GLOSSARY

Numerics

10BaseT 10-Mbps baseband Ethernet specification using two pairs of twisted-pair cabling (Categories 3, 4, or 5): one pair for transmitting data and the other for receiving data. 10BASET, which is part of the IEEE 802.3 specification, has a distance limit of approximately 328 feet (100 meters) per segment.

A

A-law ITU-T companding standard used in the conversion between analog and digital signals in PCM systems. A-law is used primarily in European telephone networks and is similar to the North American µ-law standard. See also companding and µ-law.

AVT tones Out-of-bound signaling as defined in RFC 2833.

C

category-3 cable One of five grades of UTP cabling described in the EIA/TIA-586 standard. Category 3 cabling is used in 10BaseT networks and can transmit data at speeds up to 10 Mbps.

CED tone detection Called station identification. A three-second, 2100 Hz tone generated by a fax machine answering a call, which is used in the hand-shaking used to set the call; the response from a called fax machine to a CNG tone.

CELP code excited linear prediction compression. Compression algorithm used in low bit-rate voice encoding. Used in ITU-T Recommendations G.728, G.729, G.723.1.

CLIP Calling Line Identification Presentation. Shows your identity to callers with Caller ID.

CLIR Calling Line Identification Restriction. Hides your identity from callers with Caller ID.

CNG Comfort Noise Generation.

codec coder decoder. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.

companding Contraction derived from the opposite processes of compression and expansion. Part of the PCM process whereby analog signal values are rounded logically to discrete scale-step values on a nonlinear scale. The decimal step number then is coded in its binary equivalent prior to transmission. The process is reversed at the receiving terminal using the same nonlinear scale. Compare with compression and expansion. See also a-law and µ-law.

compression The running of a data set through an algorithm that reduces the space required to store or the bandwidth required to transmit the data set. Compare with companding and expansion.

CoS Class of service. An indication of how an upper-layer protocol requires a lower-layer protocol to treat its messages. In SNA subarea routing, CoS definitions are used by subarea nodes to determine the optimal route to establish a given session. A CoS definition comprises a virtual route number and a transmission priority field.

D

DHCP Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

dial peer An addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.

DNS Domain Name System. System used on the Internet for translating names of network nodes into addresses.

DSP digital signal processor. A DSP segments the voice signal into frames and stores them in voice packets.

DTMF dual tone multifrequency. Tones generated when a button is pressed on a telephone, primarily used in the U.S. and Canada.

E

E.164 The international public telecommunications numbering plan. A standard set by the ITU-T which addresses telephone numbers.

endpoint A SIP terminal or gateway. An endpoint can call and be called. It generates and/or terminates the information stream.

expansion The process of running a compressed data set through an algorithm that restores the data set to its original size. Compare with companding and compression.

F

firewall Router or access server, or several routers or access servers, designated as a buffer between any connected public networks and a private network. A firewall router uses access lists and other methods to ensure the security of the private network.

FoIP Fax over IP

FQDN Fully Qualified Domain (FQDN) format “mydomain.com” or “company.mydomain.com.”

FSK Frequency shift key.

FXO Foreign Exchange Office. An FXO interface connects to the public switched telephone network (PSTN) central office and is the interface offered on a standard telephone. Cisco FXO interface is an RJ-11 connector that allows an analog connection at the PSTN central office or to a station interface on a PBX.

FXS Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. Cisco's FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

G

G.711 Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. Described in the ITU-T standard in its G-series recommendations.

G.723.1 Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This Codec has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility. Described in the ITU-T standard in its G-series recommendations.

G.729A Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. Described in the ITU-T standard in its G-series recommendations.

gateway A gateway allows SIP or H.323 terminals to communicate with terminals configured to other protocols by converting protocols. A gateway is the point where a circuit-switched call is encoded and repackaged into IP packets.

H

H.245 An ITU standard that governs H.245 endpoint control.

H.323 H.323 allows dissimilar communication devices to communicate with each other by using a standard communication protocol. H.323 defines a common set of CODECs, call setup and negotiating procedures, and basic data transport methods.

I

ICMP Internet Control Message Protocol

IP Internet Protocol. Network layer protocol in the TCP/IP stack offering a connectionless internetwork service. IP provides features for addressing, type-of-service specification, fragmentation and reassembly, and security. Defined in RFC 791.

IVR Interactive voice response. Term used to describe systems that provide information in the form of recorded messages over telephone lines in response to user input in the form of spoken words or, more commonly, DTMF signaling.

L

LDAP Lightweight DirectoryAccess Protocol

LEC local exchange carrier.

Location Server A SIP redirect or proxy server uses a location server to get information about a caller’s location. Location services are offered by location servers.

M

MGCP Media Gateway Control Protocol.

MWI message waiting indication.

μ-law North American companding standard used in conversion between analog and digital signals in PCM systems. Similar to the European a-law. See also a-law and companding.

N

NAT Network Address Translation. Mechanism for reducing the need for globally unique IP addresses. NAT allows an organization with addresses that are not globally unique to connect to the Internet by translating those addresses into globally routable address spaces. Also known as Network Address Translator.

NSE packets Real-Time Transport Protocol (RTP) digit events are encoded using the Named Signaling Event (NSE) format specified in RFC 2833, Section 3.0.

NAT Server Network Address Translation. an Internet standard that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic.

NTP Network Time Protocol. Protocol built on top of TCP that assures accurate local time-keeping with reference to radio and atomic clocks located on the Internet. This protocol is capable of synchronizing distributed clocks within milliseconds over long time periods.

P

POTS
Plain old telephone service. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the PSTN.

Proxy Server
An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it.

PSTN
Public switched telephone network.

Q

QoS
Quality of Service. The capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics.

R

Redirect Server
A redirect server is a server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client. It does not initiate its own SIP request nor accept calls.

Registrar Server
A registrar server is a server that accepts Register requests. A registrar is typically co-located with a proxy or redirect server and may offer location services.

router
Network layer device that uses one or more metrics to determine the optimal path along which network traffic should be forwarded. Routers forward packets from one network to another based on network layer information. Occasionally called a gateway (although this definition of gateway is becoming increasingly outdated). Compare with gateway.

RTP
Real-Time Transport Protocol. One of the IPv6 protocols. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides services such as payload type identification, sequence numbering, timestamping, and delivery monitoring to real-time applications.

S

SCCP
Signaling connection control part.

SDP
Session Definition Protocol. An IETF protocol for the definition of Multimedia Services. SDP messages can be part of SGCP and MGCP messages.

SIP Session Initiation Protocol. Protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.

SIP endpoint A terminal or gateway that acts as a source or sink of Session Initiation Protocol (SIP) voice data. An endpoint can call or be called, and it generates or terminates the information stream.

SLIC Subscriber Line Interface Circuit. An integrated circuit (IC) providing central office-like telephone interface functionality.

SOHO Small office, home office. Networking solutions and access technologies for offices that are not directly connected to large corporate networks.

T

TCP Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.

TFTP Trivial File Transfer Protocol. Simplified version of FTP that allows files to be transferred from one computer to another over a network, usually without the use of client authentication (for example, username and password).

TN power systems A TN power system is a power distribution system with one point connected directly to earth (ground). The exposed conductive parts of the installation are connected to that point by protective earth conductors.

TOS Type of service. See CoS.

U

UAC User agent client. A client application that initiates the SIP request.

UAS User agent server (or user agent). A server application that contacts the user when a SIP request is received, and then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.

UDP User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.

user agent See UAS.

V

VAD Voice activity detection. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

voice packet gateway Gateway platforms that enable Internet telephony service providers to offer residential and business-class services for Internet telephony.

VoIP Voice over IP. The capability to carry normal telephony-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP enables a router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the DSP segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets. VoIP is a blanket term, which generally refers to Cisco's standard-based (for example H.323) approach to IP voice traffic.

X

XML eXtensible Markup Language. Designed to enable the use of SGML on the World-Wide Web. XML allow you to define your own customized markup language.