

Voice over IP (VoIP) Basics for IT Technicians

VoIP brings a new environment to the network technician that requires expanded knowledge and tools to deploy and troubleshoot IP phones. This paper provides an introduction to VoIP technology and operation.

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Voice over IP Basics for IT Technicians

Executive summary

The IP phone is coming – or has arrived – on desk near you. The IP phone is not a PC, but does have a number of hardware and software elements also resident in PCs. Voice over IP (VoIP) brings a new environment to the network technician that requires expanded knowledge and tools to deploy and troubleshoot IP phones. The LAN diagnostic tools have to analyze Ethernet, but must now support VoIP signaling protocols and voice transmission. Understanding VoIP will become mandatory for the network technician. This paper provides an introduction to VoIP technology and operation.

What is VoIP?

Voice over Internet Protocol (VoIP), also called IP Telephony, is rapidly becoming a familiar term and technology that is invading enterprise, education and government organizations. VoIP is designed to replace the legacy TDM technologies and networks with an IP-based data network. Digitized voice will be carried in IP data packets over a LAN and/or WAN network. Installing and testing the VoIP network of IP phones, gateways and servers requires new tools and expanded knowledge. This paper discusses VoIP operation and testing.

The legacy telephone network has provided reliable and high-quality voice communications for many years. It delivers voice and speech over a standardized 64 Kbps channel. The 64 Kbps bandwidth is guaranteed for each call and the speech path carries the voice as a continuous digital stream. Digital voice is NOT carried in packets. Enterprise and residential callers use DTMF (Touch Tone), TI channel and ISDN D channel signaling to set up and manage the call. Within ISDN, signaling is carried in packets on a separate signaling channel over the Basic Rate Interface (BRI) and Primary Rate Interface (PRI) connections to the carrier. The carrier then translates the signaling (all forms) in an internal signaling protocol called Signaling System #7. Signaling protocols are primarily active at the beginning and end of a call.

In a VoIP network, there is a signaling protocol and a speech transmission protocol. Both protocols require all information be carried in IP packets. Several standards-based choices are available for signaling protocols, including H.323, SIP, MGCP and H.248 (MEGACO). Most IP PBX vendors have developed their own proprietary signaling protocol, the most common of which is Cisco's SCCP (Skinny) protocol. RTP is the standard speech transmission protocol used with VoIP networks. The speech is digitized, placed in packets, and transmitted through the IP network. Multiple packets are required to carry a single spoken word. The voice is digitized using one of the G.7xx standards. Each of these standards will be discussed in detail later in this paper.

Why, then, do organizations move from a TDM-based telephone to an IP-based telephone? This is a major change in the underlying technology, yet we are still making phone calls as we have for years. One or more of the following may justify the move to VoIP:

- Reducing long distance charges, especially international long distance
- Reducing staff by combining voice-network and data-network management and eliminating redundant functions
- Adding expanded applications that are not offered by TDM-based systems
- Having one common network for different forms of communication
- TDM vendors are not offering new systems, thus forcing customers to eventually adopt IP-based telephone systems

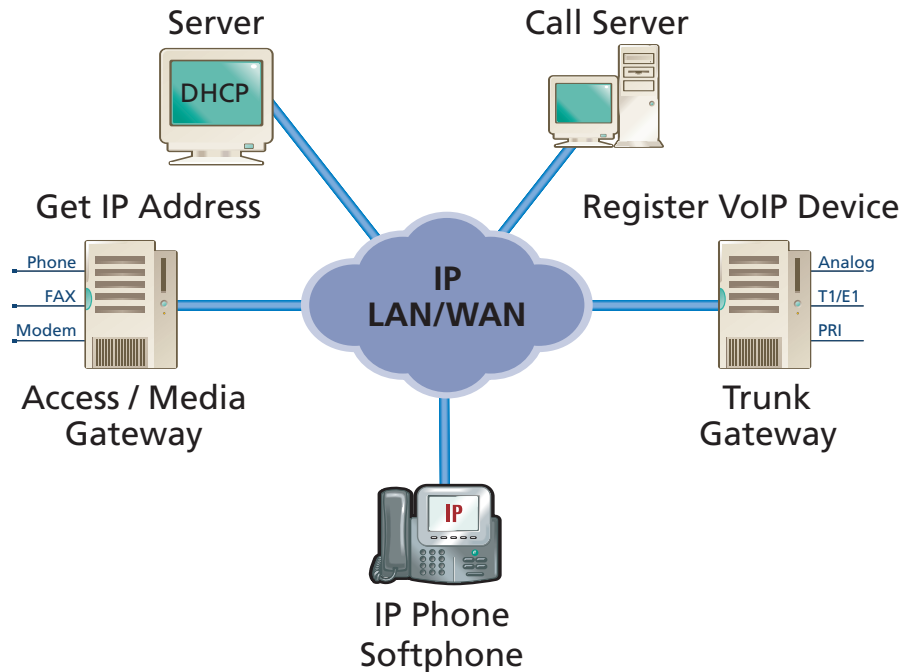


Figure 1
Typical VoIP configuration

There are two forms of a VoIP call. If you have Microsoft's NetMeeting, you can set up a PC-to-PC call without working with a call server. This is typically how the early users of VoIP made calls. However, the prevalent enterprise VoIP solution requires a call server (the standards community calls this a "gatekeeper") to be part of the network configuration. Although it is called a server, the server does not operate like a traditional server. An e-mail server and a PC are in constant contact for the e-mail operations. In VoIP, the call server (see Figure 1) controls all the services offered, provides control over the call, supports the telephone features, authenticates and authorizes the caller and implements security. The call server is NOT the telephone switch. Once the call server sets up a phone (peer-to-peer) call, the server becomes dormant during the speech transmission unless the phones contact the server to indicate a change in status or the call server wants to change the call configuration, such as indicating there is a call waiting. The server is there to process the signaling, but does not switch the speech. The speech packets are passed directly from phone to phone.

There are two major categories of IP phone implementations: hard phone and soft phone. The hard phone contains all the hardware and software to implement VoIP. It is not a PC, but is specifically designed as a phone. Hard phones can be simple in their functions, but can also have color displays with touch sensitive screens and may even support web browsing. There is no typical hard phone on the market. The second category, the softphone, is a headset connected to a PC with all the telephone features implemented by the sound card and software resident in the PC.

Another piece of hardware, the gateway, is usually part of the VoIP network. Most organizations will have legacy phones, fax machines, modems, connections to the PSTN, and other devices that originally connected to the organization's telephone switch, called a PBX. When migrating to VoIP, these devices and interfaces will have to be connected to a conversion system that supports the legacy devices and interfaces on one side and connects to the IP network on the other. The legacy devices will be connected to an access/gateway and the PSTN interface connection will be terminated on a trunk gateway.

Standards for VoIP

“Standards are great, I have so many to choose from” is a quote that amply describes VoIP. There are multiple signaling standards.

- H.323, the ITU standard that was published in 1995, started the development of VoIP products and services. There are four versions available. V.1 is obsolete and has been discontinued in virtually all products. Versions 2, 3 and 4 are used in today’s products. These three versions are similar in design and are upwardly compatible. This is the dominant installed signaling protocol for use with hard and softphones.
- The Session Initiation Protocol (SIP) was produced by the IETF as an IP standard. Although SIP is gaining considerable attention, it will not become the dominant installed protocol for a few years. The attractions of SIP are better interoperability among vendors, easier application development, operation that is close to other existing IP protocols, and easier operation through firewalls. It is usually part of hard and softphones, but it may also be used with gateways. SIP is a completely different design when compared to H.323.
- MGCP is a protocol used primarily with gateways, although an occasional hard phone may support MGCP.
- MEGACO/H.248, another standard protocol, is a combined effort of the ITU and IETF. It can be used with gateways and server-to-server communications. It is not found in hard or softphones.

In addition to the standards, nearly every IP PBX vendor has produced a proprietary signaling protocol. The most commonly found protocol is Cisco’s SCCP, or “Skinny,” protocol. These proprietary protocols may be variations of the standards or may be uniquely designed. They each provide the call control found in the standard protocols. An IP PBX vendor usually supports one or more of the standard signaling protocols plus their proprietary protocol. All the signaling protocols follow the same path for control as shown in Figure 2. The H.323, and most proprietary signaling, is carried over TCP, while SIP operates over UDP.

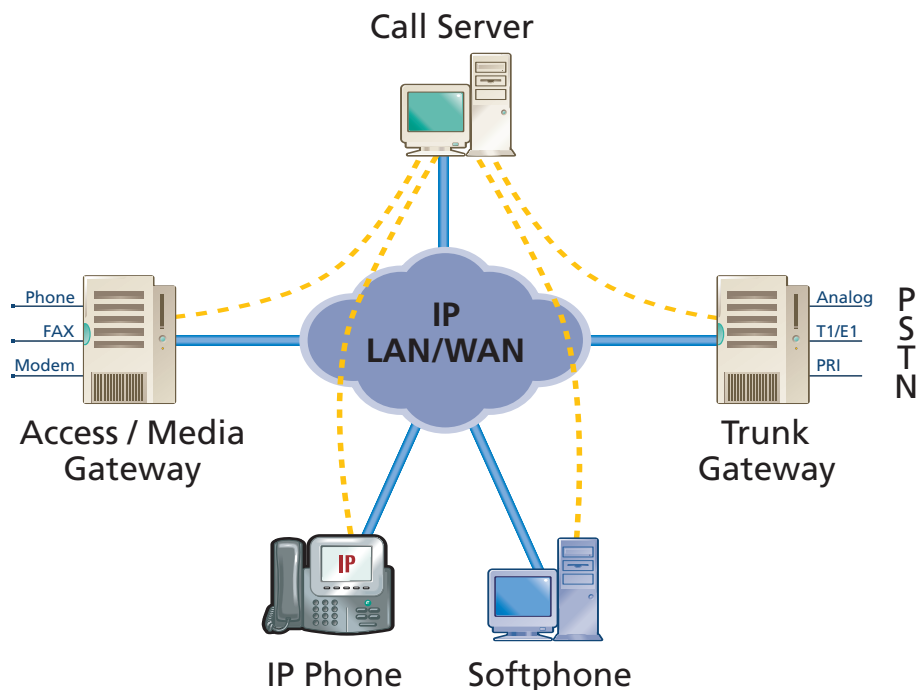


Figure 2
H.323, SIP and proprietary signaling

Speech is carried in packets that use the Real Time Protocol (RTP) standard. Each RTP packet contains a piece of a digitized word. Multiple RTP packets, when combined at the receiving IP phone, produce a spoken word. The RTP IP PBX vendors commonly implement RTP. Proprietary protocols that operate like RTP are uncommon. The speech paths connect directly between phones and gateways; speech does not pass through the server, nor is speech carried by the signaling protocols as shown in Figure 3.

There are several voice digitization standards and some proprietary techniques in use. Most vendors support one or more of the following ITU standards and avoid proprietary solutions:

- G.711 is the default standard for all vendors, as well as for the PSTN. This standard digitizes voice into 64 Kbps and does not compress the voice
- G.729 is supported by many vendors for compressed voice operating at 8 Kbps. With quality just below that of G.711, it is the second most commonly implemented standard.
- G.723.1 was once the recommended compression standard. It operates at 6.3 and 5.3 Kbps. Although this standard reduces bandwidth consumption, voice is noticeably poorer than with G.729 and is not very popular for VoIP.
- G.722 operates at 64 Kbps but offers high-fidelity speech. Whereas, the three previously described standards deliver an analog sound range of 3.4 kHz, G.722 delivers 7kHz. This version of digitized speech will become common in the future

In all cases, the IP phones and gateways collect about 10 to 30ms of digital speech and place it in the RTP packet for transmission.

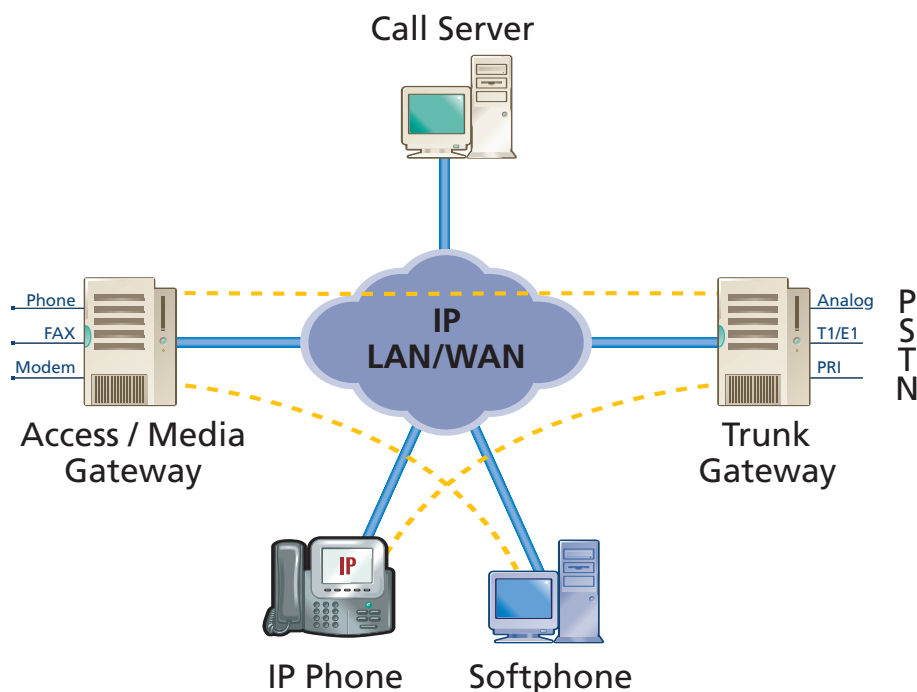


Figure 3
RTP speech/talk paths

VoIP networks

Local (LAN) and wide area networks (WAN) can both support VoIP operation. However, there are significant differences between LAN and WAN performance that affect signaling execution speed and voice quality. The LAN implementation uses Ethernet as the transport network. VoIP devices operate following the IEEE 802.3 standards. No proprietary changes have been made to the Ethernet protocols and architecture. The Ethernet LAN operates at 10 and 100 Mbps on networks with very short delay, no jitter (delay variance), few

errors and no packet loss. Although VoIP traffic could share the LAN with data users, it is recommended that the voice and data devices run on separate VLANs (IEEE 802.1q) on the LAN switches for both performance and security reasons. Voice quality and signaling execution speeds are as good as the TDM PBXs.

The WAN presents a number of performance-hindering issues. Bandwidth is limited, end-to-end delays are much longer, packet jitter occurs and, finally, packets are lost. The transmission error rate is low enough that signaling packets do not often require retransmission, and voice quality is not impaired by errors. The end-to-end delay goal between IP phones is 150ms. The receiving IP phone must compensate for jitter by waiting for all the packets of a word to arrive before the word can be converted back into analog sound. The receiving IP phone must also insert simulated voice packets to eliminate holes in a word, which occurs when packets are lost. Correcting jitter and packet loss causes extra delay between IP phones.

Extra bandwidth and Quality of Service (QoS) techniques can solve these problems. Voice calls consume bandwidth: about 80 Kbps when no voice compression is used (G.711) and about 25 Kbps (G.729) when the voice is compressed. The actual bandwidth consumption will vary based on compression type and packet size. Bandwidth is probably not an issue on the LAN as most LANs operate at low utilization, probably less than 10% to 20% utilization. If bandwidth is not increased on the WAN, then voice and data users will suffer call degradation.

QoS can be supported on LAN switches using the IEEE 802.1p standard. This standard requires that IP phones and gateways also support the 802.1p standard. Routers produce QoS through the implementation of DiffServ, which must be supported by the IP phones and gateways, as well as MPLS in the routers. RSVP was an early technique for VoIP QoS, but is less frequently used in today's products.

How VoIP works

A VoIP-based PBX starts up (boots up) like other servers. Once the booting up is complete, the IP phones and gateways can register with the call server (see Figure 1). The IP phone and/or gateway must first access a DHCP to obtain an IP address. The DHCP may be part of the data network, or it may be a separate server, or it can be integrated with the call server. Once an address has been assigned, the IP device contacts the call server to register. The call server may have a common set of privileges and restrictions for IP devices or an administrator can make the feature assignments.

The call server or another assigned server also adds this device and its phone number(s) to the DNS to support directory services. A permanent H.323 TCP session is established between the VoIP device and the call server. This is true for most proprietary signaling protocols. SIP uses UDP for the signaling path.

When a user picks up the phone, the dial tone can be generated locally by the phone or by the call server. The IP device then sends one or more packets requesting a connection and the features to be implemented during the connection, such as a conference call. The call server then determines whether the other device is available or busy. If available, the call server contacts the receiving device and instructs both the caller and called devices to establish a peer-to-peer UDP path to carry the RTP speech. The call server becomes dormant during the call until one of the devices terminates the call. The call server then breaks the peer-to-peer connection and records the call event as part of the Call Detail Record (CDR).

Security of VoIP has become a major concern for VoIP adopters. Most firewalls do not support VoIP except through VPN connections. As a default, data firewalls will probably prevent the operation of VoIP by users on the untrusted network when they call devices on the trusted network. Several vendors are now offering encrypted signaling and encrypted speech behind the firewall. This design prevents security problems produced by users of the trusted network. The encryption functions must be implemented in the IP phones, gateways and call servers.

Conclusions

VoIP forces an expanded role for the network technician. Not only will the physical and LAN deployment and troubleshooting tasks continue to exist, but there will be new responsibilities. These will include VoIP protocol operation, call server interaction, network performance measurement and IP phone configuration. Compounding these responsibilities are the vendors who have chosen to implement multiple standards as well introduce proprietary solutions to the VoIP mix. The job description for the future network technician will have to expand to meet these new challenges.

About the author

Gary Audin has 40 years of computer, communications and security experience. He has been involved with VoIP since 1998. Mr. Audin has been published extensively and many of his VoIP related articles can be found at www.webtorials.com. He has made more than 200 presentations at trade shows and user conferences. He is the exclusive instructor for the BCR VoIP seminar series, presenting 180 VoIP seminars.

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