VoIP Glossary

Analog audio signals: Analog audio signals are used to transmit voice data over telephone lines. This is done by varying or modulating the frequency of sound waves to accurately reflect the pitch of the sound. The same technology is used for radio wave transmissions.

ATA: ATA or the analog telephone adaptor is the hardware device that connects the conventional telephone to the Internet through a high speed bandwidth line, provides the interface to convert the analog voice signals into IP packets, delivers dial tone and manages the call setup.

Bandwidth: Bandwidth is the volume of data that can be transmitted over a communication line in a fixed amount of time. It is expressed in bits per second (bps) or bytes per second for digital devices and in cycles per second, or Hertz (Hz) for analog devices. Bandwidth can also be defined as the difference between a band of frequencies or wavelengths.

Broadband: It is a term used to define high speed Internet connection, generally provided by cable TV, DSL or dedicated telecom lines. The high speeds are achieved by the carrying capacity of the cable that can carry multiple messages simultaneously.

Cable modem: The cable modem is a device that is used to connect a computer to the high speed coaxial cable run by cable TV companies to provide access to the Internet. The connection is made through an Ethernet port, which is a shared medium and can affect download speeds if too many users log on simultaneously to the Internet on that particular cable segment. However, despite this cable modems provide extremely fast access to the net.

Circuit switched networks: These networks have been used for making phone calls since 1878. They use a dedicated point-to-point connection for each call. This reduces their utility because no network traffic can move across the switches that are being used to transmit a call.

Client (Softphone client): The software installed in the userâ€™s computer to make calls over the Internet.

Codec: Codec is a term that arises from the Compressor-Decompressor or enCOder/DECoder process. It is used for software or hardware devices that can convert or transform a data stream. For instance, at the transmitting end codecs can encode a data stream or data signal for easy transmission, storage or encryption. At the receiving end, they can decode the signal in the appropriate form for viewing. They are most suitable for videoconferencing and streaming media solutions.

Compression: This is a term that is used to indicate the squeezing of data in a format that takes less space to store or less bandwidth to transmit. It is very useful in handling large graphics, audio and video files.

Data compression: This is the process that is used to compress large data files into mall files so that they use less bandwidth during transmission and less disk space when stored. The compression depends upon the repeatable patterns of binary 0s and 1s. The higher the number of repeatable patters, the higher is the compression. The right compression codes can compress data files to 40% of their original size. The graphics files can be compressed even more â€“ from 20% to 90%.

DSL modem: A DSL modem is a device that is used to connect one or more computers to the high speed DSL line provided by a DSL operator to gain access to the Internet. The customers use these modems to log on the net to download or transmit data. Since the DSL lines have high bandwidth capacity the data transfer speeds are very high.

E911: **E911** is the short form of the term Enhanced 911, and is used for providing emergency service on cellular and Internet voice calls.

Emergency 911 calls: This is an emergency telephone number that handles all calls related to police, fire or medical emergencies. The number, which is allotted under the North American Numbering Plan (NANP), is answered by either a telephone operator or an emergency service dispatcher, who, in turn, alerts the appropriate emergency service.

H.323: An ITU standard that lays down guidelines for real time voice and videoconferencing utilities on the Internet. The H.323 standard supports voice, video, data, application sharing and whiteboarding and defines media gateways for conversion to packets.

Internet congestion: Internet congestion occurs when a large volume of data is being routed on low bandwidth lines or across networks that have high latency and cannot handle large volumes. The result is slowing down of packet movement, packet loss and drop in service quality.

IP address: An IP address, also known as **Internet Protocol** address, is the machine number used to identify all devices that are connected to the net. Each device has its own unique number which it uses to communicate. This number is fixed in the case of those computing devices that have a fixed IP address. The rest are allotted a dynamic IP address, which is valid for the period they are connected to the net. The numbers range from 0.0.0.0 to 255.255.255.255.

IP mapping: IP mapping is the process of identifying IP addresses on the basis of their geographical locations. The mapping enables web administrators to pinpoint the location of any computing device connected to the Internet.

IP Phone: An **IP phone** is one that converts voice into digital packets and vice versa to make phone calls over Internet possible. It has built-in IP signaling protocols such as H.323 that ensure that the voice is routed to the right destination over the net. The IP phones come with several value added services like voicemail, e-mail, call number blocking etc.

IP telephony: IP telephony refers to the two-way transmission of voice over Internet. The voice is transmitted in real time by using the packet-switched technology over the IP network. Some of the applications that use IP telephony are IP-based phone services, voice over instant messaging and videoconferencing.

IP: IP, which is the acronym for **Internet Protocol**, defines the way data packets, also called datagrams, should be moved between the destination and the source. More technically, it can be defined as the network layer protocol in the TCP/IP communications protocol suite.

ITU: ITU, which is the acronym of International Telecommunication Union, is a telecommunications standards body based in Geneva. It works under the aegis of the United Nations and makes recommendations on standards in telecommunications, information technology, consumer electronics, broadcasting and multimedia communications.

Jitter: It is a term used to indicate a momentary fluctuation in the transmission signal. This happens in computing when a data packet arrives either ahead or behind a standard clock cycle. In telecommunication, it may result from an abrupt variation in signal characteristics, such as the interval between successive pulses.

Kbps: Kbps is the acronym for kilobits per second and is used to indicate the data transfer speed. If the modem speed, for instance, is 1 Kbps then it means that the modem can route data at the speed of one thousand bits per second.

Lag: Lag is the term used to indicate the extra time taken by a packet of data to travel from the source computer to the destination computer and back again. The lag may be caused by poor networking or by inefficient or excessive processing.

Latency: Latency is the time that elapses between the initiation of a request for data and the start of the actual data transfer. This delay may be in nanoseconds but it is still used to judge the efficiency of networks.

Mapping: The process of identifying all related data fields or data streams and putting them in an easily identifiable context. For example, IP mapping enables users to pinpoint the geographical location of any computing device on the Internet.

MGCP: Acronym of Media Gateway Control Protocol. Used for a Voice over IP system. It consists of a Call Agent and a set of gateways, of which at least one works as the “media gateway” and performs the conversions.

NANP: Stands for North American Numbering Plan. A telephone numbering system that has evolved the way area codes and numbers are allotted. The system was established in 1947 and covers the United States, Canada and a few neighboring areas. It uses a three-digit area code and seven-digit telephone numbers. Its fiat is, however, limited to the public switched telephone networks only.

Net Phone: A net phone uses the Voice over IP technology to make voice calls. These calls are made by converting analog sound signals into digital data packets, and then moving the packets to their destination over the net.

Packet loss: Packet loss is the term used to indicate the loss of data packets during transmission over a computer network. This may happen on account of high network latency or on account of overloading of switches or routers that are unable to process or route all the incoming data.

Packet switched networks: These are networks that break messages into small digital packets, stamp each packet with the destination IP address, and route them across different channels to their destination where they are reassembled in their proper sequence. This is done to avoid network congestion and speed up data movement from multiple sources.

Packet: A packet is a unit of data transmitted over the network in a packet-switched system. It consists of a header that stores the destination address, a data area which carries the information that is being transmitted, and a trailer which contains information to prevent errors during transmission.

Peer-to-Peer (P2P): The term peer-to-peer is used to indicate a form of computing where two or more than two users can share files or CPU power. They can even transmit real time data such as telephony traffic on their highly ad hoc networks. Interestingly, the peer-to-peer network does not work on the traditional client-server model but on equal peer nodes that work both as “clients” and “servers” to other nodes on the network.

POTS: POTS is the short form of plain old telephone service. It transmits voice as analog data on communication lines that are much slower when compared to todayâ€™s ISDN or FDDI lines. However, not long ago POTS, which is also known as the public switched telephone network, was the standard telephone system across the world.

Processor drain: This is a term used to indicate a drop in the quality of VoIP phone service when a user opens several applications on his computer simultaneously.

Protocol: It is a convention or standard that defines the procedures to be adopted regarding the transmission of data between two computing end points. These procedures include the way the sending device should sign off a message or how the receiving device should indicate the receipt of a message. Similarly, the protocols also lay down guidelines for error checking, data compression, and other relevant operational details.

PSTN: PSTN, which stands for Public Switched Telephone Network, refers to the telephone system that transmits analog voice data. Till recently, PSTN was the heart of all phone systems worldwide. However, most of the developed world is now switching to or has switched to telephone networks that are based on digital technologies, such as ISDN and FDDI. RJ45: RJ45, which is the acronym of Registered Jack-45, is a telephone connector that is used in Ethernet and Token Ring Type 3 devices. It has eight “pins” or electrical connections.

Router: A router is a network device that that handles message transfer between computers that form part of the Internet. The messages, which are in the form of data packets, are forwarded to their respective IP destinations by the router. A router can also be called the junction box that routes data packets between computer networks.

Sampling: This is a methodology used to measure the value of an analog signal at regular intervals, and encoding it into a digital format for VoIP phone services.

Service provider: A service provider is a business entity that provides a communication, storage or processing service for a fee. Some of the service providers in the digital world are the Internet service provider (ISP), application service provider (ASP), storage service provider, mobile phone service provider, web hosting provider, and of course,**VOIP service provider**.

SIP phone: A SIP phone is a telephone that uses the SIP (Session Initiation Protocol) standard to make a voice call over the Internet. The SIP phones come with several value added services like voicemail, e-mail, call number blocking etc. There are no charges for making calls from one SIP phone to another, and negligible charges for routing the call from a SIP phone to a PSTN phone.

SIP: SIP, which is the acronym of Session Initiation Protocol, is an IP telephony signaling protocol. It is primarily used for voice over IP (VoIP) calls, though with some extensions it can also be used for instant messaging. It is less complex than H.323, the other IP telephony protocol.

Skype: **Skype** is a peer-to-peer Internet telephony company that revolutionized the way voice calls are made by using VoIP technology. The company, which has been acquired by eBay, was founded by Niklas ZennstrÃm and Janus Friis. Skype users can speak to other Skype users for free, but have to pay a small fee for calling or receiving calls from conventional phones.

Soft switch: It is a software application that is used to keep track of, monitor or regulate connections at the junction point between circuit and packet networks. This software is loaded in computers and is now replacing hardware switches on most telecom networks.

Softphone: This is a software application that is installed in the userâ€™s PC. It uses the Voice over IP technology to route voice calls over the net and provides several value added features, such as call forwarding, conference calling, and integration with applications such as Outlook for automatic dialing The audio is provided through a microphone and speakers plugged into the sound card. The only limitation of a Softphone is that the phone call has to made through a PC. Many softphone are **free VOIP software** downloads.

Voice chat: This is an application that enables two or more than two individuals to carry on a verbal conversation over the Internet. Voice chat is also known as audio-conferencing or telephone conferencing on the net.

Voice over IP (VOIP): VoIP or **Voice over IP** is the technology that is used to transmit voice over the Internet. The voice is first converted into digital data which is then organized into small packets. These packets are stamped with the destination IP address and routed over the Internet. At the receiving end the digital data is reconverted into voice and fed into the userâ€™s phone.

Voicemail: It is a telephone messaging system that digitizes the analog voice signals and stores them on disk or flash memory in a central computer. These messages can then be retrieved by users by logging on to the server or forwarded to another voice mailbox. Most voice mail systems have auto attendant capabilities, that is they can use prerecorded messages to route callers to the appropriate person or mailbox. **Voicemail** is usually a free feature in VOIP service plans

IM: IM, which stands for Instant Messenging, is software that allows users to exchange messages in real time. However, to do so both the users must be logged on to the instant messaging service at the same time. Some of the popular IM services are: MSN Messenger, AOL Instant Messenger, Yahoo! Messenger, Google Talk and ICQ.

VOIP Gateway: This device provides the conversion interface between the public switched telephone network (PSTN) and an IP network for voice and fax calls. Its primary functions include: voice and fax compression/decompression, packetization, call routing and control signaling. It also provides an interface to Gatekeepers or Softswitches, billing systems, and network management systems.

VOIP PBX: VoIP PBX, which stands for Voice over Internet Protocol Private Branch eXchange, is a telephone switch that converts IP phone calls into traditional circuit-switched TDM connections. It also supports traditional analog and digital telephones.

VOIP Phone: A **VoIP phone** is one that uses the Internet to route voice calls by converting the voice data into IP packets and vice versa. The phones come with built-in IP signaling protocols such as H.323 or SIP that help in the routing of data to the right destination. A VoIP phone can also be a software application that is installed in the user's PC. In this case it is known as the Softphone. Also, the calls in this case have to be made from the PC, and not through a telephone instrument.

VOIP services: The **VoIP services** are packet-based services that use the Internet to move voice data. These services are much cheaper than the traditional PSTN services because the investment in infrastructure is low. They also come with several value added features which make them more lucrative than the conventional landline phone services.

Web phone: A web phone is a device that allows users to make voice calls over the Internet.

WiFi Hotspot: An area where a wireless access point enables users carrying wireless-enabled laptops to log on to the Internet. The limiting condition is that the access point is configured to broadcast its presence and does not require authorization for access. Generally, WiFI hotspots are located in public places like airports, train stations, libraries, marinas, convention centers, coffee shops and hotels.

WiFi phone: A WiFI phone is one that enables users to make phone calls from public WiFi hotspots or residential WiFI network environments. Besides voice calls, these phones can be used to send e-mails wirelessly.