Glossary and Terms

802.1p
An IEEE standard for providing QoS using three bits (defined in 802.1q) to allow switches to reorder packets based on priority level.

802.1q
An IEEE standard for providing virtual LAN (VLAN) identification and QoS levels. Three bits are used to allow eight priority levels, and 12 bits are used to identify up to 4,096 VLANs.

AHT (Average Hold Time)

The average length of time between the moment a caller finishes dialing and the moment the call is answered or terminated.

ANI (Automatic Number Identification)

A telephone function which transmits the billing number of the incoming call (Caller ID, for example).

ANSI (American National Standards Institute)

The American standardization body known for interface recommendations and standardization of programming languages. ANSI is a non-profit making, government-independent organization.

AS (Autonomous System)

A group of networks under mutual administration that share the same routing methodology.

ASP (Application Service Provider)

An independent, third party provider of software-based services delivered to customers across a wide area network (WAN).

ASR (Answer-Seizure Ratio)

The ratio of successfully connected calls to attempted calls (also called 'Call Completion Rate').

ATA (Analogue Telephone Adapter)

Used to connect a standard telephone to a high-speed modem to facilitate VoIP and/or fax calls over the Internet.

ATM (Asynchronous Transfer Mode)

A technology for switched, connection-oriented transmission of voice, data and video. It makes high-speed dedicated connections possible between a theoretically unlimited number of network users and also to servers

Backbone

A high-speed network spanning the world from one major metropolitan area to another.

Bad Frame Interpolation

Interpolates lost/corrupted packets by using the previously received voice frames. It increases voice quality by making the voice transmission more robust.

Bandwidth

The maximum data carrying capacity of a transmission link. For networks, bandwidth is usually expressed in bits per second (bps).

Billing Increment

A call duration measurement unit, usually expressed in seconds.

BLI (Busy Lamp Indicator)

A light or LED on a telephone that shows which line is in use.

Broadband

A descriptive term for evolving digital technology that provides consumers a single switch facility offering integrated access to voice, high-speed data service, video demand services, and interactive delivery services.

Call deflection

Call Deflection allows a called endpoint to redirect the unanswered call to another endpoint.

Call Detail Record (CDR)

Information regarding a single call collected from the switch and available as an automatically generated downloadable report for a requested time period. The report contains information on the number of calls, call duration, call origination and destination, and billed amount.

Codec (Compression-decompression)

In VoIP it is a voice compression-decompression algorithm that defines the rate of speech compression, quality of decompressed speech and processing power requirements. The most popular codecs in VoIP are ITU-T G.723.1 and G.729 (AB).

Compression

Compression is used at anywhere from 1:1 to 12:1 ratios in VOIP applications to consume less bandwidth and leave more for data or other voice/fax communications. The voice quality may decrease with increased compression ratios.

Congestion

The situation in which the traffic present on the network exceeds available network bandwidth/capacity.

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Dial-peer (Addressable Call Endpoint)

A software structure that binds a dialed digit string to a voice port or IP address of the destination gateway. Several dial peers always exist on each router in the network, and at least two will be involved in making a call across the network, one on the originating end and one on the terminating end. In Voice over IP, there are two kinds of dial peers: POTS and VoIP. VoIP peers point to specific VoIP devices.

Dial-peer hunting

Process when the originating router tries to establish call on different dial peers if the originating router receives a user-busy invalid number or an unassigned-number disconnect cause code from a destination router.

DiffServ (Differentiated Services)

A quality of service (QoS) protocol that prioritizes IP voice and data traffic to help preserve voice quality, even when network traffic is heavy.

DNIS (Dialed Number Identification Service)

A telephone function which sends the dialed telephone number to the answering service.

DTMF (Dual-Tone Multi Frequency)

The type of audio signals generated when you press the buttons on a touch-tone telephone.

Dynamic Jitter Buffer

Collects voice packets, stores them, and shifts them to the voice processor in evenly spaced intervals to reduce any distortion in the sound.

E&M (Ear and Mouth)

Is the interface on a VOIP device that allows it to be connected to analog PBX trunk ports (tie lines).

E.164

The international public telecommunication numbering plan. An E.164 number uniquely identifies a public network termination point and typically consists of three fields, CC (country code), NDC (national destination code), and SN (subscriber number), up to 15 digits in total.

E1

A wide-area digital transmission scheme (European): 2,048 Mbits/s; 31 channels, 64 Kbps each.

Endpoint

SIP or H.323 terminal or Gateway. An endpoint can Call and be Called. It generates and terminates the information stream.

Firewall

A system designed to prevent unauthorized access to or from a private network. Firewalls can be implemented as hardware, software, or a combination of both. All messages entering or leaving the intranet pass through the firewall, which examines each message and blocks those that do not meet the security criteria specified on the firewall.

FoIP (Fax Over Internet Protocol)

The term used for the technology that transports facsimiles over the Internet.

Forward Error Correction

Increases voice quality by recovering lost or corrupted packets.

FXO (Foreign Exchange Office)

Is the interface on a VOIP device for connecting to an analog PBX extension.

FXS (Foreign Exchange Station)

Is the interface on a VOIP device for connecting directly to phones, faxes, and CO ports on PBXs or key telephone systems.

G.711

An ITU-T PCM half-duplex codec that uses either A-law or ?-law compression (64 kbps, high quality, minimum processor load).

G.723.1

An ITU-T double rate CELP codec (6.4/5.3 kbps, medium quality, high processor load).

G.726

An ITU-T ADPCM wave form codec (16/24/32/40 kbps, good quality, low processor load).

G.728

An ITU-T low delay CELP codec (16 kbps, medium quality, very high processor load).

G.729

An ITU-T ACELP codec (8 kbps, medium quality, high processor load).

G.7xx

A family of ITU standards for audio compression.

Gatekeeper

The central control entity that performs management functions in a Voice and Fax over IP network and for multimedia applications such as video conferencing. Gatekeepers provide intelligence for the network, including address resolution, authorization, and authentication services, the logging of Call Detail Records, and communications with network management systems. Gatekeepers control bandwidth, provide interfaces to existing legacy systems, and monitor the network for engineering purposes as well as for real-time network management and load balancing.

Gateway

In IP telephony, a network device that converts voice and fax calls, in real time, between the public switched telephone network (PSTN) and an IP network. The primary functions of an IP gateway include voice and fax compression/ decompression, packetization, call routing, and control signaling. Additional features may include interfaces to external controllers, such as Gatekeepers or Softswitches, billing systems, and network management systems.

Grace Period

The time interval at the beginning of a call, measured in seconds that is not billed.

H.225

Protocols (RAS, RTP/RTCP, Q.931 call signaling) and message formats for H.323.

H.245

A protocol for capability negotiation, messages for opening and closing channels for media streams, etc. (i.e. media signaling).

H.323

An ITU-T "umbrella" of standards for Packet-based multimedia communications systems. This standard defines the different multimedia entities that make up a multimedia system - Endpoints, Gateways, Multipoint Conferencing Units (MCUs), and Gatekeepers -- and their interaction. This standard is used for many Voice-over-IP applications, and is heavily dependent on other standards, mainly H.225 and H.245.

Hairpin

Telephony term that means to send a call back in the direction that it came from. For example, if a call cannot be routed over IP to a gateway that is closer to the target telephone, the call typically is sent back out the local zone, back the way from which it came.

Hop off

Point at which a call transitions from H.323 to non-H.323, typically at a gateway.

IETF (Internet Engineering Task Force)

One of two technical working bodies in the Internet Activities Board. The IETF meets three times a year to set technical standards for the Internet.

IMS

IP Multimedia Subsystem

- AS – Application Server
- SCIM - Service Capability Interaction Manager
- MRFC - Multimedia Resource Function Controller
- MRFP - Multimedia Resource Function Processor
- CSCF- Call Session Control Function
- BGCF - Breakout Gateway Control Function
- MGCF - Media Gateway Control Function
- MGW - Media Gateway
- HSS - Home Subscription Server
- HLR - Home Location Register

Integrated T-1

Comprised of 24 64Kbps channels, T1 lines can be used for a diverse number of applications. Commonly referred to as an integrated T1 or channelized T1, this highly flexible circuit is designed for businesses that need to run multiple services over the same line. Common applications for integrated T1 service include, Frame Relay/dedicated long distance and Internet/point-to-point. Often confused with a fractional T1, integrated service is made up of multiple fractional T1 services.

IP Centrex

IP Centrex delivers such services as call hold, call transfer, last number look-up and redial, call forward, three-way calling, but does it on a packet-based network.

IP Telephony

The transmission of voice and fax phone calls over data networks that uses the Internet Protocol (IP). IP telephony is the result of the transformation of the circuit-switched telephone network to a packet-based network that deploys voice-compression algorithms and flexible and sophisticated transmission techniques, and delivers richer services using only a fraction of traditional digital telephony's usual bandwidth.

ITSP (Internet Telephony Service Provider)

Provider of telephony based services.

ITU-T

ITU standards for telecommunications.

Jitter

The variation in the amount of Latency among Packets being received.

LAN (Local Area Network)

A LAN is a group of computers and associated devices that share a common communications line or wireless link and typically share the resources of a single processor or server within a small geographic area (for example, within an office building).

Latency

Also called Delay. The amount of time it takes a Packet to travel from source to destination. Together, Latency and Bandwidth define the speed and capacity of a network.

MGCP Media Gateway Control Protocol
A protocol for IP telephony that enables a caller with a PSTN phone number to locate the destination device and establish a session.

MGCP (Media Gateway Control Protocol)

A protocol complementary to H.323 and SIP, designed to control media gateways from external call control elements in decomposed gateway architectures. MGCP is meant to simplify standards for the new Voice over Packet technology by eliminating the need for complex, processor-intense IP telephony devices, thus simplifying and lowering the cost of these terminals.

P2P

A peer-to-peer (or P2P) computer network relies primarily on the computing power and bandwidth of the participants in the network rather than concentrating it in a relatively low number of servers. P2P networks are typically used for connecting nodes via largely ad hoc connections. Such networks are useful for many purposes. Sharing content files containing audio, video, data or anything in digital format is very common, and real-time data, such as telephony traffic, is also passed using P2P technology.

Packet

In data communication, the basic unit of information transferred.

PBX (Private Branch Exchange)

An in-house telephone switching system that interconnects telephone extensions to each other, as well as to the outside telephone network.

PRI (Primary Rate Interface)

An ISDN service that provides 23 64-Kbps B (Bearer) channels and one 64-Kbps D (Data) channel (23 B and D).

PSTN

Public Switched Telephone Network.

Q.931

ISDN connection control protocol, roughly comparable to TCP in the Internet protocol stack. Q.931 doesn't provide flow control or perform retransmission, because the underlying layers are assumed to be reliable and the circuit-oriented nature of ISDN allocates bandwidth in fixed increments of 64 kbps. Q.931 does manage connection setup and breakdown. In H.323 scenario, this protocol is encapsulated in TCP and sent to port 1720.

QoS (Quality of Service)

Measure of performance for a transmission system that reflects it's transmission quality and service availability. Standards based QOS for VoIP usually involves the implementation of Ethernet standards 802.1p and 802.1q at layer 2 across an Ethernet.

QSIG (Q (point of the ISDN model) Signaling)

Signaling standard. Common channel signaling protocol based on ISDN Q.931 standards and used by many digital PBXs.

RAS (Registration, Admission, Status)

A management protocol between terminals and Gatekeepers.

Redundant

Redundant describes computer or network system components, such as fans, hard disk drives, servers, operating systems, switches, and telecommunication links that are installed to back up primary resources in case they fail.

RSVP (Resource Reservation Protocol)

A 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.

RTP (Real-Time Transport Protocol)

Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time stamping, and delivery monitoring to real-time applications.

SIP (Session Initiation Protocol)

An application-layer control protocol, a Signaling protocol for Internet Telephony. SIP can establish sessions for features such as audio/videoconferencing, interactive gaming, and call forwarding to be deployed over IP networks thus enabling service providers to integrate basic IP telephony services with Web, e-mail, and chat services. In addition to user authentication, redirect and registration services, SIP Server supports traditional telephony features such as personal mobility, time-of-day routing and call forwarding based on the geographical location of the person being called.

Softswitch

Also called a Proxy Gatekeeper, Call Server, Call Agent, Media Gateway Controller, or Switch Controller. Software used to bridge a public switched telephone network and voice over Internet by separating the call control functions of a phone call from the media gateway (transport layer). Softswitch performs call control functions such as protocol conversion, authorization, accounting and administration operations.

T1

1.544-Mbps point-to-point dedicated digital circuit provided by the telephone companies consisting of 24 channels.

TAPI (Telephony API)

A programming interface that allows Windows client applications to access voice services on a server.

TCP (Transmission Control Protocol)

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

Trunk

A communications channel between two points, typically referring to large-bandwidth telephone channels between switching centers, that handle many simultaneous voice and data signals.

Trunking

Trunking means that several connections in a network may be established simultaneously, and that setup of connections proceeds automatically using the channels available at the time in question. In this way many users may share a few connections, and if the number of connections is increased, the capacity of the network is increased more than proportionally. This means that an optimal trunking effect is obtained in very large networks.

VoIP (Voice Over Internet Protocol)

Transportation of voice calls across the Internet.

VPDN (Virtual Private Dial-up Network)

Also known as virtual private dial network. A VPDN is a network that extends remote access to a private network using a shared infrastructure. VPDNs use Layer 2 tunnel technologies (L2F, L2TP, and PPTP) to extend the Layer 2 and higher parts of the network connection from a remote user across an ISP network to a private network. VPDNs are a cost effective method of establishing a long distance, point-to-point connection between remote dial users and a private network

VPN

Virtual Private Network. Enables IP traffic to travel securely over a public TCP/IP network by encrypting all traffic from one network to another. A VPN uses “tunneling” to encrypt all information at the IP level.