VoIP Frequently Asked Question (FAQ) sheet

What is VoIP?
Voice over IP or VoIP is the transport of digitized speech traffic over an IP based network instead of over a traditional telephone network. The speech may be part of a real-time conversation or a non-real-time transaction such as voice mail. Speech is digitized by any of a large number of standard or proprietary voice coding schemes; however, compatible coding is required at both ends of the connection. The IP network may be the public Internet or a private IP-based network. Voice transport service could be phone-to-phone, computer-to-phone or computer-to-computer. Phone connections require an interface to the Public Switched Telephone Network (PSTN) via a gateway. The presence of a Gatekeeper is optional.

What voice compression or coding does VoIP use?
There are three main voice compression schemes used in VoIP networks. G.711 is the same compression used on traditional wired telephone networks and used 64 kbps coding. G.729 compresses voice to an 8 kbps rate but still delivers near toll quality voice over most networks. G.723 compresses voice at either 6.3 kbps or 5.3 kbps, but has the lowest quality of the three compression algorithms mentioned here.

What signaling protocol (for setting up calls) is used?
VoIP networks typically use one of two signaling methods for setting up and tearing down calls. The most widely deployed is referred to as H.323, which is an international standard designed to deliver multimedia services over wired networks. Multimedia services include voice, video, and data. H.323 is related with and depends on other protocols to deliver these services. A second protocol is SIP, which is an Internet Protocol compliant signaling standard set by the IETF.

What is a VoIP Gateway?
In H.323, an endpoint on the network that supports real-time communication between other endpoints or terminals that have dissimilar capabilities. This includes supporting voice communication between terminals on a packet, e.g., Internet Protocol (IP) network and terminals on a circuit (e.g., Public Switched Telephone Network (PSTN) network. Gateways translate protocols between telephony and packet networks to deliver voice and similar services.

What is a VoIP Gatekeeper?
In H.323, provides address translation, bandwidth control, and access control to a network of VoIP terminals and gateways. The network of all elements (gateways, gatekeepers, VoIP terminals) under control of a gatekeeper is defined as an H.323 Zone.

What are the critical factors in delivering good quality voice services on IP networks?
There are many factors that contribute to voice quality. The three main factors are latency, jitter, and voice coding or compression.

What is Latency?
Latency refers to the delays encountered in delivering voice packets from the originating to terminating end of a voice call. Delays are contributed by the voice coding algorithms, packetization of voice packets, equipment used in the delivery of these packets over an Internet Protocol network, and ability to prioritize voice packets through such networks.

What is Jitter?
Jitter is the variations in the delays delivering voice packets from originating to terminating ends.

What equipment is needed in a VoIP system?
Generally, a VoIP system needs a network gateway, premises gateway, and gatekeeper. The size of business served will dictate the type of premises gateway used. For example, small or home-based businesses could be well served by a premises gateway that provides a couple of analog telephone ports and a LAN port to provide simultaneous voice and data services. A medium sized business could be well served by a premises gateway that provides up to eight analog telephony ports. These ports could be used for traditional analog telephones or a small PBX. Larger enterprises would be better served by a premises gateway that provides T1 / E1 services.

What types of network topologies can be supported?
There are two general topologies, point- to-point and point to multipoint. Enterprise customers are likely to employ point-to-point topologies while service providers are likely to employ point to multipoint. Wi-LAN Inc. can support both.

Is billing supported on VoIP systems?
RADIUS/Billing in VoIP systems varies widely. The most common system for collecting billing information in VoIP networks is a RADIUS server. Originally developed to manage access in dial-up networks they have evolved to collect call information, which is uploaded to a billing system.

What is RADIUS?
The RADIUS server allows centralized administrative control over remote access to a network service. For example, a RADIUS server will typically hold the database of user logins and passwords, and will be queried when a user dials into an Internet Service Provider or a corporate network. Some RADIUS servers collect usage information, which can be used for billing purposes.

Can regular phones be used?

Most premises gateways support the use of analog telephones and Fax machines. Depending on the VoIP network equipment, vertical features can be provided; conferencing, call forwarding, call waiting, calling number ID, among others.

Can VoIP phones be used?

VoIP phones, although significantly more expensive, are supported on VoIP networks using WiLAN radios.

Can I send a Fax over a VoIP system?

Wi-LAN VoIP networks support Group 3 Fax over Internet Protocol services. To do this, both the premises and network Gateways need to support T.38 protocols. T.38 defines an Internet facsimile protocol consisting of messages and data exchanged between Facsimile gateways connected via an IP network. Facsimile gateways must exist at both ends of the IP network.

What is Packet Saver Technology?

PacketSaver™ Technology saves bandwidth, and bandwidth can be an expensive. By configuring the Gateways with PacketSaver, many (up to 30) IP phone calls share the same IP Packet Header (which is the largest chunk of a VoIP call). By combining PacketSaver and other integrated technologies, you can reduce the amount of IP Bandwidth used by over 50%! At 100% capacity on an E1 trunk, that means using 192Kbps instead of 397Kbps (Kilobits per second).

Are separate Gateway and Gatekeeper needed for Wi-LAN's VoIP networks?

Most other VoIP Gateways are only equipped to turn voice signal into IP Packets and start the call setup process. Wi-LAN's VoIP solutions don't stop there. Integrated within every Gateway is the ability to run Gateway, Gatekeeper and Border Element processes. The Gatekeeper's function is to match together Phone Numbers on your network with the coinciding IP addresses of where each Phone Number terminates. The Border Element's function is to communicate with each Gatekeeper and to furnish updates of the matching Phone Numbers & IP Addresses to all of other Gatekeepers dynamically as changes are made.

What are the differences between FXO and FXS ports?

A FXS port is a transmission equipment interface that emulates the line-side interface of a switching system such as a PSTN Central Office and can be connected directly to a telephone set. A FXO port is a transmission equipment interface that emulates subscriber equipment such as a telephone or a key system and passes standard signaling indications to a remote location. Premises Gateways provide FXS ports to which the end user connects an analog telephone or key telephone system.