this project nor any part of this project has been submitted elsewhere for award of any degree or diploma.

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Abstract

VoIP is a world-shattering technology, which has become a key topic in both the growing Internet industry and established telecommunications industry. In VoIP technology telephone calls are to be made over computer networks like the Internet where packet switching technique is used instead of circuit switching and analog signals are converted into the digital signals in the real time two-way transmission of conversation. VoIP has become a potential alternative to the existing PSTN technology due to its reduced costs. Despite of its reduced cost, it has major challenges, which are affecting its adoption.

This project, which consists are several segment to explore recourses allocation to establish the positive views of VOIP-Perspective Bangladesh. First chapter and second chapter are to bring out the theoretical knowledge on VoIP for reader, third chapter is about VoIP soft Switch (VPS) which can give understanding about VoIP system manage (VSM), fourth and fifth chapter are fully experimental study where we have tried to conduct to understand the potentiality of VoIP in Bangladesh. We have executed some data analysis like connected call ratio, calls per hour in a day, call rate comparison VoIP vs. BTCL, call duration of VoIP calls ,PDD of VoIP calls, successful and failed calls of a VoIP provider, difference type user ratio .

In summary, this study investigates the major causes for the slower adoption of VoIP and also proposes the suggestions for improving those causes in the perspective of Bangladesh.

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CHAPTER-1

Introduction

In this chapter reader will be introduced with background of the Voice over Internet protocol technology. Problem discussion, purpose of the research and research questions will also be discussed. Finally line out the purpose of the report will also be presented.

Ever since its advent VoIP has opened new doors for telephony bringing forward immense possibilities. The basic reason for the popularity of VoIP is the cost which is very low as compared to the conventional telephony services. The concept of transmission of voice over data stream makes it possible to have VoIP transmitted and received using anything that uses IP - laptops, PC's, Wi-Fi enabled handsets etc. [10]

1.1 Background

The global evolution of the Internet and the wide spread growth of networks have been made the Internet part of our everyday life. This is the reason why the interest and demand on different applications has been increased. The raise in demand has produced many new applications. Voice over Internet Protocol (VoIP) technology has become a potential alternative to and supplement of the traditional telephony systems over the Public Switched Telephone Network (PSTN), providing a versatile, flexible and cost-effective solution to speech communications. Internet telephony is a revolutionary technology that has the potential to completely rework the world's phone systems. Internet telephony is the transmission of voice signals from one party to other party digitally i.e., usage of packet switched data network (PSDN). The first documented internet telephony experiments were conducted on the then ARPANET (the forerunner of the Internet) by researchers at MIT in the mid 1970s, resulting in the publication of an Internet protocol specification, RFC741, for the 'Network Voice Protocol', in 1977[1]. These experiments resulted in audio transmission on packet networks but were limited to academic environments only. As computers of that age did not have the power to compress the audio data below 64 kbps or 56 kbps and sound input and output devices have also to be made because there were none to be bought. But later when the computing power to compress the speech below 14.4 kbps by 1993 then first commercial Internet phone Application appeared.[1]

The public switched telephone network (PSTN) has been evolving ever since Alexander Graham Bell made the first voice transmission over wire in 1876. In traditional telephones, devices are limited to communicating with those devices, which are connected directly, and the telephony companies and their protocols must handle all location and routing features. Traditional telephone uses circuit networks.[1]

As PSTN works on circuit switching technique in which network establishes a dedicated end to end connection between two hosts. The resources needed for communication between these end systems are reserved for the duration of communication session. The main disadvantage of circuit switching is the dedicated circuits are idle during silent periods and thus network resources are wasted during these contemplation periods.[6] Unlike PSTN, VoIP uses packet switching, which sends digitized voice data packets over the internet using many possible paths. The packets are reassembled at their destination to generate the voice signals.[9]

VoIP uses packet switching where the network resources are not reserved; a session's messages use the resources on demand, and as a consequence, may have to wait for access to a communication link. In packet switching the packet is sent into network without reserving any bandwidth. If one of the links is congested because other packets need to be transmitted over the link at the same time, then that particular packet has to wait in queue at the sending side of the transmission link and hence suffer a delay. The internet makes its best effort to deliver packets in a timely manner but it does not make any guarantees.[9]

Table 1: A qualitative comparison of voice over PSTN and over VoIP.

Concept	Voice Over PSTN	Voice Over VoIP
Switching	Circuit switched (end-to-end dedicated circuit set up by circuit switches)	Packet switched (statistical multiplexing of several connections over links).

Bit rate	64kbps pr 32kbps	14kbps with overhead*
Latency	< 1 00ms	200-700ms depending on the total traffic on the IP network. Lower latencies possible with private IP networks.
Bandwidth	Dedicate	Dynamically allocated
Cost of access/billing	Business customers. Monthly charge for line, plus per-minute charge for long distance cost of PBX, and other telephony	Business customers. Cost of IP infra-structure, Hybrid IP/PBX, and IP phones. Residential customers. Monthly charge for
	Equipment. Residential customers. Monthly charge for line, plus per-minute charge for long distance, cost of simple phone.	Line, plus monthly charge for ISP, cost of computer, and other equipment.
Equipment	Dumb terminal (less expensive); intelligence in the network	Integrated smart programmable terminal (expensive); intelligence not in the network.
Additional features and services	Requires reprogramming or changes in the network design but fast enough to add if advanced intelligent networks (AIN) are in use.	Easy to add without major changes, due to flexible protocol support, but standards are needed for traditional. user services
Quality of service	High (extremely low loss)	Low and variable, but traffic is sensitive depending on packet loss and delay experienced.
Authorization and authentication	Only once when the service is installed	Potentially required, per-call basis

Regulations	Many at federal and state levels	Few yet, but regulatory uncertainty; future regulations may reduce the cost advantages of VoIP.
Network availability	99.999% up time	Level of reliability is not known
Electrical power failure at customer premise	Not a problem; powered by a separate source from phone company.	Will have problems, as equipment may be down. Power from other sources is not easy to obtain
Security	High level of security because one line is dedicated to one call	Possible eavesdropping at routers.

Source: Communications of the ACM January 2002/Vol. 45, No. 1

1.2 How does VoIP work?

VoIP uses Internet Protocol for transmission of voice as packets over IP networks. The process involves digitization of voice, the isolation of unwanted noise signals and then the compression of the voice signal using compression algorithms/codes. After the compression the voice is packetized to send over an IP network, each packet needs a destination address and sequence number and data for error checking. The signaling protocols are added at this stage to achieve these requirements along with the other call management requirements. When a voice packet arrives at the destination, the sequence number enables the packets to be place in order and then the decompression algorithms are applied to recover the data from the packets. Here the synchronization and delay management needs to be taken care of to make sure that there is proper spacing. Jitter buffer is used to store the packets arriving out of order through different routes, to wait for the packets arriving late. [10]

1.3 Problem Discussion

The largest incentive to replacing your traditional phone line with a VoIP line is of course the price. A VoIP phone line can save you a great deal of money over your traditional phone service, especially on long distance and international charges It is not yet widely known, but the second key benefit of VoIP over a traditional phone line are the additional calling features available. The features offered by VoIP are quite varied and can be useful for both business and residential customers. Following are the new calling features offered by VoIP providers. [1]

Online call logs. Provide you with an online log of all your incoming and outgoing calls. This feature is useful for easily tracking your expenditures on long distance calls.

- **Locate me.** You can set up your VOIP service to ring multiple phone lines simultaneously or in sequence. This feature is very useful for someone with multiple phone lines or if you only want to have to give out one phone number. For example, you could set up your phone service to ring both your home phone and cell phone at the same time. With this feature, you'll never miss that important call again.

- **Online voicemail.** This feature takes your traditional voicemail service and adds a number of new elements. Not only can you access your online voicemail from any phone, you can access it from any PC as well. With online voicemail, you can access your voicemail account from an Internet browser and listen to your messages using your computer. You can also have your voicemail messages sent to your e-mail account.

- **Conference calling.** Although you could create conference calls with a traditional phone line, VOIP service makes this feature much easier. Some providers allow you to create a conference call between up to 10 people and include this server in your monthly package.

- **Free in network calling.** Long distance rates from VOIP providers are already much cheaper than the rates of traditional phone or long distance companies. Some VOIP providers take these discounts a step further and offer free long distance service if the other person you are calling is also using the same provider.

- **Call Transfer.** Allows you to easily transfer calls to another number by pressing a few numbers on your phone.

- **Computer Dial.** Allows you to store phone numbers on your computer and use your computer to dial them for you. This feature is very useful for dialing calling lists. For example, say you wanted to call all of the members in your book club. You can organize all of the members into a contact list on your computer. You can then use your computer to dial all of the contacts sequentially. When using computer dial, your computer will first ring your phone, wait for you to pick up, and then dial the number you are calling.

Voice over internet protocol technology was emerged by many years now not too many businesses or individual users are using this new technology. According to, Technologies and mechanisms for the transmission of voice over IP Networks were very promising some years ago and they stay the same today but the adoption rate is rather modest. The three main reasons in adopting the VoIP technology are low call cost, lower cost of infrastructure and integration of voice and data applications.

Still, the adoption of VoIP has been slower than expected; mainstream subscribers and corporations have been reluctant to adopt the new technology, despite its cost savings, flexibility, and new functionality. The issue comes down to trust: until recently, service providers have lacked the network intelligence needed to ensure appropriate reliability and Quality of Service (QoS) levels, and, in the absence of anything proving otherwise, subscribers rightly believed that they could not get the same service from VoIP as from PSTN.

1.4 Purpose and Research Questions

The purpose of this study is to find out the causes/ reasons which are affecting the adoption of this technology at a greater rate and at the same time to tell the reader that which of these reasons of slower adoption of VoIP are most important for consideration and how those reasons or problems can be rectified. Finding out the major causes for the slower adoption of this technology and the suggestions for improving those causes or to rectify those problems provide the opportunity to the service provider of this technology that how the adoption of this technology can be improved. Keeping the above purpose in mind we have the following research question.

Research question 1: What are the causes for slower adoption of VoIP?
Research question 2: How the problems can be rectified
Research question 3: How can VoIP step forward by falling back the adoption obstacles?
By analyzing the above research question we can support and clarify the research problem.

CHAPTER-2

Literature review

2.0 Traditional Telephony Systems

2.1 Terminals and Access Network:

A local telephone operator builds the access network, consisting of local loops and access switches, and connects the access switches together to form the local carrier network. Various telephone operators connect their networks to form the global telephone network, and the subscribers can make local, long distance and international calls on reasonable charges. With these charges, telephone companies get their investments paid and their operations costs covered. In PSTN, signals in the local loop carry analogue signals. In most cases, analogue voice signal is converted to digital form in the local access switch, as shown in figure 2.1. In most countries, excluding developing countries and some regions in Eastern Europe, the telephone network is fully digital. [2]

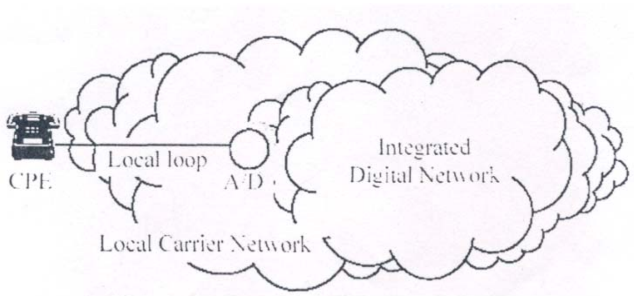

Figure 2.1 The local carrier network consists of local loops, access switches, local telephone exchanges and the transmission system between switches.

The analogue to digital (A/D) conversion is done in phases, as shown in Figure 2.2 in the next section. The G.711 PCM voice coding includes the following steps:

- The telephone network is designed to carry signals between 300 and 3400 Hz. To eliminate aliasing and noise, the frequency of the voice signal is limited with a low pass filter.
- Next, samples are taken from the analogue signal in fixed time intervals. The sampling theorem says, that to be able to reproduce a 3400 Hz signal, the sampling

frequency must be at least 6800 samples/s. 8000 1/s was chosen as the sampling rate, producing one sample every 125 us. The sampling rate is the same in all countries.

- The analogue samples are then expanded to make small samples near the zero level larger. The meaning of the expansion is to make the relative quantization noise even for all signal strengths. When the samples are compressed with a reverse law at the other end, the original signal can be reproduced. The European expansion/compression follows *A* law, while *A* law is used in North American networks.

- Now each expanded sample is converted to a digital number. The European system uses eight bits per sample, producing a bit stream of,
8000 samples/sec x 8 bit/sample = 64000 bits/sec = 64 Kbit/sec.

With eight bits, 256 distinguished levels can be coded. The MSB represents the sign of the signal, zero indicating a negative value.
Also the Northern American system uses eight bit sampling, but the LSB is once in a while borrowed for signaling, leading to a bit rate of 56 - 64 kbit/s. Both coding variations are defined in the CCITT standard G.711.[2]

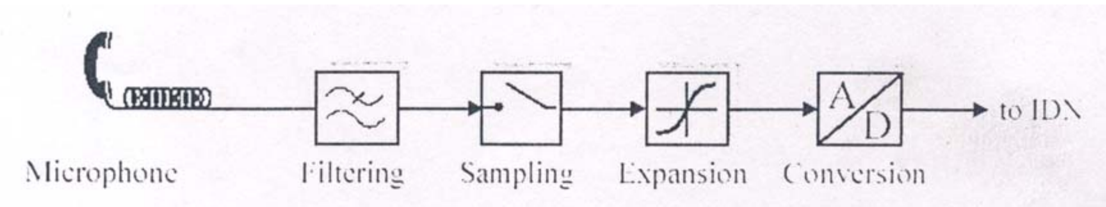

Figure 2.2: Analog to digital conversion and PCM voice coding.
PCM voice coding produces a 64 kbit/s bit stream, and it is transmitted in the digital switching and transmission system. IRL, the access switch also converts incoming digital numbers from the opposite direction into analogue signals to be reproduced in the ear piece. For two way operation, a 64 kbit/s Full-Duplex channel is needed in the Integrated Digital Network (IDN).[9]

Often companies and public organizations decide to buy or lease a Private Area Branch Exchange. The telephone sets are connected to the PBX and calls within the organization are switched locally with no charge. The PBX is connected to the PSTN, normally with physical El or T1 interfaces, and subscribers compete of the limited number for outside lines.

2.2 Telephone Switches

Telephone switches listen to the signaling requests, make a circuit through the telephone network and handle digital signals. At the end of the call, the terminal signals a disconnect message and intermediate switches release the circuit. Switches also handle counting and logging, the call log and summary information is kept at the local exchange and transferred to the billing system periodically.

Telephone switches form a hierarchical tree: concentrator multiplex local loops to a single trunk connected to a regional switch, and regional transit level switches cover a whole terrifying region. International calls are handled by international transit level switches and there are national transit level switches to handle calls between different telephone operators within a single country.

A block diagram of a modern digital telephone switch is presented in figure 2.3. Subscriber lines are normally concentrated into few trunks by concentrators, located near the subscribers. The switch has multiple E or T series trunk interfaces, which are used for concentrators, customer PBX's and for inter switch connections through the transmission system. The contents of the time slots are written to the RAM memory of the connection matrix or a Group Switch, and read from the right place at the right time to be sent to another interface. Reading and writing is controlled by the processor of the control part.[2]

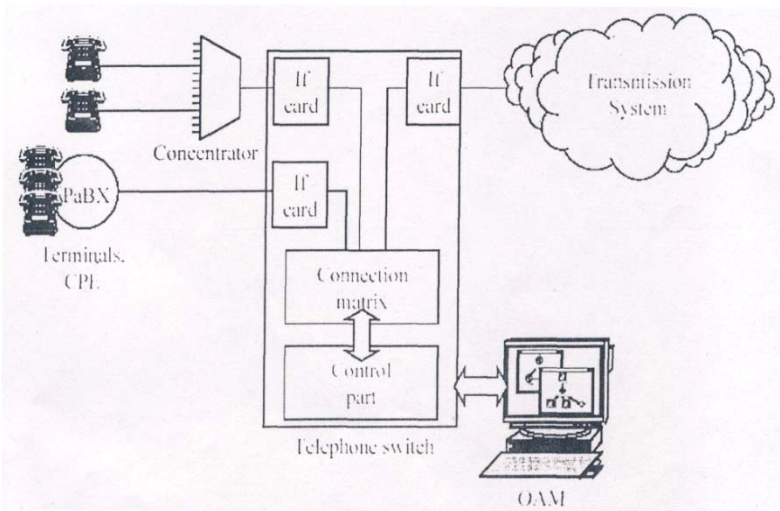

Figure 2.3: Block diagram of a digital telephone switch.

2.3 Transmission Procedure

Two switches are connected with a transmission system, which consists of the physical media and the multiplexers. For the physical connection, single mode fiber optic cables are often used, but also radio links and satellite connections have their role. The emitter sends an optical signal to the fiber, and the signal is converted into bits by a detector. The capacity of a SM fiber is enormous, so it should not be wasted on a single El or Tl channel. A multiplexer can take numerous channels and concentrate theft on a single high capacity trunk. Normally, time division multiplexing is used. SDH (Synchronous Digital Hierarchy) specifies a hierarchical system to multiplex lower capacity synchronous channels into a single trunk. Modern transmission systems include Wavelength Division Multiplexers and optical Cross Connect devices, which can multiplex and switch optical channels transparently. Circuit switched switching and transmission systems are dimensioned based on blocking. A call is blocked, if the network cannot make the connection. The refusal can be based on lack of switching or transmission capacity. According to the national regulations, the nominal blocking level on the public telecommunication network must be kept under 1 % for long distance trunks and under 0.2 % for feeder trunks to the long distance switches. The network is dimensioned to handle also the peak load. The amount of traffic during the busiest full hour is used as the dimensioning value. This can be read from the traffic statistics or estimated as a multiple of the characteristic traffic value and the number of subscribers. The network is dimensioned to offer a certain nominal blocking level, by looking, for the dimensioning value, the minimum number of switching elements that will lead to the nominal blocking level.

Erlang's first formula [2] gives the blocking probability of a call (under certain conditions), when the traffic intensity and the number of lines is known. The formula is the following:[1]

$$\mathrm{E}_1(n, A) = \frac{\dfrac{A^n}{n!}}{1 + A + \dfrac{A^2}{2!} + \cdots + \dfrac{A^n}{n!}}$$

Where E is the blocking probability, A is the traffic intensity and n is the number of lines. Often pre-calculated tables are used, giving the number of lines for a given blocking probability.

A significantly higher block level can be accepted, if a leased line capacity between two PBXs with a PSTN connection is dimensioned, because extra spill traffic can be redirected through the PSTN .If a regional office has a PBX or an interface module with 100 extensions and 50 % of the incoming and outgoing traffic is between the headquarters and the regional office, the dimensioning value for the inter PBX trunk will be 5 Erl, needing only 9 lines, with a block level of 5 % [2].

2.4 Quality of Service in Telephone Networks

A telephone user expects good quality service block free availability and good quality connection at low price. If unhappy, rationally anyone might select another carrier the next time. The following technical details are expected by the user of public fixed line voice services [2]

- High availability, meaning low blocking probability. The service is judged as successful, if the caller hears a ringing tone, indicating that the target telephone is alerting. Most of the unsuccessful call attempts are caused by network blocking. The network is redundant, so device faults, that prevent the service, are very rare.

- Low end to end delay. Delay disturbs interactive conversation, because parties may start to talk at the same time. ITU-T regulations state the upper limit of 150 ms for end to end delay for calls, which are not routed through a satellite link, and 400 ms for international lines. The delay is sensed subjectively, but normally delays over a few hundred milliseconds are regarded as disturbing.

- Low echo level and low echo delay, most often generated by an imperfect' hybrid, which converts the ^duplex local loop to two simplex IDN channels. With a speakerphone, the voice can leak also acoustically from the receiving

2.5 VoIP Architecture

The VoIP infrastructure is built as a virtual network for a company network and is called the VoIP Connectivity Layer. This is a Core Network which is surrounded by a

Multi-Service Access/Multi-Service Edge network that supports all popular access technologies including TDM, ATM, Frame Relay and Ethernet. Thus, the VoIP infrastructure can be reached via any of these access technologies. Moreover, the architecture provides capabilities to support various access protocols such as, H.323, MGCP, MEGACO, SIP, TDM/SS7, as well as any VoIP protocol may emerge in the future. This is achieved by surrounding the VoIP Connectivity Layer with Border Elements (BEs), which mark the trust boundaries of the VoIP Infrastructure and translate the specifics of various VoIP access protocols into Session Initiation Protocol (SIP) - the single common internal protocol used by all VoIP infrastructure components [4].

Voice over IP networks are complex. They represent the converging worlds of tele- and data communications, and therefore a typical VoIP architecture shown in figure 3.1 has presented. Some essentials for implementing VoIP are:

1. Integration to traditional telecom infrastructure
2. Integration to billing systems
3. Many add-on services
4. Large variety of protocols
5. Quality is an issue
6. Network specialists are expensive and scarce
7. Reliability is a must
8. Multiple High Quality Services: voice, fax, video, unified messaging, call centers.

A typical VoIP network includes the following components:

a. Media gateways
b. Signaling gateways
c. Gatekeepers
d. Class 5 switches
e. SS7 network
f. Network management system ,
g. Billing systems

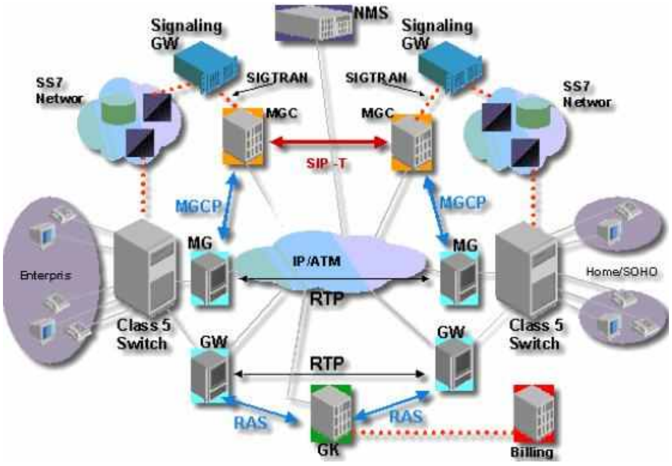

Figure 2.4: Typical VoIP Architecture. [4]

2.6 VoIP Layers

The VoIP architecture is devised in a series of Layers, as illustrated in Figure. The layers follow the principle of information hiding; that is, each layer has a well-defined role and provides a set of well-defined capabilities to the layer to which it is sending information, by using the capabilities of the layer it received information.[22]

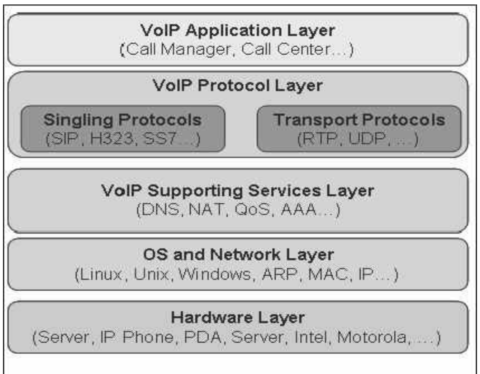

Fig 2: Layers of VoIP network

Figure 2.5: VoIP Layer

The Access Layer interfaces with customer equipment and provides customer connectivity using access-specific protocols.

The IP/MPLS Core Network Layer provides IP connectivity for all elements of the VoIP infrastructure.

The Connectivity Layer provides the VoIP infrastructure needed to process basic calls, support high performance network functions, send network primitives, provide media services, interact with Application Servers for more advanced calls, and support Call Detail Recording.

The Applications Layer consists of several Application Servers, each providing one or more services.

The Resource Layer provides an environment for the creation of service logic and the management of services including customer record maintenance and billing plans.

The Operations Support Layer consists of multiple applications, databases and a supporting data communications network, which are used by internal and external personnel to manage the VoIP network and its elements.

2.7 H.323 System

H.323 is an ITU-T standard for multimedia conferencing system and signaling in packet switched networks. It includes voice, video, and data conferencing, for use over packet-switched networks. The H.323 was first published in 1996 and the latest version (v5) was completed in 2003.

H.323 was the first standard for VoIP, defines five components of a multimedia network, figure 3.3 describes h323 architecture:

a) Terminals
b) Multipoint Control Units (MCUs)
c) Gateways
d) Gatekeeper
e) Border Elements

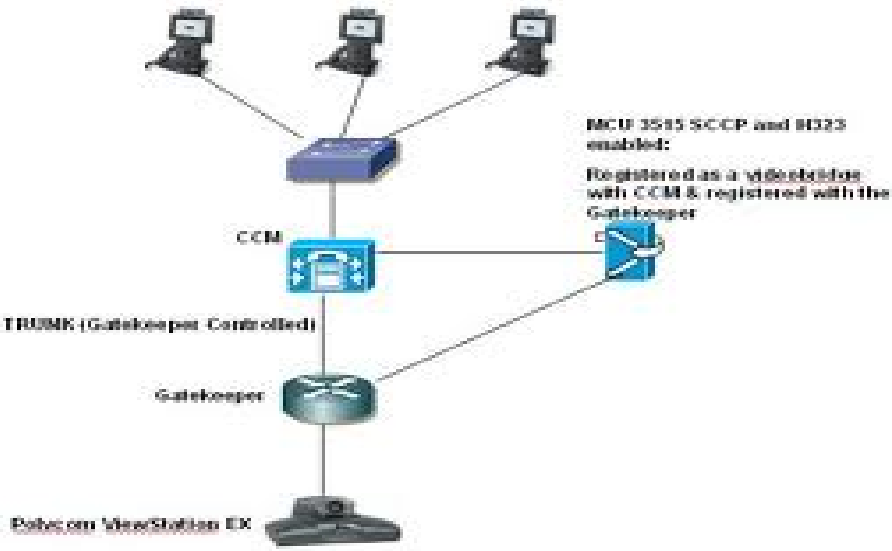

Figure 2.6: H.323 System Architecture.

There are five types of information exchange enabled in the H.323 architecture:[29]

1. Audio (digitized) voice
2. Video (digitized)
3. Data (files or image)
4. Communication control (exchange of supported functions, controlling logic channels)
5. Controlling connections and sessions (setup and tear down)

Terminals are telephone and PC equipment which connect end-users to the H.323 network.
MCUs are responsible for managing conferences. MCU's consist of a Multipoint Controller (MC) and an optional Multipoint Processor (MP). The MC manages signaling and the MP manages media mixing and switching.
Gateways interface the H.323 network with other networks, including PSTN (Public Switched Telephone Network) and other H.323 networks. Gateways consist of a Media Gateway Controller (MGC) and a Media Gateway (MG). The MGC is responsible for call signaling functions and the.MG is responsible for media-related functions.
Gatekeepers are responsible for admission control and address resolution. Gatekeepers are able to provide advanced services such as normally found in PBX's.
Border Elements are positioned between two H.323 networks and assist in call routing and call authorization.

2.8 H.323 System Terminals

An H.323 terminal provides a bidirectional audio, video or data conversation. All terminals must support audio, with the ITU-T G.711 PCM coding. If video is supported, the video terminal must include H.261 coding. Intelligent terminals may also include T.I20 data conferencing services, like a common whiteboard, file sharing and document sharing without common application software. The standard offers optional lower bit rate G.723.1 and G.729 audio coding and H.262, H.263 and

MPEG-4 video coding. Other optional parts are security, multipoint connections, camera control and channel aggregation.

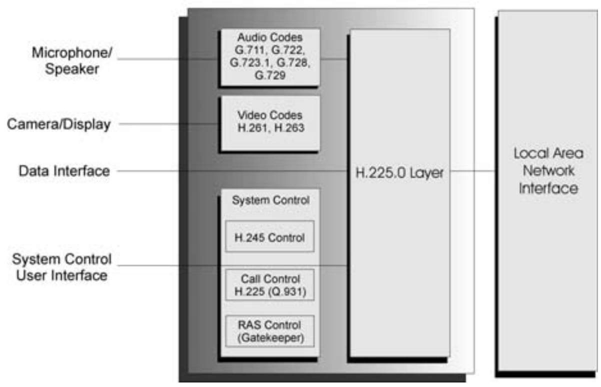

Figure 2.7: H.323 Terminal.

An example of H.323 audio terminal is a Cisco 7940 feature phone, which is attached directly to an Ethernet LAN. The unit gets the power from a separate mains adaptor or from a switch feeding DC power to the terminals with unused pairs. While a workstation may be attached to the internal switch of the IP telephone set, a single twisted pair link is enough for the workplace. The device includes a large LCD display, multiple feature buttons, a speaker and all common feature phone functions.[2]

2.9 H.323 System Gatekeeper

A gatekeeper is an optional H.323 component. If it exists on the network, all terminals must use it. A gatekeeper is responsible for its H.323, zone, consisting of terminals, gateways and MCUs.[29]

2.10 H.323 System Gateway

The gateway is also an optional H.323 component. It is only used, if the H.323 system is connected to a non-H.323 system, like H.320 ISDN video conferencing systems, PSTN or ISDN telephone networks or terminals, H.310 ATM video conferencing systems or V.70 audio & video systems.[29]

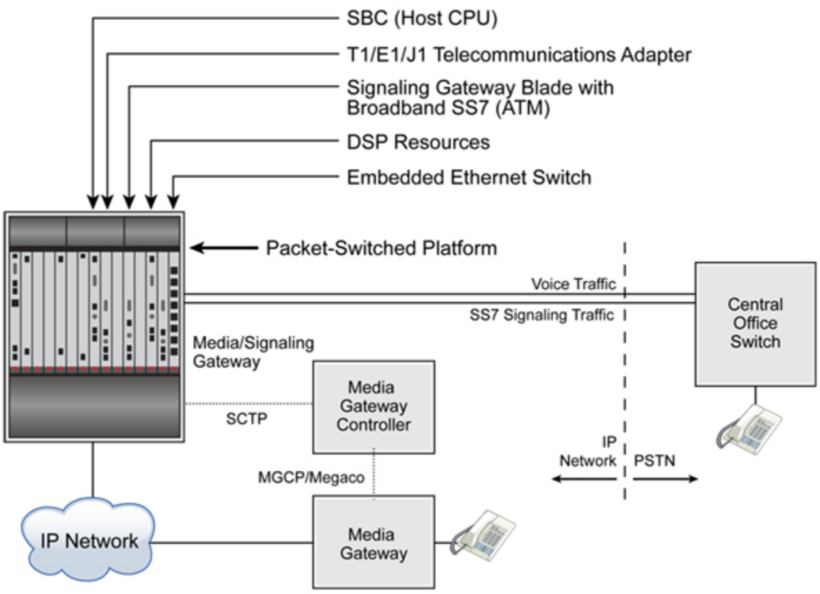

Figure 2.8: Gateway signaling functions. [24]

As shown in the figure 3.8, a gateway has two interfaces: an H.323 interface and a non-H.323 interface. The gatekeeper initiates a call to the PSTN, and the gateway sets up the call and provides signaling conversions (Q.931/H.245 to SS#7), as well as coding translations, if needed. The PSTN network sees the gateway as a digital PBX.

2.11 VoIP Protocols

H323 protocol is used, for example, by Microsoft Net meeting to make VoIP calls. This protocol allows a variety of elements talking each other:[11]

1. Terminals, clients that initialize VoIP connection. Although terminals could talk together without anyone else, we need some additional elements for a Scalable vision.
2. Gatekeepers, that essentially operates:

    a) Address translation service, to use names instead IP addresses
    b) Admission control, to allow or deny some hosts or some users
    c) Bandwidth management,
3. Gateways, points of reference for conversion TCP/IP - PSTN.
4. Multipoint Control Units (MCUs) to provide conference
5. Proxies Server also is used.

H323 allows not only VoIP but also video and data communications.

Concerning VoIP, h323 can carry* audio codec G.711, G.722, G.723, G.728 and G.729 while for video it supports h261 and h263.

It completes some missing UDP functions; which are critical to real-time applications. RTF offers timing information between the parties and detection of lost packets, and it returns the correct packet order, even after reordering by the network. An RTF message, shown in figure 3.13, consists of a header and a data field and holds the following fields:[11]

- RTF protocol version, currently version 2. This field is two bits long.
- P and X bits, indicating the existence of padding or an extension header. If padding is added, the last padding byte holds information about the length of the padding, which should be removed by the receiver.
- CSCR Count (Contributing Sources), which indicates the number of CSCR identifiers after the fixed header. The CSCR count field is four bits, making it possible to have up to 15 contributing sources.
- A Marker bit, which can be used to mark important events. The precise interpretation of the marker is defined by a profile.
- Payload Type, containing seven bits of information about the RTF user data media and coding.
- Two byte Sequence Number, which presents the number of the RTF message. The initial value is random, and each message increments this value.
- Four byte Timestamp, holding information about the time of the first sample in the message. The initial value of the timestamp is random and it will be incremented linearly. For multimedia conferencing, successive RTF messages can hold equal timestamp values.

- Four byte Synchronization Source Identifier for the source. This value is also random, to ensure the uniqueness of the synchronization sources during an RTF session.
- A 0 - 15 word list of Contributing Sources, if any. Multiple contributing sources are used by an RTF mixer, which resynchronizes converts and combines packets from various incoming RTF sessions.

- Variable length data, containing voice or video information.

Figure 3.12: An RTF message, carrying low bit rate video. The RTF messages don't use any standard TCP port, but the parties agree on the port during session negotiation. A free unassigned port with an even number is used in both directions, and the next odd port is used for RTCP. Different RTF sessions and ports are used for different media, like voice and video, between the same parties. Each source sends RTF messages, which are not acknowledged.

RTF offers means to detect lost or reordered packets, but it doesn't ensure timely delivery of packets or guarantee any Quality of Service. The unreliable UDP transport only detects bit errors on the UDP header, but doesn't offer any reliability or QoS guarantees. To ensure sufficient quality, external solutions, like prioritization or service differentiation must be used. [11]

2.12 RTCP

Real-time Transport Control Protocol (RTCP) is used for providing feedback on the RTF transport, for RTF source identification and for observing the number of participants. Optionally, RTCP can also carry minimal session control information, like participant identification, which will be displayed on the user interface. The RTF sources use random and unique CSCR numbers, but to be able to synchronize different streams on a single application, a distinguished CNAME (Canonical Name) is distributed by the RTCP. The RTCP message rate is adapted to the bandwidth usage and to the number of participants, but the total bandwidth usage should be 5 % of the RTP bandwidth. [2]

2.13 Voice and Video Coding Protocols

Voice Coding

Traditional analogue telephone network carries frequencies from 300 to 3400 Hz. The digital PSTN takes 8000 samples a second and codes every sample with eight bits, after nonlinear decompression. The G.711 PCM coding [Appendix C] produces a continuous bit stream of 64 kbit/s and offers good voice quality.[12]

G.711 is the international standard for encoding audio on a 64 kbps channel. It is a pulse code modulation scheme operating at 8 kHz sample rate, with 8 bits per sample. There are two variants of the G.711 protocol: ,4-law and //-law. The difference between the two is that *A*- law represents smaller signals with greater fidelity. , 4-law is the standard for international circuits. The /v-law is usually used in North America and Japan and, 4-law is in the rest of the countries.

Terminal suppliers may provide other audio compression algorithms aside from G.711. Other protocols such as G.722, G.728, G.729, MPEG1 audio, and G.723.1 may also be used. H.323 has also provided a new audio algorithm: G.723. G.723 is a dual-rate speech-coding standard. But this standard is not required in our project. Table-1 shows different audio codec along with their algorithm bit rate, bandwidth and their application. Among all these codec only G.711 and G.729 is supported by our devices that we will use in our project. [11]

Table-2: Audio Codec

ITU Standard (year approve d)	Type of Algorithm	Bit Rate (kbit/s)	Bandwidth (kHz)	Typical End-To-End Delay (ms) (excludes channel delay)	Application
G.711 (1977)	PCM	48, 56, 64	3	«1	GSTN telephony, H.323 & H.320 videoconferencing

G.723. 1 (1995)	MPE/ACELP	5.3, 6.3	3	67-97	GSTN video-telephony, H.323 telephony
G.728 (1992)	LD-CELP	16	3	«2	GSTN, H.320 Videoconferencing
G.729 (1995)	ACELP	8	3	25-35	GSTN telephony, wireless/PCS/FPLMTS, GSTN modem DVSD V.70, H.324 GSTN videophone
G.729 annex A (1996)	ACELP	8	3	*t* 25-35	GSTN modem DVSD V.70, H.324 GSTN videophone
G.722 (1988)	Sub band ADPCM	48, 56, 64	7	<2	ISDN telephony/videoconferencing, commentary audio

The G.723 is an ITU-T standard, which uses ADPCM (Adaptive Differential Pulse Code Modulation), and produces 24 kbit/s or 32 kbit/s bit stream with quite small quality degradation. As shown in figure 3.14, a signal channel exists between the transmitter and the receiver. Waveform coding produces shown in figure 3.14 is a constant bit stream, but voice samples can be stored in a transmission buffer and a suitable cluster of samples is sent on an RTP message. [17]

2.14 Signaling Protocols

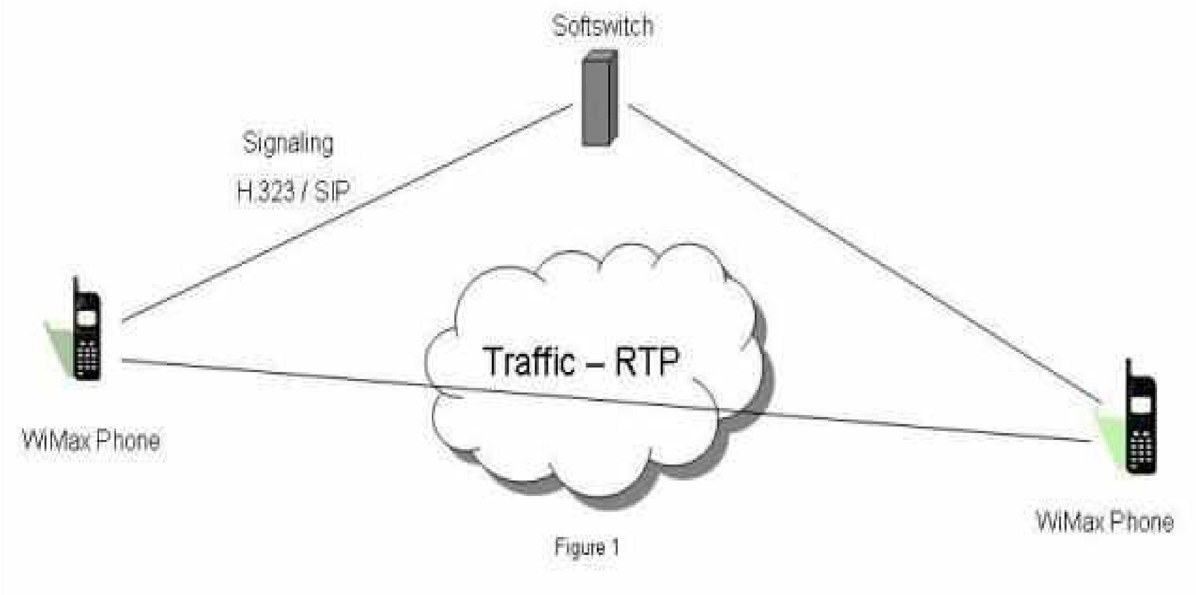

Figure 2.9: Signaling Protocol

The signaling protocols H.323, SIP are used to setup the route for the transmission over the IP network, the Gateway protocols like the Media Gateway Control Protocol are used to establish control and status in the media and signaling gateways. Routing (UDP, TCP) and transport protocols (RTP) are used once the route is established for the transport of the data stream as shown in Figure [10]

H.323 Signaling Protocols

H.323 is the ITU-T standard for packet based multimedia communication, though originally developed for multimedia conferencing over LAN's it was later modified for VoIP as well. With versions coming out in 1996 and 1998, the standard has faced stiff competition from the other protocol SIP which was specifically designed for VoIP, but is more used because of its wide existence in the already installed networks. The standard is interoperable and has both point to point and multipoint capabilities. H.323 uses a number of other sub protocols for the various functions.

- H.255.0 - Registration, Admission, Status, Call Signaling, Control
- H.245 - Terminal Capability Exchange, Media Description, Control of Logical Channel

Also H.323 offers specifications for call control, channel setup, codecs for the transmission of Real time video and voice over the networks where the QoS and guaranteed services are not available. For the transport RTP is used for real time audio and video streaming. [10]

Session Initiation Protocol (SIP)

SIP is the IETF standard for VoIP signaling. It is based on the existing protocols like SMTP and HTTP, and uses a text based syntax that is comparable to HTTP uses in web addresses. A web address is comparable to a telephone number in a SIP network, also the PSTN phone numbers are also compatible in a SIP network ensuring interfacing with PSTN systems. SIP also provides a mobility function to the users. SIP also supports multiple media sessions during a single call hence users can - share a game, use instant message (IM), and talk at the same time. SIP works with most protocols like RTP, Session Description Protocol (SDP), Session Announcement Protocol (SAP). A lot of other protocols are also needed when it comes to the transport and signaling with the PSTN networks - RSVP, LDAP, RADIUS. SIP works on a client server architecture, where the clients are referred to as User Agents (UA). UA interact with the server mainly through a PC with a telephony agent or IP phone. The servers are of four types:

- Registration
- Redirect Server
- Proxy Server
- UA server

The SIP messages used for communication between the client-server are INVITE, ACK, OPTIONS, REGISTER, CANCEL, and BYE. [10]

2.15 Challenges Faced

Voice Quality

Presently, the telephone companies are losing so much of their business to the VoIP counterparts, but still the speculation exists on the kind of voice quality that VoIP service providers would give to customers. The techniques used to measure the voice quality of a VoIP call are the Mean Opinion Score (MOS) and Perceptual Speech Quality Measurement (PSQM)[10]

- **MOS**

  MOS follows the measurement techniques specified in ITU-T P.800, where different people are made to listen the voice signals and are made to rate the factors like distortion, delay, echo, noise etc on a scale of 1 to 5 where 1 is the minimum and 5 being the maximum. The mean opinion score is then calculated. A value of MOS 4 is considered as the toll quality. The conventional codec for fixed line telephones G.711 has a MOS of 4.0 at 64kbps. The modified codecs mainly used for VoIP are G.729 and G.723. G.729 is widely used for VoIP because of its low bandwidth requirements, it has a MOS value of 4.0 operates at 8kbps. The G.723 which is mainly used for Video Telephony, has a MOS 3.8 and operates at 5.3 /6.8 kbps [10]

- **PSQM**

  Based on ITU-T P.861 standard, this technique uses artificial speech, to provides numeric values of approximate speech intelligibility taking into account effects such as noise, coding errors, packet reordering, phase jitter, and excessive bit error rate. PSQM=0 signifies no impairment, while PSQM=6.5 indicates that the signal is totally unusable. Although PSQM values do not have any direct correlation with MOS values, but roughly PSQM value of 0 corresponds to MOS of 5 and PSQM value of 6.5 to MOS of 1.

The other factors affecting the voice quality are:

- **Delay**

  Voice transmission over wireless brings along with a it a big problem of packet delay or latency. The factors that add to the delay are the Propagation delay, the serialization delay, channel coding delay at the physical layer and the Medium access delay at the MAC layer. Similarly at the Network layer the Forwarding and the Queuing delays are encountered, and at the application layer Packetization / depacketization, delay, Algorithmic delay and look-ahead delay, decoding delay are inherently caused. Studies have proved that packet delay of 100ms doesn't do any harm, but if the delay increases to 150ms the voice signal is unusable. The service providers have to ensure that the delay caused does not exceed 100ms by any ways. Table 1 shows the various CODECs along with their MOS and delay time. [1]

CODEC	Bit Rate Kbps	MOS	Delay ms
G.711	64	4.5	0.125
G.723.1	5.3 6.3	3.6 3.98	30
G.726	16-24-32-40 Most commonly used 32	4.2	0.125
G.728	16	4.2	2.5
G.729	8	4.2	10

Table 1

Table 3: Various CODECs along with their MOS and delay time

- **Packet Loss/ Dropped Packets**

Packet loss does excessive damage to the voice signal, as retransmission cannot be considered as an option while transmitting voice. Loss of voiced frames at unvoiced/voiced transition causes significant degradation of the signal. Advanced error detection and correction algorithms are used to fill the gaps created by the dropped packets. A sample of the speaker's voice is stored and is used to create a new sound from an algorithm which tries to approximate the contents of the dropped packets or lost packets. [1]

- **Jitter**

when the transmission times of the arriving packets varies as a result of different queuing times or different routes it is referred to as Jitter. Jitter can be taken care of by using an adaptive jitter buffer which adapts itself according to the delay encountered over the networks, to provide a smooth voice stream at the output [1].

CHAPTER-3

SoftSwitch

3.1 Overview

A **Soft switch** is a central device in a telecommunications network which connects telephone calls from one phone line to another, entirely by means of software running on a computer system, typically the internet. This work was formerly carried out by hardware, with physical switchboards to route the calls. Although the phrase Soft switch is sometimes used to refer to any such device, it is more common to use the term Soft switch to refer to a device that handles IP-to-IP phone calls, while the phrase "access server" or "media gateway" is used to refer to devices that either originate or terminate traditional "land line" (hard wired) phone calls. Often times, in practice, such devices can do both. As a practical distinction, a Skype to Skype phone call is entirely IP (internet) based, and so uses a softswitch somewhere in the middle connecting the origating caller with the receiving caller. In contrast, access servers might take a mobile call or a call originating from a traditional phone line, and then convert it to IP traffic, then send it over the internet to another such device which terminates the call by reversing the process and converting the IP call back to digital or analog and connecting it to a destination phone number.[17]

A Softswitch is typically used to control connections at the junction point between circuit and packet networks. A single device containing both the switching logic and the switching fabric can be used for this purpose; however, modern technology has led to a preference for decomposing this device into a Call Agent and a Media Gateway.

The Call Agent takes care of functions including billing, call routing, signalling, call services and so on and is the 'brains' of the outfit. A Call Agent may control several different Media Gateways in geographically dispersed areas over a TCP/IP link.

The Media Gateway connects different types of digital media stream together to create an end-to-end path for the media (voice and data) in the call. It may have interfaces to connect to traditional PSTN networks like DS1 or DS3 ports (E1 or STM1 in the case of non-US networks), it may have interfaces to connect to ATM and

IP networks and in the modern system will have Ethernet interfaces to connect VoIP calls. The call agent will instruct the media gateway to connect media streams between these interfaces to connect the call - all transparently to the end-users.

The Softswitch generally resides in a building owned by the telephone company called a central office. The central office will have telephone trunks to carry calls to other offices owned by the telecommunication company and to other telecommunication companies (aka the Public Switched Telephone Network or PSTN).

Looking towards the end users from the switch, the Media Gateway may be connected to several access devices. These access devices can range from small Analog Telephone Adaptors (ATA) which provide just one RJ11 telephone jack to an Integrated Access Device (IAD) or PBX which may provide several hundred telephone connections.

Typically the larger access devices will be located in a building owned by the telecommunication company near to the customers they serve. Each end user can be connected to the IAD by a simple pair of copper wires.

The medium sized devices and PBXs will typically be used in a business premises and the single line devices would probably be found in residential premises.

At the turn of the 21st century with IP Multimedia Subsystem (or IMS), the Softswitch element is represented by the Media Gateway Controller (MGC) element, and the term "Softswitch" is rarely used in the IMS context, rather it is called AGCF (Access Gateway Control Function).[17]

3.2 VoIP SoftSwitch features:

- A highly scalable SIP and H323 Softswitch with integrated billing
- Class 5 features included, follow-me, call forwarding, Voicemail and more
- Video support
- A Calling card platform with IP IVR including pin or pin-less scenarios
- Callback solution supports all types of triggering methods: SMS, missed-call, web Callback

- Web Callshop interface
- Web self-care portal for end users and support for Online payments
- SMS features for both wholesale and retail services, using HTTP, SIP and SMPP protocols
- Mobile SIP for Symbian, Windows Mobile, IPhone, Android and BlackBerry users
- Windows communicator with Instant Messaging, SMS, Voicemail and many other modules
- Multi-tenant, multi-user IP PBX platform for offering hosted services
- Instant messenger functionality, XMPP protocol support, presence, chat, file transfer
- Unified messaging, Voicemail/SMS forwarding to SMS or email, Voicemail transcriptions.
- Calls recording.[17]

3.3 Resellers

Reseller's module is an additional module that enhances VoIP Switch services by enabling reseller accounts structure.

As shown on a picture below the resellers are divided on three levels with different rights. The system enables for selling VoIP services through network of service distributors/agent, which can provision their end-users themselves through the web interface which is part of the module.

The module architecture allows resellers to have own, unique Web Portal module, Online Shop and branded Soft phone. Thus, they become in fact virtual VoIP providers based on one, central VoIP Switch; with possibility to manage the Web Portal (under their own domain name) and integrate with own online payment processors (e.g. PayPal accounts)

The billing system behind the reseller structure is based on separate rates sheets associated with each reseller account. The resellers can create own rates sheets for their users or sub-resellers. [13]

Main characteristics of the Resellers module:

- Multi-level structure
- Web-based management interface
- Active calls monitoring
- Advanced reporting
- Multi-currency
- Multi-lingual
- Own, branded Web Portals for resellers
- Own E-Shop account
- Branded softphone (Vippie)

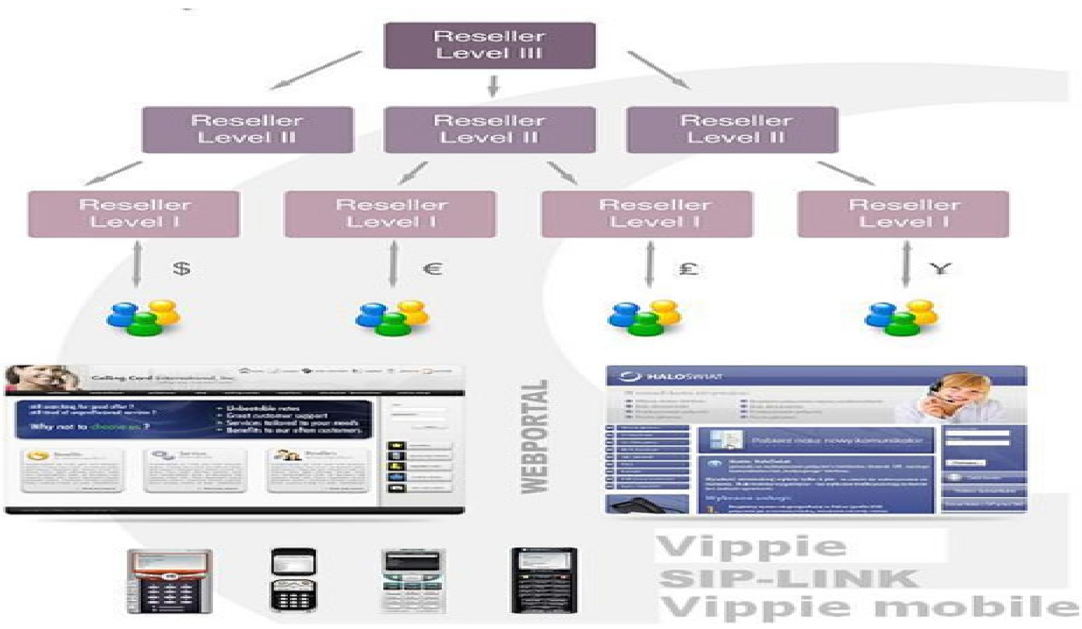

Figure 3.1: Resellers in order

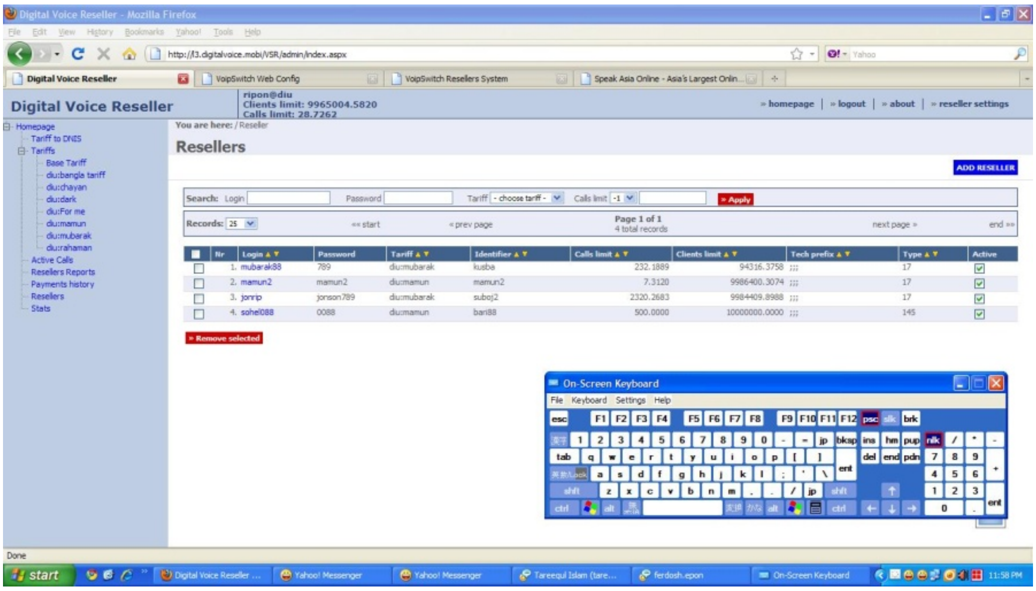

Figure 3.2: A reseller overview of digital telecom

3.4 Call shop

The Callshop module is an additional module, it is intended for Callshops facilities, where customer can step in and make international phone calls. Phones are usually installed in cabins (booths). After making a call or many calls, the customer comes to the cash desk where he receives a bill for made connections. In this scenario, regular phones are replaced by VoIP clients which can be for example multiple lines FXS gateways, SIP/H323 IP phones (adapters) or even softphones installed on personal computers (for example in internet cafes).

The whole communication is carried out over VoIP directly between the end-points in Callshop and VoipSwitch.

The Callshop module consists of two parts:

- Server side Callshop listener integrated in VoipSwitch main application
- Web flash interface; calls status is shown in real time.[17]

3.5 VoIP Tunnel

The VoIP Tunnel technology has been developed by VoipSwitch in order to enable making and receiving VoIP calls for users who are behind firewalls that block VoIP traffic, especially in countries like UAE, Oman and some other which have recently decided to delegalize internet telephony.

The VoIP Tunnel is part of the main package thus it's granted with the purchase of VoipSwitch main platform. The VoIP Tunnel reduces the number of ports needed for VoIP communication to only one. The VoIP Tunnel is compatible with any hardware or software SIP clients. It is also embedded in our softphones, both SIPLink and Vippie. The communication between the client and the VoIP Tunnel Server can be either in TCP or UDP protocol and can use any port (to be specified by VoipSwitch administrator).

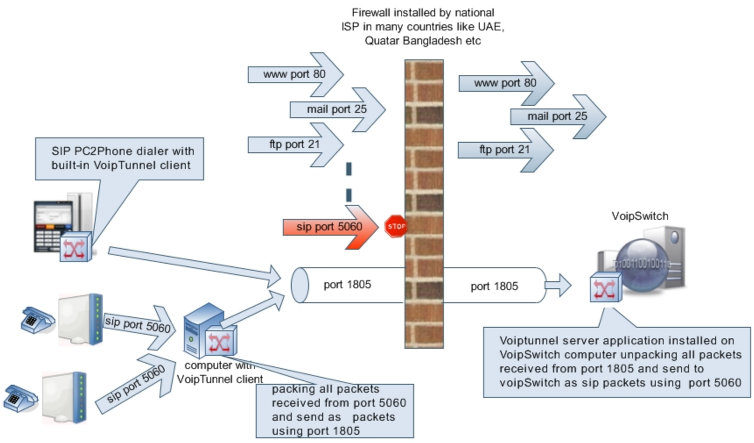

Figure 3.3: VOIP tunnel system

3.6 Web Portal

Web Portal is a web interface that is offered with the VoIP Switch main platform and consists of two parts:

- Web interface for logged users
- Admin part

The admin part provides advanced tools for managing the entire portal's appearance and functionality, delivering ready to use objects that enables users to utilize fully all the features offered by VoIP Switch. The Web Portal is offered as a part of the VoIP Switch package and comes with a set of exemplary design templates. As an optional service we provide also custom-made designs.

The main characteristics of the Web Portal include:

- Customizable web interface for all types of end-users
- Admin section with built-in tools for modifying the portal's look
- Predefined web components e.g. hot rates box, rates, sign up…
- Contacts (phone book) stored on the server side and accessible from the web as well as the mobile and PC SIP clients.
- Calls history with export to various formats (Excel, XML, PDF and others)
- Payments history
- Invoices
- User's profile (Address information, special settings)
- Auto-recharge on low account balance or for monthly payments (Monthly plans, DIDs, E911 and other special services)
- Call forwarding management, conditional follow me/find me
- Voicemail management, personalized welcome greetings dependent on the caller ID
- Missed calls details
- Web callback/bridge call applet with real-time call status messages (connect two service)
- Virtual numbers management with automatic orders placing procedures with support for DIDs providers APIs (*Voxbone*, *DIDX*), also support for DID numbers stored in local database
- Registering phone numbers (caller IDs) for calling cards or callback services
- Sign-up and recharge functions, support for credit cards, PayPal, moneybookers…
- Recharge by vouchers/PINs
- Online shop integration for selling goods online, multilingual interface support
- Instant messenger (Jabber) with video conferencing
- Web based softphone integrated with the Contacts
- SMS module allowing for sending and receiving short text message

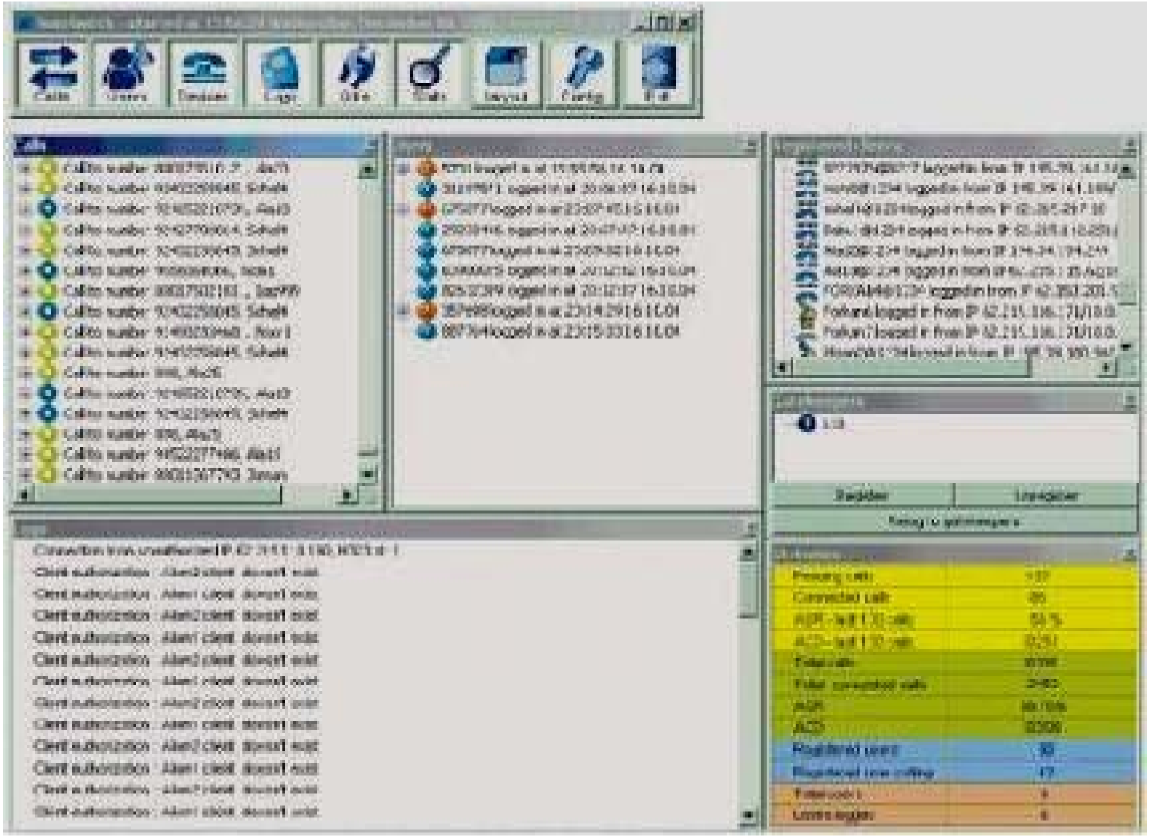

Figure3.4: VoIP Softswitch

3.7 The VoIP Soft switch for starting ITSPs

IP Softswitch is a platform that allows to implement various types of Voice Over the Internet Protocol (VOIP) services, with retaining shared, uniform management interface. The feature that distinguishes this platform is the implementation of an integrated, embedded billing system that cooperates with SQL – MS SQL or MySQL databases' servers.

This solution results in the simplicity of preparing the system, by the operator, to be fully functional for providing services and also for administrating it in the future.

Softswitch is the main element of the platform, which merges the functionality of the following VOIP architecture's elements:

- H323 switch
- H323 gatekeeper

- SIP Proxy
- SIP registrar

Each of the described elements can operate simultaneously with the others. Moreover, the clients, regardless of the protocol, or the way they transfer connections, can connect between one another. This option allows connecting the networks, which because of the differences in implemented protocols or dialects inside the particular protocol, cannot directly transfer connection between one another. Implementing VoIP Switch as a central traffic controller also introduces a number of additional management, supervision and network security facilitations. Interop tested with audio Codes Mediant1000 and Mediant 2000 lines of Digital gateways.

3.8 VSM Management system

VoIP switch Real-time monitor:
Real-time monitoring, live calls information, statistics at a glance (number of connections, connected calls, registered users, ASR/ACD/ PDD and others)

- Registered users details, gatekeepers/sip registrars status, system logs
- Debug functionality for selected connections

VoipSwitch Manager (VSM) – windows graphical interface for managing the entire platform.[30]

Figure3.5: VoIP Softswitch Manager

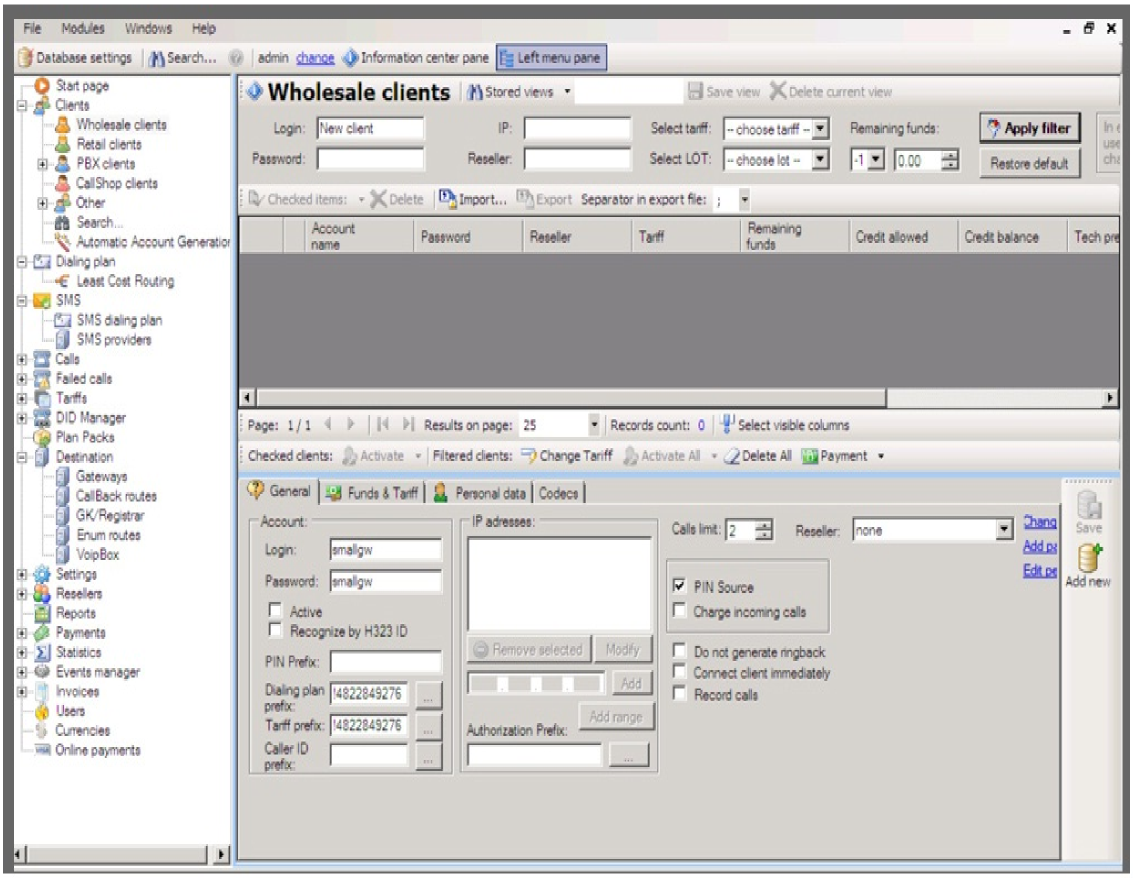

Figure 3.6: Wholesale Clients Manage by VSM

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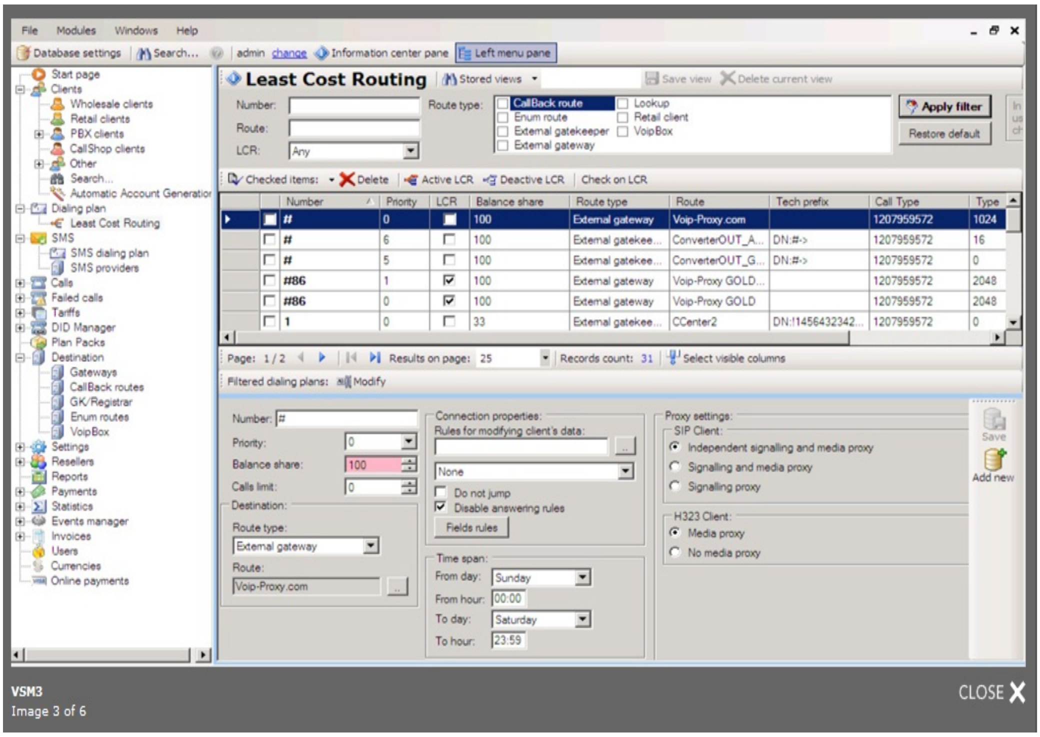


Figure 3.7: Cost Manage by VSM

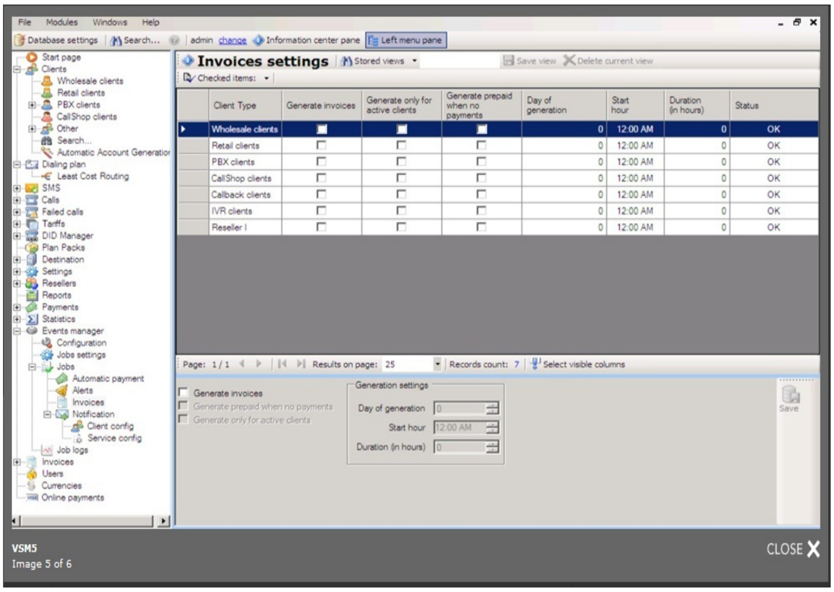


Figure 3.8: Invoice Settings by VSM

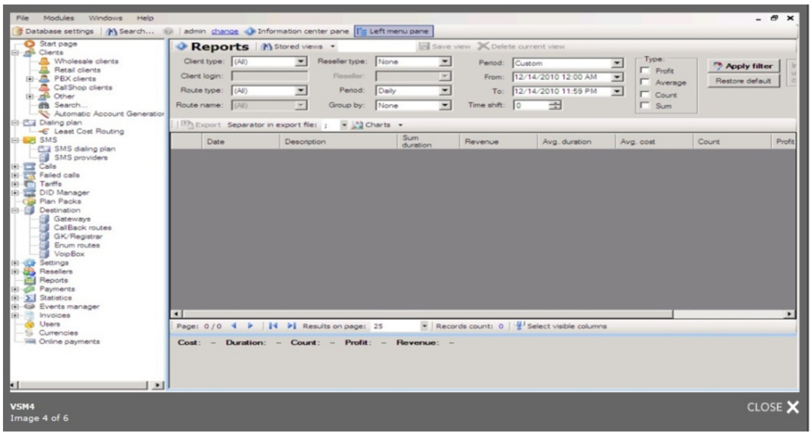


Figure 3.9: Report makes by VSM

3.9 The main functionality of VSM

- Managing user's accounts – Divided on types that limit users to particular services (callback, calling cards, wholesale, broadband calling clients, Callshops), also support for one account for all services.
- Managing rates sheets.
- Dialing plan.
- Managing resellers accounts.
- Protocol settings.

- Alerts, system services, recurring payments, special tasks, watchdog.
- Advanced reporting, detailed information on the traffic, CDRs, export to various formats, filters (per route, destination, client, date/time, duration, cost etc.).
- Statistics – ASR, ACD, PDD and other parameters.
- Balance reports.
- Payments management.
- Invoicing.
- PIN generator.

CHAPTER-4
Methodology

The purpose of this chapter is to provide reader a brief introduction to the research approach and methods. A method is a tool, a way to solve a problem and research new knowledge. The procedure of the research will be presented.

4.1 Research Purpose

The purpose of this study is to provide the opportunity to the reader that how the purpose of VoIP can be improved in Bangladesh and at the same time it will also tell the reader that which reasons of slower adoption of VoIP are most important for consideration and how those reasons can be rectified. The purpose with research is to state what is to be accomplished by conducting research and how the results of the research can be used to improve the VoIP adoption in Bangladesh.

4.2 Research Approach

Different approaches can be taken such as deductive or inductive and quantitative and qualitative approach. Deductive research starts with existing theories and concepts and formulates hypotheses that are subsequently tested; its advantage point is received theory. Inductive research starts with real-world data, and categories, concepts, patterns, models, and eventually, theories emerge from this input. After the initial stages, all types of research become iteration between the deductive and the inductive. This is sometimes referred to as adductive research.

We started our research by studying existing theories relating to our problem area which will be later compared with reality. So, our research is mostly deductive.

The quantitative approach is also characterized by studying few variables on a large number of entities. To find answers to its research problem, this is normally done in a broad sense by using surveys with already set answering alternatives. Furthermore, this approach is considered especially useful when conducting a wide investigation that contains many units.

Characteristics of qualitative studies are that they are based largely on the researcher's own description, emotions and reactions. The qualitative approach also includes a great closeness to the respondents or to the source that the data is being collected from. It is characterized by gathering abundant information and to investigate several variables from a few numbers of entities. To make use of the possibility to gather high quality data, the most common way to do this is with the use of case studies and interviews where no set answering alternatives are being offered.

As our purpose of this study is to provide reader a better understanding of the reasons of slower adoption of VoIP and then how that adoption can be improved. So, we believe
that both the qualitative and quantitative approaches were found to be more suitable for the purpose of this report.

4.3 Research Strategy

The research strategy conditions are

- The type of research question posed.
- The extent of control an investigator has over actual behavioral events.
- The degree of focus on contemporary, as opposed to historical, events.

The first and most important condition for differentiating among the various research strategies is to identify the type of research question being asked. A basic categorization scheme for the types of questions is the familiar series: "who", "what", "where", "how", and "why", "How" and "why" questions are more explanatory and likely to lead to the use of case studies, histories, and experiments as the preferred research strategies.

A case study is an empirical inquiry that investigates a contemporary phenomenon within its real-life context, especially when the boundaries between phenomenon and context are not clearly evident. The case study allows an investigation to retain the holistic and meaningful characteristics of real-life events. A case study can involve a single and a multiple-case study. The single case study makes an in-depth investigation regarding only one entity but in multiple-case study two or more entities are being investigated which gives the opportunity of comparisons.

We have chosen a multiple case studies as our research strategy and also survey. This is more appropriate for our study because it fits to achieve our goals of this report. This strategy also helps us in comparing different cases.

4.4 Data Collection

The data that will be collected is expected to be mainly of a qualitative nature, due to the chosen units of analysis. Documents are important in the data collection stage in a case study, due to their overall value. However, care must be taken in the interpretation of documents, since they are often prepared for another purpose and audience than that of the case study.

The data collection methods that will be used for this research are interviews and documentation. For this research, interviews will be performed by emails, telephone calls and may also in person. We will collect the data from different companies that are providing VoIP services and also from the end users.

To collect data from companies [NIRBAIN VOICE, operation code 0630,SIP dialer, DIGITAL VOICE] and end users, we will build a questionnaire website and will email web link to companies to participate in the interview. To build an Internet based dynamic questionnaire, we will use Dreamweaver questionnaire, PHP (hypertext processor) as a scripting language and MySQL as database. The web-based questionnaire would be simple and consists of both open and close-ended questions.
We will approach data collection technique through personal contacts from MSN and yahoo friend list. Furthermore there is no restriction of age and gender of the individual end users which we are selecting for the data collection of our research.

4.5 Data Analysis

Data analysis consists of three concurrent flows of activities. These three are data reduction, data display, and conclusion drawing and verification.

- Data reduction should not be considered to be separate from analysis, but as a part of it. This reduction of the data is analysis that helps to sharpen sort, focus, discard, and organize the data in a way that allows for "final." conclusion to be drawn and verified. Data can be reduced and transformed through such means as selection, summary, paraphrasing, or through being subsumed in a large pattern.

- Data display is the second major activity which the research should go through, and this means taking the reduced data and displaying it in an organized, compressed way so that conclusions can be 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 humans' information-processing capabilities". It is further explained that good display are, "a major avenue to valid qualitative analysis". In conclusion, with data reduction, the creation and use of display is not separate from analysis, but it a part of it.

Conclusion drawing and verification is the final analytical activity for the qualitative researcher. It is here that the researcher begins to decide what things mean. They do this by noting regularities, patterns, explanations, possible configurations, causal flows, and propositions. (Miles & Huberman, 1994) However, Miles and Huberman (1994) also add that competent researcher should hold such conclusions lightly, while maintaining both openness and degree of scepticism.

Above three steps will be included in the data analysis. So, after reducing the data, only relevant data is discussed. As this research is a multi case study, then the relevant data is displayed for comparison of different case studies. In the end conclusions and results will be drawn.

4.6 Validity and Reliability

Validity and reliability helps to measure the research and add strength to the findings. Validity is the most important requirement on a measurement instrument. Three sorts of validity need to be considered. there are three forms of validity: construct validity, internal validity, and external validity.

- Construct validity: establishing correct operational measures for the concepts being studied.
- Internal validity: establishes a causal relationship, whereby certain conditions are shown to lead to other conditions, as distinguished from spurious relationships.
- External validity: establishing the domain to which a study's findings can be generalized.

Reliability is necessary for validity and is easier to achieve than validity. Although reliability is necessary in order to have a valid measure of a concept, it does not guarantee that a measure will be valid. It is not a sufficient condition for validity.

every case study project should strive to develop a formal, presentable database, so that, in principle other investigators can review the evidence directly and not be limited to the written reports. In this manner, a case-study database markedly increases the reliability of the entire case study. For case studies, notes are likely to be the most common component of a database. The notes may be a result of an investigator's interviews, observations, or document analysis.

In order to increase the validity of this research e-mail request are going to send to the company’s and end-user to participate the survey through the application in advance about the matters that are going to be discussed. To increase the external validity and replication logic in multiple-case studies, an survey guide is developed and followed through the study. To make sure that we are contacting right person who has good knowledge over the technical and non-technical aspects of the VoIP technology. We are going to survey the systems administrator, chief executive officer, marketing manager etc.

CHAPTER-5

Experimental Study & Data Analysis

5.1 Experimental Study

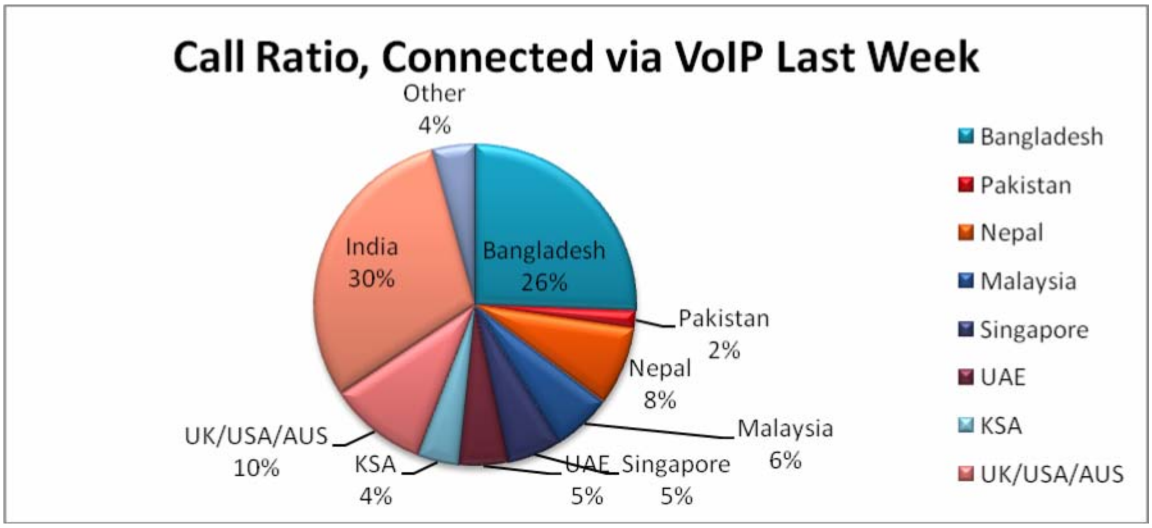
In this chapter we will be presenting the experimental data from individual end users and also from VoIP companies [NIRBAIN VOICE Telecom, operation code 0630, SIP dialer, products of www.e-softbilling.com/products, DIGITAL VOICE Telecom, operation code 81485,ITEl dialer product of www.revesoft.com]. General characteristics of the individual end users are presented and also each VoIP company is first introduced. Afterwards in the next chapter, we will present the data collected from both individual users and VoIP companies and at the same time we will do the analysis part in respect to our research question.

5.2 **experimental data from provider companies'**

We collect raw data from NIRBANVOICE VOICE and DIGITAL VOICE, we collect one week calls data from VSM call report, collect a hole day call depend on time ,collect client time data ,got the VoIP rate and highest and lowest duration of call and call PDD etc.

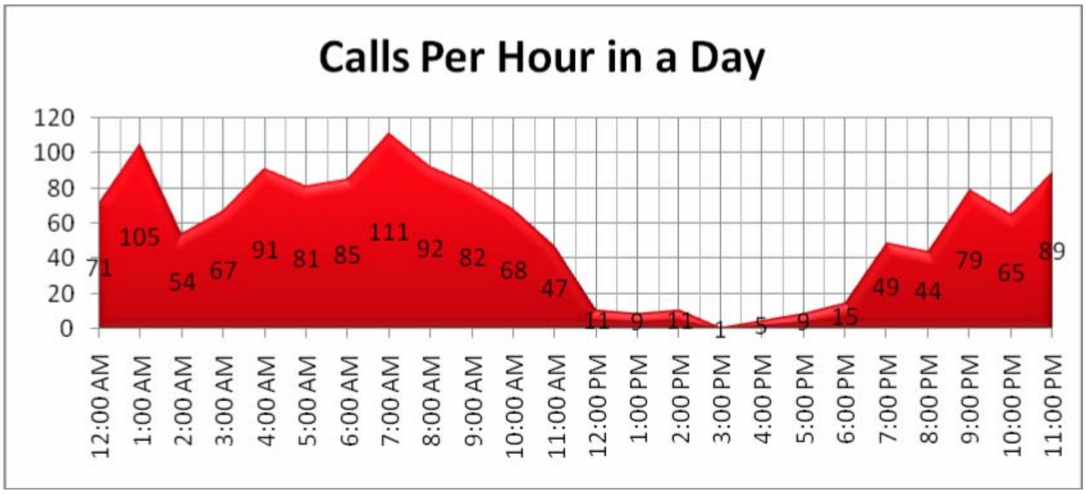
We process the raw data to make the desired output, below the some experimental output:

Cause study#1: Call Ratio, Connected via VoIP Last Week



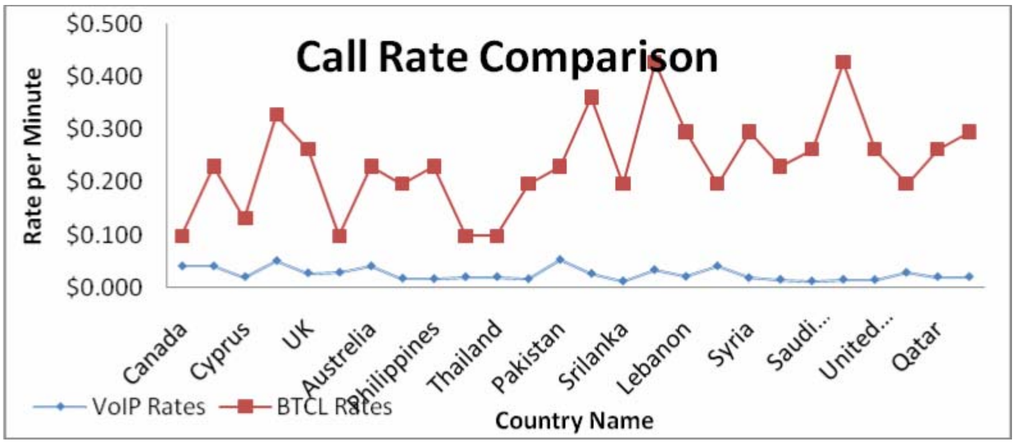
We have collected those data from one of the renowned VoIP Dialer ‘Nirban Voice’ duration of last one week. Here we can see that the call connection per week value is appx. 22698. The percentage ratio makes the sense that maximum call goes to India and the second highest calls goes to amazingly in Bangladesh.

Cause study#2: Calls per Hour in a Day via VoIP



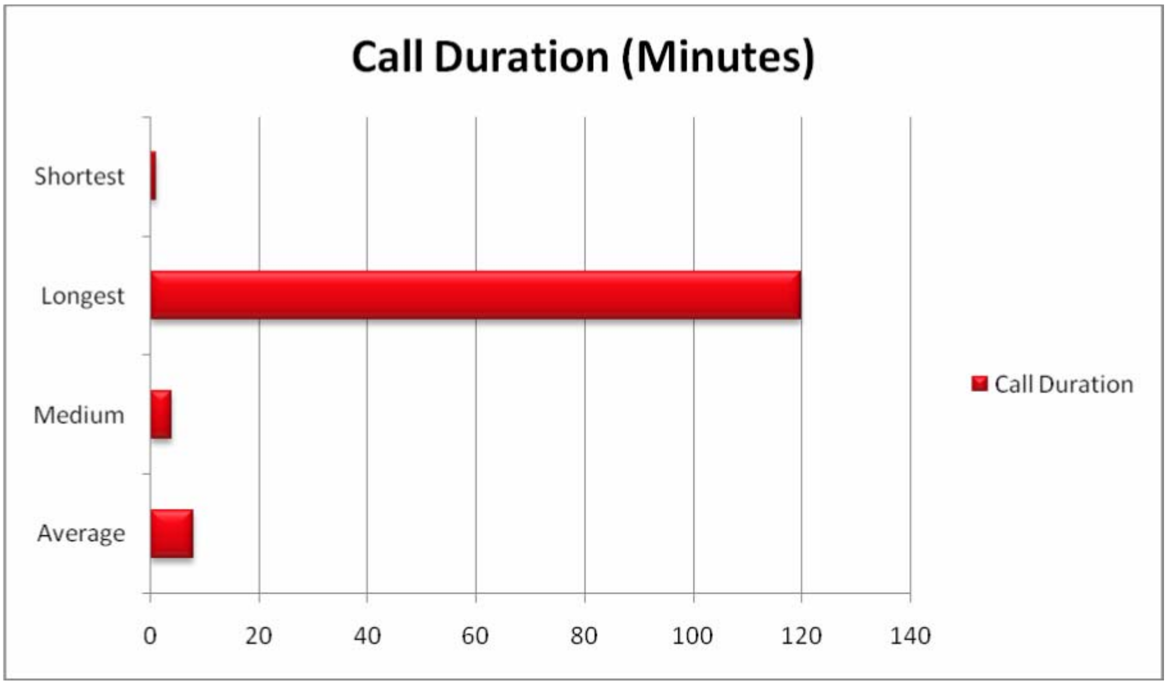
Here we can find the ratio of calls connected via VoIP within a time frame of a specific day (16 January 2011). The figure make the sense that maximum calls are connected between 6 o’clock at morning to 8 o’ clock of morning and the minimum calls are at 3 o’clock at afternoon. Here the second highest calls are made when it is 1 o’clock at night.

Cause study#3: Call Rate Comparison [BTCL VS VOIP]



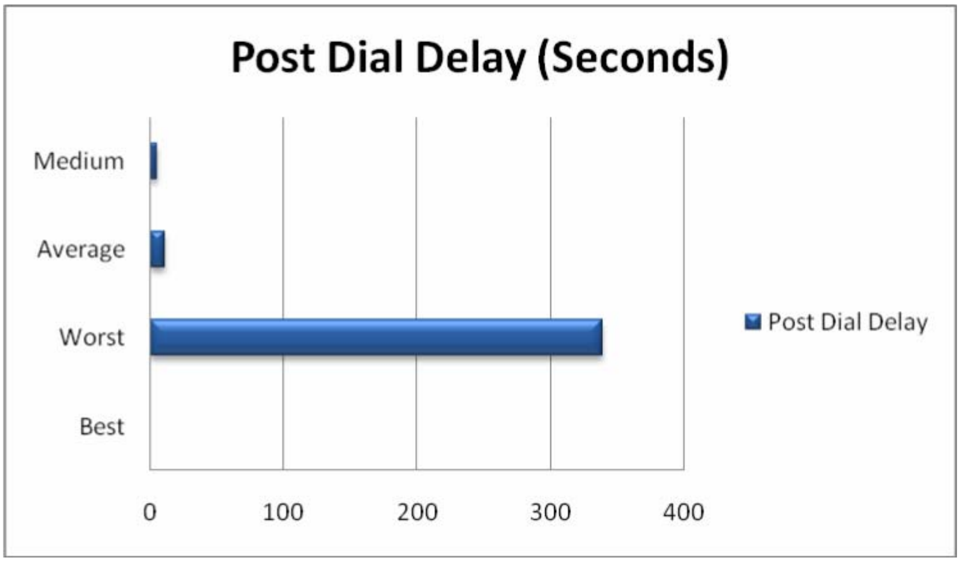
Here the chart figured out the comparison of call rates per minutes among the BTCL rates and VoIP call rates. The blue lines are showing the call rate of VoIP against country. It amazingly seems that the rate is like flat rate for all the possible countries, where the BTCL rates are huge.

Cause study#4: call duration of VoIP calls



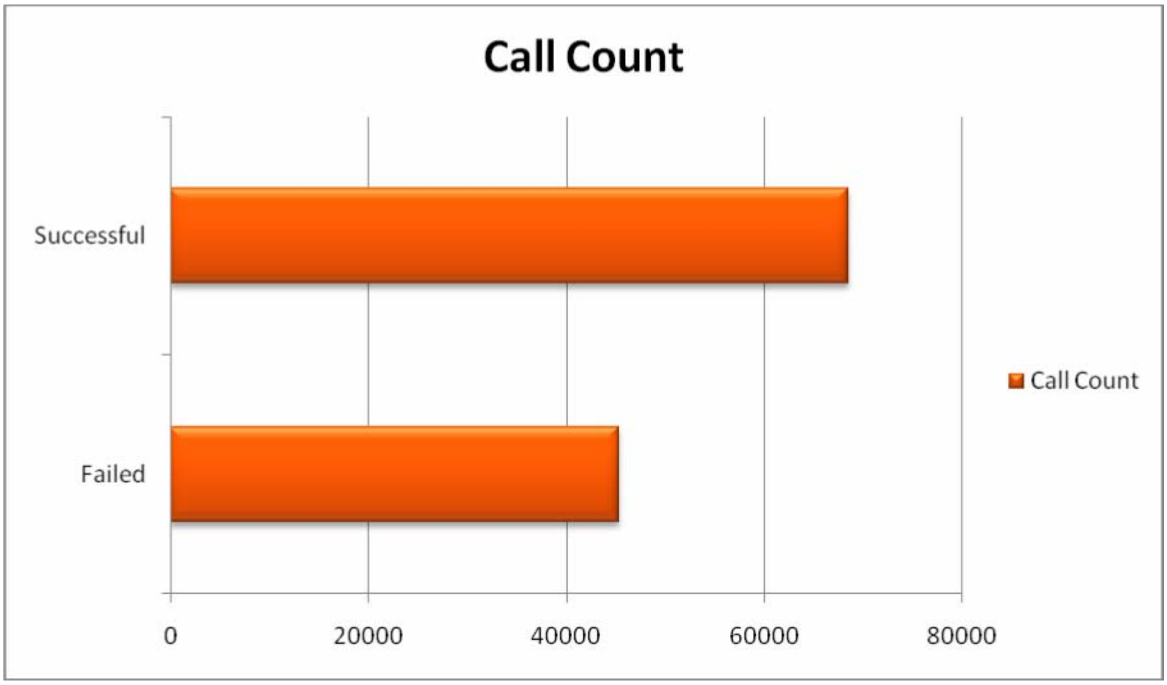
Here we see the call duration, from the graph the highest call duration is 120 minute and the lowest call duration is approximately 0-1 minute.

Cause study#5: Post Dial Delay of VoIP calls



Above this graph show the call Post Dial Delay of VoIP call, maximum PDD of a call is 340 second and lowest PDD of a call is 2/3 second [app.] Average PDD of VoIP calls are 5/7 second.

Cause study#6: Successful and failed call count

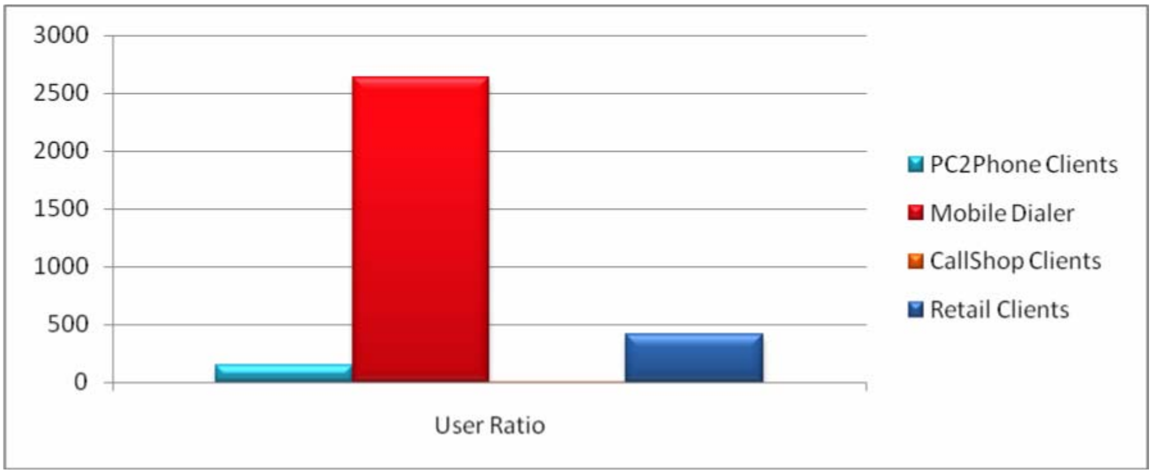


Above this graph show the success and failed calls of digital voice on one day.

5.3 experimental data from end user

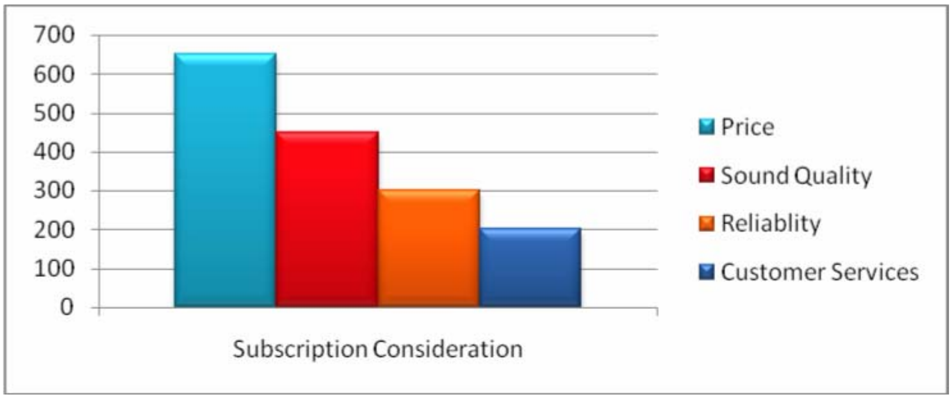
We make a survey among the subscribers of Digital Voice Ltd. a growing up VoIP service provider in Bangladesh. And bring out some synopsis of VoIP user experiences ratio.

Cause study#7: End user ratio



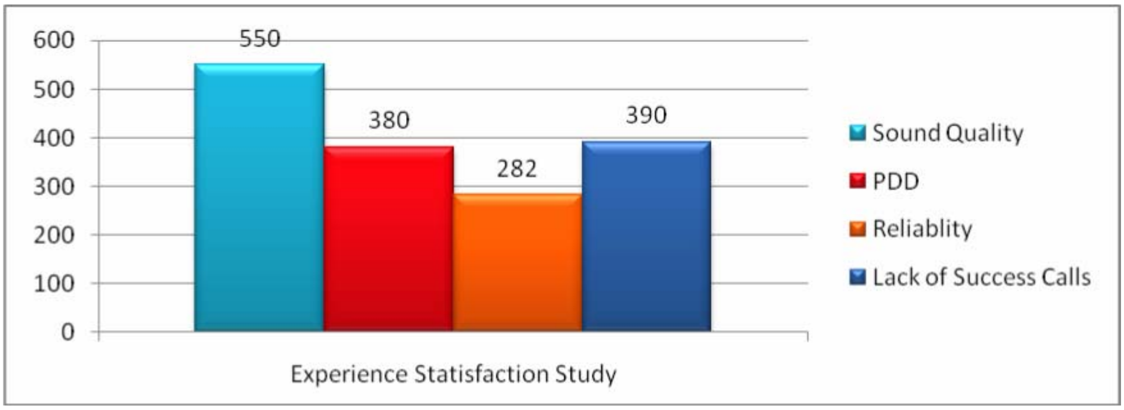
Here the User Ratio chart shows that out of 3204 subscriber of Digital Voice Ltd. 149 uses their PC2Phone service, whether the highest count for 2634 is for Mobile Dialer users and the lowest are Calls Shop Users and lastly 418 subscribers are retail clients. Among the 50% of randomly selected users are questioned by a survey of what are the issue that comes first in their mind when they subscribe a service.

Cause study#8: consider issues of End user



Here 650 subscribers first bring their consideration about pricing and accordingly less priority to sound quality (450 users), reliability (300 Users) and lastly 200 users for customer services. And among the same count of random selected users we have asked their satisfactory level of their consider issues.

Cause study#9: subscriber's satisfaction issue



550 subscribers are satisfied about sound quality, 380 subscribers are satisfied about PDD, and 282 users are for reliability and 380 for lack of success calls.

CHAPTER-6

Findings and Conclusions

6.1 Findings:

In this chapter we will present the outcome from this study in the form of findings and the conclusions. Initially findings and then general conclusions will be presented with implications for future research.

As we want to know that how the adoption of the voice over internet protocol (VoIP) can be improved so that more and more people will use this technology and will get the benefits associated with it rather than the existing expensive PSTN technology and for achieving that target initially. We have to focus on the major causes or the problems, which are hindrances in the better adoption of this technology by the people and then try to improve those factors or to rectify those problems in order to achieve our purpose. So from the previous chapter we see that the major causes for slower adoption are

- Price
- Sound Quality
- Marketing problems
- Reliability
- Latency
- Packet Loss
- Jitter
- Regulatory challenges

After identifying the major factors and giving the proper attention to these factors next we see that how these factors can be improved or rectified for the overall improvement of the adoption of this technology.

Price:

As we have seen that in our research individual end users and the service provider companies consider the price very major factor, but practically we see that the price in this service is much lesser than that of the existing PSTN price. So, wethink that actually service provider companies are not marketing their products and the prices of the their products because of which end users perceive that prices are high so in that regard we suggest the service providers that they should properly market the prices of their products because most of the active users who use this service frequently are satisfied with the price factor as evident from empirical data however before availing this service they gave too much importance to the price.

Sound Quality:

Sound quality is another important consideration from end users perspective. We see that most of the users are not satisfied with the quality of the sound however the end users said that Skype has the much better sound quality than that of the other service providers and in case of Skype we see that they are using their own proprietary protocols so for others service providers it is also recommended that they should also build their own proprietary protocols to get better sound quality.

Marketing Problems:

Marketing is also a big issue for the service providers. However, presently most of the service providers marketing strategies are not working properly because of which existing customers and potential customers of these service providers are not getting proper information regarding the products/ services and also about their prices. So it is suggested that the service providers should allocate the more budget for the marketing campaigns for getting the proper customer response and to provide necessary information to the end users because proper marketing of this service can greatly improve the adoption of this service in general messes.

Reliability:

To improve the reliability of VoIP, we believe that the following things are to be Considered.

- Constraint-based routing and traffic engineering: selecting different paths on which calls are routed with minimum acceptable delays and also reserving the sufficient capacity for the expected load and implementing advanced routing.
- Bandwidth management and admission control: managing the requests for the allocation of bandwidth for VoIP calls and limits those calls.
- QoS for established voice calls: ensuring that accepted calls are prioritized to maintain the QoS level even if a link is congested.

- Adaptive Coding: reduce bandwidth consumption in congestion state to still provide good QoS to active calls.
- Monitoring the network

Packet Loss:

In case of internet the media which is used for VoIP did not guarantee the 100% packets reached the destination but important thing is here that for better quality of service the packet loss should be as minimum as possible and for that minimum loss case we suggest that it is better to not send the silences especially in low speed networks or networks with congestion. There are many silences period in conversion so if we only send the voice information; we can use the available bandwidth in better way.

Jitter:

Jitter refers to variation in the amount of the 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. To overcome this problem the suggestion which we recommend is the jitter buffer as the solution and this buffer receives the packets and transmits to the receiver with a small delays. The limitation with the jitter buffer is that if length of the buffer is large it results in more delays and if buffer size is too small it will result in packet loss. So, service provider has

Regulatory Challenges

The effect of regulation on how VoIP providers market their solutions is important. The commission stated that voice communication services using IP that provide access to PSTN and utilize phone numbers that conform to the North American Numbering Plan have characteristics that are functionally the same as circuit-switched voice telecommunications services.2 As a result, the existing regulatory framework and tariffs should apply to VoIP. Although long distance phone calls are currently free using VoIP, the CRTC might implement charges and taxes which will result in an increase in price. The players offering VoIP solutions have to consider these changes.

With in this 2008 Bangladesh government is going to give a general rule for this VoIP providers and users and they will also have some effecting networking system is coming up as they are going to established the Submarine cable with in this year

Proposals and suggestions:

Voice over IP (VoIP) is revolutionary technologies by all means of modern time which change the attributes of communications dramatically. VoIP has been established as potential alternative to tradition public switched telephone network (PSTN) technology. Since VoIP technologies have shown its existence in today's market, wider view of effort in country's economy of this technologies is necessary and hence this are being deployed but the question is whether these technologies will be able to serve up to the same level of expectations on economic development of Bangladesh and print it successor foot step in contribution of economic development.

VoIP includes QoS, Security, user behavior, regulations, and last but not least current economic downturn. It would be a difficult task to explain all these issues with regards to the world market along with Bangladesh market. In this report so throughout our research, Bangladesh market is our preliminary focus. First, we discussed briefly look into the future services, applications and trends which will be beneficial in supporting the growth rate of VoIP in near future and then investigated all those trouble making factors causing a slow adoption rate of VoIP in Bangladesh. At the end of the report we have lined out that, there will be some proposals and suggestions to improve the growth rate which belong prospect of VoIP in Bangladesh as below:

It would be safe to conclude that VoIP is here to stay. It will continue to flourish greatly even in this era of economic concern and cost cutting. Lots of companies will be more excited to get the best growth, despite the economic crisis.

It is pretty safe to assume that one benefit of this economic down turn is that more and more companies would start to pay attention to taking appropriate measures for cost reduction. Using technology with special features as we mentioned earlier for cost reduction would indeed by coming their top priorities. In upcoming years not only the

SMEs will get the benefit from these services but the bigger organizations which have not consider these options until today, will find this opportunity quite exciting to adopt. Other than that we have very specific solutions for customer contact centers which are more intelligent. For instance some solutions offer the customers some great easy-to-use contact options of their choice; this will in turn force the companies to make the most of the reduced agent resources.

The companies will be able to save huge dollars by offering customers and employees, the kinds of sophisticated self- and assisted-service they have come to expect. These services may include, but are not limited to, options like advanced speech, web, and streaming video capabilities. These services enable the customers to discover how they can create an environment which is alive and thriving regardless of the economy – by using market-proven agent resource management strategies.Here we can make a reference of a recent seminar on VoIP Outsourcing Prospect in Bangladesh held on February 3, 2011 in Bangabandhu International Conference Center, Dhaka. Outsourcing business opportunities in the VoIP Industry is expected to reach more than USD 1 billion by 2015. This is a huge scope for the local ICT industry to increase its presence in the global market, said the industry insiders at a seminar organized by Reve Systems at softexpo2011.In his keynote speech, Sanjit Chatterjee, director, marketing, Reve Systems outlined the various opportunities in the VoIP industry, which can be explored by the Bangladesh entrepreneurs.M. Rezaul Hassan, CEO, Reve Systems disclosed that Bangladeshi entrepreneurs are leading the VoIP service industry and he mentioned some examples of Japan and USA.They said Bangladesh is the best place in the world to tap billion dollars through VoIP.

6.2 Conclusions

This report explores the main factors affecting the deployment of VoIP and its adoption of VoIP by Small Medium Enterprises (SMEs) and general consumers in Bangladesh which line out in these report papers subjected “A Feasibility study on VoIP-Perspective Bangladesh ".

Chapter1 and two have provided the motivation of the report, brief overview theory on VoIP which the results of this report are base on and the outline of the report.

Chapter 3 has proposed VoIP Soft Switch (VPS) and its management approaches for manage the VoIP reseller, balling, dialing plan, protocol setting, services, payment management statistics of ASR, ACD, PDD and other report making.

Chapter4 and chapter5 have focused on all of experimental study and its analysis like connected call ratio, calls per hour in a day, call rate comparison VoIP vs. BTCL, call duration of VoIP calls ,PDD of VoIP calls, successful and failed calls of a VoIP provider, difference type user ratio, and other two is end-user consideration and satisfaction study. It was also shows the present condition with benefit of VoIP in Bangladesh and Bangladeshi users and providers. Which will also boost growth of the country's telecommunication sector and economic development through VoIP .This report provides insight to the potentiality of development of VoIP under a legal way in Bangladesh.

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Appendix

A A L-ATM Adaptation Layer

ACD (Automatic Call Distribution)

ACM-Association for Computing Machinery

ADC- Analog to Digital Converter

ADPCM-Adaptive Differential Pulse Code Modulation

ADSL- Asymmetric Digital Subscribers Line

AIN- Advanced Intelligent Networks

ARPANET- Advanced Research Projects Agency Network

ASP-Active Server Pages

ATA- Analog Telephone Adaptor

ATM-Asynchronous Transmission Mode

Bps- Bits per second

CEO-Chief Executive Officer

CTI -Computer Telephony Integration

DSL-Digital Subscriber line

DTel-Deepija Telenetworks

H.323- Application Protocol by International Telecommunication Union (ITU)

IEEE- Institute of Electrical and Electronics Engineers

IETF- Internet Engineering Task Force

IP- Internet Protocol

IP/PBX-Internet Protocol Public Branch Exchange

IPv4 & IPv6-Internet Protocol version 4 and 6

ISDN-Integrated Service Digital Network

ISP- Internet Service Provider

ITU- International Telecommunication Union

IVR- Interactive Voice Response

Ms- Milli seconds

MIT- Massachusetts Institute of Technology

NAT-Network Address Translation

OSI (Open Systems Interconnection)

PBX- Public Branch exchanges

PC-Personnel Computer

PCM- Pulse Code Modulation

PPP-Point to Point Protocol

PSDN- Packet Switched Data Network

PSTN- Public Switched Telephone Network

QoS- Quality of Service

RFC- Request For Comment

RSVP- Resource Reservation Protocol

RTCP-Real Time Control Protocol

RTP- Real Time Protocol

RTSP-Real Time Streaming Protocol

SIP-Session Initiation Protocol

SONNET-Synchronous Optical Networking

TCP-Transmission Control Protocol

TTS- Text To Speech

UDP-User Datagram Protocol

US-United States

VoIP-Voice over Internet Protocol

WAN- Wide Are Networks

Wi-Fi- Wireless Fidelity denotes a set of Wireless LAN/WLAN standards

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