Project Report
on
IP Telephony

Executive Summary

This paper examines the issues regarding the advent of IP telephony, which is one of numerous good inventions that technology has provided to enhance the welfare of human beings. IP telephony represents an efficient and inexpensive technology that will make a difference in the way that companies approach their clients, and will in the near future allow people to communicate in a more efficient way.

The paper examines the evolution of IP Telephony as an industry and the hurdles it has overcome to gain its current state of acceptability. We shall be focussing on three key aspects in the paper.

Firstly, the nature of the industry makes it highly vulnerable to technological changes. It is thus important for us to look at the technical backbone, which has been responsible for setting up this unique paradigm. We will examine the continuously changing nature of the technology, its comparison with other Voice transfer technologies and the future tend in this field.

Secondly, we shall aim to highlight the pricing benefits of IP telephony by examining a model developed for costing the service. The paper shall also examine the economic aspects of putting an IP telephony network in place and its impact on communication services.

Lastly, the focus of this paper will be on examining the various regulatory stances adopted by governments across the world and the reasons for their divergent views. It will also look at the issues facing the regulators in various countries in the future. The

major challenge to developing countries' regulators will be to continue the solid expansion of the basic telecommunications infrastructure while allowing new technologies like IP telephony to develop in a competitive environment.

The paper will finally summarize the growth potential for this industry and the effect it will have on the communications business.

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INTRODUCTION

Since the 1960's when digital voice communication first emerged, the Public Switched Transfer Network (PSTN) has been supported worldwide as the primary means of voice communication. The PSTN is a connection-oriented, circuit-switched network in which a dedicated channel (or *circuit*) is established for the duration of a transmission. Originally transmitting only analog signals, the PSTN ultimately switched to using digital communication, which offered solutions to the attenuation, noise and interference problems inherent in the analog system. The modern PSTN uses Pulse Code Modulation (PCM) to convert all analog signals into digital transmissions at the calling end office that initiated communication and reverses the processes at the receiving end office.

Although highly rated for reliability and Quality of Service (QoS), the PSTN has two significant disadvantages:

?? Expensive bandwidth, which results in high telephone bills for individuals and businesses alike;

?? Inefficient use of networking channels, which results from dedicating an entire channel for each conversation.

Internet Protocol (IP)-based networks are the latest innovation to offer solutions to such problems and are increasingly being used as alternatives to the traditional circuit-switched telephone service. IP Telephony provides alternative means of originating, transmitting, and terminating voice and data transmissions that would otherwise be carried by the public switched telephone network (PSTN).

IP telephony has a relatively short history, which began with Vocaltec Inc.'s introduction of its Internet Phone software in February 1995.[1] For the first time, this software enabled analog signals converted into digital IP packets to be transmitted over the Internet. By using the Internet, which is connectionless-oriented and packet-switched based,

[1] http://isds.bus.lsu.edu/cvoc/Projects/TechLibrary/VoiceOver/history.htm

businesses could eliminate much of their communications expenses. In addition, they could use the bandwidth of their communications channels more efficiently through multiplexing of audio, video and data. The new software quickly captured industry interest, created market demand for the communications capability and sparked competition among IT vendors.

Like virtually all-new technology, the first VoIP software had problems. First, there was a "lack of connectivity between an IP telephony network and the PSTN"; IP telephony could successfully occur only when both parties were using the same software. However, this problem was remedied by the development of gateways, which contained voice-processing cards and provided the necessary interface between the IP telephony software and the PSTN. The second hurdle was the QoS standard that most businesses expected for communications. Unlike PSTN, the packet-switched nature of the IP protocol hindered reliability and posed a significant concern for the level of quality. Although newly developed network architecture components, such as gateways and routers, have improved overall quality of communications for intranets, the lack of standards governing VoIP and the current status of the Internet architecture continue to present problems. Nonetheless, VoIP is still considered a powerful communications tool, and new advances in the technology are being made quickly. Thus, the perpetual search for competitive advantage continues to serve as the primary motivation behind increased investment and research into this new technology.

Internet Telephony Service Providers (ITSPs) can provide voice and fax services that are close to becoming functionally equivalent to those provided by public telecommunication operators (PTOs). However, national authorities license few ITSPs and they generally do not have any universal service obligations. Many countries ban IP telephony completely, yet IP calls can be made to almost any telephone in the world. Many PTOs are establishing their own IP telephony services, and/or using IP-based networks as alternative transmission platforms.

Internet Service Providers v/s Public Telecommunications Operators

In order to approach the regulatory discussion, is important to indicate the general comparative advantages of Internet Service Providers (ISPs) over Public Telecommunications Operators (PTOs). While some of these advantages are technical or economical, a great part is based on regulatory discrepancies that will be analyzed in the next section of this paper.

Licensing Advantages

While the PTO has to compete for a license issued by the national regulator or ministry, to be an ISP rarely requires a license. Further, frequently the PTO has to pay in order to get that license, which also often comes with regulatory obligations like price control and universal services contributions. This is a major regulatory concern that will be analyzed below. However, it is not a common problem worldwide, rather it is a major concern for the countries that have not committed themselves to a total liberalization.

Networking Requirements.

The PTO has to establish a network of fiber, copper, microwave or satellite links, while the ISP can lease a circuit to the closest PoP. Moreover, the PSTN technology obligates the PTO to have excess of capacity to meet peak loads, in comparison to the ISP who can add extra bandwidth as demand grows.

Pricing Concerns.

The PTO has to connect each subscriber at the same price, regardless of location and in conformity with the regulator price structure. In contrast, the ISP can introduce numerous price structures depending on the customer and whether the customer is willing to lease his or her own line. Further PTO has to bill customers according to their level of usage while ISP can charge customers at a flat rate depending on the bandwidth capacity of their connection.

Traffic Concerns.

The PTO has to establish a hierarchy of central office exchanges and an international gateway for international traffic. In the other hand, the ISP can lease routers and servers, adding modems as needed.

Accounting Rate Concerns.

The PTO has to pay accounting rates and negotiate settlement arrangements with carriers in each country where the traffic is directed. Instead, the ISP can negotiate peering and transit arrangements with local point of presence. However, in the case of computer to phone or phone-to-phone IP telephony the ISP will have to negotiate arrangements for termination with other ISP. In any case, the ISPs do not pay accounting rates. This is a second major issue, especially for developing countries. The FCC under ITU and bilaterally has been pushing hard in order to lower the accounting rates to cost levels. However, the process has been slow due to the big opposition from the regulators and operators in these countries.

What is IP Telephony?

Internet Protocol (IP) Telephony is the transmission of voice, fax and related services over packet-switched IP-based networks. In the longer term, as more and more voice traffic becomes IP data traffic, there will be little to distinguish between IP telephony and circuit-switched telephony. However, many telecommunications regulatory schemes depend upon such a distinction, both physically and as a matter of policy and law.

In order to address the regulatory battle between traditional PSTN, Public Switched Telephone Network, voice services and IP telephony, it is important to underline the differences between these two technologies. The bases of future regulations will have to attend to not just the final product to the user, but also to the technological structure of the services provided. The following description provides a basic introduction for those not familiar with voice technologies.

Traditional Telephony

The widespread traditional telephone service is based on the PSTN, relying on circuit switching to reach end users. Every time that the user makes a call a dedicated circuit transports the sound waves in form of electric signals. In user’s terms a "line" is dedicated exclusively for the communication between the two actual users, regardless how efficiently they use it. In other words, even if the users keep silence for a period of time, the line remains dedicated to these particular users, at the same time is unavailable to others. This inefficiency is unavoidable under PSTN.

Long distance service based on PSTN begins at the user's phone, then the signal travels through local telephone company lines to the long distance company’s point of presence (PoP), where the signal is carried by the long distance company to the PoP based on the callee's local area. The local telephone company carries the signal finally to the other user’s phone.

IP Telephony

IP Telephony rather than circuit switching is based on packet switching. The voice transmission, instead of using a dedicated line, is broken in numerous packets. These packets look for the most efficient route individually, and later are reorganized at the IP address where the voice was originally sent. This technology allows numerous users to send information over the same line, providing with a more efficient utilization of the telecommunications infrastructure.

IP Telephony – Comparative Advantage

The overall advantage of IP telephony comes from treating voice as another form of data. While claims that the PSTN is dying are premature and unfounded, the advantages presented by IP telephony are clearly visible today:

More Bandwidth: One advantage of IP telephony is that it dramatically improves efficiency of bandwidth for real-time voice transmission, in many cases by a factor of 6 or more. This increase in efficiency is a real long-term driver for the evolution from circuit-switched to packet-switched technology.

New Services: Another advantage IP telephony has over the PSTN is that it enables the creation of a new class of service that combines the best characteristics of real-time voice communications and data processing, such as web-enabled call centres, collaborative white-boarding, multimedia, telecommuting, and distance learning. This combination of human interaction and the power and efficiency of computers is opening up an entirely

Progressive Deployment: The final advantage of IP telephony is that it is additive to today's communications networks. IP telephony can be used in conjunction with existing PSTN switches, leased and dial-up lines, PBXs and other customer premise equipment (CPE), enterprise LANs, and Internet connections. IP telephony applications can be implemented through dedicated gateways, which in turn can be based on open standards platforms for reliability and scalability.

Internet Telephony v/s VoIP

The most important threshold issue relating to IP Telephony concerns definitions. Divergent views are expressed over the definition of IP Telephony, whether as a technology or as a service concept. IP Telephony can be subdivided into two major groups: Internet Telephony and Voice-over-IP (VoIP). The distinction between Internet Telephony and VoIP lies in the nature of the underlying means of transmission or the underlying IP Network. IP Telephony is generally employed as a generic term covering both. The following definitions are offered as a means of interpreting the many different terms that are thrown about in this field:

?? **Internet Telephony**: IP Telephony in which the principal transmission network is the public Internet. (Internet Telephony is also commonly referred to as “Voice-over-the-Net” (VON), “Internet Phone,” and “Net Telephony” – with appropriate modifications to refer to fax as well, such as “Internet Fax”).

?? **Voice-over-IP (VoIP)**: IP Telephony, in which the principal transmission network or networks are private, managed IP-based networks (of any type). (Depending on the type of network, you can have “Voice-over-frame relay,” “Voice-over-cable,” and “Voice-over-DSL” or “VoDSL,” as examples).

Even within these two broad groups, there is a potentially infinite number of ways to use IP technology to provide different services in different ways. Therefore, services are further classified according to the nature of the terminal devices used (e.g., computer or telephone).

Types of IP Telephony

While the emergence of IP Telephony is often associated with the rise of the Internet itself, it is important to appreciate that IP Telephony often does not involve the public Internet at all – but rather only its underlying technology, the Internet Protocol suite. Different types of IP Telephony can be identified according to the type of terminal used, where gateways are located, and the underlying transmission means.

IP telephony has four main categories:

- ?? Personal Computer (PC) to Personal Computer,
- ?? Personal Computer to Telephone or Fax,
- ?? Personal Computer to a content provider's call center or website
- ?? Telephone-to-Telephone communications via the Internet.

PC-to-PC Internet Telephony

This is the most well known form of IP telephony. It has been developing since 1994 and is used mostly by young cyber addicts. A microphone connected to the computer receives the voice, and then the computer transmits the voice through the Internet. The key component for this service is software that processes the voice into IP data, later reconstructed into voice waves at the end user's computer. The quality of the service was initially low and it needed the two users to be on-line at the same time. This form of IP telephony has not been a real contender to traditional PSTN telephony service.

PC to Telephone or Fax Telephony

The first steps towards developing this technology began around 1996. It is based on the same concept that PC-to-PC IP telephony but uses a gateway bridge between the PSTN and the Internet. The basic understanding of gateways is a computer that changes the circuit-switched signal into packets and from packets into a PSTN signal again. This technology allows users to call from a computer interconnecting with the PSTN to someone who is using a regular phone.

PC to a Content Provider's Call Center or Website

The ITU considers this form of Internet Telephony "the largest application market in the long term because offers a degree of voice/data integration which has long been an elusive goal for the telecommunications industry." The user can access a company's website and request to speak to a representative at any time by just clicking the appropriate icon in the screen. The architecture of this service can be based either on the PC to PC IP telephony format if the representative is using a computer, or PC to Telephone IP telephony format if the representative is part of a call centre. The natural advantages of this service included the interaction of voice and data in the same communication, which will permit a much better customer service.

Telephone-to-Telephone IP Telephony

Using this technology the caller uses a normal phone, and then the signal travels through the local telephone infrastructure to a gateway based on the local area. The gateway sends the signal converted into IP protocol packets over the Internet. The voice packets travel via the Internet related to the other gateway located at the callee's local area. Then the signal travels over the local telephone infrastructure to the callee's regular phone. This paper's analysis will focus principally in this IP telephony category, which is the most similar to the traditional telephone service and is the critical regulation conflict in the short term. From the perspective of the user it is almost the same service that the traditional PSTN voice telephony provides, while from provider's perspective it is much less expensive and deregulated.

TECHNOLOGY STANDARDS

To the user, the fact that a particular call travels for part of its journey via the Internet or another IP network is irrelevant, as long as the price is low and the quality is acceptable. For IPTSPs, the main motivation is to reduce costs, particularly on the international leg of a call. Fax-to-Fax services work in substantially the same way as Phone-to-Phone voice.

Enhanced IP Telephony Applications

Beyond merely an alternative means of making long distance and international calls, IP Telephony technology enables a wide range of other services. As international accounting rates and PSTN calling rates come down, these enhanced or “value-added” applications may be the main source of revenues for IPTSPs.

The basic IP Telephony technology can be extended to create limitless possibilities for the transmission of voice alone, or in combination with any other digitisable information. Drawing regulatory distinctions between what is voice, what is telecommunications, what is computing, and what is Internet is therefore difficult.

Technical Aspects of IP Telephony

Circuit Switching vs. Packet Switching: Two different switching techniques can be used in a telephone system: circuit switching and packet switching.

The concept behind circuit switching is to establish a dedicated path between the two ends that will continue until the call is terminated. When a user or a computer places a telephone call, the switching equipment within the telephone system seeks out a physical path all the way from the user’s telephone to the receiver’s telephone. This technique is shown in Exhibit 1.

Each of the rectangles in the figure represents a carrier switching office. When a call passes through a switching office, a physical connection is (conceptually) established between the line on which the call came in and one of the output lines, as shown by the dotted lines. By contrast, on circuit-switched networks operating under a protocol such as Signalling System 7 (SS7) a call is routed through a hierarchy of local, inter-urban and international switches to establish a circuit between caller and called party.

The second switching mechanism is Packet Switching. Internet technology chops up electronic transmissions into packets of varying numbers of bytes. Each packet is given a ‘header’ or address label, and sent from one network node “towards” another. The packets are (theoretically) bounced along from one router to another, armed at each ‘hop’ with only enough information to get them safely to another router, where the process is repeated.

The key difference in the two methods is that telephone networks have been carefully engineered to provide extremely reliable, high-quality voice transmission, making real-time, two-way conversations possible from almost any two points on earth. IP networks, on the other hand, were originally designed for two-way, asynchronous (not real-time) *text-based* communication. A second difference lies in the way the services have traditionally been priced: on a per-minute usage basis for circuit-switched calls and on a flat-rate basis for IP traffic. The desire to make these two types of networks interconnect and interoperate without the user being able to tell the difference has prompted enormous research and development efforts in both the telecommunication and computer industries.

The differences between Circuit Switching and Packet Switching are[2]:

Feature	Circuit Switched	Packet Switched
Dedicated Path	Yes	No
Bandwidth Available	Fixed	Dynamic
Potentially wasted bandwidth	Yes	No
Store-and–forward transmission	No	Yes
Each packet follows the same route	Yes	No
Call Setup	Required	Not Needed
When can congestion occur	At setup time	On every packet
Charging	Per minute	Per Packet

As IP Telephony development has proceeded, the primary goal has been to replicate the functionality, reliability, and service quality of circuit-switched telecommunications networks, and to do so using a protocol suite which was not optimised to facilitate real-time communications. While Internet communication is indeed "connectionless", current development trends seek to make IP Telephony much more connection-oriented than other types of IP communications. In practice, in a digital network with use of data compression, the real differences between the two network types are becoming less and less significant.

[2] Source: Pg. 134, Computer Networks, Andrew S. Tanenbaum, Prentice-Hall India.

Gateways

The link between an IP network and a telephone network is called a **gateway**. This is the point at which voice signals become digitised and packetized, or at which digitised packets are converted back to voice. In the case of Phone-to-Phone IP Telephony (including calling card), the gateway is a server which customers dial, much as they might dial an ISP's server from their computer modem. The originating server prompts the user to input an account code and the desired telephone number, encodes the ensuing call into IP packets, each with a header directing it to another gateway server somewhere else, where the process will be reversed, and the IP call forwarded to an ordinary telephone. On the other end, the terminating server, which is located as close as possible to the called party's exchange, dials the called party's telephone number and, when a connection is made, starts sending the caller's speech and transmitting the called party's speech in the other direction.

Gateways allow long distance or international calls to 'look' to the billing systems of PSTN operators at either end like local calls. Not all originating gateways redirect PSTN calls onto the Internet, and not all terminating gateways receive calls from the Internet. Gateways can be connected to any kind of IP network, and, in the case of commercial IPTSPs like DeltaThree.com, the network in not the public Internet.

Many IPTSPs, however, *do* use the public Internet for some or all of their call routing, and this has implications for call quality. Once on the Internet, the packets are forwarded like any other Internet packets, whether they carry text, graphics or video. When they reach the terminating gateway, the packets are 'dumped' onto the local PSTN. From this point on, a true Internet Telephony call and an IP Telephony call would be indistinguishable from the 'point of view' of the local telephone network.

Gateway operators prefer to locate their equipment in major metropolitan centres, where the largest number of PSTN subscribers can be reached with or can make a local call. If a gateway server must dial a long distance call to terminate an IP call, the cost savings otherwise available can be lost. Terminating gateway operators must generally pay for local access lines to the PSTN, which are frequently the same lines leased by ISPs, so that their dial-up Internet access customers can dial into their Web servers.

IP Telephony users who are connected to ‘always-on’ LANs do not dial into gateways. Instead, their networks are constantly connected to one or more gateways. On enterprise internal IP Telephony networks, calls may never pass through a gateway at all, remaining ‘on-net’ throughout their entire path.

Quality of Service

An important issue in Internet Telephony is Quality of Service (QoS). The advantages of reduced cost and bandwidth savings of carrying voice over data networks are associated with some Quality of Service (QoS) issues unique to packet networks. Delivering quality voice signals from one point to another cannot be considered successful unless the quality of the delivered signal satisfies the recipient. Providing a level of quality that at least equals the PSTN (usually referred to as "toll quality voice") is viewed as a basic requirement. Although QoS usually refers to the fidelity of the transmitted voice and facsimile document it can also be applied to network availability, telephone feature availability, and scalability. Many factors have been identified that play a big role in determining the quality of service. They are as follows:

Latency

Latency is the time delay incurred in speech by the IP Telephony system. Latency is typically measured in milliseconds from the moment that the speaker utters a word until the listener actually hears the word. This is termed as "mouth-to-ear" latency or the "one-way" latency that the users would realize when using the system. The round-trip latency is the sum of the two one-way latency figures that make up a telephone call. In the traditional Public Switched Telephone Network, the round-trip latency for domestic calls is virtually always under 150 milliseconds. At these levels, the latency is not noticeable to most people. Many international calls (especially calls carried via satellite) will have round-trip latency figures that can exceed 1 second, which can be very annoying for users.

Effect of Latency - Two problems that result from high end-to-end delay in a voice network is echo and talk over lap. Echo is caused by signal reflections of the speaker’s voice from the far end telephone equipment back into the speaker’s ear. Echo becomes a problem when the round trip delay exceeds 50 milliseconds. Since echo is perceived as a significant quality problem, Voice over IP systems have to address the need for echo control and implement means for echo cancellation. Talkers overlap is the problem of one caller stepping on the other talker's speech. This becomes significant if the one-way delay becomes greater than 250 milliseconds. Delay can be subdivided into two sub-components. They can be fixed delay components as well as variable delay components. Fixed delay components include propagation, serialization, and processing. The variable delay components include the queuing delay, dejitter buffers as well as variable packet sizes.

The following are sources of delay in an IP Telephony call:

1) Accumulation delay, which is also called algorithmic delay at times, is caused by the need to collect a frame of voice samples to be processed by the voice coder. This depends on the type of voice coder used and varies from a single sample time, which is .125 microseconds, to many milliseconds.

2) The actual process of encoding and collecting the encoded samples into a packet for transmission over the IP network causes processing delay. The encoding delay is a function of both the processor execution time and the type of algorithm used.

3) Network delay, is caused by the physical medium and protocols used to transmit the voice and data, and by the buffers used to remove packet jitter on the receiver side. This delay is a function of the capacity of the links in the network as well as the processing that happens as the packets pass through the network. The jitter buffers add delay used to remove the packet delay variation that each packet experiences as it transits the packet network. This delay can be a significant part of the overall delay as packet delay variations can be as high as 70msec to 100msec in IP networks.

Jitter (Delay Variability)

Jitter is the variation in inter-packet arrival time as introduced by the variable transmission delay over the network. Removing jitter requires collecting packets and holding them long enough to allow the slowest packets to arrive in time to be played in the correct sequence, which in turn causes additional delay. The conflicting goals of minimizing delay and removing jitter has led to the development of various schemes to adapt the jitter buffer size to match the time varying requirements of network jitter removal.

Packet Loss

IP networks cannot provide a guarantee that packets will be delivered at all, much less in order. Packets will be dropped under peak loads and during periods of network congestion. But due to the time sensitivity of voice transmissions, however the normal TCP- based retransmission schemes are not suitable. Approaches that compensate for packet loss have to be developed to overcome this problem.

Bandwidth Availability

Bandwidth is the portion of the network that is available to an application to transfer information on the network. The level of reliability and sound quality that is acceptable among users has not yet been reached and this is primarily because of bandwidth limitations and this also leads to packet loss. In voice communications, packet loss shows up in the form of gaps or periods of silence in the conversation, thus leading to a "clipped speech" effect that is unsatisfactory for most users and unacceptable in business communications.

Proposed Solutions

Maintenance of acceptable voice quality levels despite inevitable variations in network performance is achieved using a variety of techniques. These techniques and solutions to problems that have been detailed above with regard to transmission of voice over IP are as follows:

?? One of the main problems of a very big end-to-end delay is the problem of echoes. This happens whenever the round-trip delay exceeds 50 milliseconds. Echo in a telephone network is caused by signal reflections generated by the hybrid circuit that converts between a 4-wire circuit (a separate transmit and receive pair) and a 2-wire circuit (a single transmit and receive pair). Echo in a telephone network is acceptable because the round-trip delays through the network are smaller than 50 milliseconds. Echoes are a problem in Voice over IP as the round-trip delays are almost always greater than 50 milliseconds. A way to improve speech quality is to implement some kind of echo cancellation mechanism. The ITU standard G.165 defines performance requirements for echo cancellers. The way the echo cancellers work is that when the echoes are generated from the telephone network toward the packet network, the echo canceller compares the voice data received from the IP network to the voice data that is being transmitted to the IP network. The echo from the telephone network is removed by a digital filter on the transmit path to the IP network.

?? As mentioned earlier the task of solving the problem of jitter in transmitting voice over IP has two conflicting goals. They are of minimizing delay as well as removing jitter. This has led to the development of various schemes to adapt the jitter buffer size to match the time varying requirements of network jitter removal. These schemes have the explicit goal of minimizing the size and delay of the jitter buffer while at the same time preventing buffer underflow caused by jitter. One approach that is used in IP networks to adapt the jitter buffer size is as follows. This approach involves counting the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ration is then in turn used to adjust the jitter buffer to target a predetermined allowable late packet ratio. This approach works best with networks with highly variable packet and inter-arrival intervals such as IP.

?? Lost packets are a big problem in networks. In current IP networks all voice frames are treated like data. Under peak loads and congestion, voice frames will be dropped equally with data frames. Data frames are not time sensitive like voice frames and there is no point in retransmission of lost frames as in voice transmission, if a packet is late, it is as good as not reaching the receiver at all. If the lost packets are left untreated, the listener hears annoying pops and clicks. Some schemes called lost packet compensation schemes used to overcome the problem of lost packets are as under:

1) Interpolate for lost speech packets by replaying the last packet received during the interval when the last packet was supposed to be played out. This works well when the incidence of lost frames is infrequent. It does not work very well for bursty loss of packets.

2) Another way is to send redundant information at the expense of bandwidth utilization. The basic approach replicates and sends the nth packet of voice information along with the (n+1) th packet. This method has the advantage of being able to exactly correct for the lost packet. However, this approach uses more bandwidth and creates greater delay.

3) An alternative approach is to develop an algorithm in the digital signal processor that detects missing packets, and then replays the last successfully received packet at a decreased volume in order to fill the gaps.

4) Another problem is that of Out of Order Packets. When an out of order condition is detected in the network, the missing packet is replaced by its predecessor, as if it was lost. When the late packet finally arrives, it is discarded.

Quality levels that are tolerated by the users are also achieved using such techniques such as compression, silence compression. The way silence compression works is that whenever it detects a gap in speech, it suppresses the transfer of things like pauses, breaths, and other periods of silence. This can amount to 50%-60% of the time of a call, resulting in considerable bandwidth conservation. Since the lack of packets is interpreted as complete silence at the output, another function is needed at the receiving end to add "comfort noise" to the output.

Network support for QoS

Voice processing involves handling greater and variable delays, jitter, and cancel echoes introduced from the telephony side. It will also involve masking the gaps caused by dropped packets using an appropriate algorithm. A protocol needs to be implemented which guarantees bandwidth for the duration of a session and also better compression technologies need to be put in place. Internet Protocol (IP), however, is a best effort service and therefore provides no guarantees on delivery and data integrity.

There are two ways in which sound quality can be improved – special quality of service algorithms and more bandwidth. Under current Internet routing conditions, IP voice packets do have a small advantage over email and Web packets, but not much. Internet Telephony packets are not transmitted using the Transmission Control Protocol (TCP), but via another called User Datagram Protocol (UDP).

UDP is used for Internet Telephony packets to minimize delay to some degree – at least to keep them from being slowed due to traffic conditions. However, UDP only compensates for the Internet's single service class to a limited extent – and not enough to facilitate PSTN-quality phone calls. Quality of service in IP networks and the public Internet will improve with innovations in routing protocols and improvements in the physical networks that carry IP traffic. One limitation that may take longer to ameliorate, however, is the poor quality of telecommunication infrastructure in developing countries. Even where IP infrastructure exists, congestion can be a major problem.

Many different techniques and protocols are used to improve the quality of service in networks. Some of these techniques and protocols that provide better network support in terms of reliability, availability, etc. are discussed below and in the next section.

?? Providing a controlled networking environment in which capacity can be preplanned and adequate performance can be assumed (at least for a majority of the time). This would generally be the case with a private IP network like an Intranet that is owned and operated by a single owner.

?? Using management tools to configure the network nodes, monitor performance, and manage capacity and flow on a dynamic basis. Most internetworking devices such as routers and switches include a variety of mechanisms that can be used in supporting voice. Traffic can be prioritized by protocol, location, and application type, thereby allowing real-time traffic to be given precedence over non-critical traffic. Queuing mechanisms may also be manipulated to minimize delays for real-time data flows. Recent research has shown that tag switching and flow switching can also improve overall performance and reduce delays.

?? Adding control protocols and mechanisms that help avoid or alleviate the problems inherent in IP networks. Protocols such as Real-time Transport Protocol (RTP), Real-time Transport Control Protocol (RTCP), and Resource Reservation Protocol (RSVP) are being implemented to provide greater assurances of controlled Quality of Service within the network. These protocols and the way they provide solutions to the problem of QoS in IP networks are discussed in the next section (Section 8). Other mechanisms such as admission controls and traffic shaping may also be used to avoid performance deterioration and overloading the network

?? Some other Networking tools to provide Quality of Service include

?? Congestion Management (Weighted Fair Queuing)

?? QoS signaling (IP Precedence and RSVP)

?? Packet Residency (MLPPP/MTU Size Reduction)

?? RTP header compression

?? Generic Traffic Shaping

?? Weighted Random Early Detection (WRED)

Standards and Protocols

H.323 Protocol

Originally proposed by Intel and PictureTel, H.323 defines a flexible means for multi-media teleconferencing equipment to communicate and provide application-sharing features over an IP protocol stack. The standard can be used on a variety of devices including videophones, desktop PCs, and large multi-port gateways. H.323 is part of a family of ITU-T recommendations called H.32x that provides multimedia communication services over a variety of networks that provide a non-guaranteed Quality of Service (QOS). This recommendation is based on the real-time protocol/real-time control protocol (RTP/RTCP) for managing audio and video signals. One of the primary goals in the development of the H.323 standard was the interoperability with other multimedia-services networks. This interoperability is achieved through the use of a gateway. A gateway performs any network or signalling translation required for interoperability. H.323 is considered an umbrella standard that encompasses a number of subordinate standards. Therefore, the International Telecommunications Union (ITU) can define the standard by re-using a number of other data and telecommunications standards such as Q.931, G.711, G.723.1, etc.

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. These components are:

?? Terminals

?? Gateways

?? Gatekeepers

?? Multipoint Control Units (MCU’s)

H.323 terminals:

Used for real-time communications, an H.323 terminal can be either a personal computer or a stand-alone device, running an H.323 and the multi-media applications. It supports audio communications and optionally supports video and data communications. The primary goal of H.323 is to interwork with other multimedia terminals. H.323 terminals are compatible with H.324 terminals on switched circuit networks and wireless networks, H.320 on ISDN and H.322 terminals on guaranteed QoS LAN’s. H.323 terminals may be used in multipoint conferences.

H.323 Gateways:

A gateway connects two dissimilar networks. A H.323 gateway provides connectivity between an H.323 network and a non-H.323 network. For example an H.323 gateway can provide connectivity between a circuit switched network, such as the PSTN and an H.323 terminal. The connectivity of these dissimilar networks however has to be achieved by using translation protocols for call set up and release, and transferring information between the networks connected by the gateway. A gateway is although not required for communicating between two terminals on an H.323 network.

The way the gateway works is that on the H.323 side a gateway runs H.245 control signalling for exchanging capabilities, H.225 call signalling for call set-up and release, and H.225 registration, admissions and status (RAS), for registration with the gatekeeper. On the SCN side the gateway runs SCN specific protocols such as ISDN and SS7 protocols.

Gatekeepers:

A gatekeeper can be considered to be the controller of an H.323 network. It provides call control services such as address translation and bandwidth management as defined within RAS. The H.323 standards both define mandatory services and optional services that the gatekeeper supports. The mandatory services of the gatekeeper include address translation, admission control, bandwidth control, and zone management.

The address translation function of the gatekeeper translates E.164 telephone numbers (e.g. 214-768-1234) or the alias into the network address (e.g. 47.41.56.123 for IP) for the destination terminal. Calls that originate within a H.323 network may use an alias to address the destination terminal, whereas calls that originate outside the H.323 network and received by a gateway may use the E.164 telephone number to address the destination terminal.

The gatekeeper also performs admission control of the end points into the H.323 network. It uses RAS messages, admission request (ARQ), confirm (ACF), and reject to achieve this.

The gatekeeper also provides support for bandwidth control by using the RAS messages, bandwidth request (BRQ), confirm (BCF), and reject (BRJ). The way this works is that if a network manager has specified a threshold for the number of simultaneous connections on the H.323 network, the gatekeeper can refuse additional connections once that threshold limit has been reached.

Optional gatekeeper functions include call-control signalling, call authorization and call management.

H.225 is part of the H.323 recommendation and it involves call control messages including signalling, registration and admission, and for the packetization and synchronization of media streams. The H.225 RAS is used between H.323 endpoints for the following reasons.

?? Gatekeeper discovery

?? Endpoint registration

?? End point location

?? Admission control

?? Access tokens

The disadvantage of RAS messaging is that these messages are carried on a RAS channel that is unreliable. Hence RAS message exchange may be associated with timeouts and retry counts.

The gatekeeper discovery process is used by the H.323 endpoints to determine the gatekeeper with whom the endpoint must register. The process of gatekeeper discovery may be done statically or dynamically. Endpoint registration is a process used by the endpoints to join a zone and inform the gatekeeper of the zones' transport and alias address. All endpoints automatically register with a gatekeeper as part of their configuration. Endpoint location is a process by which the transport address of an endpoint is determined and given its alias name or an E.164 address.

Multipoint Control Units (MCUs)

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 H.323 protocol is specified so that it interoperates with other networks. The most popular H.323 interworking is IP telephony, when the underlying network of H.323 is an IP network and the interoperating network is SCN. SCN includes PSTN and ISDN networks.

H.323 Strengths

?? The key strength of H.323 is its maturity, which has allowed a number of software vendors to develop robust implementations.

?? The standard's maturity also has allowed the various vendors to eliminate interoperability issues, permitting a widely compatible range of H.323 capable devices to be introduced into the market.

?? Because the H.323 standard includes an adaptation of the Q.931 protocol for call control, many developers with experience in existing ISDN telephony are familiar with the call control model. In fact, the events and parameters often can be directly passed from H.323 into applications that previously operated with ISDN.

H.323 Weaknesses

?? The original H.323 Version 1 recommendation suffered from slow call setup because many messages were interchanged between end devices before the voice path was established. The fast call setup features allowed in Version 2 have overcome this problem.

?? Because of the complexity of the standard, many products that require basic "quick and dirty" inter-gateway call control find H.323 too complex or expensive.

?? When defining H.323, the designers worked from the prospective of an end device, not a device that would reside within the existing PSTN. As a result, H.323 cannot integrate with SS7 or leverage the powerful capabilities that SS7 has to offer.

?? H.323's scalability also has proven to be an issue in very large applications. Designers using deployed gateways that included thousands of ports found the centralized state management to be a bottleneck.

?? The cost of implementation has been an issue when the end device needs to be very low cost. The complexity of the standard requires reasonable processing capability at the end device, which has prevented implementation on devices such as set-top cable boxes and hand-held wireless devices

Session Initiation Protocol (SIP)

The Session Initiated Protocol (SIP) provides a means to communicate call control information from end devices or proxy servers to each other or to gateway devices. It is an application layer signalling control protocol for creating, modifying, and terminating sessions with one or more participants. The sessions that SIP is concerned with include Internet telephone calls, multimedia conferences, and multimedia distribution. SIP can invite parties to join both unicast as well as multicast sessions, and it also has the added functionality that the initiator does not necessarily have to be a member of the session to which it is inviting. SIP is also capable of adding media and participants to existing sessions.

This protocol can be used to initiate sessions as well as invite members to sessions that have previously been advertised and established by other means. SIP facilities also enable personal mobility. It therefore gives the end users the ability to originate and receive calls and access subscribed services on any terminal in any location, and the network should also be able to identify the users as they move. SIP supports five aspects of establishing and terminating multimedia calls and communications. These are:

1) Locating the user that involves determining the end system to be used for communication.

2) Identifying the capabilities of the user such as the media and the media parameters that will be used in the communication.

3) Determining user availability and the willingness of the called party to engage in communications.

4) Setting up the call and establishing the parameters at both the calling and called parties.

5) Call handling which involves the transfer and the termination of calls.

The Session Initiation Protocol has many similarities to the H.323 signalling protocol. SIP also has the capability of handling and initiating multi-party calls using Multipoint Control Units, as does H.323 to handle multi-party calls. Internet telephony gateways that connect to the Public Switched Telephone Network (PSTN) can also use SIP to set up call between them.

The operation of SIP is as follows: Callers and callees are identified by SIP addresses. When making a call over SIP, a caller first locates the appropriate server and then sends a SIP request. The most common operation performed by SIP is the invitation to users to join a session. Instead of directly reaching a callee, a SIP request may be redirected or may trigger a chain of new SIP requests by proxies. Users register their location or locations with the SIP servers. The SIP servers facilitate the connection between the caller and the callees.

SIP Strengths

?? The expandable nature of the protocol allows future capabilities to be easily defined and quickly implemented.

?? It is simple and easy to embed into inexpensive end-user devices.

?? The protocol was designed to ensure interoperability and enable different devices to communicate.

?? Non-telephony developers find the protocol easier to understand.

SIP Weaknesses

- SIP is very new, so most application are in the prototype stage.
- The protocol has a narrow scope and thus has limited applications by itself; however, it gains flexibility when used with other protocols.
- SIP is only one small piece of a complete solution. Numerous other software components are required to build a complete IP/Telephony product

Media Gateway Control Protocol (MGCP)

Media Gateway Control Protocol (MGCP) is a protocol that provides the means to interconnect a large number of IP telephony gateways, allowing them to work together as one. MGCP assumes that a Call Agent (CA) performs the intelligence of all call control operations and that Media Gateway Controller (MGC) carries out all media processing and conversion. This is a master-slave protocol, where the gateways are expected to execute commands sent by the Call Agents. MGCP assumes a connection model where the basic constructs are end points and connections. Endpoints are sources or sinks of data, and they may be virtual or physical.

MGCP Strengths

- Because MGCP was defined to solve a specific problem with very large deployed systems, it is particularly suited to large deployed application.
- Use of MGCP allows for good integration into the SS7 network, which gives greater control and throughput in handling calls.
- MGCP splits the media handling and signalling functions, thus providing a simpler implementation, which can be developed by multiple vendors.

MGCP Weaknesses

- MGCP is too complex for smaller applications.

?? Carriers that require media gateway control may elect to use either MGCP or H.248. Therefore, H.248 implementations may ultimately replace earlier MGCP versions.

Interworking of H.323 and MGCP:

MGCP is designed as an internal protocol within a distributed system that appears to the outside as a single VoIP gateway. This system is composed of a call agent, that may or may not be distributed over several platforms, and a set of gateways, including at least one media gateway, whose purpose is to perform the conversion of media signals between signals and packets, and at least one signalling gateway when connecting to an SS7 controlled network. In a typical configuration, this distributed gateway system will interface on one side with on or more circuit switches on the PSTN side, and on the other side with H.323 conformant systems. Since in an MGCP model the gateways focus on audio signal translation function, the call agent then has to handle the call processing and signalling. Therefore in this model the call agent has to implement the signalling layers of the H.323 standard and presents itself as a H.323 gatekeeper or an H.323 end point to the H.323 systems. The distributed gateway systems and MGCP will enable PSTN telephony users to access sessions set up using SAP, SIP, or RTSP.

Comparison of H.323 and Session Initiation Protocol

The reason for a comparison of H.323 and Session Initiation Protocol is because both are standards for signalling and control for Internet Telephony. H.323 on the one hand embraces the more traditional circuit switched approach to signalling whereas SIP on the other hand favours the more lightweight Internet approach. In both cases the multimedia data that needs to be exchanged will be done via RTP, which means that the protocol suite does not influence the Internet telephony QoS.

H.323 is a very complex protocol compared to SIP. This is because H.323 defines hundreds of elements while SIP has only 37 headers, each with a small number of values and parameters but these contain more information. H.323 on the other hand uses several protocol components which make it all the more complex.

Complexity is just one metric for comparison but another metric which plays a big role in comparing the two protocols is extensibility. SIP is very easily extendible, as it has taken information from both HTTP and SMTP. Though H.323 also provides a number of extensibility mechanisms, these have a number of limitations such as including non-standard parameters and there is no mechanism for allowing terminals to exchange information about which extensions each supports. Another area where SIP is more extendable than H.323 is in codecs. SIP can work with any codec whereas in H.323 each codec must be centrally registered and standardized.

ECONOMIC ASPECTS OF IP TELEPHONY

Liberalization of telecom sector in developing countries is going to bring down the tariff of the long distance calls. Consumers in the developing countries will be benefited immensely because of long distance calls become cheaper. With the strong middle class base in India, it is expected that this will fuel a demand, which may far exceed the fall in revenue. This phenomenon has been observed already in the cellular market, where reduction in tariff has resulted in an explosive demand growth. IP Telephony can be used by operators to take advantage of the above fact, because implementation of this technology is easy and at a lower cost. Economic aspects of IP telephony can be considered from the perspective of consumers, carriers and countries.

Consumers

The economic advantages of IP Telephony for *consumers* can be described very simply: it is invariably *cheaper* than the traditional alternative, i.e., the POTS. For the moment, the other factors such as quality, reliability and convenience are not equal. Therefore consumers are forced to make a trade-off between price and quality. In the long run, it is likely that IP telephony will be equivalent in quality and reliability, and in some circumstances might be more convenient (e.g., for unified messaging) than traditional telephony. Thus for consumers price will be the main distinguishing factor.

Carriers

The impact of IP telephony on the operations of sector members can be considered on the basis of following aspects:

?? It is likely to increase the competition by reducing barriers to entry in the long distance voice market, as well as decreasing the bandwidth cost. The competition is expected to drive all the operators to resort to cost based pricing, thus eliminating the cross-subsidization that has been prevalent in most of the developing countries.

?? The incumbent operators will face loss of income from international calling, both direct (loss of collection charges) and indirect (loss of settlement payments). But arguably this is expected to happen even without IP Telephony. Markets for international calling are shrinking in value as, the prices are falling precipitously, and the traffic is also getting routed through lower cost routes. Operators in developing countries may be better advised to embrace IP Telephony, and bear the consequences of reduced per-minute revenues from long-distance and international services, than to risk missing the opportunity to develop revenues in future growth areas.

?? Incumbent operators will face a financial challenge to see that the service offered remains competitive. The new service providers are likely to deploy low cost infrastructure of IP telephony for providing services at low costs. The incumbent operator can also deploy IP telephony to effectively use the existing infrastructure in a more optimum and efficient way to carry more traffic and generate more revenue. A convergent media, which ensures a single media for all the multi media services including voice also enables to reduce the infrastructure cost including the staff, to a great extent.

?? The operators will have to focus on offering value added services to the customers. More penetration of Internet along with these new value added services will in turn create more demand and lead to increase in the telephone penetration. The introduction of IP telephony will create many new opportunities.

?? IP telephony will lead to calls getting routed through cheapest routes. Future operators will need to ensure that their local access networks are self-financing. This will require substantial and urgent tariff rebalancing to bring the price of local and international calls much closer. As IP telephony makes access to the Internet even more popular, it will actually increase the volume of local calls and the demand for second lines. Already, in some economies, as much as a third of all local calls are to the Internet and around 15 per cent of all local lines are used primarily for Internet access[3]. Furthermore, dial-up Internet access is on a steeply rising curve while international traffic growth is slowing down (Refer Exhibit 2).

The incumbent carriers generally have existing revenue streams that they fear might get cannibalised by a shift to IP telephony. Even if IP telephony offers a cheaper alternative for those existing revenues, it might not be economically rational to move immediately towards providing telephony services over an IP platform. The speed of this transition will be dictated by the following factors:

?? The regulatory environment.

?? The degree of competition the carrier experiences in its domestic and in foreign markets. The greater the level of competition, the faster the shift will be towards lower cost services.

?? Whether a particular carrier is an incumbent or a new market entrant. New market entrants, with no legacy network to defend, are likely to be the first movers towards IP telephony.

?? The anticipated level of price elasticity in the demand for voice-based services. Where price elasticity is high, then the shift towards IP telephony will be quicker.

Countries

For *countries,* the question of whether to permit or ban different forms of IP Telephony is a sensitive one. For countries in which the interests of regulators are aligned with those of consumers, it would appear to be Luddite to ban IP Telephony. Generally a liberal approach offers the best prospects for consumer welfare. A less liberal approach might be expected in countries where the interests of regulators are more closely aligned with those of the incumbent carrier (e.g., where the carrier is state-owned). Some carriers might restrict the offer of IP Telephony to a limited range of licensed carriers, reinforcing existing restrictions on market entry for voice communications.

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[3] Anonymous, "Background Issues Paper," IP Telephony Workshop, webpage in www.itu.int, 29 May 2000.

A more nuanced approach might be to permit (or even encourage) use of the Internet to carry outgoing international calls (thereby bypassing the accounting rate system) while insisting that carriers making incoming international calls pay the full inward settlement rate. Asymmetric policies of this nature are being applied in some developing countries with a view to *maximising incoming settlement payments*. (refer Exhibit 3).

Size, Substitutability and Settlements

From an economic viewpoint, the significant questions to ask about Internet Telephony are:

- The size of the market?
- To what extent is IP Telephony generating *new* traffic or is it substituting for that which already exists?
- What is the impact of IP telephony on the business models of existing carriers?

Size

Looking first at the issue of size, market estimates vary widely. The market research company, IDC, estimates that the IP Telephony market generated traffic worth 2.7 billion minutes in 1999 and will expand to around 135 billion minutes, with revenues of US$19 billion, by 2004[4]. Deltathree.com forecasts that IP Telephony will generate around 16 billion minutes of international traffic in 2000 and will account for some 35 per cent of the total by 2005.

One reason that the market estimates differ so much is because the studies use different definitions. Market forecasts, such as those put out by IDC, are based mainly on traffic reported by IP Telephony service providers (IP TSPs). They do not generally include traffic that is being carried over IP (for at least some of the route) by the major public telecommunication operators. This is particularly difficult to estimate. Already, the number of international circuits which are used for leased lines outnumber those that are used for the PSTN, especially on the busiest routes, for instance between the United States and Europe (see Exhibit 4). These figures suggest that, within a few years, a significant share of international telephony traffic will be carried over IP for at least part of its route.

[4] Anonymous, "Background Issues Paper," IP Telephony Workshop, webpage in www.itu.int, 29 May 2000.

Substitutability

The issue of substitutability is more complex. Here we need to look into the mode of communications:

PC-to-PC Internet Telephony - Clearly, much of the traffic carried over *PC-to-PC* Internet Telephony will be "new" traffic that would not otherwise have existed.

PC-to-Phone services - Much of the discount traffic generated over *PC-to-Phone services*, especially that which is "free", is also likely to be new traffic.

Settlement

The motivation for sending traffic outside the accounting rate system is to reduce the level of settlements that are due to partner countries. Under the international settlements system, the operator(s) in the country that originates a call have traditionally made a compensatory payment to the operator(s) in the country that terminate the call. Payments are made when traffic in one direction is greater than traffic in the return direction. The level of payment is based on bilaterally negotiated "accounting rates".

A *net settlement payment* is usually made on the basis of excess traffic minutes, multiplied by half the accounting rate (the accounting rate share, or settlement rate). Net settlement payments, primarily from developed countries, have grown larger as traffic flows have become less balanced. ITU estimates that, between 1993-98, net flows of settlement payments from developed counties to developing ones amounted to some US$40 billion. The top ten net settlement surplus countries (i.e., which receive more money than they spend) are illustrated in Exhibit 3.

Operators that send more traffic than they receive have an incentive to develop alternative routing procedures. They do this to avoid having to make settlements based on above-cost accounting rates and instead pay interconnect fees at local call rates or below. This is one reason for using international IP backbones instead of PSTN circuits to deliver traffic. Analysis of individual country's traffic data appears to confirm that this is happening to an increasing extent. The settlement rates between the United States and Argentina and Colombia on 1 March 2000 stood at 27 and 32.5 US cents per minute respectively[5]. Increasing volumes of traffic from US carriers have been routed outside the accounting rate mechanism, for instance via the Internet (see Exhibit 5) or via refile

[5] Source: ITU "Americas Telecommunication Indicators," ITU/TeleGeography Inc. "Direction of Traffic" Database.

through other countries. In the case of Argentina, estimated bypass traffic[6] amounts to almost the same as the total reported volume of traffic on the route to the United States in 1998 (i.e., just over 200 million minutes). In the case of Colombia, where call-turnaround was historically less significant, estimated bypass traffic amounts to around 160 million minutes (Exhibit 5). The losses incurred from bypass traffic by Argentina and Colombia were over US$ 60 million for each country, at 1998 settlement rates.

[6] *"Estimated call-turnaround"* traffic is the volume of traffic on a particular route that has been re-routed so that it appears that it is coming from the United States. This includes call-back, calling card and home-country direct traffic. It is estimated by applying the ratio between incoming and outgoing traffic that applied before 1992 to the subsequent traffic balance. *"Estimated bypass traffic"* is the volume of traffic on a particular route which is estimated to be rerouted via a least cost route (e.g., refile) or outside the accounting rate mechanism (e.g., via the Internet) such that it is not reported in official traffic statistics. It is estimated by comparing the projected growth in the total volume of traffic on the route, based on trends before 1996, with what actually happened after that date.

Currently, connecting to a long distance carrier’s makes long distance phone calls point-of-presence (POP) through a local central office. The long distance carrier's POP serves as a local interconnection point to its long distance network, which is made up of 5 levels of hierarchical switching offices, in a tree-like structure[7]. A simple diagram of the Public Switched Telephone Network (PSTN) is shown in Exhibit 6.

Long distance phone calls using IT are made by connecting through the central office to the Internet Access Provider (IAP) or Gateway (depending on whether the call is being made through a computer or a standard phone). This IAP/Gateway can be considered an Internet Point-of-Presence (IPOP), analogous to the long distance carrier's POP. Similarly, the IPOP serves as a connection point to the Internet, through Internet Service Providers (ISPs) that serve multiple IAPs. The ISPs themselves are connected through Network Access Points (NAPs). Exhibit 7 shows a simple diagram of the Internet telephony architecture.

Cost Model for Internet Telephony

We will lay a general framework for a cost model of Internet telephony, which can be contrasted with the costs of providing a telephone call on the Public Switched Telephone Network (PSTN). The intention is not to provide an exhaustive cost comparison between PSTN and Internet telephony, but simply to draw distinctions between the various cost components to contrast the cost of delivering a call through the Internet and the PSTN.

All long distance calls include costs for the local loop, a point of presence (POP), and an upstream or "long distance" portion. To better understand the costs of delivering Internet telephony, it is useful to contrast the differences between calls placed through the PSTN and the Internet. Exhibit 8 shows the parallel cost components of Internet telephony and PSTN.

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[7] “Comments of Students of the Telecommunications Modeling and Policy Analysis Seminar Massachusetts Institute of Technology” web page in http://itel.mit.edu:/itel/Docs/ACTA/TPP91.HTM

While the models of the two networks appear quite similar, there are differences in the costs of each of the components, in particular:

1. **Local Loop Costs (C_{local}):** Local loop costs are similar because they use essentially the same lines. The only difference between these costs is in the method of interconnection between the local loop and the POP. For the PSTN, the long distance carrier's POP is collocated in the local telephone company's facility. This allows users to make long distance calls with one number dialing. In contrast, users of Internet telephony must first call the local IPOP, which will then route the call to the destination. For the purpose of this model, local loop costs for Internet telephony include the cost of the incoming lines (through which they are connected to their users) and the costs of the leased lines that connect the IPOP to its upstream Internet Service Provider.
2. **POP Costs (C_{IPOP}):** The POP in the PSTN model consists of the LD carrier's interconnection equipment that is collocated in a LEC facility. For Internet telephony, the IPOP is an Internet Access Provider. For the model, this includes all operating costs of an IAP except the cost of leased lines and interconnection with an upstream Internet Service Provider.
3. **Upstream/LD Costs ($C_{upstream}$):** The long distance component of the PSTN consists of circuit switched lines. This includes the cost of the physical lines and the cost of the digital circuit switches. The upstream component for the Internet telephony model consists of Internet Service Providers (ISPs), which are connected by Internet routers and ATM switches. .

In summary, the costs considered for Internet telephony can be loosely described by the following formula:

$$C_{total} = C_{local} + C_{IPOP} + C_{upstream}$$

A Cost Model Of Internet Service Providers: Implications For Internet Telephony

The MIT Internet Telephony Consortium (ITC) and the MIT Research Program on Communications Policy (RPCP) have developed a cost model of Internet service providers (ISPs). The model quantifies the impact on an ISP's costs due to an increased use in Internet telephony.

Two scenarios are modeled: a baseline scenario representing current ISPs in which the principal use of the network is for web browsing and there is essentially no Internet telephony; and an Internet telephony (IT) scenario in which the ISP sees a substantial increase in use of computer-to-computer Internet telephony by its subscribers.

The model is used to identify the costs of end-to-end Internet service for various types of users (dial-in, leased-line, etc.). These costs are broken down into five categories:

?? *Capital Equipment* - the hardware and software of the network

?? *Transport* - the leased-lines of the network and interconnection costs

?? *Customer Service* - staff and facilities for supporting the customers

?? *Operations* - billing, equipment and facilities maintenance, and operations personnel

?? *Other Expenses* - sales, marketing, general and administrative

Principal Cost Categories

The ISP's costs are separated into five principal categories: capital equipment, transport, customer service, operations and other expenses (which include sales/marketing and general/administrative). Each cost element (e.g., router, billing or marketing costs) is determined based on assumptions about how large the costs would be for the given number of subscribers. Once the total cost of an element is known, its cost is allocated to each type of subscriber based on the relative amount of use by each type of subscriber. Carrying out similar calculations for each cost component permits the model to determine the cost per subscriber for each type of subscriber.

Capital Equipment

Capital equipment includes that which is found in the Tier 1 and Tier 2 POPs. Capital investments are converted from a one-time, fixed cost to a leveled, annual cost by using a cost of capital rate[8].

Each piece of capital equipment is sized based on assumptions of users' access patterns and bandwidth requirements. Once the total cost of a piece of equipment is known, its cost is allocated to the various types of subscribers based on the relative amount of use by each type of subscriber.

Transport

The transport costs of the ISP are comprised of costs due to leased-lines to connect the Tier 1 and Tier 2 POPs (T1s, T3s and OC-3s) and costs due to incoming analog and ISDN phone lines (T1 and PRI) to connect the dial-in subscribers. In addition, monthly costs for the ISP to interconnect at a network access point (NAP) are included in transport costs.

As with the capital equipment costs, transport costs are allocated to the various types of subscribers.

Customer Service

Customer service is furnished by representatives who provide technical support via the telephone to the subscribers. For the model, the perspective taken is that customer service is outsourced by the ISP. Hence, instead of determining how large a staff is needed, one determines how many call-minutes there are and what is the cost per minute charged to the ISP.

Operations

Operations correspond to the routine tasks necessary to keep the ISP functioning. Operations costs fall into three principal sections: network operations and maintenance, facilities, and billing.

Network operations and maintenance costs include those for maintaining the hardware and software of the network, as well as the personnel needed to carry out these responsibilities. The costs for the maintenance are based on a percentage of the total costs for the capital equipment, and the personnel costs are based on the number of people

[8] "A cost model of internet service providers: Implications for internet telephony and yield management" by Brett A. Leida, Web page in http://rpcp.mit.edu/Pubs/Theses/leida.pdf

needed to maintain the given number of POPs. Facilities costs are those associated with maintaining a physical space for each POP. Included are such costs as building rent, electricity, heat, etc. The costs are based on an expenditure in $/month for each type of POP. Network operations and maintenance costs include those for maintaining the hardware and software of the network, as well as the personnel needed to carry out these responsibilities. The costs for the maintenance are based on a percentage of the total costs for the capital equipment, and the personnel costs are based on the number of people needed to maintain the given number of POPs.

Facilities costs are those associated with maintaining a physical space for each POP. Included are such costs as building rent, electricity, heat, etc. The costs are based on an expenditure in $/month for each type of POP.

Other Expenses

The remaining costs seen by an ISP are included in an Other Expenses category. These costs include sales/marketing and general/administrative.

Sales and marketing costs are those used to attract and retain subscribers. These costs are based on a percentage of revenue.

General and administrative (G&A) expenses consist primarily of salaries and occupancy costs for administrative, executive, legal, accounting and finance personnel. Similar to sales and marketing costs, G&A costs are based on a percentage of total costs.

Effect of Internet telephony on the ISPs

Comparison of baseline scenario with Internet Telephony scenario for the US case

Using the above theory Leida had modeled a U.S.-based Internet Service Provider that controls a nationwide Internet backbone with 9 Points of Presence in the 9 largest U.S. cities and 450 Points of Presence through the entire country[9]. Comparing the Internet telephony scenario to the baseline scenario for the US case, it was found that the costs in all categories increase in the Internet telephony scenario; however, some categories are affected more than others are. The bottom line for an ISP is that revenues will increase

[9] "Internet Telephony: Costs, Pricing, and Policy" by Dr. Lee W. McKnight, and Bett Leida, web page in http://itel.mit.edu:/itel/pubs/itel.tprc97.pdf

slightly, while costs will increase substantially with only a moderate use of Internet telephony. Hence, ISPs need to consider how to minimize the cost impact of Internet telephony and/or how to recover additional revenue if they hope to operate at profitable levels. The comparative results for the baseline and Internet telephony scenarios for the US case are shown in Exhibit 9.

Various cost component were found to be affected:

1) At 28% of total costs, transport costs become the largest cost category in the Internet telephony scenario. The implication for ISPs based on this result is that an ISP that operates its network most efficiently will have a competitive advantage over other ISPs if the Internet telephony scenario takes place. Such efficiencies could come from scale economies, facilities-based networks or network optimization techniques. However, if one believes that the market for transport is already efficient and that transport is essentially a commodity, then there would be fewer opportunities for competitive advantage resulting from owning a network. Even so, network optimization techniques would prove advantageous whether or not the ISP owns or leases its network.
2) Subscriber costs are impacted in different ways. . Exhibit 10 shows the percentage increase in each cost category for each subscriber type. For example, transport costs increased by 75% for the analog dial-in subscribers. In general, transport costs are substantially impacted for each subscriber type. Costs in the other expenses category increase for the leased line subscribers (56kb and T1) due to an increase in sales and marketing costs. This is based on the assumption that leased-line subscribers would purchase enough capacity to maintain their circuit at the same level for both scenarios. Hence, additional revenue is received from the leased-line subscribers in the Internet telephony scenario. Because sales and marketing costs are based on a percentage of revenue, these costs also increase.

Drawing conclusions based on comparing the change in costs between the different types of subscribers is not valid because the revenue is also changing, but only for the leased-line subscribers. One method of comparing the impact on the different types of subscribers is to consider a cost/revenue ratio for both scenarios, which is presented in Exhibit 11. Here one can see that the dial-in subscribers become particularly unprofitable, but the leased-line subscribers remain at about the same level of profitability as in the baseline scenario.

Thus the Internet telephony scenario analysis in US yield the following conclusions:

1) In the IT scenario, the increase in the ISP's costs is double the revenue increase. Hence, ISPs will lose even more money if they do not attempt to recover additional costs.
2) Transport costs become the largest cost category in the IT scenario.
3) Non-technical costs still remain a large portion of per-user costs, especially for dial-in subscribers.
4) Dial-in subscribers' costs are sensitive to access patterns and customer service costs. T1 subscribers' costs are sensitive to bandwidth usage patterns.

Internet Telephony Pricing

As described in the previous section, ISPs face potential increased cost pressure due to Internet telephony. For ISPs to remain in business, they will need to recover these increased costs. The Resource ReSerVation Protocol (RSVP) developed by the Internet Engineering Task Force (IETF) could be used as a mechanism for implementation of usage-sensitive pricing to recover those costs.[10] However, whether and where the current specification of RSVP is useful has not yet been determined. It is clear that RSVP by itself is not capable of resolving the host of network architecture and Quality of Service constraints on Internet pricing models. For example, how RSVP traffic could cross multiple networks has not been resolved.

When developing pricing schemes, service providers will have to look beyond the Internet telephony service and consider how to price differentiated and/or integrated Internet services generally. We argue elsewhere that an integrated regulatory framework will be required to permit the provision of such integrated services (Neuman, McKnight, and Solomon, 1997).

Alternatives for pricing include flat rate, or the introduction of usage-sensitive pricing. In McKnight and Bailey (1997), a variety of proposed approaches to Internet pricing,

[10] IETF rfc2205 (http://reference.nrcs.usda.gov/ietf/rfc2300/rfc2205.txt)

including approaches at the infrastructure level for network interconnection, are presented (see Bailey and McKnight, 1997). In the next few years, we anticipate experimentation with a variety of pricing models that permit service guarantees for multiple qualities of service, including guarantees for both real-time multimedia and multicast conferencing.

Employment of yield management techniques, which may enable use of innovative service definitions in the face of highly variable demand to maximize revenue, should also be considered[11]. Yield management, which originated in the airline industry and is discussed, further in (Leida, 1998), uses a combination of service definition, pricing and admission control. The fundamental principle of yield management is that different classes of service, be it Internet access or Internet telephony, are defined and only the high priority classes are served during peak periods of demand. During low periods of demand, discount classes are intended to attract an increased level of demand. The consequence of such techniques is that the system's capacity is more full on average and revenues are higher.

Additionally, one must consider the state of technology when considering cost-recovery alternatives. Usage-sensitive pricing will not be an option until protocols that monitor the use of Internet telephony are deployed widely.

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[11] 27 For example, see (Paschalidis, Kavassalis, and Tsitsiklis, 1997). Ideally, a company using yield management wants to maximize its profit, not just its revenue. However, in most cases where yield management is currently used, the marginal cost of providing service is very small vis-a-vis fixed costs. Hence, maximizing revenue is essentially the same as maximizing profit.

REGULATORY ISSUES

Very significant portions of the economic advantages that IP telephony has over traditional PSTN services are based on regulatory discrepancies. The matters are further complicated by the fact that regulators around the globe have reacted differently to new technology making it difficult to predict the outcome of the future regulatory system. Initially, many regulators lauded the introduction of a wide range of services via the Internet as innovative and pro-competitive developments. IP telephony services were granted a fair amount of leeway by regulators, many of who were reluctant to impose the traditional burdens of telecommunications regulation on these new services. This "hands off" approach is clear from the rulings of the European Commission and the US FCC to date.

Regulatory issues, like local access fees or any kinds of tax imposed to the long distance carrier, cause discrepancies that affect national and international long distance service under IP telephony. However, international services face other regulatory difficulties like accounting rates, which are the international access charges that the long distance operators pay to the company receiving the call in the callee's country.

Further, one issue that scholars have not discussed is the different licensing regime between IP and long distances operators. Governments worldwide have charged important fees to the long distances carriers in order to issue the respective licenses, often granting monopoly or oligopoly market conditions for certain period of time. On the other hand, IP telephony has been using value-added licenses that have been more liberalized and less expensive, despite delivering almost the same service to the end user.

Basic Telecommunications - Value Added Services – Enhanced Services

In order to understand the huge regulatory differences present between IP telephony and the traditional voice service under PSTN, it is critical to identify the various criteria used to divide telecommunications services.

In the United States the FCC has classified telecommunications services as either basic or enhanced. The general understanding of basic services is "the transmission without manipulation or change en route from sender to receiver." Whereas, enhanced services are defined as "services, offered over common carrier transmission facilities used in interstate communications which

1) Employ computer processing applications that act on the format, content, code, protocol or similar aspects of the subscriber’s transmitted information;
2) Provide the subscriber additional, different, or restructured information; or
3) Involve subscriber interaction with stored information."

At the World Trade Organization (WTO) under the General Agreement on Trade in Services (GATS) the countries divided "basic telecommunications services" and "value added" services, or "telecommunications for which suppliers add value to the customer’s information by enhancing its form or content or by providing for its storage and retrieval"
Traditionally, "basic" services have been hyper-regulated while the "value added" and "enhanced" services have enjoyed a much more liberalized regulatory environment. The common problem is that while traditional long distance is considered one of the main examples of basic services, IP telephony providers have been using the "value added" or "enhanced service" status in order to provide the service. In this way, IP telephony can shield itself from most of the regulation designed specifically for the basic service of international long distance.

However, some regulators have started to redefine the concept of "basic" services in order to embody IP telephony. The European Commission stated "…due to the development of specific software it has become possible to code, compress and transmit voice communications in such a way that it has become viable to send them via the Internet to other Internet subscribers using the same or interoperable software and via gateways to standard telephones. This is a new issue and the Commission should therefore adopt a Supplement to the Communication on these services, often described as 'Internet telephony.'"

The European Commission has taken the most clear approach to defining what IP telephony is by interpreting the application of the original definition of "voice telephony" to embrace Internet telephony when it meets the following requirements. It has to be subject to a commercial offer, for the public, to and from public switched network termination points and it has to involve direct transport and switching of speech in real time. Consequently, the Commission stated, "to the extent that Internet Voice is considered not to be a 'voice telephony service' within the meaning of the Directive, a requirement for an individual license may therefore not be imposed on Internet access/service providers". Further, indicated that universal service contributions will not be applied to the ISP unless their service fits the conditions indicated above.

Along these lines the Commission ruled that "when all the criteria of the 'Voice Telephony' definition are satisfied, those Internet service providers offering a dial out service to any telephone number - and only those - could then be considered providers of voice telephony services under Community law." However, the European Commission indicated that its position regarding IP telephony could change depending on further developments of the technology. In other words, the regulatory framework does not seem to be a solid one, rather it is flexible in order to adapt to the new technologies.

Notwithstanding these particular efforts to delimit the spectrum of traditional voice telephony regulation over IP telephony, numerous countries will face legislative and political obstacles to accomplishing this task. Further, some time will be needed in order to observe if these efforts are efficient with regard to a daily changing technology. Along these lines, the FCC "decided to refrain from formally classifying IP telephony as a regulated basic service, leaving the question open for future regulatory proceedings."

Local Access Fees

Local access charges are the interconnection fees that long distance companies pay to Local Exchange Companies (LECs) in order to complete a call. Generally, these fees have been over priced to subsidize the development of the countries' telecommunications infrastructure, or just to make the local communications more affordable to the consumer.

The basic claim of the Long Distance Companies is that Internet Services Providers should be charged this kind of fee, especially now that IP telephony is providing services very similar to the traditional voice telephony. The analysis of this issue was well discussed internally in the United States. Initially the FCC had classified enhanced services providers as "end users" while the traditional long distance companies were indicated as "carriers". For this reason, ISPs did not pay the access charges that carriers should pay to the LEC. In the access reform order adopted in 1997, the FCC determined that the ISPs would continue to be exempt from these access charges.

However, the FCC let the door open to the States impose such fees, and the final status of the ISP is as a hybrid between a long distance carrier and a local final user. Despite the fact that this decision was well received by Internet supporters, a lack of political determination and technical clarity was noted. This illustrates some of the ambiguity and political difficulties that regulators face when regulating IP telephony.

The fact is that IP telephony is still exempt from this kind of access charges within the US but the regulatory support seems to be poor and confusing. However, a local access fee is not the major issue in other countries. "In the rest of the world, notably in Western Europe, local calls are priced and have tended to rise in recent years as PTOs have rebalanced their tariffs to reduce the distance element and to eliminate perceived cross-subsidies." Furthermore, countries where the governments have guaranteed market systems, like what happened in Latin America, the focus is on other regulatory conflicts, principally accounting rates and licenses.

Accounting Rates

The accounting rates as indicated above are the payments that Long Distance Carriers have to pay to the Carriers that receive the call in the respective country, in order to complete the call. In other words, accounting rates are the international interconnection fees. This has been a very sensitive issue the last decade, especially to developing countries operators. In order to understand IP telephony regulatory repercussions in this area, it is important to take an overview of the accounting rates system and politics.

Historically the international traffic has been bigger from industrialized countries to developing countries, rather than the traffic from these developing countries to abroad. Further, the United States has an upstanding surplus in its traffic balance to other countries. Due to the accounting rates system this surplus means a payment deficit of approximately $5 billion dollars per year.

For that reason, the accounting rates were initially overpriced as an aid to the developing countries in order to build their communications infrastructure.

However, once the telecommunication industry entered in the liberalization era, the US carriers initiated a strong process towards the reduction of these rates. The FCC has been the representative of the US interest, negotiating bilaterally and multilaterally the decrease of the accounting rates to their cost levels. The International Telecommunications Union (ITU) and the Inter American Telecommunications Commission (CITEL) are the two principle forums where the multilateral negotiations have taken place, without strong practical repercussions. On the other hand bilateral negotiations seem to have been more efficient towards rate reductions. Despite this efficiency, the goal of reducing these rates to cost is not close to being attained yet, especially in Latin America.

Here is where IP telephony comes into the picture. The FCC gladly received the introduction of IP telephony to avoid the international accounting rates. Further, the FCC indicated that the arrival of this technology would pressure the countries into accepting lower settlements rates. "Internet telephony has the potential to be a significant alternative to the accounting rate system. Calls made over the Internet are not subject to the accounting rate system, and as a result, we anticipate that charges for Internet telephony will be substantially closer to the actual cost of providing service, and much lower than most collection rates for international service."

The response from foreign countries to the FCC ruling was mixed, with an orientation against IP telephony as technology, and to the US policy it self. Countries like the Czech Republic, Iceland, Hungary, and Singapore "have indicated that they intended to prohibit or regulate Internet Telephony" Further, other countries have insinuated they would file suit against the FCC decision.

As these actions indicate, developing countries care a great deal about accounting rates. They support their telecommunications infrastructure on these fees more than on local universal service contributions. From this perspective, IP telephony is not just out of the accounting rates system from the economic contribution point of view, but also from the competitive perspective. While Long Distance carriers have to pay and deal with expensive settlements rates, ISPs and other IP telephony providers are exempt by default.

The regulatory disparity between IP telephony and PSTN long distance is clearly expressed in the following example. In 1997 the standard rate for a telephone connection between USA and Germany was US$1.36 per minute, and the economy rate was US$0.78 per minute. At the same time, IP telephony rates from the US to Germany were between US$0.10 and US$0.45 per minute. The interesting and imbalancing issue is that among the cost of the long distance operator was an accounting rate of US$0.20, while the IP telephony provider did not have such an obligation.

Is very likely that those countries that have not lowered significantly their accounting rates will try to incorporate IP telephony into their settlements system, despite the fact that the FCC has declared that such system does not apply to IP telephony. The argument will be similar to the one expressed after the 1997 decision. The FCC is acting unilaterally, and international telephony interconnections are a bilateral issue, rather than a univocal determination.

Licenses

Traditionally basic telecommunications services have been heavily regulated by governments, and until few years ago were the exclusivity of established monopolies. Once the liberalization process began, the regulators became aware of universal service goals and profit viability of the operators. Licenses are the most common administrative way of limiting participation in the market to certain levels, as well as constraining the operators to comply with basic commitments on universal service. Further, some governments take the opportunity to charge significant fees for the licenses, in order to invest in telecommunications infrastructure or simply to obtain new revenues for the public budget.

In contrast, value added or enhanced services, have been generally liberalized and their license process if any, has been really simple and inexpensive. Most of the times these services are exempt from universal service contributions and there have not been any limitations for the number of competitors in the market. Further, usually no charges have been attributed to these kinds of licenses.

From the IP telephony perspective, licenses will not be a big issue for those countries that have liberalized their international long distance service. It would be similarly easy to get a basic telecommunications license or a value added license, depending on how the respective regulator classified IP telephony. It does not mean that one kind of license is not more beneficial in terms of regulation than another. It simply means that it will be possible to enter into the market under similar conditions with either license.

In contrast, those countries that have not liberalized their international voice telephony service will face short-term difficulties regulating IP telephony from the license perspective. Until now, ISPs have been providing services as value added services or simply under no specific license regulation. At the same time, traditional international voice telephony operators have competed, negotiated and paid for licenses that in most of the cases guaranteed specific duopolies or oligopolies.

Until ISPs started to provide voice services over the Internet, every day with improved quality, there had been no conflict between the ISPs and the PTOs. In the case of phone to phone IP telephony, the ISPs claimed that they had to compress the voice signal from the PSTN to packets in order to send it through the Internet, a process that made the service a value added one. Further, they could claim they were not delivering public voice services because their service was limited to a close user group, which is integrated by the ISP's subscribers.

This argument can be used not just to avoid the market access limitation, but also to escape the other repercussions of being considered a basic telecommunications operator. At this point we are pushed to a vicious circle, where the basic key regulatory issue will be defining IP telephony status. As mentioned earlier, the European Commission notice seems to be the most aggressive and complete move towards defining IP telephony in the regulatory context, but it is far from a definitive solution.

There are two schools of thought regarding IP telephony's international regulatory future. Many governments may not worry about it for a while because IP telephony makes up only a small portion of telecom revenues, and they think it very well may stay that way. Some countries, however, may be concerned because traditional carriers are using their long distance revenues to invest in telephone network build outs. When IP telephony carriers undercut those fees with their 3 cents-a-minute services, the regulators may conclude, it will affect telecom network investments adversely.

The analysis of some specific countries' telecommunications liberalization is important to illustrate which will be the regulators that may engage in a restrictive understanding of IP telephony. Despite the fact that some countries' agendas involve only short-term goals, it does not mean that the next step will be a full liberalization as noted above. Notice that the commitments are restrictive; in other words, when a government commits to liberalize national long distance and does not say anything about international long distance, it implies that, at least for a while, such service will not be liberalized.

Stands of Various Countries

USA

In the United States telephony services are split between Local Exchange Carriers (LECs) and InterExchange Carriers (IXCs). The generally monopolistic LECs provide local telephone service, whereas the IXCs provide long-distance service between LECs, making up a highly competitive, albeit regulated, industry.

Most long-distance phone calls in the U.S. involve an LEC connection on both ends (with the long-distance carrier as the bridge). Each time an IXC terminates or originates a call through an LEC, the IXC pays the LEC an access charge of roughly 3 cents per minute on each end. This access charge is greatly inflated but it covers Universal Service obligations. However, in the early 1980's the FCC ruled that providers of 'enhanced services', like Internet Service Providers (ISPs), need not pay these access charges. ISPs are treated, as "end users" who can purchase lines that have no per minute charge for receiving calls from their customers.

America's Carriers Telecommunication Association (ACTA)[12] filed a petition with U.S. Federal Communications Commission (FCC) to prevent companies from selling Internet Telephony software and to "institute rulemaking proceedings defining permissible communications over the Internet".

ACTA argued that it is not public interest to permit long distance services through Internet Telephony thus depriving those who maintains infrastructure for the same and also it is not in public interest for these selected communication operators to operate outside regulatory requirements that are applicable to all other telecommunication provider. ACTA argued that FCC has regulatory control over the Internet and should take action with regard to technology of long distance calling. ACTA further argued that goals such as Universal Service and fair competition in the telecommunications market were being thwarted by Internet Telephony.

In May 1997, while not explicitly ruling on the ACTA petition, the FCC ruled against requiring ISPs to pay per-minute access charges - instead an increase in fixed charges on each phone line for business users was implemented, ISPs included. This was in accordance with The U.S. Telecommunications Act of 1996 which clearly states that it is the policy of the United States Government "to preserve the vibrant and competitive free market that presently exists for the Internet and other interactive computer services, unfettered by Federal or State regulation".

The ACTA petition merely identified producers of software for the output and input of audio, some of who may have coincidentally been offering IP Telephony services as providers of IP Telephony services. The initial definition of Internet Telephony (as

[12] ACTA's membership consists primarily of small to medium-sized resellers of long-distance services; larger companies like AT&T, MCI and Sprint are not concerned with ACTA or their petition since they are 'wholesalers of capacity'. Internet telephony is not a form of competition in their market. ACTA's main corporate purpose is to represent these small resellers of long-distance services in legal and political spheres

merely the use of the Internet to transmit 'real-time' audio either from PC to PC or from PC to phone) neglected a third, next generation type of Internet Telephony - phone to phone.

While the first two types of Internet Telephony are inherently tied to the PC (including necessary software) and Internet Service Providers, the third type, however, is not. In phone-to-phone Internet Telephony the customer, using an ordinary telephone, dials an access code and then the telephone number; the access code then routes the call to a special computer gateway (the IP network). The trouble is that local computer gateways for companies offering this type of service must be placed in strategic geographic areas. For instance, if a customer using phone-to-phone Internet Telephony plans to call London (England) from Toronto (Canada), then local gateways must be located in both London and Toronto. The gateways convert audio into data for transmission across the IP network and then convert incoming data back into analog signals.

The FCC's definition of phone-to-phone IP Telephony requires that such services:

1) Hold themselves out as providing voice telephony or facsimile transmission service;
2) Do not require the customer to use CPE different from that CPE necessary to place an ordinary touch-tone call (or facsimile transmission) over the public switched telephone network;
3) Allow the customer to call telephone numbers assigned in accordance with the North American Numbering Plan (and associated international agreements); and
4) Transmit customer information without net change in form or content.

In the FCC's report to Congress it states, "when an IP telephone provider deploys a gateway within the network to enable phone-to-phone service, it creates a virtual transmission path between points on the public switched telephone network. From a functional standpoint, users of these services obtain only voice transmission, rather than information services such as access to stored files. Routing and protocol conversion within the network does not change this conclusion, because from the user's standpoint there is no net change in form or content". Given this, together with the

Telecommunications Act's (1996) definitions of a telecommunications carrier, telecommunications service and 'telecommunications' - as the transmission, between or among points specified by the user, of information of the user's choosing, without change in the form or content of the information as sent and received - it seems readily apparent that phone-to-phone IP Telephony companies should be required to pay access charges for connecting to and/or the usage of the local phone companies' systems. In the absence of a more detailed case-by-case investigation, however, the FCC withheld any definitive conclusion regarding whether phone-to-phone IP Telephony should be properly considered a 'telecommunications' rather than an 'information' service.

With regards to specifically PC-to-PC Telephony, the FCC held that "Internet service providers over whose networks the information passes may not even be aware that particular customers are using IP telephony software, because IP packets carrying voice communications are indistinguishable from other types of packets (in which case the) Internet service provider does not appear to be 'providing' telecommunications to its subscribers." While it would only be fair to presuppose this will also apply to PC-to-phone IP Telephony, big business cannot make that assumption.

EUROPE

In the wake of the European Union’s January 1, 1998 telecom liberalization, the European Commission ("EC") has adopted a directive (under article 1 of Directive 90/388/EEC) declining for now to regulate voice communications over the Internet. This issue is confronting regulators across the globe. The Commission issued a draft notice on the status of voice on the Internet as a supplement to Directive90/388/EEC on competition in the markets for telecommunications services. Directive 90/388/EEC defined in detail the services which Member States may continue to reserve to their telecommunications organisations.[13]

The Commission identified three categories of Internet telephony:

1) Computer to computer voice services -- voice communications transmitted via the Internet between computers;

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2) Computer to phone voice services -- voice communications transmitted via the Internet between a computer and a telephone; and
3) Phone-to-phone voice services -- voice communications transmitted via the Internet between telephones connected to a public switched telephone network ("PSTN").

According to the directive, none of these categories can yet be considered as voice telephony because each fails to meet the EU's definition of voice telephony.

The EU says that Internet telephony can only be considered voice telephony so long as:

1) It is offered commercially;
2) To the public;
3) From and to public switched network termination points - meaning that it must connect two network termination points on the PSTN; and
4) Uses direct transport and real-time switching, thus, providing a similar level of reliability and quality as produced by voice communications services using the PSTNs. When this designation is made, Internet telephony will be subject to the voice telephony regulations that it is currently immune from.

Given that no service has yet been offered that meets these conditions, the Commission has ruled that Internet telephony services cannot yet be considered 'voice telephony' and is not subject to formal licensing requirements at this time. The EC's position, however, may be amended pending developments in the market and in the technology that Internet telephony employs. Were the technology to evolve to the point where the provision of Internet telephony would meet the conditions set forth in the directive, it would then be designated as 'voice telephony' and subject to the EU's regulatory regime, including licensing requirements and universal service contribution obligations. The Commission will review the situation again before January 1, 2000.

Implications of the Notification

Licensing

As non-voice telephony services, Internet voice services fall within the liberalized area for telecommunications services. Accordingly, even in the five countries (i.e. Greece, Spain, Luxembourg, Portugal and Ireland), which have yet to fully liberalize their telecommunications market, Internet telephony services may be offered on a commercial, competitive basis.

Moreover, pursuant to Directive 90/388/EEC, the provision of telecommunications services other than voice telephony, may be subject to a general authorization or a declaration procedure. As long as Internet Voice is considered not to be a voice telephony service, a requirement for an individual licence may, therefore, not be imposed on providers of such services.

Universal Service

To the extent that Internet Voice is considered not to be a "voice telephony service", no universal service charges or contributions may be required from Internet telephony providers.

Interconnection

As non-"voice telephony service" providers, ISPs are not subject to the terms and conditions of the EU's Interconnection Directive. For example, ISPs need not provide interconnection to their networks at any technically feasible point or at cost-based rates. At the same time, however, ISPs may not access cost-based interconnection rates offered exclusively to voice telephony providers. Given that interconnection is one of the most significant costs for ISPs, it may, however, be preferable for Internet Voice providers to operate under the voice telephony regulations in order to benefit from more advantageous interconnection and access conditions.

Future

The key element that will determine the regulatory classification of Internet Voice services is quality of service. The Status notice also says that Internet telephony providers will "be subject to the regulatory regime applicable to voice telephony in the future, as soon as they will offer a quality of service equivalent to traditional voice telephony"(emphasis added). It is not clear how the Commission will assess, on a going-forward basis, whether Internet voice services offer a quality of service equivalent to traditional voice telephony. Given the difficulty in assessing and comparing the quality of service of various service offerings, it is likely that the Commission will rely on qualitative, and not quantitative, measures to determine the future regulatory classification of Internet voice services.

JAPAN

Japan is the world's second largest telecommunications market after the U.S. In August 1997 in response to the proliferation of callback services in Japan,[14] the Japanese government removed all restrictive regulations on Internet telephony service, and ten companies have since entered the market and offer services in competition with incumbent telephone companies. In its August 8, 1997 ruling, Settlement on the Guideline to Liberalize the Provision of International Internet Telephony Services, MPT announced - "From today, international Special Type II carriers who wish to provide international Internet telephony services must first form, or make any necessary changes to, a non-tariff contract with an international Type I carrier. Then, after obtaining the approval of the Minister of Posts and Telecommunications, Special Type II carriers or other international Type II carriers connected with them will be able to provide services."[15]

Therefore, although Internet telephony services have been liberalized in Japan, the provision of these services requires the prior approval of MPT. Rather than adopting a "hands-off" policy in respect to Internet telephony services, MPT has imposed some traditional regulatory requirements and obligations on Internet telephony providers.

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[14] See www.totaltele.com, July 9, 1997. It has been reported that AT&T and other providers that offer call-back in Japan are undercutting the incumbent operator's (i.e. KDD) rates by up to 90% on calls placed to the U.S.

[15] See www.mpt.go.jp/pressrelease/english/telecomm/news8-26.html. In Japan, carriers are classified as either Type I or Type II. Type I carriers provide services using their own telecommunications circuit facilities. Type I carriers need the permission of MPT to enter the market. Type II carriers provide service through circuits leased from Type I carriers. Type II carriers include Special Type II carriers (i.e., carriers that provide service to an unspecified number of general subscribers and of a scale that exceeds the criteria established by the applicable cabinet order or that provide international VAN service using an international leased line) and General Type II carriers (i.e., all other Type II carriers). Special Type II carriers must register with MPT and General Type II carriers must file a notification with MPT.

Prior to the Japanese government's decision to allow Internet telephony, many of these carriers petitioned Japan's Ministry of Posts and Telecommunications (MPT, the key regulatory agency in Japan), to regulate Internet telephone services in order to maintain an "orderly market." This euphemism is generally understood to mean, "change which is gradual enough to allow domestic incumbents time to adapt or catch up." Most of these incumbent carriers are now offering or preparing to offer Internet telephony services themselves. KDD, for instance, has already established a subsidiary named KDD Communications (K-Com) to offer international Internet services including voice services.

As communications expenses represent a large percentage of business cost in Japan, reducing them has become a high priority. Many information management teams in Japanese multinational corporations have been tasked to examine the feasibility of employing their intranets for voice communications, and one large trading house recently conducted an experiment using Internet VPN (Virtual Private Network) for voice communications between its Tokyo headquarters and its office in New York City.[16]

Some industry analysts predict that eventually the market for Internet telephony will saturate as facility-based carriers such as KDD reduce international telephone rates to levels comparable to those offered by Internet telephony providers. They further predict, however, that Internet telephony will be increasingly integrated with PCs to view, transmit, and manipulate any variety of multimedia applications, any one of which could prove to be a significant market niche.

MEXICO

Mexico committed to competition in all public telecommunication services, including voice telephone services. However, it has foreign equity limitation of 49% for the telecommunications operators. The Mexican government has indicated unofficially that it

[16] This Internet VPN works as a closed private network so that all calls from Tokyo to New York are extension calls. The company found that it costs only about 70 yen per three minutes compared to KDD’s PSTN (Public Switched Telephone Network) service, which costs some 450-yen for the same three minutes.

would apply traditional telephone regulations and restrictions on phone-to-phone IP voice services, but it has not taken any official action either.

TURKEY

The Government of Turkey (GOT) policy is to encourage Internet usage in Turkey. Under current laws and regulations, the Turkish Ministry of Transport and Communications (MOTC) regulates the market with the advice of State-Owned Turk Telekom, Inc., (TTAS). Current laws and regulations give TTAS a monopoly right on voice through the end of 2005.

A draft Telecom Law, when passed, will establish an independent regulatory body. The new law includes EU Directives. If unchanged, the new law will terminate TTAS's voice monopoly on January 1, 2004.

With the exception of voice over Internet (IP telephony) there are no Internet access restrictions. IP telephony is prohibited at the moment. However, TTNET (Turkish Telekom's Internet backbone) has started test marketing IP telephony in early 2000.

INDIA

VSNL, India's monopoly national telecommunications provider is believed to have taken the stance against Internet telephony in response to the potential threat of its customers using the Internet to make cheaper national and international phone calls.

But a number of industry experts, including the chairman of VSNL, say there is no way VSNL could detect people using the Internet for telephony, and prosecutions made against offenders probably wouldn't stand up in court.

VSNL is once again changing its view on Internet telephony as the time for its divestment keeps coming nearer. The current approach is in sharp contrast to the threats made by VSNL against its subscribers earlier (Exhibit 12) VSNL has not employed any device to detect the illegal use of voice transfer over the Internet. As a result, "Even if VSNL files a case against those providing such services, there is no way it can prove IP telephony was being provided," said an ex Indian Department of Telecommunications official, who asked not to be named.[17]

While the Department of Telecommunications has been quick to respond to the advent of Internet telephony, India's telecom regulators have yet to address the subject.

The problem VSNL has, detecting IP telephony users arises because its gateway access for Internet services, the node that routes Internet traffic, cannot differentiate between voice and data, said "I doubt whether it's possible for anyone today to regulate the use of technology too much,'*' a VSNL official said.[18]

India committed to competition of international long distance service in 2004. It allows one new operator in addition to the actual monopoly to provide long distance service. However, India submitted a MFN Exception List to allow the Government to apply differential measures such as accounting rates.

Latest developments

The expert committee on Internet telephony set up by the Government has given its green signal for allowing the service from April, 2002, once Videsh Sanchar Nigam Ltd (VSNL) loses its monopoly on international voice traffic. As a rider, it has sought that cost-based tariffs for basic services should be in place for a year. In its final report submitted to the Department of Telecommunications (DoT), it has recommended that any licensed PSTN/PLMN/NLD/ILD operator may be permitted to offer Internet telephony service as per licence conditions of the respective services. However, till the cost-based tariffs are put in place, all types of Internet telephony viz. PC to PC, PC-to-phone and phone-to-phone should remain illegal.

It has pointed out that the national long distance operation (NLDO) have already been opened to private participation. In case Internet telephony is offered by a different set of

[17] http://www.chennaionline.com/business/features/itele3101.asp

[18] http://www.chennaionline.com/business/features/itele3101.asp

operators, this is likely to affect the operations including return on investment of NLDOs. ``In India there are cross-subsidies i.e. the existing tariff and rental are not cost-based. Long distance revenue is used to subsidise local access. Cost-based tariffing would therefore have to be started before introducing Internet telephony," it has said.

The committee - comprising top officials from DoT, TEC, BSNL, VSNL and MTNL - has also ruled out separate licences for the IP telephony service provider (IPTSP). ``It was felt that any new category of service providers will lead to technical problems such as interconnections, numbering, routing, charging and tariff plans, which may be difficult to sort out as the new service provider will have to interconnect with all the existing operators including national long distance operators and may lead to litigations," it added.

In another significant recommendation, it has noted that the Internet service providers (ISP) should not be allowed to offer Internet telephony. ``The telecom service providers like basic, cellular and long distance operators have the universal service obligation. In the case of ISPs, they do not contribute towards the universal access levies; neither do they pay any entry/licence fees. In case Internet telephony is offered by a service provider who do not have universal service obligation or does not contribute towards universal access levies, then raising the resources for universal access and telecom infrastructure development will be a problem," it said. In this context, it has also noted that there would be an impact in the revenue of the incumbent operators in case the operator offering Internet telephony has lighter regulations. There should not be any difference in the regulatory conditions for carrying of telephony traffic through different means.

The Telecom Commission is at present considering the recommendations, following which the Telecom Regulatory Authority of India (TRAI) is expected to start the due process of consultation regarding terms of quality of service, interconnection and numbering. The telephone call starts from a PC that has special software to convert the sounds into digital codes. These are passed on to the Internet Service Provider (ISP), which breaks the digital messages up into packets - pieces of the message each encoded with a destination address. The packets go to the Internet. Using packets allows multiple parties to share digital lines so data transmission is much more efficient than traditional phone conversations, each of which requires a line. The packets go to the Internet Telephone Service Provider (ITSP), which reassembles the packets as they arrive and converts them to speech. This then goes to the local public telephone network, which directs the call to the right phone number. The ITSP charges for the local call and a handling fee.

OTHERS

Following are noted WTO countries that still have market restrictions on international long distance voice telephony and possibly would have complications in accepting a total liberalized IP telephony service.

ANTIGUA & BARBUDA offered to liberalize international voice telephony services by 2010. Other basic telecommunication services such as data transmission and private leased circuit services will be liberalized by 2012. While these countries have open full competition in Internet and Internet access services, they have excluded expressible Internet voice services.

BANGLADESH committed to issue two licenses to provide domestic long distance, in addition to the Government owned operator. It has full competition in voice transmission over closed user groups and also for Internet access services. However, Bangladesh indicated that it would review the possibility of adding regulatory principles in the future. Further, it submitted a Most Favored Nation (MFN) Exemption List in order to allow the Government to apply differential measures, such as accounting rates.

BULGARIA has a monopoly for the public voice telephony. It committed to liberalize this service in 2005. However, it will be partially liberalized in 2003 on the condition to use the monopoly's infrastructure.

CZECH REPUBLIC committed to full competition in all forms of voice telephony after 2000. It has open markets for voice over closed user groups. It is specified that the Internet provider must be awarded a license for IP telephony operations.[19]

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[19] Zang Rong, *"Current Status and Future of IP Telephone"*, available at www.telecomn.com

DOMINICA has an exclusive telecommunications operator. It has full competition for value added telecommunication services, Internet and Internet access services. However, it excluded expressible voice over Internet.

GHANA has a duopoly for domestic and international long distance telephony service. However, it committed to have a policy review in 2003. It has full competition in Internet and Internet access. However, it excluded voice over Internet.

GRENADA has an exclusive operator for basic telecommunications sector. It committed to initiate a liberalization process of voice telephony by 2006. It has full competition in value added, Internet and Internet access service. However, it excluded voice over the Internet.

HUNGARY committed to initiate competition in domestic long distance and international public voice telephony in 2003. Also it offered to introduce competition to the local voice service on 2004. It is specified that the Internet provider must be awarded a license for IP telephony operations.[20]

INDONESIA has a number of operators with exclusive rights for public voice telephony. The exclusivity expires in 2005 for international long distance service and in 2006 for domestic long distance. It has competition for Internet access services, subject to using networks of a duopoly for the international traffic.

ISRAEL committed to three international voice service operators. Further, Israel will review competition policy on this service in 2001 and on domestic voice services in 2002. It has competitive market access for voice over closed user groups and international private lease circuit services. However, it excluded voice from this last service.

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[20] Zang Rong, *"Current Status and Future of IP Telephone"*, available at www.telecomn.com

PAKISTAN has exclusive operator for the international voice telephony. However, it committed to competition in 2004. It allows competition in domestic voice telephony, value added services and Internet. However, it submitted a MFN Exemption List to allow the government to apply differential measures, such as accounting rates. The government has prohibited personal computer (PC)-based IP voice services, but it doesn't appear to be enforcing that ban.[21]

SINGAPORE committed to initiate competition in facilities-based telecommunication services in April 2000. It would license two additional operators at a first stage, allowing additional operators at a later time. However, it reserved the right to limited foreign equity participation to 49% for facilities based supply. It has domestic and international resale of public-switched capacity with exception of the connection of leased lines to public network, for basic services, including voice. The IP telephone is considered as both the basic telecommunications service and Internet application, and different regulation methods for different definitions are adopted.

SLOVAK REPUBLIC committed to competition in public voice services and network infrastructure by 2003. It has competition for voice telephony within closed user groups and leased circuits services with no connection to the public network.

SURINAME committed to duopoly competition of public voice telephone services. It has liberalized non-public voice services and Internet services. However, it excluded voice over Internet.

TRINIDAD & TOBAGO have a monopoly for voice telephony and private leased circuit services for public use. However, it committed to competition by 2010. It committed to review its policy in order to liberalize Internet and Internet access for private use.

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[21] Kim Sunderland, *"The 1999 Regulatory Outlook for IP Telephony"*, www.soundingboard.com.

FUTURE TRENDS OF INTERNET TELEPHONY

It is difficult to make a prediction about the future of IP telephony as there is no proven success model in the market, there is little historic evidence to project future developments, and both the market and the competitive environment are in a process of rapid growth and change. This induces a set of regulatory, technical and economic challenges. Estimated market size by 2003 is around $2.1billion, which is 3% of the US long distance and international market[22]. It is the fast growth rates (80% CAGR over 1997-2003) and the potential for future exponential growth (similar to the Internet) that makes the industry attractive.

The public Internet will be able to handle voice and video services quite reliably within the next three to five years once two critical changes take place:

?? An increase by several orders of magnitude in backbone bandwidth and access speeds, stemming from the deployment of IP/ATM/SONET and ISDN, cable modems, and xDSL technologies respectively.

?? The 'tiering' of the public Internet, in which users will be required to pay for the specific service levels they require.

Throughout the remainder of this decade, video conferencing (H.323) with data collaboration (T.120) will become the normal method of corporate communications, as network performance and interoperability increase and business organizations appreciate the economics of telecommuting.

Several factors will influence future research and developments in IP telephony. Currently the most promising areas for the implementation of IP telephony are in corporate Intranets and commercial extranets. Their existing IP based networks enable operators to control who can and cannot use the network. Another element that is going to play a big role in IP telephony are the gateways which provide the interoperability between voice and data networks. The economies of placing all traffic- data, voice, and video – over an IP based network will pull organizations and corporations in this direction, simply because IP will act as a unifying agent, regardless of the underlying

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[22] *Deanna K. Dokey and John S. Rushing, "Future Trends of IP telephony" a webpage in* Center for Virtual Organization and Commerce website, *September 26, 1999*

architecture. IP telephony products and services transported via the public Internet will be niche markets that can tolerate varying performance levels of that transport medium.

Now we will first look at the critical success factors for the development of IP telephony, next these findings will be condensed into a framework of drivers and restraints, and finally an attempt will be made to make an assessment about the future of IP telephony.

Critical success factors for IP telephony development

The critical success factors related to IP telephony can be grouped in the following three areas:

Defining a sustainable business model - The emergence of a sustainable and profitable business model in IP telephony will depend on the following factors:

?? *Advertisement-supported model vs. customer revenue supported model.*

This will largely depend upon consumer acceptance of banners and other advertising tools on the web, their willingness to share key data when subscribing for IP telephony services which could then be used for data mining services, and on the improvement of voice quality of public internet routed calls (standard for ad supported models) vs. private IP network routed calls (standard for customer revenue supported model). Already today, in a market with 64% of consumers saying that they would accept lower quality of service for a 50% price reduction, advertisement supported models are a threat to revenue based models. Some analysts predict that this business model will dominate future IP telephony.

?? *The sustainability of the cost advantage vs. circuit telephony*

The estimated cost advantage of IP telephony is approximately 50% over traditional telephony. However, up to 36% of these depend on US regulations, where IP telephony is defined as data streams and exempt from fees that are levied on circuit telephony. With increasing price pressure from circuit long-distance telephony, cost advantages will become even more important.

?? *The definition of higher value revenue streams* to soften the price pressure from telephony services. This includes the development of multi country conferencing services, adding voice to current community and entertainment oriented services (e.g. chats), wholesale of capacity to traditional telephony providers, and licensing agreements for IP telephony technology abroad.

?? *The ability to tackle the corporate market*. In general, corporations demand a higher level of reliability and service than private consumers and are less price sensitive, since many of them have preferred rate agreements with circuit telephone companies in place. Areas of improvement include better voice quality, system reliability and connectivity to support large-scale call center operations.

Increasing the consumer base - As with all networks, the value of IP telephony will increase exponentially with the number of nodes, i.e. the number of consumers who make use of its services (Metcalfe's Law). The relevant factors are as follows:

?? *Internet penetration* this acts as a cap on the IP telephony consumer base. Since access to IP telephony sites necessitates access to the internet, growing internet penetration grows the IP telephony consumer universe. This is even more important in countries other than the US with comparatively low internet usage and connection.

?? *Hardware requirements* Consumers need headsets and microphones to make use of IP telephony. Smart distributor deals (e.g. as free giveaways to computer purchasers) could overcome this bottleneck.

?? *User friendliness of surfaces:* Loging in and placing a call must become as simple and convenient as traditional calling. Moves like Net2phone's agreement to be embedded in Netscape's next browser are a step in this direction, for they make access much easier.

?? *Value added services:* Value added services like phone book synchronization with electronic diaries, voice messaging, caller ID, call forwarding etc will go a long way in increasing the consumer base.

Solving technical issues - The pertinent issues related to technology are as follows:

?? *Clarity of transmission* This is linked to the nature of package routing, and problems in prioritizing voice packages. Interestingly, this problem is grounded in the design of IP communication and the system backbones and not in insufficient access bandwidths, i.e. IP telephony is expected to only mildly benefit from broader bandwidth.

?? *The expected growth of wireless internet access.* While internet access on a cell-phone makes access to IP telephony sites geographically universal, the question is whether consumers will switch to IP telephony service once they are linked into the Internet or stay with the phone provider who gave them access.

Industry drivers and Restraining Factors

Based on these critical success factors, we can identify the following industry drivers and restraints:

Drivers of growth:

?? *Evolution of Common Standards:* Common Standards evolved over the past years help in assuring interconnectivity of business similar to internet.

?? *Constant Technology Improvement:* Improvement in clarity, advances in echo cancellation, silence suppression and data compression will help in making IP telephony more acceptable to the consumers.

?? *High consumer price elasticity:* Consumers highly value the ability to deliver services at significantly lower prices. Thus, IP telephony will be able to attract a substantial number of people given its cost advantage over traditional telephony.

?? *Economic Efficiencies:* IP telephony has significant cost advantage over circuit telephony. Substantial savings accrue due to lower infrastructure cost and network sharing technology.

?? *Investments in Infrastructure:* The consumer revenue based companies' extensive private data networks enable faster, more reliable and better sharing of voice and data.

?? *Money from buy-ins* The big telecom players have bought into a number of IP telephony providers and brought capital and knowledge. (on the forefront is AT&T's stake in Net2Phone).

Restrainting Factors:

?? *Absence of Established Names* No broad-scale brand equity- 34% consumers would prefer to buy IP telephony from long distance carriers.

- ?? *Low Voice Quality:* IP telephony still suffers from low quality of transmission. Class-bass queuing and IP telephony guidelines are expected to improve this.
- ?? *Commoditization of Long Distance Services:* Price-based competition in long-distance market has led to low price consumer expectations.
- ?? *Inconvenient service:* Placing a call via IP telephony is more difficult than via circuit telephony – one needs to dial access code, use of prepaid cards, etc.
- ?? *Inadequate Functionality:* Lack of compatibility with the SS7 switching protocol hinders the implementation of value added calling features (call forwarding, CLID etc.)
- ?? *Data Communication Cost Decrease:* Rapid decline in data communication cost can improve the cost structure of circuit switching telephony.

The Future

Continued Growth

IP telephony enables the integration of voice, data and video streams on the Internet. As such, we expect it to continue to grow rapidly. It is a primarily 'bits' business and will consequently continue to be impacted as the Internet grows and evolves further.

Bifurcation

The market is likely to undergo a bifurcation (parallel to what happened in the first years of Internet Brokerage) into a low end and a high end IP telephony service segment catering to different segments.

- ?? *Cheap lower end IP telephony Services* targeted at specific price-sensitive niches who will care less about lower quality (international calling for example). The prevalent business model for this segment is likely to be ad-revenue based with cheap (or free) long distance calling.
- ?? *High End Services* sub-market that will appeal both to the mass consumer market as well as the corporate sector. In addition to higher quality and reliable IP Telephony, providers will supply value added services such as conference facilities, Voice Mail, speech recognition and E-commerce solutions (such as Click2Talk options on e-tailers, etc.). The revenue model is likely to be either subscription-based or pay-per-use.

Among the competitors, partnerships and revenue sharing are likely to prevail, as a network of services will need to be provided to acquire customers and build a customer base.

Development of the Competitive Landscape

As the industry becomes more mature, we expect big network companies (like cable companies, utilities) to become increasingly involved in IP telephony, joining the big telecom players, which have already started to acquire stakes in the business. Both possess a good understanding of networking technologies and have the necessary funds to provide large-scale leverage.

CONCLUSION

IP telephony represents an efficient and inexpensive technology that will make a difference in the way that companies approach their clients. Telecommunications operators worldwide have been expanding the global telephone network based on government regulations aimed at supplying basic social needs. European Union Countries, the United States and few others have completed the first stage, represented in very acceptable phone penetration rates. However, numerous countries worldwide have not accomplished such a task. Around forty WTO countries still have different kinds of market limitations in order to allow the telecommunications operators to be profitable and at the same time continue expanding the basic network where it is not naturally lucrative.

The fact that IP telephony emerges in a liberalizing environment provides this new technology with a promising development. Yet, IP telephony will not be able to provide any social good, or expand where there are no outlets to connect to the Internet. In this sense, PSTN is a necessary and good social prerequisite to the benefits of IP telephony.

Over time, some forms of IP telephony, particularly those that originate and terminate on a user's telephone set which is connected to the PSTN are more closely approximating conventional voice telephony, not only in terms of quality and reliability, but also in their use of local PSTN access facilities. As a result, even those regulators that have maintained an ambiguous approach towards IP telephony are starting to re-consider the application of telephone subsidy charges. The Universal Service Report published by the FCC signals a preparedness to levy local access and universal service fund charges on IP telephony services.

The EU has signalled a similar willingness to reconsider the application of universal service charges and other regulatory burdens on IP telephony services once their quality and reliability develop to a level where they are similar to conventional "real time" voice telephony. Canada has moved to impose traditional regulatory subsidies on IP telephony faster than the US or EU, since the CRTC has ruled that local access contribution charges are payable on Internet access lines used for voice telephony purposes.

Several other countries, mostly in the developing world, have gone further than trying to "level the playing field" between IP and conventional voice telephony. These countries have moved to prohibit IP telephony services outright. A few other countries, which are in the minority so far, have gone the other way, and have assertively authorized the provision of IP telephony services subject to relatively light-handed forms of regulation. This is the case in Japan.

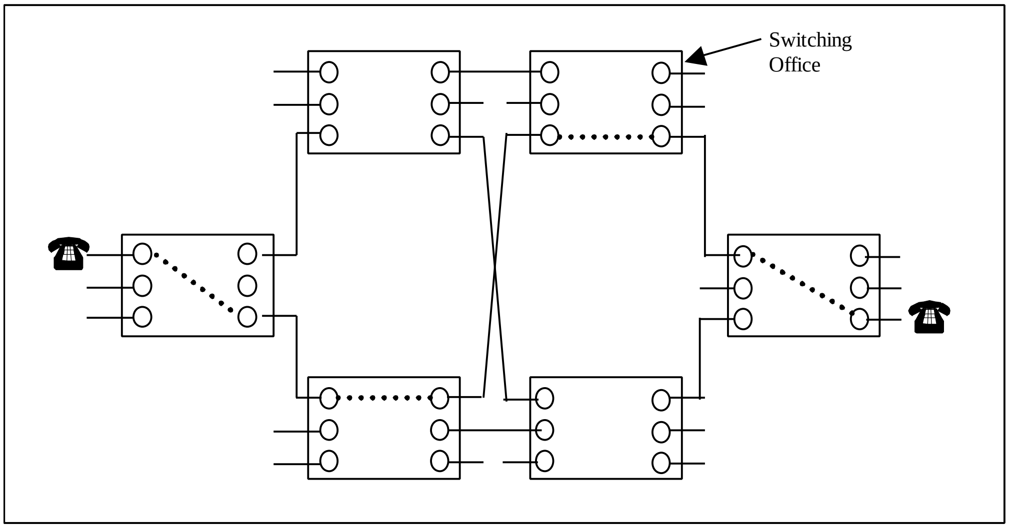
In summary, in most industrialized countries, providers of IP telephony services are starting to face a somewhat more regulated and costly operating environment than in the early days of their services. The initial advantages accruing from the avoidance of subsidy payments and other regulatory obligations will likely diminish in those markets. IP telephony services will then have to rely on other factors to compete effectively with conventional voice telephony services. These factors may include cost advantages inherent in packet switched services, service innovations such as multimedia voice applications, Internet "voice-buttons" and other value added features, and the quality, functionality and ubiquity of the international networks of individual service providers. However, in the longer run, the regulatory burdens and restrictions imposed upon all international and domestic voice service providers (including IP telephony providers) will inevitably continue to decrease.

It will never be possible for regulators to identify, restrict and tax all IP telephony providers, any more than they have been able to do so with call-back providers, refilers, switched-hubbing providers, or other operators that bypass current accounting rates. Consequently, IP telephony will add to the pressures to deregulate and simplify international and domestic regulation of the telecommunications sector.

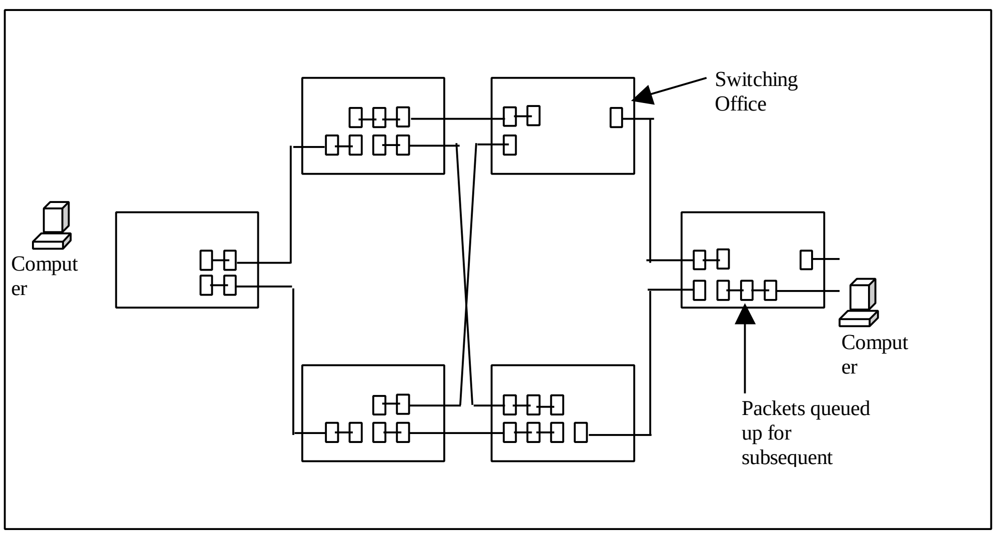
The major challenge to developing countries' regulators will be to continue the solid expansion of the basic telecommunications infrastructure while allowing new technologies like IP telephony to develop in a competitive environment.

EXHIBITS

Exhibit 1



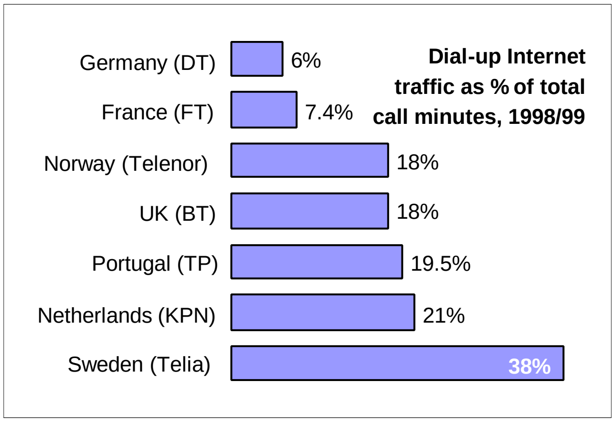
(a) Circuit Switching



(b) Packet Switching

Exhibit 2

Dial-up Internet traffic contributing to carrier revenue streamsDial-up Internet traffic as a percentage of total traffic, selected European carriers, 1998/99, and trends in dial-up Internet traffic and international traffic, April 1998 – March 2000, Hongkong SAR



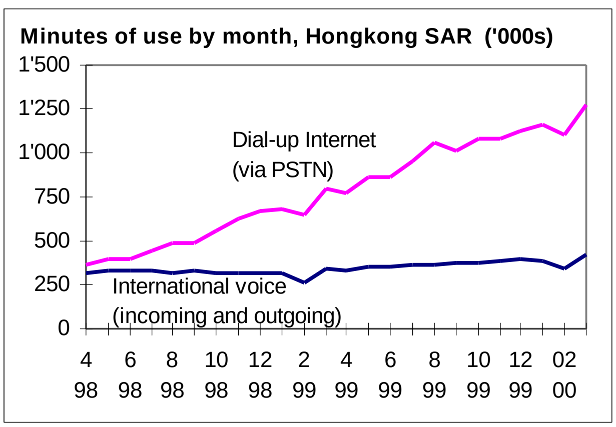


Figure 4.3: Dial-up Internet traffic as a percentage of total traffic, selected European carriers, 1998/99, and trends in dial-up Internet traffic and international traffic, April 1998 – March 2000, Hongkong SAR[23]

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[23] Carrier annual reports, OFTA, webpage in www.ofta.gov.hk.

Exhibit 3 - Top ten net settlement surplus countries

As measured by estimated net settlements from the rest of world, in US$ million, 1998

Country	Outgoing traffic 1998, million minutes	Incoming traffic 1998, million minutes	Imbalance (outgoing minus incoming)	Imbalance as % of total traffic	US settlement rate, 1998 (US cents per minute)	Estimated net settlement, 1998 (US$m)
India	436.2	1'498.8	-1'062.6	-54.9%	64.0	*680*
Mexico	1'307.6	3'060.5	-1'752.9	-40.1%	35.0	*620*
Philippines	286.4	681.2	-394.7	-40.8%	36.5	505.0
China	1'711.5	2'400.0	-688.5	-16.7%	70.0	*480*
Pakistan	87.5	640.4	-552.9	-76.0%	60.0	*330*
Viet Nam	56.0	334.0	-278.0	-71.3%	55.0	*240*
Lebanon	70.0	300.0	-230.0	-62.2%	85.0	201.3
Egypt	127.3	475.3	-348.0	-57.8%	87.5	*150*
Poland	602.4	1'144.4	-542.0	-31.0%	65.0	*145*
Dominican Rep.	157.5	730.5	-573.0	-64.5%	10.5	*130*

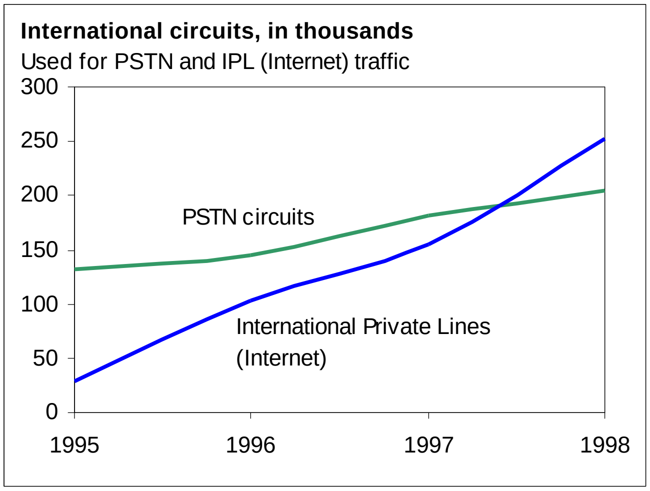
Notes: Figures shown in italics are estimates. All other figures are as reported by the countries concerned. The methodology used for estimation of net settlement is as follows: Where the country reports this indicator, it is calculated as incoming payments minus outgoing payments; where the country does not report this indicator, it is estimated by multiplying the traffic imbalance fir each country by its settlement rate to the United States during 1998.

Sources ITU/TeleGeography Inc. “Direction of Traffic Database”, FCC.

Exhibit 4

IP capacity overtaking voice capacity

Number of international circuits used for private lines (Internet) and PSTN traffic, worldwide, 1995-98, and in selected regions, 1998



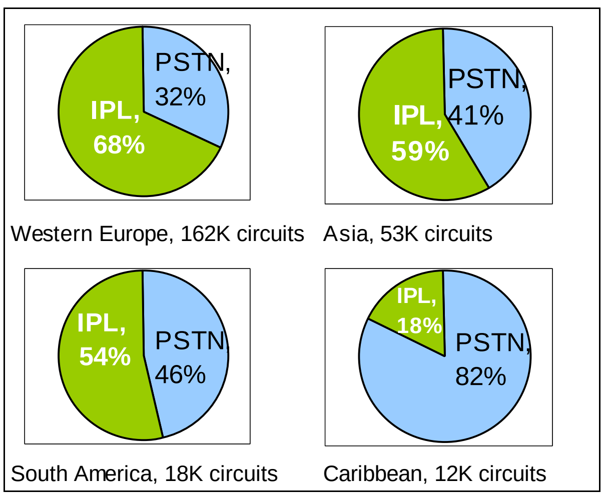


Figure 4.1: IP capacity overtaking voice capacity: Number of international circuits used for private lines (Internet) and PSTN traffic, worldwide, 1995-98, and in selected regions, 1998[24]

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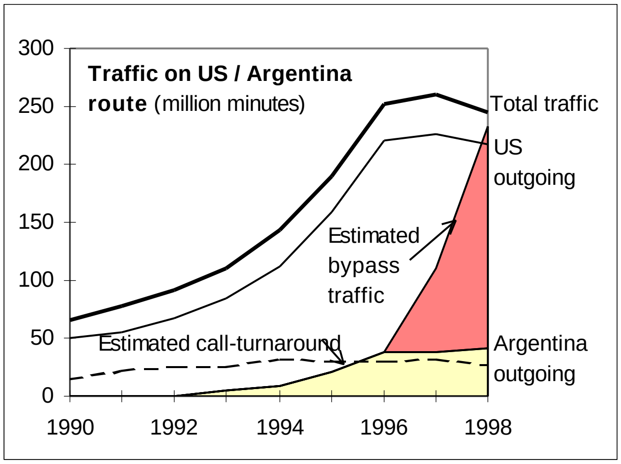
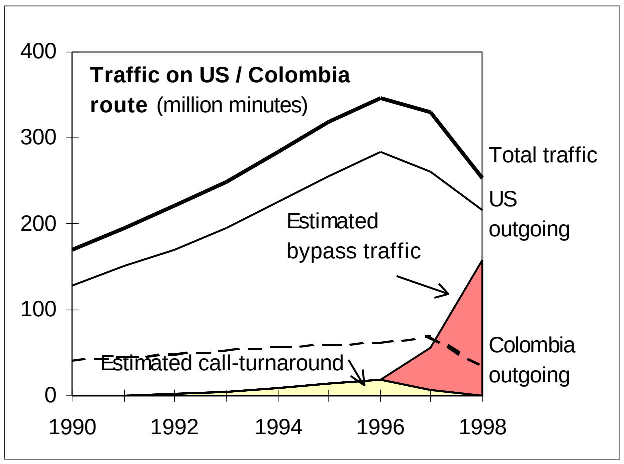
[24] *Sources:* ITU, adapted from FCC

Exhibit 5 : Where did all that traffic disappear to?

Traffic balance on routes between US and Argentina and between US and Colombia, including estimates of call-turnaround and bypass traffic





Note: "Estimated call-turnaround" traffic is the volume of traffic on a particular route that has been re-routed so that it appears that it is coming from the United States. This includes call-back, calling card and home-country direct traffic. It is estimated by applying the ratio between incoming and outgoing traffic that applied before 1992 to the subsequent traffic balance. "Estimated bypass traffic" is the volume of traffic on a particular route which is estimated to be rerouted via a least cost route (e.g., refile) or outside the accounting rate mechanism (e.g., via the Internet) such that it is not reported in official traffic statistics. It is estimated by comparing the projected growth in the total volume of traffic on the route, based on trends before 1996, with what actually happened after that date.

Exhibit 6 : Calls through the PSTN

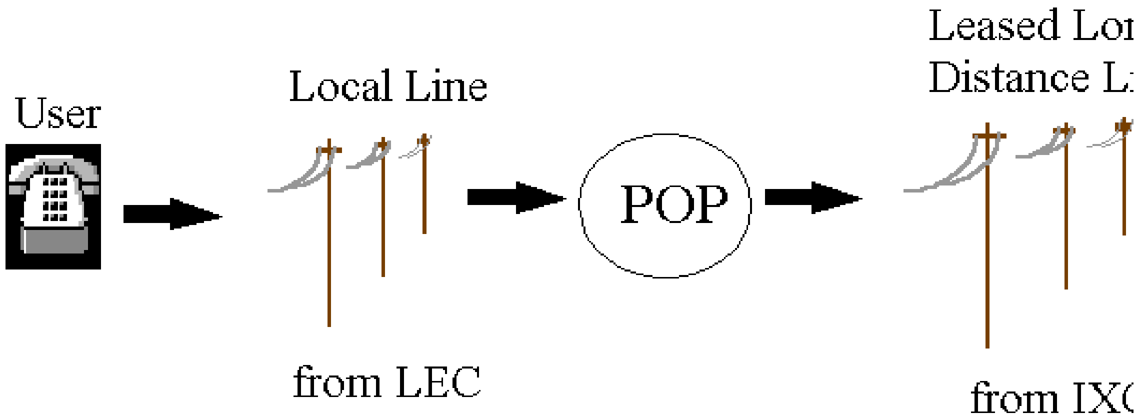


Exhibit 7 : Calls through Internet Telephony

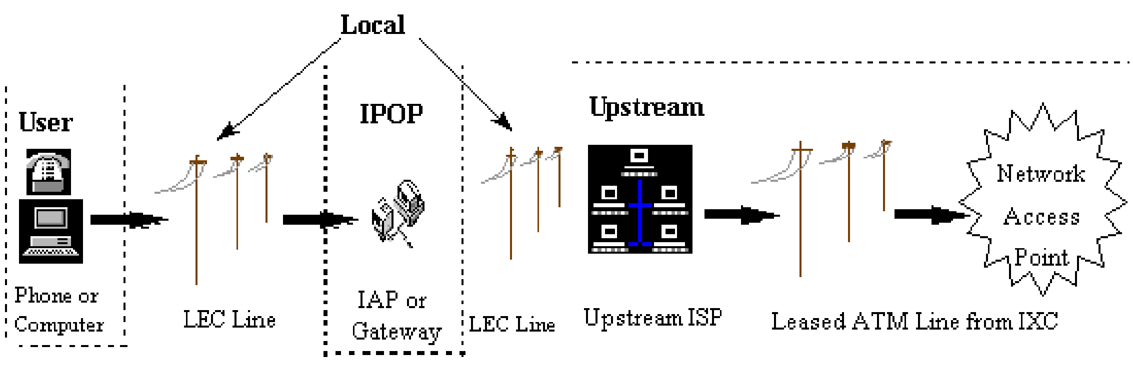


Exhibit 8 : PSTN and Internet telephony costs

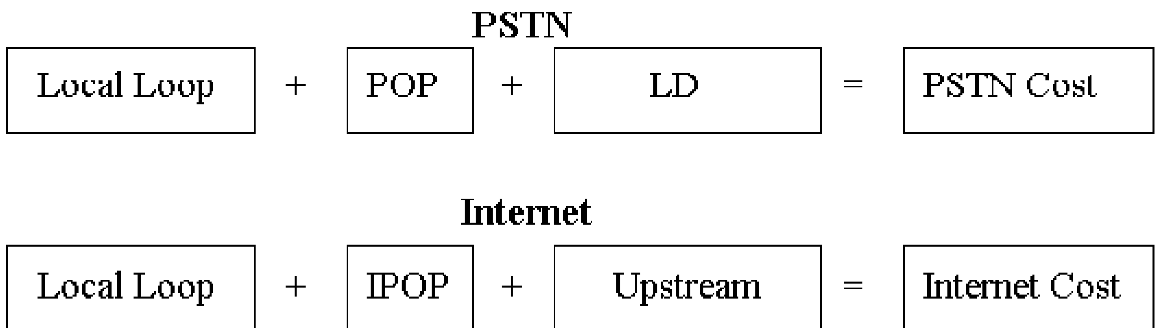
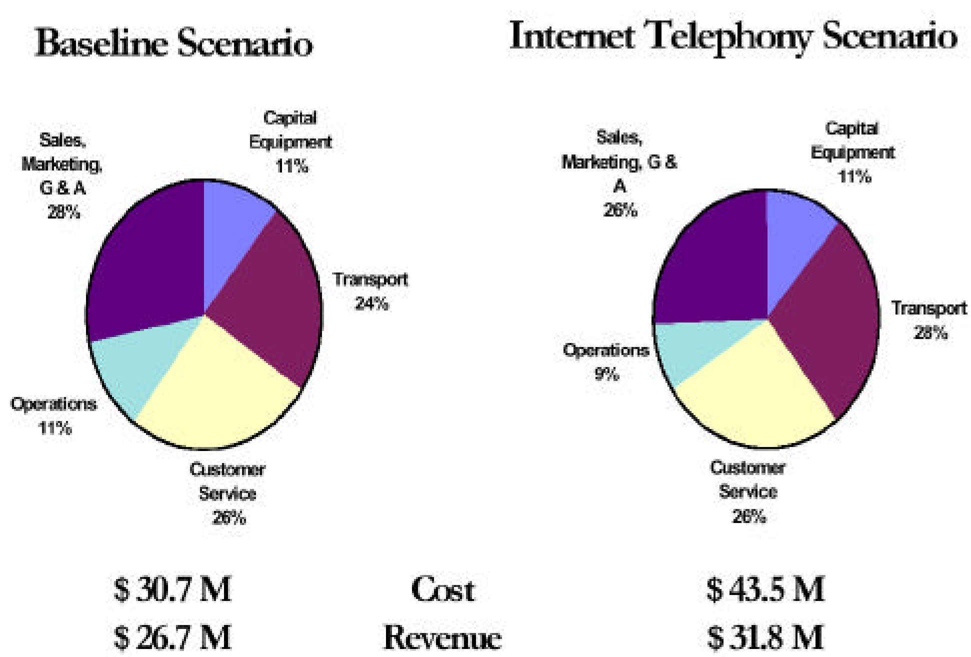


Exhibit 9: Comparative cost results for US



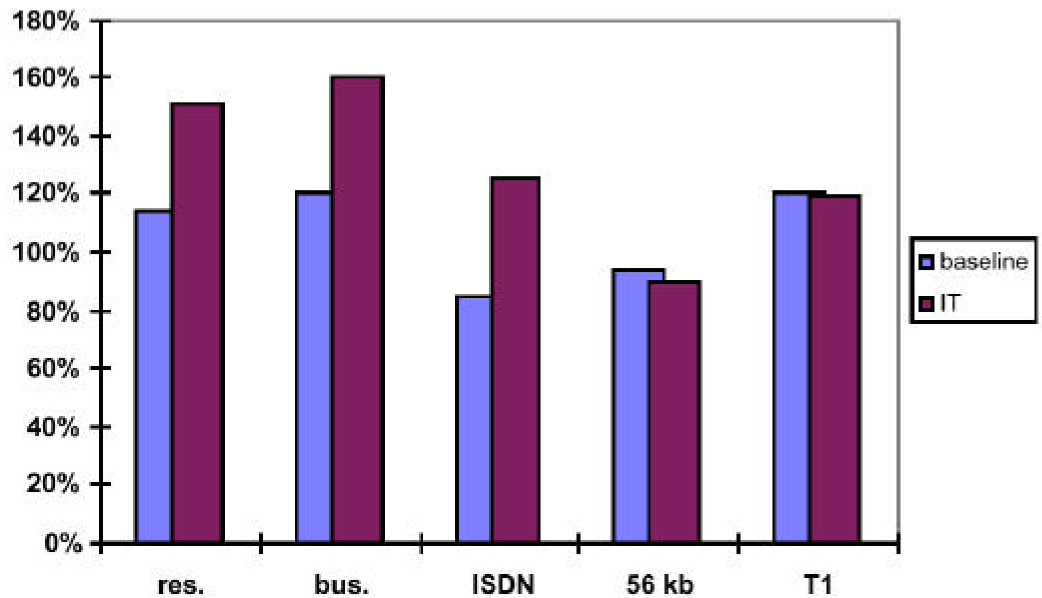
Source: “Internet Telephony: Costs, Pricing, and Policy” by Dr. Lee W. McKnight, and Bett Leida, web page in http://itel.mit.edu:/itel/pubs/itel.tprc97.pdf

Exhibit 10 : Subscriber cost increase

	RES.	BUS.	ISDN	56 KB	T1
Capital Equipment	45%	45%	80%	66%	63%
Transport	75%	75%	85%	64%	64%
Customer Service	44%	44%	44%	43%	44%
Operations	7%	7%	30%	26%	25%
Other Expenses	7%	7%	7%	78%	78%
Total	33%	34%	48%	59%	64%
Cost	$30	$32	$126	$745	$2,375

Source: "Internet Telephony: Costs, Pricing, and Policy" by Dr. Lee W. McKnight, and Bett Leida, web page in http://itel.mit.edu:/itel/pubs/itel.tprc97.pdf

Exhibit 11: Cost/Revenue Ratio for Baseline and IT Scenarios



Source: "Internet Telephony: Costs, Pricing, and Policy" by Dr. Lee W. McKnight, and Bett Leida, web page in http://itel.mit.edu:/itel/pubs/itel.tprc97.pdf

Exhibit 12

(Letter written by VSNL to its Internet consumers)

"Dear Internet Customer,

As you are aware, the usage of Telephony on the Internet is not permitted as per the terms and conditions of your Internet subscription and the Indian rules and regulations.

It has come to our notice that some agents are actively selling Internet Telephony by offering low tariffs. We would like to inform our customers that this type of usage of Internet is illegal and violative of the terms and conditions of the Internet subscription.

You are advised not to use the Internet connection for Telephony or Fax applications. VSNL would be monitoring the use of Internet and those subscribers who are found to be violating the conditions of subscription, would be permanently debarred from using Internet services."

***Source**: http://www.pulver.com., Pulver Report, (February 18, 1998).*

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