VOIP ADOPTION AND BUSSINESS PROSPECTIVE IN BANGLADESH

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DECLARATION

I hereby declare that this thesis is based on the results found by myself. Materials of work found by other researcher are mentioned by reference. This thesis, neither in whole nor in part, has been previously submitted for any degree.

Signature of Supervisor	Signature of Author

ABSTRACT

Voice over Internet protocol - VoIP, or IP telephony is a technology by which the routing of voice communications are done through Internet or any other Internet Protocol (IP) based networks. Here the voice data is transmitted over a general purpose packet-switched network instead of dedicated traditional circuit-switched voice transmission lines. Voice Over Internet Protocol (VoIP) is a telephony technology used to transmit ordinary telephone calls over the Internet. VoIP takes analogue audio signals and turns them into digital signals (packets) that are transmitted using Internet Protocol (IP) networks. VoIP's advantages include low cost, flexibility, and mobility. Conversely, VoIP's disadvantages include sound quality such as latency (delay), jitter, and packet loss. VoIP has a number of cultural, social, and regulatory impacts that solution providers must consider when marketing their services.Our target is to learn the basics of VoIP, to focus on the current problems of VoIP, to find out the solutions, to Setup a VoIP business.

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4.0 What is VoIP

4.1 An Overview

Voice over Internet protocol - VoIP, or IP telephony is a technology by which the routing of voice communications are done through Internet or any other Internet Protocol (IP) based networks. Here the voice data is transmitted over a general purpose packet-switched network instead of dedicated traditional circuit-switched voice transmission lines.

VoIP is a part of the group of technologies called voice over packet networks. Other network protocols like asynchronous transfer mode (ATM) can perform similar functions. Though the concept of VoIP is simple, the implementation and applications of it is a bit complicated. In order to send voice, the information has to be separated into packets just like data. Packets are chunks of information broken up into the most efficient size for routing. From there, the packets need to be sent and put back together in an efficient manner. For more efficient use, the voice data can be compressed so that it require less space and will certainly record only a limited frequency range. There are many ways to compress audio, the algorithm for which is referred to as a compressor/de-compressor (CODEC). Many a number of CODECs exist depending on the application (e.g., conversations, music, movies and sound recordings). The CODECs are optimized for compressing voice, which significantly reduce the bandwidth used compared to an uncompressed audio stream. Speech CODECs are optimized to improve spoken words at the expense of sounds outside the frequency range of human speech. Recorded music and other sounds do not generally sound very good when passed through a speech CODEC.

4.2 Technology Overview

VoIP is a new form of communication that takes analogue audio signals and turns them into digital signals, or packets. This is an innovative alternate to the traditional circuit-switched method of telecommunication, where a dedicated circuit between two parties is maintained. In order to set up a traditional phone call between two telephones, the switched and the intervening network establish a dedicated route from one end of the call to the other. Conversely, VoIP uses a packet-switched method where audio signals are converted into digital data at the originating end, which is then transmitted over the Internet and converted back to analog signal at the receiving end. In other words, VoIP digitizes voice, inserts the digitized data into discrete packets, and sends them over the IP network. The packets have a destination address, but no fixed path through the network. The packets arrive at the address, where they are put back together and converted back to analog audio signals. VoIP integrates voice and data communications and turns any Internet connection into a phone call. VoIP is a revolutionary technology that has the potential to drastically change the way people communicate and talk on the phone around the world.

5. How does VoIP works

When you speak at the handset or a mike or a microphone, your voice generates electrical signals inside the gadget. These are analog signals i.e. the voltage level can take up any value within a range.

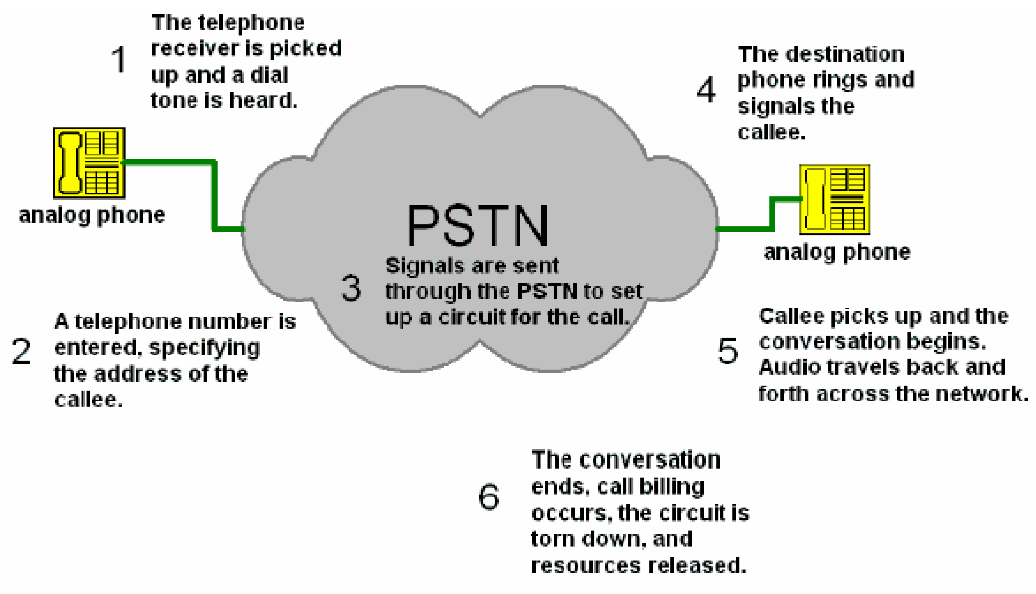


Fig: Typical PSTN form.

The analog signal is converted to a digital signal using an algorithm implemented by the device you are using. It can be a stand-alone VoIP phone or a softphone running on your PC. If you are using an analog phone, you will need a Telephony Adapter (TA) for this purpose. The digitized voice is arranged in packets (i.e. collection of bytes) and sent over the IP network.

The data is channeled through gateways and servers to the destination. If the called number is on the PSTN, the server opens a connection to the PSTN and routes your call there.

While going to the PSTN or at the end device of a VoIP connection, the voice is gain brought back to its analog form so that it is perceptible to a human ear. During the entire process a protocol like SIP or H.323 is used to control the call (e.g. setting up connection, dialing, disconnecting etc.) and

RTP is used for reliable transmission of data packets and maintain Quality of Service.

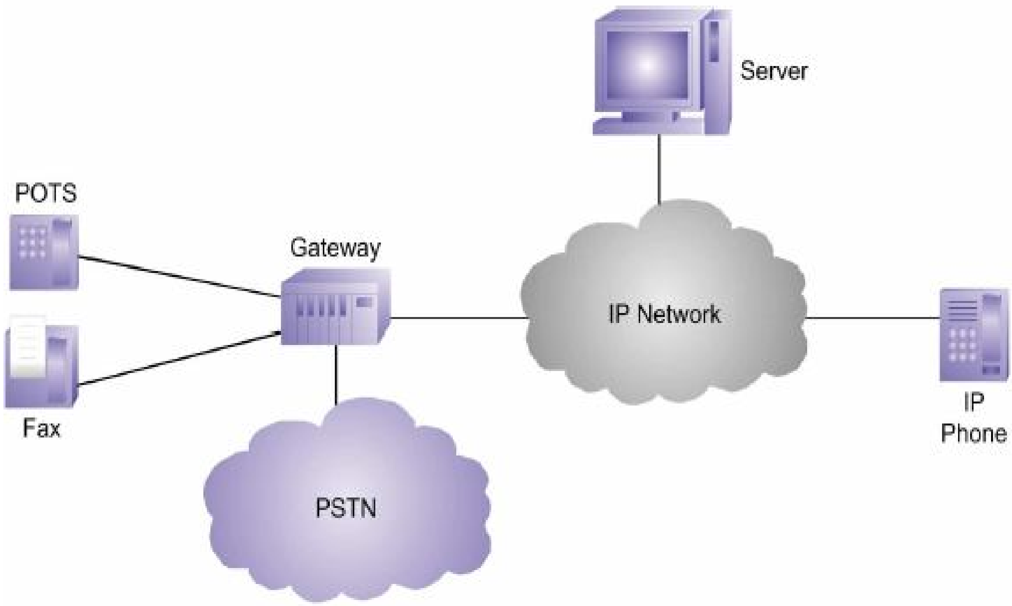


Figure 1. VoIP Components

5.1 The digitization of analog voice signals

The digitization of analog voice signals is a must to transmit voice over the digital IP network. It can be done in several ways:

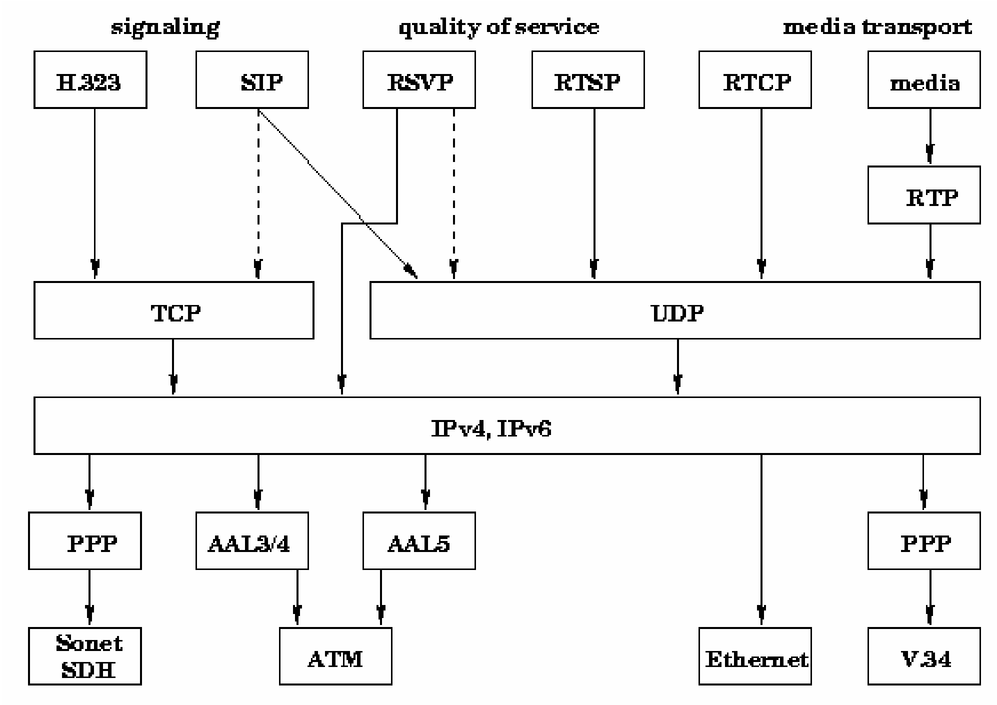
PCM (Pulse Code Modulation) is a simple technique of sampling the sound signal at a fixed rate (8000 Times/second) and generate a number corresponding to each sample. It assumes no specific property of the signal. So it works reasonably well with all types of sounds.

LPC (Liner Predictive Coding) assumes specific properties of human voice and uses a more complex algorithm to digitize and compress voice data. It works well for sending human utterances offering a low data rate but is not suitable for transmitting music or fax.

SBC (Sub Band Coder) uses a different approach of representing sounds in terms of frequencies rather than sampling at regular intervals.

Hybrid coders like the CELP (Code Excited Linear Prediction) use a mixture of the techniques to transmit sound of adequate quality.

5.2 Protocol Architechture



5.3 RTP

Real-time Transfer Protocol (RTP) is the Internet-standard protocol for the transport of real-time data, including audio and video. It can be used for media-on-demand as well as interactive services such as Internet telephony. RTP consists of a data and a control part. The latter is called RTCP.

The data part of RTP is a thin protocol providing support for applications with real-time properties such as continuous media (e.g., audio and video), including timing reconstruction, loss detection, security and content identification.

RTCP provides support for real-time conferencing of groups of any size within an internet. This support includes source identification and support for gateways like audio and video bridges as well as multicast-to-unicast translators. It offers quality-of-service feedback from receivers to the multicast group as well as support for the synchronization of different media streams.

While UDP/IP is its initial target networking environment, efforts have been made to make RTP transport-independent so that it could be used, say, over CLNP, IPX or other protocols. RTP is currently also in experimental use directly over AAL5/ATM. RTP does not address the issue of resource reservation or quality of service control; instead, it relies on resource reservation protocols such as RSVP .

Other applications, such as real-time control and distributed simulation, are also targets.

RTP was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889 . It was originally designed as a multicast protocol, but has since been applied in many unicast applications. It is frequently used in streaming media systems (in conjunction with RTSP) as well as videoconferencing and push to talk systems (in conjunction with H.323 or SIP), making it the technical foundation of the Voice over IP industry. It goes along with the RTP Control Protocol (RTCP) and it's built on top of the User Datagram Protocol (UDP) (in OSI model).

According to RFC 1889 , the services provided by RTP include:

- Payload-type identification
- Sequence numbering
- Time stamping
- Delivery monitoring

5.4 SIP

Session Initiation Protocol (SIP) is a protocol developed by IETF to assist in providing advanced telephony services across the Internet. Basically is a signaling protocol used for establishing sessions in an IP network. A session could be a simple two-way telephone call or it could be a collaborative multi-media conference session. The ability to establish these sessions means that a host of innovative services become possible, such as voice-enriched e-commerce, web page click-to-dial, Instant Messaging with buddy lists, and IP Centrex services.

SIP is modeled upon other Internet protocols such as SMTP (Simple Mail Transfer Protocol) and HTTP (Hypertext Transfer Protocol.) It is used to establish, change and tear down (end) calls between one or more users in an IP-based network. In order to provide telephony services there is a need for a number of different standards and protocols to come together - specifically to ensure transport (RTP), signaling inter-working with today’s telephony network, to be able to guarantee voice quality (RSVP, YESSIR), to be able to provide directories (LDAP), to authenticate users (RADIUS, DIAMETER), and to scale to meet the anticipated growth curves.

SIP can be regarded as the enabler protocol for telephony and voice over IP (VoIP) services. The following features of SIP play a major role in the ennoblement of IP telephony and VoIP:

* Name Translation and User Location - Ensuring that the call reaches the called party wherever they are located. Carrying out any mapping of descriptive information to location information. Ensuring that details of the nature of the call (Session) are supported.

* Feature Negotiation - This allows the group involved in a call (this may be a multi-party call) to agree on the features supported – recognizing that not all the parties can support the same level of features. For example video may or may not be supported; as any form of MIME type is supported by SIP, there is plenty of scope for negotiation. Call Participant Management - During a call a participant can bring other users onto the call or cancel connections to other users. In addition, users could be transferred or placed on hold.

* Call feature changes - A user should be able to change the call characteristics during the course of the call. For example, a call may have been set up as 'voice-only', but in the course of the call, the users may need to enable a video function. A third party joining a call may require different features to be enabled in order to participate in the call

* Media negotiation – The inherent SIP mechanisms that enable negotiation of the media used in a call, enable selection of the appropriate codex for establishing a call between the various devices. This way, less advanced devices can participate in the call, provided the appropriate codex is selected.

Below is are some of other SIP features that distinguish it among new signaling protocols

* SIP messages are text based and hence are easy to read and debug. Programming new services is easier and more intuitive for designers.

* SIP re-uses MIME type description in the same way that email clients do, so applications associated with sessions can be launched automatically.

* SIP re-uses several existing and mature internet services and protocols such as DNS, RTP, RSVP etc. No new services have to be introduced to support the SIP infrastructure, as much of it is already in place or available off the shelf.

* SIP extensions are easily defined, enabling service providers to add them for new applications without damaging their networks. Older SIP-based equipment in the network will not impede newer SIP-based services. For example, an older SIP implementation that does not support method/ header utilized by a newer SIP application would simply ignore it. SIP is transport layer independent. Therefore, the underlying transport could be IP over ATM. SIP uses the User Datagram Protocol, (UDP) as well as the Transmission

Control Protocol (TCP) protocol, flexibly connecting users independent of the underlying infrastructure.

* SIP supports multi-device feature leveling and negotiation. If a service or session initiates video and voice, voice can still be transmitted to non-video enabled devices, or other device features can be used such as one way video streaming.

SIP sessions utilize up to four major components: SIP User Agents, SIP Registrar Servers, SIP Proxy Servers and SIP Redirect Servers. Together, these systems deliver messages embedded with the SDP protocol defining their content and characteristics to complete a SIP session.

5.5 H.323

H.323 is a standard that specifies the components, protocols and procedures that provide multimedia communication services : real-time audio, video, and data communications over packet networks, including Internet protocol (IP) based networks. H.323 is part of a family of ITU-T recommendations called H.32x that provides multimedia communication services over a variety of networks.

H.323 was originally created to provide a mechanism for transporting multimedia applications over LANs but it has rapidly evolved to address the growing needs of VoIP networks. One strength of H.323 was the relatively early availability of a set of standards, not only defining the basic call model, but in addition the supplementary services, needed to address business communication expectations. H.323 was the first VoIP standard to adopt the IETF standard RTP to transport audio and video over IP networks.

The H.323 standard specifies four kinds of components, which, when networked together, provide the point-to-point and point-to-multipoint multimedia-communication services:

- **Terminals:** Used for real-time bi-directional multimedia communications, an H.323 terminal can either be a personal computer (PC) or a stand-alone device, running an H.323 and the multimedia applications. It supports audio communications and can optionally support video or data communications. Because the basic service provided by an H.323 terminal is audio communications, an H.323 terminal plays a key role in IP–telephony services. An H.323 terminal can either be a PC or a stand-alone device, running an H.323 stack and multimedia applications. The primary goal of H.323 is to inter-work with other multimedia terminals. H.323 terminals are compatible with H.324 terminals on SCN and wireless networks, H.310 terminals on B–ISDN, H.320 terminals on ISDN, H.321 terminals on B–ISDN, and

H.322 terminals on guaranteed QoS LANs. H.323 terminals may be used in multipoint conferences.

- **Gateways:** A gateway connects two dissimilar networks. An H.323 gateway provides connectivity between an H.323 network and a non–H.323 network. For example, a gateway can connect and provide communication between an H.323 terminal and SCN networks (SCN networks include all switched telephony networks, e.g., public switched telephone network [PSTN]). This connectivity of dissimilar networks is achieved by translating protocols for call setup and release, converting media formats between different networks, and transferring information between the networks connected by the gateway. A gateway is not required, however, for communication between two terminals on an H.323 network.
- **Gatekeepers:** A gatekeeper can be considered the brain of the H.323 network. It is the focal point for all calls within the H.323 network. Although they are not required, gatekeepers provide important services such as addressing, authorization and authentication of terminals and gateways; bandwidth management; accounting; billing; and charging. Gatekeepers may also provide call-routing services.
- **Multipoint Control Units:** MCUs provide support for conferences of three or more H.323 terminals. All terminals participating in the conference establish a connection with the MCU. The MCU manages conference resources, negotiates between terminals for the purpose of determining the audio or video coder/decoder (CODEC) to use, and may handle the media stream. The gatekeepers, gateways, and MCUs are logically separate components of the H.323 standard but can be implemented as a single physical device.

Key Benefits of H.323 :

Codec Standards: H.323 establishes standards for compression and decompression of audio and video data streams, ensuring that equipment from different vendors will have some area of common support.

Interoperability: Users want to conference without worrying about compatibility at the receiving point. Besides ensuring that the receiver can decompress the information, H.323 establishes methods for receiving clients to communicate capabilities to the sender. The standard also establishes common call setup and control protocols.

Network Independence: H.323 is designed to run on top of common network architectures. As network technology evolves, and as bandwidth-management techniques improve, H.323-based solutions will be able to take advantage of those enhanced capabilities.

Platform and Application Independence:H.323 is not tied to any hardware or operating system. H.323-compliant platforms will be available in many sizes and shapes, including video-enabled personal computers, dedicated platforms, IP-enabled telephone handsets, cable TV set-top boxes and turnkey boxes.

Bandwidth Management:

Video and audio traffic is bandwidth-intensive and could clog the corporate network. H.323 addresses this issue by providing bandwidth management. Network managers can limit the number of simultaneous H.323 connections within their network or the amount of bandwidth available to H.323 applications. These limits ensure that critical traffic will not be disrupted.

Flexibility: An H.323 conference can include endpoints with different capabilities. For example, a terminal with audio-only capabilities can participate in a conference with terminals that have video and/or data capabilities. Furthermore, an H.323 multimedia terminal can share the data portion of a video conference with a T.120 data-only terminal, while sharing voice, video, and data with other H.323 terminals.

Inter-Network Conferencing: Many users want to conference from a LAN to a remote site. For example, H.323 establishes a means of linking LAN-based desktop systems with ISDN-based group systems. H.323 uses common codec technology from different videoconferencing standards to minimize transcoding delays and to provide optimum performance.

6.0 PSTN vs VoIP

Many factors in the past have slowed the anticipated growth of Voice over IP (VoIP). Now, VoIP solutions that achieve quality and reliability, close to what we are used to from the Public Switched Telephony Network (PSTN), are emerging as the market is quickly growing. However, as will be shown in this article, there is no reason to limit the expectations to achieve only the same level of quality as in PSTN. It is quite well known that, by deploying wideband voice codecs, much better quality can be achieved. However, a little known fact is that there are ways to achieve better quality than a standard PSTN solution, even when using narrowband codecs. For example, the full available spectral bandwidth is not typically utilized in traditional PSTN solutions, something that can easily be done in a VoIP system. Implementing a wideband codec or expanding the bandwidth of narrowband codecs does not automatically guarantee great quality. There are many potential pitfalls when deploying VoIP. In this article we will also discuss implementation issues related to VoIP that will impact the final voice quality. We will discuss what level of quality can be achieved and describe how this can be implemented.

Speech Signals and Speech Coding Sampled digital signals can contain frequency content up to half the sampling frequency. Typically, a young adult has a hearing span from about 20 to 20,000 Hz. Consequently, the sampling frequency of CD audio is chosen to be 44.1 kHz, which is more than double that of the highest frequency perceivable by most humans. Legacy telephony solutions are narrowband, which seriously limits the achievable quality. Wideband codecs could potentially be used in digital telephone systems, but this has never been practical enough to gain any real interest

In fact, in traditional telephony applications, the speech bandwidth is restricted much more than the inherent limitations of narrowband coding. Typical telephony speech is bandlimited to 300 – 3400 Hz (listen to Sound Pure digital connections are typically only found in enterprise environments. Due to poor connections or old wires, significant distortion is often generated in the analog part of the phone connection, a type of distortion that is entirely absent from VoIP implementations. The cordless phones so popular today also generate significant amounts of analog distortion due to radio interference and other implementation issues.

On the Public Switched Telephone Network (PSTN), calls between two parties are set up by a series of private and public switches. The resulting fixed communications link is dedicated for the duration of the call. When an individual makes a phone call over a circuit-switched network, a connection is made between a company's PBX and the local telephone company, also known as the PSTN. Depending on the destination, this connection might extend to the national or international exchange before reaching another local exchange, where it will be passed on to the PBX and the person who receives the call. This end-to-end link, established by a series of public and private switches, is 100% dedicated on a single, per-call basis and cannot be shared or used for another function as long as the call is in progress. For this reason, these dedicated circuits cannot be shared and the carrier bills the call on a time and distance rate. The Internet does not use switches to link calling parties. Instead, the analog voice signal is digitized by an Internet Protocol (IP) and broken up into thousands of small data packets by a router - the VoIP equivalent to a switch. These data packets are sent, or routed, over the public Internet to their destination, enabling calls to bypass the PSTN entirely.

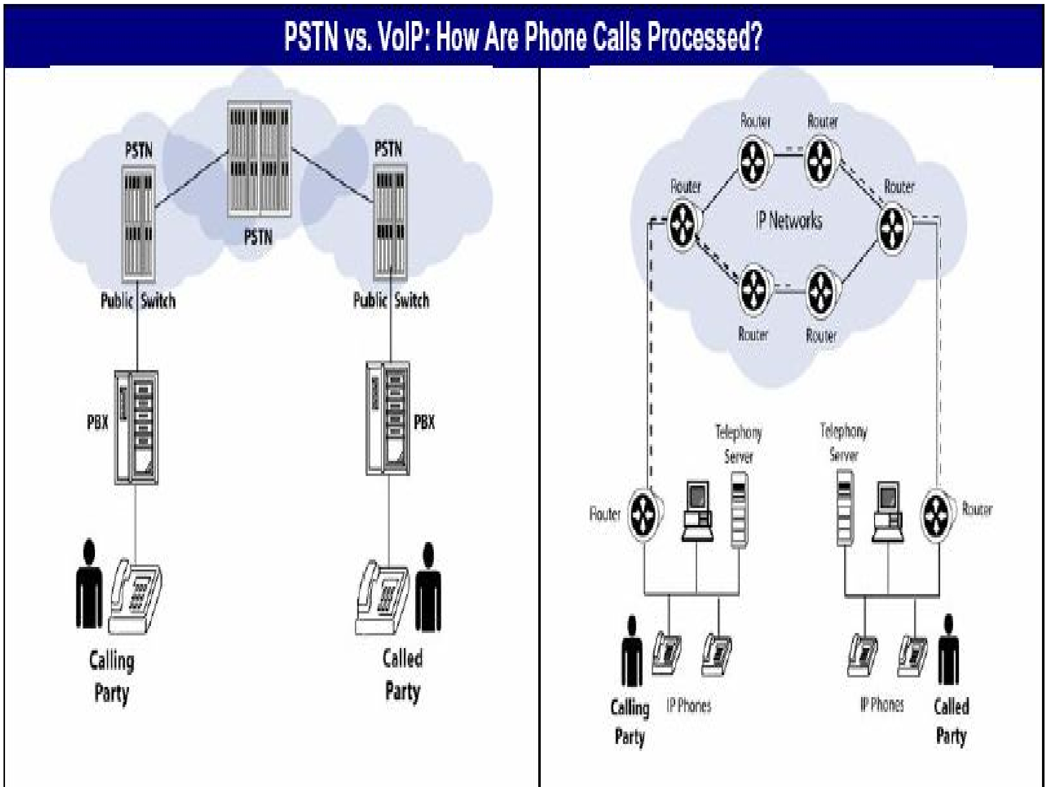


Figure 1: Traditional Circuit-Switched Network

Figure 2: VoIP Networks

7.0 REASON FOR SLOWER ADOPTION OF VoIP

VoIP has its share of disadvantages when compared to the functionality of PSTN. A major disadvantage of VoIP is that it is a new technology. As a result, the long-term benefits and risks are not yet known. These risks include unknown service life of hardware and infrastructure, and details surrounding reliability and quality. The factors that affect the sound quality during transmission include latency (or delay), jitter, and packet loss.[i] In terms of latency, human ears can withstand a delay of 150-250 ms and not be able to notice the delay. The PSTN meets this standard with a nominal delay of 150; however, VoIP cannot meet this standard of delay for a consistent period of time.[ii] Jitter is defined as the variability in packet arrival at the destination.[1] Voice packets are transmitted over the same IP network as normal data packets and therefore voice packets have to compete for bandwidth with data packets. When a situation arises whereby there is a burst of network traffic (mostly in the form of data packets) voice packets arrive at sporadic times to the destination. The consequence of sporadic arrival time of the packets is sound distortion at the receiver's end - jitter. Lastly,

the issue of packet loss occurs when voice packets that are transmitted over the network do not arrive at the destination. Along the same lines as the causes of jitter, the IP network is to blame for this drawback because it does not guarantee delivery of any packets (data or voice). The consequence of packet loss is distortion at the receivers end as sounds and words may actually never reach the receiver.

With regards to availability, VoIP must meet the "five nines" availability demanded of phone services (i.e. VoIP must be available at least 99.999% of the time). A common misconception is that VoIP will have lower dependability and availability than standard PSTN systems because of power failures, internet service provider 'down-time', security issues, etc. Nevertheless, it is has been demonstrated that it is possible to build VoIP systems that are more reliable than circuit based PSTN platforms. Adaptive routing ensures that packets reach their designation using multiple network lines.

Overall, the disadvantages of VoIP are not significant enough to hamper its ability to compete with traditional PSTN. In addition, advances are being made for the technology to get over some of these stumbling blocks. For example, the problem of jitter has been shown to decrease by using specialized gateways that determine whether large network data bursts are

currently affecting throughput and the gateway adjusts to decrease jitter. The technology has matured to a state where major players are now offering VoIP solutions as alternatives to traditional telecommunication solutions.

Packet loss, the most important component of a good route for a VoIP call, is the percentage of transmitted packets that never reach the intended estination. Packet loss of just 1-2% can affect service, resulting in confusion and frustration for VoIP users. This is because VoIP technology utilizes User Datagram Protocol (UDP) rather than Transmission Control Protocol (TCP) to route packets.† TCP is a connection-oriented protocol that automatically re-transmits lost packets in the event that they fail to be delivered, which is ideal for data transmissions. UDP, however, is not connection-oriented and does not re-transmit lost packets. In a real-time communications environment, old voice packets are of no use to callers and only cause confusion if they arrive after the conversation has progressed beyond the point where the packet fits. When UDP packets are lost, callers hear clipping, or short losses of conversation. The second most important component is jitter. Jitter causes packets to arrive at their destination in uneven patterns, which degrades the call and causes inconsistent voice quality. Depending on the type of VoIP codec equipment the enterprise uses, VoIP can be extremely jitter-sensitive. The codec will deal with jitter by buffering the call, which creates an audible delay. When the delay goes above 100 milliseconds (ms), it is noticeable and begins to interfere with the conversation. If jitter exceeds the levels for which codecs can buffer, the call will begin to clip and may be dropped. Although thresholds vary by equipment manufacturer, jitter should be less than 15 ms. The last metric is latency. Although thresholds for latency are much larger than for packet loss, latency varies significantly for domestic and international calls. Also, there is a lower expectation on the user's part when it comes to the delay on international calls. Ideally, latency should be below 200 ms round trip internationally and less than 100 ms round trip domestically. When latency is more than 100 ms, listeners hear a slight pause in the conversation which may be acceptable for international calls but not for domestic. When the delay is 250 ms, conversations become very stilted and experience many awkward silences and accidental interruptions. The biggest contribution to latency internationally is the use of satellite links. A single satellite hop will add up to 500 ms of delay to a round trip connection making the best paths for international calls undersea fiber connections.

7.1 Cultural and Social Impact

The cultural and social impacts are important in order to understand the customer base and must be taken into account in order to increase market penetration. VoIP brings in a change of social behaviour patterns similar to that of text messaging on mobile phones. Individuals can take their VoIP connection along with them as long they have a broadband connection available at their location. In addition, the long distance VoIP charges are cheap enough to open up new market segments such as small businesses, students, teenagers, and even children. For example, youth TV channels could adopt VoIP and video streaming to deliver interactive shows to youth. The availability of new products and services which leverage the use of VoIP will be a major driving force in increasing adoption of VoIP by the general public.

Security offered by VoIP systems vs. traditional phone systems is still an issue. This will affect the confidence of the users in adopting a VoIP system, especially in a business environment where confidential matters may be discussed. Also, other risks and fears of adoption include complicated installation, and the threat of viruses and hackers. These fears and risks must be addressed in order to improve the likelihood of adoption.

As work groups continue to become distributed across several time zones, they increasingly need to interact and conduct business in a virtual environment. Studies have revealed that alternative method of communication such as email is not very effective for bringing people together. The combination of instant messaging with VoIP has been found to be very effective in such situations. The addition of “presence awareness” enables employees.

8.0 Some Bussiness Solution

8.1Calling Cards Solution

Business Opportunity

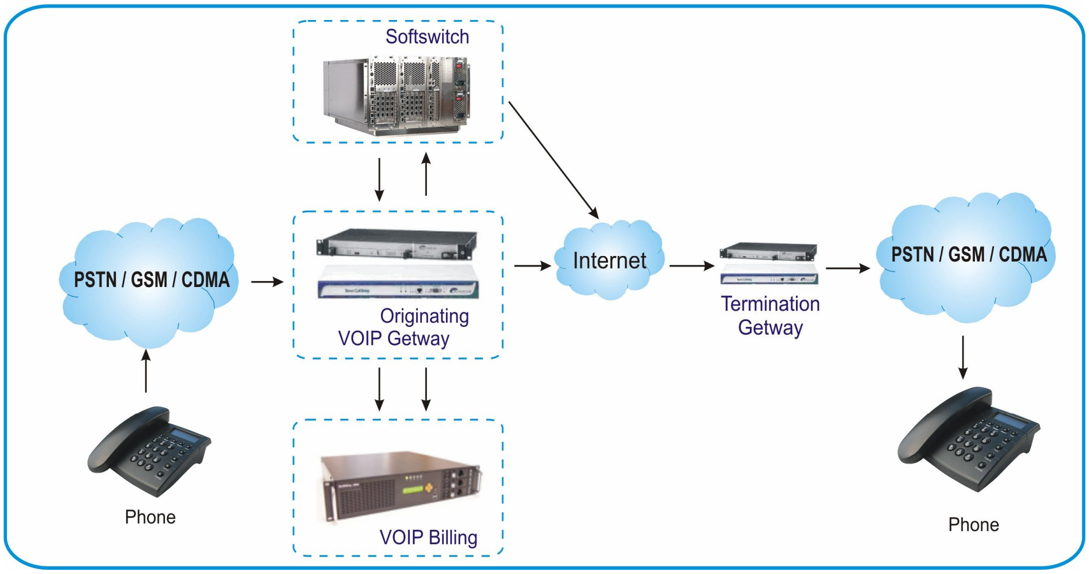
Traditionally, the telecom sector has been one of the most lucrative market segments in both emerging and developed economies. But also traditionally, that sector has been protected by heavy local regulation. Nowadays, however, many countries have either started or are considering deregulation of their telecom sectors. Such strong trend towards liberalization of the sector, coupled with the emergence of new technologies for cost effective transport of voice data, and namely Voice-over-IP, opens up new revenue generating opportunities for entrepreneurs all over the world.

The calling cards business model offers a relatively low cost entry into the lucrative telecom market segment. The essence of the calling cards business model is to creatively segment customers by various demographic and/or behavioral characteristics and to design calling cards offerings to meet their specific calling needs. The calling cards business typically attracts entrepreneurs who want to enter the VoIP market, businesses with established retail distribution channels, and service providers who want to diversify their revenue streams.

Business Solution

Aiming PROVIDER as a leading VoIP service provider, PROVIDER have to offer integrated, scalable, and cost-effective calling cards solutions. Such solutions will feature powerful billing capabilities, intuitive CRM (Customer Relation Managent) web portals, and proven interoperability with equipment from other VoIP providers. Because all solution components are developed by the engineers of PROVIDER and Softswitch Billing company, that reduces the need for integration costs at the PROVIDER VOIP service level which saves money for PROVIDER and improves our return on investment.

Calling Card Solution



How will the solution work?

User experience

- The user will purchase a calling card from a retail or online store.
- The user will dial the local or toll-free access number printed on the card.
- The user will hear a voice prompt, asking him to enter his PIN number.
- The user will then enter the PIN number printed on the calling card using the phone dial pad.
- He will hear his account balance and is invited to make a call.
- The user will dial the number and will get connected.

Provider experience

- When the user calls, PROVIDER Gateway, which is connected to the access number, will answer the call and send IVR prompts to the caller, inviting him to enter his PIN number.
- PROVIDER Gateway will read the PIN and will send an authorization request to PROVIDER VoIP Billing

- PROVIDER VoIP Billing will verify account information and will authorize the call if the user has sufficient balance.
- Then PROVIDER Gateway will announce to the caller his account balance and will invite him to dial a destination number.
- Then PROVIDER Gateway will passe the destination number to PROVIDER Softswitch and will request routing information.
- PROVIDER Softswitch will return to PROVIDER Getway the IP address of the remote gateway and Getway will connect the call.
- During the conversation, Getway will convert voice signal to data packets and will route them to the terminating gateway (owned by the call termination provider) and vice versa.
- Upon call completion, PROVIDER VoIP Billing will record complete CDR information for the call and then debit the user account with accumulated service charges.

Solution components are required

Core elements of PROVIDER's Calling Cards Solution include:

- PROVIDER VoIP Billing, will be a robust billing server that provides us with all necessary tools to successfully implement a wide spectrum of VoIP business models.
- VoIP Gateway, will be a flexible switching device that provides us with universal IP-PSTN switching, high flexibility and remote feature upgradeability.
- PROVIDER Softswitch will be an advanced VoIP softswitch that provides secure and reliable peering between our own networks and the VoIP networks of our business partners.

8.2 CallShop Solution

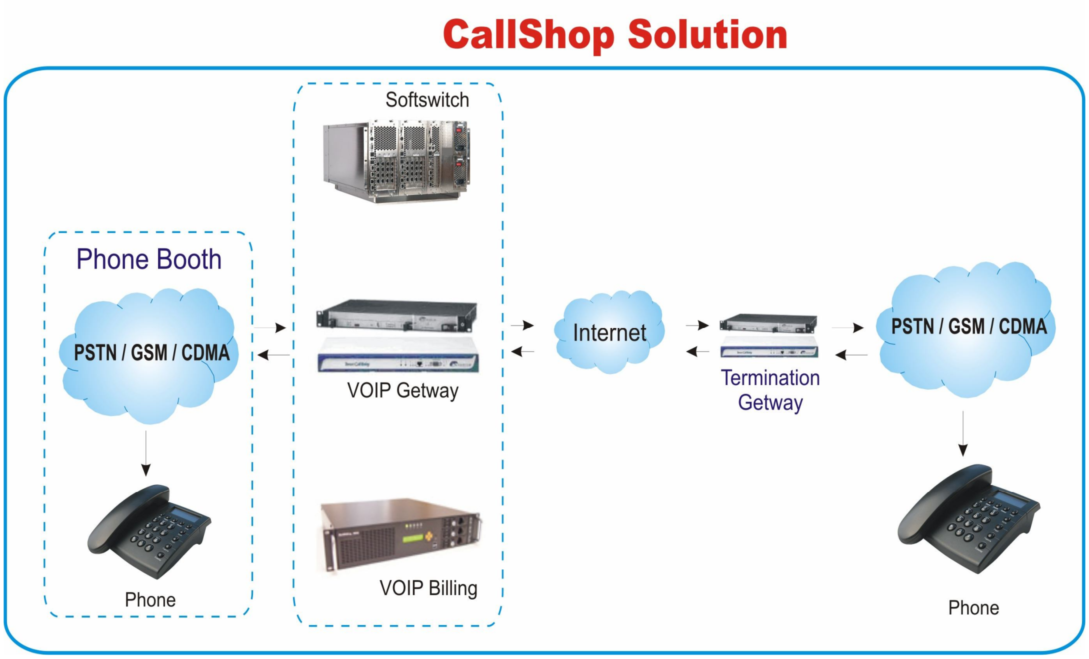
Business Opportunity

The rapid development of Voice-over-IP technology in the early 2000s has lead to fast innovation in the telecom sector. New telecom providers have introduced many innovative services utilizing VoIP technology, such as calling cards, callshops, broadband VoIP and others. Because such services are delivered over broadband lines, countries with well established last mile Internet infrastructure tend to adopt faster VoIP services where end-users install VoIP equipment at their premises (e.g. IP Phones), such as broadband VoIP . Alternatively, countries with limited or expensive last mile Internet infrastructure tend to adopt VoIP services that do not require end-users to install equipment, such as calling cards and callshops.

The Callshop business model appeals to providers in countries with limited bandwidth or expensive access to the Internet. Callshop services also offer lucrative opportunity in countries with limited number of telecom providers and heavy telecom regulation. Consumers, in such countries, typically do not have access to low cost long distance and international telephony services. Finally, Callshops tend to flourish in areas which attract international tourism. In any of the above cases, Callshop operators can capitalize on VoIP technology and offer competitively priced calling services to any part of the world.

Business Solution

PROVIDER will offer end-to-end, cost-effective and scalable Callshop solution. The



solution will feature powerful billing, flexible routing, and proven interoperability with VoIP equipment from other leading voip provider. Because PROVIDER will develop all solution components, callshop owners benefit from reduced integration costs and improved return on investment.

How will the solution work?

User experience

- When a customer visits the Callshop. customer will chooses a vacant telephony booth, will enters and place a call by dialing a destination number.
- The call will get connected.
- Upon call completion, the callshop operator will present the customer with an invoice for accumulated call charges.
- Then the customer will pay the callshop operator.

Provider experience

- When the user picks up the phone and dials a destination number, VoIP Gateway will send an authorization request to PROVIDER VoIP Billing Server.
- Billing Server will verify whether a call can be placed from that particular booth and will authorize the call.
- Then VOIP Getway will pass the destination number to PROVIDER Softswitch and will requests routing information.
- PROVIDER Softswitch will return VOIP Getway the IP address of the remote (termination) gateway and VOIP Getway will connect to it. Then the remote gateway will terminate the call to the destination party.
- Upon call completion, PROVIDER VOIP Billing Server will records complete CDR information for the call and will make it available to the operator for billing, reporting and monitoring purposes.

Solution components are required

Core elements of SysMaster's Callshop Solution include:

- PROVIDER VOIP Billing Server that supports a wide spectrum of pre- and post-paid VoIP services.
- VOIP Getway will be a carrier grade gateway that provides universal IP-PSTN switching, high flexibility and remote feature upgradeability.
- PROVIDER Softswitch is a flexible VoIP Softswitch that offers routing and reliable peering between VoIP networks and capable for high volume callshop implementations.

8.4 Wholesale VoIP Solution

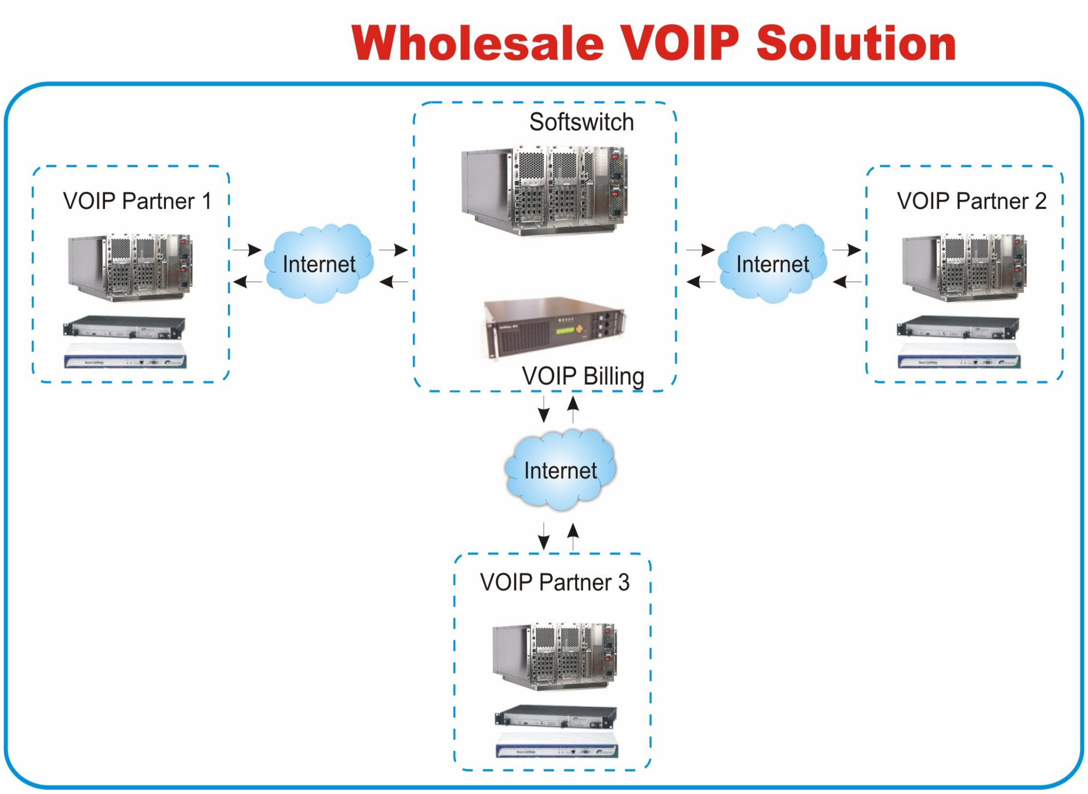
Business Opportunity

The birth of Voice-over-IP technology in late 1990s and its rapid growth in early 2000s, have resulted in the proliferation of many alternative telecom providers. Those providers have introduced a number of innovative communications services such as calling cards and callback which relied on call origination and termination over the Internet. The growing number of new telecom providers led to increased demand for traffic aggregation services and as a result the Wholesale VoIP business was born.

Wholesale VoIP providers act as traffic aggregators and/or traffic exchanges for their customers. They sign up contracts with both retail and wholesale VoIP providers and act as a middle man for call origination and termination services. When partners send traffic to the Wholesale VoIP provider, he/she reroutes it to other partners for termination and makes profit from the difference in negotiated rates.

Business Solution

PROVIDER Softswitch will offer end-to-end, cost-effective and scalable Wholesale VoIP solution. That solution will feature powerful billing, flexible routing, and proven interoperability with equipment from other leading VoIP vendors. Because all solution components will be developed by PROVIDER and Softswitch manufacturer we can eliminate integration issues and for that we can quickly start and/or expand our Wholesale VoIP business while enjoying high return on investment.



How will the solution work?

PROVIDER Experience

- PROVIDER will receive VOIP Traffic request from Partner 1 for termination.
- Then PROVIDER Softswitch will accept the traffic and will send authorization request to PROVIDER VoIP Billing server.

- The VoIP billing server will verifiy the account balance of Partner 1 and will authorize call termination.
- PROVIDER Softswitch will re-route the traffic to Partner 2, hiding traffic source information.
- Upon completing the call, the PROVIDER VoIP Billing server records will debit the account of Partner 1 and will record CDR record of the call.

Solution components are required

Core elements of PROVIDER Wholesale VoIP Solution include:

- PROVIDER VOIP Billing Server is a robust billing server that provides our customers with all necessary tools to successfully implement a wide spectrum of VoIP business models.
- PROVIDER Softswitch is an advanced VoIP softswitch that provides our customer with secure and reliable peering between their own networks and the VoIP networks of their business partners.

9.0 Some Important Componants

9.1 Softswitch

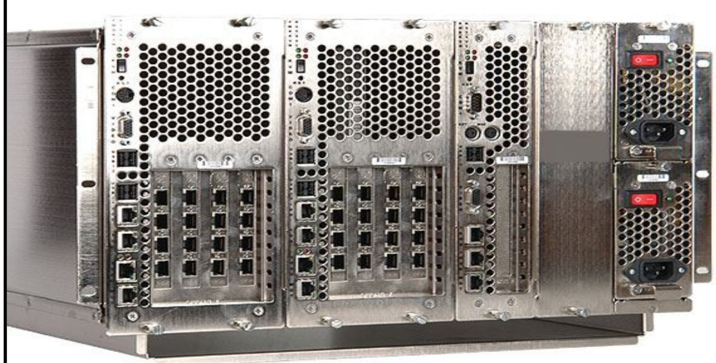
Softswitches are carrier-class servers that control voice phone calls across circuit- and packet-switched networks, which promise to support intriguing new services by dint of their open application programming interfaces for service software. The attraction softswitches hold is that they cost about a tenth of regular local phone switches, take up much less space, and have open APIs that enable third-party vendors to write complex services the switches control. They also enable completion of calls between circuit-switched and packet networks, enabling carriers to use a single packet backbone for voice and data traffic.

Switching lies at the core of all telecommunications networks, allowing efficient point-to-point communications without direct connections between every node. The Softswitch is a new software-based switching solution that runs on standard hardware to supplement or replace central office switching functions. Softswitches execute the same functions as traditional switches and are completely transparent to end-users. Telecommunications companies are embracing softswiches because they are functionally equivalent to conventional phone switches, only better, faster, and cheaper.

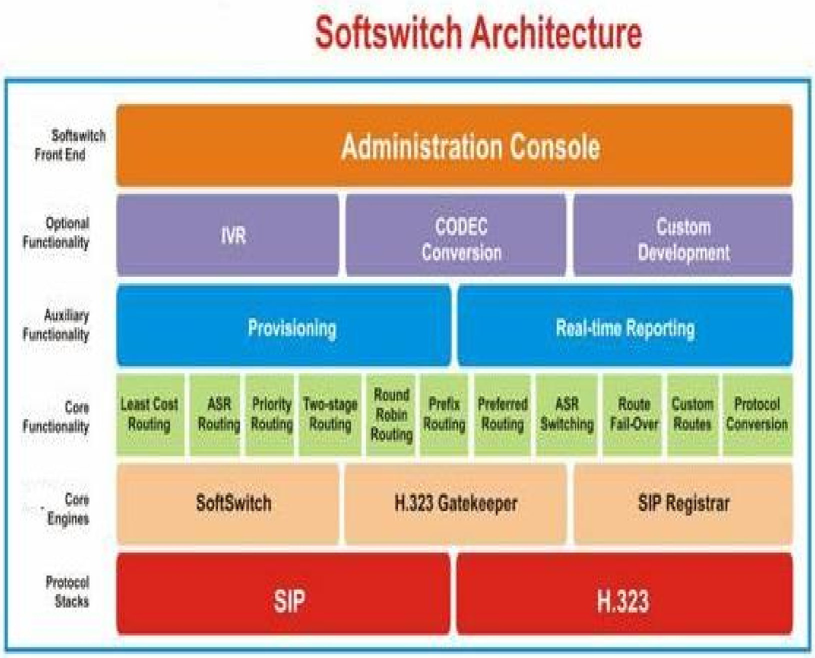
One characteristic of Softswitches is their open architecture, which provides great flexibility for carriers to develop custom solutions based on best-of-breed hardware and software components. Softswitches also tend to be modular, smaller, and less expensive than their conventional switching counterparts. This modularity makes scaling easy, critical when telephony markets and technologies can change overnight. As with conventional switches, high availability and reliability remain the top priority. Softswitches must also maintain compatibility with PSTN networks (Public Switching Telephone Network) and compliance with switching standards.

Softswitches provide several advantages over traditional switching alternatives. They reduce the cost of providing service by using less expensive IP networks. They allow carriers to differentiate their offerings through value-added services, often by simply adding a new server that delivers the desired functionality. Softswitches also allow telecommunications companies to leverage their existing investment in switching, preserve interoperability with PSTN networks, and assure a smooth transition to packet-based IP technology.

Provider SOFTSWITCH



The PROVIDER Softswitch that are going to install will have to offer flexible routing between VoIP networks, carrier grade reliability and high scalability. It will support multiple routing methods, including Least Cost Routing, ASR Routing, and Priority Routing which will enable PROVIDER to select the most profitable and high quality routes for each call, and increase call completion rates, revenues and profits. With this type of Softswitch, PROVIDER's will benefit from improved call completion rates, higher revenues, less revenue leakage, and improved network security and availability.

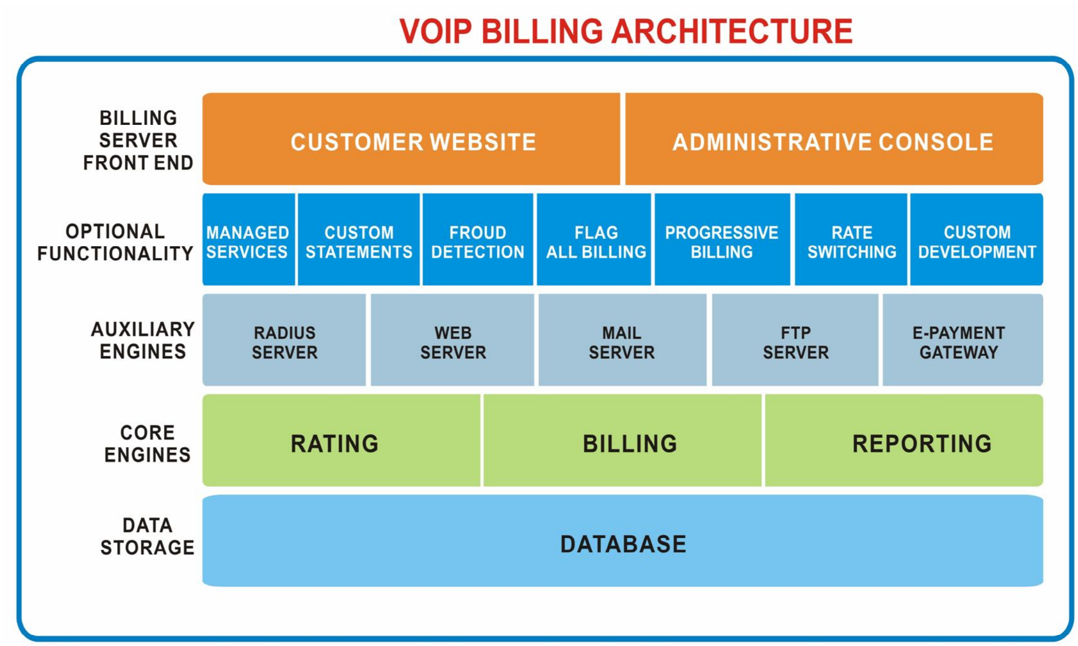


9.2 Billing Server

PROVIDER VoIP Billing

Product Overview

PROVIDER VOIP Billing will empowers service providers with all necessary tools to successfully implement a wide variety of business models. The Server will feature advanced VoIP billing capabilities, carrier grade robustness and high scalability to meet the needs of both emerging and established telecoms and service providers. With unlimited call billing capabilities, PROVIDER VOIP Billing will enable providers to fine-tune their service offerings and capture the most lucrative market segments.



Key Features

- Standard and Advanced VoIP Billing Functionality
- Advanced Management of Calling Cards
- Multiple Authentication Methods
- Selection of Call Legs for Billing Purposes
- Real-time Monitoring and Alerts
- Comprehensive Reporting
- Virtual Server Partitioning
- Carrier Grade Reliability
- Modular Architecture
- Unlimited Rate Tables
- High Call Capacity

9.3 PROVIDER Softphone

Product Overview

SoftPhone is a state-of-the-art telephony application which offers the functionality of a high-end telephone system from the convenience of user's PC desktop. The PROVIDER softphone what we are planning for can be used in place of IP phones and is designed to be an integral element of several solutions, including IP Centrex, Virtual Office, Broadband VoIP, and more. Embedding cutting edge technologies, like NAT traversal, centralized setup and provisioning, and centralized address book management, PROVIDER SoftPhone ensures simple and intuitive telephony experience for all end users.

Key Capabilities of the softphone

- SIP/H.323 PROVIDER Softphone Versions with NAT Traversal
- Customizable PROVIDER Softphone Skins and Phone Profiles

- Centralized PROVIDER Softphone Setup, Provisioning, and Management
- Direct Address Book Import
- USB Phone Support
- Centralized Address Book Management
- Direct Access to User Account Balance and Call History
- Access to Voice Mail, Conference, and Advanced PBX Services
- Customizable Soft Buttons
- History of Outgoing, Incoming and Missed Calls
- PBX Functionality Support and Soft Buttons
- Caller ID/PIN Authentication and Authorization

SIP/H.323 PROVIDER Softphone Versions with NAT Traversal

PROVIDER SoftPhone will have both SIP and H.323 version, both of which will support multiple voice codecs.

Customizable PROVIDER Softphone Skins and Phone Profiles

PROVIDER SoftPhone will support web based customizable skin templates. The functionality will allow clients to update the underlying HTTP and image files and can create customized/private label softphones. Additionally, clients can create multiple phone profiles with different design, language, and functionality and will use them to target different customer groups.

10.0 System Requirements & Costing

	System	Quantity
1	Softswitch	1
2	VOIP Billing Server	1
3	Central Database Server	1
4	CDR Backup Server	1
5	SIP Proxy Server	1
6	IVR Server	1

7	Gateway	(as per requirement)
8	Phones for Termination	15 E1
9	Colocation	1
10	Dedicated IP Block from APNIC with Very Small Membership	1
11	NOC & Customer Care Workstation	10

The Charge for installing the system:

	System	**Cost (USD)**
1	Softswitch	20000
2	VOIP Billing Server	10000-12000
3	Central Database Server	5000
4	CDR Backup Server	3000
5	SIP Proxy Server	12000
6	IVR Server	2000
7	Gateway	50000
8	Phones for Termination & Origination	50000
9	Colocation (1/2 Rack , 5 T1 , 10 MBPS of BW , /28 13 usable IPs and 16 Port Switch etc. monthly $1850)	4500
10	Dedicated IP Block	1250
11	NOC & Customer Care Workstation	10000

Profit Analysis

Project	**Target Call**	**Profit Margin per call (BDT)**	**Net Profit**
Origination	**28,00,000**	**.50**	**14,00,000**
Termination	**67,50,000**	**.75**	**50,62,500**
Carrier Whole Sale	**50,00,000**	**.20**	**10,00,000**

Note : The profit margin varies depends on the clients demand and market condition.

Payback:

Current forecast says the payback should be two month for Origination & termination packages prepared here.

Risk factor:

No Risk if legal made by Government.
Carrier Payment should be fair.
Any-user payment system should be clear.
Pay-rate are to be competitive.

Time Requirement:
The Total Project will take at least 2-3 months to give it the real structure.

Management:

We need skilled personnel to manage the system full time. The Softswitch and other solutions providers will train our executives to manage the system. We will manage our VOIP Network centrally so that we don't need any executive to manage each POP office location.

Telephones:

We will use PSTN, GSM, CDMA telephones channels for our origination and termination solutions. we will take E1/T1 connectivity from our local telecom company.

Expansion:

Initially we are starting 5 E1 for origination and 10 E1 for our termination solution. So with this network structure we can handle up to 300 calls for termination and 150 origination calls at a time. We will setup necessary POP depends on PSTN/ GSM/ CDMA connectivity for our network. Our company has to bear the costs of handling moves, adds and changes anywhere in our company.

11.0 Regulation

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 Submerine cable with in this year.

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