VoIP Glossary

ADSI - Analog Display Services Interface
A complex set of standards for the telecom industry. Session based applications can be used when the phone is online, or scripts can be preprogrammed into the phone for when no ADSI connection has been established (on or off hook).

ATA - Analog Telephone Adapter
A device used to connect a standard telephone to a computer or network so that the user can make calls over the Internet. The ATA is an analog-to-digital converter. An ATA that connects telephones to a local area network (LAN) is sometimes called a VoIP gateway.

ATM - Asynchronous Transfer Mode
A network technology based on transferring data in cells or packets of a fixed size. The cell used with ATM is relatively small compared to units used with older technologies. The small, constant cell size allows ATM equipment to transmit video, audio, and computer data over the same network, and assure that no single type of data hogs the line.

CBR - Constant Bit Rate
An ATM bandwidth-allocation service that requires the user to determine a fixed bandwidth requirement at the time the connection is set up so that the data can be sent in a steady stream. CBR service is often used when transmitting fixed-rate uncompressed video. (A Class A quality of service.)

CLEC - Competitive Local Exchange Carrier
A telephone company that competes with an incumbent local exchange carrier (ILEC) such as a Regional Bell Operating Company (RBOC), GTE, ALLNET, etc.

CPE - Customer Premise Equipment
Communications equipment that is located on the customer's premises (i.e., is owned or leased by the customer) rather than on the service provider's premises. Telephone handsets and Digital Subscriber Line (DSL) routers are examples.

CODEC - Coder-decoder
Integrated circuits or chips that perform analog-to-digital conversion and digital-to-analog conversion functions. The process of converting analog waveforms to digital information is done through a codec algorithm. Output from a codec is a data stream that is put into IP packets and transported across the network to an endpoint.

Delay/Latency
Excessive end-to-end delay makes conversation inconvenient and unnatural. 150 ms (milliseconds) is the maximum desired one-way latency to achieve high-quality voice transmission. Excess latency can make the call incomprehensible.

DMZ - Demilitarized Zone
A computer host or small network inserted as a “neutral zone” between a company's private network and the outside public network. It prevents outside users from getting direct access to a server that has company data.

DNS - Domain Name System (or Service or Server)
An Internet service that translates domain names into IP addresses.

ENUM - Electronic Numbering

The ITU ENUM allocates a specific zone, namely e164.arpa for use with E.164 numbers. Any phone number, such as +1 555 42 42 can be transformed into a hostname by reversing the numbers, separating them with dots and adding the e164.arpa suffix, like so: 2.4.2.4.5.5.5.1.e164.arpa.DNS can then be used to look up Internet addresses for services such as SIP VoIP telephony. For example, NAPTR records are used to "translate" E.164 addresses to SIP addresses.

Gateway

A network point that acts as an entrance to another network. When using H.323 protocol, gateways serve to establish connections with the telephone network or a PBX system. The function of the gateway is to convert the various data formats in transport, process control and audio/video processing.

Gatekeeper

A Gatekeeper is used on an H323 network to provide user registration and call routing capabilities analogous to the SIP Server on a SIP network. It also lets network managers set policies and control network resources such as bandwidth utilization. An H323 Gatekeeper is necessary in order to provide routing on an H323 VoIPTalk network.

H.248

A collaborative standard by the IETF and ITU, Megaco/H.248 is used similarly to MGCP in providing control over Media Gateways (MGs) by Media Gateway Controllers (MGCs). (Megaco is the IETF name and H.248 is the ITU-T name.) Megaco/H.248 leverages and extends the IETF's Media Gateway Control Protocol (MGCP) and is It is the result of continued cooperation between the IETF Megaco Working Group and ITU Study Group 16.

H.323

H.323 is an umbrella recommendation from the ITU-T that defines the protocols to provide audio-visual communication sessions on any packet network.

IETF - Internet Engineering Task Force

The main standards organization for the Internet.

ILEC - Incumbent Local Exchange Carrier

A telephone company that was already providing local service when the Telecommunications Act of 1996 was enacted.

IP PBX - IP Private Branch Exchange

A PBX that switches calls between VoIP users on local lines while allowing all users to share a certain number of external phone lines. Because an IP PBX employs converged data and voice networks, Internet access, VoIP communications and traditional telephone communications are all possible using a single line to each user.

ITU - International Telecommunication Union

An intergovernmental organization through which public and private organizations develop telecommunications. It is responsible for adopting international treaties, regulations and standards governing telecommunications. The ITU-T (for Telecommunication Standardization Sector) is the primary international body for fostering cooperative standards for telecommunications equipment and systems.

Jitter

The variation in the time between packets' arrival, caused by network congestion, timing drift or route changes. Packets transmitted at equal intervals from one gateway arrive at the other gateway at irregular intervals; as a result, they may be reconstructed out of sequence, and some packets that arrive too late may be discarded. Excessive jitter makes speech choppy and difficult to understand.

MCU - Multipoint Control Unit
A multi-port device by which two or more audiovisual terminals may communicate in a conference call. MCUs allow for audio/video conferencing by collecting information about the capabilities of the systems at each of the conference endpoints and setting the conference at the lowest common denominator so that everyone can participate.

MG - Media Gateway
Any device, such as a circuit switch, IP gateway or channel bank that converts data from the format required for one type of network to the format required for another.

MGCP- Media Gateway Control Protocol
A new standard developed by Telcordia and Level 3 Communications, a control and signal standard to compete with the older H.323 standard for the conversion of audio signals carried on telephone circuits (PSTN) to data packets carried over the Internet or other packet networks.

MOS - Mean Opinion Score
A common benchmark used to determine the quality of sound produced by a specific codec. The mean opinion score (MOS) is rated on a scale of 1 (bad) to 5 (excellent).

MPLS - Multi-Protocol Label Switching
A standards-approved technology for speeding up network traffic flow and making it easier to manage. MPLS is an IETF initiative that integrates Layer 2 information about network links (bandwidth, latency and utilization) into Layer 3 (IP) within a particular autonomous system or Internet service provider (ISP) in order to simplify and improve IP-packet exchange. In addition to moving traffic faster overall, MPLS makes it easy to manage a network for quality of service (QoS).

MTU - Maximum Transmission Unit
The largest physical packet size, measured in bytes, that a network can transmit. Any messages larger than the MTU are divided into smaller packets before being sent.

NAPTR - Naming Authority Pointer
A way to encode rule-sets in DNS so that the delegated sections of a Uniform Resource Identifier (URI) <RFC 2396> could be decomposed, changed and re-delegated over time. The result was a Resource Record that included a regular expression that would be used by a client program to rewrite a string into a domain name.

Packet Loss
The misdirected or out-of-sequence arrival of one or more packets within a voice (or data) transmission. When a packet is not reassembled in the correct sequence for delivery to the (analog) listener, part of the conversation is missing or garbled. Packet loss typically occurs either in bursts or periodically due to a consistently congested network.

PBX - Private Branch Exchange
A private telephone network used within an enterprise. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX.

PCM - Pulse Code Modulation
A sampling technique for digitizing analog signals, especially audio signals. PCM samples the signal 8000 times a second; each sample is represented by 8 bits for a total of 64 Kbps. There are two standards for coding the sample level. The Mu-Law standard is used in North America and Japan while the A-Law standard is use in most other countries. PCM is used with T-1 and T-3 carrier systems. These carrier systems combine the PCM signals from many lines and transmit them over a single cable or other medium.

PESQ - Perceptual Evaluation of Speech Quality
PESQ is an objective measurement tool that predicts the results of subjective listening tests on telephony systems. PESQ uses a sensory model to compare the original, unprocessed signal with the degraded signal from the network or network element. The resulting quality score is analogous to the subjective “Mean Opinion Score” (MOS) measured using panel tests according to ITU-T P.800.

PSQM - Perceptual speech-quality measurement
Originally created to evaluate speech codecs, the PSQM algorithm provides a method by which speech within the voice bandwidth of 300 to 3400 Hz can be objectively measured for distortion, the effects of noise, and overall perceptual fidelity. Simply put, PSQM is an automated human listener.

PSTN- Public Switched Telephone Network
PSTN refers to the traditional international telephone system based on copper wires carrying analog voice data.

QoS - Quality of Service
The idea that transmission rates, error rates and other characteristics can be measured, improved and to some extent guaranteed in advance. QoS is of particular concern for the continuous transmission of high-bandwidth video and multimedia information. Transmitting this kind of content dependably is difficult in public networks using ordinary “best effort” protocols.

RTP - Real Time Transport Protocol
An Internet protocol for transmitting real-time data such as audio and video. RTP itself does not guarantee real-time delivery of data, but it does provide mechanisms for sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol, although the specification is general enough to support other transport protocols.

RTCP - Real Time Transport Control Protocol
RTCP is the optional companion protocol to RTP; it is not needed for RTP to work. Its primary function is to provide feedback on the quality of the data distribution transmitted by RTP.

SBM - Subnet Bandwidth Manager
A signaling technique used to manage network resources (bandwidth) on legacy and newer local area network (LAN) topologies. SBM manages network resources and uses Admission Control Services (ACS) to make traffic-flow decisions. A subnet is a portion of a network that shares a common address component. On TCP/IP networks, subnets are defined as all devices whose IP addresses have the same prefix.

SBC - Session Border Controller
A device that sits between two administrative domains (e.g., between two service providers) and is responsible for routing VoIP calls between the two domains and ensuring QoS. The SBC may also serve as a protocol interworking function between various VoIP protocols, including H.323 and SIP.

Sequence Errors
Congestion in packet switched networks can cause packets to take different routes to reach the same destination. Packets may arrive out of order resulting in garbled speech.

Softswitch - Software Switch
Any open application program interface (API) software used to bridge a traditional PSTN and VoIP by separating the call control functions of a phone call from the media gateway (transport layer) and managing traffic that contains a mixture of voice, fax, data and video. Softswitches cost about a tenth of regular local phone switches, take up much less space and enable carriers to use a single packet backbone for voice and data traffic.

SVZ - Secure Voice Zone
Like a DMZ, an SVZ establishes a protected zone for VoIP infrastructure and segregates the voice traffic from the data traffic.

SIP - Session Initiation Protocol
An IETF standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality. SIP establishes call parameters at either end of the communication, and handles call transfer and termination. Like HTTP or SMTP, SIP works in the Application layer of the Open Systems Interconnection (OSI) communications model.

TDM - Time-Division Multiplexing
A method of putting multiple data streams into a single signal by separating the signal into many segments of very short duration, each with a fixed time slot. Each individual data stream is reassembled at the receiving end based on the timing. Within T-Carrier systems such as T-1 and T-3, TDM combines Pulse Code Modulation (PCM) streams created for each conversation or data stream.

Telephony
The science of translating sound into electrical signals, transmitting them, and then converting them back to sound; that is, the science of telephones. The term is used frequently to refer to computer hardware and software that performs functions traditionally performed by telephone equipment.

UDP - User Datagram Protocol
A communications protocol that offers a limited amount of service when messages are exchanged between computers in a network that uses the Internet Protocol (IP). UDP is an alternative to the Transmission Control Protocol (TCP). UDP doesn't provide sequencing of the packets in which the data arrive. This means that the application program that uses UDP must be able to make sure that the entire message has arrived and is in the right order.

VBR - Variable Bit Rate
An ATM bandwidth-allocation service that allows users to specify a throughput capacity (i.e., a peak rate) and a sustained rate, but data is not sent evenly. VBR is often used when transmitting compressed packetized voice and video data such as videoconferencing. (A Class B quality of service.)

VoIP -Voice over Internet Protocol (IP)
A category of hardware and software that uses the Internet as the transmission medium for telephone calls by sending voice data in packets using IP rather than by traditional (analog) circuit transmissions of the public switched telephone network (PSTN). VoIP also is referred to as Internet telephony, IP telephony (IPT) or Voice over the Internet (VOI).

Definitions provided by online dictionaries including http://www.pcwebopedia.com/, http://whatis.techtarget.com/, http://searchnetworking.techtarget.com and http://www.techabulary.com.

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