Next-Generation IP Phones Arriving

Allan Sulkin

The new desksets remain expensive, but over the next few years, deployment should increase.

The market for IP-PBXs is still defining itself, and shipments of IP telephones are a very small percentage of total PBX station shipments. That said, however, IP telephone technology and offerings are evolving at a fast pace—there have been three generations of IP phones since IP PBXs hit the market about three years ago.

The first generation closely resembled traditional digital telephones in appearance and basic functions. The second generation delivered a new set of features and function attributes, such as pixel displays and integrated Ethernet switch ports.

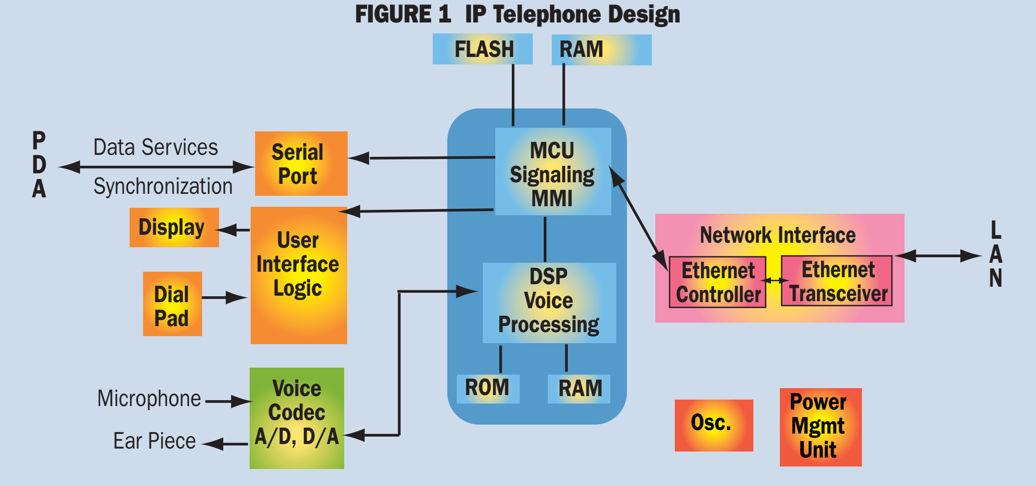
As the third generation of IP phones begins to make its appearance, it's clear that the familiar desktop telephone is evolving from a voice-only instrument to a mixed-media IP appliance. During the next year, most IP-PBX suppliers will offer models that support features and capabilities such as Web browsers, touch screens, 1/4 VGA monitor color display fields, WAP and infra-red (I/R) interfaces and USB interface ports.

Design Basics

All the system manufacturers base their IP telephone on proprietary design schematics and circuitry, but there are several common design elements (Figure 1):

- **User interface:** Provides the classic telephone functions—keypad for dialing numbers, keys for line and feature access, a display for user prompts, caller feedback, messages and other call processing information. In a growing number of products there is also a serial interface to external devices, such as a PDA, to allow synchronization of telephone information, speed dialing, custom programming, etc. An audible indicator, e.g., ringer, is also included to announce incoming calls.
- **Voice interface:** Converts analog voice signals into 8-bit digital bit samples. These signals are sampled at an 8-kHz rate, to create a 64 kbps digital bit stream to the processor, using a standard PCM codec.
- **Network interface:** Allows transmission and reception of voice packets to/from the telephone terminal, based on 10BaseT or 10/100BaseT Ethernet. Some IP telephones are equipped with multiple RJ-45 Ethernet connector ports, and an integrated Ethernet hub/switch to support connections to both the customer premises LAN and desktop PC clients. Newer generation IP telephones may also be designed with a USB connector port.
- **Processor complex and associated logic:** The processor complex performs voice processing, call processing, protocol processing and network management software functions. The processor complex consists of a digital signal processor (DSP) for voice-related functions and a micro

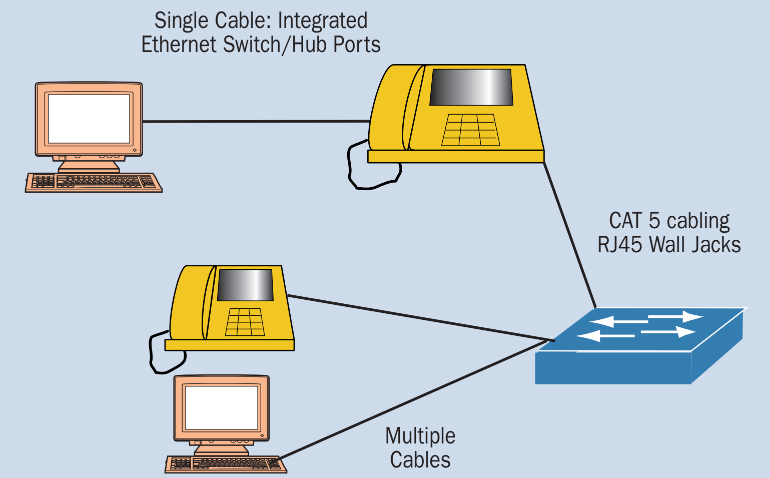
FIGURE 1 IP Telephone Design



Source: Telogy Networks

Allan Sulkin is president, TEQConsult Group. He is a member of the BCR *Board of Contributors, developed and is the instructor for* BCR*'s seminar, "Understanding PBX Systems and Technology," and is a featured speaker at* BCR*'s annual Voice-Con conference (www.voicecon.com). He can be reached at amsulkin@aol.com.*

FIGURE 2 IP Phone LAN Wiring Options



controller unit (MCU) for the remaining control and management functions. The DSP and MCU each have associated memory; DSP memory usually includes both RAM and ROM elements, and MCU memory usually includes both RAM and Flash elements. The Flash memory element is used to support software upgrades.

Basic IP phone software modules include a variety of user interface drivers (display, keypad, ringer, user procedures), voice processing modules, telephony signaling gateway modules, network management modules and system service modules. The voice processing software modules include:

- A PCM interface unit.
- A tone generator—Call progress tones, in-band DTMF signaling digits.
- A line echo canceller unit—ITU G.168-compliant echo cancellation on sampled, full-duplex voice port signals.
- An acoustic echo canceller for terminals equipped with a speakerphone.
- Voice activity detector (VAD).
- Voice codec unit—compression and packetization of the 64-kbps digital stream received from the station user based on a variety of algorithms (e.g., G.711, G.723.1, G.729a, etc.).
- Packet playout unit—Compensation for network delay, jitter and packet loss.
- Packet protocol encapsulation unit, based on Real-time Transport Protocol (RTP), which runs directly on top of User Datagram Protocol (UDP).
- Voice encryption.
- A control unit, which coordinates the exchange of monitor and control information between the voice processing module and the telephony signaling and network management modules.

A telephony signaling gateway subsystem performs basic call set-up and tear-down procedures. The software modules used by this subsystem include call processing, address translation and parsing and network signaling.

The most widely implemented network signaling standard is H.323, and the main H.323 standards used in an IP telephone include H.225—Call Signaling Protocol (based on Q.931); H.245—Control Protocol; RAS—Registration, Admission and Status Protocol; and RTCP—Real-time Transport Control Protocol. But H.323 is being supplanted, slowly but surely, by the Session Initiation Protocol (SIP), which is being implemented for network-hosted services such as IP-Centrex. H.323 was defined by the ITU, SIP by the IETF.

Distinct Features/Functions

Some features and functions in IP telephony systems that affect end stations include:

- **Integrated Port Interfaces:** Several models of IP phones integrate a multi-port Ethernet hub/switch into the phone, allowing devices to share a single connector port to the Ethernet switched network, which also reduces requirements for outlets and inside wiring. In contrast, most early-generation IP telephone models are equipped with two Ethernet port connectors—one for the Ethernet network and another for a desktop PC client (Figure 2).

Cisco was the first supplier to incorporate an integrated Ethernet switch into its 7900 series IP telephones. Mitel followed Cisco's approach by including integrated Ethernet switch ports in its second generation of IP telephones, and promises that its next-generation models will have a external connector port to support two Ethernet devices. Avaya, still marketing its first-generation IP phones, also offers an integrated Ethernet hub.

The integrated switch or hub plays an important role in voice quality, and must be programmed to prioritize voice over data and other non-real time communications. Voice quality of service (QOS) at the desktop can be supported by 802.1 p/Q and COS programming (by switch or hub port). For example, each of the internal Ethernet ports on a Cisco 7900 IP phone can be programmed for different classes of service, although these default levels can be overridden by the system administrator.

Some IP telephones, e.g., the Siemens Optipoint 600, also support peripheral data devices—printers, scanners or digital cameras—through a USB port or infrared interface to a PDA. IP phones from Avaya and Mitel can be equipped with an infrared interface, and Don Smith, Mitel's CEO, announced at the recent Society of Telecommunications Consultants conference that his firm would introduce a new model with a PDA docking station interface. The PDA would function as the instrument's display field, as well as providing data download capabilities for call processing and handling applications.

- **Ethernet Power Distribution:** Traditional PBXs use internal power supplies to distribute power over inside telephony wiring to analog and digital phones. Traditional circuit-switched PBXs that have been IP-enabled cannot distribute power across integrated IP gateway circuit cards to the LAN, and neither can LAN-connected telephony servers used in client/server IP-PBXs.

The first-generation IP phones delivered power via an AC/DC transformer connected to a local AC power outlet. Each IP telephone required its own transformer and a dedicated UPS for emergency power support.

Approaches for in-line power are now available. With in-line power, an Ethernet switch is equipped with either an integrated or external power patch module, and power is transmitted over unused Ethernet wire pairs to only those Ethernet ports identifying themselves to the switch as IP telephone devices. IP telephones identify themselves to the LAN switch during an automatic self-discovery installation method, or through manual programming by the system administrator. The Ethernet switch queries the IP telephone as to how much power is required, or a default power level is assumed.

The current price for in-line power options ranges between \$50 and \$100 per Ethernet port, but is expected to decline. Although an IEEE subcommittee has been working on a standard—802.3af—for in-line power over an Ethernet LAN, proprietary solutions are available from Cisco (Mitel, NEC and Siemens also support the Cisco solution), 3Com and Alcatel. There are also third-party power suppliers, like PowerDsine, which Avaya uses for its solution.

■ **Compressed Voice:** The most common digital encoding schemes for sending voice over Ethernet and IP WANs are G.711 (64 kbps), G.723.1 (5.3–6.3 kbps), and G.729a (8 kbps). The compression function can be implemented by either the phone or a standalone gateway. In either case, you need to account for IP overhead (destination address and signaling), which adds about 16 kbps to the required bandwidth. That additional overhead can affect voice quality, but the trade-off is the potential for more efficient utilization of WAN circuits.

Voice Activity Detection (VAD) and silence suppression also conserve bandwidth. No matter how talkative you think a particular caller is, as much as 50 percent of total call time actually is silence. With circuit-switched technology, during those silent periods 8-bit "words" are still being transmitted, thus consuming bandwidth. By contrast, during silent periods on IP phone systems equipped with VAD and silence suppression, no packets are sent and thus the bandwidth is available for other applications. When voice activity resumes, a signaling packet is forwarded to inform the destination IP address that incoming voice packets are on their way.

■ **Web Browser:** As of early November, Cisco, Avaya and Mitel had announced and shipped IP phones with integrated Web browsers. Of course, it only makes sense to ask why someone with a PC on their desk needs another Web browser.

The manufacturers' response has been that their products are not intended as a replacement for a full-function PC client. Instead, they maintain it's a supplemental device for access to information when data processing is not required, and they're positioning the browser-equipped IP phones as a "portal," that combines telephony with access to information servers distributed on a network. Among the applications foreseen for this capability are:

- Access to directories external to an IP-PBX's directory database.
- Messaging—voice, text, fax.
- Web page information screens.
- Personal calendar.
- Conference planning.
- Transportation schedules and reservations.
- Financial data—real-time stock quotes, investor information, etc.

Using a telephone for email or calendar access may seem strange if a PC is only inches away, and whether this takes off remains to be seen. But it may be useful in selected vertical markets where there are stations that don't have a desktop computer—for example, in health care, retail and hospitality settings. Many nursing stations, for

Browser-equipped IP phones have yet to prove their value to the market

Mitel Networks IP Appliance

Avaya 4630 IP Screen-phone

IP phones may increase productivity, reduce costs and enable new functions

example, only have dumb CRT terminals for information access; most retail Point-of-Sale (POS) terminals don't have Web server access. Guest rooms at hotels offer telephones and, increasingly, Ethernet ports, but no computers. Accordingly, Avaya will position its browser-equipped 4630 IP phone as a "low cost" information kiosk for common areas—e.g., corporate offices, retail malls and transportation terminals (airports, bus stations).

An IP telephone that incorporates an integrated operating system, memory storage and Web browser also eliminates the need for a standalone computer for CTI applications, such as call screening, screen pop, directory dialing, and forms/menus display. An IP telephone can connect directly to a Web server loaded with software that provides CTI-like features and functions. A Cisco "bake-off" for third-party software application developers featured applications for contact center operations, personal productivity and systems management, all of which previously required a separate PC client.

Issues: The Good, The Bad And The Ugly

The good news about IP phones involves their potential to increase productivity, reduce installation and maintenance costs and create new features, functions and applications. An IP telephone can be installed and implemented anywhere a station user has LAN/WAN access to their own premises communications; users can leverage existing investments in LAN wiring and hardware to support the new telephones.

One valuable application involves remote stations for fixed desktop or road-warrior users, because IP telephony is usually less costly than alternatives that require expensive signaling conversion equipment between the PBX and the PSTN to support proprietary digital telephones.

There are, however, downsides to today's IP telephones. First and foremost, they're expensive—generally 25–50 percent more than a digital telephone with a comparable number of programmable line/feature buttons. The Cisco 7960 is priced at $695, but is equipped with only six programmable buttons for line/feature access; its new 14-button expansion module adds $425 to the price.

Moreover, many IP telephones require expensive options for power and protocol software licenses. The software license is a "right to use" (RTU) fee, that the IP-PBX supplier passes back to the original software programmer of the H.323 protocol stack. But in the meantime, the customer pays—for example, the list price of the Avaya 4630 touch-screen phone with Web browser is $995, excluding RTU fee and in-line power option. Even with a price discount, the cost is more than double the median cost of an installed digital telephone.

The H.323 and SIP situation discussed above also is cause for concern. Customers who place their bets on H.323 today may become "orphaned" in the future if SIP becomes dominant. Some suppliers claim that they can just do a firmware download to upgrade their instrument from H.323 to SIP, but there's no information available about how much that will cost.

Voice codecs in IP telephones also continue to evolve. Only two years ago, the ITU's recommended voice codec was G.723.1, but that algorithm's inherent 30-millisecond delay factor makes it a poor choice if voice-grade QOS levels are to be maintained. So, today, G.729a is the voice compression standard *du jour*. Who knows what tomorrow's will be?

Power support and management also need to be carefully considered. Customers who purchased first-generation IP telephones from Cisco and Siemens discovered that their phones did not support any in-line power option. Customers who buy an IP telephone that does not currently support the IEEE 802.3af power standard may find themselves locked into a proprietary solution: Change out the LAN switch, and you may need to change the power module option.

Conclusion

During 2000, an estimated 250,000 IP stations were shipped in the U.S., and that figure is likely to increase by at least 50 percent during 2001. While most of the installations were in small—less than 100-station—environments, the impact of IP phones is starting to be felt, particularly in the market for key/hybrid systems, which declined by about 15 percent during the past year.

IP phones—and, for that matter, IP PBXs—still have important obstacles to overcome. As noted above, they're not cheap, and so breaking into the market for large systems remains a challenge. Indeed, they seem best suited to new, "greenfield" installations, but with the economy being what it is, there are likely to be fewer of those, at least for a while.

But those issues notwithstanding, the long-term trend is clear: Sometime after 2005 IP telephones will become the dominant desktop voice terminal.

What does that mean for you? Well, if your PBX system is less than five years old, your digital telephones are likely to remain around for a while. But after your next upgrade or system replacement, you'll probably be using IP telephones□

Companies Mentioned In This Article
Avaya (www.avaya.com)
Cisco (www.cisco.com)
Mitel (www.mitel.com)
NEC (www.necamerica.com)
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