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The Open Systems Approach to Voice Over IP

A Practical and Future-Proof Way to Embrace VoIP Technology

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About Artisoft

Artisoft is a leading developer of software-based phone systems for small-to medium-sized businesses and corporate branch offices. Artisoft's TeleVantage product has received more than 25 industry awards, including "Product of the Year," "Best of Show," and "Editors' Choice" from Network magazine, Computer Telephony magazine, Communication Solutions magazine and CTI magazine, among others. The company distributes its products world-wide.

For more information about Artisoft products, visit Artisoft's World Wide Web site at www.artisoft.com or contact the company at 1-800-914-9985.

As demand for Voice over IP (VoIP) accelerates, developers of switching systems are offering different approaches to deploying IP phone systems in small- to medium-sized businesses. One approach, called the open systems approach, capitalizes on the cost and applications development advantages of IP technology to bring productivity improvements to end users. Because an open system uses both circuit-switching and packet-switching, an expanding array of converged voice and data applications are available to end users, strengthening each end user's ability to create an efficient work environment and customized interface with customers and suppliers.

Benefits of IP Telephony

As the mobile workforce expands, and companies establish virtual offices, home offices, and small branch offices, IP telephony provides applications' functionality to all users, allowing them to maximize their productivity whether they are in the home office or working remotely. Deploying Voice over IP (VoIP) provides infrastructure flexibility for growing businesses and improved communications functionality for end users, even as it reduces facilities expense. As VoIP technology advances, these benefits will become compelling for an ever-widening spectrum of businesses.

Reduction in Infrastructure Expenses

A key advantage of deploying IP telephony is that it allows a company to standardize on a single communications infrastructure. Today, businesses typically purchase and maintain two separate communications systems: one for voice communications and one for data communications.

The voice communication system includes a telephone switch (PBX), wiring, and a connection to the public service telephone network (PSTN), and the data communication system includes a router, Ethernet wiring, Web server technology, and a connection to the Internet. These two communications systems require two different sets of purchases. With VoIP, however, an organization can start reducing costs immediately by utilizing the same infrastructure for both voice and data communications. For example, by routing phone calls over Ethernet lines to the desktop, a company can eliminate the need to purchase and install separate wiring.

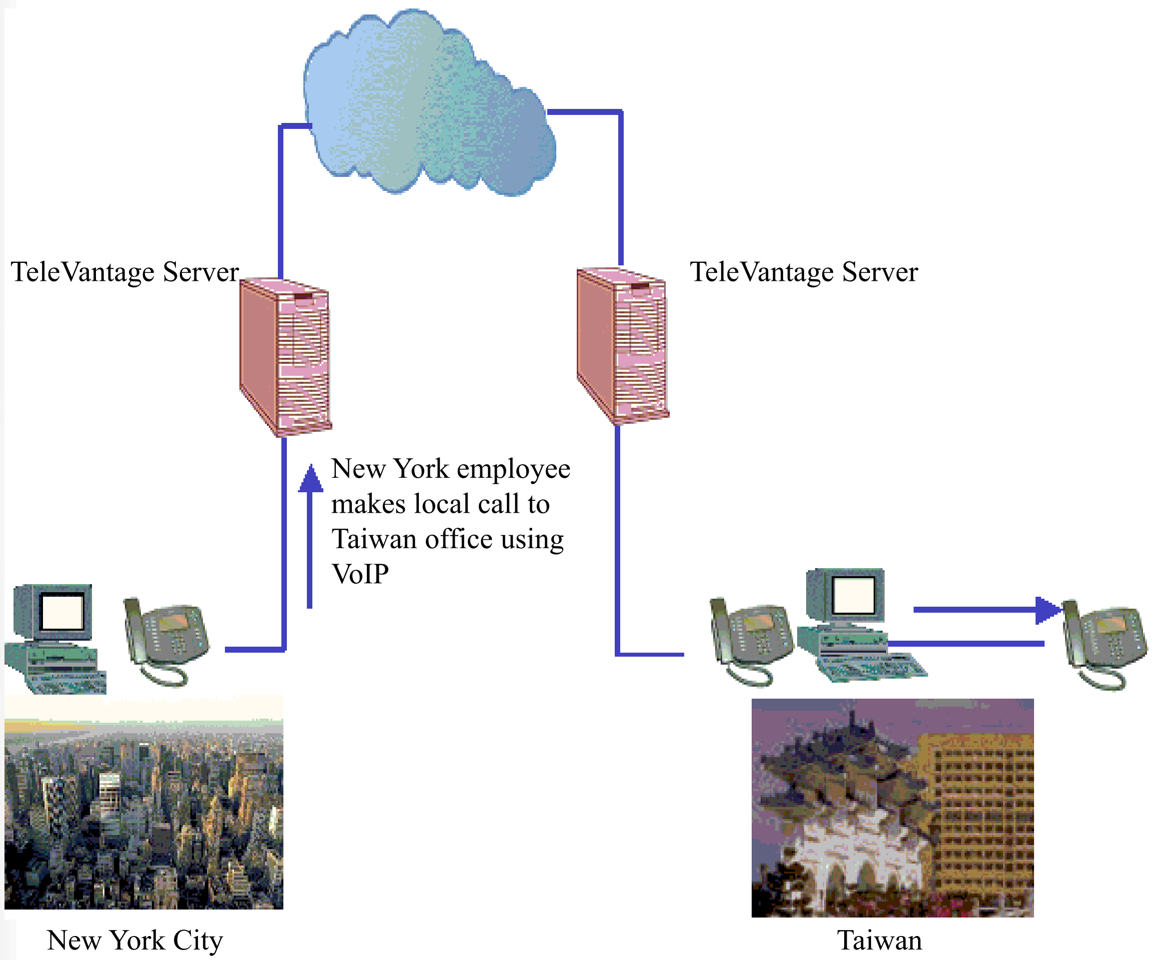
In addition, supporting and administering phone systems requires a specialized skill set that is altogether different than the skill set required for supporting and administering a data network As a result, businesses traditionally incur what is essentially a duplication of overhead expenses when they support separate infrastructures for voice and data communications. They must either hire or subcontract the labor required to administer each of these systems separately. As the computer industry has shifted away from proprietary technologies and settled on de facto industry standards such as Microsoft Windows, it has become easier for businesses to hire employees to administer their data systems internally. With VoIP, these same administrators can manage their phone systems as well.

Linking Remote Offices

Another benefit of VoIP is its potential to reduce long-distance telephone bills. Using packet-switched Wide Area Networks (WANs) to carry voice traffic can offer significant savings, because companies can fully utilize leased data connections with excess bandwidth. With an Internet gateway, two phone switches (private branch exchanges, or PBXs) at different locations can be connected over the Internet or a private IP network. The two switches are unified into a single user environment in which all users appear to be local. Connected with VoIP, users at one location can make calls as if they were physically present at another location. For example, an employee in New York can dial an internal extension and, using an Internet connection, reach a colleague who is located in a branch office in Taiwan. The company does not incur long-distance toll charges and pays only for the Internet connection.

For further cost savings, users can place long-distance calls through remote switches that are closer to the destinations of their calls. For example, a New York employee can call a customer in Taiwan by placing the call through the switch that is located in their Taiwan office. Although the call originated in New York, the company pays only for a local call within Taiwan. (See Figure 1)

Figure 1
Linking Remote Offices



Virtual Office

With an increasingly mobile workforce, businesses are continually evaluating ways to improve communication capabilities in home offices and for traveling professionals. VoIP allows remote users to enjoy the same advanced PBX features that are available in their corporate offices, while at the same time reducing costs.

VoIP capabilities allow remote users to place and receive calls from H.323-based terminals such as Microsoft NetMeeting, a real-time conferencing and collaboration tool included in Windows 2000, or from IP phones based on the H.323 standard. This gives users the ability to give out a single phone number and operate as if they were physically located in the corporate office, regardless of their actual location. The user can be part of the directory, the Automatic Call Distribution (ACD) hunt group, and has the ability to transfer calls, forward voice messages, and access all of the switch features. All of this is accomplished using a remote user's standard data connection, which eliminates the fees normally associated with rerouting calls over the PSTN.

Expanding Set of VoIP Applications

Beyond cost savings and flexibility, perhaps the greatest benefit of VoIP is its ability to enable converged voice and data applications. Companies can use VoIP to support a number of exciting new features such as Web-enabled call centers, click-to-talk Web site buttons, Internet call waiting, workplace collaboration initiatives, and Web-casts. VoIP also provides greater flexibility for employees and call center agents. For example, agents can view customer records on their PCs during phone conversations and combine e-mail and voice mail for improved productivity. In an open development environment, new VoIP applications that target the unique requirements of vertical markets, specific CRM integrations, and customized phone handling and internet transactions are being introduced at a rapid pace.

Approaches to IP

The company's specific requirements for VoIP infrastructure are determined by the quality of service (QoS) required for the IP network. To overcome packet loss and latency issues, protocols that reserve bandwidth or prioritize packets to guarantee consistent QoS for VoIP call traffic have been developed. Each router in the IP network, including ISP routers, must be set up to honor bandwidth and priority requests. Establishing QoS for real-time voice applications need not be a complicated and costly process, provided that the right approach and the appropriate PBX architecture are adopted. By optimizing the approach, the company is protected against unanticipated requirements to upgrade hardware and software in the future. When VoIP is planned as an integral part of the PBX architecture decision, the IT manager can anticipate future staff requirements and budgets at the time of the initial capital investment.

When considering the adoption of VoIP technologies, companies can choose from three alternative implementation approaches. The first approach, called the adjunct approach, consists of using an external gateway to interface a circuit-switched PBX to the IP network. The second approach is the proprietary IP-PBX approach, which extends data networks to handle voice traffic. The third approach is the open systems approach, which provides a standards-based infrastructure for both traditional circuit-based and VoIP switching.

Adjunct Approach

The adjunct approach consists of an external gateway that is connected to a proprietary PBX and that interfaces with the IP network. The adjunct approach is often selected because it appears to be the least expensive and the company believes it will be simple to install.

In fact, the adjunct approach creates functional limitations and excessive overhead costs because it forces the company to maintain multiple complex hardware infrastructures. Also, with an adjunct approach, the PBX does not handle Internet addressing, so the company must deal with two dial plans: the PBX dial plan and the routing guide administered through the gateway. Installation of the gateway does not simplify administration. Instead, administration becomes more time-consuming.

As the company grows and requires more IP capabilities, application functionality from either the gateway manufacturer or the PBX vendor may be limited. The adjunct approach does not address the company's initial or ongoing requirements for VoIP. This architecture does not natively handle Internet addressing or provide new application functionality for end users.

Proprietary IP Approach

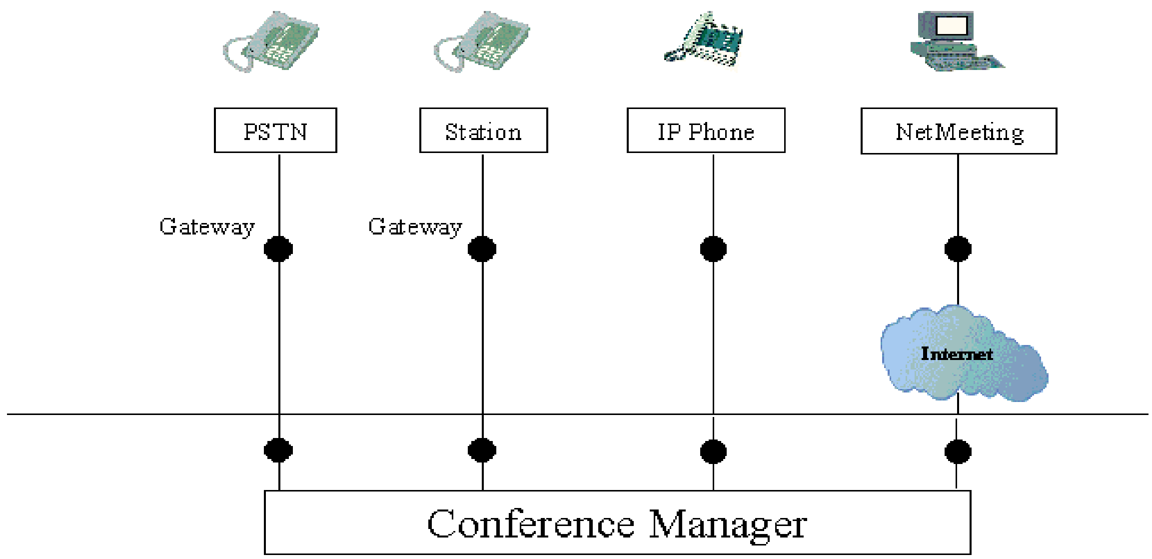
In the past few years, several vendors have introduced IP-PBX products that are designed to manage all communications over the IP network. With this approach, called the proprietary IP-PBX approach, *all voice processing* is brought into the existing data-switching architecture. This approach introduces significant technical complexities that must be addressed successfully. If they are not, users may face a potential degradation in the quality of their phone service.

Quality of Service

With a proprietary IP-PBX, all voice traffic is packetized, even in cases when it is unnecessary or undesirable to do so, such as when setting up conference calls, faxes, and calls over the PSTN. Typically, this is the majority of a company's calls. By packetizing the call, the IP-PBX is converting a telephone conversation of continuous speech into distinct segments, or packets, and then reassembling them as audio operations are performed in the data network.

The assembling and reassembling of voice packets can often result in degradation in the quality of service of the call. In some scenarios, such as conferencing, this process of multiple packet compressions and decompressions can have a particularly damaging impact. With more routers and more hops than a circuit-switched architecture, some latency (the time it takes for a packet to transfer from one IP node to another) or packet-loss issues can develop. This causes a "delay" in the conversation and a "push-to-talk" feel when one speaker perceives an interruption of the other speaker. In some cases, this latency can even cause the conversation to sound distorted. Figure 2 shows the packet assemblies associated with the proprietary IP approach.

Figure 2
Proprietary IP Approach



For intra-company communications (or home consumer use), the cost savings of carrying the call over the toll-free intranet may outweigh the potential loss in quality. However, for the majority of a company's calls, which originate from the PSTN or are transferred back to the PSTN, this conversion has no benefit and may result in an unacceptable loss of quality.

Limited Choice in Applications

One of the most important criteria in selecting a VoIP system is whether it will improve end-user productivity and create a positive customer experience. The key to this is the quality and availability of the software applications that interface with the system.

Although many proprietary IP PBX vendors claim to support "open systems," the reality is that their products have been built on proprietary architectures and are based on either vendor-specific interfaces or loose interpretations of industry standards. The vendors' goal is to restrain customers from replacing components of their IP PBXs with third-party products.

While this approach may serve the interests of the IP PBX vendors and its resellers, it has a negative effect on its customers. In addition to locking customers into proprietary hardware, the proprietary architectures of IP PBXs discourage third-party developers from creating add-on applications. Instead of being able to choose from the wealth of applications that flourish in an open systems environment, customers are thus limited to the small set of applications that a single vendor and a small group of vendor-specific partners can provide.

Open Systems Approach

A third approach to VoIP is the open systems approach. Unlike the other two approaches, an open system requires no proprietary hardware. Instead, it uses industry-standard interfaces and runs on commercial, off-the-shelf hardware.

By utilizing a comprehensive set of standards, an open system can control switching and voice-processing hardware without becoming entangled in the hardware implementation details. Underlying hardware, signal processing, and communications links can be changed without forcing users to discard the software that defines voice and IP communications at their companies. Because the software is compatible with industry-standard telephony hardware and computing platforms, it provides plug-and-play compatibility among software, hardware, and all peripherals, which permits great flexibility in implementation.

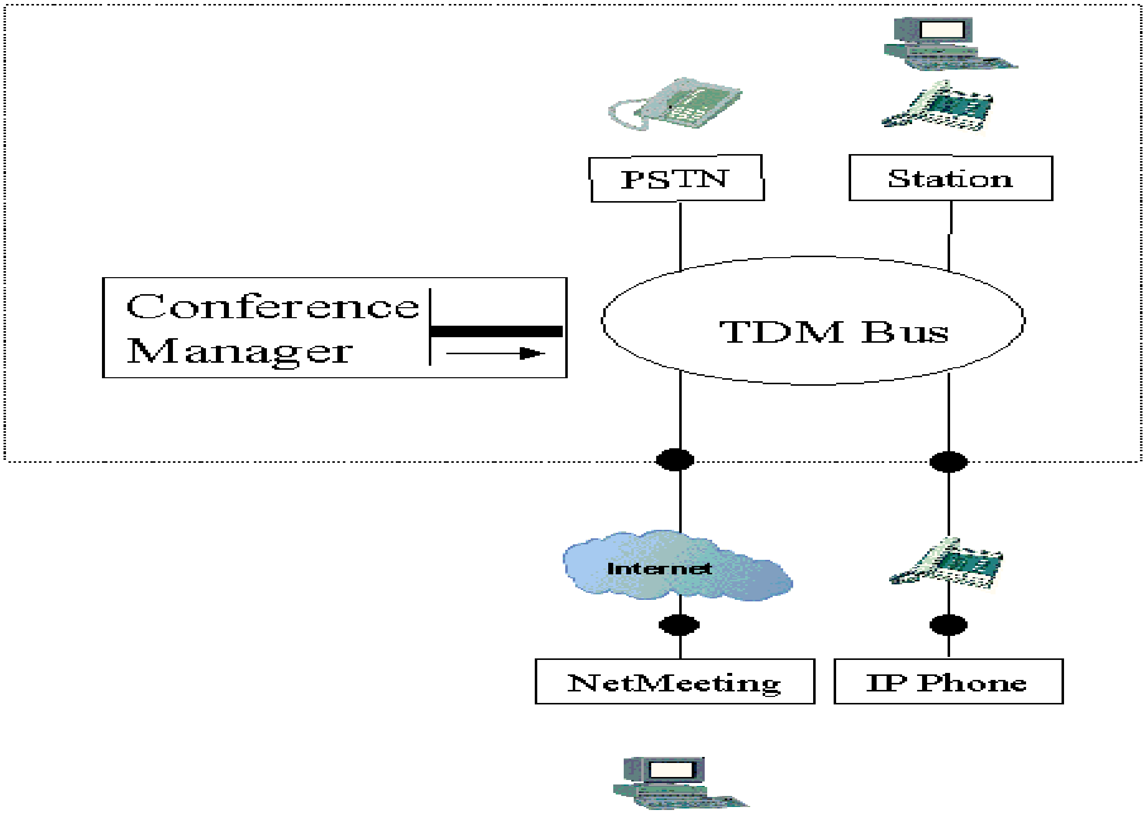
The advantages of the open systems approach are threefold: First, its flexible architecture is optimized to eliminate the latency and QoS issues associated with multiple-packet assemblies. Second, it provides multiple economical choices for phones and terminal devices. Finally, as the ideal infrastructure for expanding application choice in the future, it provides businesses with tremendous investment protection.

Practical Utilization of VoIP

An open-system architecture utilizes VoIP when it is cost-effective or when it supports specific features that make use of the Internet. Because most corporate or organizational WANs are not homogeneous, the open system is often deployed as part of a corporate network that has been optimized to utilize the least-cost alternative for each connection. For example, a frame relay link may be superior to a point-to-point fractional T1, or frame relay encapsulation might be used over a point-to-point T1 link to provide the capability for multiple virtual circuits. With an open system, an IT manager can make the decision to upgrade bandwidth on any particular link in the network, if required, to support VoIP traffic. The quality and cost isssues associated with VoIP can be constantly weighed and optimized.

The flexibility of an open system allows companies to evolve to VoIP rather than implementing it all at once. For example, a company may utilize VoIP for remote workers and facilities where only data networks are wired, while handling the remaining calls through circuit switching. Features that are difficult or expensive to implement over VoIP, such as call recording, supervisory monitoring, or music on hold, can continue to use circuit switching while Web-intensive call center applications make use of VoIP. Figure 3 shows a conference call established using the open systems approach. Packet assemblies are utilized to interface with the Internet and IP phones.

Figure 3
Open Systems Approach



Choice in Telephones

In addition to analyzing the quality and cost issues associated with company networks, a company must consider its choice in telephone handsets as part of the decision to purchase a VoIP system. Over the past few years, IP telephones have become less expensive and begun to offer more features than they did when first introduced. However, they are still more expensive than existing analog or digital sets with comparable features, and they may not meet all employees' needs.

An open system gives companies a multitude of choices in handsets and allows them to use a combination of non-IP and IP handsets. They can also mix and match handsets from different vendors. For non-IP handsets, there are analog, digital, 900 MHz, 2.4 GHz, cellular, conference phones, overhead speaker devices, and fax machines that range from simple, inexpensive offerings to high-end, feature-rich executive phones. For IP phones, there are a variety of choices that offer a range of feature sets and price points. Because the market for phones is competitive, manufacturers continue to enhance their products and bring new features to market that provided wider choices to customers. Customers can add IP telephones incrementally, rather than throughout the company, reducing up-front costs in the transition to VoIP.

Industry Standards Provide Customer Choice

To realize the full benefit of VoIP, businesses need access to a broad range of multimedia applications that have been written for both circuit-switched and IP networks. By implementing a VoIP system based on industry standards, customers can choose from a wide variety of third-party application vendors and employ best-of-breed applications that meet their ever-changing needs.

To ensure plug-and-play compatibility among different applications, the telecommunications industry has adopted a standard set of interfaces. The most important of these standards is the S.100 standard. Approved by the Enterprise Computer Telephony Forum (ECTF) and endorsed by leading technology vendors such as Intel, Microsoft, Compaq, H-P, and IBM, the S.100 standard provides a common way for applications to interact. In addition, the S.100 standard allows customers to replace a single component in their communications infrastructures instead of upgrading the entire system. It also ensures competition between multiple vendors, which drives down pricing and motivates vendors to deliver new features.

Artisoft's TeleVantage – A Smart VoIP Investment

As the industry-leading software PBX, Artisoft's TeleVantage is designed to accommodate the requirements of growing businesses as they adopt VoIP technology. Built on industry standards, TeleVantage uses the open systems approach, allowing customers to implement a practical VoIP strategy and choose from a wide variety of handsets and third-party applications.

Artisoft's TeleVantage is designed to meet the VoIP business needs of the future. Its hardware independence provides customers with maximum flexibility and makes it easy for them to embrace advances in technology. Because the functionality of TeleVantage is contained in software, customers continually have access to new features through inexpensive software upgrades. By giving customers the flexibility to enhance and customize their communications infrastructures over time, TeleVantage provides a great investment in the future.

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