ISOA: An Architecture for PSTN-IP Interworking

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Certificate

This is to certify that the work contained in the dissertation entitled "ISOA: An Architecture for PSTN-IP Interworking" has been carried out under our supervision and the work has not been submitted elsewhere for any degree.

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dedicated to the heroes of Kargil...

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CHAPTER 1

INTRODUCTION

1.1 The Evolution of Telecommunication Networks

"Mr. Watson, come here, I want you." With these historic words Alexander Graham Bell called to his assistant Thomas Augustas Watson over the so-called "telephone", and an industry was born.

The place: 5 Exeter Place, Boston, Massachusetts

The time: evening, March 10, 1876

As with all inventions, the road had not been smooth. After many years the concept of Network came. If there were only three or four telephones in an area, it would make sense to connect each phone to all other phones and find a simple method of selecting the desired one. However, if there are three of four thousand phones in an area, such a method is out of the question. Then it is appropriate to connect each phone to some centrally located offices and perform switching there. This switching could be a simple manual operation using plugs and sockets or could be done with electronic devices. This "central office" solution is the one that has been adopted by the telecommunications industry.

Earlier networks used primitive signaling schemes, such as R2 (a series of specifications that refer to European analog and digital trunk signaling) and E&M (a trunking arrangement), to set up and tear down calls node by node. Due to limitations in signaling and transmission technology the subscribers to these networks could only be provided with a limited set of services.

As transmission technology advanced, the analog transmission infrastructure was replaced by digital transmission systems. This change resulted in better quality of service (QoS) and lower cost for service providers. Signaling evolved from primitive forms of Channel Associated Signaling (CAS) based to Common Channel Signaling (CCS) based SS7. CCS had several advantages over CAS in terms of ease of implementation, centralized control and lower equipment costs. Additionally, the high reliability of SS7, coupled with faster operations and increased capabilities, proved to be an important point in the evolution. SS7 provided the signaling capability with which the users could specify QoS requirements- in terms of bandwidth from the network. End to end digital connections could now be set up and torn down dynamically. SS7 signaling also allowed services other than voice.

1.2 What is VoIP?

"Internet" is a buzzword today. One can see many people talking about this technology. Recently, one more technology is becoming very popular and that is speaking over the Internet.

VoIP (Voice over IP), this technology comes with two important points and they are:

- Unbelievable low cost.
- Unbelievable powerful features.

These promises are what today's customer is looking for. Above all these are not only promises, but also is being delivered as of this date.

Before describing the wonders of VoIP, let us know some background of how it was and lay the foundation for how it is and will be.

1.2.1 The Traditional Way of Calling

Almost everyone has a phone installed in his or her houses. Lets suppose, Amit wants to place a call to Kamal, so he dials a particular number, say 6984359. There is a beeping noise, then follows a ringing tone and finally, someone on the other side picks up the phone. All this is O.K., but now let us understand this being some more technical.

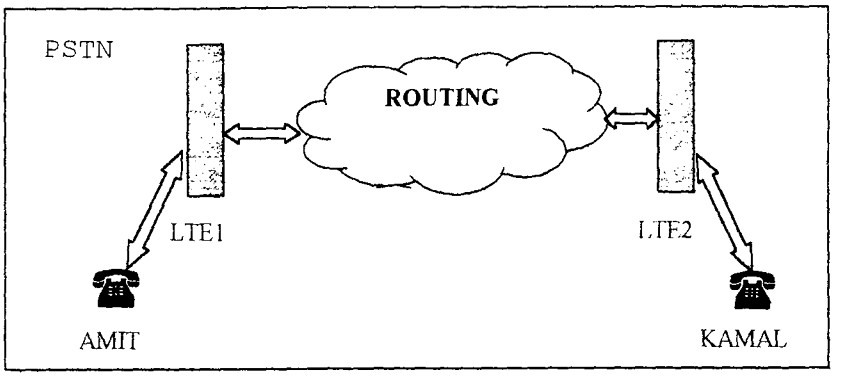

Figure 1.1 PSTN Based Connection

When Amit picks up his phone, in Telecommunication terminology it is called "off-hook". Amit's phone receives a signal that we know as a dial tone. Amit now dials Kamal's phone number 6984359 - this number is used by his local telephone exchange (shown as LTE1) to decide where to direct this call to. This decision is very similar to a network routing table. The concept is simple - Lets assume that there were only a couple of hundred telephone numbers in the world. Then, its possible that each exchange can know the destination of each and every end user phone number. However it is not the case, as the number of telephone numbers increases, it becomes impossible for each exchange to know the destination routes of every possible number.

So instead what is usually done is, that Exchanges keep a track of which other Exchanges can handle routing for which range of numbers. For example, LTE1 may be handling numbers beginning with 686, LTE2 may be handling numbers beginning with 698 and LTE3 handles numbers with prefixes 724. So now, when Amit dials 698… his local exchange (that is, the exchange to which his phone is connected to) realises that this is outside its own domain, but figures out that LTE2 is the correct Exchange for any numbers beginning with 698 and so routes this call to LTE2. LTE2 now receives the number 6984359 and realises that this is a number within its own domain and goes about contacting kamal's phone.

At many times, before ringing, one can hear a beeping noise - this is mostly to tell the user that 'I am trying to reach your destination'. Once LTE2 manages to connect to the terminal (phone) whose number is 6984359, it sends a ringing tone to the caller - this is its way of saying 'I think I have reached your destination, now I am alerting the destination of your call'. Once Kamal picks up his handset (i.e. going off hook) the communication is established.

The above description is very loose and is only a basic description of what is going on - there are more in between signaling etc. happening, which are not being discussed here.

The important thing to note here is that once Amit and Kamal are connected, a dedicated line is established between the two points. What this means is that a physical connection is established from Amit to LTE1 to LTE2 to Kamal for the entire duration of the call - what this means is that while Kamal and Amit are talking, this connection is reserved entirely for them and nobody else can use this established connection to talk.

Now lets suppose that one has to make an ISD call from India to somewhere in the US. So what would happen is that a dedicated line would have to be established from your Exchange all the way to the correct exchange in the US. And this connection could well traverse across multiple exchanges till an exchange is reached which can directly connect to the destination number. The above diagram showed only two exchanges - however, its is very much possible that many such nodes might have to be traversed to complete a call.

So its obvious then that since the end to end connection must be dedicated (that is it can't be used for more than one connection at a time), the expense of making calls will vary depending on the distance (and some other factors too, which we will not be expanding here). It is the only reason why, making calls to our neighbour is so much cheaper than speaking to our friend across the seven seas.

The above scenario is what happens when a regular phone call is made - and this is what is known as the PSTN world (Public Switched Telephone Network). This sort of a network is called a Circuit Switched Network - this is so as a complete end to end circuit needs to be established during connection from source to destination.

1.2.2 The VoIP way of calling

Now what kind of a network is the Internet? The Internet is based on a technology called TCP/IP (Transmission Control Protocol / Internet Protocol) where the IP part is essentially Packet Based. What this means is that multiple connections can be transmitted over one connection at any instant of time, unlike the traditional Circuit switched way. This is possible by dividing each connection into small packets and multiplexing them across the same connection. The following diagram compares the two technologies:

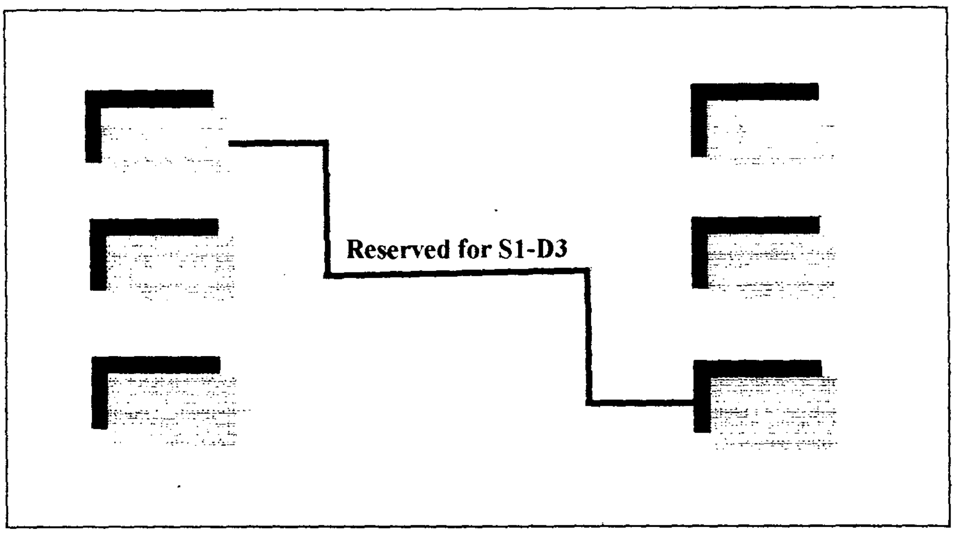

Figure 1.2 PSTN World

PSTN World: When S1 & D3 are speaking, no one else can use the same connection and must wait till this connection is released (or communicate using a new physical connection).

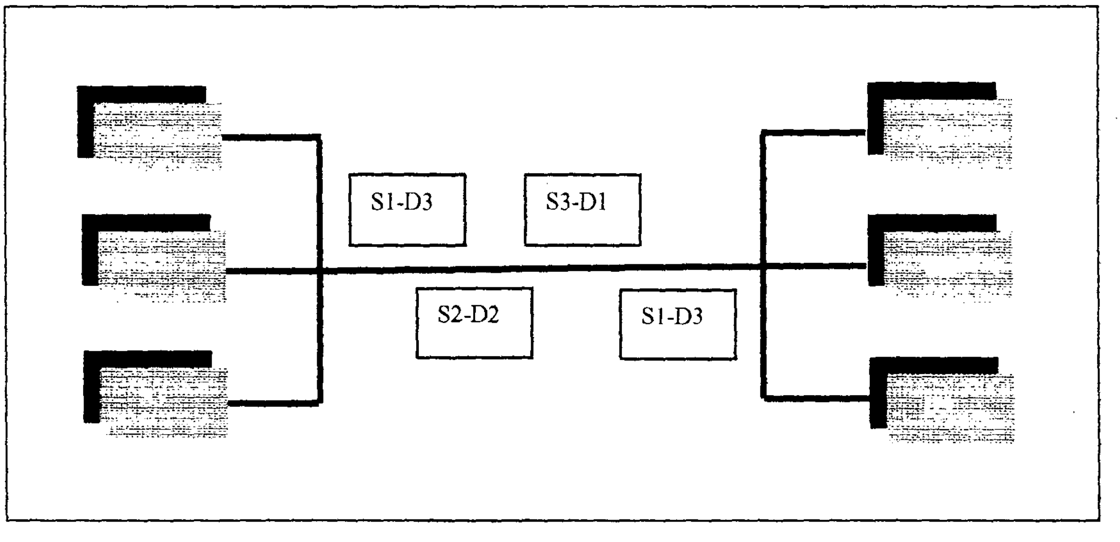

Figure 1.3 IP World

IP World: There is no dedicated connection - every one can communicate using the same connection - this is achieved by breaking each session into discrete packets and multiplexing all the packets one by one across the same connection.

So what all does this amount to? Supposing we utilise this existing packet based network as the backbone of our calls instead of the expensive dedicate lines that PSTN offers us, cant we reduce calling charges? Of course! Look at the following diagram, with the same old example.

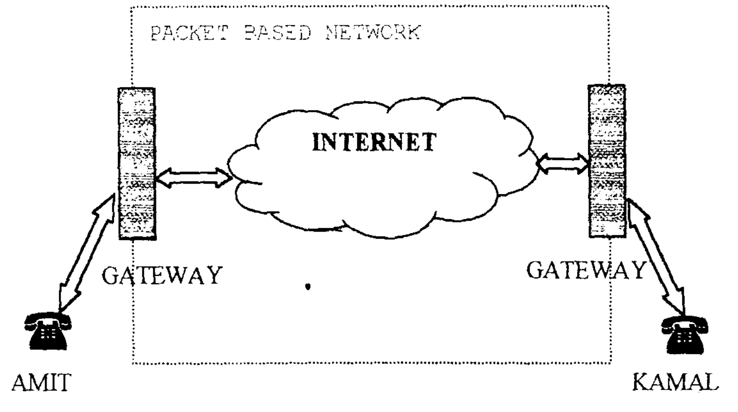

Figure 1.4 IP Based Connection

So lets describe the above. Amit calls 6984359. There is a device called a 'Gateway' that receives this request, converts it into the appropriate packet based format and forwards this request to some node in the Internet, which looks at the number, decides which destination Gateway can handle this number and forwards the request to the correct gateway, which in turn contacts the destination phone (Kamal's phone) and the call is set up.

Some important points:

- **What is this gateway and why is it required?**

Well, the phones through which we communicate using a set of protocols that is suited for the PSTN world - their signaling format is different from what is used in the Internet and hence there needs to be some conversion to and from the PSTN to IP world in order for them to inter operate. Compare this to two people. One who speaks Chinese and the other Hindi - left to their own, they will not be able to communicate to each other - we need a translator who understands both languages and can communicate with both. This is exactly what a Gateway is - it is capable of translating or Gatewaying between different protocols.

Shown above is a PSTN-IP gateway.

- **So what have we achieved?**

Notice that now there is no need for a dedicated connection between Kamal and Amit. The main backbone of communication is now through the Internet and as mentioned before, it is a Packet Based network where dedicated connections are not used.

To elaborate it more lets assume we in India connect to the Internet (using VSNL as our gateway). Now we launch up our browser and hit http://www.aajtak.com. A little later, we hit http://www.ieee.org. We get both the sites on our browser. But do we realise where the information on our screen is coming from? In the former case, the information on our screen is coming from a site somewhere in India (may be even close to where we stay) and in the latter case, this information is being transmitted to us from some where in the US. But does that mean when we are connecting to http://www.ieee.org, our phone bills show an ISD call?? No way! What we see is a charge for connecting to our local Gateway (VSNL) which is a local call and that's all (plus some fixed charge for using this service).

Now we can see the possibilities?? Is it then, not possible to extend this concept and use the existing Internet services and backbone to talk across the Internet just like we browse the Internet (and hence all over the world) at the cost of a local call??

Of course !!! And that is a part of what VoIP is all about !!! Transmit Voice across the Internet just like you transmit data - and use the already present backbone to talk to our friend across the world and forget all about STD and ISD rates.

But of course, not everything is so simple. There are legal and technical constraints to this. Many countries do not yet allow using this technology (since this means the existing PSTN providers stand to lose a lot of money amongst many other reasons). Also, passing voice across the Internet, which is essentially an unreliable network bring in all the complexities of voice getting lost somewhere in the way, delay of transmission and other such factors form technical challenges which must be met to make this a truly commercially viable technology.

- **So finally, what is VoIP?**

Simply put VoIP is a technology that aims to provide real time transmission of voice across the Internet.

Actually, the 'V' in VoIP is very misleading - VoIP does not necessarily talk only about speech transmission - it also deals with Video transmission - in general any Multimedia content across the Internet in real time.

The above diagram illustrated a case of PSTN-IP-PSTN communication. It is also possible to have pure IP-IP communication (where we don't need a Gateway), IP-PSTN communication or PSTN-IP communication.

1.3 Need for PSTN-IP Interworking

The switching technology at the core of telecommunication networks continued to be circuit switched, which has limited the spectrum of services that could be provided efficiently over these networks. In a circuit switched connection, the entire bandwidth is dedicated for the entire duration of the call. The cost of carrying voice and video over these networks is very high and the other option could be to install separate networks (for example, frame relay networks) to carry traffic for these premium services. This arrangement is also not a cost effective one. The cost of such an arrangement, in terms of duplicating physical paths, transmission facilities, and management capabilities, were substantial. These problems propelled the search for a multiservice technology that could efficiently carry all types of traffic at a very high speed on a single transmission system.

IP has emerged as one of the key technologies in such a multiservice network. It represents a milestone in the convergence of voice and data networks. In spite of growth in IP networks, PSTN networks will continue to exist. Among the major reasons for this are huge investments that have been made in these networks and the fact that in a large number of geographical areas around the world little need exists for services other than the basic ones. It is evident that even as IP networks come up, PSTN based narrow band networks will continue to exist and grow.

An interworking solution is required to bridge the gap between the two networks (PSTN and IP) and allow seamless integration. The interworking solution could be in the form of a gateway at the boundary of the two networks, which interprets the traffic and signaling going in either direction and performs the appropriate mapping and conversion between the networks on either side. The gateway takes care of the differences between the two networks such as differences in signaling, addressing and transport technology. The gateway can either be located at the edge of the IP network or as a stand-alone system between the edges of the two networks.

Some of the important requirements for such an interworking solution are the following.

- Bearer service interworking should be possible between the two networks.

- The interworking should take place transparently. A user in the IP network is not expected to invoke any special procedure or install additional hardware to place a call to a PSTN subscriber. Similarly, a user in a PSTN network should be able to place calls to the IP subscriber using existing procedures and user equipments.

- There should be no loss in service quality when the call transits from one network to another.

- The existing PSTN and IP networks should not require any hardware and software upgrades or changes.

Here, in this work we have proposed an Interworking Signaling Open Architecture (ISOA) that defines the functions and protocols to solve the problem of signaling incompatibilities between PSTN and IP networks. This is a new open and scalable architecture, which is different from the proposed architectures in the past. Almost all those architectures provide the small-scale solutions, aimed primarily at the enterprise market. Those are normally some forms of interconnections between the two networks.

ISOA is a step towards PSTN-IP interworking and it provides a new generation of IP based signaling protocols and functions required to carry the necessary variants of SS7 signaling messages rapidly and reliably in the IP networks.

ISOA is an open architecture, which satisfies the need for interoperability. ISOA provides following benefits over other similar architectures.

- **Application Portability**

Application portability benefits the end user to run the same application across a broad range of platforms/environment.

- **Interoperability**

Interoperability means users will be able to interconnect all of their systems into networks and communicate effectively within and among networks.

- **Flexibility**

ISOA distributed open architecture can easily be reconfigured to meet changing needs of the industry. It is possible to use large numbers of small systems in a variety of combinations to create a large communication environment.

There are many standardization approaches that are being carried out for PSTN-IP interworking. Internet Engineering Task Force (IETF), International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) has initiated various standardization activities. Related to these standards, the European Telecommunication Standards Institute (ETSI) project Telecommunication and Internet Protocol Harmonization Over Networks (TIPHON) undertook the effort to identify additional technical agreements required for interoperability between IP and circuit switched networks. Some industrial consortia such as the International Multimedia Teleconferencing Consortium (IMTC), through its Voice over IP (VoIP) group, also provide recommendations related to the interoperability.

This dissertation is organized as follows. The second chapter discusses the Voice over IP (VoIP) Technology, its applications standards and technical barriers in its implementation. The third chapter describes the SS7 signaling architecture. The fourth chapter provides the functional overview of the PSTN, SS7 and IP signaling environments and basic concepts associated with each. The fifth chapter describes an IP telephony signaling model designated ISOA (Interworking Signaling Open Architecture). The sixth and the last chapter conclude the dissertation.

CHAPTER 2

VoIP TECHNOLOGY : AN OVERVIEW

2.1 Definition

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

2.2 Overview

This chapter discusses the ongoing but rapid evolution of Internet telephony, the forces fueling that evolution and the benefits that users can realize, as well as the underlying technologies. It also examines the hurdles that must be overcome before Internet telephony can be adopted on a widespread basis.

Voice over the Internet. Voice over ATM. Voice over frame relay. Thinking back just a few years, most of us can admit we would have found it difficult to imagine that these telecommunications applications would be growing at the pace as it is growing today. The forecasts for the year 2002 estimate that close to 20% of all domestic phone traffic will be carried over data lines, up from less than just one percent now.

Everyone around the world wants to reduce rising communications costs. The consolidation of separate voice and data networks offers an opportunity for significant savings. Accordingly, the challenge of integrating voice and data networks is becoming a rising priority for the network industry. Organizations are pursuing solutions that will enable them to take advantage of excess capacity on broadband networks for voice and data transmission as well as utilize the Internet and company Intranets as alternatives to costlier mediums.

Truly wonderful services still require a voice engineered access gateway linking the data and telephony networks. Inside will be a comprehensive technology set that reduces the impairments caused by sending voice over data networks that were not designed to handle it. Voice processing will need to handle greater and variable delays and cancel the echoes that will be introduced from the telephony side so the voice will not sound mechanical. It also will need to mask the gaps caused by dropped packets during congestion. The packet processing (data) side of the gateway will have to adapt to variable networks and conditions and ensure the right end-to-end connections. Also, an understanding of how to handle call set up translation for different types of networks, connections, and interworking is essential for competent handling of every call.

A voice over packet application meets the challenges of combining legacy voice networks and packet networks by allowing both voice and signaling information to be transported over the packet network.

2.3 Applications Enabled by the Transmission of Voice over Packet Networks

A wide variety of applications are enabled by the transmission of voice over packet networks. Here we will explore three examples of these applications.

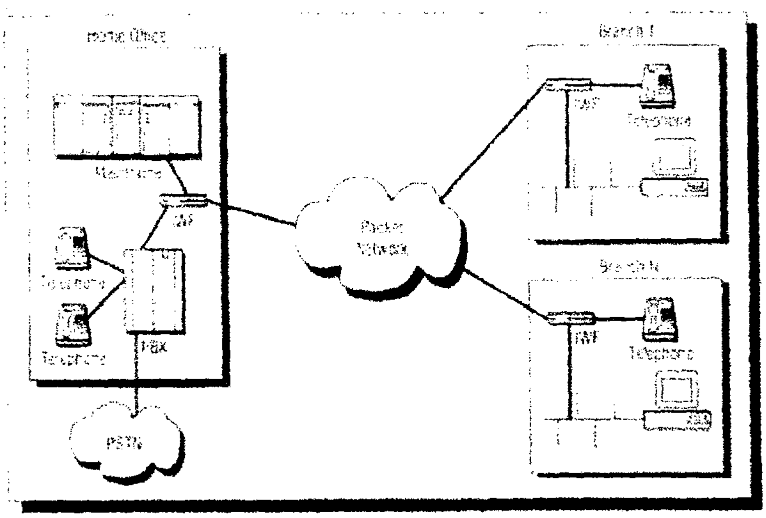

Figure 2.1 Branch Office Application

The first application, shown in Figure 2.1, is a network configuration of an organization with many branch offices (e.g., a bank) that wants to reduce costs and combine traffic to provide voice and data access to the main office. This is accomplished by using a packet network to provide standard data transmission while at the same time enhancing it to carry voice traffic along with the data. Typically, this network configuration will benefit if the voice traffic is compressed due to the low bandwidth available for this access application. Voice over packet provides the interworking function (IWF), which is the physical implementation of the hardware and software that allows the transmission of combined voice and data over the packet network. The interfaces the IWF must support in this case are analog interfaces that directly connect to telephones or key systems. The IWF must emulate the functions of both a private branch exchanges (PBX) for the telephony terminals at the branches as well as the functions of the telephony terminals for the PBX at the home office. The IWF accomplishes this by implementing signaling software that performs these functions.

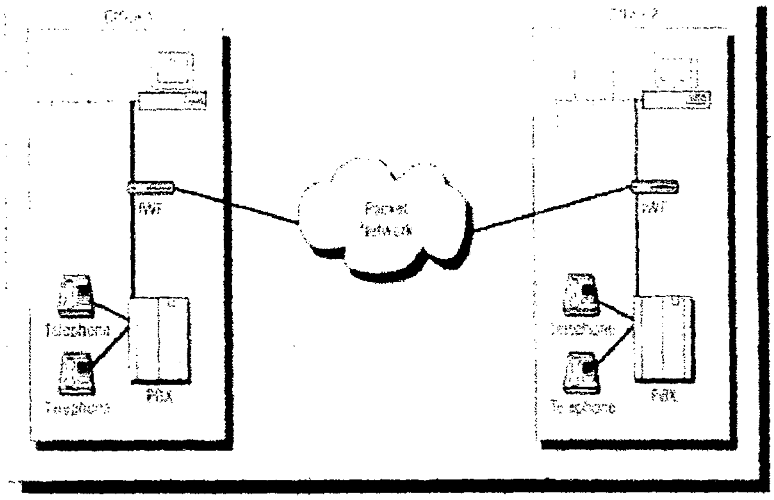

Figure 2.2 Interoffice Trunking Application

A second application of voice over packet, shown in Figure 2.2, is a trunking application. In this scenario, an organization wants to send voice traffic between two locations over the packet network and replace the tie trunks used to connect the PBXs at the locations. This application usually requires the IWF to support a higher capacity digital channel than the branch application, such as a T1/E1 interface of 1.544 or 2.048 Mbps. The IWF emulates the signaling functions of a PBX, resulting in significant savings in companies' communications costs.

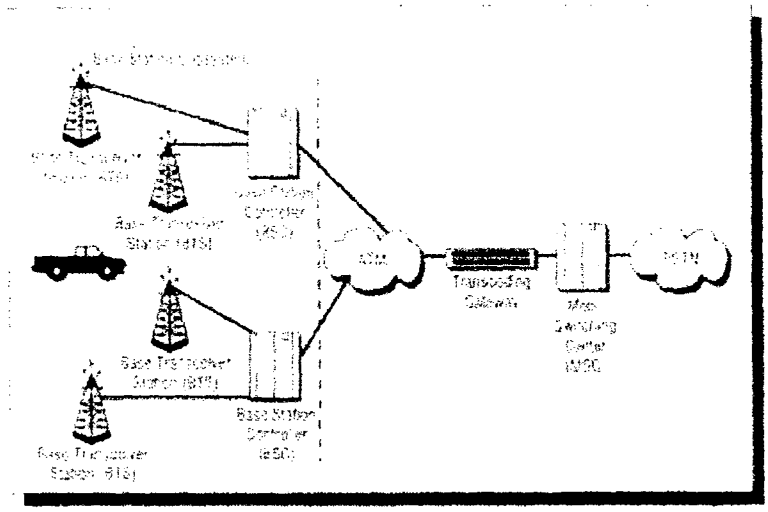

Figure 2.3 Cellular Network Interworking Application

A third application of voice over packet software is interworking with cellular networks, as shown in Figure 2.3. The voice data in a digital cellular network is already compressed and packetized for transmission over the air by the cellular phone. Packet networks can then transmit the compressed cellular voice packet, saving a tremendous amount of bandwidth. The IWF provides the transcoding function required to convert the cellular voice data to the format required by the public switched telephone network (PSTN).

2.4 Quality of Service Issues Unique to Packet Networks

The advantages of reduced cost and bandwidth savings of carrying voice over packet networks are associated with some quality of service (QoS) issues unique to packet networks. These issues are explored below.

2.4.1 Delay

Delay causes two problems—echo and talker overlap. Echo is caused by the signal reflections of the speaker's voice from the far end telephone equipment back into the speaker's ear. Echo becomes a significant problem when the round-trip delay becomes greater than 50 milliseconds. Because echo is perceived as a significant quality problem, voice over packet systems must address the need for echo control and implement some means of echo cancellation. Talker overlap (or the problem of one talker stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 milliseconds. The end-to-end delay budget is, therefore, the major constraint and driving requirement for reducing delay through a packet network.

The following are sources of delay in an end-to-end voice over packet call.

- **accumulation delay (sometimes called algorithmic delay)**

This delay is caused by the need to collect a frame of voice samples to be processed by the voice coder. It is related to the type of voice coder used and varies from a single sample time (.125 microseconds) to many milliseconds. A representative list of standard voice coders and their frame times follows:

G.726-ADPCM (16, 24, 32, 40 kbps)-0.125 microseconds
G.728-LD-CELP (16 kbps)-2.5 milliseconds
G.729-CS-ACELP (8 kbps)-10 milliseconds
G.723.1-Multi Rate Coder (5.3, 6.3 kbps)-30 milliseconds

- **processing delay**

The actual process of encoding and collecting the encoded samples into a packet for transmission over the packet network causes this delay. The encoding delay is a function of both the processor execution time and the type of algorithm used. Often, multiple voice coder frames will be collected in a single packet to reduce the packet network overhead. For example, three frames of G.729 codewords, equaling 30 milliseconds of speech, may be collected and packed into a single packet.

- **network delay**

This delay is caused by the physical medium and protocols used to transmit the voice data and by the buffers used to remove packet jitter on the receiver side. Network delay is a function of the capacity of the links in the network and the processing that occurs as the packet transit the network. The jitter buffers add delay that is used to remove the packet delay variation that each packet is subjected to as it transits the packet network. This delay can be a significant part of the overall delay because packet-delay variations can be as high as 70 msec to 100 msec in some frame-relay networks and IP networks.

2.4.2 Jitter

The delay problem is compounded by the need to remove jitter; a variable inter-packet timing caused by the network a packet traverses. 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. This causes additional delay. The two conflicting goals of minimizing delay and removing jitter have engendered various schemes to adapt the jitter buffer size to match the time varying requirements of network jitter removal. This adaptation has the explicit goal of minimizing the size and delay of the jitter buffer while at the same time preventing buffer underflow caused by jitter.

Two approaches to adapting the jitter buffer size are detailed below. The approach selected will depend on the type of network the packets are traversing.

- The first approach is to measure the variation of packet level in the jitter buffer over a period of time and to incrementally adapt the buffer size to match the calculated jitter. This approach works best with networks that provide a consistent jitter performance over time (e.g., ATM networks).

- The second approach is to count the number of packets that arrive late and create a ratio of these packets to the number of packets that are successfully processed. This ratio is then used to adjust the jitter buffer to target a predetermined allowable late packet ratio. This approach works best with the networks with highly variable packet inter-arrival intervals (e.g., IP networks).

In addition to the techniques described above, the network must be configured and managed to provide minimal delay and jitter, enabling a consistent QoS.

2.4.3 Lost Packet Compensation

Lost packets can be an even more severe problem, depending on the type of packet network that is being used. Because IP networks do not guarantee service, they will usually exhibit a much higher incidence of lost voice packets than ATM 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. The data frames, however, are not time-sensitive and dropped packets can be appropriately corrected through the process of retransmission. Lost voice packets, however, cannot be dealt with in this manner.

Some schemes used by voice over packet software to address the problem of lost frames are listed below.

- Interpolate for lost speech packets by replaying the last packet received during the interval when the lost packet was supposed to be played out. This is a simple method that fills the time between noncontiguous speech frames. It works well when the incidence of lost frames is infrequent. It does not work very well when there are a number of lost packets in a row or a burst of lost packets.

- 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.

- A hybrid approach uses a much lower bandwidth voice coder to provide redundant information carried along in the (n+1)th packet. This reduces the problem of the extra bandwidth required but fails to solve the problem of delay.

2.4.4 Echo Compensation

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). These reflections of the speaker's voice are heard in the speaker's ear. Echo is present even in a conventional circuit-switched telephone network. However, it is acceptable because the round-trip delays through the network are smaller than 50 msec and the echo is masked by the normal side tone every telephone generates.

Echo becomes a problem in voice over packet networks because the round-trip delay through the network is almost always greater than 50 msec. Thus, echo cancellation techniques are always used. ITU standard G.165 defines performance requirements that are currently required for echo cancellers. The ITU is defining much more stringent performance requirements in the G.IEC specification.

Echo is generated toward the packet network from the telephone network. The echo canceller compares the voice data received from the packet network with voice data being transmitted to the packet network. The echo from the telephone network hybrid is removed by a digital filter on the transmit path into the packet network.

2.5 Technical Barriers

The ultimate objective of Internet telephony is, of course, reliable, high-quality voice service, the kind that users expect from the PSTN. At the moment, however, that level of reliability and sound quality is not available on the Internet, primarily because of bandwidth limitations that lead to packet loss. In voice communications, packet loss shows up in the form of gaps or periods of silence in the conversation, leading to a "clipped-speech" effect that is unsatisfactory for most users and unacceptable in business communications.

The Internet, a collection of more than 130,000 networks, is gaining in popularity as millions of new users sign on every month. The increasingly heavy use of the Internet's limited bandwidth often results in congestion, which, in turn, can cause delays in packet transmission. Such network delays mean packets are lost or discarded.

In addition, because the Internet is a packet-switched or "connectionless" network, the individual packets of each voice signal travel over separate network paths for reassembly in the proper sequence at their ultimate destination. While this makes for a more efficient use of network resources than the circuit-switched PSTN, which routes a call over a single path, it also increases the chances for packet loss.

Network reliability and sound quality also are functions of the voice-encoding techniques and associated voice-processing functions of the gateway servers. To date, most developers of Internet-telephony software, as well as vendors of gateway servers, have been using a variety of speech-compression protocols. The use of various speech-coding algorithms—with their different bit rates and mechanisms for reconstructing voice packets and handling delays—produces varying levels of intelligibility and fidelity in sound transmitted over the Internet. The lack of standardized protocols also means that many Internet-telephony products do not interoperate with each other or with the PSTN.

2.6 Standards

Over the next few years, the industry will address the bandwidth limitations by upgrading the Internet backbone to asynchronous transfer mode (ATM); the switching fabric designed to handle voice, data, and video traffic. Such network optimization will go a long way toward eliminating network congestion and the associated packet loss. The Internet industry also is tackling the problems of network reliability and sound quality on the Internet through the gradual adoption of standards. Standards-setting efforts are focusing on the three central elements of Internet telephony:

the audio codec format; transport protocols; and directory services.

In May 1996, the International Telecommunications Union (ITU) ratified the H.323 specification, which defines how voice, data, and video traffic will be transported over IP-based local area networks; it also incorporates the T.120 data-conferencing standard (see Figure 2.4). The recommendation is based on the real-time protocol/real-time control protocol (RTP/RTCP) for managing audio and video signals.

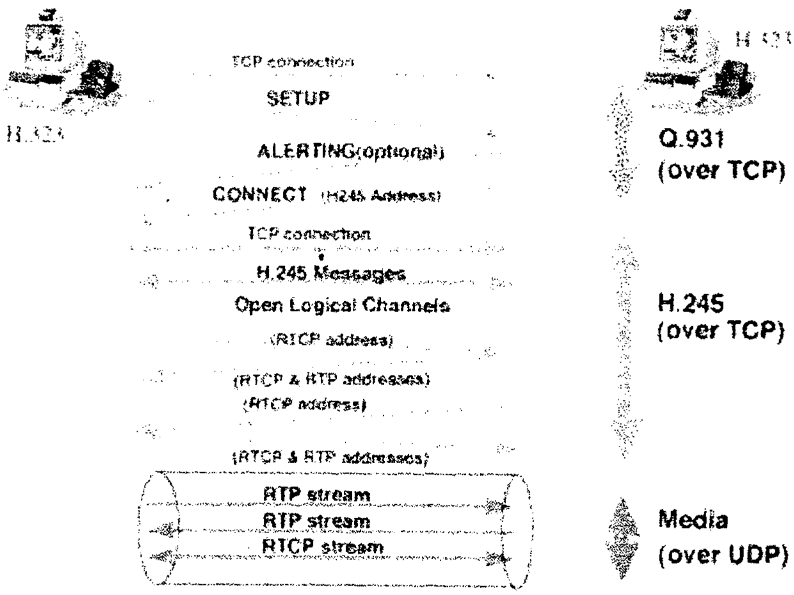

Figure 2.4 H.323 Call Sequence

As such, H.323 addresses the core Internet-telephony applications by defining how delay-sensitive traffic, (i.e., voice and video), gets priority transport to ensure real-time communications service over the Internet. (The H.324 specification defines the transport of voice, data, and video over regular telephony networks, while H.320 defines the protocols for transporting voice, data, and video over ISDN networks.)

H.323 is a set of recommendations, one of which is G.729 for audio codecs, which the ITU ratified in November 1995. Despite the ITU recommendation, however, the Voice over IP (VoIP) Forum in March 1997 voted to recommend the G.723.1 specification over the

G.729 standard. The industry consortium, which is led by Intel and Microsoft, agreed to sacrifice some sound quality for the sake of greater bandwidth efficiency—G.723.1 requires 6.3 kbps, while G.729 requires 7.9 kbps. Adoption of the audio codec standard, while an important step, is expected to improve reliability and sound quality mostly for Intranet traffic and point-to-point IP connections. To achieve PSTN-like quality, standards are required to guarantee Internet connections.

The transport protocol RTP, on which the H.323 recommendation is based, essentially is a new protocol layer for real-time applications; RTP-compliant equipment will include control mechanisms for synchronizing different traffic streams. However, RTP does not have any mechanisms for ensuring the on-time delivery of traffic signals or for recovering lost packets. RTP also does not address the so-called "quality of service" (QoS) issue related to guaranteed bandwidth availability for specific applications. Currently, there is a draft signaling-protocol standard aimed at strengthening the Internet's ability to handle real-time traffic reliably (i.e., to dedicate end-to-end transport paths for specific sessions much like the circuit-switched PSTN does). If adopted, the resource reservation protocol, or RSVP, will be implemented in routers to establish and maintain requested transmission paths and quality-of-service levels.

Finally, there is a need for industry standards in the area of Internet-telephony directory services. Directories are required to ensure interoperability between the Internet and the PSTN, and most current Internet-telephony applications involve proprietary implementations. However, the lightweight directory access protocol (LDAP v3.0) seems to be emerging as the basis for a new standard.

2.7 Future of Voice over Internet Protocol (VoIP) Telephony

Several factors will influence future developments in VoIP services. Currently, the most promising areas for VoIP are corporate intranets and commercial extranets. Their IP-based infrastructures enable operators to control who can—and cannot—use the network.

Another influential element in the ongoing Internet-telephony evolution is the VoIP gateway. As these gateways evolve from PC-based platforms to robust embedded systems, each will be able to handle hundreds of simultaneous calls. Consequently, corporations will deploy large numbers of them in an effort to reduce the expenses associated with high-volume voice, fax, and videoconferencing traffic. The economics of placing all traffic— data, voice, and video—over an IP-based network will pull companies in this direction, simply because IP will act as a unifying agent, regardless of the underlying architecture (i.e., leased lines, frame relay, or ATM) of an organization's network.

Commercial extranets, based on conservatively engineered IP networks, will deliver VoIP and FAXoIP services to the general public. By guaranteeing specific parameters, such as packet delay, packet jitter, and service interoperability, these extranets will ensure reliable network support for such applications.

VoIP services transported via the public Internet will be niche markets that can tolerate the varying performance levels of that transport medium. Telecommunications carriers most likely will rely on the public Internet to provide telephone service between/among geographic locations that today are high-tariff areas. It is unlikely that the public Internet's performance characteristics will improve sufficiently within the next two years to stimulate significant growth in VoIP for that medium.

However, 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

On the other hand, FAXoIP services via the public Internet will become economically viable more quickly than voice and video, primarily because the technical roadblocks are less challenging. Within two years, corporations will take their fax traffic off the PSTN and move it quickly to the public Internet and corporate Intranet, first through FAXoIP gateways and then via IP-capable fax machines.

Now,videoconferencing (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.

CHAPTER 3

SIGNALING SYSTEM 7 (SS7): AN INTRODUCTION

3.1 Definition

Signaling System 7 (SS7) is an architecture for performing out-of-band signaling in support of the call-establishment, billing, routing, and information-exchange functions of the public switched telephone network (PSTN). It identifies functions to be performed by a signaling-system network and a protocol to enable their performance.

3.2 What is Signaling?

Signaling refers to the exchange of information between call components required to provide and maintain service.

As users of the PSTN, we exchange signaling with network elements all the time. Examples of signaling between a telephone user and the telephone network include: dialing digits, providing dial tone, accessing a voice mailbox, sending a call-waiting tone, dialing *66 (to retry a busy number), etc.

SS7 is a means by which elements of the telephone network exchange information. Information is conveyed in the form of messages. SS7 messages can convey information such as:

- I'm forwarding to you a call placed from 212-555-1234 to 718-555-5678. Look for it on trunk 067.

- Someone just dialed 800-555-1212. Where do I route the call?

- The called subscriber for the call on trunk 11 is busy. Release the call and play a busy tone.

- The route to XXX is congested. Please don't send any messages to XXX unless they are of priority 2 or higher.

- I'm taking trunk 143 out of service for maintenance.

SS7 is characterized by high-speed packet data and out-of-band signaling.

3.2.1 What is Out-of-Band Signaling?

Out-of-band signaling is signaling that does not take place over the same path as the conversation.

We are used to thinking of signaling as being in-band. We hear dial tone, dial digits, and hear ringing over the same channel on the same pair of wires. When the call completes, we talk over the same path that was used for the signaling. Traditional telephony used to work in this way as well. The signals to set up a call between one switch and another always took place over the same trunk that would eventually carry the call. Signaling took the form of a series of multifrequency (MF) tones, much like touch-tone dialing between switches.

Out-of-band signaling establishes a separate digital channel for the exchange of signaling information. This channel is called a signaling link. Signaling links are used to carry all the necessary signaling messages between nodes. Thus, when a call is placed, the dialed digits, trunk selected, and other pertinent information are sent between switches using their signaling links, rather than the trunks which will ultimately carry the conversation. Today, signaling links carry information at a rate of 56 or 64 kbps. It is interesting to note that while SS7 is used only for signaling between network elements, the ISDN D channel extends the concept of out-of-band signaling to the interface between the subscriber and the switch. With ISDN service, signaling that must be conveyed between the user station and the local switch is carried on a separate digital channel called the D channel. The voice or data, which comprise the call, is carried on one or more B channels.

3.2.2 Why Out-of-Band Signaling?

Out-of-band signaling has several advantages that make it more desirable than traditional in-band signaling.

- It allows for the transport of more data at higher speeds (56 kbps can carry data much faster than MF outpulsing).

- It allows for signaling at any time in the entire duration of the call, not only at the beginning.

- It enables signaling to network elements to which there is no direct trunk connection.

3.3 Signaling Network Architecture

If signaling is to be carried on a different path from the voice and data traffic it supports, then what should that path look like? The simplest design would be to allocate one of the paths between each interconnected pair of switches as the signaling link. Subject to capacity constraints, all signaling traffic between the two switches could traverse this link. This type of signaling is known as associated signaling, and is shown below in Figure 3.1.

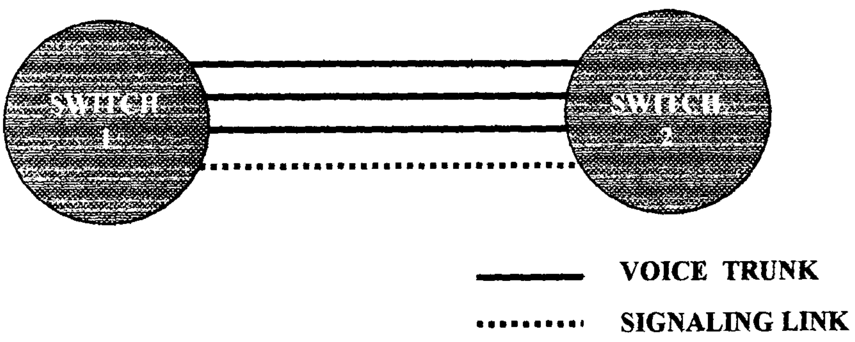

Figure 3.1 Associated Signaling

Associated signaling works well as long as a switch's only signaling requirements are between itself and other switches to which it has trunks. If call setup and management was the only application of SS7, associated signaling would meet that need simply and efficiently. In fact, much of the out-of-band signaling deployed in Europe today uses associated mode.

The North American implementers of SS7, however, wanted to design a signaling network that would enable any node to exchange signaling with any other SS7-capable node. Clearly, associated signaling becomes much more complicated when it is used to exchange signaling between nodes, which do not have a direct connection.

From this need, the North American SS7 architecture was born.

3.4 The North American Signaling Architecture

The North American signaling architecture defines a completely new and separate signaling network. The network is built out of the following three essential components, interconnected by signaling links:

- **signal switching points (SSPs)**

SSPs are telephone switches (end offices or tandems) equipped with SS7-capable software and terminating signaling links. They generally originate, terminate, or switch calls.

- **signal transfer points (STPs)**

STPs are the packet switches of the SS7 network. They receive and route incoming signaling messages towards the proper destination. They also perform specialized routing functions.

- **signal control points (SCPs)**

SCPs are databases that provide information necessary for advanced call-processing capabilities.

Once deployed, the availability of SS7 network is critical to call processing. Unless SSPs can exchange signaling, they cannot complete any interswitch calls. For this reason, the SS7 network is built using a highly redundant architecture. Each individual element also must meet exacting requirements for availability. Finally, protocol has been defined between interconnected elements to facilitate the routing of signaling traffic around any difficulties that may arise in the signaling network.

To enable signaling network architectures to be easily communicated and understood, a standard set of symbols was adopted for depicting SS7 networks. Figure 3.2 shows the symbols that are used to depict these three key elements of any SS7 network.

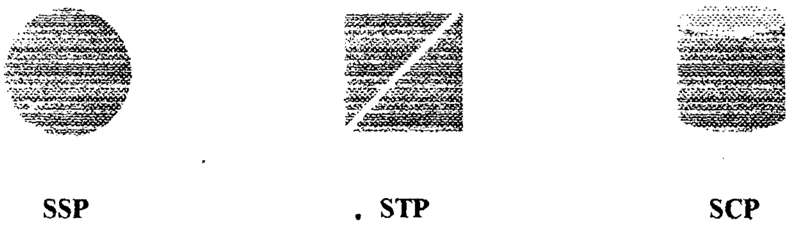

Figure 3.2 Signaling Network Elements

STPs and SCPs are customarily deployed in pairs. While elements of a pair are not generally co-located, they work redundantly to perform the same logical function. When drawing complex network diagrams, these pairs may be depicted as a single element for simplicity, as shown in Figure 3.3

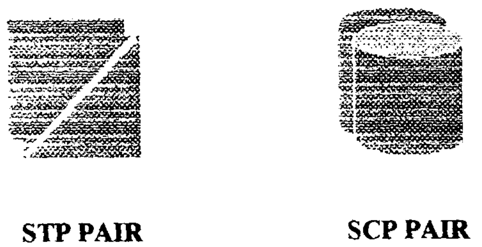

Figure 3.3 STP and SCP Pairs

3.5 Basic Signaling Architecture

Figure 3.4 shows a small example of how the basic elements of an SS7 network are deployed to form two interconnected networks.

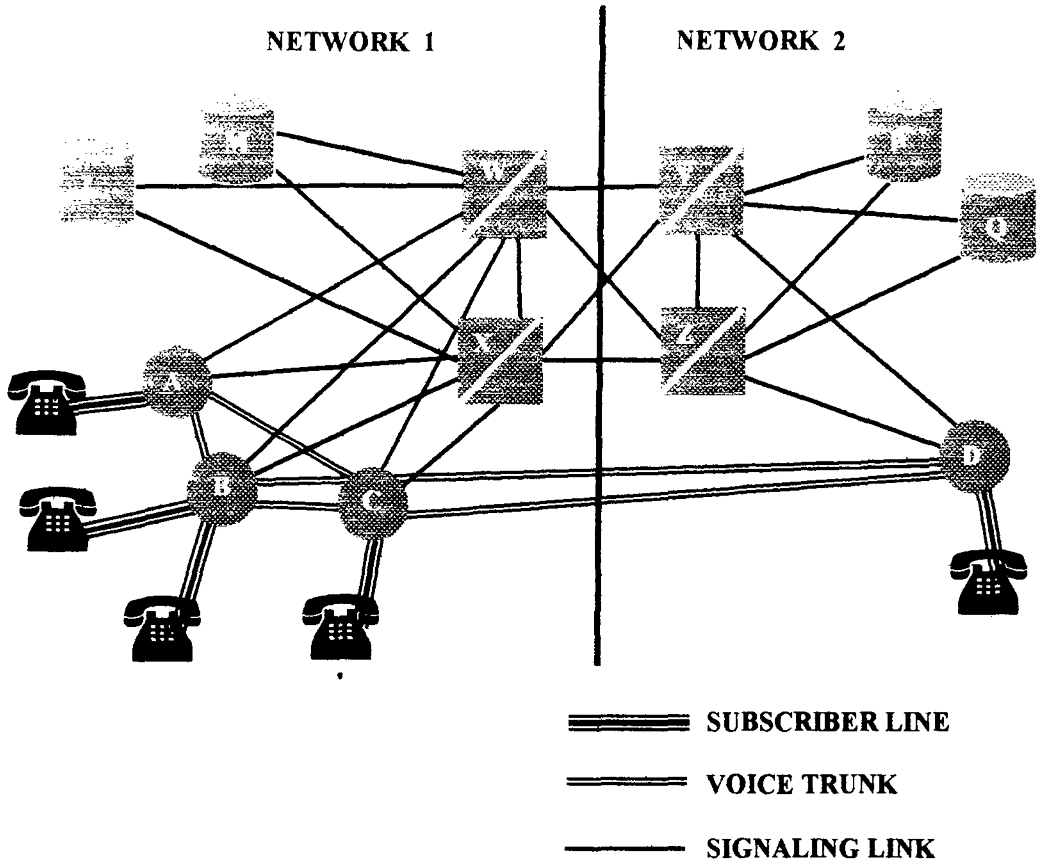

Figure 3.4 Sample Network

The following points should be noted:

- STPs W and X perform identical functions. They are redundant. Together, they are referred to as a mated pair of STPs. Similarly, STPs Y and Z form a mated pair.

- Each SSP has two links (or sets of links), one to each STP of a mated pair. All SS7 signaling to the rest of the world is sent out over these links. Because the STPs of a mated pair are redundant, messages sent over either link (to either STP) would be treated equivalently.

- The STPs of a mated pair are joined by a link (or set of links).

- Two mated pairs of STPs are interconnected by four links (or sets of links). These links are referred to as a quad.

- SCPs are usually (though not always) deployed in pairs. As with STPs, the SCPs of a pair are intended to function identically. Pairs of SCPs are also referred to as mated pairs of SCPs. Note that they are not directly joined by a pair of links.

Signaling architectures such as this, which provide indirect signaling paths between network elements, are referred to as providing quasi-associated signaling.

3.6 Basic Call Setup Example

Before going into much more detail, it might be helpful to look at several basic calls and the way in which they use SS7 signaling (see Figure 3.5).

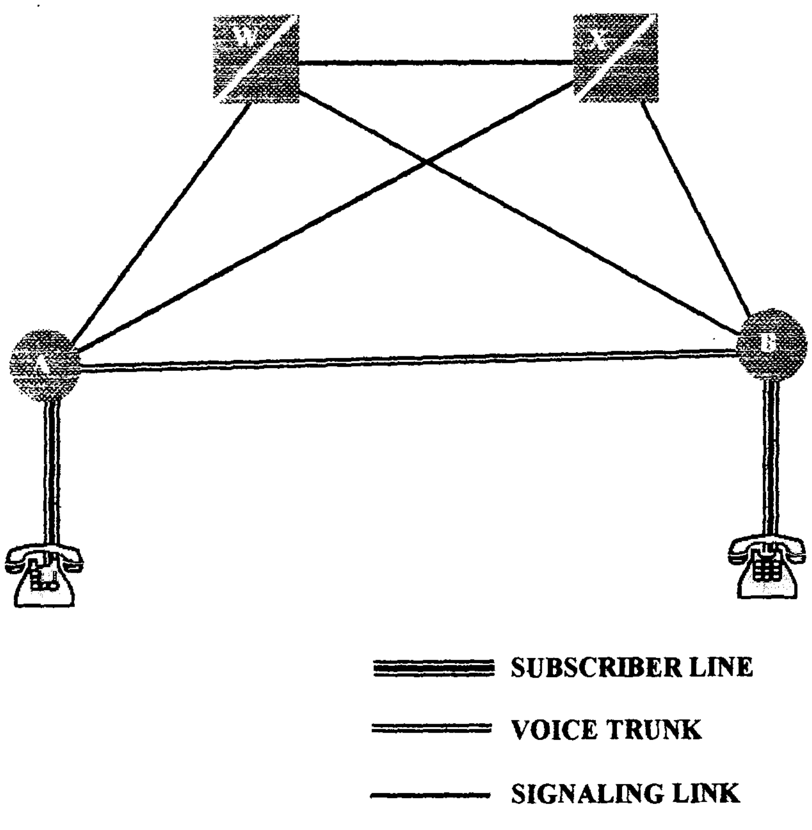

Figure 3.5 Call Setup Example

In this example, a subscriber on switch A places a call to a subscriber on switch B.

- Switch A analyzes the dialed digits and determines that it needs to send the call to switch B.

- Switch A selects an idle trunk between itself and switch B and formulates an initial address message (IAM), the basic message necessary to initiate a call. The IAM is addressed to switch B. It identifies the initiating switch (switch A), the destination switch (switch B), the trunk selected, the calling and called numbers, as well as some other information.

- Switch A picks one of its A links (e.g., AW) and transmits the message over the link for routing to switch B.

- STP W receives a message, inspects its routing label, and determines that it is to be routed to switch B. It transmits the message on link BW.

- Switch B receives the message. On analyzing the message, it determines that it serves the called number and that the called number is idle.

- Switch B formulates an address complete message (ACM), which indicates that the IAM has reached its proper destination. The message identifies the recipient switch (A), the sending switch (B), and the selected trunk.

- Switch B picks one of its A links (e.g., BX) and transmits the ACM over the link for routing to switch A. At the same time, it completes the call path in the backwards direction (towards switch A), sends a ringing tone over that trunk towards switch A, and rings the line of the called subscriber.

- STP X receives the message, inspects its routing label, and determines that it is to be routed to switch A. It transmits the message on link AX.

- On receiving the ACM, switch A connects the calling subscriber line to the selected trunk in the backward direction (so that the caller can hear the ringing sent by switch B).

- When the called subscriber picks up the phone, switch B formulates an answer message (ANM), identifying the intended recipient switch (A), the sending switch (B), and the selected trunk.

- Switch B selects the same A link it used to transmit the ACM (link BX) and sends the ANM. By this time, the trunk also must be connected to the called line in both directions (to allow conversation).

- STP X recognizes that the ANM is addressed to switch A and forwards it over link AX.

- Switch A ensures that the calling subscriber is connected to the outgoing trunk (in both directions) and that conversation can take place.

- If the calling subscriber hangs up first (following the conversation), switch A will generate a release message (REL) addressed to switch B, identifying the trunk associated with the call. It sends the message on link AW.

- STP W receives the REL, determines that it is addressed to switch B, and forwards it using link WB.

- Switch B receives the REL, disconnects the trunk from the subscriber line, returns the trunk to idle status, generates a release complete message (RCL) addressed back to switch A, and transmits it on link BX. The RCL identifies the trunk used to carry the call.

- STP X receives the RCL, determines that it is addressed to switch A, and forwards it over link AX.

- On receiving the RCL, switch A idles the identified trunk.

3.7 Layers of the SS7 Protocol

As the call-flow examples show, the SS7 network is an interconnected set of network elements that is used to exchange messages in support of telecommunications functions. The SS7 protocol is designed to both facilitate these functions and to maintain the network over which they are provided. Like most modern protocols, the SS7 protocol is layered.

Physical Layer

This defines the physical and electrical characteristics of the signaling links of the SS7 network. Signaling links utilize DS-0 channels and carry raw signaling data at a rate of 56 kbps or 64 kbps (56 kbps is the more common implementation).

Message Transfer Part—Level 2

The level 2 portion of the message transfer part (MTP Level 2) provides link-layer functionality. It ensures that the two end points of a signaling link can reliably exchange signaling messages. It incorporates such capabilities as error checking, flow control, and sequence checking.

Message Transfer Part—Level 3

The level 3 portion of the message transfer part (MTP Level 3) extends the functionality provided by MTP level 2 to provide network layer functionality. It ensures that messages can be delivered between signaling points across the SS7 network regardless of whether they are directly connected. It includes such capabilities as node addressing, routing, alternate routing, and congestion control.

Collectively, MTP levels 2 and 3 are referred to as the message transfer part (MTP).

Signaling Connection Control Part

The signaling connection control part (SCCP) provides two major functions that are lacking in the MTP. The first of these is the capability to address applications within a signaling point. The MTP can only receive and deliver messages from a node as a whole; it does not deal with software applications within a node.

While MTP network-management messages and basic call-setup messages are addressed to a node as a whole, other messages are used by separate applications (referred to as subsystems) within a node. Examples of subsystems are 800 call processing, calling-card processing, advanced intelligent network (AIN), and custom local-area signaling services (CLASS) services (e.g., repeat dialing and call return). The SCCP allows these subsystems to be addressed explicitly.

Global Title Translation

The second function provided by the SCCP is the ability to perform incremental routing using a capability called global title translation (GTT). GTT frees originating signaling points from the burden of having to know every potential destination to which they might have to route a message. A switch can originate a query, for example, and address it to a STP along with a request for GTT. The receiving STP can then examine a portion of the message, make a determination as to where the message should be routed, and then route it.

For example, calling-card queries (used to verify that a call can be properly billed to a calling card) must be routed to a SCP designated by the company that issued the calling card. Rather than maintaining a nationwide database of where such queries should be routed (based on the calling-card number), switches generate queries addressed to their local STPs, which, using GTT, select the correct destination to which the message should be routed. Note that there is no magic here; STPs must maintain a database that enables them to determine where a query should be routed. GTT effectively centralizes the problem and places it in a node (the STP) that has been designed to perform this function.

In performing GTT, a STP does not need to know the exact final destination of a message. It can, instead, perform intermediate GTT, in which it uses its tables to find another STP further along the route to the destination. That STP, in turn, can perform final GTT, routing the message to its actual destination.

Intermediate GTT minimizes the need for STPs to maintain extensive information about nodes that are far removed from them. GTT also is used at the STP to share load among mated SCPs in both normal and failure scenarios. In these instances, when messages arrive at a STP for final GTT and routing to a database, the STP can select from among available redundant SCPs. It can select a SCP on either a priority basis (referred to as primary backup) or so as to equalize the load across all available SCPs (referred to as load sharing).

ISDN User Part (ISUP)

ISUP user part defines the messages and protocol used in the establishment and tear down of voice and data calls over the public switched network (PSN), and to manage the trunk network on which they rely. Despite its name, ISUP is used for both ISDN and non-ISDN calls. In the North American version of SS7, ISUP messages rely exclusively on MTP to transport messages between concerned nodes.

Transaction Capabilities Application Part (TCAP)

TCAP defines the messages and protocol used to communicate between applications (deployed as subsystems) in nodes. It is used for database services such as calling card, 800, and AIN as well as switch-to-switch services including repeat dialing and call return. Because TCAP messages must be delivered to individual applications within the nodes they address, they use the SCCP for transport.

Operations, Maintenance, and Administration Part (OMAP)

OMAP defines messages and protocol designed to assist administrators of the SS7 network. To date, the most fully developed and deployed of these capabilities are procedures for validating network routing tables and for diagnosing link troubles. OMAP includes messages that use both the MTP and SCCP for routing.

CHAPTER 4

INTERWORKING CONCEPTS

4.1 Introduction

This chapter describes the functional overview of the PSTN, SS7 and IP signaling environments and the basic concepts associated with each.

The functional overview of the PSTN, SS7 and IP signaling environments described here revolves around the model proposed in this work for the next generation of unified networks. The model provides an open and scalable architecture to accommodate multi-vendor and multi-protocol variants under a single signaling scheme.

If one gets the solid rock reliability of the PSTN with the freedom, scope and flexibility of the Internet, then what could be better than this? and that too at very attractive prices.

The work towards scalable, reliable interconnection between the worlds of PSTN and IP is going with high speed and determination. Voice over IP (VoIP), Fax over IP (FoIP) and other interworking solutions already exist today, but most of them are small-scale solutions. Many solutions are just some form of interconnections between the two worlds in which media and its related control information are carried in the same interface, such as the ISDN Primary Rate Interface (PRI). As such, they do not offer a practical solution to the key issue of achieving network-scale reliability, performance and efficiency.

The proposed model provides truly scalable, carrier-class interworking solutions for media transportation of high bandwidth. Each of the network (PSTN, SS7, IP) comes with its unique set of strengths and weaknesses that have to be accommodated. Getting them work together is the need of the hour. Now the three networks are illustrated below:

4.2 Circuit Switched Networks

Public Switched Telephone Network (PSTN) which is a Circuit Switched network carries media streams on the hard wired, landline-based connections that provides constant bandwidth to the users. These media streams flow over dedicated connections, typically between two telephones. Circuit Switched network was originally designed and optimized for voice traffic, but today a considerable amount of fax traffic is also carried over these same voice-quality connections. With the emergence of Internet, modem data traffic occupies a significant portion of the Circuit-Switched network as it goes to IP-based data networks.

In circuit switched network the connection occupies bandwidth even when no real information is flowing. This is the worst characteristic of the circuit switched network. In normal human conversation, there is a pause between the words, even during this pause the bandwidth is consumed at the same rate throughout the connection.

There are many inefficiencies that result from using the PSTN networks as a be-all solution, which leads to finding an alternate network. PSTN has large degree of centralized functionality. They are highly reliable and statically configured. Circuit Switched network topology is rigidly engineered, thus significant coordination is required to add or modify network elements. In PSTN 'in-band' signaling is being performed.

4.3 Signaling System 7 (SS 7)

SS7 was developed to improve the efficiency and throughput capacity of the PSTN circuit-switching infrastructure. SS7 network performs 'out-of-band' signaling unlike PSTN in which all the set up procedures, as well as the call itself move together. SS7 separates these two functions. It provides a separate control-messaging network, comprising high-speed 'out-of-band' links. These links deliver only the control related messages, enabling switching elements to quickly communicate call related control information necessary to establish, bill and terminate connections.

Even though the SS7 network communicates messages that are really only data packets, any unused bandwidth is filled with 'filler' packets, which generate constant streams to help ensure the reliability of constant bandwidth dedicated connections. A key advantage of this type of approach is that any disturbance in the SS7 network is quickly detected and routed around to ensure reliable, rapid message delivery. This dedicated messaging network not only reduces call setup costs but also frees up trunk capacity to carry optimal traffic. Thus delivering on SS7 network promises increased network capacity and better network efficiency. SS7 network also implement storage of information for handling various calling scenarios or subscriber services. This information storage capability has led to the description of some SS7 enabled services, such as Advanced Intelligent Network (AIN) services. AIN is at the heart of many of today's most powerful new network management features and capabilities.

SS7 was designed in support of the circuit switched network, so it is also a connection-oriented and statically configured. Because of critical role of SS7 network, it is over engineered with critical internal elements deployed in mated pairs.

SS7 network must support protocol variations as countries implement telephony differently. This further complexity and the requisite high levels of reliability, make the implementation more complex.

4.4 Internet Protocol Networks

Earlier when Internet was developed by United States Department of Defence, was used almost exclusively for academics to exchange their research files. Now people everywhere could exchange entire libraries of information, pictures and all, with a simple modem, a computer, and a hook-up service provided by many Internet service providers (ISPs) of today and that too at a very low cost.

Like SS7, IP is also a packet-oriented network. It was designed primarily for the efficient delivery of data among devices sharing a network. IP network has some characteristics that are different from both the SS7 and PSTN networks. IP networks are configured dynamically, not statically and they are not centrally managed rather are designed to be self-learning.

In IP networks, the data is broken up into manageable sized packets and these packets are thrown to the IP network. These packets have many information like address of both the sender and receiver. These packets are dynamically guided along one of many possible paths to their final destination.

Packets for the same destination and from the same source may take different routes to reach their destination and these packets may arrive in random sequence. IP was designed such that if any of an individual network element is relocated or destroyed, the network itself survives by remaining accessible by all of its remaining elements. The actual path taken by any given packet of data is selected dynamically by any of the network transport elements still functioning.

In the IP environment, it is very well expected and known that due to various congestion or failure situations in the network, a few IP packets won't arrive at all or may get damaged. Certain high level protocols such as Transmission Control Protocol (TCP), are utilized to guarantee information delivery in the IP networks.

Most of the applications of IP networks are based on distributed model of the network rather than closed, centralized systems. They rely on cooperating distributed elements in an open architecture. This open and distributed IP model make it easy to grow incrementally with little coordination required beyond basic addressing scheme.

4.5 Convergence

The convergence of PSTN, SS7 and IP technologies can produce tremendous new openings in the telecommunication world. This requires the right combination of network interfaces and the ability to direct and manage these interactions. This converged unified network which has the strengths of these networks, will be capable of delivering unbelievable service innovations.

Next chapter describes the distributed model for interworking of PSTN and IP worlds, which will provide the most cost effective scalable systems. Unlike the PSTN, in which reliability is typically achieved by centralized elements, the IP network model tends to rely on cooperating, multiple instances of independent servers.

CHAPTER 5

THE DESIGN OF ISOA

5.1 Introduction

This chapter describes an IP telephony signaling model designated Interworking Open Signaling Architecture (ISOA) proposed by us. This is an open, scalable architecture to accommodate multi-vendor and multi-protocol variants within a single signaling framework.

To take the full advantage of the strengths of IP based networks for the new integrated IP telephony environment, a new telephony architecture was required, one that is different from the classical models typically found in the PSTN. To achieve truly scalable, interworking solutions PSTN and IP networks needs to be prepared to work together. A new generation of IP based signaling protocols and functions are required to carry the necessary variants of SS7 signaling messages rapidly and reliably in IP networks.

5.2 ISOA (Interworking Signaling Open Architecture)

ISOA includes three basic functional elements. Two of these basic functional elements are interface elements: The Circuit Switched Gateway, for interfacing media flows between IP and circuit switched networks and the SS7 Signaling Gateway, for interfacing SS7 signaling information. The third basic network element , the Media Controller, provides decision making and coordination functionality. All these elements provide the basic building blocks for a distributed architecture approach to provide 'XoIP'- Voice, Fax, and Video over IP networks.

There are also some optional elements in this model, which comprise a variety of applications and database servers that enable the addition of new services. The flexibility to add new service architectures incrementally, outside the basic media control elements in unified networks, is one of the great promises of this distributed network model. This ability to add application servers incrementally, decouples the development of new services from the basic media control functionality, thus enabling new services to be developed relatively independently.

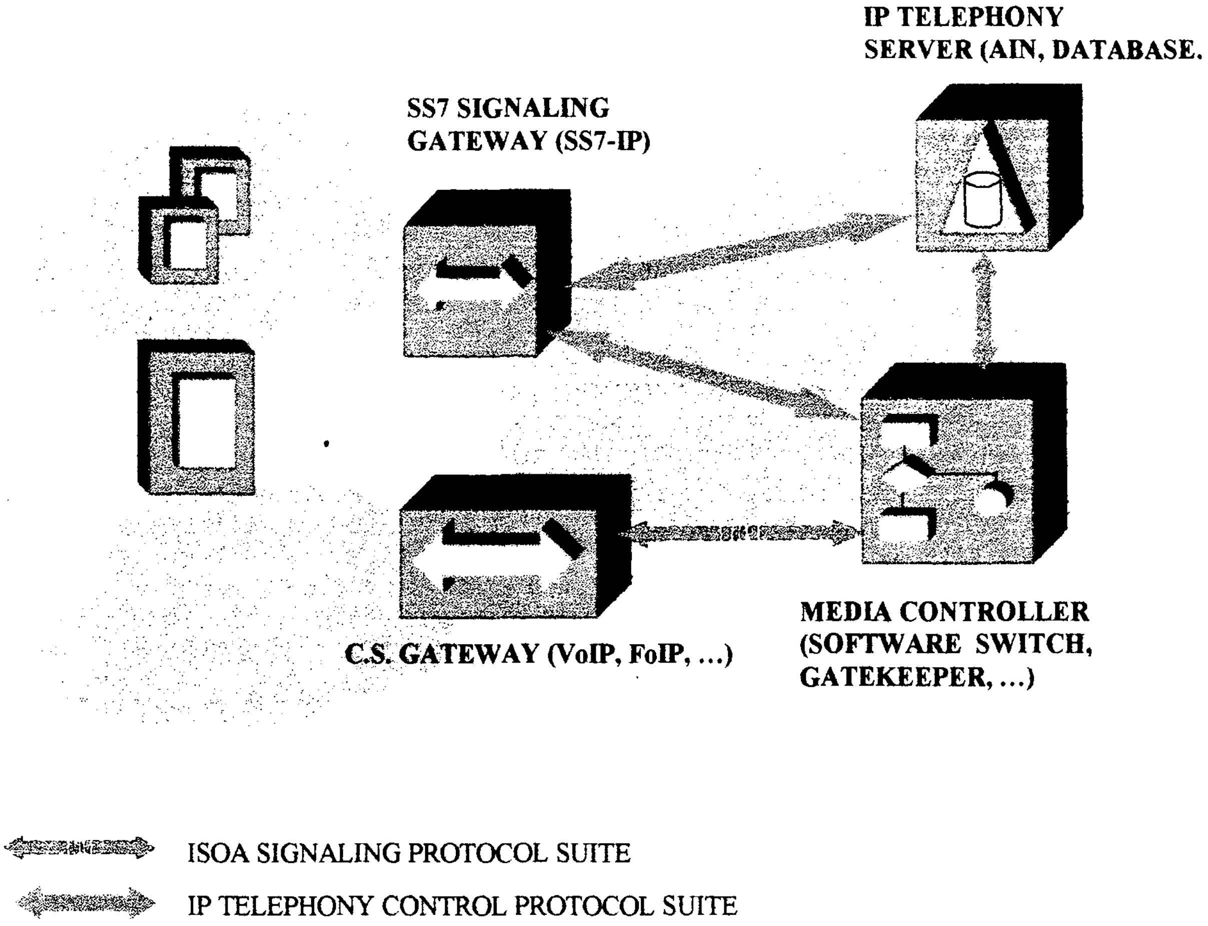

Figure 5.1 ISOA Open Architecture

5.2.1 Circuit Switched Gateway

The main task of the Circuit Switched Gateway is to allow the media of different types like voice, fax, video and modem data- to be transported as packets. These packets must be transportable, both in the IP network and as digital or analog streams in the circuit switched network. They must also be able to move without loss of integrity or degradation of quality. These various criteria are met by different-different types of coding, compression, echo cancellation and decoding schemes.

The circuit switched gateway function provides a bi-directional interface between a circuit switched network and media related elements in an IP network. Typically, circuit switched gateway will interact either with end user applications residing in computers attached to the IP network, or with other circuit switched gateways.

Circuit switched gateway can implement a variety of physical interface to the PSTN. For example, highly scalable circuit switched gateway systems can implement high speed Time Division Multiplexing (TDM) trunk interfaces, which are commonly used between switch elements in the circuit switched network.

5.2.2 Media Controller

The key responsibilities of the Media Controller are to make decisions on flow-related information and to provide associated instructions on the interconnecting of two or more 'IP elements' so that they can easily exchange information. Media Controller maintains current status information of all information flows, and they generate the administrative records necessary for activities such as billing. Most commonly, media controllers provide coordination of circuit switched gateways.

Media controllers implement much of the call control functionality found in switching elements in the PSTN. Media controllers instruct the circuit switched gateways on how to setup, handle and terminate individual media flows. They also provide the parameters associated with bandwidth allocation and quality of service characteristics. Media controllers are also used by sophisticated end user interface applications. In some protocol implementations, such as H.323, significant media controller functions are performed in network elements called Gatekeeper.

5.2.3 SS7 Signaling Gateway

The SS7 signaling gateway function implements bi-directional interfaces between an SS7 network and various call control related elements in an IP network. The key responsibilities of the SS7 signaling gateway are to repackage SS7 information into formats understood by elements in each network, and to present an accurate view of these elements in the IP network to the SS7 network. Typically, the associated IP network elements will implement media controller functions, database storage, or query functions.

The SS7 network has reliability constraints on all devices directly attached to it. Thus, one key attribute of any SS7 signaling gateway is that it needs to be always available in reporting to the SS7 network an appropriate status of all IP network elements for which it is the SS7 proxy.

By definition, SS7 signaling gateway must implement highly reliable SS7 messaging that obeys all the rules of the SS7 network, while also accommodate a variety of bahaviors in the IP network. These bahaviors may be entirely appropriate in an IP world, but not necessarily conventional, or even acceptable by PSTN standards. To actually enable IP elements, like media controller, to perform their designated administrative functions, the SS7 signaling gateway repackages the information contained in various high level SS7 message protocols- such as Integrated Services Digital Network User Part (ISUP) and Transaction capabilities Application Part (TCAP) - into formats that can be understood by IP elements.

SS7 signaling gateway need to understand each of the many variants of SS7 protocols used throughout the world. It is also highly desirable for SS7 signaling gateways, in the process of repackaging SS7 information, to be able to minimize the amount of variation presented to IP network elements. Finally, since an IP network is a shared medium lacking physical security, it is essential that SS7 signaling gateways filter out inappropriate traffic.

5.3 A New Signaling Approach

Unlike the PSTN model, in which reliability is typically achieved by centrally based elements; the IP telephony network model tends to rely on multiple independent, cooperating servers. The 'distributed' model has the potential to provide the most cost effective scalable systems. This non-linear modular approach provides, among other things, more flexibility in building system implementations. Since multiple controllers can be deployed on the basis of need, reliability can be achieved with less equipment. Different networks with unique service architectures add only the server capabilities essential for their particular needs. In some cases, even specialized servers according to particular profiles can be easily integrated into unified networks.

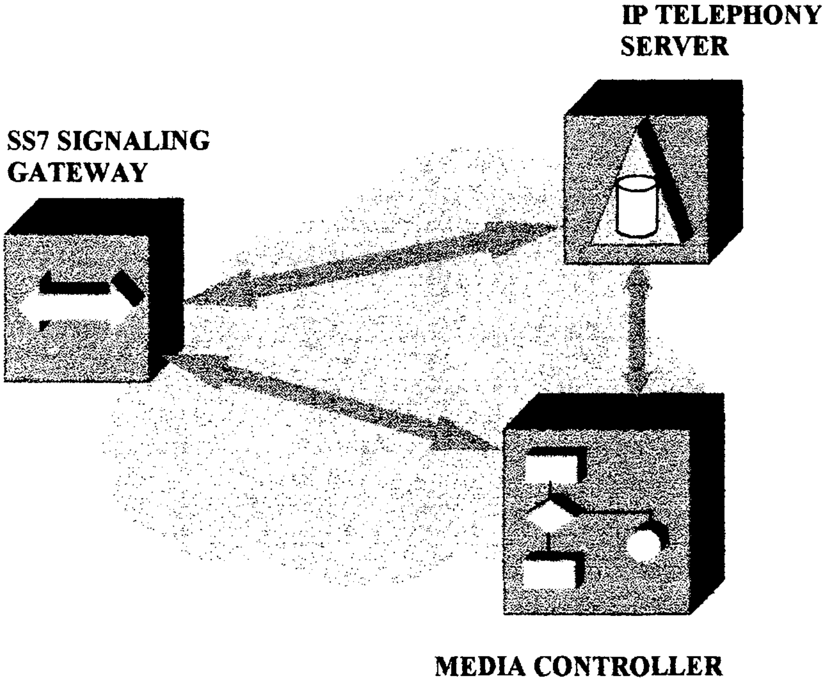

Figure 5.2 ISOA Protocol Mapping

At present, the signaling messaging protocols necessary to reliably exchange existing SS7 based signaling information between elements in the new distributed IP telephony model do not exist in classical IP protocol infrastructure. Various timing and reliability considerations unique to SS7 behavior make the use of standard IP protocols available for reliable information delivery, like TCP, less than optimum for exchanging all of the messaging that needs to flow between SS7 and IP network elements. Most SS7 protocol layers depends on guaranteed, timely, ordered delivery of messages, along with rapid detection of disturbances of any kind in the network carrying signaling information.

The ISOA architecture provides the protocols necessary for SS7 signaling gateways to communicate effectively with media controllers and other IP networks elements requiring SS7 related information.

5.4 ISOA : A New Signaling Architecture

ISOA is a new 'open architecture' definition that encompasses all SS7 related signaling protocols and functions necessary to communicate SS7 information effectively in IP networks. ISOA architecture is aimed to provide a set of common standards that supports multiprotocols, unified PSTN-IP network environments.

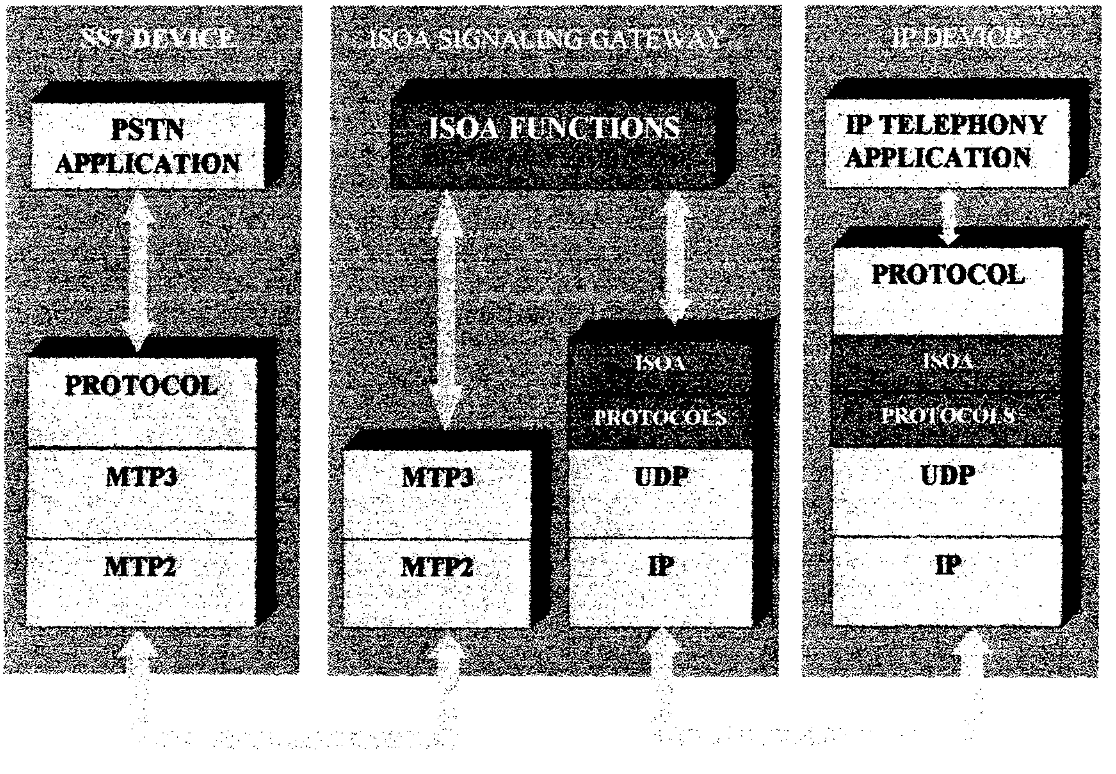

Figure 5.3 ISOA Protocols/Functions

ISOA provides for the implementation of Voice over IP (VoIP) by fully integrating about fifteen new protocols and functions. The ISOA 'open architecture' supports bi directional signaling between the two different networks. Thus VoIP networks can easily be assembled to take advantage of the full range of PSTN SS7 based 'intelligent network' capabilities.

5.5 ISOA Functional Components - Protocols/Functions

The protocols and functions of ISOA 'open architecture' are defined below. As other IP telephony initiatives, the ISOA architecture can also be enhanced according to the future needs.

- **ISOA SS7 Network Management**

It is the protocol to support SS7 MTP signaling network management principles to media controllers and other IP telephony elements to enable total network management.

- **ISOA IP Node Management**

Protocol to support media gateways and other IP telephony elements to coordinate availability with SS7 signaling gateways.

- **ISOA TCAP Transaction over IP**

Protocol to support the transport of TCAP transactions directly over IP between media controllers and IP telephony elements.

- **ISOA ISUP Messages**

SS7 signaling gateway-media controller protocol to support the encapsulation and transfer of ISUP messages between SS7 signaling gateway and media controllers.

- **ISOA SCCP/TCAP Messages**

Protocol to support the encapsulation and transfer of SCCP/TCAP messages between SS7 signaling gateways and media controllers or other IP telephony elements.

- **ISOA Transport Path Management**

Protocol to support management of the transport paths between SS7 signaling gateways and IP telephony elements.

- **ISOA Service Control**

Function to provide Intelligent Network (IN) services functionality between the IP and PSTN domains.

- **ISOA OMAP Messages**

Protocol supporting transfer of Operation, Maintenance and Administration Part (OMAP) messages between SS7 signaling gateways and media controllers.

- **ISOA Load Distribution**

Protocol allowing the distribution of signaling traffic across multiple transport paths to distribute the load.

- **ISOA Secure Communication**

IP network level function enabling secure communication between elements involved in signaling.

- **ISOA QoS**

SS7 signaling gateway functionality enabling utilization of IP Quality of Service (QoS) mechanisms.

- **ISOA Reliable Transport of Signaling Payloads**

An IP transport protocol to support the reliable, ordered delivery of PSTN signaling payloads.

- **ISOA Protocol Conversion**

Function at the application layer that translates signaling messages across the SS7 gateway to the protocols supported by IP elements.

- **ISOA SS7 Links**

Adaptation protocol to enable the use of IP transport between SS7 nodes.

- **ISOA Control**

Protocol between a media controller or other IP telephony element and a SS7 signaling gateway supporting the SS7 signaling gateway control of the offered call from the PSTN.

- **ISOA Authentication and Authorization**

Function in media controller to authenticate signaling messages and authorize requests for services on a call.

- **ISOA Address Translation**

Function that performs the translation between the E.164 and IP addresses.

- **ISOA Firewall Function**

SS7 sigaling gateway function that supports "application firewall" operating at the ISUP/TCAP level.

CHAPTER 6

CONCLUSION

Internet Telephony promises to combine our separate data and voice networks into a single transport mechanism.

Building a robost IP-based telephony service requires coordination between PSTN and IP based networks. The ISOA (Interworking Signaling Open Architecture) proposed in this work incorporates explicit coordination between the two networks. ISOA interfaces IP based networks with the PSTN through a SS7 Signaling Gateway and a Circuit Switched Gateway. ISOA is designed 'to deliver seamless intelligent interworking services between SS7 based PSTN and IP worlds. In one direction, it repackages SS7 variants, high level control information (such as ISUP and TCAP message protocols) in specific IP formats, and delivers them to dynamically addressed IP network elements. In the other direction, it formats signaling information IP packets as standardized SS7 messages, addressed and delivered with complete SS7 robustness. The ISOA 'open architecture' enables classical PSTN attributes in IP telephony networks. Thus ISOA is a contribution towards enabling a multi-vendor environment for PSTN-IP interworking and brings the SS7 reliability and scalability of IP telephony.

A number of open issues remain. There are significant barriers to acceptable QoS that must be overcome. Many of these barriers are in the form of trade-offs; finding the best combination of codec, access technology, and end-to-end architecture is challenging. Also, the mechanism for authorization and usage recording must be developed that allow service providers to charge users for telephony services.

Acronym Guide

Acronyms used throughout this dissertation are listed below.

ACM	Address Complete Message
AIN	Advanced Intelligent Network
ANM	Answer Message
ATM	Asynchronous Transfer mode
CAS	Common Associated Signaling
CCS	Cômmon Channel Signaling
CLASS	Custom Local Area Signaling Service ETSI European Telecommunication Standards Institute
FoIP	Fax over Internet Protocol
GTT	Global Title Translation
IAM	Initial Address Message
IETF	Internet Engineering Task Force
IMTC	International Multimedia Teleconferencing consortium
IP	Internet protocol
ISDN	Integrated Services Digital Network
ISOA	Interworking Open Signaling Architecture
ISPs	Internet Service Providers
ISUP	ISDN User Part

ITU-T	International Telecommunication Standardization Sector
IWF	Interworking Function
Kbps	Kilobits per second
LDAP	Lightweight Directory Access Protocol
LTE	Local Telephone Exchange
MF	Multifrequency
MTP	Message Transfer Part
OMAP	Operation, Maintenance and Administration Part
PBX	Private Branch Exchange
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
QOS	Quality of Service
RCL	Release Complete Message
REL	Release Message
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
SCCP	Signaling Connection Control Part
SCP	Signal Control Point
SONET	Synchronous Optical Network
SS7	Signaling System 7
SSP	Signal Switch point
STP	Signal Transfer Point

TCAP	Transaction Capabilities Application Part
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TIPHON	Telecommunication and Internet Protocol Harmonization Over Networks
VoIP	Voice over Internet Protocol
VSNL	Videsh Sanchar Nigam Limited
xDSL	x Digital Subscriber Line (e.g., x = A for "asymmetric", x = H for "High bit rate")

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