VoIP Glossary

VoIP Terms and Definitions

Let's Connect!™

07VOIPGLOSS-REF2

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Introduction—What is VoIP?

Voice over IP or *VoIP* is a term used in IP telephony for a set of facilities that use the Internet Protocol (IP) to deliver voice information. In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that they avoid the tolls charged by ordinary telephone service.

The term *VoIP* derives from the VoIP Forum, an effort by major equipment providers, to promote the use of ITU-T H.323, the standard for sending voice (audio) and video using IP on the public Internet and within an intranet. The Forum also promotes the user of directory service standards so that users can locate other users and the use of touch-tone signals for automatic call distribution and voice mail.

In addition to IP, VoIP uses the real-time protocol (RTP) to ensure that packets get delivered in a timely way. Because the nature of public networks such as the Internet makes it difficult to guarantee Quality of Service (QoS), better service is usually possible with private networks managed by an enterprise or by an Internet telephony service provider (ITSP).

A technique used by at least one equipment manufacturer, Adir Technologies (formerly Netspeak), to help ensure faster packet delivery is to use the ping utility to contact all possible network gateway computers that have access to the public network and choose the fastest path before establishing a Transmission Control Protocol (TCP) sockets connection with the other end.

Enterprises use VoIP gateways to enter into the VoIP environment. A gateway receives packetized voice transmissions from users within the company and then routes them to other parts of the company's intranet (local area or wide area network) or—using a T-carrier system or E-carrier interface—sends them over the public switched telephone network.

VoIP Standards

- ITU-T H.320 Standards for Video Conferencing
- H.323 ITU Standards
- H.324 ITU Standards
- VPIM Technical Specification

VoIP Glossary

A-law

A-law is an audio codec companding format in which the signal is compressed on input and expanded back to its original form on output. A logarithmic mapping between 8-bit data space and 13-bit sample space as described in the CCITT G.711 recommendation.þA-law has 13 bits of dynamic range (78 dB). See also µ-law or u-law on page 21 and G.711 on page 11.

ADSL

Acronym for *Asymmetric Digital Subscriber Line*. A type of DSL utilizing DMT providing a higher downstream (to the customer site) than upstream (to the service provider). A single twisted-pair offers up to 12 Mbps downstream and 1.8 Mbps upstream.

ADSL2+

The latest ADSL technology. Like the previous versions of ADSL and ADSL2, ADSL2+ uses DMT providing a higher downstream (to the customer site) than upstream (to the service provider). A single twisted pair offers up to 24 Mbps downstream and 3.3 Mbps upstream.

Aggregate

Combining two or more bit streams into a single bit stream.

American wire gauge

See AWG on page 7.

Analog telephone adapter

See ATA on page 7.

Asymmetric digital subscriber line

See ADSL.

Asynchronous communication
A data communications method in which bits are sent without using a clock signal for synchronization. Instead, each character is transmitted surrounded by a start and stop bit that designates the beginning and ending points of the information. This as opposed to synchronous communication where blocks of data are transmitted using a synchronizing clock.

Asynchronous transfer mode
See ATM.

ATA
Acronym for *analog telephone adapter*, an ATA enables you to use analog phones for VoIP without having to purchase an IP phone. But an ATA cannot provide many of the added features (such as transfer and hold buttons) that an IP phone provides.

ATM
Acronym for *Asynchronous Transfer Mode*. A connection-oriented switching and multiplexing technique using 53-byte cells to transmit different types of data concurrently across a single physical link. ATM framing carries between 15–40% overhead depending on packet size.

Audio menu
A verbal choice provided by a recording over the phone. Audio choice menus are common in automated attendant, IVR and fax-on-demand systems. They are prompts for caller input. Audio menus can instruct you to speak commands or press keys on a touch-tone keypad as commands.

ARU
Acronym for *Audio Response Unit*. A computer telephony system incorporating voice store-and-forward technology. There are passive and interactive ARUs. Passive ARUs simply play out messages while interactive ARUs play messages based on input from callers.

Audio response unit
See ARU.

Audio teleconferencing or audio conferencing
The original technology used for audio teleconferencing was based on PBX (Private Branch Exchange) conferencing circuits. Setting up conference calls through the PBX is cumbersome, the voice quality degrades as the number of people on a call increases, and there are capacity limitations, so specialized conference bridges were developed to improve capacity and voice quality. Conference bridges, however, require trained operator intervention to schedule and invoke most features. As a result, individual corporations found the cost of ownership prohibitive, and the market for such products has been concentrated on service bureau providers. Today's PC-based systems provide the freedom of conference bridges. By installing a conference server on your voice networks, you can set up, attend, and manage your own conferences over any touch-tone telephone. Additionally, users can schedule meetings using desktop software from their e-mail systems, or from a web browser.

AWG
Acronym for *American Wire Gauge*. An indication of wire diameter. The heavier the gauge, the lower the AWG number, and the lower the impedance.

BLEC
Acronym for *Building Local Exchange Carrier*. A telephone service provider whose equipment and network are contained within a building or complex.

Broadband
Broadband is short for broad bandwidth. When used in a VoIP context, broadband refers to the information capacity (2 Mbps or higher) of a communication channel. This high-capacity, two-way medium supports a wide range of frequencies, typically from audio up to video frequencies.

Building local exchange carrier
See BLEC.

Cat 5
Short for *Category 5*. A level of unshielded twisted-pair wiring performance as defined by EIA/TIA-568.

Cat 5 cable is used for transmission speeds up to 1000 Mbps (1 Gbps).

CCITT

Acronym for the *Comité Consultatif Internationale de Téléphonie et de Télégraphie*, an organization based in Geneva, Switzerland that develops world-wide data communications standards. CCITT is part of the ITU. Three main sets of standards have been established: CCITT Groups 1-4 standards apply to facsimile transmissions; the CCITT V series of standards apply to modems and error detection and correction methods; and the CCITT X series standards apply to local area networks.

Central office

See CO.

CLE

Acronym for *Customer Located Equipment*.

CLEC

Acronym for *competitive local exchange carrier*, telecommunications carriers that provide local exchange service in competition with an ILEC, using just the CLEC's own switching and network or a combination of the CLEC's switching facilities and the CLEC's network facilities or an ILEC's unbundled network facilities. Examples of CLECs are Teleport, GTE, and AT&T.

CLI

Acronym for *command line interface*. Configuration interface for devices such as CPE', IAD's, etc.

CO

Acronym for *central office*, a telco facility that handles the switching of telephone calls on the PSTN for a small regional area. The CO houses servers, storage systems, switching equipment, emergency power systems, and related devices that are used to run telephone systems.

Codec

An acronym for *coder/decoder* (also defined as *compressor/decompressor*). In a VoIP context, a codec is a DSP software algorithm that converts an audio signal into a digital format suitable for transportation using a specific set of protocols. Equipment on the receiving end, which must also use these same protocols, restores the digital signals back to audio (or analog) format. Codecs differ based on the quality of the sound they produce, how much bandwidth they require, the processing power needed, and so on. The main voice compression schemes used in VoIP networks are: G.711, which is the same compression used on traditional wired telephone networks and used 64-kbps coding; G.729, which compresses voice to an 8-kbps rate but still delivers near toll quality voice over most networks; and G.723, which compresses voice at 6.3 or 5.3-kbps, but also has the poorest quality compared to the other two codecs.

Coder/decoder

See Codec.

Comfort noise

See Silence suppression on page 19.

Comité Consultatif Internationale de Téléphonie et de Télégraphie

See CCITT.

Command line interface

See CLI.

Competitive local exchange carrier

See CLEC.

Conference bridge

A device used to connect multiple parties over the phone. A proctor or operator can man conference bridges or they can be supervised. There are standalone conference bridges and conference bridge functions built in to some PBXs (Private Branch Exchange). These systems have circuitry for summing and balancing the energy (noise) on each channel so everyone can hear each other. More sophisticated conference bridges have the ability to "idle" the transmit side of channels of non-speaking parties.

CPE
Acronym for *customer premise equipment*, a CPE is a networking device, such as a modem, POTS splitter, or other device, that is installed at a customer site. CPEs, which terminate the telco or broadband network, pass communications streams to VoIP gateways, switches, routers, PBXs, DSUs or CSUs, telephone sets, personal computers, set-top boxes, and other devices at the customer's premises.

Customer located equipment
See CLE on page 8.

Customer premise equipment
See CPE.

DDNS
An acronym for *dynamic domain naming system* or *dynamic DNS*. When you connect via broadband or ISDN to your ISP, you are normally assigned a dynamic IP address. Depending on how your ISP has set it up, this address can change each time you connect to the ISP or when a time limit expires. If you are running a local server, when the IP address changes, remote users will not be able to locate your new IP address. Some people pay their ISPs for a static IP address that never changes. Others use various free services on the Internet that provide dynamic DNS service. Once registered, you choose a name (myhostname.dyndns.org for example) that is mapped to your current IP address so remote users can get to your network via that address. Thereafter, when a new IP address is assigned, software on your server automatically notifies the DDNS service of the change, at which the service remaps to the new address, so remote users never notice the change.

DHCP
Acronym for *dynamic host configuration protocol*, a method for automatically assigning IP addresses to devices on a TCP/IP network. When a new device connects to the network, the DHCP server allocates an IP address from a list of available addresses. The device keeps this IP address until the session ends. After the device disconnects, the IP address is released and available for use.

Digital subscriber level 1 channel
See DS1 on page 9.

Digital subscriber level zero
See DS0.

Digital subscriber line
See DSL on page 10.

Digital subscriber line access multiplexer
See DSLAM on page 10.

Discrete MultiTone
See DMT.

DMT
Acronym for *Discrete MultiTone*. DSL technology using digital signal processors to divide the signal into 256 sub-channels. (Ex: ADSL, VDSL)

DNS
Acronym for *domain name system* or *domain name service*, DNS is a process that maps hostnames (which are more easily remembered than numbered addresses) to IP addresses.

Domain name service
See DNS.

Domain name system
See DNS.

Downstream
In the direction toward the customer premises.

DS0
Short for *Digital Subscriber Level Zero*. A 64-kbps unit of transmission bandwidth. A worldwide standard speed for digitizing one voice conversation, and more recently, for data transmission. Twenty-four DS0s (24 x 64 kbps) equal one DS1.

DS1
Short for *Digital Signal 1*. DS1 is the primary digital telephone standard used in the United States and

Japan and is able to transmit up to 24 multiplexed voice and data calls over telephone lines.

DSL

Acronym for *digital subscriber line*, a high speed digital switched service that uses existing copper pairs to connect subscriber CPE (customer premises equipment) to the CO (central office). DSL handles more data downstream (data flowing towards the subscriber) than upstream (flowing towards the network).

DSLAM

Acronym for *digital subscriber line access multiplexer*. Also known as an *IAC (Integrated Access Concentrator)*. A piece of equipment located in the Central Office (CO) that combines (or *multiplexes*) multiple DSL subscriber lines into a single high-speed connection (usually an OC-3 or OC-12 optical trunk). When the telco receives a DSL signal, an xDSL modem with a POTS splitter detects and routes voice calls and data. It sends voice calls to the PSTN and data to the DSLAM, where it passes through the ATM to the Internet, then back through the DSLAM and xDSL modem before returning to the customer's computer

DTMF

Acronym for *dual tone multi-frequency*, the signal a telephone company receives when a telephone's touchpad keys are pressed. Also known as *Touch Tone*.

Dual tone multi-frequency

See DTMF.

Dynamic DNS

See DDNS on page 9.

Dynamic domain naming system

See DDNS on page 9.

Dynamic host configuration protocol

See DHCP on page 9.

E1

A 2.048 Mbps signal that supports thirty-two 64 kbps timeslots, at least 30 of which can transmit and receive data or digitized voice. The most common configurations for E1 lines are E1 PRI, and unchannelized E1

EFM

Acronym for *Ethernet in the First Mile*. Uses one of the Ethernet family network protocols between a telecommunications company and a customer's premises. From the customer's point of view it is their "first" mile, although from the access network's point of view it is known as the "last mile".

Element management system

See EMS.

EMS

Acronym for *Element Management System*. An EMS consists of systems and applications for managing network elements (NE) on the network element-management layer (NEL) of the telecommunications management network (TMN).

Enterprise session border controller

See eSBC.

eSBC

Acronym for *Enterprise Session Border Controller*. Controls the communication between different networks, typically between customers LAN and WAN.

Ethernet

A type of network that supports high-speed communication among systems. It is a widely implemented standard for LANs.

Ethernet demarcation

Provides a clear separation between the user and the network, allowing carriers to extend network visibility up to the user premises.

Ethernet extenders

Ethernet signals begin to degrade beyond 100 meters (328 feet). To achieve longer distances, an Ethernet extender is required. Ethernet Extenders can be used to drive Ethernet up to 10 kilometers (6.4 miles) over copper. Extenders are also commonly used to provide Ethernet over a single voice-grade twisted-pair as opposed to Cat 5 or higher rated cables. These devises are typically transparent to higher layer protocols.

Ethernet in the first mile
See EFM.

Ethernet telephone
See IP phone on page 14.

Fax server
A computer based fax machine. Fax servers are "shared use" devices, typically installed on a LAN. Clients on the LAN can use the fax server from their PCs in much the same way they share a network-based (shared) printer. Faxes can be generated by users at their workstations and "printed" to the fax server for transmission. Likewise, fax servers can route incoming faxes to printers, file server directories, or to individual users. Fax servers save users from having to print documents, carry them to the fax machine, and subsequently wait for them to be transmitted after creating a cover page.

First mile
Sometimes referred to as Local Loop, the final leg of delivering communications connectivity to a resident or customer.

Foreign exchange office
See FXO.

Foreign exchange station
See FXO.

Frame relay
In data communications, Frame Relay is a packet switching method that uses available bandwidth only when it is needed. This fast packet switching method is efficient enough to transmit voice communications with the proper network management.

Full duplex
In telephony and data communications, full duplex means the ability for both ends of a communication to simultaneously send and receive information without degrading the quality of the content.

FXO
Acronym for *foreign exchange office*. A telephone signaling interface that generates off-hook and on-hook indications at the foreign exchange station (FXS) at the end of a telephone circuit.

FXS
Acronym for *foreign exchange station*. Telecom equipment instance that delivers line power, generates dialtone and ringing voltage to a subscriber device (telephone).

G.711
G.711 is an ITU-T standard for audio compression that is mostly used in telephony. The standard defines two algorithms: *μ-law* algorithm (used in America) and *a-law* algorithm (used in Europe and the rest of the world). Both are logarithmic, but a-law was specifically designed for improved computer processing.

G.723.1
The G.723.1 codec is commonly used in VoIP applications because of its low-bandwidth requirements. G.729 compresses voice audio in 30-millisecond segments. Music or tones (DTMF Touch Tone or fax tones, for example) cannot be transmitted reliably with this codec, so G.711 or out-of-band methods are used instead to carry these signals. G.723.1 supports 6.3-kbps data rates (in 24-bytes segments) and 5.3-kbps in 20-byte segments.

G.729
The G.729 codec is commonly used in VoIP applications because of its low-bandwidth requirements. G.729 compresses voice audio in 10-millisecond segments. Music or tones (DTMF Touch Tone or fax tones, for example) cannot be transmitted reliably with this codec, so G.711 or out-of-band methods are used instead to carry these signals. Standard G.729 operates at 8 kbps, but G.729 has two variants: Annex A (G.729A) which is less processor intensive and allows double the number of calls as plain G.729 and Annex B (G.729B) which adds *voice activity detection* (VAD) and *comfort noise generation* (CNG) which work together to reduce bandwidth used. You can combine Annex B with G.729A to give G.729AB.

The G.729 variants can generally interoperate with each othe but there are modified versions that provide and 11.8-kbps rates for better speech quality. Also very common is G.729A which is compatible with G.729, but requires less computation (at the cost of degraded speech quality).

G.dmt
A name for the line modulation specified by ITU recommendation G.992.1.

G.lite
A name for the line modulation specified by ITU recommendation G.992.2.

G.SHDSL
G.SHDSL, or SHDSL, is a standardized method (ITU-T G.991.2) to transport symmetrical data rates on copper pair access lines. G.SHDSL offers bit rates from 192 kbps to 15.3 Mbps over a 2-wire single pair and up to 60 Mbps over four bonded pairs

Gatekeeper
This component of H.323 manages the bandwidth inbound and outbound from the LAN. Gatekeepers register clients and coordinate communications with other gatekeepers. A gatekeeper performs the following functions: admission control for authorizing clients' access to the LAN; bandwidth control for each network segment managed; client network address translation so users can dial network locations with easily remembered aliases (such as e-mail addresses) instead of IP addresses; and call management that monitors H.323 calls and tracks rejected calls.

Gateway—VoIP
Device that is converting traditional telephony in to VoIP. For instance, T1 ISDN lines to be converted over to VoIP using SIP or H.323

GRE
Acronym for *Generic Routing Encapsulation* A protocol that encapsulates other protocols in order to route them over IP networks

GSM 6.10
Short for *Groupe Speciale Mobile 6.10* or *Global System for Mobile Communications 6.10*, GSM 6.10 is a lossy CBR (constant bit rate) codec in which data is compressed into a stream at a fixed rate. GSM 6.10 has a sample rate of 8k samples per second and a data rate of 13 kbps. GSM is also a popular global mobile phone standard.

GUI
Acronym for *Graphical User Interface*. Mostly used for CPE type device configuration

H.245 tunneling
H.245 tunneling is the encapsulation of H.245 messages within H.225/Q.931 messages. If you have a firewall and enable H.245 tunneling, there is one less TCP port that you need to allow for incoming connections.

H.323
H.323 is the ITU standard for the transmission of real-time audio, video and data information over packet switching-based networks. Such networks include IP-based Internet packet exchange-based local area networks, enterprise networks, and metropolitan and wide area networks.

HTTP
Acronym for *Hyper Text Transfer Protocol*. An application-level protocol for the transmission of information in distributed systems.

HTTPS
Acronym for *Hyper Text Transfer Protocol Secure*. The secure version of HTTP, the protocol over which data is sent between your browser and the website to which you are connected.

I-Phone
See IP phone on page 14.

IAC
Acronym for *Integrated Access Concentrator*. An access gateway providing aggregation and switching for multiple DSL connections.

ILEC

Acronym for *incumbent local exchange carrier*, the U.S. local telephone companies that comprised the seven Bell Operating Companies (such as Pacific Bell or GTE) until the 1996 Telecommunications Act established CLECs. Also called *Baby Bells* or *dominant carriers*. See also RBOC on page 18.

Incumbent local exchange carrier

See ILEC.

Integrated access concentrator

See IAC.

IVR

Acronym for *Interactive voice response*. In computer telephony, IVR is a horizontal application wherein computer-based information is accessed over the phone by using a telephone instead of a computer. An IVR platform uses computer telephony components to translate callers' touch-tones or voice commands into computer queries after the callers listen to an audio menu. For example: "Please enter your account number using the touch-tones on your telephone." These queries are then "fetched" by the IVR platform from the host computer. In some cases, the information resides in the same platform (self-hosted). The information is converted into voice commands that are spoken over the phone to the caller.

Interactive voice response

See IVR.

International Telecommunication Union

See ITU on page 14.

Internet

The Internet consists of the world's combined public IP-based packet-switched networks. The Internet is an outgrowth and combination of a variety of university and government sponsored computer networks. Federal and private sector subsidies supported the DARPA-NET, NSFnet (National Sciences Foundation,) and thousands of other subnetworks, which were used to do inter-agency research and communication. Today, the Internet is made up of millions upon millions of computers and subnetworks—almost entirely supported by commercial funds except in countries where deregulation has not occurred. The Internet is the substrate and chief communications backbone for the WWW.

Internet protocol

See IP.

Internet service provider

See ISP on page 14.

Internet telephone

See IP phone on page 14.

Internetwork

An interconnected group of networks (also called an internet).

Interoperability

The ability of equipment from different vendors to communicate using common protocols.

Intranet

A private network or internet using Internet standards and software, but protected from public access.

Internet telephony

Any means of transmitting the human voice (real-time or near real-time) over the Internet. There are several components: 1) On the client side, a multimedia-equipped PC with special client software will digitize your voice. This can be done with a voice modem or other voice encoding method; 2) A direct or dial-up connection to the Internet allows your voice to be transmitted in packet form to its destination; 3) Connection with the far side is achieved by IP address search, common servers or beacons to identify the called party (and to "ring" that person's phone); 4) A similar arrangement on the far end completes the call and allows both parties to speak. There are also PSTN/Internet gateways that allow regular telephone callers to make phone-to-Internet-to-phone connections. There are PC-to-phone connections and phone-to-PC connections.

IP
Acronym for *Internet Protocol*. An open networking protocol used for internet packet delivery, IP specifies the format of packets, also called datagrams, and the addressing scheme. IPv4 is for IP version 4 (RFC 791) and IPv6 is for IP version 6 (RFC 2460)

iPhone
See IP phone.

IP-PBX
An IP-PBX is a business phone system that manages telephones throughout the enterprise and acts as a gateway to data and voice networks. An IP-PBX is a combination switch/router and PBX that handles VoIP so you can place calls using over a packet-switched network instead of the circuit-switched PSTN.

IP address
The Internet protocol (IP) address (also called the *Internet address*) is the numeric address—formatted as four sets of numbers separated by periods (*204.171.64.2*, for example)—of a computer attached to a TCP/IP network. Every client and server station must have a unique IP address. Client workstations have a permanent (or *static*) address or one that is dynamically assigned to them each dial-up session.

IP phone
IP phones work and look like regular telephones, but they have the ability to connect to the Internet and make cost-saving long-distance calls.

IP streaming video
See VioIP on page 22.

IP telephony
See Internet telephony on page 13.

IPsec
Short for *Internet Protocol Security*. IPsec is a network protocol suite that authenticates and encrypts the packets of data sent over a network.

ISP
Acronym for *Internet service provider*, a business that provides subscriber-based access to the Internet. Subscribers can be individuals or businesses. According to Jack Rickard, publisher of Boardwatch Magazine, ISPs operate at the fourth or lowest level of the Internet. At the third level, regional providers aggregate traffic from lower-order ISPs to the second, backbone level. The highest level in North America is the NAP (network access point), which acts as peer-to-peer interconnection points for the largest backbones. There are three ”official” NAPs located in San Francisco, California; Chicago, Illinois; and Pennsauken, New Jersey. ISPs use Internet routers, servers and Rrack-mounted modems to provide a variety of services, including web site hosting, FTP service, e-mail accounts, unified messaging, audio and video broadcasting, and—in some cases—Internet telephony and fax gateway services.

ITU
Acronym for the *International Telecommunication Union*, an agency of the United Nations based in Geneva, Switzerland. The ITU sets standards related to facsimile communications. Previously the ITU was known as the *CCITT*.

Jitter
Jitter is a measure of variations in the delays that occur while delivering voice packets from sending to receiving ends.

Latency
Latency or *delay*, is a measure of how much time it takes for a packet of data to get from one point to another. Together, latency and bandwidth define the speed and capacity of a network.

Layer
A logical level in the Open Systems Interconnection model.

LEC
Acronym for *local exchange carrier*, A company that provides intra-LATA (local access transport area) telecommunications services.

Local exchange carrier
See LEC.

Loopback
A diagnostic procedure that sends a test message back to its origination point. Used to test various portions of a data link in order to isolate an equipment or data line problem.

Main distribution frame
See MDF.

Management information base
See MIB.

MCU
Acronym for *Multi-Commercial Unit*. A building or complex with commercial tenants, such as an office building or shopping mall.

MDF
Acronym for *Main Distribution Frame*. The point where all local loops are terminated at a central office.

MDU
Acronym for *Multi-Dwelling Unit*. A building housing multiple residences, such as an apartment building.

Mean opinion score
See MOS.

Messaging
In computer telephony, any means of storing and forwarding messages. This includes fax mail, voice mail, and broadcast messaging. This horizontal application is the most popular of all voice solutions. Messaging systems provide for the storing and forwarding of "non-real time" communication. For example, a recorded voice message can be stored for later playback either locally or remotely, or a fax can be received and stored before it is re-transmitted to the ultimate recipient. Messages can vary in content and media type—the distinction being that they are recorded or stored for pick up in the future.

MHU
Acronym for *Multi-Hospitality Unit* or *Multi-Hotel Unit*. A hotel or motel.

MIB
Acronym for *Management Information Base*. A database of managed objects used by SNMP to provide network management information and device control.

Microfilter
A microfilter (also called a *splitter*) filters out the broadband signal from the normal phone signal, enabling you to use both on the same line. It comes with two sockets, one for the broadband router or modem and one for the phone, fax, set top box, or answering machine.

Modem
A modem (modulator/demodulator) is equipment that converts digital signals to analog signals and vice-versa. Modems are used to send data signals (digital) over the telephone network, which is usually analog. A modem modulates binary signals into tones that can be carried over the telephone network. At the other end, the demodulator part of the modem converts the tones to binary code.

Modulator/demodulator
See Modem.

MOS
Acronym for *mean opinion score*. Voice quality is reported as an MOS on a scale from 1 to 5 where 1 is the worst and 5 the best quality. Originally, telephone companies would hire people to listen to test phone calls and rank the quality of those calls. They would then take the average or the *mean* opinion of the listeners and assign that as the call's quality. Today, MOS scores can be arrived at based on network performance measures such as jitter and latency using an international telephony standard called the E-Model (the ITU-T G.107 standard).

MSAR
Acronym for *Multi-Service Access Router*. Device that integrates at least one WAN technology such as VDSL for multiple services at the customer premise.

MSBR
Acronym for *Multi-Service Business Router*. Customer premise device for multi service purposes, but without WAN interface.

MTU
Acronym for *Multi-Tenant Unit*. A building housing multiple tenants, such as an office building. Sometimes used as a generic term to encompass Multi-Commercial Units (MCUs), Multi-Dwelling Units (MDUs), and Multi-Hospitality Units (MHUs).

Multiplex
Multiplexing combines input signals from many sources onto a single communications path, thereby enabling a single communications channel to carry several messages simultaneously.

Multi-commercial unit
See MCU.

Multi-dwelling unit
See MDU.

Multi-hospitality unit or multi-hotel unit
See MHU on page 15.

Multi-service access router
See MSAR.

Multi-service business router
See MSBR.

Multi-tenant unit
See MTU.

Network functions virtualization
See NFV.

Network management system
See NMS.

Network operations center
See NOC.

Network service provider
See NSP.

NFV
Acronym for *Network Functions Virtualization*. An emerging networking technology in which functionality conventionally carried out in dedicated network elements is performed in software hosted on computer hardware or virtual machines.

NMS
Acronym for *Network Management System*. A computer system used for monitoring and controlling network devices.

NOC
Acronym for *Network Operations Center*. The point at which a network is monitored and controlled.

Node
A connection or switching point on the network.

NSP
Acronym for *Network Service Provider*. A local telephone company or ISP that provides network services to subscribers.

OAM
Acronym for *Operations, Administration and Maintenance*.

OC-3
A fiber optic line capable of 155 Mbps of bandwidth.

OC-12
A fiber optic line carrying 622.08 Mbps of bandwidth.

Optical Carrier-3
See OC-3.

Optical Carrier-12
See OC-12.

PABX
See PBX.

Packet
A logically grouped unit of data. Packets contain a payload (the information to be transmitted), originator, destination, and synchronization information. The idea with packets is to transmit them over a network so each individual packet can be sent along the most optimal route to its destination. Packets are constructed on one end of the communication and deconstructed on the receiving end based on the header addressing information at the front of each packet. Routers in the network will store and forward packets based on network delays, errors, and re-transmittal requests from the receiving end.

Packet switching
A means of economically sending and receiving data over multiple network channels. Packet switching takes data and breaks it down into packets—small bundles of information containing the payload and routing information. The packets are then transmitted to the receiving end, where they are converted back to the original data format. One feature of packet switching is that packets can be received out of order and then be quickly arranged into the correct order. There are slow packet switching networks—like the old SNA networks—and fast packet networks based on Frame Relay and ATM. Although traditionally used for data, packet networks—especially well-managed ones—are suitable for real-time transmission of voice and video.

Packet transfer mode
See PTM on page 18.

PBX
In telephony, a *private branch exchange* (PBX) (also called a *private automatic branch exchange* or *PABX*) system behaves as a customer's premises over trunk lines (thus the term branch). At first, PBXs mimicked a small telephone company switchboard. Users would use an operator to make telephone calls to the PSTN (public switched telephone network). Now, users dial directly, without using an operator; computer telephony platforms such as automated attendants are able to route incoming calls automatically, too.

Plain old telephone system
See POTS.

Point of presence
See PoP.

Point-to-point protocol
See PPP.

Point-to-point protocol over Ethernet
See PPPoE.

PoP
An acronym for *point of presence.* The Network Service Provider's access point on a Network Access Provider's network.

POP
Short for *post office protocol*, an Internet standard for storage and retrieval of email messages.

Post office protocol
See POP.

Post, telegraph, and telephone
See PTT.

POTS
An acronym for *Plain Old Telephone Service*. Standard telephone service over the PSTN, with an analog bandwidth of less than 4 kHz. See also PSTN.

POTS splitter
A device that filters out the DSL signal and allows POTS frequencies to pass through.

PPP
An acronym for *Point-to-Point Protocol.* A protocol for packet transmission over serial links, specified by RFC 1661.

PPPoE
An acronym for *Point-to-Point Protocol over Ethernet*. A method for establishing sessions and encapsulating PPP packets over an Ethernet, specified by RFC 2516.

Private automatic branch exchange
See PBX on page 17.

Private branch exchange
See PBX on page 17.

Protocol
A set of rules that determines the behavior of devices in achieving and maintaining communication.

PSTN
An acronym for *Public Switched Telephone Network*. A network shared among many users who can use telephones to establish connections between two points. Also known as *dial network*.

PSTN backup
See PSTN failover.

PSTN failover
PSTN failover lines are used as backup connections in case your Internet connection is not available. These are optional ports on ATA devices or IP phones that connect directly to the analog PSTN lines coming from the telco. This setup requires having both a regular analog telephone lines and an account with a VoIP Service Provider.

PSTN (POTS) gateway
Software that enables H.323 clients to make outgoing calls, and for incoming calls to be routed to H.323 clients.

PTM
An acronym for *Packet Transfer Mode*. PTM is a type of data communication that is not synchronized by a clock. Data is broken into variable-size units of data, and those packets are relayed from one node to another, often in multiple parallel paths, until they reach their final destination. PTM framing caries between 2-9% overhead depending on packet size.

PTT
An acronym for *Post, Telegraph, and Telephone*. A national communications authority, sometimes government-controlled and monopolistic, which acts as a common carrier.

Public switched telephone network
See PSTN.

Punchdown block
An array of connectors used for connecting cable circuits of a network interface.

QoS
Measure of how well a transmission system performs in terms of transmission quality and service availability. As a concept, QoS means that transmission rates, error rates, and other characteristics can be measured, improved, and, within limits, guaranteed in advance. QoS incorporates bandwidth, latency, and jitter to describe a network's ability to customize the treatment of specific classes of data. For example, QoS can be used to prioritize audio transmissions over Web browsing traffic.

Quality of service
See QoS.

RBOC
Short for *Regional Bell Operating Company*, the seven companies formed to manage the local exchanges originally owned by AT&T.

Real-time
Communications wherein perceptible delays between the sender and receiver are minimal and easily tolerated are considered to take place in real-time. Regular telephone calls are real time. Point-to-point fax transmissions are near to real-time. Voice messaging is not real-time.

Regional Bell operating company
See RBOC.

Registered jack-11
See Registered jack-11.

Registered jack-45
See RJ-45.

RJ-11
The designation for connecting a tip and ring circuit to a standard, modular, 6-position jack.

RJ-45
Eight-position modular connector used for data transmission over standard twisted or flat pairs.

Router
A device that connects LANs by dynamically routing data according to destination and available routes.

RSVP
Short for *resource reservation protocol*, RSVP is a protocol used in VoIP to manage QoS. Defined in *RFC 2205: Resource ReSerVation Protocol (RSVP)*, RSVP works by requesting that required bandwidth and latency be "reserved" for the VoIP telephone call by every network device between the two endpoints.

RTP
Acronym for *real-time transport protocol*, RTP encapsulates VoIP data packets inside UDP packets. *Defined in RFC 3550—RTP: A Transport Protocol for Real-Time Applications*, RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality.

SDH
An acronym for *synchronous digital hierarchy*. This is the European standard for metropolitan fiber rings, used by ILECs. SDH handles multiple data types (voice, video, and so on). The minimum rate for SDH is 155 Mbps, the maximum is 2.488 Gbps.

Service level agreement
See SLA on page 20.

Service provider
A company that provides services to Internet, telephone, and mobile phone users.

Session initiation protocol
See SIP on page 20.

SFP
An acronym for *Small Form-factor Pluggable*. A specification for modular optical transceivers.

Sidetone
Sidetone is a technique in which the speaker's voice is played through the telephone earpiece locally so that the speaker, subconsciously hearing his or her voice instead of silence, will use the telephone more comfortably.

Signaling system #7 (SSY7)
The basis for routing traffic with out-of-band signaling. Its forerunner, CCIS (Common Channel Interoffice Signaling), used 4.8 kbps data links to transmit call set up and tear down messages to switching office adjunct computers and packet switches. SS7 in itself is not a network service offering, but rather the underlying infrastructure upon which many existing and proposed offerings are based. For example, local Basic Rate ISDN (BRI) services can tap into SS7, so 64 kbps packetized data can be routed with the help of the network's out-of-band signaling capability. In addition, nationwide Primary Rate ISDN (PRI) services can use the same backbone.

Silence suppression
70–80% of a phone conversation consists of silence, the rest being used for talking. To save bandwidth, silence activity detection (SAD) and voice activity detection (VAD) techniques are used to avoid transmitting the silent portions of the conversation, thereby saving up to 35% of the bandwidth. A packet

is sent called a silence indicator (SID) to notify the other end that the voice activity power level has dropped below a certain threshold (-50 dbm, for example) thereby marking a silent period. VAD requires a 5-ms look-ahead buffer so this adds delay to the voice path, so VAD is generally used only on WAN circuits. One problem with SAD/VAD is that the natural silences that occur between talking actually contains sounds such as breathing and other background noises that users are not consciously aware of. Therefore, when VAD replaces these subliminal noises with pure silence, it is off-putting to the users; so techniques are employed to add white (or pink) random noise (called comfort noise) locally to simulate this background noise.

SIP

Short for *session initiation protocol*, SIP is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality. Like HTTP or SMTP, SIP works in the Application layer of the Open Systems Interconnection (OSI) communications model. The Application layer is the level responsible for ensuring that communication is possible. SIP can establish multimedia sessions or Internet telephony calls and modify or terminate them.

SLA

An acronym for *Service Level Agreement*. The contract between the subscriber or operator and service provider specifying the agreed to service level commitments and related business agreement.

Softphone

A softphone is a software application that loads the VoIP service onto your desktop computer or laptop. Softphone applications enable you to place VoIP calls from your computer to any location if you have access to a broadband connection.

Softswitch

Short for *software switch* (also referred to as a *media gateway controller*, *gatekeeper*, and *call agent*), a softswitch is a software-based switching platform that links IP networks to the PSTN and manages traffic containing a mix of voice, fax, data, and video.

SOHO or SoHo

Acronym for *small office, home office*, a term that refers to people who work in very small businesses or home offices.

SONET

An acronym for *synchronous optical network*. This is the ANSI standard for metropolitan fiber rings, used by ILECs. Traffic is carried in an electronic packet transported over fiber, using dual counter-rotating rings. The SONET standard defines a hierarchy of interface rates that allow data streams at different rates to be multiplexed. It establishes Optical Carrier (OC) levels from 51.8 Mbps to 2.48 Gbps (10 Gbps for OC-192).

Speech recognition

Speech recognition describes a technology that enable callers to speak words that are used to control applications.

Store and forward

The method for storing a message or transmission for later playback or transmission. As opposed to real-time communication, store and forward is the basis for all messaging systems, including email, fax-on-demand, unified messaging, etc. In data communications, store and forward applies to momentary buffering of packets or other data strings.

T1

A digital transmission with a capacity of 1.544 Mbps used in North America. A T1 may be channelized into 24 DS0s, each capable of carrying a single voice conversation or data stream. T1 may be carried on coaxial cable or two twisted pairs.

T3

North American standard for DS-3. Operates at a signaling rate of 44.736 Mbps, or the equivalent of 28 T1s.

TC-PAM
An acronym for *Trellis-Coded Pulse-Amplitude Modulation*, Modulation format that is used in HDSL2 and G.SHDSL. It is a variant of trellis coded modulation (TCM) which uses a one-dimensional pulse-amplitude modulation (PAM) symbol space, as opposed to a two-dimensional quadrature amplitude modulation (QAM) symbol space. TC-PAM is spectrally compatible with all other forms of DSL.

TCP
An acronym for *transmission control protocol*, The transport layer protocol developed for the ARPAnet which comprises layers 4 and 5 of the OSI model. TCP controls sequential data exchange in TCP/IP for remotely hosts in a peer-to-peer network. See also IP on page 14.

Telco
Telephone company.

Telephony
Taken from Greek root words meaning far sound, telephony means the process of converting or transmitting voice or other signals over a distance, and then re-converting them to an audible sound at the far end.

Touch tone
See DTMF on page 10.

TR-069
Technical Report 069, is a specification for remote management of end user devices (CPEs).

Traffic shaping
Traffic shaping (also known as *metering*, *shaping*, and *smoothing*) refers to the use of queues to reduce surges that can clog a network. Data is buffered in a queue and then released to the network in set amounts to ensure that the traffic will fit within the promised traffic limits for the particular connection. Traffic shaping is used in ATM, Frame Relay, and other types of networks.

Transmission control protocol
See TCP.

µ-law or u-law
µ-law or u-law is an audio codec companding format in which the signal is compressed on input and expanded back to its original form on output. More correctly known as mu-law, it performs logarithmic mapping between an 8-bit data space and 14-bit sample space as described in the CCITT G.711 recommendation.þµ-law has 14 bits of dynamic range (84 dB).þSee also A-law on page 6 and G.711 on page 11.

UNIX
A multi-user, multi-tasking operating system originally developed in 1969 by Ken Thompson of AT&T Bell Laboratories. UNIX is used in telephone company and mission-critical applications.

Upstream
In the direction of the telephone network or server.

UTP
An acronym for *Unshielded Twisted Pair*.

VAD
See Silence suppression on page 19.

VDSL2
An acronym for *Very High Bit Rate Digital Subscriber Line 2*. Similar to ADSL, VDSL2 is a type of DSL utilizing DMT. At longer distances the transmission provides a higher downstream (to the customer site) than upstream (to the service provider). On short loops VDSL2 can offer symmetrical performance. A single twisted pair offers up to 200 Mbps downstream. A new 35b profile allows for up to 400 Mbps downstream.

Vectoring
Also known as *G.vector* or *VDSL2 vectoring*. Vectoring is a transmission method that employs the coordination of line signals for the reduction of self-FEXT (far-end crosstalk) levels in both the downstream and upstream directions.

Very high bit rate digital subscriber line 2
See VDSL2 on page 21.

VioIP
Short for *video over IP* (also called *IP streaming video*), VioIP enables video signals to be transported and managed over IP networks. With VioIP, live or pre-recorded video content is captured and compressed, then transmitted via the network to one or more receiving stations for viewing.

Video-over-IP
See VioIP.

Voice activity detection
See Silence suppression on page 19.

Voice quality
Human speech uses a bandwidth of 100 Hz to 10 kHz (if you include harmonics) with most of the speech occurring in the 100 Hz to 3 kHz range. The more bandwidth that is allocated to reproduce speech, the more faithfully the sound compares to the original, which is known as *fidelity*. Voice quality is a measure of how well a person can hear the voice on the opposite end of the connection after it has been subjected to signal compression and the effects of *echo*, *latency* (or *delay*), and *jitter*. Voice quality is most commonly stated as a metric called the mean opinion score (MOS). See also MOS on page 15.

VoIP gateway
A VoIP gateway translates protocols between telephony and packet networks to deliver voice and similar services. In compliance with the H.323 standard, gateways serve as network endpoints that support real-time communication between other endpoints. VoIP gateways support voice communication between terminals on a packet network (usually IP) and terminals on a circuit network (the PSTN, for example).

VoIP router
A VoIP router connects IP networks together. When an IP phone or ATA connects to the network, it does so through a router. Some IP phones and ATAs have built-in routing capabilities and their use is recommended because they support VoIP prioritization, which means they are alotted bandwidth to ensure VoIP quality, regardless of other traffic.

VoIP telephone
See IP phone on page 14.

VoIP switch
A VoIP switch is very much like an ordinary PBX in that it enables users to make telephone calls to the PSTN. The difference is that devices that connect to a PBX are hardwired (or connected permanently) to it, whereas with a VoIP switch, the IP protocol enables connections to be softwired and easily modified.

VPN
Acronym for *virtual private network*, a network that is constructed by using public wires to connect nodes. A VPN is a private data network that makes use of the public telecommunication infrastructure, using security measures that involve encrypting data before sending it across the Internet and decrypting the data at the other end.

WAN
An acronym for *Wide Area Network*. A network that spans a large geographic area.

Web browser
Client software used to view information on WWW servers. Web browsers are also packaged with email clients, newsreaders and IP telephony clients.

Web server
On the world wide web, a server dedicated to storing data (such as web pages in HTML format) and distributing it to users. Web browsers are able to download video, text, still images and audio from web pages. Some servers support *unified messaging*.

Wide area network
See WAN.

World wide web
See WWW on page 23.

WWW

Short for *World Wide Web*, WWW (also called *W3*) is an information space in which the items of interest, referred to as resources, are identified by global identifiers called *uniform resource identifiers* (URI). Although the WWW is often considered to be the same thing as the Internet, it is actually part of the Internet. The WWW comprises the resources that can be accessed using Gopher, FTP, HTTP, telnet, Usenet, WAIS and other tools, and the network of hypertext (HTTP) servers.

X.21

An ITU-T standard for a high-speed, 15-position, DCE/DTE interface.

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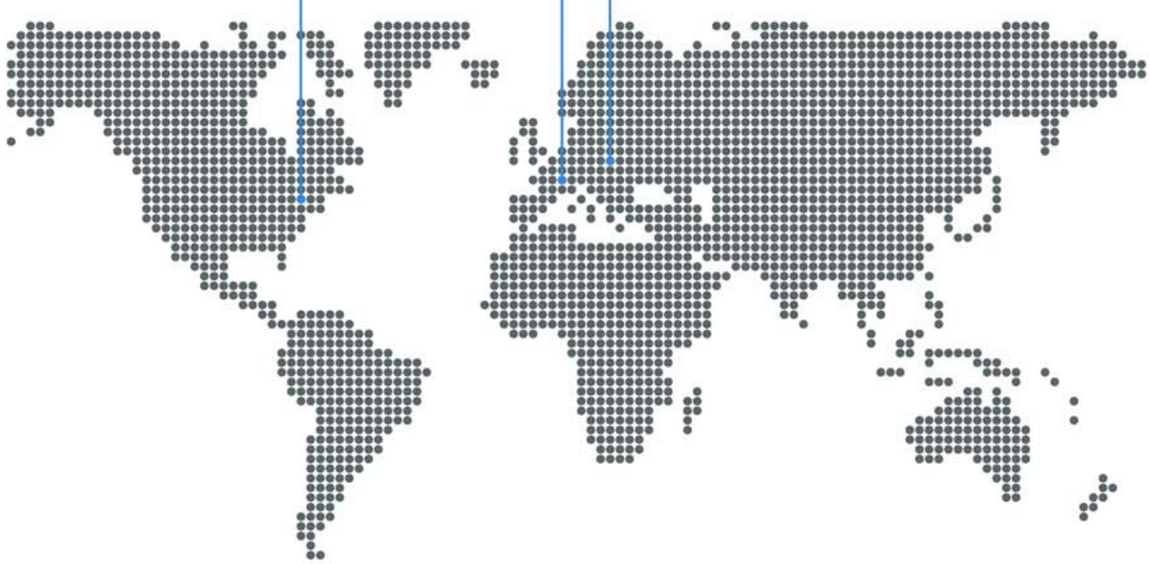

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