



NACT VoIP Industry Tutorial

Introduction

This tutorial gives background information on the VoIP industry as it exists at the time of this writing, and the view of NACT Telecommunications, Inc. as it pertains to how a large VoIP network should be set up and managed, including the interconnection of multiple VoIP carrier networks.

The first section is a glossary of terms that are used throughout the VoIP industry and in this tutorial. The terms are organized in alphabetical order. Reading this section first is highly recommended so the other sections can be understood. Many definitions contain acronyms, abbreviations or other terms that are defined elsewhere in the glossary in order to minimize the glossary size, which means that fully understanding a term's definition may require searching out the definitions of other terms.

The second section is a short description and illustration of the current telephone network (the "PSTN") with its advantages and disadvantages.

The third section covers the H.323 protocol view of VoIP networks and the different configurations the H.323 protocol supports. The advantages and disadvantages of an H.323 protocol-based VoIP network are discussed.

The fourth section describes the IETF (Internet Engineering Task Force) and ISC (International Softswitch Consortium) views of a VoIP network and how the different components of their VoIP network model interact. This model's advantages and disadvantages are discussed.

The fifth and last section describes the NACT VoIP-7[®] network model. This discussion uses NACT IPAX[®] gateways with VoIP-7[®] signaling in a VoIP network to illustrate and discuss the advantages and disadvantages of this VoIP network model, including the interconnection of multiple VoIP carrier networks.

Diagrams

In the following discussion, references to figures containing diagrams of various network models, their components, and interconnections are given. Text following the figures contains a discussion of what each figure illustrates.

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Glossary of Terms

100baseT	The 100/10 megabits/second Ethernet standard that uses twisted pair wires with RJ-45 connectors.
Access Tandem	An Access Tandem switch interconnects with CO switches and other access tandem switches to form the PSTN network. It is often called a “class 4” switch, a reference to the original AT&T hierarchal network structure.
Application Server	An application server is an IETF concept. It is most often a computer that implements applications, such as prepaid calling, “class 5” services, automated operator services, etc., and interfaces with a softswitch in order to permit interactions with a caller.
Bandwidth	Bandwidth is a term used to describe the capacity of a physical circuit or piece of equipment to transport or switch information and is often expressed in terms of bits per second. Common abbreviations are Kbits, Mbits, and Gbits for kilobits (thousands of bits), megabits (millions of bits), and gigabits (billions of bits), respectively.
Bearer Circuits	Bearer circuits refer to specific circuits that carry only voice and are never used to carry data messages for call signaling, such as the ISDN or SS7 data links.
CLEC	A Competitive Local Exchange Carrier is a local telephone company available to subscribers within a geographic area that is not the original local telephone company. See ILEC.
CO	Central Office refers to the type of PSTN switch to which telephones and PBX systems are attached. A CO switch implements features such as call waiting, call forwarding, etc. and interconnects with tandem switches. It is often called a “class 5” switch, a reference to the original AT&T hierarchal network structure.
DTMF	Dual-Tone Multi-Frequency is a set of 16 tones used for in-band signaling between PSTN switches and telephones. The tones can also be used for signaling between switches.
G.711	This vocoder is used in the PSTN, and is commonly called PCM (Pulse Code Modulation). There are two flavors of its algorithm; mu-law used in North America and Japan, and A-law used in the rest of the world. This vocoder algorithm produces a bit rate of 64 Kbits/second and has no silence suppression, meaning silence is transmitted and occupies bandwidth.
G.723	This vocoder is commonly used in VoIP gateways and is the vocoder in Microsoft’s NetMeeting software. It is called the Multi-rate Coder and has two bit rates, 5.3 and 6.4 Kbits/second. This vocoder algorithm has silence suppression, meaning silence is not transmitted and does not occupy bandwidth, and is patented, the use for which a license must be paid.
G.726	This vocoder, commonly called ADPCM (Adaptive Differential Pulse Code Modulation), is used in the NACT IPAX gateway for the playback of voice messages such as authorization code and destination number prompts. This vocoder algorithm can produce bit rates of 16, 24, 32, and 40 Kbits/second. The NACT IPAX voice prompts are recorded at 32 Kbits/second. This vocoder has no silence suppression, meaning silence is transmitted and occupies bandwidth.
G.727	This vocoder, commonly called Variable-Rate ADPCM, is available in the NACT IPAX VoIP vocoder suite. This vocoder algorithm allows bit rates of 16-40 Kbits/second for sending and receiving voice. The bit rates can be different in each direction. It has no silence suppression.
G.728	This vocoder, sometimes known by the abbreviation LD-CELP (Low-Delay Code Excited Linear Prediction), is used in some VoIP gateways. It has a bit rate of 16 Kbits/second.
G.729	This vocoder, sometimes known by the abbreviation CS-ACELP (Conjugate Structure Algebraic-Code Excited Linear Prediction), is used in many VoIP gateways. It has a bit rate of 8 Kbits/second. This vocoder algorithm has silence suppression, meaning silence is not transmitted and does not occupy bandwidth. This is a patented vocoder algorithm for which a license fee must be paid.
Gatekeeper	A gatekeeper is an H.323 concept and a device or computer that controls one or more gateways within a “zone”. It is responsible for routing all calls to/from the gateways in a non-point-to-point situation. Gatekeepers can communicate with other gatekeepers, but usually only between those of the same manufacturer.
Gateway1	A gateway in the circuit-switched network is a switch that bridges or interfaces two separate PSTN networks, for example., the mu-law PCM, SS7 signaling-based, North American network with the A-law PCM, C7 signaling-based, European network. Such switches are often called international gateway switches because they interface with and connect the national networks of two or more countries. Examples of such switches are the Nortel DMS 300 and the Lucent 5ESS.
Gateway2	A gateway is a hardware and software system that converts one type of media to another type of media (e.g., G.711 T1/E1 (TDM-based) to G.723 RTP (IP-based)). In general telephony terms it converts voice from one form to another,

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usually from circuit-switched to packet-switched. In the ISC/IETF VoIP model, a gateway is a dumb device that mainly converts the voice media.

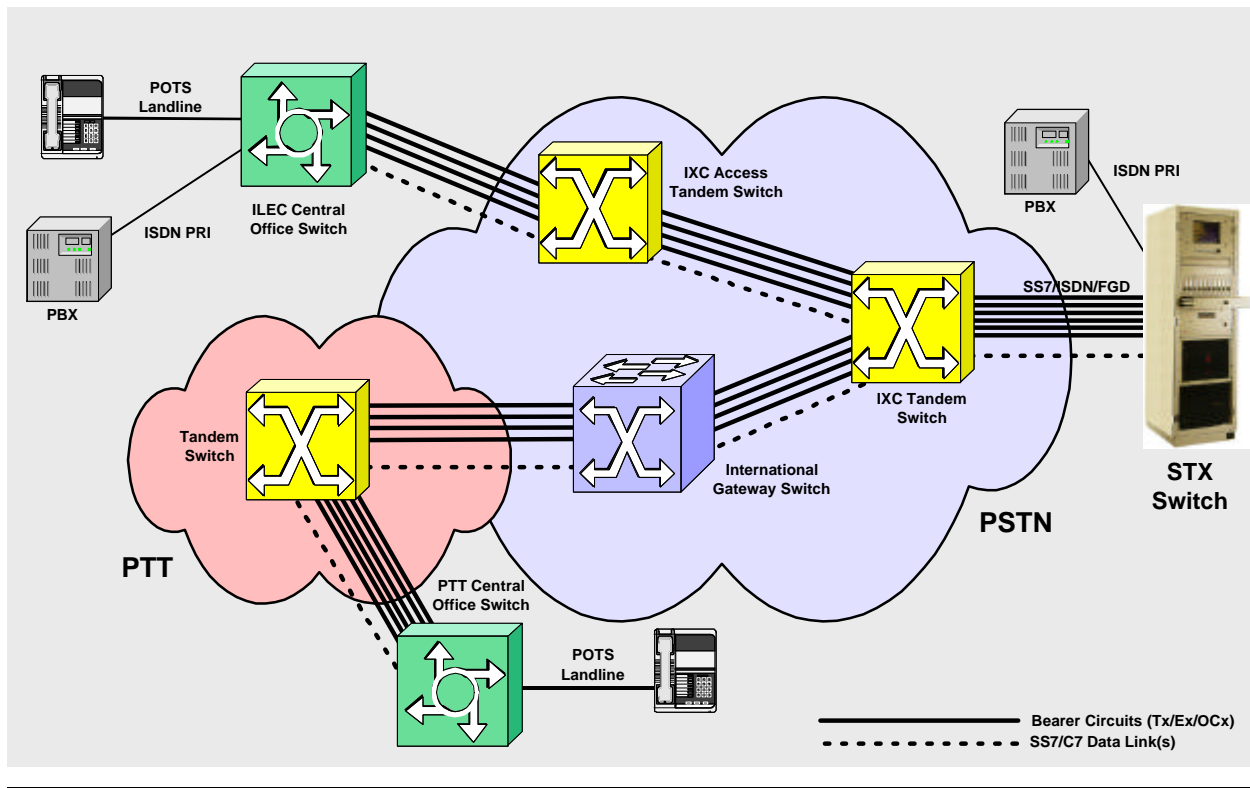
Gateway3	A gateway is an H.323 device that converts one form of media into another form, for example, PSTN voice into VoIP, has a fair amount of intelligence, and can function on its own in a point-to-point situation.
H.248	The ITU-T protocol recommendation for the control of media gateways by media gateway controllers. See Megaco.
H.323	An ITU-T standard for handling video, data and voice call information. This standard was originally proposed for the PictureTel system, which allowed video/voice calls, but has been enlarged to encompass many types of media devices. This “protocol” is actually a umbrella for multiple protocols, each responsible for different items such as packet format and transmission, video codecs, voice codecs, call signaling, conferencing, etc. Version 1 was formally approved in 1996 and the latest, version 4, was approved in November 2000.
Header	The addressing and routing information contained at the beginning of a data packet. An IP packet header contains the IP address of the source and destination systems along with hop counts, packet type, data bytes count, etc. An Ethernet packet header contains the MAC (Media Access Control) addresses of the sending and receiving computers on the Ethernet LAN along with a packet type and a count of the number of data bytes in the packet.
IETF	The Internet Engineering Task Force is the organization responsible for defining standards that affect or use the IP network.
In-band Signaling	The use of tones to convey addressing information (ANI, destination number and information integer digits) in conjunction with changes in the state of the circuit (T1 “A” and “B” bits, E&M E-lead and M-lead, etc.) Tones may be MF or DTMF.
ILEC	An Incumbent Local Exchange Carrier is the local telephone company originally granted a monopoly by the government. See CLEC.
Interoperability	Interoperability refers to the ability of a piece of equipment to interoperate with other equipment from different vendors. For example, some VoIP gateways cannot interoperate with other vendors’ gateways even though they all purpose to use the same signaling and transport protocols. This is often due to incomplete specifications or poor design and implementation. Many gateways do not interoperate because the existing VoIP protocols were more concerned with the basic issue of voice transport and do not consider higher-level issues such as call control and/or applications.
IP	Internet Protocol is the base protocol upon which the Internet packet-based network operates. TCP, UDP, RTP and SCTP all operate on top of IP and use it as their transport protocol. The IP protocol is not a reliable protocol, meaning it does not guarantee delivery and receipt of a packet, which is what makes the public Internet IP network unreliable.
IP Network	An Internet Protocol network is a packet-oriented network designed to transport packets of data between systems. Systems are assigned IP addresses that identify them to the IP network. This network is not designed to reliably transport packets between systems, i.e., it does not guarantee packets transmitted by one system will arrive at the intended recipient system. It is left to higher-level protocols such as TCP to guarantee correct and reliable delivery. IP packets are not related as far as the IP network is concerned.
ISC	The International Softswitch Consortium is an organization of companies and other organizations that work on a set of open standards for voice call control by softswitches. NACT is a member of this organization.
ISDN	Integrated Services Digital Network is a point-to-point signaling protocol designed to interface PBX equipment with central office switches. It uses out-of-band signaling on a data link that is carried with the bearer circuits. A U.S. based ISDN PRI circuit is sometimes referred to as a 23B+D circuit, which means it is a T1 circuit that has 23 Bearer circuits (DS0 channels) used for voice transport plus one Data circuit used to transport the ISDN signaling protocol message data packets.
ISUP	ISDN User Part is the network signaling protocol that sets up and tears down calls in the PSTN network using messages transmitted between switches over the SS7 data network. The ISUP protocol is a connection-oriented or circuit-oriented protocol, meaning that ISUP messages between systems are about connections or circuits between them. ISUP messages do not stand on their own, meaning multiple messages will be sent and received concerning the same connection or circuit in a specific order, each time the circuit is used for a call. ISUP messages are transported over the SS7 network by the MTP3 or M3UA transport protocol layers.
ITU	International Telecommunications Union—the international standards organization for all types of radio and telephony telecommunications.
ITU-T	International Telecommunications Union—Telecommunications sector. The part of the ITU responsible for both circuit-switched and packet-switched telecommunications.
IXC	An Interexchange Carrier is normally a long distance telephone company in the United States. It interconnects ILEC, CLEC and other IXC switches to form the PSTN.

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LAN	A Local Area Network is a network within a small geographic area to which various devices such as computers and printers are attached. Examples of LAN technologies are Ethernet, Arcnet, and IBM Token-ring.
M3UA	M3UA is the MTP3 User Adaptation Layer protocol that adapts SS7 MTP3 messages for transport over an IP network using the SCTP protocol. This is a very recent protocol adopted in September 2000. This protocol is aware of SS7 point codes and circuits. It translates point codes into IP addresses for SS7 message transport across the IP network.
Media Gateway	A media gateway is an IETF and MGCP concept. It is a device that converts one form of media into another form, for example, PSTN voice into VoIP, and has very little intelligence.
Media Gateway Controller	A media gateway is an IETF and MGCP protocol concept. It is a device that controls one or more media gateways and has enough intelligence to be able to control nearly all aspects of each media gateway, but does not have enough intelligence to control an entire call.
Megaco	Megaco is the IETF name for the ITU-T H.248 protocol standard recommendation for controlling a media gateway when connecting telephone calls between a LAN and the PSTN. It handles many telephony issues such as redundant MGC systems that MGCP does not address. This recommendation was announced in August 2000. The Megaco protocol is similar to the MGCP protocol but is not backward-compatible with it, meaning a system using Megaco will not interoperate with a system using MGCP.
MGCP	Media Gateway Control Protocol is a protocol designed to control various devices that can handle different types of media, e.g., voice, video, data, etc. It is designed to allow a Media Gateway Controller (MGC) to control one or more unintelligent Media Gateway (MG) devices.
MTP2	MTP2 is the Message Transfer Part [layer] 2 protocol that transmits and guarantees receipt of a SS7 message between two adjacent systems in the SS7 data network using the traditional SS7 A-links or F-links.
MTP3	MTP3 is the Message Transfer Part [layer] 3 protocol that routes SS7 messages given to it over the SS7 data network. It is aware of SS7 point codes and how to route messages to them. When it has selected a route to a particular point code, it uses the MTP2 protocol to transmit the message to the next system in the SS7 data network.
Packet-switched	A technique for routing data through a network by encapsulating the data in packets, labeling them with addresses and routing information (the packet header), and then routing the packets over various physical circuits (oftentimes different for each packet) until they reach their intended destination. Packets from the same source are not considered related to each other and thus each packet must make its own way through the network. A packet may not reach its intended destination due to network congestion and can be discarded. This network characteristic cannot guarantee quality of service, the lack of which the public IP network known as the Internet is famous.
PBX	A Private Branch Exchange is a very small, specialized switch similar to a CO switch. It permits the attached telephones to call each other using shorter numbers, and requires the caller to select an "outside line" in order to call through the PSTN.
PCM	Pulse Code Modulation is the name of the G.711 voice codec algorithm that is used to convert analog voice into digital data. It converts analog voice into 8,000 8-bit data samples every second. It has two implementations; A-law used throughout most of the world and mu-law used throughout North America and Japan.
POTS	Plain Old Telephone Service is the term applied to the normal telephone service delivered by a PSTN CO switch.
PRI	Primary Rate Interface. This term is always used in connection with ISDN, as in "ISDN PRI". It refers to a digital circuit, such as a T1 or E1, that carries multiple calls and uses the ISDN signaling protocol.
PSTN	The Public Switched Telephone Network is the public, wire-line, switched network based on TDM circuits and used to transport PCM-encoded voice calls. It is often times referred to as the SCN, or Switched Circuit Network, by those in the packet-switched data network world.
PTT	The PTT network is a term used to refer to the PSTN network of other countries. It comes from the abbreviation for Post, Telephone and Telegraph, which used to be the government-owned company within other countries that owned and managed the national postal, telephone and telegraph systems.
Router	A router is a hardware and software system that directs data packets entering the router toward their intended destinations. Routers have various physical interfaces such as 100baseT Ethernet, T1, E1, DS3, OC-3, etc. Routing is based upon the destination address contained within the header of the IP data packet.
RTCP	The Real-time Transport Control Protocol is used to report on the performance of a particular RTP transport session. It delivers information such as the number of packets transmitted and received, the round-trip delay, jitter delay, etc. that are used to measure Quality of Service in the IP network.
RTP	The Real-time Transport Protocol is a protocol that is used to transport real-time data, such as voice or video. This is an "unreliable" protocol built on top of the UDP protocol that does not guarantee delivery of packets, but which has little overhead.

SCCP	SCCP is the Signaling Connection Control Part protocol used to deliver transaction messages that do not perform call setup and teardown, which is the responsibility of the ISUP protocol. It is connection-less, meaning each message stands on its own and does not depend on any messages sent before or after it.
SCTP	SCTP is the Stream Control Transmission Protocol created for transmitting SS7 messages over the IP network. Since the IP network does not guarantee the delivery of messages through the network, nor provides for redundant physical paths through the network, the SCTP protocol performs these functions.
Signaling Gateway	A signaling gateway (SG) is an IETF and ISC concept. It is a device or computer that interfaces a softswitch with a signaling system such as SS7.
SIP	Session Initiation Protocol is a protocol designed to allow personal computers to place telephone calls on the Internet. It is often used by devices on the Internet and is currently used for telephone calls, most often "SIP-enabled" telephones. It is proposed as a protocol for gateways, and especially for softswitches. Currently there are gateways that interface with SIP-enabled telephones, but the SIP protocol really is not well-suited for control of a gateway by a softswitch, which currently is a source of raucous debate within the VoIP community.
Softswitch	A softswitch is an IETF and ISC concept and a device or computer that controls the setup and teardown of calls in media gateways. A softswitch has no network interface cards and call control software that tracks and controls calls in other computers such as media gateways or media gateway controllers.
SS7	Signaling System 7 is the international data network and signaling protocols that control the PSTN voice circuits and calls. These protocols have country-by-country variations. ITU SS7 is the base on which the national variants are based. ANSI SS7 is the North American SS7 variant.
SS7 Network	The Signaling System 7 network is a data network that transmits messages between switches and other computers used in the PSTN. More than one message-based protocol can use the SS7 data network for transport of messages between switches and computers. Examples of message-based protocols that use the SS7 data network are ISUP and TCAP. SS7 messages are typically transmitted over redundant data links, often over dual point-to-point telephone lines using modems (A-link) or through a DS0 channel on two T1 or E1 digital spans (F-link).
SS7 Point Code	The SS7 data network uses "point codes" to identify switches and computers connected to the SS7 data network.
Subscribers	The customers of a telephone company that are signed up for service.
Tandem Switch	A tandem switch interconnects with other PSTN switches. It is often called a "class 4" switch, a reference to the original AT&T hierarchal network structure.
TCAP	Transaction Capabilities is a message-based protocol that is used to perform non-call services, such as Local Number Portability database lookups, portable 800-number translation, etc. It transports its messages over the SCCP connectionless-oriented protocol.
TCP	The Transmission Control Protocol is the IP protocol that guarantees delivery of a data packet from the sender to the receiver. This is a "reliable" protocol that guarantees delivery and order of packets, but which has a lot of overhead and can take a long time guaranteeing packet transmission. It is the protocol used on the public Internet with web browser software. It is highly unsuitable for the transport of real-time data such as voice and video.
TDM	Time Division Multiplexing is the technique by which circuit-switched networks (PSTN) carry multiple voice calls over a single physical circuit. This is accomplished by transmitting individual PCM samples from multiple calls in a serial fashion 8,000 times per second for 24 calls on a T1 circuit and for 30 or 31 calls on an E1 circuit.
UDP	The User Datagram Protocol is the IP protocol that delivers data in the same manner in which it was sent (i.e., if the sender transmits 20 bytes in a packet, they are delivered to the receiver as 20 bytes together). This is an "unreliable" protocol that does not guarantee delivery or order of delivered packets, but which has little overhead.
Vocoder	Vocoder is an acronym for voice coder/decoder. A synonym is software codec. This is the term given to the process of encoding and decoding voice using an algorithm implemented in software and/or hardware. The process is often referred to as voice compression and decompression. Common vocoder algorithms are G.711 (PCM), G.723.1 (used by Microsoft's NetMeeting), G.726 (ADPCM, used by the NACT STX switch/IPAX gateway for voice message playback), G.728 and G.729. This function is usually performed by hardware or DSP software.
VoIP	Voice over the IP network is a general term that refers to any means of converting voice calls into voice data packets that are transmitted over an IP network, either public or private.
WAN	A Wide Area Network is a network that covers a large geographic area and interconnects many devices such as computers, telephones, etc. A WAN often cannot use the same technologies as a LAN due to electrical or other physical limitations. The two most well-known circuit-switched and packet-switched WANs are the PSTN and Internet, respectively.
WAN Interface	A WAN interface is the hardware and software that converts data packets on a LAN to data packets on a WAN. This is most often found in routers that convert from Ethernet LAN technology to T1/E1/OC3/OC12 technology.

Figure 1 – PSTN Circuit-switched Model



The existing PSTN model shown in Figure 1 is composed of CO switches (green box) owned by ILECs to which telephones and PBX systems (gray) are attached. These CO switches are interconnected via Access Tandem switches (yellow box), either owned by the ILECs or by IXC's. These switch together make up the PSTN (blue cloud). International calls are routed through the tandem switches (green) to international gateway switches (blue box) that interface with tandem switches (green box) in other countries. The PSTN network in other countries is often referred to as the PTT network (rust cloud). The connections (black lines) between the switches are TDM circuits such as T1, E1, T3, E3, OC-3, OC-12, etc.

Note that a cloud is often used to represent the many interconnected switches that make up a network when illustrating the connection(s) between a switch and the PSTN.

Call setup and teardown signaling is performed by SS7, ISDN or in-band protocols such as Feature Group D. SS7 and ISDN are out-of-band signaling protocols that use a data link to pass the messages controlling the setup and teardown of voice calls on the bearer circuits, while the FG-D protocol uses inband MF tones within the bearer circuits to perform the call setup and teardown, which is much slower than SS7 or ISDN. Voice is carried over bearer circuits to passing voice full-time regardless of whether or not the line is silent.

Advantages of the PSTN network are the voice quality (G.711 PCM) and its low-delay delivery, which is typically less than one or two milliseconds.

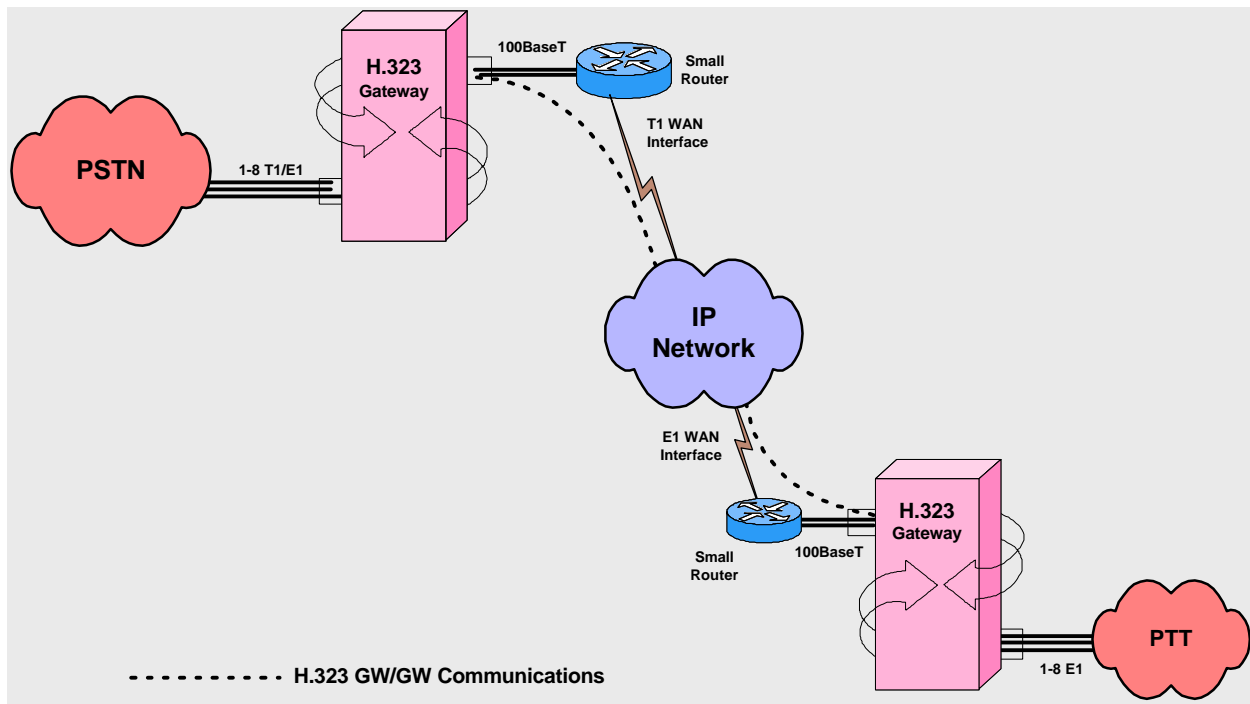
The disadvantages are the large quantity of circuits and equipment required to carry voice calls, the complexity of most switch manufacturer's equipment, the cost of powering the equipment, and the difficulty of new feature implementation in most switches.

Billing is often difficult due to the large quantities of call event records that often must be matched to create call records, and the paucity of information supplied by the switches.

Interconnection with other carriers' networks require large numbers of circuits and equipment and is prone to human error during configuration and maintenance.

The NACT STX switch has typically been used as an Access Tandem switch (yellow box) in support of wholesale and retail telephony products, such as prepaid cards, payphones, automated operator, etc.

Figure 2 – H.323 Gateways in a Point-to-Point Configuration



The H.323 network model allows multiple configurations. The configuration illustrated in Figure 2 shows a peer-to-peer relationship between two standalone H.323 gateways (purple box) connected via the IP network (blue cloud). They are self-sufficient and can be devices such as a gateway, personal computer, or H.323-enabled telephone. The figure shows gateways in an international scenario that illustrates connections to PSTN and PTT networks (red cloud).

The typical H.323 gateway handles a single DS0 (personal computer or H.323-enabled telephone) or one, two, four, eight or twelve T1/E1 circuits. Examples of the latter type of H.323 gateway are those produced by Nuera and Clarent.

Advantages of the H.323 protocol are few. It does permit standalone gateway operation, illustrated in the figure above. It was the first VoIP protocol that had general acceptance. It was formally approved in 1996 and, even though but five years old, is already labeled as a “legacy” VoIP protocol. Few manufacturers are using it as the sole protocol of their gateway products. It is, however, in quite a few VoIP installations and is the largest percentage of installed VoIP systems today. It has a fairly good interoperability record at the voice transport level. The H.323 specification requires use of the G.711, G.723 and G.729 vocoders under certain conditions, but the availability of multiple vocoders allow vocoder choices that can be tailored to WAN bandwidth capacities.

General disadvantages with the H.323 protocol itself include slow call setup and teardown, large software size and complexity, and software implementation difficulty. Later versions (especially versions 2 and 3) of the protocol specification have improved the call setup and teardown times, reducing them to 1-3 seconds.

The specific disadvantage in peer-to-peer H.323 gateways is that they are essentially point-to-point systems that can interconnect two sets of PSTN T1/E1 circuits over an IP network and do not perform switching, thus dedicating their entire resources to a single use, whether utilized or not.

The H.323 specification makes no mention of billing. Billing may or may not be an issue in an H.323 point-to-point configuration. Enough information should be produced by the gateways that billing should be possible. The issue will be one of implementation by their manufacturer.

Figure 3 – H.323 Gatekeeper/Gateways in a Point-to-Multipoint Configuration

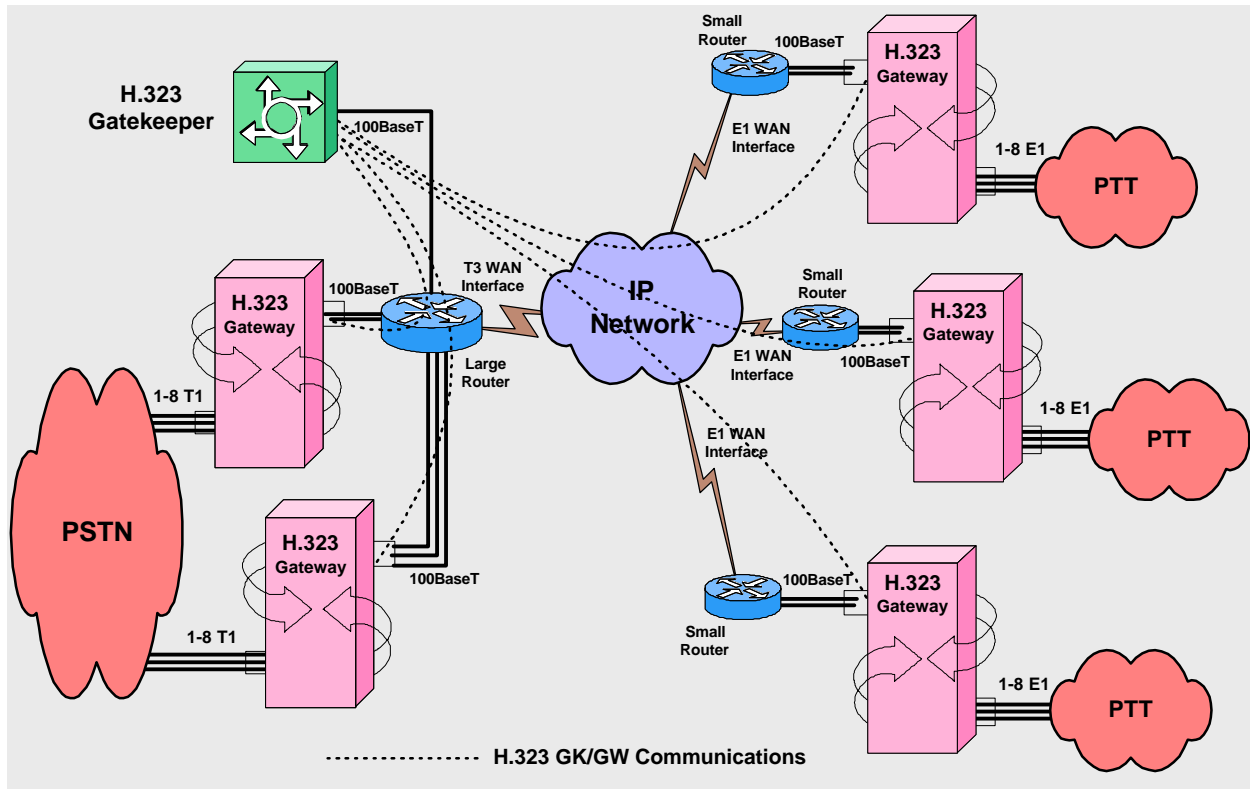


Figure 3 illustrates multiple H.323 gateways under the management of a single H.323 gatekeeper (green box). The H.323 gatekeeper is responsible for registering and tracking H.323 gateways and authenticating and routing each call handled by each gateway. The H.323 protocol defines the messages used between the gatekeeper and the gateways in a point-to-multipoint configuration.

An H.323 gatekeeper simplifies the management of multiple H.323 gateways by allowing the central administration of routing and registration information. The gatekeeper defines one or more “zones” within which are located one or more gateways. When a call enters a gateway from the PSTN network, the gateway informs the gatekeeper, which then decides the gateway that should terminate the call. The gatekeeper routes the call by consulting routing tables, queries the determined terminating gateway for resource availability, and then tells the originating gateway through which gateway it will terminate the call. The originating gateway then sets up the call with the terminating gateway.

Advantages of the gatekeeper concept are that it can control multiple gateways, and management of routing and some gateway control is centralized in one place.

One of the major disadvantages of the gatekeeper concept is that it cannot scale up to the global, or even a national, level; that is, the centralized gatekeeper can control only so many gateways before it becomes the bottleneck in the network. Gateway call setup and teardown is slower because it must go through the gatekeeper in order for calls to be routed to their terminating gateways.

Billing is almost always an issue because it has not been addressed by the gatekeeper and gateway manufacturers or they have a poor implementation. Often the billing information is derived from multiple “event” records produced by each piece of equipment that participates in a call, thus requiring the collection and matching of records from the gatekeeper and the originating and terminating gateways to create a single, complete billing record. This is oftentimes a difficult task that does not produce 100% results (i.e., not all calls will have billing records) due to problems such as time synchronization, lack of knowledge of network topology and interconnections, insufficient information to identify individual calls, etc.

Clarent has an H.323 gatekeeper product that appears to fulfill the H.323 protocol specification, but commonly uses the Clarent Command Center as the “brains” (call control) of their product offering since the gatekeeper, based upon the H.323 specification, cannot do enough to handle calls that are some form of telephony application.

Figure 4 – Multiple H.323 Gatekeepers in a Multipoint-to-Multipoint Configuration

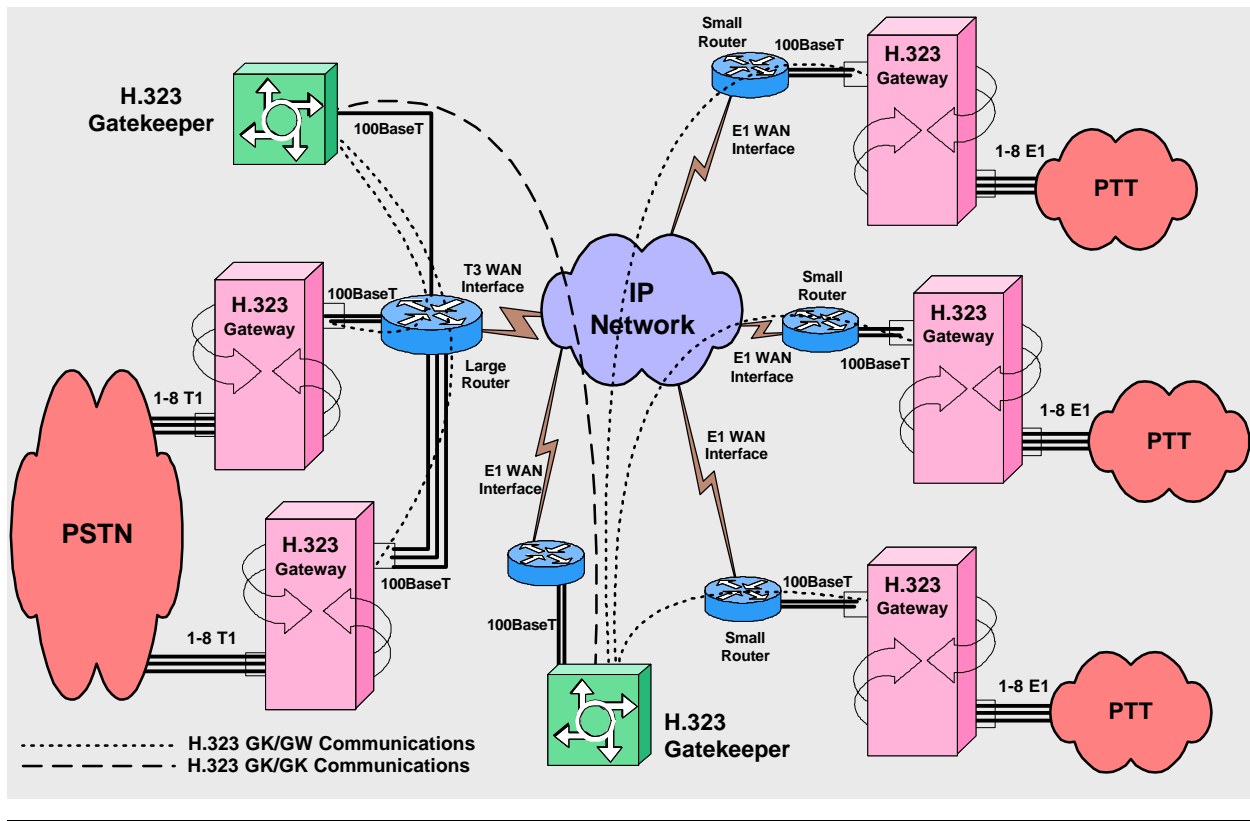


Figure 4 illustrates multiple H.323 gateways under the management of multiple H.323 gatekeepers (green boxes). This is a multipoint-to-multipoint configuration. Such a configuration may be implemented because a single gatekeeper cannot handle the number of gateways in a network, and thus more than one gatekeeper is required. This configuration also will exist when different VoIP carriers have their own gateways and gatekeepers.

The H.323 protocol defines the messages between the gatekeeper and the gateways, but does not really define many messages or interactions between gatekeepers. This means that two H.323 carriers, each with their own gatekeeper(s), cannot easily pass a call between them. If this is currently done, it is usually done only between gatekeepers from the same manufacturer, which is one of the reasons why H.323 inter-operability is difficult and usually not available today, or in the case of different manufacturers, the gateways of one carrier are placed under the control of another carrier's gatekeeper, which is most undesirable due to fraud and mismanagement potential.

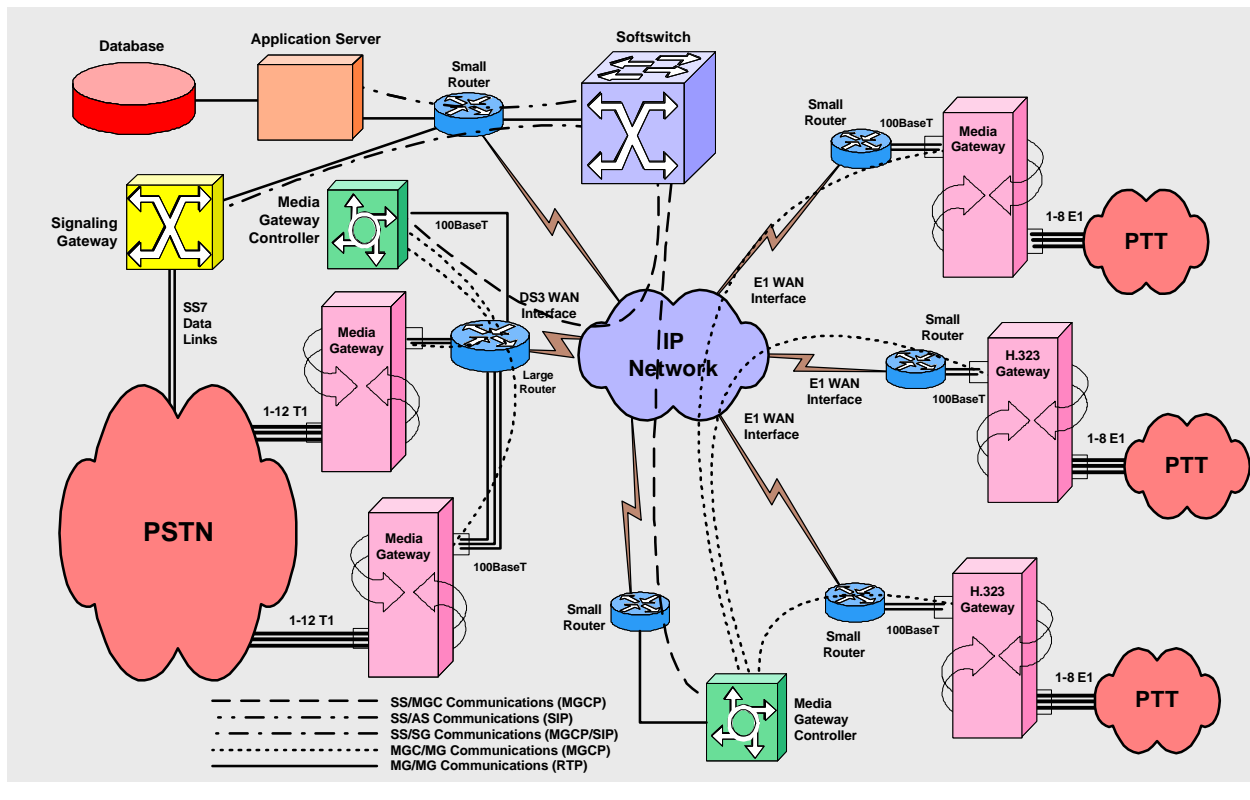
An advantage of this network model is that upward scalability is possible, although the H.323 protocol specification really does not handle this situation well, and national and global networks really are still not possible. Theoretically interoperability among multiple H.323 VoIP carriers is possible, although in practice it has not fulfilled its promise.

The major disadvantage of this network model is that a gatekeeper in one carrier must control the gateway in another carrier if they are to pass calls between them. This means that one carrier would have to cede control of its network to another carrier for the setup and teardown of a call between them. No carrier wants to do this, and thus interoperability between carriers is not usually done. If it is, it is performed by a carrier, such as GRIC, that also performs mediation of the calls between carriers and functions as a clearinghouse between them in order to reconcile the billing.

Billing becomes a much more serious problem in this model because the gateways and gatekeepers usually contribute various "event" records that must be consolidated into a single call record. If the gateways and gatekeepers are owned by different carriers, they have to exchange and match the event records, which becomes a serious problem that is difficult and sometimes impossible to resolve.

Clarent has a "reconciliation" product that attempts to reconcile the various event records from multiple gateways and gatekeepers because their existing gatekeeper and gateway software cannot record and reconcile "event" records when more than two gatekeepers are involved.

Figure 5 – ISC/IETF Network Model



The International Softswitch Consortium (ISC) and the Internet Engineering Task Force (IETF) both propose the VoIP network model illustrated in Figure 5. This network model assigns what are essentially “servers” to handle different responsibilities in the VoIP network. The illustration depicts a Softswitch (SS) (blue box) controlling the call setup and teardown of VoIP calls. It is the intelligent system that “knows” about call control. It communicates with and uses a Signaling Gateway (SG) (yellow box) to interface with the PSTN SS7 network, and controls the Media Gateways (MG) (purple boxes) through the Media Gateway Controllers (MGC) (green boxes). Applications are handled by Application Servers (orange box) that interface with the softswitch and that use database servers (red cylinder) to store information.

This network model is the ultimate in function decomposition, with each function handled by a separate system. Since this network model came from the organization responsible for the Internet, it has a client/server architecture. The softswitch is the “server” and uses the MGCP or Megaco/H.248 protocol to control the media gateway controller systems, which are its clients, and they function as “servers” using MGCP or Megaco/H.248 to control the media gateway systems, which are their clients. The softswitch uses MGCP to communicate with the signaling gateway, which may function as a “server” and/or a “client”, and uses the SIP protocol to communicate with the application server, which may function as a “server” and/or a “client”.

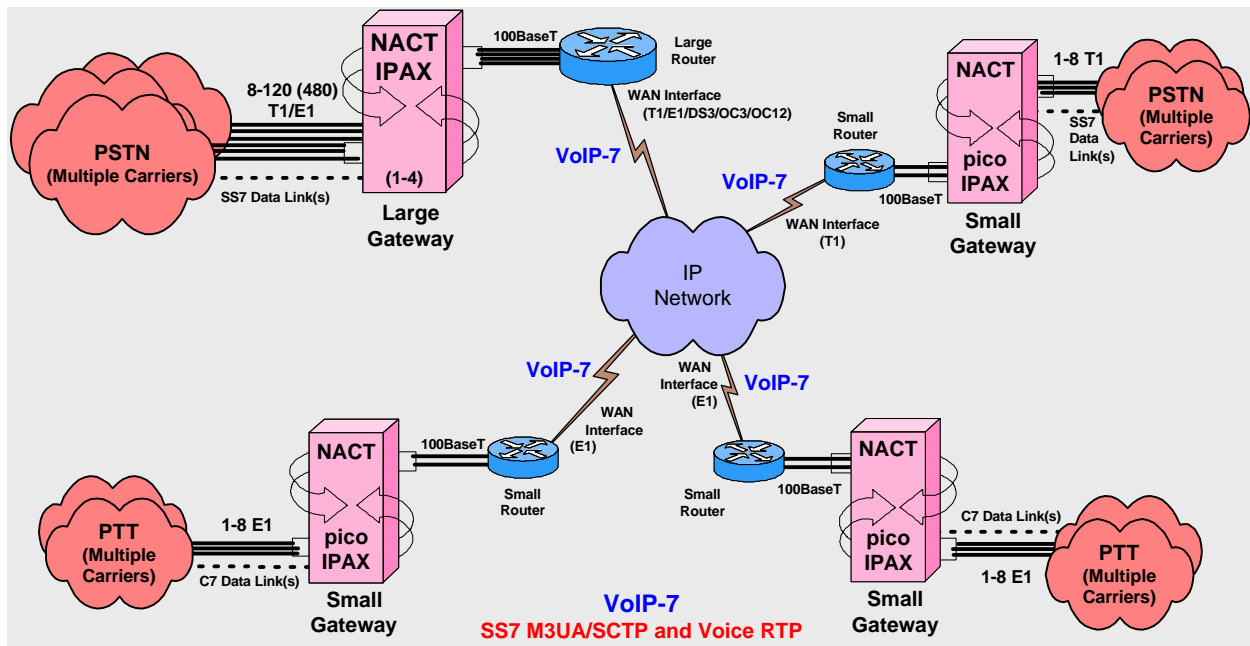
This approach draws fine distinctions between the various components of a network, but there are currently no standard protocols for the application server and softswitch to communicate, nor for softswitch-to-softswitch communication. SIP has been proposed for these communications, and some early softswitch systems are using it. The network model also does not define any protocol of any kind for the Application Server relative to applications themselves.

This network model suffers from the same communication and scalability problems of H.323 gatekeepers. A softswitch can handle only so many calls, MGCs and MGs before the messages passing back and forth become the bottleneck and slow the entire network down. The model also suffers from the same carrier interconnection problems of the H.323 multiple gatekeeper model. It may support end-to-end SS7 signaling, but only if the softswitches support it and use it end-to-end (i.e., no H.323, MGCP, H.248/Megaco, or SIP protocol is used exclusively between systems), which no systems do.

Billing can be a larger issue than in any other network model if any of the VoIP protocols are exclusively used at any point in the call path, since end-to-end SS7 is not present, and the softswitch does not handle the creation of call records well. It will most likely be a very large issue because it will be highly unlikely that all components will be produced by the same manufacturer, and billing information is usually an afterthought in the creation of these types of network components. Since there are no standard protocols for the higher level functions, such as SS-to-SS or SS-to-AS communication, billing will become difficult to do since there will be many “event” records produced by the various pieces of equipment.

This model is being promoted by the IETF and ISC but actually does not yet fully exist. Some companies are trying to produce various products of this network model but will most certainly be hampered by the lack of standards when it comes to interoperability of applications.

Figure 6 – NACT VoIP-7 Multipoint-to-Multipoint Network Model



The NACT network model illustrated in Figure 6 closely parallels the current PSTN model in that the VoIP gateways are intelligent, self-sufficient, and interconnected end-to-end via SS7. In this model the gateways (labeled NACT IPAX) communicate SS7 messages through the IP network using the M3UA and SCTP protocols. The SS7 messages have been enhanced with the VoIP information needed to set up the VoIP portion of the call through the IP network. NACT has named this VoIP information-enhanced SS7 protocol VoIP-7, which in reality is still standard SS7. The VoIP-7 software simply uses the “access transport” field of certain select SS7 messages to pass the VoIP information, which is what the access transport field is for, passing information between adjacent systems.

This network model teams the strengths of the PSTN model with the strengths of the IP network. The strengths of the PSTN model are signaling integrity and scalability. The PSTN model can scale upward to truly national and global networks and permits carriers to retain control of their networks because they control their signaling interfaces with other carriers. The strength of the IP network is the transport of voice directly from originating gateway to terminating gateway without the maintenance of many physical circuits and their corresponding expense.

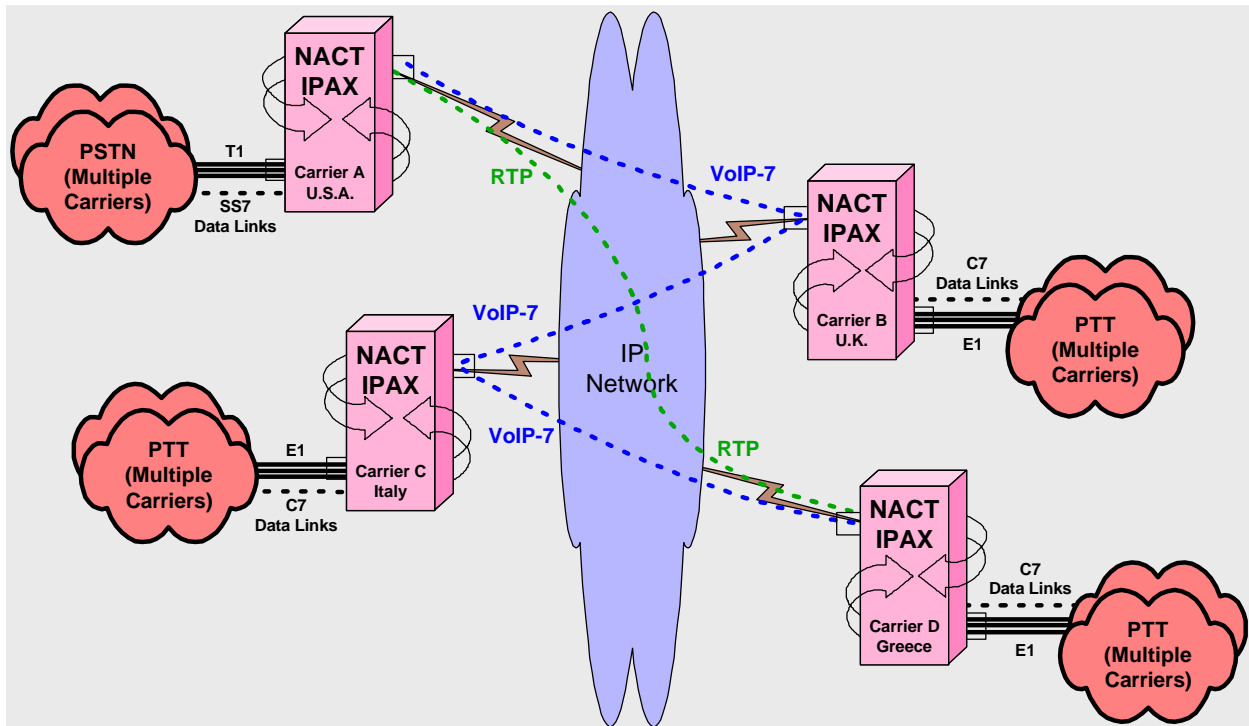
A VoIP-7 network model gateway is intelligent, can stand alone, and performs all of the functions that appear in the ISC/IETF model, albeit almost entirely within the gateway. The gateway software actually follows the function decomposition of the ISC/IETF model, but the messages between the software components never leave the gateway computer and are communicated entirely between the software components, which means the messages will not become a bottleneck since they are passed and processed at internal speeds.

The VoIP-7 model permits end-to-end SS7 signaling naturally because it is the protocol exclusively used to set up and tear down the VoIP portion of the call as well as the PSTN portions. This model solves the carrier-to-carrier interaction problem because no carrier has to cede control of its gateways to another carrier. A carrier remains in control of its gateways at all times and communicates with neighboring carriers through SS7. Since SS7 generates the events and information that cause the creation of the billing information, the billing problems the current VoIP protocols have are not an issue because each gateway can generate its own, full call record (i.e., no “event” records must be collected from multiple systems and reconciled).

A disadvantage of this network model is that each gateway must be fully configured with the information needed to perform all signaling functions as well as all application functions. This will require a large amount of configuration management, which can, however, be performed by a centralized management that disseminates the information to the gateways where it is used. The centralized management system mitigates this disadvantage and changes it into an advantage on par with that of the ISC/IETF model.

The following two scenarios illustrate the strengths of the VoIP-7 network model.

Figure 7 – NACT VoIP-7 Carrier Signaling Interconnection and IP Network Transport



The VoIP-7 network model illustrated in Figure 7 shows the synergy derived from the strengths of the PSTN model and the technology of the IP network.

Each of the gateways pictured above is intelligent and can function in a standalone fashion. Management of the configuration information for each gateway can be performed at a central location, but the information is disseminated to each gateway so they do not have to query the central database for information on how to handle each call.

In this example, four carriers, A, B, C, and D, have gateways in the U.S.A., the United Kingdom, Italy and Greece, respectively. However, carrier A has an interconnection agreement only with carrier B to provide termination services to Europe. Carrier B has an interconnection agreement with carrier C to provide termination services to the European continent, and carrier C has an interconnection agreement with carrier D to provide termination services to Greece.

For a call placed to Greece through carrier A, the U.S.A. gateway sends a VoIP-7 (SS7 message enhanced with VoIP information) message to the U.K. gateway, which routes it to the Italy gateway, which in turn routes the message to the Greece gateway, which will terminate the call to a Greek circuit-switched network. The U.K. and Italy gateways pass on the original U.S.A. gateway VoIP information contained in the VoIP-7 message sent by the U.S.A. gateway since they are not terminating the call to a circuit-switched network. Since the Greek gateway will terminate the call to a circuit-switched network, it passes its VoIP information back to the Italy gateway in a VoIP-7 message, which the Italy gateway passes back to the U.K. gateway, which passes it on to the U.S.A. gateway. At this point, the U.S.A. and Greece gateways start their voice RTP packet flows directly to each other through the IP network using the VoIP information passed in the two VoIP-7 messages.

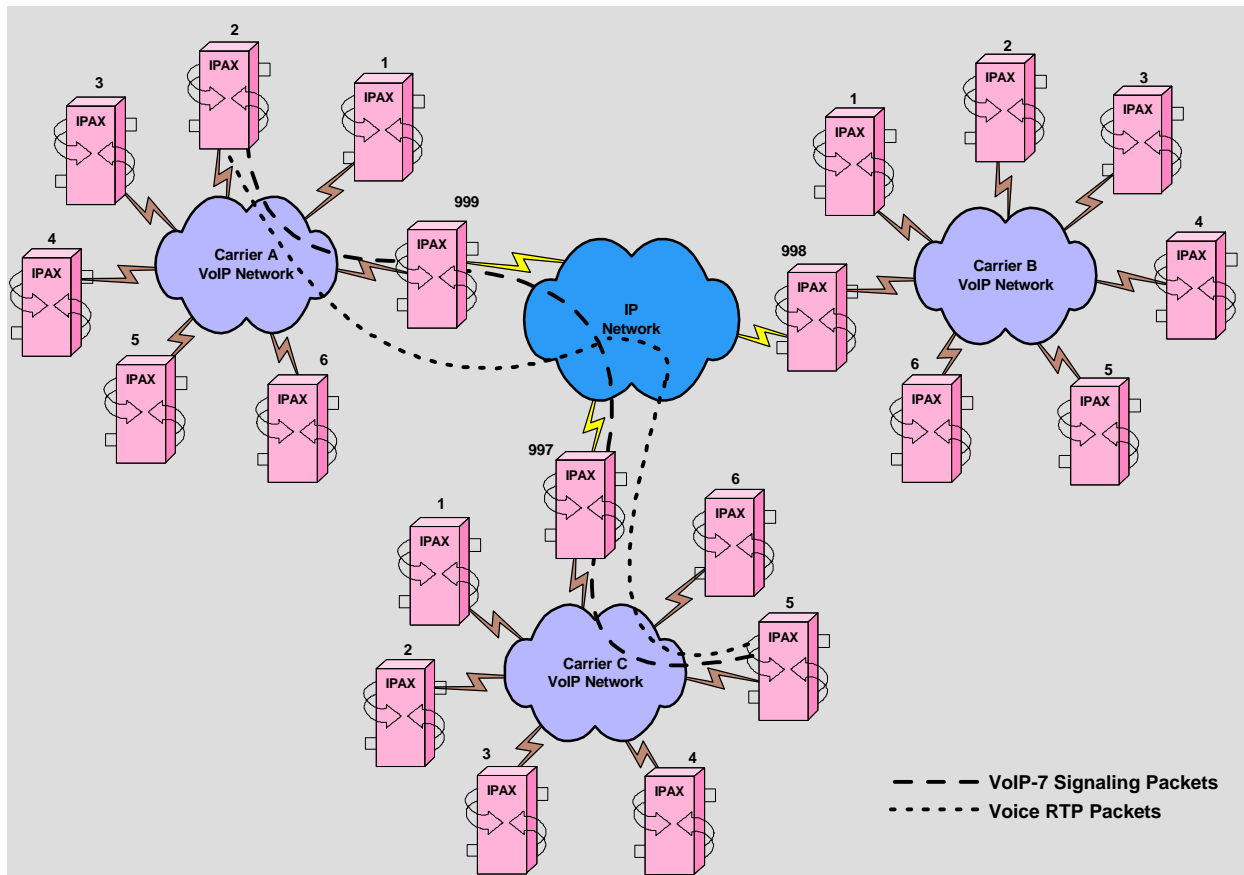
This scenario takes advantage of a strength of the IP network, which is the voice packets can travel directly to their intended destination without going through intervening gateways, and simultaneously uses an advantage of the PSTN network, which is the VoIP-7 signaling messages travel through the gateways that parallel the interconnection agreements between the carriers.

Since these messages are passed through intervening gateways, each gateway can use the information in the messages to construct an accurate billing record reflecting its involvement in the setup and teardown of the call, thus fulfilling one of the purposes of an interconnection agreement. Each gateway has a complete call record that can be used to bill termination services back to the originating carrier and to audit services billed by the terminating carrier. Billing between the four carriers is greatly simplified since they only have to deal with the carriers for which there are interconnection agreements, which in this example is no more than two carriers.

In the H.323 or ISC/IETF scenarios, all carriers would have to bill all other carriers, which quickly grows in a so-called N^2 problem that would become prohibitively expensive and complicated to resolve. The current VoIP industry is trying to resolve this problem through the use of mediation servers and services or clearinghouses, whose sole job is to reconcile the various charges between all carriers on a per-call basis. This service simply adds cost and complexity to a VoIP-based carrier.

Another strength of the VoIP-7 model is that each carrier remains in control of its own gateways, which control is not relinquished in order to pass a call through its network. This is not the case for the current VoIP network implementations. If two carriers desire to interconnect their networks and permit their gateways to handle calls that originate or terminate in the other carrier, one carrier must allow the other carrier to control its gateways. This situation is abhorrent to the carriers because they do not want to cede control of their network equipment to another carrier. The other carrier could defraud them, favor its own call traffic to the detriment of their call traffic, etc. The VoIP-7 model prevents this control imbroglio and allows a carrier to remain in control of its facilities at all times.

Figure 8 – NACT VoIP-7 Carrier Network Signaling Portals



To illustrate the “arm’s length” at which the VoIP carriers can separate themselves so as to retain control of their own network, but simultaneously use the strength of the IP network to carry the voice call, consider the following interconnection example illustrated in Figure 8.

The diagram depicts three carrier networks, A, B and C, each of which have seven VoIP gateways. Each carrier network has six gateways numbered 1-6. The carriers, A, B and C, each have a seventh gateway, numbered 999, 998 and 997, respectively, that are used as the signaling interconnection point, or “signaling portal”, between their VoIP networks.

VoIP gateway 999 is visible to all gateways in the local VoIP network of carrier A, but it is also visible to the interconnection gateways or signaling portals of carriers B (998) and C (997) because they contain a definition for gateway 999. The other gateways within the local VoIP networks of carriers B and C do not have a definition for gateway 999. To terminate calls in the network of carrier A, they must send their SS7 signaling messages to their own interconnection gateway or signaling portal, which then passes the message on to gateway 999.

In this manner gateway 999 is the signaling interconnection point or portal for all calls originated by gateways in the networks of carriers B and C and terminated by carrier A. This means the seven gateways in the network of carrier B shall send their SS7 messages to gateway 998, which passes them through to gateway 999, for all calls that terminate in the network of carrier A. Likewise the gateways in carrier C’s network send their SS7 messages to gateway 997, which passes them through to gateway 999, for all calls that terminate in carrier A’s network. Carrier A’s gateway 999 passes the SS7 messages on to the other gateways in its own VoIP network where the calls are terminated.

Using this method, all gateways can terminate voice calls into all other gateways, but only have to be aware of their own single “signaling portal” gateway through which all SS7 messages flow. This means the six local network gateways in any of the three VoIP networks above only have



NACT VoIP-7 Network Model

definitions for seven (7) gateways, those of their own network, while the seventh gateway or signaling portal in each network has definitions for nine (9), its own seven plus the interconnection gateways or signaling portals of the other two networks. This is far better than the 21 gateway definitions that would be necessary if all gateways had to be visible to all other gateways. This scheme permits the interconnection of multiple VoIP carrier networks and allows the carriers to retain control of their own gateways and monitor and control the voice calls they accept from other networks.