



Glossary of Terms

AAA—authentication, authorization, and accounting. AAA is a suite of network security services that provide the primary framework through which access control can be set up on your Cisco router or access server.

AAL—ATM adaptation layer.

ABCD signalling—4-bit telephony line signalling coding in which each letter of “ABCD” represents one of the 4 bits. ABCD signalling is often associated with channel-associated signalling or robbed-bit signalling on a T1 or E1 telephony trunk.

ACOM—Term used in G.165, “General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers.” ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.

ADPCM—adaptive differential pulse code modulation. A process by which analog voice samples are encoded into high-quality digital signals.

a-law—A voice compression technique commonly used in Europe.

ANI—Automatic Number Identification. Feature of signalling System 7 (SS7) in which a series of digits, either analog or digital, are included in the call, identifying the telephone number of the calling device. In other words, ANI identifies the number of the calling party.

ARQ—Admission repeat request.

broadband—Transmission system that combines multiple independent signals onto one cable. In the cable industry, broadband refers to the frequency-division multiplexing of many signals in a wide bandwidth of RF frequencies using a hybrid fiber-coaxial (HFC) network.

cable modem—A modulator-demodulator device that is placed at subscriber locations to convey data communications on a cable television system. The Cisco uBR900 series cable access router is also a cable modem.

cable router—A modular chassis-based router optimized for data-over-CATV hybrid fiber-coaxial (HFC) applications.

call leg—A logical connection between the router and either a telephony endpoint over a bearer channel, or another endpoint using a session protocol.

carrier—A signal on which another, lower-frequency signal is modulated in order to transport the lower-frequency signal to another location.

Carrier-to-Noise—C/N (also CNR). The difference in amplitude between the desired radio frequency (RF) carrier and the noise in a portion of the spectrum.

CAS—channel-associated signalling. In E1 applications, time slot 16 is used to send CAS information. Each frame's time slot 16 carries signalling information (ABCD bits) for two of the B channel time slots.

CATV—Originally stood for Community Antenna Television. Now refers to any coaxial or fiber cable-based broadband transmission facility.

CELP—Code Excited Linear Prediction. A compression algorithm used in low bit-rate voice encoding. CELP is used in ITU-T Recommendations G.728, G.729, and G.723.1.

CEPT—Conférence Européenne des Postes et des Télécommunications. Association of the 26 European PTTs that recommends communication specifications to the ITU-T.

channel—A specific frequency allocation and bandwidth. Downstream channels used for television are 6 MHz wide in the United States, 8 MHz wide in Europe.

CID—channel ID. Designates the Frame Relay subchannel ID for Voice over Frame Relay.

CIR—committed information rate. The average rate of information transfer a subscriber (for example, the network administrator) has stipulated for a Frame Relay PVC.

Cisco-trunk (private line) call—A Cisco-trunk (private line) call is established by the forced connection of a dynamic switched call. A Cisco-trunk call is established during configuration of the trunk and stays up for the duration of the configuration. It optionally provides a pass-through connection path to pass signalling information between the two telephony interfaces at either end of the connection.

CM—cable modem.

CMTS—Cable Modem Termination System. A termination system located at the cable television system headend or distribution hub that provides complementary functionality to the cable modems, enabling data connectivity to a WAN.

coaxial cable—The principal physical media over which CATV systems are built.

codec—coder-decoder. Device that typically uses pulse code modulation to transform analog signals into a digital bit stream and digital signals back into analog signals. In Voice over IP, it specifies the voice coder rate of speech for a dial peer.

CPE—customer premises equipment

CS-ACELP—Conjugate Structure Algebraic Code Excited Linear Prediction. A CELP voice compression algorithm specified in ITU-T Recommendation G.729, providing 8 kbps, or 8:1 compression.

CSI—called subscriber identification. An identifier whose coding format contains the telephone number from the remote fax terminal.

CSO—Composite Second Order Beat. The peak of the average level of distortion products due to second-order non-linearities in cable system equipment.

CTB—Composite Triple Beat. The peak of the average level of distortion components due to third-order non-linearities in cable system equipment.

dB—decibel. A measure of the relative strength of two signals.

dBm—Decibels with respect to 1 milliwatt. A unit of RF signal strength used in satellite work and other communications applications.

dBmV—Decibels with respect to one millivolt in a 75-ohm system. The unit of RF power used in CATV work in North America.

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

DID—direct inward dial. Allows a user outside a company to dial an internal extension number without needing to pass through an operator or attendant. The dialed digits are passed to the PBX, which then completes the call.

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

DLCI—Data-Link Connection Identifier. Frame Relay virtual circuit number corresponding to a particular destination. The DLCI is part of the Frame Relay header and is usually 10 bits long.

DNIS—dialed number identification service. Feature of trunk lines where the called number is identified; this called number information is used to route the call to the appropriate service.

DNS—Domain Name System used to address translation to convert H.323 IDs, URLs, or e-mail IDs to IP addresses. DNS is also used to assist in the location of remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

DOCSIS—Data Over Cable Service Interface Specification. Defines technical specifications for equipment at both subscriber locations and cable operators' headends.

downstream—Frequency multiplexed band in a CATV channel that distributes signals from a headend facility to subscribers.

DS0—A 64-bps channel on an E1 or T1 WAN interface.

DSN—delivery status notification. Message returned to the originator indicating the delivery status of an e-mail message. Three types of delivery status notifications can be requested by a sender: delay, success, and failure. Specifications for DSN are described in RFC 1891, RFC 1892, RFC 1893, and RFC 1894.

DSP—digital signal processor.

DTMF—dual tone multifrequency. Use of two simultaneous voice-band tones for dial (such as touch tone).

DTMF relay—Enables the generation of FRF.11 Annex A frames for a VoFR dial peer. The DSP generates Annex A frames instead of passing a DTMF tone through the network as a voice sample.

DVM—digital voice module (Cisco MC3810).

dynamic switched call—A telephone call dynamically established across a packet data network based on a dialed telephone number. In the case of VoFR, a Cisco proprietary session protocol similar to Q.931 is used to achieve call switching and negotiation between calling endpoints. The proprietary session protocol runs over FRF.11-compliant subchannels.

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

E1—Wide-area digital transmission scheme. E1 is the European equivalent of a T1 line. The E1's higher clock rate (2.048 MHz) allows for 32 64-kbps channels, which include one channel for framing and one channel for D-channel information.

E&M—recEive and transMit (or Ear and Mouth). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. The Cisco E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines).

endpoint—An H.323 terminal or gateway. An endpoint can call and be called. It generates or terminates the information stream.

ESMTP—Extended Simple Mail Transfer Protocol. Extended version of the Simple Mail Transfer Protocol (SMTP), which includes additional functionality such as delivery notification and session delivery. ESMTP is described in RFC 1869, *SMTP Service Extensions*.

FDM—frequency division multiplexing. A data transmission method in which a number of transmitters share a transmission medium, each occupying a different frequency.

FEC—Forward Error Correction. In data transmission, a process by which data is added that is derived from the payload by an assigned algorithm. It allows the receiver to determine if certain classes of errors have occurred in transmission and, in some cases, allows other classes of errors to be corrected.

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.

FRF—Frame Relay Forum. An association of corporate members consisting of vendors, carriers, users, and consultants committed to the implementation of Frame Relay in accordance with national and international standards. Refer to the web site <http://www.frforum.com>.

FRF.11—Frame Relay Forum implementation agreement for Voice over Frame Relay (v1.0 May 1997). This specification defines multiplexed data, voice, fax, DTMF digit-relay and CAS/robbed-bit signalling frame formats, but does not include call setup, routing, or administration facilities. Refer to the web site <http://www.frforum.com>.

FRF.11 Annex C—See FRF.12.

FRF.11 trunk—A point-to-point permanent voice connection (private line) conforming to the FRF.11 specification.

FRF.12—The FRF.12 Implementation Agreement (also known as FRF.11 Annex C) was developed to allow long data frames to be fragmented into smaller pieces and interleaved with real-time frames. In this way, real-time voice and nonreal-time data frames can be carried together on lower speed links without causing excessive delay to the real-time traffic. Refer to the web site <http://www.frforum.com>.

FXO—foreign exchange office. An FXO interface connects to the PSTN's central office and is the interface offered on a standard telephone. The Cisco FXO interface is an RJ-11 connector that allows an analog connection to be directed at the PSTN central office. This interface is of value for off-premises extension applications.

FXS—foreign exchange station. An FXS interface connects directly to a standard telephone and that 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.

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

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

G.726—Describes ADPCM coding at 40, 32, 24, and 16 kbps. ADPCM-encoded voice can be interchanged between packet voice, PSTN, and PBX networks if the PBX networks are configured to support ADPCM. Described in the ITU-T standard in its G-series recommendations.

G.728—Describes a 16-kbps low-delay variation of CELP voice compression. CELP voice coding must be translated into a public telephony format for delivery to or through the PSTN. Described in the ITU-T standard in its G-series recommendations.

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

gatekeeper—A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup, and request admission to a call from the gatekeeper. The gatekeeper is an H.323 entity on the LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper may provide other services to the H.323 terminals and gateways, such as bandwidth management and locating gateways.

gateway—A gateway allows H.323 terminals to communicate with non-H.323 terminals by converting protocols. A gateway is the point at which a circuit-switched call is encoded and repackaged into IP packets. An H.323 gateway is an endpoint on the LAN that provides real-time two-way communications between H.323 terminals on the LAN and other ITU-T terminals in the WAN, or to another H.323 gateway.

Group 3—Standard created by the International Telecommunication Union Telecommunications (ITU-T) relating to fax devices. A Group 3 fax device is a digital machine containing a 14400 baud modem that can send an 8 1/2 by 11-inch page in approximately 20 seconds with a resolution of either 203 by 98 dots per inch (dpi) or 203 by 196 dpi (fine), using Huffman code to compress fax data. Group 3 faxes use a standard dial up telephone line for transmission.

H.323—An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. 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.

H.323 RAS—registration, admission, and status. The RAS signalling function performs registration, admissions, bandwidth changes, status and disengage procedures between the VoIP gateway and the gatekeeper.

headend—Central distribution point for a CATV system. Video signals are received here from satellite (either collocated or remote), and frequency is converted to the appropriate channels, combined with locally originated signals, and rebroadcast onto the HFC plant. For a CATV data system, the headend is the typical place to create a link between the HFC system and any external data networks.

HFC—hybrid fiber-coaxial cable. Distribution cabling concept using both fiber-optic and coaxial cable. Older CATV systems were provisioned using only coaxial cable. Modern systems use fiber transport from the headend to an optical node located in the neighborhood to reduce system noise. Coaxial cable runs from the node to the subscriber. The fiber plant is generally a star configuration with all optical node fibers terminating at a headend. The coaxial cable part of the system is generally a trunk-and-branch configuration.

hookflash—A short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone wishes to perform dial-tone recall from a PBX. Hookflash is often used to perform call transfer.

host—Any end-user computer system that connects to a network. In this document, the term host refers to the computer system connected to the LAN interface of the cable access router.

HSRP—Hot Standby Routing Protocol. HSRP is a Cisco proprietary protocol that provides a redundancy mechanism when more than one router is connected to the same segment/subnet of an Ethernet/FDDI/Token Ring network.

IF—intermediate frequency. Intermediate electromagnetic frequencies generated by a superheterodyne radio receiver.

ingress noise—Over-the-air signals that are inadvertently coupled into the nominally closed coaxial cable distribution system. Ingress noise is difficult to track down and intermittent in nature.

ISDN—Integrated Services Digital Network. ISDN is a communications protocol, offered by telephone companies, that permits telephone networks to carry data, voice, and other traffic.

ITU—International Telecommunication Union. An organization established by the United Nations to set international telecommunications standards and allocate frequencies for specific uses.

ITU-T—Telecommunication standardization sector of ITU.

IVR—interactive voice response. A software feature that allows the use of one of several interactive voice response scripts during the call processing functionality.

LCD—liquid crystal display. An alphanumeric display on computers and fax devices using liquid crystal sealed between two pieces of glass.

LD-CELP—low-delay CELP. A CELP voice compression algorithm specified in ITU-T Recommendation G.728, providing 16 kbps, or 4:1 compression.

LEC—local exchange carrier.

LRQ—location request.

MAC layer—Media Access Control sublayer. Controls access by the cable access router to the CMTS and to the upstream data slots.

MCU—multipoint control unit.

MCNS—Multimedia Cable Network System Partners Ltd. A consortium of cable companies providing service to the majority of homes in the United States and Canada. This consortium has decided to drive a standard with the goal of having interoperable cable access routers.

MDN—message disposition notification. Message returned to the originator of an e-mail message indicating that the e-mail message has been opened. Specifications for MDN are described in RFC 2298.

MEL CAS—Mercury Exchange Limited (MEL) channel-associated signalling. CAS is a voice signalling protocol used primarily in the United Kingdom.

MFT—Multiflex Trunk (Cisco MC3810).

MICA—Modem ISDN channel aggregation. Modem module and card used in the Cisco AS5300 universal access servers. A MICA technologies modem provides an interface between an incoming or outgoing digital call and an Integrated Services Digital Network (ISDN) telephone line; the call need have be converted to analog as it does with a conventional modem and an analog telephone line. Each line can accommodate, or aggregate, up to 24 (T1) or 30 (E1)calls.

MIME—Multipurpose Internet Mail Extension. MIME is the standard for sending nontext data (or data that cannot be represented in plain ASCII code) in Internet mail, such as binary, foreign language text (such as Russian), audio, or video data. MIME is defined in RFC 2045.

MMoIP—Multimedia Mail over Internet Protocol. Dial peer specific to Store and Forward Fax. The MMoIP dial peer is the vehicle you use to assign particular line characteristics (such as a destination telephone number) to the connection between the Cisco AS5300 and the SMTP mail server during on-ramp faxing.

MPEG—Moving Pictures Experts Group, a joint committee of ISO and the International Electrotechnical Commission. MPEG is more commonly known as the series of hardware and software standards involving the reduction of storage requirements (compression schemes) for full-motion video.

MSO—Multiple System Operator. A cable service provider that operates in more than one geographic area, thus having multiple headend facilities.

MTA—Message Transfer Agent. Software that implements SMTP and provides storage for mail messages to be forwarded or delivered to a local user. MTAs implement SMTP (RFC 821).

multicast—A process of sending PDUs from one source to many destinations. The actual mechanism (that is, IP multicast, multi-unicast, and so on) for this process may be different for LAN technologies.

Multilink PPP—Multilink Point-to-Point Protocol. This protocol is a method of splitting, recombining, and sequencing datagrams across multiple logical data links.

multipoint-unicast—A process of transferring protocol data units (PDUs) where an endpoint sends more than one copy of a media stream to different endpoints. This may be necessary in networks that do not support multicast.

narrowband—A single RF frequency.

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

NTSC—National Television Systems Committee. A United States TV technical standard, named after the organization that created the standard in 1941. Specifies a 6 MHz-wide modulated signal.

OOS—Out of service signalling.

PAL—Phase Alternating Line. The TV system used in most of Europe, in which the color carrier phase definition changes in alternate scan lines. Utilizes an 8 MHz-wide modulated signal.

PBX—private branch exchange. Privately owned central switching office.

PDU—protocol data units. Used by bridges to transfer connectivity information.

permanent calls—Permanent calls are private line calls used for fixed point-to-point calls, connections between PBXs (E&M to E&M), or for remote telephone extensions (FXO to FXS).

PLAR—private line auto ringdown. This type of service results in a call attempt to some particular remote endpoint when the local extension is taken off-key.

POTS—plain old telephone service. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the Public Switched Telephone Network.

POTS dial peer—Dial peer connected via a traditional telephony network. POTS peers point to a particular voice-port on a voice network device. Also, a “plain old telephone service” dial peer used as a vehicle to assign particular line characteristics to the connection between the user and the receiving Cisco AS5300 during inbound (or on-ramp) faxing and to the connection between the Cisco AS5300 and the receiving fax device during outbound (or off-ramp) faxing.

PRI—Primary Rate Interface. PRI is an ISDN interface to primary rate access. Primary rate access consists of a single 64-kbps D channel plus 23 T1 or 30 E1 B channels for voice or data.

PSTN—Public Switched Telephone Network. PSTN refers to the local telephone company.

PVC—permanent virtual circuit.

QAM—Quadrature Amplitude Modulation. A method of modulating digital signals onto a radio-frequency carrier signal in which the value of a symbol consisting of multiple bits is represented by amplitude and phase states of the carrier. QAM is a modulation scheme mostly used in the downstream direction (64-QAM, 256-QAM); 16-QAM is often usable in the upstream direction. Numbers indicate number of code points per symbol. The QAM rate or the number of points in the QAM constellation can be computed by 2 raised to the power of <number of bits/symbol>. For example, 16-QAM has 4 bits per symbol, 64-QAM has 6 bits per symbol, and 256-QAM has 8 bits per symbol.

QoS—quality of service. QoS refers to the measure of service quality provided to the user.

QPSK—Quadrature Phase-Shift Keying. A digital modulation technique used in modems and wireless networks, allowing the transmission of 2 bits per symbol. QPSK provides a 2:1 compression ratio, resulting in double efficiency for the circuit being used.

ranging—The process of acquiring the correct timing offset such that the transmissions of a cable access router are aligned with the correct minislot boundary.

RAS—Registration, Admission, and Status Protocol. This is the protocol that is used between endpoints and the gatekeeper to perform management functions.

RBS—robbed-bit signalling.

Redialer—Interface hardware device that interconnects between a fax device and a Public Switched Telephone Network (PSTN) network. A redialer is used to forward a dialed number to another destination. Redialers contain a database of referral telephone numbers. When the user dials a specific number, the redialer collects the dialed digits and matches them to a listing in its database. If there is a match, the redialer dials the referral number (transparent to the user) and forwards the call to the referral number.

RF—radio frequency. The portion of the electromagnetic frequency spectrum having wavelengths between the audio and light range, usually between 5 MHz and approximately 860 MHz.

RRQ—registration request.

RSVP—Resource Reservation Protocol. This protocol supports the reservation of resources across an IP network.

SECAM—TV system used in France and elsewhere, utilizing an 8 MHz-wide modulated signal.

SID—service ID. A number that defines (at the MAC sublayer) a particular mapping between a cable access router (CM) and the CMTS. The SID is used for the purpose of upstream bandwidth allocation and class-of-service management.

signal-to-noise—S/N (also SNR). The difference in amplitude between a baseband signal and the noise in a portion of the spectrum.

SMTP—Simple Mail Transfer Protocol. Application-level protocol in the TCP/IP protocol suite involved with the transmission and reception of electronic mail. Specifications for SMTP are described in RFC 821 and RFC 822.

spectrum reuse—CATV's most fundamental concept. Historically, the over-the-air spectrum has been assigned to many purposes other than that of carrying TV signals. This has resulted in an inadequate supply of spectrum to serve the needs of viewers. Cable can reuse spectrum that is sealed in its aluminum tubes.

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

SU—subscriber unit. An alternate term for cable access router.

SVC—switched virtual circuit.

switched calls—Switched calls are normal telephone calls in which a user picks up a telephone, hears dial tone, enters the destination telephone number to reach the other telephone. Switched calls can also be private line, auto-ringdown (PLAR) calls, or tie-line calls for fixed E&M to E&M fixed point-to-point connections.

symbol—Phase range of a sine wave.

tandem switching—The dynamic switching of voice calls between VoFR, VoATM, or VoHDL PVCs and subchannels; also called tandeming. Tandem switching is often encountered in multihop VoFR call connection paths.

T1—Digital WAN carrier facility. T1 sends DS1 formatted data at 1.544 Mbps through the telephone-switching network, using AMI or B8ZS coding. T1 is the North American equivalent of an E1 line.

TCL—Tool Command Language. An interpreted script language developed by Dr. John Ousterhout of the University of California, Berkeley, and now developed and maintained by Sun Microsystems Laboratories.

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.

trunk—Service that allows quasitransparent connections between two PBXs, a PBX and a local extension, or some other combination of telephony interfaces with signalling passed transparently through the packet data network.

TSI—transmitting subscriber information. A frame that can be sent by the caller with the caller telephone number that can be used to screen calls.

UIO—universal I/O serial port (Cisco router).

U-law—A companding technique commonly used in North America. U-law is standardized as a 64-kbps codec in ITU-T G.711.

upstream—Frequency multiplexed band in a CATV channel that distributes signals from subscribers to the headend. In this instance, upstream refers to the data flow from a cable modem to the Cisco MC11 modem card in a Cisco uBR7200 series.

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

VBR—variable bit rate.

VoATM dial peer—Dial peer connected via an ATM network. VoATM peers point to specific VoATM devices.

VoFR—Voice over Frame Relay.

VoFR dial peer—Dial peer connected via a Frame Relay network. VoFR peers point to specific VoFR devices.

Voice over Frame Relay—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.

Voice over IP—Voice over IP enables a router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In Voice over IP, 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—Voice over IP.

VoIP dial peer—Dial peer connected via a packet network; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.

VTSP—Voice telephony service provider.

zone—A collection of all terminals (tx), gateways (GW), and Multipoint Control Units (MCUs) managed by a single gatekeeper (GK). A zone includes at least one terminal, and may include gateways or MCUs. A zone has only one gatekeeper. A zone may be independent of LAN topology and may comprise multiple LAN segments that are connected using routes or other devices.

