VoIP Solutions – getting ahead of the game

by Jeff Wen

All over the Internet banner ads tout "Free Long Distance Calls" or "Talk for Free over the Internet." While Voice over Internet Protocol (VoIP) is nothing new, corporate America has been watching the steady leaps in the technology and is now poised to take B2B communications to a new platform level. The integration of telephony services into the data network has the potential to drive major changes in institutions, delivering information more efficiently than today's dual-network approach.

VoIP, converged voice, video and data networks are improving customer care capabilities, employee productivity, enhancing corporate agility, and reducing costs. This is especially true for call centers that need to support a wide variety of e-com sites as well as brick and mortar companies with both telephone and live Internet support.

Companies also benefit in voice communication cost reduction. Because VoIP calls travel as data over a private IP or the public Internet, instead of over telephone lines, calls are not bound by regional calling charges. Companies with multiple business locations that are already linked can easily add VoIP to reduce costs of standard voice systems at a fraction of the cost of traditional tie-line services.

VoIP gateways provide predictable, real-time, toll-quality voice and fax communication using existing IP data networks or the Internet. This article will overview how, in the present applications, VoIP gateways bring affordable business communication solutions to small and medium-sized businesses especially for businesses with geographically dispersed workgroups.

Designing a Voice over IP gateway

Almost all VoIP solutions are true open systems built on industry standard hardware and developed to meet industry software standards. The use of industry standard interfaces allows different manufacturers' gateways to interface with other products without being locked into a proprietary system.

The gateway server converts telephone signals into digitally compressed information packets. The server then routes the packets using the Internet Protocol (IP) on a private or public network. The gateway at the other end of the network receives the incoming packets, and reverses the above operation. It then distributes the telephone signals through the receiver's telephone network.

The simplest method of applying VoIP technology is the use of a multi-line PCI card plugged into a desktop computer. This small, but comparably limited expansion solution satisfies the home office and small office application; however, most solutions require the use of an enterprise server (Dell, Compaq, Gateway, IBM, et al.) or more rugged industrial computer chassis Advantech, ICS, MiTAC, Texas Microsystems, et al.). Industrial chassis, such as the one shown in Figure 1, tend to offer much higher flexibility – number of expansion slots, backplane, or motherboard solutions – ensuring future growth and added features.

Figure 1

With a variety of plug-in application cards and built around a CompactPCI or ISA/PCI chassis, these PC-based phone switching systems, newly marketed as CommServers, retain basic PBX functions while providing enhancements beyond standard phone features. The basic system consists of a motherboard computer with 7, 10, or 14 slots or a rack mount chassis with a slot-one controller and 5, 10, 14, or 20 slots for application boards.

CommServers are scalable, allowing expansion by adding another VoIP adapter for 2, 4, 8, or 16 voice/data channels, or even another chassis overcoming the fixed maximum number of users for traditional PBX systems, which typically require a "forklift upgrade" when the maximum number of users has been depleted.

If premium data safety, accessibility, reliability, and scalability are an issue, one may also want to install a RAID system. This redundancy prevents the risk of lost data and minimizes system downtime. Some chassis have the RAID capability built into the controller card and have sufficient disk available to do this. Others have an Ultra Wide/Ultra2 External SCSI RAID subsystem to add hot-pluggable disk drives online making it easy to expand the system.

Software

Once a reliable computing system has been selected and the number of extensions has been configured, application, management and operating system software must be selected. The OS is a relatively easy choice of LINUX or Windows NT/2000. This choice is more dependent on the application and management software but most software packages are supporting both Windows and LINUX.

As ease-of-use is important, the software should have an intuitive GUI format that allows full viewing of all calls in process and the control of features such as call blocking, compression selection, intelligent routing instructions, and other enhanced services. This application software gives global enterprises the ability to route voice calls and faxes over a LAN, WAN, or Intranet/Internet circuit. In addition, at each location, a gateway server locally routes call and data directly from one location to another, while maintaining full support for PBX functions such as conference calling, follow-me, voicemail, and tone recognition.

These integrated gatekeepers add robustness to large multi-site networks while providing centralized remote management for full control of every unit. However, individual customized Call Detail Records can be written locally for each gateway and copied to a central database.

These software packages also support a dial up connection, whereby a remote user can send and receive email, fax, and voicemail, plus obtain a dial tone and place calls (which originate from the gateway) – all from anywhere on the Internet.

One of the more modern conveniences most companies must have is an automatic call distributor (ACD). Most VoIP solutions provide this as an option, either in software or combination hardware/software package. Other VoIP software applications allow users to dynamically handle incoming calls. These let the user predefine call-handling actions so that when incoming VoIP calls arrive, the application launches a pop-up window that offers:

- **Answer the call via PC:** answers the call while maintaining the current Internet session.
- **Answer the call via PSTN:** answers the call by interrupting the current Internet session and routing the incoming call to a traditional telephone.
- **Send to voice mail:** sends the call to a traditional message machine or records the voice mail message as a WAV file and attaches it to e-mail for later retrieval.
- **Redirect:** forwards the call to a user-defined number, such as a cellular phone.
- **Reject:** plays an announcement to the caller stating that the person is unavailable.

Complex skill-based call routing and data and call management are also important. Because VoIP systems are built around a powerful CPU, the software, and associated customer information allows routing to the most skilled operator.

Features and benefits

Figure 1 details a basic VoIP system, an open system built using industry standard and readily available hardware and software. It has many advantages over previous systems including:

Reliable – Reliability is everything. Revenue producing nodes must stay up. In the selection of an IP telephony gateway, carriers must research the reliability of the product being evaluated, as well as the response to be expected when a problem does arise. This unit is an industrial PC-based system capable of 99.999% reliability. All components factor into the reliability so it is more important to select quality products than price.

Scalable – Scalability further contributes to reliability. Multiple systems can be linked as one node for expansion and redundancy. Traffic can be redirected so preventive maintenance or upgrades can be performed with zero downtime. These systems racks are then cascaded in a rack enclosure. Literally thousands of ports can exist in one location, as one node, with fully automated load transfer between systems in constant operation.

Easily managed – The full functionality of gatekeeper software should be incorporated into every unit. By giving every system in the network the same abilities, the network can be easily managed from any point. In a typical gatekeeper/gateway topography, if the gatekeeper goes down, so does management of the entire network. By removing this single point of failure and giving each unit full management and functionality, if a single unit does fail, full management of the network is simply performed from any other unit.

Interoperable – H.323 protocol is the leading standard. All products, including the chassis suppliers, are beginning to support H.323 above all others. This standard allows users, from anywhere on the Internet, access to the gateway via a fixed IP address and pick up a dial tone or transfer to another gateway.

When implementing a system, it is always important to choose a vendor who understands and can customize each machine to function in the country and city where it is being used. Companies can build a good product that performs well in one country yet fails in the particular location where it is most needed. Interfacing with the changing protocols of a local PTT is a critical part in any Internet Telephony network.

Cost effective – Businesses can now achieve higher efficiency from their data network while cutting their overall communications costs by up to 85%. And with measurement and billing reports, business can be generated more flexibly, and can be further processed by other application software.

Conclusion

The Information Age has produced the Personal Computer, the Internet, and digital technology. These innovations have changed the way we work and play, and are now beginning to impact the way in which we conduct voice communications. Today's open systems are built using industry standard hardware and software and delivering high quality voice and data exchanges over an IP network.

VoIP is dramatically changing the marketplace. Many companies have begun the transition and are finding that they are improving customer and employee communications, support, and service while reducing their overall telephone costs. Current and next generation solutions are delivering real-time, toll-quality voice and fax communication over IP data networks and the Internet bringing affordable business communication solutions to businesses.

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Advantech has available three different system configurations highlighted below.

Low-end system:

- IPC-610F ~ 4U 14-slot rackmount Industrial Computer with 300W ATX power supply (Figure 2)
- PCA-6114P4 ~ Backplane (9 ISA/ 4 PCI/ 1 CPU)
- PCA-6178F ~ Socket 370 Pentium® III Coppermine SBC (up to 850 MHz) with HISA/ LAN/ VGA/ SCSI on-board
- 256 Mbyte SDRAM memory (minimum size required)
- IBM 9.1 Gbyte Ultra2 SCSI HDD
- SONY 48X IDE CDROM
- 1.44 Mbyte FDD

Software: Windows NT4 enterprise w/ cluster support
Additional LAN card is recommended for private network in clustering basic requirement (Suggest 3COM LAN card).

Figure 2: IPC-610F

Mid-range system configuration:

- IPC-623 ~ 4U 20-slot fault-resilient Industrial Computer with 400W ATX redundant power supply (Figure 3)
- Backplane options; PCA-6120P4 (15 ISA/ 4 PCI/ 1 CPU), PCA-6119P7 (11 ISA/ 7 PCI/ 1 CPU), PCA-6119P10 (8 ISA/ 10 PCI/ 1 CPU)
- PCA-6178F ~ Socket 370 Pentium® III Coppermine SBC (up to 850 MHz) with HISA/ LAN/ VGA/ SCSI onboard
- 256 Mbyte SDRAM memory (minimum size required)
- SCSI HDD is recommended.
- The other peripherals like FDD, CDROM are the same as the 610.

Software: Windows NT4 enterprise w/cluster support, or Windows NT Workstation, or Windows 2000 Professional
Additional LAN card is recommended for private network in clustering basic requirement (Suggest 3COM LAN card).

Customers may require an external SCSI card instead of onboard SCSI, for improved performance. We recommend using PCA-6178E w/ external SCSI card such as Adaptec.

Figure 3: IPC-623

High-end system:

- SPC-530 ~ 5U 20-Slot rackmount Industrial CT Server (Figure 1)
- PCA-6178VE ~ Socket 370 CeleronTM SBC (up to 600 MHz) with HISA/VGA/ LAN onboard (Figure 4)
- 10/100 Mbyte PCI Network Adapter (3 COM)
- V.90/Fled 56 K modem for remote service accessibility
- Display: S3 PCI display adapter on board ATI RAGE Pro Display
- 8-Channel voice/fax VoIP adapter
- Digital Signal Processor (DSP): AC 4801 F-M
- Analog network interface: Onboard loop start interface circuits
- Telephone interface: Loop start trunk type; 600 ohms nominal impedance; loop current range 20 to 80 mA
- Audio signal: Silence detection, transmit level, transmit volume control
- Audio Digitalizing-law or A-law PCm at 6.3 kpbs per voice channel

Software: Microsoft NT 4.0, Service Pack 6
CpIP Voice Gateway Manager
CpIP Billing Manager

Figure 4: PCA-6178