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Paper: Voice over Internet Protocol & Its Business & Growth Trends in Indian Market

Authors
Siddarth Kaul[1], Rajesh Saini[2]
Senior Unified Communication Engineer Polycom India
Assistant Professor Dept. of Computer Science and Engineering, Govt College Ateli Haryana

Abstract
Voice over Internet protocol (VoIP), there is an existing way of communication over any network. The Users can make the telephone calls over an IP network using this technology. This paper will describe Voice over Internet Protocol (VoIP) to a level that allows discussion of security issues and concerns. There are two kinds of spoofing attacks are possible, first one is IP spoofing attack and another is URI spoofing attack, which are described in this review paper. The Implementation of VoIP concerned by businesses, components of a VoIP system, and relevant security issues. The business concerns will be those which are used to affect the Quality of Service (QoS). The network components call processors, gateways and two of the more common architectures are held by VoIP.
*Keywords- VoIP, H.323, SIP, MGCP, QoS, Spoofing Attacks***.**

I. Introduction

To transmit voice conversations over a data network using IP, VoIP technology is used. Such data network may be the Internet or a corporate Intranet or managed networks which are specially used by long distance and local service traditional providers and ISPs (Internet Service Provider).

Voice over Internet Protocol (VoIP) is a form of communication that allows end-user to make phone calls over a broadband internet connection. Basic VoIP access usually allows you to call others who are also receiving calls over the internet. Interconnected VoIP services also allow you to make and receive calls to and from traditional landline numbers, usually for a service fee. A special type of adapter is used in some VoIP services which required a computer and a dedicated VoIP telephone. Other services allow to end-users to use own landline phone, it is used to replace VoIP calls. All these paradigms are held by a special adapter.

Voice over IP refers to the diffusion of voice traffic over internet-based networks. Internet Protocol (IP) was originally designed for data networking for purpose of its success, VoIP protocol has been adapted to voice networking.

The history of VoIP began with conversations by a few computer users over the Internet. Initially, VoIP required a headset to be plugged into the computer, and the participants could only speak with others who had a similar set up. They had to phone each other ahead or sent a text message, in order to alert the user at the other end of the incoming call and the exact time [2].

In November 1977, the IETF published the Specifications for the NVP (network voice protocol)'. In the preface to this document, the objectives for the research were explained as the development and the demonstration of the 'feasibility of secure, high-quality, low-bandwidth, real-time, full-duplex1 digital voice communications over packet-switched computer communications networks [3].

In the mid-90s, IP networks were growing, the technology had progressed and the use of personal computers had grown extensively. The belief that VoIP could start to make some impact on the market resulted in high expectations and the distribution of the first software package.

In its early stages, the VoIP technology was not sufficiently mature. There was a big gap between the marketing structure and the technological reality. It results in an overall agreement that technical shortages stopped any major transition to VoIP. However, VoIP is continued to make technical and commercial progress. The most of the technical problems have been solved by VoIP technology. There are no restrictions in the limited market conditions [4].

The communications network providers are used to adopt IP in their infrastructure, enterprises are adopting IP for private corporate networks. The communication between employees facilitate by using VoIP technique, whether working at corporate locations, working at home, or travelling. VoIP can also augment corporate efficiencies.

There are several enterprises which are used to test VoIP, doing a tryout, or engaging in incremental upgrades. The majority of multinational corporations use VoIP instead of remote possibility. The business opportunity will be a major part of their business operations in the near future [5].

This paper is divided into seven parts. Starting with introduction (Section-I), next section covers the implementation of VoIP (Section-II). Moving ahead, Configuration of VoIP is discussed (Section-III). After that VoIP attacks are discussed (Section-IV), How to Protect against Risks are discussed (Section-V). More over Requirements, Availability and Service Limitations are discussed (Section-VI) and finally, conclusions summarizes the last section (Section-VII).

II. Implementation of VOIP

In this section first we will discuss VoIP protocols and after that data processing in VoIP, at last we will discuss about quality of service in VoIP systems.

a. *Protocols*

There are currently three types of protocols which are widely used in VoIP implementations: the H.323 family of protocols, the Session Initiation Protocol and the media Gateway Controller Protocol (MGCP). The discussion of these protocols is as follows:

- *H.323 Family of Protocols*

H.323 [8], [9] is a set of recommendations from the International Telecommunication Union (ITU) and consists of family of protocols that are used for call set-up, call termination, registration, authentication and other functions. These protocols are transported over TCP or UDP protocols. The following figure.1 shows the various H.323 protocols with their transport mechanisms. H.323 family of protocol consists of H.225 which is used for registration, admission, and call signaling. H.245 is used to establish and control the media sessions. T.120 is used for conferencing applications in which a shared white-board application is used. The audio codec is defined by G.7xx series by H.323, while video codec is defined by H.26x series of specifications. H.323 uses RTP for media transport and RTCP is used for purpose of controlling RTP sessions. The following figure.2 & figure.3 shows the H.323 architecture and call set-up process.

- *Session Initiation Protocol (SIP)*

The modification and termination sessions between two or more participants the IETF is used which is defined by SIP (session initiation protocol) [9]. These sessions are not limited to VoIP calls. The SIP protocol which is a text-based protocol, it is similar to HTTP and offers an alternative to the complex H.323 protocols. SIP protocol become more popular in comparison to H.323 family of protocol because it is more similar than it. The following figure.4 and figure.5 shows the SIP architecture, call set-up and tear down process.

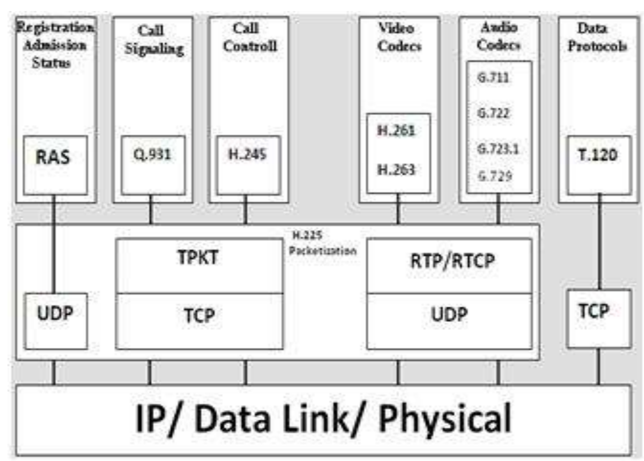

Fig.1 H.323 Protocol family [19]

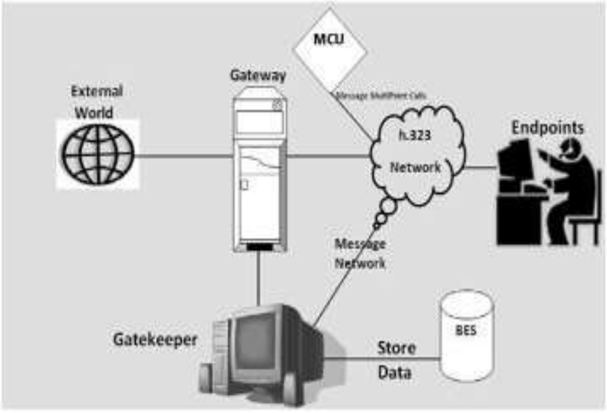

Fig. 2 H.323 Architecture [20]

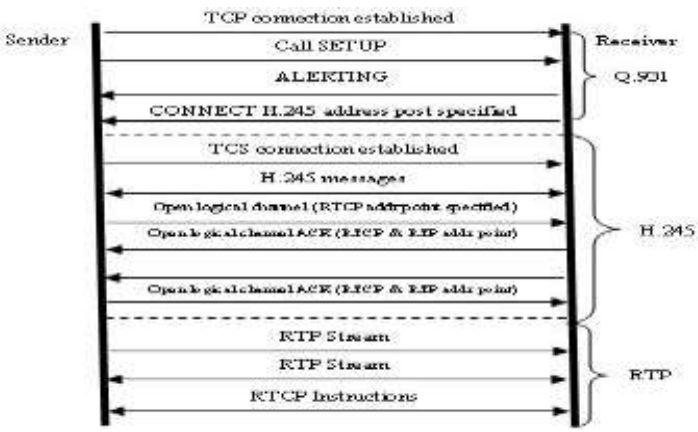

Fig. 3 Call Setup Process in H.323 [20]

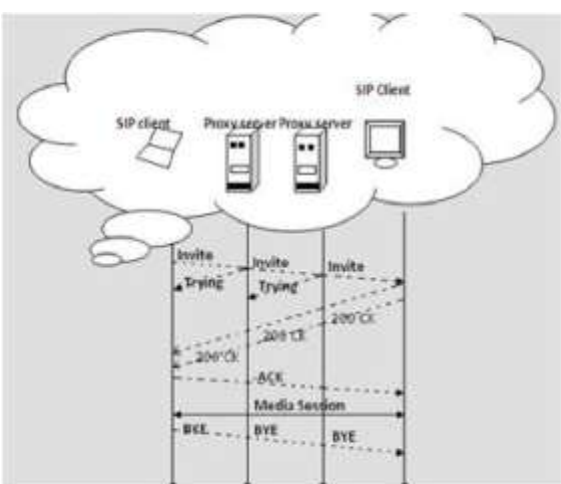

Fig. 4 SIP Network Architecture [9]

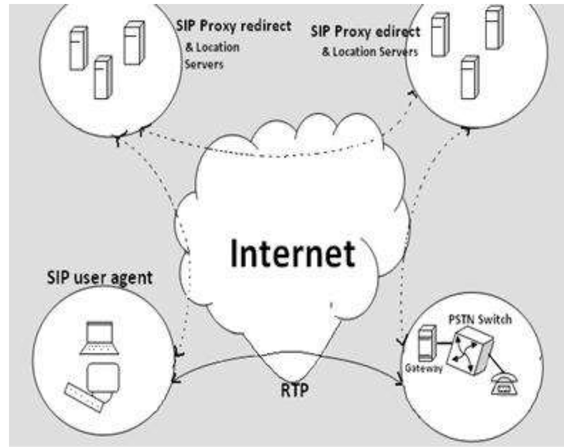

Fig. 5 Call setup and tear down in SIP [9]

- *Media Gateway Control Protocols (MGCP)*

The communication between the separate components of a decomposed VoIP gateway is done by media gateway control protocol. It is a complementary protocol to SIP and H.323. “Call agent” is mandatory and manages calls and conferences, when we are using MGCP and MGC server (Figure 6). The MG endpoint is not responsible for calls and conferences. It does not maintain call states. MGs are responsible to execute commands sent by the MGC call agents. MGCP assumes that call agents will synchronize with each other sending coherent commands to MGs under their control. MGCP does not define a mechanism for synchronizing call agents. MGCP is a master/slave protocol with a closely coupling between the MG (endpoint) and MGC (server).

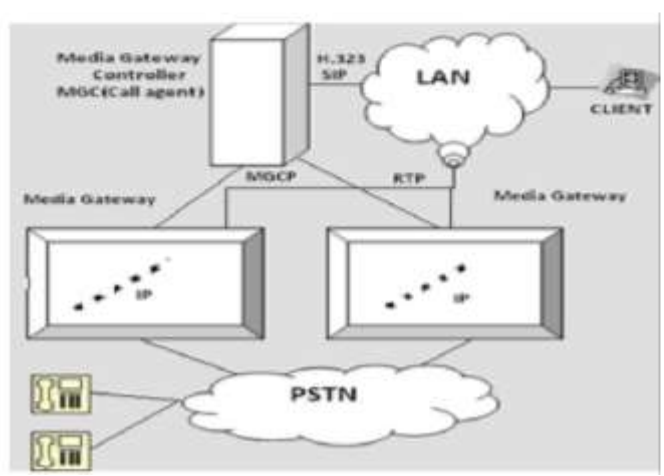

Fig. 6 MGCP Architecture [19]

b. *Data Processing in VoIP Systems*

There are three types of essential components in VoIP: CODEC (Coder/Decoder), packetizer and playout buffer [10], [11]. The analog voice signals are converted into digital signals at sender's side, after that these digital signals are compressed and then encoded into a predetermined format using voice codec. There are various voice codecs developed and standardized by International Telecommunication Union-Telecommunication (ITU-T) such as G.711, G.729, and G.723 etc. The packetization process is performed by distributing fragmented encoded voice into equal size of packets.

Furthermore, in each packet, some protocol headers from different layers are attached to the encoded voice. Protocols headers added to voice packets are of Real-time Transport protocol (RTP), User Datagram Protocol (UDP), and Internet Protocol (IP) as well as Data Link Layer header. In addition, RTP and Real-Time Control Protocol (RTCP) were designed to support real-time applications at the application layer.

Although TCP transport protocol is commonly used in the internet, UDP protocol is preferred in VoIP and other delay-sensitive real-time applications. TCP protocol is suitable for less delay-sensitive data packets and not for delay-sensitive packet due to the acknowledgement (ACK) scheme that TCP applies. This scheme introduces delay as receiver has to notify the sender for each received packet by sending an acknowledgement. The UDP protocol cannot be applied to VoIP technology. It is more suitable for VoIP applications.

The packets are then sent out over IP network to its destination where the reverse process of decoding and de-packetizing of the received packets is carried out. The time variations of packet delivery (jitter) may occur in transmission process. Hence, a play out buffer is used at the receiver end to migrate the package without any interruption. Packets are queued at the playout buffer for a playout time before being played. However, these packets continued to arrive until the playout time is discarded. The fig.7 shows the end –to- end transmission of voice in VoIP system.

Besides, there are signaling protocols of VoIP namely Session Initiation Protocol (SIP) and H.323. These signaling protocols are required at the very beginning to establish VoIP calls and at the end to close the media streams between the clients.

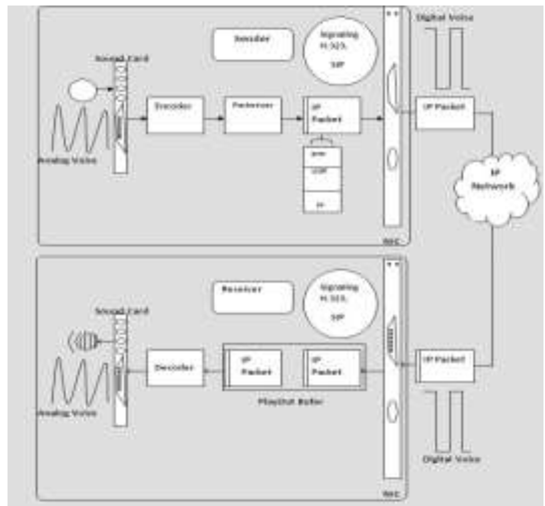

Fig. 7 End-to End Voice Transmission [3]

c. *Quality of Service (QoS) in VoIP Systems*

Quality of service (QoS) [3] can be defined as the network ability to provide good services that satisfy its customers. In other words, QoS is used for measurement of the degree of user satisfactions. When degree of user satisfactions is higher than it means the QoS is also higher. QoS are briefly described as given below:

- *Delay*

Delay can be defined as the total time it takes since a person, communicating another person, speaks words and hearing them at the other end. Delay can be categorized into three categories: delay at the source, delay at the receiver and network delay [3].

- *Jitter*

IP network does not guarantee of packets delivery time which introduces variation in transmission delay. This variation is known as jitter and it has more negative effects on voice quality [3], [4].

- *Packet Loss*

Packets transmitted over IP network may be lost in the network or arrived corrupted or late. Packets would be discarded, when they arrive late at the jitter buffer of the receiver or when there is overflow in jitter buffer or router buffer. Therefore, packet loss is equal to the total loss occurs during congestion of network and late arrival [5]. During the packet loss, the sender is informed to retransmit the lost packets. It causes more packet delay and it affects transmission QoS.

- *Echo*

In VoIP, Echo occurs when a caller at the sender side hears the reflection of his own voice after he talked on the phone or the microphone, whereas the callee does

not notice the echo. Echo is the term of the reflections of the sent voice signals by the far end. Echo could be electrical echo which exists in PSTN networks or echo of sound which is an issue in VoIP networks [6].

- *Throughput*

The throughput may be defined as the maximum number of bits received out of the total number of bits sent during an interval of time.

Internet telephony in india

The sub-committee on telecommunications, headed by Planning Commission advisor Montek Singh Ahluwalia, is likely to recommend liberalisation of Internet telephony or voice on Internet (VoI) norms, paving the way for issuing Internet telephony licenses in a year. In the New Telecom Policy 1999, the government had not permitted Internet telephony. However, the decision was kept subject to review. Similarly, in its guidelines for Internet service providers (ISP), the department of telecommunications (DoT) had said that the licence would be liable for termination following any violation. Experts believe that there may be some opposition from the Videsh Sanchar Nigam Ltd, as Internet telephony will end its monopoly. Also, Communications Minister Ram Vilas Paswan has said[9] that Net telephony would strictly be kept off the reach of private ISPs to safeguard government revenues.

A study by a London-based telecom consultancy had projected in 1997 that VSNL would lose $54 million (about Rs 195 crore) by the year 2001 if Net telephony were legalized. VSNL has until now blocked out sites on the Net which offer voice telephony services.

Small players in India (e.g., Premiere Infosystems, a Noida tech start-up) have started offering installation of software for voice services over the Net at less than 50 paise per minute. To put this in perspective: an international one-minute call on a normal basic telephone would cost some Rs 75. The infrastructure required at the subscribers' end is remarkably limited: a computer-multimedia kit and Net telephony software. Apart from a computer and a high-speed modem, users indulging in Netspeak need to install a sound-card (which converts sound into electronic signals and vice versa), a microphone, speakers and Net telephony software. The software like Net-To-Phone can be easily downloaded from the Net.

The cost of installing the software is less than Rs 2,000, after which every call domestic or international long distance would be billed as a local call: maximum of Rs 1.40 per minute. Calls could be made to other telephones or to other computers rigged up with a multimedia kit.

Significant interest has been shown by various IP telephony players in India, despite the regulations. In July'00, Israel based Arelnet Limited, leading provider of IP telephony solutions, expressed its intentions of launching VoIP gateways in India, for small and medium-sized ISPs.[10]

The modes of proliferation of VoIP in India are primarily three

1. Traditional Dial Up Internet (through an ISP)
2. Cable access (through a cable modem through existing cable TV lines)
3. Phone to Phone (Major providers could be existing established players like VSNL and DOT)

1. Traditional Dial Up Internet

Internet Service Providers connect customers to the Internet. For a particular access fee, the service provider provides an installation software, a username and password and access phone number.

National ISPs

These ISPs operate points of presence throughout the country. One category of national ISPs own the network backbone and lease the international connectivity while the other category of players lease the network and the international connectivity from other ISPs.

The main target segment for these ISPs is the corporate segment. A presence throughout the country helps these ISPs to cover all the locations for a particular corporate. ISPs owning the network backbone use the reliability of service (due to ownership of the network) as a key USP to customers. This is helpful, especially for real-time mission-critical applications.

Regional and local ISPs

These are the ISPs that either operate in the smaller towns or particular states. These ISPs serve both the business and consumer segments usually within a geographic region.

The ISP scenario in india

~~Internet subscribers in India~~ are expected to reach a figure 5,30,000 by March this year, according to IDC. This figure is expected to touch 1.3mn by March 2001, a growth of 145% over the previous year. Dataquest (DQ) estimates are slightly more optimistic. DQ expects the Internet subscriber base to reach 6,55,000 by March 2000. DQ expects this to shoot up to 1.86mn by 2001 and 3.75mn by 2002. DQ's estimates say that the ISP access market was worth Rs1.02bn in 1999 and is likely to touch Rs2.13bn in 2000. The market will expand sharply to touch Rs8.37bn in 2001 and to Rs15bn by 2002.

Falling PC prices, coupled with a drop in the access rates (due to the price war in the ISP market) has caused this growth. To some extent, future growth will also be assisted by Internet access through cable TV. Estimates are that three to four years down the line, India, with 30mn Internet users, will have the most Internet users in Asia, next only to China.

The last 12 months has seen a good growth in the usage of Internet by the corporate segment also. The factors, which have helped this, is the lowering of leased line charges by TRAI and the offering of a variety of value-added services by the ISPs. While earlier the corporates were using the Internet more as an information provider – email, surfing etc, many of them are using this to do some kind of e-commerce especially on the vendor end.

According to the IDC survey on 31st July 1999, the small / medium organizations had the largest portion of the subscriber base. Large organizations were next, with a subscriber base of nearly 97,000. The home segment showed a healthy growth in its subscriber base share over its November 1998 share of 8.9%. In the next few years, the home and small/ medium sized segments will experience unprecedented growth in terms of Internet connectivity.

PC / Internet Penetration comparison

	India	US	Asia-Pacific
Population	1000	270.3	2769.6
Number of Internet users	0.5	62.8	10.2
Net-enabled PCs	0.3	87.4	9.5
Internet users / population	0.1%	23.2%	0.4%
Net-enabled PC / household (%)[11]	0%	129%	1%

Growth of the corporate access market is also expected with large domestic computerization measures in the Government sector. Already initiatives are underway on the part of the Government to start a comprehensive move towards electronic governance. The corporate access market will also grow as the economy gathers momentum and companies realise the benefits of using the Internet.

2. Cable ISPs

These ISPs are normally owned by the cable companies that help them to get exclusive use of the cables. (The cable IP telephony technology has been explained in detail earlier)

Cable access ISPs offer broadband Internet connectivity through the coaxial fiber networks. Cable ISPs usually service the consumer segment of the market. The parent makes money from the lease charges of the cable while the ISP uses the access / e-commerce / advertising revenues to compensate for the hiring charges.

The **development of alternative means of access** is also expected to give Internet usage a boost. The most obvious way is using the TV Cable network, given the enormous TV penetration in India. Currently, 30mn Indian households have a TV. If this segment of the population can be tapped, Internet usage can explode. Currently, set-top boxes, which connect users to the Internet through the TV, are quite costly. However, with increasing penetration the cost of these boxes should fall and help increase ISP demand.

Indian Cable Television Industry

The cable industry however is growing at what some consider a chaotic rate; because entrance barriers are low, the industry is open to almost anyone. Currently there exist over 100,000 cable operators in India (compared to 10,000 in the US) employing over 1 million people. Ninety-seven percent of cable operators have less than 1,000 subscribers, most having less than 500. Presently, an initial investment of Rs.250,000 ($7,287) covering a dish, signal receiver, mixer, amplifiers and a modulator to convert frequency is all that is needed. The only other expenses are installation and small VCR movies fee: all satellite signals are received free. Eighty percent of Indian cable systems carry between six and eight channels, fifteen percent between ten and twelve, and five percent carry more than twelve. Over 30 million Indian households receive cable serving an audience of well over 125 million, the second largest cable market in Asia after China. For this reason, the Cable Television Networks (Regulation) Bill was presented to the Indian Parliament and passed December 13, 1994.

Hence, the medium will be a powerful mode of Internet penetration, especially of broadband services. Internet over cable is picking up very fast in India this year. Some of the major service providers are Hathaway in Mumbai and Spectranet in Delhi. However as compared to the traditional dial-up services, the subscriber charges are around Rs.1000 per month for unlimited access. These are in addition to the cost of a cable modem.

Once IP telephony is permitted in India, we feel that Cable IP will be a major mode of proliferation of IP telephony services. The current subscription rates of Internet over cable are targeted towards corporates and high-income consumers. However, subscribers may retail these services to other customers (who can't purchase these services themselves) in future "IP booths". Also, these services may be combined with " cyber cafes" which already provide Internet services to customers.

Lack of a proper telecom infrastructure has kept the Internet penetration down to only the major towns and cities. With the setting up of a National Telecom Backbone, it is expected that Internet will reach more areas of the country. Higher bandwidth availability is also expected to spur the usage of Internet in terms of both number of subscribers as well as the usage time by a particular user.

Key Features of the Indian Internet Policy

1. Licensee has the freedom to lease domestic backbone from DOT, basic service providers, SEBs, the power grid corporation, railways or any other authorized operator.
2. License fee is absent for the first five years and is nominal (Re.1) after that.
3. The decision on tariffs has been left to the ISP providers. However, TRAI has the right to review and fix the tariff anytime during the licensee period of a licensee.
4. The ISP policy does not allow for Internet telephony.

The government has issued in principle

clearance to all ISPs who have applied for international gateway license using satellite technology. The government has given around 225 ISP licenses so far.

Retail Segment

In the retail market, the competition is likely to be fierce because the only service provided is a plain vanilla service and, therefore, there is very little differentiation, which is possible. So the competition would be around price.

Since extent of value addition is very low here, the only way to compete is by reaching economic volumes, so it is essential to make large upfront investments in infrastructure and service and capture as much of the market as possible.

Corporate Segment

Corporate customers have relatively lower price elasticity towards access charges. As most of them use the voice and facsimile communication to support business critical applications their focus is on service rather than costs. Corporate customers are easy to retain because service disruptions due to a change in vendor can be extremely costly. Thus, entry barriers in this market are very high. This is also because a good track record is essential to get business from corporates.

With e-commerce becoming a buzzword, IP telephony helps in creating a completely new class of service such as web-enabled call centers, telecommuting and long distance learning.

Presently IP Telephony is banned in India. Unlike US there has not been any comprehensive debate on the issue. Following are some of the factors that are driving the government stance:

1. International Accounting Rate System

As per the prevalent International Accounting Rate System, India benefits from heavy incoming calls compared to outgoing calls. Under this system whichever country has net incoming calls gets paid for the same. The United States sends billions of dollars abroad as a result of such international settlement rates and a significant chunk of it comes to India. While IP telephony could save America billions of dollars, possibly a significant portion of the size of a federal universal service fund, India stands to lose out the precious foreign exchange, which it presently gets due to exorbitant rates being charged by VSNL from consumers making international calls.

2. Effect over investment in traditional telecom

After liberalisation of telecom sector in 1994, heavy investment has been made in private and public sector. These operators of basic and cellular telephony are required to pay heavy license fees in the tune of thousands of crores every year. If IP Telephony is allowed to become popular and unregulated, it will have adverse impact on the present telecom player and not only the investment made in developing telecom structure may parish but even government may lose out a major source of revenue.

3. Level Playing Field

Another major concern of Indian Regulators is to provide level playing field to traditional telephony operators and IP telephony operators.

A complete scenario of Indian telephony is not possible without considering the options before the incumbents, MTNL, DTS and DOT when voice services over the Internet are allowed. For this purpose, the strategy adopted by the largest U.S. telecommunication provider, AT&T in the wake of IP Telephony was studied.

The Telecommunications Act of 1996, which allowed local and long distance phone companies, Internet firms, and cable businesses to compete in each others' markets, immediately led way to a long list of mergers and acquisitions.

AT & T has set itself the objective of becoming the leader in end-to-end communication services. In March 1999, AT&T made a $62 billion cash bid for MediaOne, one of the largest suppliers of broadband services on cable. The driving force for this merger is that AT&T wants to take advantage of recent consolidation in the industry and changing technology. The purchase of the Cable Company was not only intended to bypass the Regional Bell Operating Companies[12], but also to deliver a full menu of phone, Internet and multimedia entertainment.

If the deal goes through, it will create the largest Cable Company in the U.S., with over 16 million subscribers. It is now planning to provide Cable IP telephony services.

Hence, it is clear from the above moves that AT&T is unfazed about the arrival of new technologies as substitutes for its services. Instead it has proactively adopted these new technologies so that it is not left behind. It plans the migration of its services from the old technology to the new as it upgrades its network across the U.S. for digital services.

We believe that it is futile for the incumbents in India to restrain the growth of IP telephony as it may not be possible to restrain users from accessing and using the services in light of the significant price differences. Given the fact that international gateways for private ISPs have been allowed, it would become more difficult to regulate the rendering of these services. In contrast, the incumbents should use their existing infrastructure and first-mover advantage to enable themselves to expedite these services on their systems. Such a process would involve the following: -

Technology assessment

Establishment of Additional Infrastructure (Gateways, Gatekeepers, etc.) Deployment of Software

Pricing of ip telephony

The issue of pricing IP Telephony is closely linked with the pricing of Internet services as a whole if viewed as capable of providing different kinds of services to the users.

One opinion is that we have survived so far. There is rather substantial experience with the current model, and it seems to meet the needs of many users. However, others believe that once commercial Internet service becomes mature, customers will start to have more sophisticated expectations, and will be willing (and indeed demand) to be able to pay differential rates for different services. Indeed, there is already evidence in the marketplace that there is a real need for service discrimination. The most significant complaint of real users today is that large data transfers take too long, and that there is no way to adjust or correct for this situation. People who would pay more for a better service cannot do so, because the Internet contains no mechanism to enhance their service.

In case of IP Telephony this debate becomes more critical because "voice packets" generated from IP Telephony users needs to be routed fast the other end to maintain semblance of real time communication. For this priority routing the "voice packets" needs to be given priority by the routing mechanisms.

In 1994 Jeffrey K. MacKie-Mason and Hal R. Varian (University of Michigan) advocated the usage based pricing of Internet and presented the concept of "smart market". They proposed a way to price network usage that they called ``smart markets." Much of the time the network is uncongested; at such times the price for usage should be zero. However, when the network is congested, packets are queued, delayed, and dropped. The current queuing scheme is FIFO. They propose instead that packets should be prioritized based on the value that the user puts on getting the packet through quickly. To do this, each user assigns her packets a bid measuring her willingness-to-pay for immediate servicing. At congested routers, packets are prioritized based on bids. In order to make the scheme incentive-compatible, users are not charged the price they bid, but rather are charged the bid of the highest priority packet that is not admitted to the network. (Vickrey Auction) It can be shown that this mechanism provides the right incentives for users to reveal their true priority.

David D. Clark (1995)[13] also discusses a scheme where bandwidth is allocated among users by creating different service classes of different priorities to serve users with different needs. The definition of priority is that if packets of different priority arrive at a switch at the same time, the higher priority packets always depart first. This has the effect of shifting delay from the higher priority packets to the lower priority packets under congestion. If this argument is pushed further by redefining priority where high priority packets (voice packets) are sent first even if they arrive a little later than the other packets (normal data packets).

Clark further says that the slowing down an individual packet does not much change the observed behavior. But the probable effect of priority queuing is to build up a queue of lower priority packets, which will cause packets in this class to be preferentially dropped due to queue overflow. The rate adaptation of TCP translates these losses into a reduction in sending rate for these flows of packets. Thus, depending on how queues are maintained, a priority scheme can translate into lower achieve throughput for lower priority classes.

This might, in fact, be a useful building block for explicit service discrimination except that priority has no means to balance the demands of the various classes. The highest priority can preempt all the available capacity and drive all lower priorities to no usage. In fact, this can easily happen in practice. A well tuned TCP on a high performance workstation today can send at a rate exceeding a 45 mb/s DS-3 link. Giving such a workstation access to a high priority service class could consume the entire capacity of the current Internet backbone for the duration of its transfer. It is not likely that either the service provider or the user (if he is billed for this usage) desired this behavior.

The other drawback to a priority scheduler (where priority is not based on price) for allocating resources is that it does not give the user a direct way to express a desired network behavior. There is no obvious way to relate a particular priority with a particular achieved service. Most proposals suggest that the user will adjust the requested priority until the desired service is obtained. *Thus, the priority is a form of price bid and not a specification of service (which is similar to the smart market pricing proposed by Hal R. Varian (1994)).* It is much more direct way to let the user directly specify the service he desires, and let the network respond.

Above concept of usage based pricing may be the fundamental of IP Telephony pricing where 1 IP Telephony voice packet transmission (with priority routing) price will be equal to x normal data packet transmission price. At times when network is uncongested and spare capacity is available the value of x will be around 1 but when the network is congested and capacity is being utilized fully, the value of x will high and cost of may even approach cost of long distance telephony on traditional network.

However, such dynamic pricing and billing for every packet may not be possible for IP Telephony service provider (ISP, Cable operator) but this can be approximated by putting different slabs for different values of x based on historical congestion (magnitude and timings). This mechanism is somewhat similar to what is being followed for peak and off-peak hours in traditional telephony.

Though it is difficult to develop the above-mentioned dynamic model, static models where values of x can be varied depending over network congestion can approximate it.

In this section the cost of IP telephony if implemented in India has been estimated based on the following explicit assumptions:

Key Assumptions

64 kbps ISDN connection for one IPT connection (this is very estimates. In fact 64 kbps should be sufficient to allow user to make two IP telephony calls at the same time. This conservative estimate has been taken in view of congested network condition in India that prevails most of the time and it is felt that 64 kbps connection will make the last mile network congestion free for good quality voice communication by providing excess bandwidth at user end. Another reason for taking ISDN costs is that capital investment very closely approximates

capital cost of IP Telephony ports for phone to phone telephony)[14].

1 data packet of IPT is equivalent to x nos. of normal data packet (as discussed earlier this variable is a function of network congestion, which may vary over time in a day).

Fixed cost recovery in 4 years (Again it appears a bit conservative but has been taken as the rate of change of technology is quite fast and may make investment obsolete).

Cost estimates are for ISP. Local call charges Rs. 1 per minutes (to make it unsubsidized)
1000 data hours have been assumed for a year and price has been taken for the same per ISDN connection

How cheap is IP Telephony compared to traditional telephony?

To find out the actual discount available from IP telephony at present, prices of per minute call from US to 180 countries for IP telephony were compared with the traditional telephony prices (Exhibit 3). Based on analyses of pricing data (US to180 countries) available from PC to phone companies (net2pnone and deltathree) with traditional telephony provider (Telecom International Ltd.), it was found that IPT is around 25% (with stdev. 13%) cheaper on average (for major destination it is 40-50%). This discount can be increased as presently part of the path is on traditional telephony network. Once a seamless phone to phone VOIP connection infrastructure is in place, this price is further expected to come down.

Major Hurdles

One of the major hurdles (other than clear regulatory framework) in roll out of VOIP network is high cost of IP Telephony port ($ 1000 - 2000 compared to $100-200 for traditional telephony port). Though it is likely to come down in near future, it is also to be considered that cost of traditional telecom equipment is also falling.

Possible Answers

One of the positive approaches to make most of existing VOIP network is to use yield management in IP Telephony.[16] It is true that providing 100% available connection to all the users for 24 hours will not be possible at a cheaper cost and the capital investment will be prohibitive. This can be overcome using different values of x for different network congestion conditions. This approach coupled with falling hardware prices is can make IP Telephony a cost effective solution for the communication for the masses.

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