

Voip Glossary

Asterisk

An Open Source "Do it all" PC based software PBX. You can set up your own highly flexible voip or analog PBX using PC hardware for very little cost.

Asynchronous Communication

A data communications method in which bits are sent one after the other with a start and stop bit used for flow control. This as opposed to synchronous communication where blocks of data are transmitted using a synchronizing clock.

Digital Subscriber Line

A high-speed digital switched service using existing copper pairs to connect subscriber CPE (Customer Premises Equipment) to the Central Office. DSL handles more data downstream (data flowing towards the subscriber) than upstream (towards the network).

E-1

The designation for the 2.048Mbps. ITU standard for Europe's 30-channel digital telephone service. It is the European version of T-1 (DS-1). The bandwidth is divided into two signaling channels (channels 15 and 31 starting from 0) and thirty bearers (voice channels). A&B bit signaling (robbed bit signaling) is not used here. E-1 uses one of the control channels for signaling and the other for clock synchronization.

Frame Relay

In data communications, a packet switching method that uses available bandwidth only when it is needed. This fast packet switching method is efficient enough to transmit voice communications with the proper network management.

Full Duplex

In telephony and data communications, the ability for both ends of a communication to simultaneously send and receive information without degrading the quality or intelligibility of the content.

FXS

Foreign Exchange Station. Provides power and ring signal to external locations. For example, your phone company provides you with an FXS interface.

FXO

Foreign Exchange Office. Receives voltage and ring signal from external source such as the phone company. A good example of this would be your standard analog phone.

G.711

This standard codec converts the analog speech signal to a digital stream using pulse code modulation (PCM), a logarithmic encoding algorithm. It delivers toll-quality voice using 64 kbps of bandwidth per call. Best choice in high-bandwidth environment.

G.723.1

Developed for multimedia, it has emerged as the standard low-bandwidth codec today. Using just 5.3 kbps or 6.4 kbps of bandwidth per call, it delivers good voice quality at low bandwidth in the presence of network impairments such as lost packets and transmission errors.

G.729

This standard codec offers toll-quality voice at lower bandwidth. Using 8 kbps of bandwidth, it offers modest processing requirements and tolerates moderate network delay. Delivers high-quality voice in low-bandwidth environments.

H.323

A popular ITU (International Telephone Union) Voip protocol that is widely in use today.

Hard Phone

A real desktop phone, Ranges in quality and price. Better than a soft phone by a long shot.

IAX

Inter Asterisk Exchange. A proprietary Asterisk Protocol that is firewall resistant and easier to use than SIP.

Interactive Voice Response

IVR. In computer telephony, Interactive Voice Response is a horizontal application wherein computer-based information is accessed over the phone - with a telephone versus a computer. An IVR platform uses computer telephony components to translate callers' touch-tones or voice commands into computer queries after the callers hear an audio menu. For exVoxisle: "Please enter your account number using the touch-tones on your telephone." These queries are then "fetched" by the IVR platform from the host computer. In some cases, the information resides in the same platform (self-hosted). The information is then converted into voice commands and then spoken over the phone to the caller. These spoken prompts can be pre-recorded, digitized speech messages that are then concatenated to form whole sentences. For exVoxisle: "Your bank balance is five hundred and sixty-three dollars". The responses to the caller also take the form of text-to-speech prompts. IVR systems can also be used for callers to change the information in a database instead of just "listen" to the information.

MGCP

Media Gateway Control Protocol. Signaling Protocol for Voip.

Packet

A logically grouped unit of data. Packets contain a payload (the information to be transmitted), originator, destination and synchronizing information. The idea with packets is to transmit them over a network so each individual packet can be sent along the most optimal route to its. Packets are assembled on one end of the communication and re-assembled on the receiving end based on the header addressing information at the front of each packet. Routers in the network will store and forward packets based on network delays, errors and re-transmittal requests from the receiving end.

Packet Switching

A means of economically sending and receiving data over alternate, multiple network channels. The premise for packet switching is the packet, a small bundle of information containing the payload and routing information. Packet switching takes data, breaks it down into packets, transmits the packets and does the reverse on the other end. Packets can be sent in order and then be received in a different order - only to be put back in the correct order in seconds. There are slow packet switching networks, like the old SNA networks - and there are fast packet networks based on Frame Relay and ATM. Although traditionally used for data, packet networks, especially well-managed ones, are becoming suitable for real-time transmission of voice and video.

POP

Point of Presence, equivalent of a local phone company's central office. The place your long distance carrier terminates your long distance lines just before those lines are connected to your local phone company's lines, or to your own direct hookup.

PSTN

Public Switched Telephone Network, Copper lines. See also: POTS.

POTS

Plain Old Telephone Service. The same service that your parents grew up with and with which we are all familiar.

Real Time

A communication wherein any perceptible delay between the sender and receiver are minimal and tolerated. Regular telephone calls are real time. Point-to-point fax transmissions are "close" to real time. Voice messaging is in non-real time.

RJ-11

The designation for connecting a tip and ring circuit to a standard, modular, six-position jack. The green and red wires go in the middle (only) pair, and the outside positions of the connector are unused.

RJ-45

Eight-position modular connector used for data transmission over standard twisted or flat pairs.

Skinny

Also called SCCP. Proprietary Cisco protocol used in Cisco VoIP products.

Service Provider

An addressable entity providing application and administrative support to the client environment by responding to client requests and maintaining the operational integrity of the server.

Signaling System #7

Or SS7. The basis for modern methods to route traffic with out-of-brand signaling. Its forerunner, CCIS (Common Channel Interoffice Signaling), used 4.8 Kbps data links to transmit call set up and tear down messages to switching office adjunct computers and packet switches. SS7 in itself is not a network service offering, but rather the underlying infrastructure with which many existing and proposed offerings are based. For example, local Basic Rate ISDN (BRI) services can tap into SS7, so 64 Kbps packetized data can be routed with the help of the network's out-of-band signaling capability. In addition, nationwide Primary Rate ISDN (PRI) services can use the same backbone.

SIP

Session Initiation Protocol. An IETF (Internet Engineering Task Force) protocol for voice, video and instant messaging. Currently the most useful and popular protocol for VoIP.

Soft Phone

A VoIP Phone completely implemented in software designed to be used from a computer with a headset or a microphone and speaker. Often because of ambient noise or poor quality microphones and sound cards yields poor results.

Soft Switch

is a call control server in a VoIP network providing user Authentication and Accounting services, independent of the VoIP protocol (H.323, MGCP, SIP).

T-1

North American digital standard for high-capacity transmission of telephony and data communications. In telephone T-1 provides a 1.544 Mbps link which is broken down in to 24 discrete, 64 Kpbs voice-grade channels. In data communications, T-1 links are used to directly connect CPE (Customer Premises Equipment) routers to the Internet and for Private Data Network or VPN circuits.

T-3

North American standard for DS-3. Operates at a signaling rate of 44.736 Mbps, or the equivalent of 28 T-1s.

T.38

is a protocol for a real-time transmission of FAX in VoIP network.

TCP

Transmission Control Protocol. The transport layer protocol developed for the ARPAnet which comprises layers 4 and 5 of the OSI model. TCP controls sequential data exchange in TCP/IP for remotely hosts in a peer-to-peer network.

Voice Messaging

An application of store and forward wherein telephone access to private messages are retrieved by users for playback. Imagine a shared tenant answering machine that handles multiple telephone lines and can record incoming messages for hundreds of people simultaneously. Imagine the intended parties being able to retrieve these messages over the phone with simple touch-tone commands. Imagine full security, so no one can pick up anyone else's messages without a special, private access code. That's voice messaging. Voice messaging systems take many forms. There are CPE (Customer Premises Equipment) versions and Service Bureau or Telco versions. The basic idea is the non real-time sending and receiving of private messages. Some systems support the broadcast of messages to multiple recipients. Some provide message waiting notification via pager, message waiting light or "outdial" telephone calls.