INTRODUCTION
Voice over Internet Protocol (VoIP) is the enabling technology for this service integration. In short, 64Kbps voice is converted into data packets and transferred over the data backbone. The savings of voice and data over a single corporate network can be considerable and the technology exists. But where does a company begin? This article reviews the enabling technologies and presents a viable private network evolution strategy that takes advantage of current technologies and standards. Savings are realized almost immediately.

VOICE AND DATA NETWORKS
Companies with private voice networks own one or more private branch exchanges (PBX)s to deliver the service. A PBX is the switching element that link two users together in a voice connection. PBXs can be broken done into the following basic functional components:

- Wiring: PBXs require dedicated wiring to each telephone in the company. This allows employees to call other employees connected to the PBX. To gain access to the public service telephone network (PSTN), PBXs require outside lines purchased from the local public telephone company.

PAYOFF IDEA
Today, voice and data are carried over separate networks. Companies with private voice and data networks typically have two sets of wiring throughout the location, different equipment to handle the two services, and two support departments. What if it was possible to integrate both services onto a single corporate network? First, consider the savings in wiring, equipment, and personnel to support a single network versus two networks. Then, consider the public network access savings if only a single pipe was required for both services instead of dedicated voice and data channels.
• Hardware: PBX hardware includes line cards terminating the local wiring, a switch network that makes the connection between two telephones, and servers for PBX software.
• Software: PBX software controls the call setup and features such as call forwarding, call transfer, and call hold; software also provides operation and maintenance support as well as per-call statistics.

Companies with multiple locations require dedicated PBXs at each site and public leased lines connecting the multiple PBXs into a single logical network.

Private data networks are delivered using equipment known as routers and hubs. Similar to voice networks, data networks also require dedicated wiring between the different network elements and to each desktop. PCs, servers, printers, and scanners are some of the data devices that can be connected to the network to deliver specific services. Once networked, these devices communicate with each other, passing information in the form of data packets. Exhibit 1 compares the voice and data network topologies.

EXHIBIT 1 — Comparison Between Private Voice and Data Networks
VOICE OVER INTERNET PROTOCOL
When a user picks up and dials the telephone, the PBX provides dial tone, collects digits, and routes the call based on entered digits. In routing the call, the PBX sets up a dedicated connection between the two users. As the users speak into their phones, the PBX converts the analog voice into digital signals and sends the digitized voice across the connection at a rate of 64Kbps. The PBX converts the analog signal to digital format by sampling the analog voice every 125 microseconds and converting the voice sample to an 8-bit digital representation. Digitized voice is converted back to an analog signal at the terminating end of the connection.

IP data networks, on the other hand, handle device communication by packaging information into logical envelopes known as packets, address-
ing the packets with destination information, and routing the packets through the network. Each network node looks at the addressing information and passes the packet onto the next network node until the packet arrives at its final destination. This routing method is connectionless because there is no dedicated connection between origination and termination as there is with a voice connection. A side benefit of connectionless communication is the fact that bandwidth is only required when there is information to send.

VoIP merges voice and data technologies into one ubiquitous voice service. Analog voice signals are sampled and digitized the same as the steps performed by the voice network. However, rather than send the 64Kbps digitized signals over dedicated voice channels, the digitized voice is packetized and sent over the IP data network. Furthermore, bandwidth efficiencies can be realized if the voice signal is compressed prior to packetization. Voice compression can lower bandwidth demands as low as 4Kbps while maintaining near toll-quality voice. This means that a single 64Kbps channel that normally carries a single voice conversation in the voice network can carry up to 16 voice conversations in the data network.

To support call control, the International Telecommunication Union (ITU-T) has standardized the H.323 protocol. H.323 describes terminals, equipment, and services for multimedia communication over a local area network (LAN), voice being just one service supported. Other recommendations within the H.323-Series include H.225 packet and synchronization, H.245 control, H.261 and H.263 video codecs, G.711, G.722, G.728, G.729, and G.723 audio codecs, and the T.120-series of multimedia communications protocols. Taken as a whole, these recommendations provide the standards to which many backbone, access, and customer premise equipment vendors are developing VoIP components and guaranteeing interoperability. Cisco, for example, has integrated H.323 protocols and voice capabilities into each of their newest routers. Microsoft and Intel also have H.323-compatible products for voice and data communications. The next section provides an overview of product categories.

VoIP PRODUCTS

H.323 VoIP products can be broken out into the following general product categories that loosely map to network layers:

- **Customer premise equipment (CPE):** Originally, H.323 CPE was mostly developed for multimedia communication. Products such as Microsoft Netmeeting, Intel Proshare®, and PictureTel LiveLAN™ were marketed for their desktop video and data sharing capabilities. More recently, however, CPE products have been developed that are
more focused on IP-voice communication. The Selsius Ethernet phone, for example, resembles a standard phone but connects to an Ethernet port rather than a PBX port. Symbol Technologies has developed a wireless H.323 telephone that also connects directly into an Ethernet port.

- **Network infrastructure**: Network equipment includes standard routers, hubs, and switches. Voice, however, is very susceptible to network delays and packet loss. Router features such as random early drop/detect, weighed fair queuing, reservation protocol (RSVP), IP precedence, compressed real-time protocol, and multi-class multi-link PPP have been developed to address these issues and are prescribed in a converged network. Asynchronous transfer mode (ATM) network equipment also supports quality of service parameters and enables the administrator to set up dedicated private virtual connections (PVCs) between sites.

- **Servers**: A major VoIP benefit is the fact that it builds on the Internet model where there is a clear separation between network infrastructure and network applications. While network infrastructure provides the packet transport, servers support the applications. The H.323 gatekeeper server, for example, supports call control functions; an accounting, authorization, and administration (AAA) server provides billing and accounting; and network management typically requires a simple network management protocol (SNMP) server.

- **Gateways**: It will still be a long time before one sees the data network handling all voice communication. As an interim step, gateways will be necessary to link the new IP-based voice services with existing public and private voice networks. Gateways also provide a good evolutionary step for companies to selectively choose how and where they will apply VoIP for low risk and high returns.

**VOICE AND DATA NETWORK INTEGRATION**

So, the technology, standards, and products exist to migrate voice onto the corporate data network. But where does a company begin in order to minimize network risk and optimize savings? There are two major areas for network cost savings in a voice network: PBX private service (single location and multi-location) and public service (local and long distance). There is also another category representing value-added services that are enabled due to the voice, data, and video integration. These value-added services offer potential for companies to reengineer existing services such as automatic call distribution (ACD) services and build new services such as click-to-dial, where an Internet user can access a company’s operator through the Web.

The following sections discuss each category in further detail. Included is a proposed network integration plan as well as future value-added
services to consider when building the next-generation converged communication network. Refer to Exhibit 2 for a summary of the proposed private voice network evolution.

Private Voice Service
Purchasing a PBX is a big decision for a company. For one thing, it is expensive; and once purchased, the company is locked into the specific PBX vendor for hardware/software upgrades and maintenance. Companies are also limited to the call control features supported by the PBX so that little differentiation is possible between themselves and their competition.

In an IP world, networks are built in layers. Companies first choose the underlying infrastructure, for example, routers, switches, or ATM equipment. Then, companies choose the application protocol; IP for VoIP. Finally, they decide the specific services, servers, and CPE that they wish to run over the network. Taken as a whole, the decision is just as important as the PBX decision. But taken separately, each decision is much lower in cost and risk; new technologies can be introduced with little or no impact to other network layers.

VoIP assumes that there is already a data infrastructure supporting IP services; IP data infrastructure is a mature market with many third- and fourth-generation products. VoIP, however, is in its infancy. The most mature of the products are VoIP gateways providing interworking between the voice and data networks. Gateways terminate voice interfaces from a PBX or public voice switch and convert the signaling and voice to IP protocol. Gateways then send the information across the IP network to either another gateway or a VoIP CPE.

Gateway-to-gateway communication is a good first step for a company introducing VoIP capabilities. Back-to-back gateways provide an alternative to dedicated inter-PBX trunks, typically leased lines — which are very expensive if purchased from a public carrier. Instead, gateway-to-
gateway service allows the network administrator to redirect leased-line inter-PBX traffic over the internal corporate IP network. Gateway costs can be recovered in months, depending on the number of locations and distance.

Gateway-to-gateway communication can also be used to link a remote office to the corporate office PBX. The remote office would appear as one or more extensions off the corporate office PBX, and traffic to and from the remote office would be routed through the PBX. Cisco, for example, has integrated VoIP capabilities into both its small remote office routers as well as larger corporate office routers to support such an application.

Introducing H.323 CPE to the network is a good second step in the corporate network VoIP evolution. This enables IP users to originate and terminate calls directly from their PC, Ethernet phone, or wireless IP-phone. IP devices add a level of network complexity due to the potentially large number of users, security issues, and varying IP device types. The H.323 gatekeeper provides the needed user authorization and routing capabilities. Many vendors have introduced gatekeepers, both integrated into the router and contained on separate servers. Both solutions offer viable alternatives.

With the combination of gateway, gatekeeper, and IP device, companies can begin to introduce business solutions to work at home and CTI applications. Basically, the IP device can be logged into the corporate data network at the same time as calls are originated and terminated from the device. Help Desk operators working from home, for example, would require only a single link into the corporate data network to answer customer questions and access corporate information.

The third major step in evolving the private voice network is fully replacing the PBX with a next-generation IP-PBX. As discussed earlier, PBXs rely on wiring to the desktop and provide hardware and software for call control, accounting, etc. In a VoIP network, Ethernet wiring provides the desktop wiring, the data backbone provides the transport, and servers provide the PBX call control features. The wiring and data backbone exist, while companies such as Selsius and NBX are introducing the server and CPE components. It will not be long before IP-PBX features match and surpass capabilities of current PBXs and the IP-PBX becomes a viable alternative.

Value-Added Services

One area where IP-PBX features will surpass PBX capabilities will be value-added services. These are services that take advantage of voice, data, and video integration as well as IP network and device capabilities to deliver services not possible on a PBX. Some of these services offer significant cost savings, while others offer revenue-generating opportunities.
Following are several value-added service examples to consider when implementing a VoIP network. Note that many other services are possible by adding new servers to the network similar to adding new Web sites to an IP network.

- **Unified messaging.** Most corporate managers receive voicemail, e-mail, and faxes on a daily basis. While timely information is one of the keys to a successful business, it is also difficult to manage the information, especially when it arrives in different forms and locations. Unified messaging consolidates the different information into a single mailbox accessible from a telephone, Web site, or pager. The unified messaging server supports media conversion so that e-mails can be read to phone users, faxes can be redirected to local fax machines, and voicemail can be converted to audio files and played back on a computer. Future enhancements may include videomail and interactive content.

- **IP call center/click-to-dial.** The number of Internet users and the amount of Internet commerce are growing faster than most estimates. Intel, for example, just announced that it is generating over $1 billion in revenue per month directly from the Internet. Cisco also generates over 50 percent of its revenue from direct Internet sales. VoIP offers further opportunity to reach customers through the Internet by enabling direct and immediate interactions with customer service representatives. The click-to-dial application enables users to click a Web site button to automatically set up a VoIP call between user and operator. IP call centers extend the model so that the call completes between IP user and IP attendant, thereby lowering 800 service costs.

- **Active directory services.** There is much electronic information within a corporation and contained on public servers. Active directories enable users to interact with the directory, potentially clicking an employee name and connecting to the user through the Internet. This service integrates the data server with VoIP capabilities and strengthens the argument for integrated voice and data services.

- **Virtual second line.** Eventually, VoIP devices will be in widespread use within the corporation and residence. Much of their success will depend on how they are addressed. Dialing an IP address to reach these devices is cumbersome and doomed to failure because it changes the way people are accustomed to making a call. The possibility exists, however, to address the IP device with a standard telephone number. This strategy merges seamlessly with today’s telephones and cellular phones and provides a good evolutionary path to the next-generation communication network.

- **Follow-me/find-me.** Once the IP device has an assigned telephone number, many standard telephone features can be carried over from the PSTN. Call forwarding, call transfer, and three-way calling are examples. Follow-me service allows users to predefined a search list of
telephone numbers at which they may be contacted (e.g., work number, home number, and cellular number). When an incoming call arrives, the service rings each number on the list and, once located, transfers the incoming call to one's current location. It is an intuitive step to add the IP device telephone number to the list and direct incoming calls to the IP device when active.

- **Desktop video conferencing.** VoIP specifically refers to handling voice service over the IP network. The benefits of VoIP, however, extend beyond voice. If the devices at either end of a VoIP connection support voice, video, and data, then the underlying signaling will set up a multimedia connection. No user interface change is required; the user still dials a standard telephone number. For example, a company that builds an internal VoIP network can take advantage of full multimedia communication between employees. Gateways can provide the interoperability with older telephone devices and links to the public voice network. Not only are costs reduced because voice services are moved onto the private data network, but internal services are enhanced with video and data as well.

**Public Voice Services**
Public voice carriers have also recognized the potential cost savings and new revenue from VoIP solutions. Most carriers have ongoing trials to better define market segments and decide where best to apply internal resources. With the convergence of voice, data, and video, additional carriers are entering the foray. Internet service providers (ISPs), competitive local exchange carriers (CLECs), inter-LATA exchange carriers (IXCs), and cable companies have all begun investing in VoIP solutions. Some are looking to enhance existing services, while others are looking to compete with incumbent carriers.

This is all very good news to companies looking for competitive public service rates because increased competition is expected to drive down local and long-distance service rates. Already, VoIP-based companies such as ITXC have begun to offer cut-rate international and long-distance calling card services.

It is also good news for companies evolving their networks to VoIP solutions. In the short term, they can be assured that vendor priority is kept on standards and interoperability. Longer term, as carriers begin offering VoIP public services, company voice traffic can seamlessly transfer from private to public networks without the need for gateways, moving end-to-end via VoIP.

**CONCLUSION**
The communication paradigm is on the verge of change, signaling a new generation of lower-cost voice, video, and data services. Companies have
the opportunity to take advantage of this paradigm shift in a low-risk and cost-effective manner. VoIP is the enabling technology to begin integrating voice, video, and data services onto a single data backbone network. VoIP to PSTN gateways are a good first step to begin the convergence because they are established products with many competitors. Gatekeepers and IP CPE are close behind, with the potential to introduce new value-added services. Finally, there is the opportunity to rebuild the infrastructure with IP-PBXs and public VoIP offerings. This enables ubiquitous voice, video, and data services over a data backbone and offers services one can only begin to imagine.

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