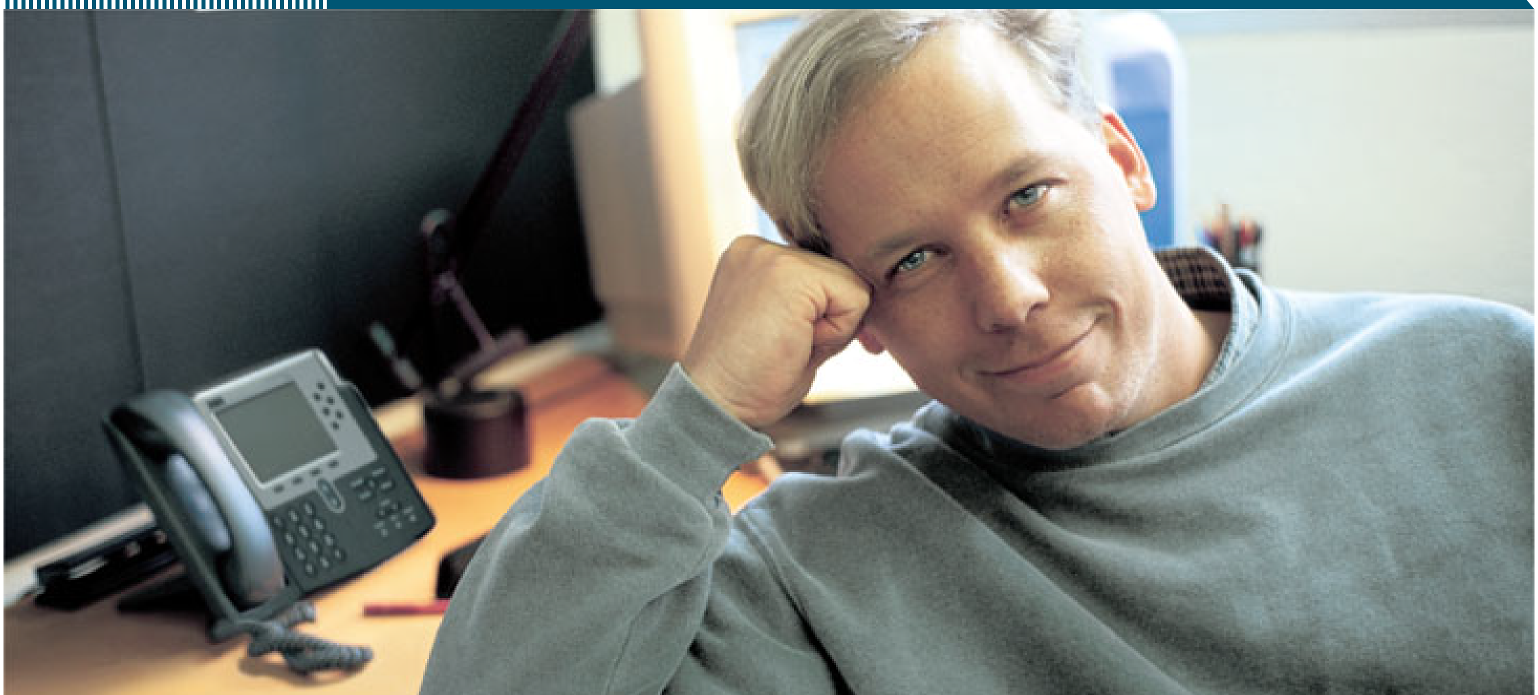
Cisco IP Telephony Glossary of Terms



A helpful guide to the terminology you need to know as you investigate the advantages of **a converged network.**

IP Telephony Glossary of Terms

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To find a specific term by letter of the alphabet, please click on the letter to the left.

A	
abandoned call	A call in which the caller hangs up before the call is answered.
access digit	On a PBX, a dialing number, such as 9, used to access an outside line. Also called access code.
access gateway	A gateway that allows the IP PBX to communicate with the PSTN or traditional PBX systems. See also *IP*, *PBX*, and *PSTN*.
access layer	Part of ISO-OSI layered protocol model.
access line	A transmission line that provides access to a larger system or network.
access link	The local access connection between a customer's premises and a carrier's point of presence (POP), which is the carrier's central switching office or closest point of local termination.
access method	The technique for moving data, voice, or video between storage and input/output devices.
access port	Connects a network device to an IP device. For example, a computer can be connected to an IP phone through an access port.
access protocol	A set of specific procedures that enable a user to obtain services from a telephone company or network.
ACD	Automatic Call Distribution. A software feature used to distribute calls to call center agents.

additional call offering	An Integrated Services Digital Network (ISDN) feature that allows multiple calls to be placed simultaneously to the same telephone number. A serving switch is programmed with the number of lines on the receiving telephone equipment. The switch will offer an additional call if there is a line available to accept it. See also *ISDN*.
alternate routing	A feature that redirects outbound calls based on predetermined criteria. For example, first choice might be the WAN with calls alternately routed to the PSTN if the WAN is busy.
ambient noise	The background noise that is present on a non-digital communications line at all times.
ANSI	American National Standards Institute. A U.S. organization chartered to accredit standards developed by a wide variety of industry groups while avoiding improper influence from any one company or organization. ANSI does not develop standards, but reviews and implements those developed by other organizations. For example, ANSI accredits standards for telephony developed by the Alliance for Telecommunications Industry Standards (ATIS) under the auspices of the T1 Committee, and standards for cellular radio developed by the Electronics Industry Association (EIA) and the Telecommunications Industry Association (TIA). ANSI is a member of the International Organization for Standardization (ISO). See also *ISO*.
ATC	Attendant Console. Also called an Operator Console. A specialized phone set used by console operators to answer and direct incoming calls.
automated attendant	An automated answering point for incoming calls to a company. Usually this is included as a feature in the voice mail system.
AVD	Alternate Voice Data. A single transmission facility used for either voice or data.
AVVID	See *Cisco AVVID*.

B	
BH	Busy Hour. The peak 60-minute period during a business day when the largest volume of traffic is handled by a network.
BHCA	Busy Hour Call Attempts. A measure of the maximum number of call attempts the system can support. Generally, BHCA and BHCC are considered to be the same.
BHCC	Busy Hour Call Completions. A measure of the maximum number of actual call completions the system can support. Generally, BHCA and BHCC are considered to be the same.
B-ISDN	Broadband Integrated Services Digital Network. A network that employs switching techniques independent of transmission speeds, and that allows a network to expand its capacity without major equipment overhauls. B-ISDNs support gigabit-speed circuits in the public network and high-speed switching of all traffic types in public and private networks. B-ISDNs also provide bandwidth-on-demand capabilities. See also *BRI* and *ISDN*.
blocked call	1. An attempted call that cannot be connected. The two most common reasons for blocked calls are that all lines or trunks to the central office are in use, or all paths through a private branch exchange (PBX) or switch are in use. 2. A service offered by 900 providers that permits users to request that their local carrier block all 900 calls in order to avoid incurring charges.
blocking	The inability to establish a new call because of restrictions or inaccessibility of facilities in the system being called. The system is called “non-blocking” if calls can always be placed.
break	1. To interrupt the sending of a message and take control of the circuit at the receiving end. 2. An interruption of a transmission or process.

BRI	Basic Rate Interface. ISDN interface composed of two B-channels and one D-channel for circuit-switched communication of voice, video, and data. The B-channels carry voice or data, the D-channel is for signaling. See also *ISDN.*
busy lamp field	A set of lights or LEDs commonly found on an attendant console (ATC) that give visual indication as to which phones on the system are busy.
busy tone	A single tone that is repeated at a 60 impulse per minute (ipm) rate to indicate that a call's terminating location is already in use.

C	
Call Admission Control (CAC)	A QoS "tool" used to protect voice traffic from being negatively affected by other voice traffic, and to keep excess voice traffic off the network.
Call Manager (CCM)	A software-based call-processing component of the Cisco IP Telephony solution. The software extends telephony features and functions to packet telephony network devices such as IP phones, media processing devices, voice-over-IP (VoIP) gateways, and multimedia applications. Additional data, voice, and video services such as unified messaging, multimedia videoconferencing, collaborative contact centres, and interactive multimedia response systems interact with the IP Telephony solution through Cisco CallManager's open telephony application programming interface (API).
caller-entered digits	Digits entered by a caller on a touch-tone phone in response to prompts.
calling line ID (CLID)	Information about the billing telephone number from which a call originated. The CLID value might be the entire phone number, the area code, or the area code plus local exchange.

carrier	A company that provides telecommunications circuits. Carriers include the local telephone company and companies such AT&T, Sprint, and MCI.
CAS	1. Channel Associated Signaling. A type of signaling on T1 digital circuits where the signaling shares the same channels as the voice or data stream. 2. Centralized Attendant Service. A feature on PBXs where operator consoles can be located at a central site and support branch locations. Commonly used in retail applications.
CC	1. Common Carrier. A government-regulated private company that furnishes the general public with telecommunications services and facilities. 2. Country Code. Part of a numbering plan.
CCAPI	Call Control Applications Programming Interface.
CCC	See *Conference Connection.*
CCITT	Consultative Committee for International Telegraph and Telephone. A telecommunications organization that recommended worldwide standards for common carrier communications services. This organization was superseded by the International Telecommunications Union, now called the ITU-T. See *ITU-T.*
CCS	Common Channel Signaling. Signaling system used in telephone networks that separates signaling information from user data. A specified channel is exclusively designated to carry signaling information for all other channels in the system.
CCSS7	Common Channel Signaling System 7. The protocol used by the AT&T signaling network. ICM software's NIC receives routing requests from the CCSS7 network and returns a routing label to the CCSS7 network.

CDR	Call Detail Recording. A stored database record containing data about a specific call. Processed as a unit and used to create billing records, a CDR contains details such as the called and calling parties, originating switch, terminating switch, call length, and time of day.
centralized call processing	A processing construct in which the central site contains all call processing resources and supports the branch offices as well. In terms of the Cisco CallManager, centralized call processing means that the central site contains a Cisco CallManager or Cisco CallManager cluster, but the branches do not have call processing servers.
character-based display	The display on the telephone is only capable of displaying characters (letters and numbers), not graphics.
Cisco AVVID	Architecture for Voice, Video and Integrated Data. AVVID is a standards-based, distributed, open network architecture that provides the framework for today's Internet business solutions and a roadmap for combining your business and technology strategies into one cohesive model. See also *AVVID*
Cisco Unity	A powerful, cost-effective unified messaging solution that allows users to access and immediately respond to voice, fax and email messages from any phone or PC. For example, you can listen to your e-mail over the telephone, check voice messages from the Internet, and forward faxes to any local fax machine. See also *Unified Messaging.*
circuit switching	Switching system in which a dedicated physical circuit path must exist between sender and receiver for the duration of the "call." Used heavily in the public switched telephone network.

Cisco IP Phone	A full-featured telephone that provides voice communication over an IP network while functioning much like a traditional PBX phone. Allows you to place and receive telephone calls, and supports features such as call forwarding, redial, speed dialing, call transfer, and conference calling. Also allows you to access voice mail, providing connectivity to Cisco IP Telephony Solutions.
Cisco IP Telephony Solutions	A software and hardware product suite offering an IP alternative to traditional PBXs. Includes Cisco IP Phones, H.323-compatible gateway clients, and server software enabling voice and data over an existing LAN or WAN infrastructure. See also *Cisco IP Phone.*
Cisco Media Convergence Servers	The Cisco MCS-7800 series server family, which includes the high-availability MCS-7830 and the Cisco AVVID IP telephony starter kits.
Classification	Marking a data packet with a specific "tag" denoting a requirement for special service from the network. i.e. a "Voice" packet or "data" packet
cluster	A group of CallManager servers that interoperate to form a single system image. This means that the servers in the cluster share databases and act as a single sytem in terms of features, administration, collection of CDR records, etc. This feature allows Cisco CallManager to scale up to 10,000 phones in a single system.
codec	Coder-decoder. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a software algorithm used to compress/decompress speech or audio signals.
computer telephony integration (CTI)	Software that integrates voice communications systems with computers for contact center and office automation applications.
Collaboration Server and E-Mail Manager	Provides customers with multiple, integrated channels for obtaining sales, service and support.

Conference Connection	Cisco Conference Connection (CCC) is an intuitive Web-based audio conference application that provides access to a conference regardless of location, utilizes a single dial-in number for all participants, and supports up to 100 simultaneous callers.
configuration file	An unformatted ASCII file that stores initialization information for an application. For Cisco CallManager, files in the .cnf format that define the parameters for Cisco IP Phone connection.
CoS	Class of Service. A collection of features, privileges, and services that are easily assignable to a group or "class" of telephones. Class of Service is used to simplify administration and maintenance tasks in complex telephony networks.
CPE	Customer Premises Equipment. Telephone equipment, such as key systems, PBXs, answering machines, etc., that reside on the customer's premises (e.g., office building, home office, or factory). Also called Customer Provided Equipment.
CTI ports	Computer Telephony Interface ports. Virtual devices that are used by Cisco CallManager applications such as Cisco SoftPhone, Cisco IP AutoAttendant, and Cisco IP Interactive Voice Response System (IVR) to create virtual lines. CTI ports are configured through the same Cisco CallManager Administration area as phones but require different configuration settings.
CTI route point	Computer Telephony Interface route point. Virtual device that can receive multiple simultaneous calls for the purpose of application-controlled redirection. Once a CTI route point has been created, lines (directory numbers) can be added and configured. Applications that use CTI route points include Cisco IP AutoAttendant, Cisco IP Interactive Voice Response System (IVR), and Cisco TAPI/JTAPI.

D	
default router	For IP devices, identifies the default gateway used by the device. Also called default gateway.
delay	Delay is the amount of time (usually measured in msecs) that it takes for a packet to travel across the network from one end of the connection to the other. Also called latency.
Dialogic card	Hardware made by Dialogic (an Intel company) that initiates dialing and voice detection that is found in many VoIP systems.
Dialogic voice board	Printed circuit board produced by Dialogic (an Intel company) containing digital signal processor (DSP) chips to digitize voice.
dial peer	An addressable call endpoint. In Voice over IP, there are two kinds of dial peers: POTS and VoIP.
dial plan	A system that allows one telephone or Cisco IP device to connect to another telephone or Cisco IP device by using dialed digits. In North America and many Caribbean nations, the dial plan is called the North American Numbering Plan.
dial-up	The use of a rotary or dual tone multiple frequency (DTMF) telephone to initiate a call over the public switched telephone network.
dial-up line	1. A communications circuit established by a switched connection. 2. Any circuit available over the public switched telephone network. See also *PSTN*.
DID	Direct Inward Dialing. A method of directly dialing the directory number of a Cisco IP Phone or a telephone attached to a PBX without routing calls through an attendant or an automated attendant console, such as Cisco WebAttendant. Compare to *DOD*.

distributed call processing	A processing construct in which each central site and branch office contains its own call processing resources. In terms of the Cisco CallManager, distributed call processing means that each central site and branch site contains its own Cisco CallManager or Cisco CallManager cluster.
DN	Directory Number. The telephone number or internal extension assigned to a Cisco IP Phone. The directory number is assigned to the phone itself, not a location or a user, so if the phone is moved, it still retains the same directory number. Also called subscriber number.
DNIS	Dialed Number Identification Service. This feature is commonly used in a call center that has multiple phone numbers to distinguish which specific number the caller dialed. The information can be used to direct the call to the appropriate answering point, such as a department or specific ACD group.
DNS	Domain Name System. System used in the Internet for translating names of network nodes into IP addresses.
DOD	Direct Outward Dialing. The ability to dial directly from Cisco CallManager or PBX extension without routing calls through an operator, attendant, or automated attendant functions. Compare to *DID*.
DoS	Denial of Service. Type of attack that prevents access to, or operation of, a device or network.
DPNSS	A protocol, developed in the 1980's, as an open standard. Used on digital trunk lines for inter-connecting PABX's.
DSCP	Differentiated Services Code Point, or DiffServe CodePoint. A marker in the header of each IP packet that prompts network routers to apply differentiated grades of service to various packet streams, forwarding them according to different Per-Hop Behaviors (PHBs). Part of DiffServe, a set of technologies proposed by the IETF that allows Internet and other IP-based network service providers to offer differentiated levels of service to customers and their information streams.

DSP	Digital Signal Processor. Specialized computer chip designed to perform speedy and complex operations on digitized waveforms. Useful in processing sound, such as voice phone calls, and video.

E

E1	An E1 circuit is the European equivalent of a T1. It carries 30 channels of 64 kbps each, along a 2.048 Mbps bearer channel.
E&M	recEive and transMit (or ear and mouth). Trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco's analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M is also available on E1 and T1 digital interfaces.
EIGRP	Enhanced Interior Gateway Routing Protocol. Advanced version of IGRP developed by Cisco. Provides superior convergence properties and operating efficiency, and combines the advantages of link state protocols with those of distance vector protocols.
Emergency Responder (E999/E911/ E110, etc.)	Enables emergency agencies to identify the location of emergency callers and eliminates the need for any administration when phones or people move from one location to another. Cisco Emergency Responder's real-time location-tracking database and improved routing capabilities direct emergency calls to the appropriate emergency services answering point based on the caller's location. This service is valid for emergency numbers around the world.
enterprise-wide call distribution	A strategy for allocating calls among several call centers or other answering locations based on real-time information about activity at each location. ICM software implements enterprise-wide call distribution and allows calls to be sent to any network-addressable location within, or outside of, an enterprise.

F	
FCC	Federal Communications Commission. U.S. government agency that supervises, licenses, and controls electronic and electromagnetic transmission standards.
FIFO	First In, First Out. In data communication, FIFO refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queuing scheme where the first calls received are the first calls processed.
FXO	Foreign Exchange Office. A trunk side connection between a central office switch and a digital telephony switching system.
FXS	Foreign Exchange Station. A line side connection between a digital telephony switching system and a POTS telephone.

G	
G.711	An audio compression standard used for digital telephones on a digital PBX/ISDN. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. G.711 uses a bandwidth of 64 Kbps. G.711-compliant devices can communicate with other G.711 devices, but not with G.723 devices. Described in the ITU-T standard in its G-series recommendations.
G.723.1	Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This codec allows dissimilar communication devices to communicate with each other using a standardized communications protocol. Used for digital telephones on a digital PBX/ISDN that produces digital audio at either 6.4 or 5.3 Kbps. The higher bit rate provides a somewhat higher quality of sound. The lower bit rate provides system designers with additional flexibility. Described in the ITU-T standard in its G-series recommendations.
G.729	ITU-T's standard voice algorithm. Describes the coding of encoding/decoding of speech at 8 Kbps using CS-ACELP methods.
gateway	The point at which a circuit-switched call is encoded and repackaged into IP packets. A gateway is an optional element in an H.323 conference and bridges H.323 conferences to other networks, communications protocols, and multimedia formats.

H

H.323	ITU-T standard that describes packet-based video, audio, and data conferencing. Allows dissimilar communication devices to communicate with each other using a standardized communications protocol. H.323 is an umbrella standard that describes the architecture of the conferencing system, and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol. For example, the Cisco IOS integrated router gateways use H.323 to communicate with Cisco CallManager. See also *gateway*.
H.323 RAS	Registration, Admission, and Status. The RAS signaling protocol performs registration, admissions, bandwidth changes, and status and disengage procedures between the VoIP gateway and the gatekeeper. See also *VoIP*.
handset	The portion of a telephone set containing the transmitter and receiver, usually designed to be hand-held when the telephone is in use. For example, lift the handset of a Cisco IP Phone to press the dial pad numbers to place a call, review voice mail messages, answer a call, and so on.
hot swappable	In service terminology, this means that a part such as a power supply, circuit board, etc., can be pulled out and replaced while this system is running (i.e., “hot”).
hunting	1. The automatic routing of calls to an idle circuit in a prearranged group when the circuit being called is busy or unavailable. 2. The movement of a call as it progresses through a group of lines. The call will try to connect to the first line of the group. If that line is busy or unavailable, it will try the second line, and then the third line, etc.

I	
ICM gateway	A construct that allows one ICM system to forward a request to another ICM. You can configure an ICM gateway in Configure ICM and reference in the ICM Gateway node in a routing script.
Intelligent Call Processing (ICP)	AT&T's name for the facility that allows third-party products such as Cisco ICM software to pre-route calls.
Intelligent Contact Management Software (ICM)	The Cisco system that implements enterprise-wide call distribution across call centers. ICM software provides pre-routing, post-routing, and performance-monitoring capabilities.
Interexchange Carrier (IXC)	A long-distance telephone company such as AT&T, MCI, or Sprint.
interflow	The ability of a switch to forward calls to another location within the switch or to another switch. Interflow between switches requires a dedicated trunk line.
Internet Service Node (ISN)	The Cisco Internet Service Node (ISN) provides Web-based, carrier-class Interactive Voice Response (IVR), queuing, and IP switching services for both IP and traditional telephony networks.
IP	Internet Protocol. Messaging protocol that addresses and sends packets across the network in the TCP/IP stack, offering a connectionless internetwork service. To communicate using IP, network devices must have an IP address, subnet, and gateway assigned to them. IP provides features for addressing, type-of-service specification, fragmentation and reassembly, and security. Standardized in RFC 791.
IP AA	IP Automated Attendant. A Cisco IP application that automatically answers and directs inbound calls.

IP Contact Centre (IPCC)	Cisco IP Contact Centre (IPCC) is an IP application that delivers intelligent call routing, network-to-desktop CTI, and multimedia contact management to contact centre agents over an IP network.
IP Integrated Contact Distribution (IP ICD)	Cisco IP Integrated Contact Distribution (IP ICD) is an inexpensive, easy-to-install, and easy-to-use automatic call distributor (ACD)
IP Phone Productivity Services	The Cisco IP Phone Productivity Services (PPS) is a suite of personal productivity applications for your Cisco IP Phone 7940 and 7960. These extensible markup language (XML)-based applications let you check your e-mail, voice mail, calendar, and personal contact information using the large, pixel-based LCD display and interactive soft keys on your phone.
IP Phone Services Software Development Kit (SDK)	The Software Developer Kit (SDK) makes it easier for Web developers to format and deliver content to the phone by providing Web server components for Lightweight Directory Access Protocol (LDAP) directory access, Web proxy, and graphics conversion. It also contains several sample applications, which show how to use the various Extensible Markup Language (XML) tags that the phone supports.
IP IVR	IP Integrated Voice Response. A Cisco IP application that provides full-featured integrated voice response capability to answer inbound calls, perform database lookups, re-direct calls automatically, etc.
IP phone	IP telephone. A phone that transports voice over a network using IP data packets instead of circuit-switched connections over voice-only networks. Full-featured IP phones can be plugged directly into an IP network and used very much like a standard private branch exchange (PBX) telephone.

IPX	Internetwork Packet Exchange. NetWare network layer (Layer 3) protocol used for transferring data from servers to workstations. IPX is similar to IP and XNS.
ISDN	Integrated Services Digital Network. Communication protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other source traffic. See also *BRI.*
IS-IS Protocol	Intermediate System to Intermediate System Protocol. A standards-based routing protocol used mainly in large ISP networks.
ISO-OSI	ISO: International Organization for Standardization. The ISO establishes global standards for communications and information exchange. OSI: A networking reference model defined by the ISO that divides computer-to-computer communications into 7 connected layers. Such layers are known as a protocol stack.
ISP	Internet Service Provider. Company that provides Internet access to other companies and individuals.
ITU	International Telecommunications Union. The telecommunications agency of the United Nations established to provide worldwide standard communications practices and procedures. Formerly known as the Comite Consultatif Internationale de Telegraphique et Telephonique (CCITT).
ITU-T	Telecommunication standardization sector of ITU. International body that develops worldwide standards for telecommunications technologies. See also *ITU*.
IVR	Integrated Voice Response unit. An application that provides full-featured integrated voice response capability to answer inbound calls, perform database lookups, re-direct calls automatically, etc.

J	
jitter	A type of distortion caused by the variation of a signal from its reference that can cause data transmission errors, particularly at high speeds.
JTAPI	Java Telephony Application Programming Interface. See also *TAPI*.

K	
KTS	Key Telephone System. A small telephone system in which the telephones have multiple buttons requiring the user to directly select central office phone lines and intercom lines. Key Telephone Systems are similar to PBX systems, but differ in that they do not provide their own switching capabilities, routing and trunking capabilities, dial plans, or feature sets. Most Key Telephone Systems support from 10 to 50 telephones.

L	
latency	Delay in the amount of time (usually measured in msecs) that it takes for a packet to travel across the network from one end of the connection to the other.
LCR	Least Cost Routing. An algorithm in digital PBXs that selects the least expensive route for a call based on factors such as the dialed digits, time of day, day of week, etc.
line side	A connection that extends from an end office (EO), central office (CO), or private branch exchange (PBX) to the subscriber's telephone or extension.
Local Exchange Carrier (LEC)	The local phone company responsible for delivering calls within a local area.
local loop	1. The communication line between a telephone subscriber and the local exchange carrier (LEC) switching center. 2. A local connection between an end user and a central office (CO) or end office (EO).

M	
media stream	The information content carried on a call. Refers to what is actually transmitted and received over the line, and can be read or written by a media stream API.

N	
NANP	North American Numbering Plan. The North American Numbering Plan (NANP) was invented in 1947 by AT&T and Bell Laboratories. It conforms to the International Telecommunications Union Recommendation E.164, the international standard for numbering plans. The NANP is the numbering plan for the Public Switched Telephone Network (PSTN) in the United States and its territories, Canada, Bermuda, and many Caribbean nations. NANP numbers are 10 digits in length, and they are in the format NXX-NXX-XXXX, where N is any digit 2-9 and X is any digit 0-9. The first three digits are called the numbering plan area (NPA) code, often called simply the area code. The second three digits are called the central office code or prefix. The final four digits are called the line number.
Network Applications Manager (NAM)	The NAM is a flexible, highly scalable software that supports a portfolio of services-from simple to complex- to capitalize on business opportunities across the customer base.
network port	Connects an IP device to the network.
network trunk group	A group of trunks to which a routing client can direct calls. A peripheral may divide its trunks into trunk groups differently than the routing client does. Simple trunk groups describe the peripheral’s view of the trunks; network trunk groups describe the routing client’s view of the trunks.
Network-Wide Quality of Service	Features that ensure voice quality and enhance the control of a converged network by allowing traffic classification and prioritization. Features provide centralized traffic management for automated service provisioning, reporting and analysis.
NTP	Network Time Protocol. Protocol that ensures that device clocks are set to the same time, relative to Greenwich Mean Time.

O	
office code	The first three digits of your seven-digit local telephone number.
off-line	Describes a device that is not permanently connected to a network.
on-hook	1. The condition that exists when a receiver or handset is resting on the hookswitch. 2. The idle state (open loop) of a single telephone or private branch exchange (PBX) line loop.
OOS	Out Of Service signaling.
OPX	Off Premises Extension. A peripheral private branch exchange (PBX) device located in a building other than the one housing the PBX system itself. Also called OPS, or Off Premise Station. See also *PBX*.

P	
PBX	Private Branch Exchange. Digital or analog telephone switchboard located on the subscriber premises, typically with an attendant console, and used to connect private and public telephone networks. A PBX is a small, privately owned version of the phone company's larger central switching office. It is connected to one or more central offices by trunks, and provides service to a number of individual phones, such as in a hotel, business, or government office. On a PBX, an outside line is normally accessed by dialing an access digit, such as 9.
Personal Assistant	Cisco Personal Assistant is a telephony application that uses voice recognition and rules-based call control to help users manage how and where they want to be reached. Users can browse voice mail, dial by name, and conference from any telephone using voice instead of the telephone keypad. The user administration interface allows users to forward and screen calls in advance or in real time.
phone button templates	Define which keys on a phone or IP device perform which functions. Use templates to customize individual IP phones and to assign common button configurations to a large number of phones. Cisco CallManager includes several default phone button templates, all of which can be modified.
Policing	Insuring that traffic uses only it's allocated amount of provisioned bandwidth.
POTS	Plain Old Telephone Service. Standard telephone service used by most residential locations. For example, POTS line connections are used to join a Cisco Analog Station gateway and an SMDI-compliant voice mail system. See also *PSTN, SMDI*.

PRI	Primary Rate Interface. An ISDN trunk type supporting 23 B-channels and 1 D-channel in North America (30 B-channels and 1 D-channel in Europe). The B-channels carry voice and data, the D-channel is for signaling.
Provisioning	Accurately calculating the required bandwidth for all application and element overhead.
PSTN	Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide.

Q	
Q.Sig	A signaling standard that allows internetworking of PBXs from different vendors and provides a basic set of feature transparency between systems (calling name and number display, transfer, etc.).
QoS	Quality of Service. Measure of performance for a transmission system that reflects its transmission quality and service availability.
queuing	A technique in which incoming calls are stored on hold until an attendant, trunk, trunk group, or station is available to accept them. Also known as camp on.

R	
RAS	Registration, Admission, and Status Protocol. Used in the H.323 protocol suite for discovering and interacting with a gatekeeper. See also *H.323*.
redundancy	Means that multiple like components are included in a system to increase the reliability and availability. Various forms of redundancy exist, but basically the duplicate component(s) back up the primary components under failure conditions.
RTP	Real-time Transport Protocol. A network protocol used to carry packetized audio and video traffic over an IP network.

S	
SMDI	Simplified Message Desk Iinterface. Analog data line from the central office containing information and instructions to your on-premises voice mail box. A required interface for voice mail systems used with Cisco CallManager. SMDI was designed to enable voice mail integration services to multiple clients. However, to use SMDI, the voice mail system must meet several qualifications, including providing database support for two PBX systems simultaneously and IP network connectivity to the voice messaging system while maintaining the existing link to the PBX. SMDI-compliant voice mail systems must be accessible with a null-modem RS-232 cable and available serial port. See also *POTS.*
Scheduling	Assigning packets to one of multiple queues (based on Classification) for expedited treatment through the network.

soft keys	On a Cisco IP Phone, buttons that activate features described by a text message. The text message is displayed directly above the soft key button on the LCD screen.
SoftPhone	Application that enables you to use a desktop PC to place and receive software telephone calls and to control an IP telephone. Also allows for audio, video, and desktop collaboration with multiple parties on a call. Cisco IP SoftPhone can be used as a standalone application or as a computer telephony integration (CTI) control device for a physical Cisco IP Phone. All features are functional in both modes of operation. See also *Cisco IP Phone*.
SS7	Signaling System 7. A telephone signaling system with three basic functions: supervising (monitoring the status of a line or circuit to see if it is busy, idle, or requesting service); alerting (indicating the arrival of an incoming call); addressing (transmission of routing and destination signals over the network).
Survivable/ Standby Remote Site IP Telephony	A cost effective, reliable mechanism for providing continuous IP Telephony services and call processing features to branch offices. Reliable back-up is provided in the event of a WAN failure.

T	
T1	A North American standard digital trunk that has 24 channels of 64 Kbps each.
TAPI	Telephony Application Programming Interface TCP/IP. A set of functions that allow Windows applications to program telephone line-based devices such as single and multi-line phones (including Cisco IP Phones), modems, and fax machines in a device-independent manner. See also *JTAPI*.
TCD	Cisco Telephony Call Dispatcher. A Cisco CallManager service that handles requests by the Cisco WebAttendant for call control, call dispatching, line status, and user directory information.
telephony	Science of converting sound to electrical signals and transmitting it between widely removed points.
Telnet	A program that lets you connect to other computers on the Internet. The terminal-remote host protocol developed for ARPA that allows you to work from your PC as if it were a terminal attached to another machine by a hardwire line.
Telnet proxy program	Links the Telnet server at the customer site to the relay server. When started by the customer, it initiates a "Telnet tunnel," establishing a TCP connection from inside the customer firewall out to the relay server on the public Internet. Then it establishes another connection to the local Telnet server, creating a two-way link between the entities.
third-party call control	If an audio stream terminates at some location or physical device other than your application or device, you have third-party call control. For example, the Cisco SoftPhone can control the Cisco IP Phones. Used in TAPI development.
tie line	A private trunk line that connects two ACDs or PBXs across a wide area.

toll bypass	A toll-free telephony call in which the relative locations of the two ends of the connection would cause toll charges to be applied if the call were made over the PSTN.
traffic	The load on a communications device or system.
trunk	Physical and logical connection between two switches across which network traffic travels. A trunk is a voice and data path that simultaneously handles multiple voice and data connections between switches. A backbone is composed of a number of trunks.
Trunk group	A group of essentially alike trunks (shared electronic characteristics) that go between the same two geographical points. A trunk group performs the same function as a single trunk but carries multiple conversations.
TSP	Telecommunications Service Priority. The regulatory, administrative, and operational system that authorizes and provides priority treatment for initiating and restoring telecommunication services.

U

Unified Messaging	A messaging solution that allows users to access and immediately respond to voice, fax and email messages from any phone or PC. See also *Cisco Unity*.

V	
VAC	Voice Activity Compression. A method of conserving transmission capacity by not transmitting pauses in speech.
VAD	Voice Activity Detection. When enabled on voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded, but the connection monopolizes much less bandwidth.
VIC	Voice Interface Card.
VoFR	Voice over Frame Relay. Enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network using FRF.12 encapsulation.
voiceband	A transmission service with a bandwidth considered suitable for transmission of audio signals. Generally 300 Hz or 500 Hz to 3,400 Hz.
VoIP	Voice over IP. Enables users to transfer voice communications over a data network using the Internet Protocol (IP). In VoIP, the DSP segments the voice signal into frames, which are then coupled in groups of two and stored in voice packets. These voice packets are transported using IP in compliance with ITU-T specification H.323.
VoIP dial peer	Dial peer connected via a packet network; in the case of VoIP, this is an IP network. VoIP peers point to specific VoIP devices.
VRU	Voice Response Unit. Also called an IVR (Integrated Voice Response unit). An application that provides full-featured integrated voice response capability to answer inbound calls, perform database lookups, re-direct calls automatically, etc.

VTSP	Voice Telephony Service Provider.
VXML	Voice-Enhanced Extensible Markup Language. Existing Web sites predominantly employ HTML. VXML is a proposed standard that will bring the power of Web development and content delivery to self-service voice applications. That is, allowing for Automatic Speech Recognition or keypad input to access Web pages that are interpreted and spoken back to the caller with recorded prompts and/or Text To Speech technology while using an industry standard programming language (VXML) to accomplish this.

W

WRED	Weighted Random Early Detection. A congestion-avoidance and QoS mechanism for IP-based networks.
Web Attendant	Cisco Web Attendant is an application that supports the traditional role of a manual attendant console. Associated with an IP phone, the application allows the attendant to quickly accept and dispatch calls to users. An integrated directory service provides traditional busy lamp field (BLF) and direct station select (DSS) functions for any line in the system. The application is Web-enabled and, therefore, portable to Windows 98, NT, and 2000 platforms. A primary benefit of Cisco Web Attendant over traditional attendant console systems is its ability to monitor the state of every line in the system and to efficiently dispatch calls. The absence of a hardware-based line monitor device offers a much more affordable and distributable manual attendant solution than traditional consoles.

X

XML	Extensible Markup Language. XML allows Web developers to define custom tags within a Web page to make the data richer in presentation. XML is used to display information on the Cisco IP Phones.