GLOSSARY

NOTES:

1. Abbreviations that appear in definitions are themselves defined in this glossary.

2. Where an abbreviation expands into multiple definitions, expansions are listed in alphabetical order.

3. For terms not included in this glossary, see the following:
– *Internetworking Terms and Acronyms* at http://www.cisco.com/univercd/cc/td/doc/cisintwk/ita/index.htm
– *Newton's Telecom Dictionary* by Harry Newton
– Tom Sheldon's Linktionary.com website at http://www.linktionary.com/linktionary.html

A

AA — Auto-attendant (or automated attendant). Software application that provides automated operator-assistance messages and prompts based on input of voice or DTMF tones, guiding callers to appropriate extensions.

AAA — Authentication, Authorization, and Accounting. Suite of network-security services that provides the framework for setting up access control on a router or gateway. See also RADIUS.

AAL — ATM adaptation layer. Service-dependent sublayer of the data link layer. The AAL accepts data from different applications and presents it to the ATM layer in the form of 48-byte ATM payload segments. AALs consist of two sublayers: the convergence sublayer (CS) and the segmentation and reassembly sublayer (SAR). AALs differ on the basis of the source-destination timing used, whether they use constant bit rate (CBR) or variable bit rate (VBR), and whether they are used for connection-oriented or connectionless mode data transfer. The four types of AAL are AAL1, AAL2, AAL3/4, and AAL5.

access server — Communications processor that connects asynchronous devices to a LAN or WAN through network and terminal emulation software. Performs both synchronous and asynchronous routing of supported protocols. Sometimes called a network access server or or universal access server.

ACD — Automatic call distributor (or distribution). Device or service that routes incoming calls to targets within a call center or geographically distributed locations served by a CO.

ACF message — Admission confirmation RAS message.

ACK message — Acknowledgment message. Message sent by one network device to another to acknowledge that some event occurred (for example, the receipt of a message).

ACOM — Loose abbreviation for "a combined loss." Combined loss achieved by an echo canceller, equal to the sum of the echo return loss, echo return loss enhancement, and nonlinear processing loss for a call.

ADSL Asymmetric digital subscriber line. One of four DSL technologies. ADSL is designed to deliver more bandwidth downstream (from the central office to the customer site) than upstream. Downstream rates range from 1.5 to 9 Mbps, whereas upstream bandwidth ranges from 16 to 640 kbps. ADSL transmissions work at distances up to 18,000 feet (5,488 meters) over a single copper twisted pair.

aggregation interval Interval during which call statistics are collected.

aggregation level Voice-signaling interface level—for example, gateway, IP, PSTN, trunk group, and voice ports (DS1 only).

AIS Alarm-indication signal. In a T1 transmission, an all-ones signal transmitted in lieu of the normal signal to maintain transmission continuity and to indicate to the receiving terminal that a transmission fault is located either at or upstream from the transmitting terminal.

Allow header Message header that lists the set of methods supported by the user agent that generates the message.

AM Amplitude modulation. Technique to modulate digital data onto high-frequency carrier tones by varying the amplitude of the signal.

AMA Automatic message accounting. Automatic collection, recording, and processing of information relating to calls for billing purposes.

ANI Automatic Number Identification (or Indication). Feature that records and sends the calling party's phone number over the network to the called party, typically for accounting and billing purposes but also for services such as enhanced 911. At the receiving end, Caller ID reads the ANI to display the caller's phone number. Often used to mean the number of the calling party.

API Application programming interface.

ARP Address Resolution Protocol. Broadcast protocol used by TCP/IP LAN-based host computers and routers to match MAC addresses with IP addresses. Operates only across a single physical network and is limited to networks supporting hardware broadcasts.

ARQ message Admission-Request RAS message.

ARS Automatic route selection. Way to provide least-cost routing.

AS Application (or access) server. Logical entity that serves a specific application instance. An example of an application server is an MGC that handles Q.931 signaling and call processing for D channels that are terminated by signaling gateways. See also access server.

ASCII American Standard Code for Information Interchange. 8-bit code for character representation (7 bits plus parity).

ASN.1 Abstract Syntax Notation One. OSI language for describing data types independent of particular computer structures and representation techniques.

ASP Application-server process. Process instance of an application server. Examples are primary or backup MGC instances.

Application-service provider (or process). Service provider that offers a unique application or service, typically hosted on another provider's core network.

ASR Automatic speech recognition. Capability of an external media server to recognize speech that is provided as input by the user of an IVR application.

ATA Analog telephone adaptor. Device that allows an analog phone to connect to an IP phone system.

ATM Asynchronous Transfer Mode. Standard for cell relay in which multiple service types (such as voice, video, or data) are conveyed in fixed-length (53-byte) cells. Fixed-length cells allow cell processing to occur in hardware, thereby reducing transit delays. ATM is designed to take advantage of high-speed transmission media such as E3, SONET, and T3.

au File extension for an audio file. See also wav.

AUCX message Audit-Connection message.

audio What the user hears on the phone. Includes both voice and DTMF digits.

AUEP command Audit Endpoint command. An MGCP or SGCP command that a call agent uses to determine the status of a given gateway endpoint.

AUEP message Audit-Endpoint message.

authentication Method for identifying users, including login and password dialog, challenge and response, messaging support, and, depending on the security protocol that you select, encryption.

AV-pair Attribute-value pair used in authentication.

B

B channel Bearer channel. DS0 time slot that carries analog voice or digital data over ISDN. In ISDN, a full-duplex, 64-kbps channel used to send user data. See also D channel.

backhaul Process in which telephony signaling is passed from a gateway to a an external media-gateway controller for processing.

backplane Physical connection between an interface processor or card and the data buses and the power distribution buses inside a chassis.

BER Bit error rate. Ratio of received bits that contain errors to the total number of transmitted bits.

BGW Business gateway.

bind In SIP and MGCP, configuring the source address for signaling and media packets to the IP address of a specific interface.

BITS Building integrated timing supply. Master timing supply for a building. Also known as a synchronization supply unit.

blind transfer Call transfer in which the transferring phone connects the caller to a destination line before ringback begins. For example, an auto-attendant uses blind transfer to redirect calls when it receives an extension number.

BOC Bell operating company.

branch A fork consists of multiple media streams. Each of these streams is referred to as a branch.

BRI Basic Rate Interface. ISDN interface composed of two B channels and one D channel for circuit-switched communication of voice, video, and data. See also ISDN Protocol and PRI.

BRQ message Bandwidth-Request RAS message.

business gateway *x*GCP media gateway. Business customer-premises equipment that connects to the VoIP network and to a user's telephony equipment (typically a PBX, corporate LAN, or WAN). Such gateways are used to eliminate or reduce the need for individual medium (voice, data, and so forth) connectivity.

buttons Physical keys on an IP phone other than those in the numbered keypad. Phone buttons are of two types: line buttons and function buttons. Line buttons can be used for extensions (ephone-dns) or for speed-dial numbers. Function buttons perform specific actions; examples are Volume, Mute, Services, Directories, and Navigation buttons. See also soft keys.

C

CA Call agent. Intelligent entity in an IP telephony network that handles call control in an MGCP VoIP network.

CAC Call-admission control. Controls whether a call can be established, according to availability of local or network resources.

call A SIP call consists of all participants in a conference who are invited by a common source and is identified by a globally unique call identifier. A point-to-point IP telephony conversation maps into a single SIP call.

call agent Intelligent entity in an IP telephony network that handles call control in an *x*GCP VoIP network. Also known as a media gateway controller.

call blocking Preventing phone users from making certain outbound calls by disallowing the dialing of specified patterns of digits.

call leg Link or hop along the route from the calling party to the destination called party. There are four call legs: leg 1 is the incoming call from PSTN to gateway; leg 2 is the originating gateway's outgoing connection to IP network; leg 3 is the incoming connection from IP network to terminating gateway; leg 4 is the outgoing call to PSTN from terminating gateway.

call manager Software-based call-processing component of an IP telephony solution that extends enterprise telephony features and functions to packet-telephony network devices such as IP phones, media-processing devices, VoIP gateways, and multimedia applications. Additional data, voice, and video services such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems interact with the solution through an API. Cisco's call manager is called Cisco CallManager.

call pickup Ability of a phone user to retrieve an incoming call from an extension on a different phone.

caller-ID blocking Feature that prevents the sending of caller-ID information.

Call-ID header Message header that uniquely identifies a particular invitation or all registrations of a particular client.

CAMA Centralized automatic message accounting. Version of AMA in which toll calls are ticketed automatically at a central location for several COs.

capabilities exchange Stage of a fax call during which the calling and called fax machines exchange messages to select compatible transmission and receipt parameters. This stage occurs after the initial voice call is established and a fax tone is sent. This stage must be completed successfully before data can be sent.

CAS Channel-associated signaling. Transmission of signaling information within the voice channel. Often is referred to as robbed-bit signaling because user bandwidth is robbed by the network for other purposes.

cause code Reason for PSTN call failure or completion.

CBR Constant bit rate. QoS class for ATM networks. Used for connections that depend on precise clocking to ensure undistorted delivery.

CBWFQ Class-based weighted fair queuing. Queueing algorithm that supports user-defined traffic classes.

CCAPI Call-control applications programming interface.

CCIT Consultative Committee for International Telegraph and Telephone. International organization responsible for the development of communications standards. Now called the ITU-T.

CCS Common-channel signaling.

CDAPI Call-distributor API.

CDR Call-detail record. A record, written to a database for use in postprocessing activities, that contains billing information for charging purposes.

CED 1. Called-station identifier. Distinctive 2100-Hz tone (on for a duration of 2.6 to 4.0 seconds) generated by a fax machine when receiving a call. CED is used in the handshaking that establishes parameters for the call.

2. Caller-entered digits. Digits entered by a caller on a touch-tone phone in response to prompts. Either a peripheral (ACD, PBX, or VRU) or the carrier network can prompt for CEDs.

CFR message Confirmation to Receive message. Message sent by a called fax machine during capabilities exchange in response to a TCF message that was received properly. Advises a calling fax machine that the modulation speed is acceptable. See also FTT message, TCF message, and training.

CHAP Challenge-Handshake Authentication Protocol. Security protocol for use on lines that use PPP encapsulation to prevent unauthorized access. Does not itself prevent unauthorized access, but merely identifies the remote end. The router or access server then determines whether that user is allowed access.

CIC Carrier-identification code. Code that identifies the network that serves a remote user.

CID AAL2 channel identifier.

CIDR Classless interdomain routing. Technique based on route aggregation that allows routers to group routes together to reduce the quantity of routing information carried by the core routers. Several IP networks appear to networks outside the group as a single, larger entity.

circuit identifier Trunk-group label or carrier ID.

Cisco RSVP Agent A Cisco IOS feature that invokes RSVP and integrates call-processing capabilities with the underlying network infrastructure to deliver call admission control and quality of service for Cisco Unified CallManager deployments. RSVP, an IETF standards-based signaling protocol for reserving resources in the IP network, is used to secure and reserve bandwidth across the WAN for calls accepted by the Cisco RSVP Agent.

CLASS Custom local-area subscriber services. Usually referred to as custom calling features.

CLEC Competitive local-exchange carrier. Company that builds and operates communication networks in metropolitan areas and provides its customers with an alternative to the local telephone company.

CLI Command-line interface. Interface that allows the user to interact with the operating system by entering commands and optional arguments. The basic Cisco IOS configuration and management interface is a CLI.

CLID Calling-line identification number. Also referred to as calling number or caller ID. Information about the billing phone number from which a call originated. The CLID value might be the entire phone number, the area code, or the area code plus the local exchange. Also called calling-line ID.

CNG 1. Calling tone or auto-fax tone. Distinctive, repeating 1100-Hz tone (on for 0.5 second, off for 3 seconds) generated by a fax machine when placing a call.

2. Comfort-noise generation.

CO Central office. Local telephone company office to which all local loops in a given area connect and in which circuit switching of subscriber lines occurs.

codec Coder-decoder. DSP device that transforms analog signals into a digital bitstream and digital signals back into analog signals. Also typically compresses and decompresses signals according to a specified algorithm.

COMET method Conditions Met method. A SIP method that indicates if the preconditions for a given call or session have been met.

complex forking Forking scenario in which at least one of the branches is transmitting using a compressed codec and at least one of the other branches is transmitting using a noncompressed (G.711) codec. See also simple forking.

conference Call in which all parties receive and send media.

configuration file See default configuration file.

connection Tying together of two streams or call legs so that the incoming voice stream of one call leg is sent as the outgoing voice stream of the other call leg.

consultative call transfer Transfer in which the transferring party either connects the caller to a ringing phone (ringback heard) or speaks with a third party before connecting the caller to the third party.

Content-Type header Message header that specifies the media type of the message body.

COR	Class of restriction. Functionality that denies certain call attempts according to the incoming and outgoing class of restrictions provisioned on the dial peers. This functionality provides flexibility in network design, allows users to block calls (for example, to 900 numbers), and applies different restrictions to call attempts from different originators. COR specifies which incoming dial peer can use which outgoing dial peer to make a call.
core router	In a packet-switched star topology, a router that is part of the backbone and serves as the single pipe through which all traffic from peripheral networks must pass on its way to other peripheral networks.
CPCS	Common-part convergence sublayer. One of the two sublayers of any AAL. Is service-independent and is responsible for preparing data for transport across the ATM network, including creation of the 48-byte payload cells that are passed to the ATM layer.
CPE	Customer-premises equipment. Terminating equipment, such as terminals, phones, and modems, supplied by the telephone company, installed at the customer sites, and connected to the telephone-company network.
crankback	Mechanism used by ATM networks when a connection-setup request is blocked because a node along a selected path cannot accept the request. The path is rolled back to an intermediate node, which attempts to discover another path to the final destination using GCAC.
CRC	Cyclic redundancy check. Error-checking technique in which the frame recipient calculates a remainder by dividing frame contents by a prime binary divisor and compares the calculated remainder to a value stored in the frame by the sending node.
CRCX message	Create Connection message. Message sent by gateways in an MGCP call sequence to exchange SDP.
CRM	Carrier resource manager.
CS	Convergence sublayer. One of the two sublayers of the AAL SSCS, which is responsible for padding and error checking. PDUs passed from the SSCS are appended with an 8-byte trailer (for error checking and other control information) and are padded, if necessary, so that the length of the resulting PDU is divisible by 48. These PDUs then are passed to the SAR sublayer of the CPCS for further processing.
CSA	H.323 call-signaling address of the gateway or endpoint.
CSeq header	Message header that identifies and orders transactions. Consists of a sequence number and a method. It uniquely identifies transactions and differentiates between new requests and request retransmissions.
CSF	Call-statistics field. Accumulative call counter or delay duration for call statistics information in a specific or periodic interval for example, number of incoming calls being answered and total post-dial-delay duration.
CSI message	Called Subscriber Identification message. Message sent by a called fax machine during capabilities exchange that provides some detail as to the identity of the called device.
CSM	Call-switching module. Card that contains digit collection logic to process incoming calls for automatic number information (ANI) and dialed number identification service (DNIS) digits.

CSR 1. Call-statistics record, composed of all defined call-statistics fields in one kind of signaling interface.

2. Carrier-sensitive routing.

CSU Channel service unit. Digital interface device that connects end-user equipment to the local digital telephone loop. Often referred to together with DSU as CSU/DSU.

customer administrator Class of user that has provisioning privileges as determined by the system administrator. Typically a customer administrator is responsible for routine phone additions and changes but is not conversant with Cisco IOS software.

D

D channel Data channel. Full-duplex, 16-kbps (BRI) or 64-kbps (PRI) ISDN channel. See also B channel.

DCN message Disconnect message. Message sent by a calling or called fax machine at the end of a fax call. On receipt of the message, the devices tear down the call and release the circuit. DCN messages do not require a response from the opposite device.

DCS Distributed Call Signaling or Digital Command Signal.

Distributed Call Signaling. A set of proposals for extending SIP.

DCS message Digital Command Signal message. Message sent by a calling fax machine during capabilities exchange that tells the called device which facilities to select for the reception of a fax transmission.

DDR Dial-on-demand routing.

default configuration file XML file that contains configuration details for IP phones. There is one shared default XML configuration file for each type of phone. When a phone goes online or is rebooted, it automatically gets information about itself from the default configuration file.

delta Incremental value, such as the difference between the current time and the time when the response occurred.

DFC Dial-feature card.

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

dial peer Addressable call endpoint that contains configuration information including voice protocol, codec type, and associated phone number. VoIP has three types of dial peers: POTS, VoIP, and MMoIP.

DID Direct Inward Dial (or Dialing). Service that allows external callers to dial an internal extension number and reach the called party without passing through an operator or attendant.

DID/DNIS Direct Inward Dial (or Dialing)/Dialed Number Identification Service. When a call arrives at an ACD or PBX, the carrier sends a digital code on the trunk line. The switch reads this code—typically the specific number dialed by the user—to determine how to dispatch the call. By mapping each possible code with an internal extension, the switch can provide DID service.

direct station select Ability to use the Transfer soft key and a monitored line button to quickly transfer a call.

DIS message Digital Information Signal message. Message sent by a called fax machine during capabilities exchange that describes the reception facilities of the called machine, such as maximum page length, scan-line time, image resolution, and error-correction mode.

disconnected procedure Procedure that the gateway may initiate when an endpoint attempts to communicate with its call agent and fails. The disconnected timer starts and RSIP messages are sent to the call agent at prescribed intervals until contact is established. This procedure ensures that an RSIP message is the first message to reach the call agent after communications are reestablished.

DN Directory number.

DND Do not disturb. Feature that causes incoming calls to not ring on an soft-key IP phone, but provides visual alerting and call information so that the call can be answered if desired.

DNIS Dialed-number identification service. Use of a called-number identifier to route a call appropriately, as when calls placed to specific toll-free numbers are routed to the appropriate area within a company to be answered. Sometimes referred to as called number.

DNIS map Table containing multiple called numbers, each individually linked to the URL of a VoiceXML document. Can reside on the gateway or an external server.

DNS Domain Name (or Naming) System. Mechanism for translating H.323 IDs, URLs, or e-mail IDs into IP addresses. Also used to assist in locating remote gatekeepers and to reverse-map raw IP addresses to hostnames of administrative domains.

DNS SRV Domain Name System server. Server that provides DNS service.

document Textual object. In HTML, documents (or pages) are single files containing HTML. In XML, documents contain markup structures and may contain content from several files.

DOD Direct Outward Dial (or Dialing).

domain 1. On the Internet, a portion of the naming-hierarchy tree that refers to general groupings of networks based on organization type or geography.

2. In security, an environment or context that is defined by a security policy, model, or architecture to include a set of system resources and the set of entities that have the right to access the resources.

Sometimes used interchangeably with ITAD. In that environment, in most cases the service provider configures the entire network to act as a single domain so that routes and routing policies are consistent across the network. Alternatively, the service provider might configure multiple domains within the network to represent major geographical regions.

DRQ message Disengage-Request RAS message.

DS0 Digital signal level 0. Single time slot on a DS1 (also known as T1) digital interface; that is, a 64-kbps, synchronous, full-duplex data channel, typically used for a single voice connection on a PBX. Alternatively, framing specification used in transmitting digital signals over a single channel at 64-kbps on a T1/E1 line.

DS3 Digital signal level 3. Framing specification for sending digital signals at 44.736 Mbps on a T3 facility.

DSL Digital subscriber line. Public-network technology that delivers high bandwidth over conventional copper wiring at limited distances. There are four types of DSL: ADSL, HDSL, SDSL, and VDSL. All are provisioned via modem pairs, with one modem located at a central office and the other at the customer site. Because most DSL technologies do not use the whole bandwidth of the twisted pair, there is room remaining for a voice channel.

DSN message Delivery Status Notification message. Message returned to the originator indicating the delivery status of an e-mail message. A sender can request three types of delivery status notifications: delay, success, and failure.

DSP Digital signal processor. Specialized electronics on a router that provide stream-to-packet signal processing functionality that includes voice compression, echo cancellation, and tone- and voice-activity detection.

DSP resource pool Collection of DSP resources available to a network module or VWIC on the router.

DSU Data service unit. Digital-transmission device that adapts the physical interface on a DTE device to a transmission facility such as T1 or E1. Also responsible for such functions as signal timing. Often referred to together with CSU as CSU/DSU.

DTD Document Type Definition. Set of rules for XML document construction. Applications use DTDs to identify the parts required of a particular document type.

DTE Data-terminal equipment. Device at the user end of a user-network interface that serves as a data source, destination, or both. Connects to a data network through a DCE device (such as a modem) and typically uses clocking signals generated by the DCE. Includes such devices as computers, protocol translators, and multiplexers.

DTMF Dual-tone multifrequency. Type of signaling that combines two distinct frequencies to generate a distinctive tone for each keypad digit or character. Sometimes referred to as “touchtone.”

DTMF-relay Mechanism for providing a reliable digit relay between VoIP gateways when a low-bandwidth codec is used. Provides a standardized means of transporting DTMF tones in RTP packets and is identified by dynamic payload types in the SDP.

dual-line ephone-dn An ephone-dn with one voice port and two channels to handle two independent calls. See also ephone-dn.

DVMRP Distance Vector Multicast Routing Protocol. Internetwork gateway protocol, largely based on RIP, that implements a typical dense-mode IP multicast scheme.

dynamic payload Payload that is defined at session initiation and is not assigned a specific RTP format. See also static payload.

E

E&M signaling	recEive and transMit (or ear and mouth) signaling. Signaling technique for two-wire and four-wire phone and trunk interfaces. One of four common forms of CAS signaling; the others are loopstart, groundstart, and EANA.
E.164	Eight-byte public-network addressing recommendation for international telecommunication numbering. Provides the number structure and functionality for the three categories of numbers used for international public telecommunication: geographic areas, global services, and networks.
E3	Wide-area digital transmission scheme, used predominantly in Europe, that carries data at a rate of 34.368 Mbps. E3 lines can be leased for private use from common carriers.
EANA signaling	Equal Access North America signaling. One of four common forms of CAS signaling; the others are loopstart, groundstart, and E&M.
ECM	Error-Correction Mode. Fax-transmission mode that provides for error-free page transmission on machines that include memory for storage of the page data (usually high-end fax machines).
EFXS	Enhanced foreign-exchange station.
element	Fundamental logical unit of an XML document. All content in a document must be contained in elements.
endpoint	Terminal or gateway that acts as a source or sink of voice data. An endpoint can call or be called, and can generate or terminate an information stream.
enterprise	Corporation with more than one location.
enterprise network	Large and diverse network that connects most major points in a company or other organization. Differs from a WAN in that it is privately owned and maintained.
ENUM	E.164 phone-number mapping.
EOL message	End of Line message. Message sent by a calling fax machine at the conclusion of each burst of line data. Six EOLs in a row constitute an RTC message that indicates the end of content transmission.
EOP message	End of Procedure message. Message sent by a calling fax machine after content transmission to indicate that page transmission is complete. Must be acknowledged with an MCF from the receiving device, after which the devices can move to the call-release phase of the call.
ephone or e-phone	Ethernet phone. Software construct that represents a physical telephone instrument or, occasionally, a port that connects to a voice-mail system. Provides the ability to configure the physical instrument using Cisco IOS software.
ephone-dn or e-phone dns	Ethernet phone-directory number. Software construct that represents the line that connects a voice channel to a phone from which a user can receive and make calls.
ERL	Echo-return loss.
ERLE	Echo-return-loss enhancement.
ESMTP	Extended Simple Mail Transfer Protocol. Extended version of SMTP that includes additional functionality such as delivery notification and session delivery.

E-TRIP TRIP communication between location servers within different ITADs.

ETSI European Telecommunications Standards Institute.

execution instance Instance of the Tcl interpreter that is created to execute a script.

extension Common equivalent for ephone-dn. Also the short form of extension number, which is the sequence of digits that is used to dial an ephone-dn from another ephone-dn in the same system.

F

failover Type of switchover that occurs when a gateway registers to a backup device because the currently active device to which it is registered has failed. Also applies to the return to service of a previously unavailable process.

fallback Type of switchover that occurs when rigorous path selection does not generate an acceptable path and a gateway attempts to determine a path by selectively relaxing certain attributes, such as delay, in order to find a path that meets some minimal set of desired attributes.

fax Transmission of an image, drawing, or document over a distance by converting it into coded electrical signals at the originating end, passing the signals from the originator to the receiver over a transmission medium, and converting the signals into a replica of the original at the receiving end. Derived from “facsimile.”

fax detection IVR application that supports the use of a single E.164 number for both voice mail and fax mail by providing the capability to detect automatically whether an incoming call is voice or fax.

fax fallback Alternate mechanism for transferring a fax across the VoIP network if T.38 fax relay cannot be dynamically negotiated.

fax mail SMTP or ESMTP e-mail message that is produced by a T.37 on-ramp gateway or an e-mail message sent to a T.37 off-ramp gateway.

fax passthrough Feature that provides repression of compression, echo cancellation, and other functions; issues redundant packets to ensure complete transmission; and provides a buffer to protect against clock skew.

fax relay Fax-transmission method in which the following occurs: An originating fax machine sends an analog signal to the originating fax gateway, which demodulates the signal into digital form, packetizes it, and sends it over the IP network. The terminating fax gateway remodulates the digital signal into analog form and sends it to the terminating fax machine.

fax rollover IVR application in which the on-ramp gateway receives fax calls at an E.164 number and attempts to route the calls using fax relay. If the attempt fails, the call is forwarded to an SMTP server by a mail-transfer agent using T.37-standard protocols for store-and-forward fax.

FCS Frame-check sequence. Extra characters added to a frame for error-control purposes. Used in HDLC, Frame Relay, and other data-link-layer protocols.

FEAC Far-end alarm code.

feature ring Audible alert with a triple-pulse cadence that differentiates incoming calls on one line from incoming calls on other lines on the phone.

FGD	Feature Group-D. Standardized service available to carriers, delivered on a channelized T1 line.
FGD-OS Protocol	Feature Group-D Operator Services Protocol. OS is a telephony-signaling protocol for calls that originate from a Bell operating company and are sent to the carrier switch.
final-recipient	User agent that is introduced into a call with the recipient.
FM	Frequency modulation. Technique to modulate digital data onto high-frequency carrier tones by varying the signal frequency.
fork	All of the media streams (branches) that are generated and transmitted by an endpoint. Only one party (the controller) receives the media from all of the other parties. The other parties receive media from the controller.
forking	Transmit function of the endpoint. Used to forward a request in several directions simultaneously—for example, to find a user who may be at one of several locations.
FPGA	Field-programmable gate array. A programmable memory device.
FQDN	Fully qualified domain name. Complete domain name including the host portion—for example, serverA.companyA.com.
Frame Relay Protocol	Switched-data link-layer protocol that handles multiple virtual circuits using HDLC encapsulation between connected devices. More efficient than X.25, the protocol that it generally replaces.
FRU	Field-replaceable unit.
FSK	Frequency-shift keying. Type of frequency modulation in which the modulating signal shifts the output frequency between predetermined values.
FSM	Finite-state machine.
FTP	File Transfer Protocol. Application protocol, part of the TCP/IP protocol stack, for transferring files between network nodes.
FTT message	Failure to Train message. Message sent by a called fax machine during a fax-capabilities exchange in response to a TCF message that was not received properly. Advises the calling machine that the modulation speed is not acceptable. See also CFR message, TCF message, and training.
fx: extension	Extension of the local-connection option used by the call agent to instruct the gateway to be in CA-controlled mode or GW-controlled mode.
FXO	Foreign-exchange office (or operator). Interface, on a standard phone, that connects to the PSTN central office. The 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	1. Foreign-exchange station. An FXS interface connects directly to a standard phone and supplies ring, voltage, and dial tone. The Cisco FXS interface is an RJ-11 connector that allows connections to basic telephone-service equipment, keysets, and PBXs. 2. Foreign-exchange subscriber (as in FXS-loop-start and FXS-ground-start). FXS loop-start and ground-start signaling are used to indicate the beginning of a call. The FXS interface is used for line-side access to the CO (for example, communication between CO and key system).

G

G.711 64-kbps PCM voice-coding technique in which encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs.

G1 fax Group 1 fax transmission. First standard for faxes, introduced in 1968. Superseded by G3 fax.

G2 fax Group 2 fax transmission. Second standard for faxes, which improved on G1 fax transmission speed and resolution but was superseded by G3 fax.

G3 fax Group 3 fax transmission, also called traditional fax. Most commonly used standard for fax transmission over circuit-switched networks in the PSTN today.

G4 fax Group 4 fax transmission. Standard for digital phone lines such as ISDN that operate at 64 kbps.

gatekeeper Entity on a LAN that provides address translation, access control, and bandwidth management. Operates by maintaining a registry of devices in the multimedia network. Devices register with the gatekeeper at startup and request admission to a call. Sometimes designated GK.

gateway Device that performs application-layer conversion of information from one protocol stack to another. Allows SIP or H.323 terminals to communicate with terminals configured to other protocols—for example, encodes a circuit-switched call and repackages it into IP packets. Sometimes designated GW.

GCAC Generic Connection Admission Control. In ATM, a PNNI algorithm designed for CBR and VBR connections. Any node can use GCAC to calculate the expected CAC behavior of another node if given that node’s advertised link metrics and the QoS of a connection setup request.

GKTMP Gatekeeper Transaction Message Protocol. Proprietary Cisco protocol that allows a third-party application to influence the operation of a gatekeeper.

groundstart signaling One of four common forms of CAS signaling; the others are loopstart, EANA, and E&M.

GTD Generic transparency descriptor. GTD objects are generically used to represent ISUP messages, parameters, and R2 signals. GTD objects are encapsulated into existing signaling protocols (for example, H.225), facilitating end-to-end transport.

GUI Graphical user interface. User environment that uses pictorial and textual representations of the input and the output of applications and the hierarchical or other data structure in which information is stored. Such conventions as buttons, icons, and windows are typical, and many actions are performed using a pointing device (such as a mouse).

GUID Globally unique identifier that identifies end-to-end calls. A new GUID is assigned to each new call. There may be numerous calls within a single session. Also called h323-conf-id.

H

H.323 Standard for packet-based video, audio, and data conferencing. Umbrella standard that describes the architecture of a conferencing system and refers to other standards (H.245, H.225.0, and Q.931) to describe its actual protocols. Defines a common set of codecs, call setup and negotiating procedures, and basic data-transport methods that allow dissimilar communication devices to communicate with each other.

H.323 RAS signaling H.323 registration, Admission, and Status signaling. Performs registration, admissions, bandwidth changes, status, and disengage procedures between a VoIP gateway and a gatekeeper.

H.450 Component of the H.323 standard that defines signaling and procedures used to provide telephony-like services. H.450 includes other standards such as H.450.2 (call forwarding) and H.450.3 (call transfer).

hairpinning Call-routing capability in which an incoming call on a specific gateway is signaled through the IP network and back out the same gateway.

handshake Sequence of messages that are exchanged between two or more network devices to ensure transmission synchronization.

HCM High-performance voice-compression module.

HDLC Protocol High-Level Data-Link Control Protocol. Bit-oriented synchronous data-link layer protocol developed by ISO. Derived from SDLC, HDLC specifies a data-encapsulation method on synchronous serial links using frame characters and checksums.

hookflash Short on-hook period usually generated by a phone-like device during a call to indicate that the phone is attempting to perform a dial-tone recall from a PBX or switch. Hookflash is often used to initiate call transfers.

HS High speed. Fax-transmission rate used for capabilities negotiation. See also LS.

HSRP Hot-Standby Router Protocol. Protocol that provides network redundancy for IP networks, ensuring that user traffic immediately and transparently recovers from first-hop failures in network edge devices or access circuits.

HSSI High-Speed Serial Interface. Network standard for high-speed (up to 52 Mbps) serial connections over WAN links.

HTML Hypertext Markup Language. The markup language used for today's web applications, providing formatting and basic document structure for presentation via browser applications.

HTTP Hypertext Transfer Protocol. Protocol used by web browsers and web servers to transfer files.

hunt group Number of phone lines that are associated together by the central office or PBX. When a call comes in to a hunt group, it cycles through the group of lines until it finds one that is not busy, then it rings that phone (or extension on a PBX system).

hunting Automatic redirecting of an incoming call from a called number that is busy or does not answer through a group of designated numbers in a designated order until the call is answered. If all the lines are busy, the caller gets a busy signal.

huntstop Ability to specify that calls should not hunt if the number that is being rung is busy or does not answer.

I

IAB Internet Architecture Board. Board of internetwork researchers who discuss issues pertinent to Internet architecture.

idle message Display that appears on an IP phone when it is not in use.

IE Information element. In ATM, portion of a signaling packet that carries information such as addresses.

IEEE Institute of Electrical and Electronics Engineers. Professional organization whose activities include the development of communications and network standards. IEEE LAN standards are the predominant LAN standards today.

IETF Internet Engineering Task Force. Task force that consists of over 80 working groups responsible for developing Internet standards. Operates under the auspices of the Internet Society.

IFP Internet Fax Protocol. Protocol used for T.38 packets.

ILEC Incumbent local-exchange carrier.

ILMI Interim Local Management Interface. Specification for incorporating network-management capabilities into the ATM user-network interface.

IMA Inverse multiplexing over ATM. Standard that provides economical, high-bandwidth ATM WAN access by enabling multiple T1 or E1 ATM links to be combined and to appear as one physical link to higher layers.

in-band signaling Transmission of signaling information within the same frequency range that is used for transmission of data.

intercom Pair of dedicated extensions that are configured to speed-dial each other when they are accessed, to provide a private, two-way speech path.

interdomain routes Routes that are exchanged between location servers in different ITADs.

intradomain routes Routes that are exchanged among location servers within a single ITAD.

INVITE request Message that initiates a SIP session. Indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides information regarding the capabilities of the called and calling parties.

IP Internet Protocol. Connectionless protocol at the network layer (Layer 3) of the OSI reference model. Provides features for addressing, type-of-service specification, fragmentation and reassemble, and security. Works with TCP and is usually identified as TCP/IP. See TCP/IP.

IPDC Internet Protocol Device Control. A device-control specification.

IPSec IP Security. Framework of open standards that provides for secure transmission of sensitive information between participating gateway and host peers over unprotected networks such as the Internet. Acts at the network layer.

IRR message Information request response RAS message.

ISDN Protocol Integrated Services Digital Network Protocol. Protocol that permits networks to carry data, voice, and other source traffic. See also BRI and PRI.

ISO International Organization for Standardization. International organization that is responsible for a wide range of standards, including those relevant to networking. ISO developed the OSI reference model, a popular networking reference model.

ISOC Internet Society. International organization that coordinates the evolution and use of the Internet.

ISP Internet service provider.

ISUP ISDN User Part. Provides interexchange signaling to support SS7 trunks that are set up for switched voice and data applications in an ISDN environment.

ITAD IP Telephony Administrative Domain. Set of resources (such as gateways and location servers) under control of a single administrative authority.

I-TRIP TRIP communication between location servers within an ITAD.

ITSP Internet telephony service provider. VoIP-based network provider that offers telephony service or voice transport. Also called an ISP.

ITU International Telecommunication Union. An international organization within the United Nations System of Organizations in which governments and the private sector coordinate global telecommunication networks and services. Formerly known as CCITT.

ITU-T International Telecommunication Union Telecommunication Standardization Sector. International body that develops worldwide standards for telecommunications technologies. Carries out the functions of the former CCITT.

IVR Interactive voice response. System that collects information from a user to process commands—for example, to retrieve voice mail, bank-account balances, or stock quotes. The system prompts the user by means of recorded IVR-script messages; the user responds by entering DTMF digits or by speaking words.

IXC Interexchange carrier. Service provider who is focused traditionally on long-distance voice transport.

IZCT Interzone ClearToken. Packet of information about a call that circulates between gatekeepers and between gatekeeper and gateway to route a call. Each successive gatekeeper stamps the IZCT's destination-gatekeeper ID with its own ID and sends the IZCT back to the originating gateway. The originating gateway passes the IZCT to the terminating gateway listed in the SETUP message. The terminating gatekeeper forwards the IZCT to the terminating gatekeeper, which then validates it.

J

jitter Interpacket delay variance. The difference between interpacket arrival and departure. Jitter is an important QoS metric for voice and video applications.

JSE JavaScript expression.

K

keepalive interval Period of time between keepalive messages sent by a network device.

keepalive message Message sent by one network device to inform another network device that the virtual circuit between the two is still alive.

keyswitch, keyswitch model In traditional telephony systems, a multibutton phone with several lines. Used on a site where the same few lines appear on all phones and are shared among all phones.

L

LAN Local-area network. High-speed, low-error data network covering a relatively small geographic area (up to a few thousand meters). LANs connect workstations, peripherals, terminals, and other devices in a single building or other geographically limited area. LAN standards specify cabling and signaling at the physical and data-link layers of the OSI reference model. Ethernet, FDDI, and Token Ring are widely used LAN technologies.

latency Delay between the time a device requests access to a network and the time it is granted permission to transmit.

Layer 1 Physical layer of the OSI reference model. Responsible for the electric signal being sent and received. This can be viewed as a bit stream coming in, and going out, of the system. Scope must be considered when using this term. For example, Layer 1 on a T1 is 1.544 Mbps but Layer 1 on a DS-0 time slot in the T1 is 64 kbps.

Layer 2 Data-link layer of the OSI reference model. Responsible for point-to-point delivery of a PDU. Layer 2 protocols have two basic classes: reliable (meaning delivery is guaranteed or an error is reported) and unreliable (meaning delivery might not occur with no indication to the upper layers).

Layer 3 Network layer of the OSI reference model. Responsible for the network routing and delivery of messages. Examples of Layer 3 protocols include X.25 and IP. Q.931 is not considered a Layer 3 protocol because it does not route or deliver messages.

LCF message Location-Confirmation RAS message.

LCS Line-control signaling. Transport of line-supervision signals in the media stream using NTEs.

LDAP Lightweight Directory Access Protocol. Protocol that provides access for management and browser applications that provide read/write interactive access to the X.500 Directory.

LEC Local-exchange carrier. Telephone company that provides customer access to the PSTN through one of its central offices.

leg 1 Call segment between the PSTN and the originating gateway (see also call leg).

leg 2 Call segment between the PSTN and the originating gateway (see also call leg).

leg 3 Call segment between the originating gateway and the IP network (see also call leg).

leg 4 Call segment between the terminating gateway and the PSTN (see also call leg).

LEX LAN extender (interface).

local route Routes injected into TRIP at a location server are considered local routes by that location server. These routes may be injected by TGREP or by manual means. Routes received from another TRIP peer through I-TRIP or E-TRIP are not considered local routes.

local speed dial Speed-dial numbers that are common to all phones and that are accessed through the Directories button on a phone. See also personal speed dial.

location server Server that provides location services. A SIP redirect or proxy server uses a location service to get information about a caller's locations.

long pound calls Calls in which the user presses the pound (#) key to make a new call—typically while making calls using a charge or debit card—without having to enter the card number or having to wait to be authenticated again.

loopstart signaling One of four common forms of CAS signaling; the others are groundstart, EANA, and E&M.

LRJ message Location Reject RAS message.

LRQ message Location Request RAS message.

LS 1. Location server. A centralized resource (such as a SIP proxy or redirect server or an H.323 gatekeeper) that identifies the next hop for the VoIP connection and communicates that to the originating gateway or terminal.

2. Low speed. Fax-transmission rate used for capabilities negotiation, defined as 300 baud. See also HS.

LT-S Subscriber-line termination.

LT-T Trunk-line termination.

M

MAC Media Access Control. Lower of the two sublayers of the data-link layer. Handles access to shared media, such as whether token passing or contention is to be used.

markup In an XML environment, structural information that is stored in the same file as the content.

MARS 1. Modular access routers.

2. Multicast-address-resolution server. Server that supports IP multicast by serving a group of nodes (known as a cluster), each of which is configured with the ATM address of the MARS.

MCF message Message Confirmation message. Message sent by a called fax machine in response to a PPS signal during the control-mode phase after content transmission.

MDCX message Modify Connection Request message. Message sent by gateways in an MGCP call sequence to exchange SDP information.

MDL Message Definition Language. High-level language for specifying protocols and protocol conversion operations on a virtual switch controller.

MDL message Maintenance Data Link message.

MDN message Message Disposition Notification message. Message returned to the originator of an e-mail message indicating that the e-mail message has been opened.

media gateway Emerging industry-standard generic term for a gateway. Equipment that connects the PSTN or a PBX with the VoIP network and is controlled by an MGCP call agent. Sometimes abbreviated MG.

MF Multifrequency or multiple frequency. Interoffice-address-signaling method in which 10 decimal digits and five auxiliary signals are each represented by a pair of the following frequencies: 700, 900, 1100, 1300, 1500, and 1700 Hz.

MGC Media gateway controller. Provides call control capability to handle signaling traffic from a variety of sources. It also manages connections and resources of its media gateways. Can also be called a call agent.

MGCP Media Gateway Control Protocol. Protocol that enables media gateway controllers and media gateways to communicate for call control on VoIP networks.

MIB Management Information Base. Database of network-management information. MIB objects are organized in a tree structure that includes public (standard) and private (proprietary) branches. An object may be, for example, a counter or a protocol status. The value of a MIB object can be changed or retrieved using SNMP or CMIP commands, usually through a network-management-system GUI. An object identifier uniquely designates any point in the tree, whether leaf object or branch point. An object identifier may be expressed as either a text string (for human use) or a numeric string (for machine use). Technically, the text form is the object descriptor and the numeric form is the object name. In practice, either is usually called an object identifier or OID. See also OID.

MIME Multipurpose Internet Mail Extension. Standard for sending nontext data (or data that cannot be represented in plain ASCII code) in Internet mail, such as binary, nonEnglish-language text (such as Russian or Chinese), audio, or video data. Store-and-Forward Fax uses image/tiff MIME content type.

minimum timer Configured minimum value for session interval accepted by SIP elements (proxy, UAC, UAS). This value helps minimize the processing load from numerous INVITE requests.

Min-SE Minimum session expiration. Minimum value for session expiration.

MIX Multiservice interchange. Feature that allows TDM connections between MIX-enabled ports.

MMoIP dial peer Multimedia-Mail-over-IP dial peer. Dial peer used for Store-and-Forward Fax and VoiceXML Voice Store and Forward. Means by which you assign particular line characteristics (such as a destination phone number) to the connection between the router or access server and the SMTP mail server.

MOH Music on hold. Feature that provides real-time or recorded sound to callers while they are on hold. Can be provided from an audio file or from a live feed from an external source.

monitor lamp, monitor mode Line button on a phone that is used to display the line status (in-use or free) of an extension as it appears on another phone. The extension cannot be used on this phone to make or receive calls.

MRCP Media Resource Control Protocol. Application-level protocol developed by Cisco and its media server partners Nuance Communications and SpeechWorks International. Used by client devices that process audio or video streams to control media resources on external media servers such as speech synthesizers and speech recognizers. The gateway that runs a voice application and the media servers that provide speech recognition and speech synthesis maintain a client/server relationship through an RTSP connection.

MSN Microsoft Network.

MTA Mail-transfer agent. Software that implements SMTP and provides storage for mail messages to be forwarded or delivered to a local user.

multicast Process of transmitting PDUs from one source to many destinations. The actual mechanism (that is, IP multicast, multi-unicast, and so forth) for this process might be different for LAN technologies.

multipoint-unicast Process of transferring PDUs where an endpoint sends more than one copy of a media stream to different endpoints. This can be necessary in networks which do not support multicast.

MWI Message-waiting indication. Notification, such as a light on the phone, from a voice-mail system that a message has been left for the owner of the phone on which the notification was left.

N

NAM Network applications management. The ICM in a two-tier service bureau architecture that receives route requests from the carrier network and forwards them to a customer ICM.

NAPTR Naming-authority pointer. Algorithm for converting an E.164 number (such as 14085551212) into a fully qualified domain name (such as 2.1.2.1.5.5.5.5.8.0.4.1.e164.arpa).

NAS Network access server. Server that connects asynchronous devices to a LAN or WAN through network and terminal-emulation software. Performs both synchronous and asynchronous routing by means of supported protocols.

NCP 1. Network control point. Process within the AT&T signaling network that sends routing requests to a customer routing point such as the network-interface controller (NIC) within the ICM.

2. Network Control Program. SNA program that routes and controls the flow of data between a communications controller (in which it resides) and other network resources.

3. Network Control Protocol. Series of protocols for establishing and configuring different network layer protocols, such as for AppleTalk over PPP. See also PPP.

NCS Network-based call signaling for residential gateways.

never-busy fax See fax rollover.

NFAS Nonfacility-Associated Signaling. ISDN service that allows a single D channel to control multiple PRI interfaces, thus freeing one B channel on each interface to carry other traffic.

NFAS group PRI channel group (the group of interfaces) that is under control of a single D channel. Can include all of the ISDN channels on multiple T1 controllers.

NFAS member PRI interface in an NFAS group. For example, an NFAS group might include serial interfaces 1/0:23, 1/1:23, and 2/0:23 if T1 controllers 1/0, 1/1, and 2/0 are configured for NFAS.

night service Service that provides, during specified hours, notification to designated phones that a particular extension on a different phone is ringing.

NLP Nonlinear processor.

NM Network module. Hardware component that stores application software.

NMS Network management system. System responsible for managing at least part of a network. Communicates with agents to help keep track of network statistics and resources.

node H.323 entity that uses RAS to communicate with the gatekeeper, for example, an endpoint such as a terminal, proxy, or gateway.

NOTIFY message NOTIFY message. Message that reports when certain events, such as DTMF events, occur in a SIP system.

NSE Named signaling (or service) event. Format of RTP packets used for applications such as modem relay and fax relay. NSEs have different payload values than NTEs. See also NTE.

NSF 1. Network-specific facilities.

2. Nonstandard facilities. Capabilities that fax manufacturers have built into fax machines to distinguish their products from others. Also a message sent by a called fax machine during capabilities exchange to advise the calling device of nonstandard facilities that are available on the called fax machine.

NSS message Nonstandard-Facilities Setup message. Message sent by a calling fax machine during capabilities exchange in response to an NSF message. The NSS message selects the extra reception parameters from the NSF message that the calling fax machine wants to use for this call.

NT Network termination. Device that provides the interface between customer-premises equipment and central-office switching equipment.

NTE Named telephony event. Event such as DTMF digits that must be encoded and transported in an RTP packet. Not all gateways use protocols capable of performing this encoding (SCCP, for instance, has no such capability). See also NSE.

NTP Network Time (or Transfer) Protocol. Protocol for synchronizing a router to a single clock on the network, known as the clock master.

O

off-ramp gateway In the Store-and-Forward Fax application, a gateway that provides an exit from the packet network to the PSTN or nonpacket device.

OGK Originating gatekeeper.

OGW Originating gateway. Voice gateway that handles calls entering the IP network.

OID Object identifier. Values are defined in specific MIB modules. The Event MIB allows a user or an NMS to watch over specified objects and to set event triggers based on existence, threshold, and boolean tests. An event occurs when a trigger is fired; this means that a specified test on an object returns a value of true. To create a trigger, a user or an NMS configures a trigger entry in a table in the Event MIB that specifies the OID of the object to be watched. For each entry type, corresponding tables (existence, threshold, and boolean tables) are populated with the information required to carry out the test. A MIB can be configured so that, when triggers are activated (fired), either an SNMP set is performed, a notification is sent out to the interested host, or both. See also MIB.

OIR Online insertion and removal. Feature that permits the addition, replacement, or removal of cards without interrupting the system power, entering console commands, or causing other software or interfaces to shut down.

on-ramp gateway In the Store-and-Forward Fax application, a gateway that provides an entrance to the packet network from the PSTN or a nonpacket device.

originator User agent that initiates a Transfer Request or Refer Request message with the recipient.

OSI reference model Open Systems Interconnection reference model. Seven-layer reference model and framework of standards for communication in a multisystem environment.

OSP Open Settlement Protocol. Client/server protocol for establishing authenticated connections between gateways and for allowing gateways and servers to transfer accounting and routing information securely. Allows service providers to roll out VoIP services without establishing direct peering agreements with other ITSPs.

OSPF Open shortest path first. Link-state, hierarchical routing algorithm. Features include least-cost routing, multipath routing, and load balancing.

OSS Operations support system.

overlay Placement of up to 10 ephone-dns on a single phone button. An overlay set is a group of ephone-dns that are overlaid onto a single phone button.

P

package Set of signals and events that define a type of voice endpoint or connection. Examples include line-package, trunk-package, dtmf-package, and atm-package.

PAD Packet assembler/disassembler. Device used to connect simple devices (like character-mode terminals) that do not support the full functionality of a particular protocol to a network. PADs buffer data and assemble and disassemble packets sent to such end devices.

paging Ability to make a one-way audio announcement to phones in a previously specified paging group.

pass-through See fax pass-through.

payload type Number from 96 to 127 that identifies the type of payload carried in the packet (for example, a payload type of 121 denotes a DTMF relay payload; a payload type of 122 denotes a fax payload). Should be identical on the gateway and call agent.

Payload-Type header Message header that defines the content, format, and encoding scheme of the RTP payload for a SIP call.

PBX Private branch exchange. Privately owned local digital or analog telephone switchboard office, either manually or automatically operated, that serves extensions in a business complex and provides access to the public network.

PBX model Private branch exchange model. A system configuration patterned after a traditional telephony site with a PBX in which most phones have a single, unique extension number.

PCM Pulse-code modulation. Technique of encoding analog voice into a 64-kbit data stream by sampling with eight-bit resolution at a rate of 8000 times per second.

PCR Preventive cyclic retransmission. Basic SS7 Message Transfer Part Layer 2 functionality that is used when signals are transmitted over satellite connections between a Cisco signaling-link terminal and a signal-transfer point.

PDU Protocol data unit. Another name for packet. Used by bridges to transfer connectivity information.

peer ephone hunt group Hunt group in which ephone-dns ring in the left-to-right order in which they were listed when the hunt group was defined. The first ephone-dn to ring is the number in the list to the right of the ephone-dn that was the last to ring when the hunt group was last called. Ringing proceeds in a circular manner, left-to-right, for the number of hops specified when the ephone hunt group is defined. See also sequential ephone hunt group.

personal speed dial Speed-dial numbers that are specific to an individual phone and are accessed through the Directories button on the phone. See also local speed dial.

pickup See call pickup.

pilot number Extension number that is the single number that is called to access a particular service, such as paging or ephone-hunt. The number is for an ephone-dn that is not associated with any phone. Calls to the pilot number are redirected to actual ephone-dns according to configuration instructions.

PIM Protocol-independent multicast. Multicast routing architecture that allows the addition of IP multicast routing on existing IP networks.

PIN Personal identification number.

PINX Private integrated services network exchange. PBX or key system in a BRI voice application that uses QSIG signaling.

PISN Private Integrated Services Network. A private ISDN that provides services to a specific set of users (contrary to public ISDN, which provides services to the general public).

PLAR Private line, automatic ringdown. Leased voice circuit that connects two single endpoints. When either telephone handset is taken off-hook, the remote telephone automatically rings.

PLL Phase-locked loop. Electronic circuit that synchronizes itself to an external reference signal. It locks onto the phase or the average frequency of an incoming signal, dynamically tracks it, and outputs a related but more useful version. Typical applications include synchronizing a system to a single clock source and jitter filtering (removing phase noise).

PM Phase modulation.

PNNI Protocol	Private Network-Network Interface Protocol. ATM routing protocol for distributing topology information between switches and clusters of switches that is used to compute paths through the network. Based on well-known link-state routing techniques and includes a mechanism for automatic configuration in networks in which the address structure reflects the topology.
POP	Point of presence. Physical location where an interexchange carrier has installed equipment to interconnect with a local exchange carrier.
POTS	Plain old telephone service. Basic phone service that supplies standard single-line phones, telephone lines, and PSTN access.
POTS dial peer	Dial peer that is connected via a traditional telephony network. Points to a particular voice port on a voice network device.
PPM	Port policy management. Handling of gateway port resources based on configured parameters that enforce specified policies.
PPMS	Port policy management server.
PPP	Point-to-Point Protocol. Successor to SLIP that provides router-to-router and host-to-network connections over synchronous and asynchronous circuits. Whereas SLIP works with IP, PPP works with several network layer protocols, such as IP, IPX, and ARA. PPP also has built-in security mechanisms, such as CHAP and PAP. PPP relies on two protocols: LCP and NCP.
PPR message	Partial Page Request message. Message sent by a called fax machine during content transmission when ECM is being used. A PPR is sent when HDLC blocks of T.4 page data are not received error-free. The PPR lists the frames that were not received and asks for them to be retransmitted.
PPS message	Partial Page Signal message. Message sent by a calling fax machine during the control mode phase after content transmission if ECM is being used and a partial page is being resent. Must be confirmed by an MCF from the receiving device.
PRACK message	Provisional Acknowledgement message. Message that allows reliable exchanges of SIP provisional responses between SIP endpoints.
preauthentication	Feature that allows a gateway to accept or reject a call before it is connected on the basis of information associated with the call, such as DNIS, CLID, or call type (also referred to as the bearer capability).
preference	Order in which a dial peer associated with an ephone-dn is selected if there are other dial peers that also match the dialed string.
PRI	Primary Rate Interface. ISDN interface composed of a single 64-kbps D channel plus 23 (T1) or 30 (E1) B channels for circuit-switched communication of voice, video, and data. See also BRI and ISDN protocol.
provisional response	Informational response that is often used for responding to an INVITE request by providing information on call progress. Reliability is not guaranteed when delivered over UDP.
provisioning	Act of allocating phones, phone features, and phone services to individuals at a customer site.
proxy	SIP UAC or UAS that forwards requests and responses on behalf of another SIP UAC or UAS.

proxy server Intermediary program that acts as both server and client for the purpose of making requests on behalf of other clients within a SIP network. 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.

PSK Phase-shift keying. Technique for modulating digital data onto high-frequency carrier tones that shifts the period of a wave.

PSTN Public switched telephone network. General term referring to the local telephone company or to the variety of telephone networks and services in place worldwide. Sometimes called POTS.

PTT Post, Telephone and Telegraph. European national version of an ILEC.

PVC Permanent virtual circuit (or connection). Virtual circuit that is permanently established. PVCs save bandwidth associated with circuit establishment and teardown in situations where certain virtual circuits must exist all the time. In ATM terminology, a PVC is called a permanent virtual connection.

PVDM Packet voice/data module. DSP that handles both voice and data.

Q

QAM Quadrature amplitude modulation. Technique for modulating digital data onto high-frequency carrier tones that is a combination of amplitude modulation and phase-shift keying.

QoS Quality of service. Measure of performance for a transmission system that reflects its transmission quality and service availability. QoS refers to the ability of a network to provide better service to selected network traffic over various underlying technologies. QoS is not inherent in a network infrastructure; rather, you must institute QoS by strategically deploying features that implement it throughout the network.

Q-series recommendations ITU-T standards that define switching and signaling. Well-known Q-series recommendations include Q,921 for the ISDN data-link layer and Q.931 for out-of-band signaling in ISDN connections.

QSIG Q (point of the ISDN model) Signaling. Common-channel signaling standard based on Q.931 and used by many digital PBXs.

R

RADIUS Remote Authentication Dial-In User Service. Database for authenticating modem and ISDN connections and for tracking connection time. Performs access control—that is, authentication, authorization, and accounting. RADIUS clients run on routers and send authentication requests to a central RADIUS server that contains all user-authentication and network-service access information. See also AAA.

RADIUS-based PPM server Server, either a Cisco RPMS or a third-party product, that manages port policy in conjunction with RADIUS and AAA.

RAI message Resources Available Indicator message.

RAS Protocol Registration, Admission, and Status Protocol. Protocol that is used between endpoints and a gatekeeper to perform management functions. See also H.323 RAS signaling.

RBS Robbed-bit signaling.

real-time fax See fax relay.

recipient User agent that receives a Refer request from the originator and is transferred to the final-recipient.

redirect server 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.

redirected call Call that is redirected by the switch or the gateway to another destination, such as a voice-mail box.

registrar SIP server that accepts REGISTER requests. A registrar is typically colocated with a proxy or redirect server and may offer location services.

re-INVITE request INVITE request that is sent on an active call leg.

rejected call Call that is dropped by the switch or the gateway because of insufficient resources or no available routes to the destination.

reliable provisional response Provisional response that guarantees reliability.

Request-URI Request uniform (or universal) resource identifier. SIP or a general URI that indicates the user or service to which a request is addressed.

RES message ISUP Resume message.

reset (phone) Full reboot of an IP phone, similar to a power-off, power-on cycle. DHCP and TFTP servers are contacted for their latest updates. Takes longer than a phone restart but must be used after changes to phone firmware, user locale, network locale, and URL parameters.

residential gateway *x*GCP media gateway (customer-premises equipment) that has one or more connections to the VoIP network and one or more connections to user telephony equipment. Sometimes designated RGW.

restart (phone) Quick reboot of an IP phone. DHCP and TFTP servers are not contacted for their latest updates. Takes less time than a phone reset and should be used after changes to phone buttons, phone lines, and speed-dial numbers.

Resume message ISDN Resume message.

RFC Request for Comments. Document series that describes, among other things, the Internet suite of protocols and related experiments.

RGW See residential gateway.

RIB Routing information base. Local routing table, maintained by the location server to reflect reachability and route attributes learned via routing-protocol updates.

ringback Signal used by an operator at the receiving end of an established connection to recall an operator at the originating end.

RIP Routing Information Protocol. Protocol that uses hop count as a routing metric.

RLM Redundant link manager. Software that provides virtual-link management over multiple IP networks so that Q.931 and other proprietary protocols can be transported on top of multiple redundant links between a Cisco signaling controller and a NAS. Opens, maintains, and closes multiple links, manages buffers of queued signaling messages, and monitors whether links are active for link and signaling-controller failover.

RLT Release-link trunking. Traditional PSTN signaling used by applications to initiate call transfer over a CAS trunk.

rollover See fax rollover.

route aggregation Summarization of routes that fall within a class-based network boundary or within a class-less (CIDR) prefix length into a single announcement from the TRIP speaker.

route server Generic term for server based applications designed to provided advanced routing logic within VoIP networks. Cisco route servers include CSR, ARS, and NAM.

router Network-layer device that uses one or more metrics to determine the optimal path along which network traffic should be forwarded. Forwards packets from one network to another based on this network-layer information. Occasionally called a gateway.

RRQ message Registration Request RAS message.

RSC Router switch controller.

RSIP command Restart in progress command. MGCP or SGCP command used to indicate that a span (or collection of spans) has come into service, has gone out of service, or is about to go out of service.

RSVP Resource Reservation Protocol. Protocol that supports the reservation of resources across an IP network. Applications running on IP end systems can use RSVP to indicate to other nodes the nature (bandwidth, jitter, maximum burst, and so on) of the packet streams they want to receive. RSVP depends on IPv6. Also known as Resource Reservation Setup Protocol.

RSVP Agent See Cisco RSVP Agent.

RTC message Return to Control message. Message sent by a calling fax machine during content transmission to signify the end of content transmission. The RTC message consists of six consecutive EOLs.

RTCP Real-Time Control Protocol. Protocol for monitoring an RTP connection and conveying information about an ongoing session.

RTP Real-Time Transport (or Routing Table) Protocol. Protocol, commonly used with IP networks, that provides end-to-end network transport functions for applications that send real-time data, such as audio, video, or simulation data, over multicast or unicast network services. Provides such services as payload type identification, sequence numbering, time stamping, and delivery monitoring to real-time applications.

RTP-NTE Real Time Transport Protocol-Named Telephony Event. Prevents generation of spurious digits at the receiving gateway. Must be used when DTMF-relay is configured in a forked call.

RTPvt Release-to-Pivot. Call-redirect feature for ISUP networks.

RTSP Real-Time Streaming Protocol. Protocol that enables controlled delivery of real-time data, such as audio and video. Sources of data can include both live data feeds, such as live audio and video, and stored content, such as prerecorded events. Designed to work with established protocols such as RTP and HTTP.

S

SAP Service access point. Field that is part of an address specification.

SAR Segmentation and reassembly. In ATM, the process of dividing PDUs into 48-byte pieces of payload data at the source for transport and then reassembling them into a stream at the destination.

SAS Special exchange subscriber (as in SAS loop-start and ground-start). Identical to FXS loop-start and ground-start in functionality, but makes different use of the A and B bits. (SAS-transmitted B bits are complementary to FXS-transmitted B bits. SAS-received A bits are complementary to FXS-received A bits.)

SCCP Skinny (or Simple) Client Control Protocol. Cisco-proprietary protocol that defines call-connection methods and signaling between IP phones and a router. Allows IP phones to coexist in an H.323 environment. Savings in memory size, processor power, and complexity makes the protocol desirable.

SCN Switched-circuit network. Network that carries traffic within channelized bearers of predefined sizes.

SCTP Stream Control Transmission Protocol. Protocol for general IP transport.

SDP 1. Session Definition (or Description) Protocol. Protocol that defines multimedia services. SDP messages can be part of SGCP and MGCP messages.

2. Session Data Protocol. Protocol that describes multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.

secondary dial tone Additional transmission of dial tone after one or more specified digits have been dialed on an outgoing call. Typically used for external calls and is intended to reassure the caller that the dialing of an external number can continue after the dialing of an access digit. Remains on until a subsequent digit is received from the phone.

SED Stream editor.

sequential ephone hunt group Hunt group in which ephone-dns ring in the left-to-right order in which they are listed when the hunt group is defined. The first number to ring is always the first number in the list (the number that is farthest left in the list). See also peer ephone hunt group.

session expiration Time at which an element considers the call timed out if no successful INVITE-request transaction occurs first.

session interval Largest amount of time that can occur between INVITE requests in a call before a call is timed out, as conveyed in the Session-Expires header.

SGCP Simple Gateway Control Protocol. Protocol that controls VoIP gateways using an external call control element (called a call agent). Used to establish, maintain, and disconnect calls across an IP network.

shared ephone-dn An ephone-dn that appears on more than one phone.

signaling gateway Gateway that supports only signaling traffic (no bearer traffic). Transmits PSTN signaling at the edge of an IP/ATM network and backhauls the signaling to a media gateway controller. For example, a gateway that terminates SS7 A-links is a signaling gateway. Sometimes designated SG.

silent ring Audible ring and call-waiting beeps are suppressed for incoming calls. Visible cues are the same as for a normal ring, including the flashing ((< icon in the phone display and the flashing red light on the handset.

simple forking Forking scenario in which all of the branches of a fork use the same voice codec compression. See also complex forking.

simple mixing Situation in which a mixer receives multiple streams of the same codec type and combines them into one stream that is sent to the telephony interface.

single-number voice and fax See fax detection.

SIP Session Initiation Protocol. Protocol, developed as an alternative to H.323, that equips platforms to signal the setup of voice and multimedia calls over IP networks.

SIP proxy server SIP proxy server. Sometimes designated SPS. See proxy server.

SIP session Session that includes a set of multimedia senders and receivers and the data streams that flow between senders and receivers. A SIP multimedia conference is an example of a session. The called party can be invited several times by different calls to the same session.

SIP URL Session Initiation Protocol Uniform (or Universal) Resource Locator. Used in SIP messages to indicate the originator, recipient, and destination of the SIP request. Takes the basic form of user@host, where user is a name or telephone number, and host is a domain name or network address.

SLA Service-level agreement. Contract between a wholesaler and a service provider that specifies the connectivity, performance, and availability levels that the wholesaler guarantees.

SLIP Serial Line Internet Protocol. Protocol for point-to-point serial connections using a variation of TCP/IP. Largely superseded by PPP.

SM Session manager.

SMTP Simple Mail Transfer Protocol. Simple ASCII protocol that describes the exchange of e-mail between two message-transfer agents using TCP/IP.

SNAP Subnetwork Access Protocol. Protocol that operates between a network entity in the subnetwork and a network entity in the end system. Specifies a standard method of encapsulating IP datagrams and ARP messages on IEEE networks. The SNAP entity in the end system makes use of the services of the subnetwork and performs three key functions: data transfer, connection management, and QoS selection.

SNMP Simple Network Management Protocol. Protocol, used in TCP/IP networks, for monitoring and controlling network devices and managing configurations, statistics collection, performance, and security.

socket listener Software provided by a socket client to receives datagrams addressed to the socket.

soft keys Patterns on an IP-phone LCD display that represent keys with labels indicating the particular actions that a phone user can take by pressing the display where they appear. Display of these keys changes according to the current activity of the phone. See also buttons.

SONET Synchronous Optical Network. Standard format for transporting a wide range of digital telecommunications services over optical fiber. Characterized by standard line rates, optical interfaces, and signal formats.

source IP group VoIP-side grouping of signaling characteristics that are associated with incoming H.323 or SIP calls.

span Full-duplex digital transmission line between two digital facilities.

SPE Service-processing element.

speech Encoded data delivered via an RTP stream. Excludes fax, modem, and DTMF relay. Includes DTMF digits that are sent as inband audio when DTMF relay is not enabled.

speed dial Ability to automatically make a call to a number by using a code or special button rather than dialing the complete number. See also local speed dial and personal speed dial.

SPI Service-provider interface.

SPRT Protocol Simple Packet Relay Transport Protocol. Simple packet-based protocol layered on UDP/IP that provides reliable in-sequence delivery of data across the IP network. Used in modem relay for fax.

SPVC Soft permanent virtual circuit.

SRST Survivable remote-site telephony.

SSCS Service-specific convergence sublayer. One of the two sublayers of any AAL. SSCS is service dependent and offers assured data transmission. The SSCS can be null as well, in classical IP-over-ATM or LAN-emulation implementations.

ssinfo Structure in a raw message that contains all decoded supplementary service information.

stateful proxy server SIP proxy server that remembers incoming and outgoing requests, provides reliable retransmission of proxied requests, and returns the best final responses.

stateless proxy server SIP proxy server that forgets all information after a request or response is processed. It merely forwards requests and responses.

static payload Payload that is assigned a specific RTP format and is generally grouped for specific applications—for example, audio and video conferencing. See also dynamic payload.

static route Routes that have been manually provisioned into a proxy server. Generally, there is no dynamic information about the existence, availability, or capacity of this route or it would have been discovered using TGREP. Based on policy within the location server, static routes may be injected into TRIP, in which case they are also considered local routes to that location server.

store and forward Function whereby a message is transmitted to some intermediate relay point and temporarily stored before forwarding to the next relay point.

store-and-forward fax See T.37.

stream One of multiple media branches that constitute a fork.

subrate Less than the standard rate of transmission, which is defined at the voice-grade rate of 64 kbps.

SUS message ISUP Suspend message.

suspend message ISDN Suspend message.

SVC Switched virtual circuit. Virtual circuit that is dynamically established on demand and torn down when transmission is complete. Used in situations where data transmission is sporadic. Called a switched virtual connection in ATM terminology. See also virtual circuit.

switchover Transfer of control from the active to a standby router when the active router fails, is removed from the network, or is manually taken down. See also failover.

T

T1 24 64-Kbps time slots on a 1.544-Mbps serial interface.

T1-CAS T1 channel-associated signaling.

T3 Digital WAN carrier facility. Sends DS3-formatted data at 44.736 Mbps through the telephone switching network.

TAC Cisco Technical Assistance Center organization.

TAPI Telephony-application programming interface. Microsoft Windows application-programming interface that allows developers to create customized IP telephony applications.

T-ASP Telephony-application service provider. Company that provides voice applications such as prepaid calling.

TBCT Two B-Channel Transfer. Call-transfer standard for ISDN interfaces.

TCF message Training Check message. Message sent by a calling fax machine during capabilities exchange that includes a stream of 0s for about 1.5 seconds through the agreed-upon HS modulation speed. The called fax machine responds with an FTT message if the speed is not acceptable or a CFR message if the speed is acceptable. See also CFR message, FTT message, and training.

Tcl Tool Command Language. Scripting language used for gateway products both internally and externally to Cisco IOS software code.

Tcl IVR Tool Command Language Interactive Voice Response.

TCP Transmission Control Protocol. Connection-oriented transport-layer protocol that provides reliable full-duplex data transmissions. Part of the TCP/IP protocol stack. See also TCP/IP and IP.

TCP/IP Transmission Control Protocol/Internet Protocol. Common name for the suite of protocols developed by the U.S. Department of Defense in the 1970s to support the construction of worldwide internetworks. TCP and IP are the two best-known protocols in the suite. See also TCP and IP.

TDM Time-division multiplexing. Technique in which information from multiple channels can be allocated bandwidth on a single wire based on preassigned time slots. Bandwidth is allocated to each channel regardless of whether the station has data to send.

TE Terminal equipment. Any ISDN-compatible device that can be attached to the network, such as a phone, fax, or computer. See also TEI.

TEI Terminal-endpoint identifier. Field in the LAPD address that identifies a device on an ISDN interface. See also TE.

TEL URL Telephone Uniform (or Universal) Resource Locator. Locator that describes voice call connections to a terminal. Can also be any connection through a voice messaging system or a service that can be operated using DTMF tones. Takes the basic form of tel:telephone-subscriber-number, where tel indicates a URL and requests the local entity to place a voice call, and telephone-subscriber-number is the number to receive the call.

TFTP Trivial File Transfer Protocol. Protocol for transferring files over IP-based systems. Has no authentication, only restrictions based upon IP address. Uses UDP services and is commonly used to update software within a product. Quickly sends files across the network with fewer security features than FTP.

TGK Terminating gatekeeper. Gatekeeper at the receiving end of a call. Sometimes designated TGK.

TGREP Telephony Gateway Registration Protocol. Protocol for sending updates to the TRIP system as local PSTN routes and resources disappear and reappear. TRIP updates may be preferable to H.323 or SIP registration for adding and withdrawing individual routes.

TGRM Trunk-group-resource manager.

TGW 1. Terminating gateway. Voice gateway that handles calls that leave the IP network.

2. Trunking (or trunk-side) gateway. *x*GCP media gateway that provides PSTN/IP functionality.

TIFF Tagged Image File Format. In T.37 Store-and-Forward Fax, the format into which faxes are transformed when they are delivered to an MTA for delivery or storage.

TIFF-F Tagged Image File Format for Facsimile. File format for the storage and interchange of fax images.

TNS IE Transit Network Selection information element. Call-setup information that identifies a transit network. May be repeated in a SETUP message to select a sequence of transit networks through which a call passes.

training Process during a fax-capabilities exchange that verifies the communication path. Consists of a TCF message from the calling fax machine that sends a stream of 0s for about 1.5 seconds at the agreed-upon HS modulation speed. The called device responds with an FTT message if the speed is not acceptable or a CFR message if the speed is acceptable. See also CFR message, FTT message, and TCF message.

Trellis-coded modulation Technique for modulating digital data onto a high-frequency carrier tone with an optimal encoding method that minimizes noise-induced distortions that can result in decoding errors.

TRIP Protocol Telephony Routing over IP Protocol. Protocol for communicating information about reachable telephony routes across a VoIP network.

trunk group PSTN-side logical grouping of multiple DS1 interfaces with the same signaling characteristics.

trunking gateway *x*GCP media gateway that provides connectivity between the PSTN and VoIP networks.

TSE Inband-telephony signaling event.

T-series recommendations ITU-T standards that define terminals for telematic (that is, wireless-data-network) services. Well-known T-series recommendations related to fax transmission include T.30 for formatting of nonpage data over PSTN networks, T.37 for Store-and-Forward "simple" fax transmission by Internet mail, T.38 for transfer of fax documents between standard Group 3 fax terminals using IP protocols, and T.4 for formatting of page data over PSTN networks. T.30 and T.4 together are referred to as Group 3 or traditional fax.

TSI message Transmitting Subscriber Identification message. Message sent by a called fax machine during capabilities exchange that provides some detail as to the identity of the calling device.

TSP Telephony service provider. TAPI application that allows integration with voice mail.

TTS Text-to-speech. Capability of an external media server to synthesize speech; output provided by reading speech to the user of an IVR application.

TUI Telephone-user's interface.

U

UA User agent. Combination of UAS and UAC that initiates and receives calls. See UAS and UAC.

UAC User-agent client. Client application that initiates a SIP request.

UAL User Adaptation Layer.

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

UC Unified communications. Convergence of multiple media types on top of a common IP infrastructure.

UDP User Datagram Protocol. Connectionless transport-layer protocol for exchanging datagrams without acknowledgments or guaranteed delivery. Part of the TCP/IP protocol stack.

UDPTL User Datagram Protocol transport layer. Transport layer that is used on top of UDP. Makes packet delivery more reliable by providing data redundancy.

ULP Upper Layer Protocol. Logical higher-layer application that uses SCTP services.

upspeed Method used to dynamically change the codec type and speed to meet network conditions.

URI Uniform (or Universal) Resource Identifier. Takes a form similar to an e-mail address, indicates the user's SIP identity, and is used for redirection of SIP messages.

URI Uniform Resource Identifier. Type of formatted identifier that encapsulates the name of an Internet object, and labels it with an identification of the name space, thus producing a member of the universal set of names in registered name spaces and of addresses referring to registered protocols or name spaces.

URL Universal (or Uniform) Resource Locator. Standard address of any resource on the Internet that is part of the World Wide Web.

URQ message Unregistration Request RAS message.

V

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

VBR Variable bit rate. QoS class for ATM networks. VBR is subdivided into a real time (RT) class and non-real time (NRT) class. VBR (RT) is used for connections in which there is a fixed timing relationship between samples. VBR (NRT) is used for connections in which there is no fixed timing relationship between samples but that still need a guaranteed QoS.

VCC Virtual-channel connection (used where it may be a PVC, SPVC, or SVC).

VFC Voice feature card. Voice-processing card that resides in a slot in a Cisco platform, holds up to five DSP modules, and performs voice processing for up to 30 B channels and potentially 60 B channels.

Via header Message header that is part of an INVITE request. Includes information about the transport paths taken by a SIP request.

VIC Voice-interface card. Card that connects the system to either the PSTN or to a PBX and provides voice-specific ports, including FXS, FXO, E&M, and BRI.

virtual voice port Addressable endpoint for a call leg. Each ephone-dn has one virtual voice port that can make one call connection.

VM Voice mail. Application that creates and maintains voice-message mailboxes.

VoFR Voice over Frame Relay. Service that enables a router to carry voice traffic (for example, phone calls and faxes) over a Frame Relay network.

VoiceXML Voice Extensible Markup Language. Technology that allows a user to interact with the Internet through voice-recognition technology. Instead of a traditional browser that relies on a combination of HTML and keyboard and mouse, VXML relies on a voice browser and/or phone. The user interacts with the voice browser by listening to prerecorded or computer-synthesized audio output and submitting audio input through the user's natural speaking voice or through a (usually phone) keypad. See also XML.

VoIP Voice over IP. Capability of carrying normal telephony-style voice over an IP network with circuit-based telephone-like functionality, reliability, and voice quality. DSPs segment the voice signal into frames, which then are coupled in groups of two and stored in voice packets. These packets are transported using IP in compliance with H.323. VoIP is a blanket term that generally refers to the Cisco standards-based (H.323 and so forth) approach to IP voice traffic.

VoIP dial peer Dial peer that is connected to an IP network.

VPDN Virtual private dial-up (or dial) network. Network that extends remote access to a private network using a shared infrastructure. Uses Layer 2 tunnel technologies (L2F, L2TP, and PPTP) to extend Layer 2 and higher parts of the network connection from a remote user across an ISP network to a private network. Is a cost-effective method of establishing a long-distance, point-to-point connection between remote dial users and a private network. See also VPN.

VPM Voice-port module.

VPN Virtual private network. Network that enables IP traffic to travel securely over a public TCP/IP network by encrypting all traffic from one network to another. Uses tunneling to encrypt all information at the IP level. See also VPDN.

VRU Voice-response unit.

VSA Vendor-specific attribute. Attribute that specifies vendor-specific options. Vendors can support their own extended attributes that may not be suitable for general use.

VSC Virtual switch controller.

VToA Voice Trunking on ATM.

VWIC Card that can operate as a VIC or as a WIC, providing physical connection to WAN or voice networks.

VXML See VoiceXML.

W

W3C World Wide Web Consortium. Standards body responsible for HTML and XML.

WAN Wide-area network. Data communications network that serves users across a broad geographic area and often uses transmission devices provided by common carriers. Examples include Frame Relay, SMDS, and X.25.

wav File extension for an audio file. See also au.

WIC Wide-area-network interface card.

WWW World Wide Web. Large network of Internet servers providing hypertext and other services to terminals running client applications, such as a browser.

X

XCSP External-call service provider. Subsystem that interoperates with external call protocols to provide services such as modem call setup and teardown.

XML Extensible Markup Language. Standard that defines a syntax to let you create markup languages that specify information structures.

X-series recommendations ITU-T standards that define data networks and open-system communications. Well-known X-series recommendations include X.25 for packet switching, X.400 for message-handling systems, and X.500 for directory services.