VOLUME 2, NUMBER 1

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The Authority On Computer, Internet And Network Telephony

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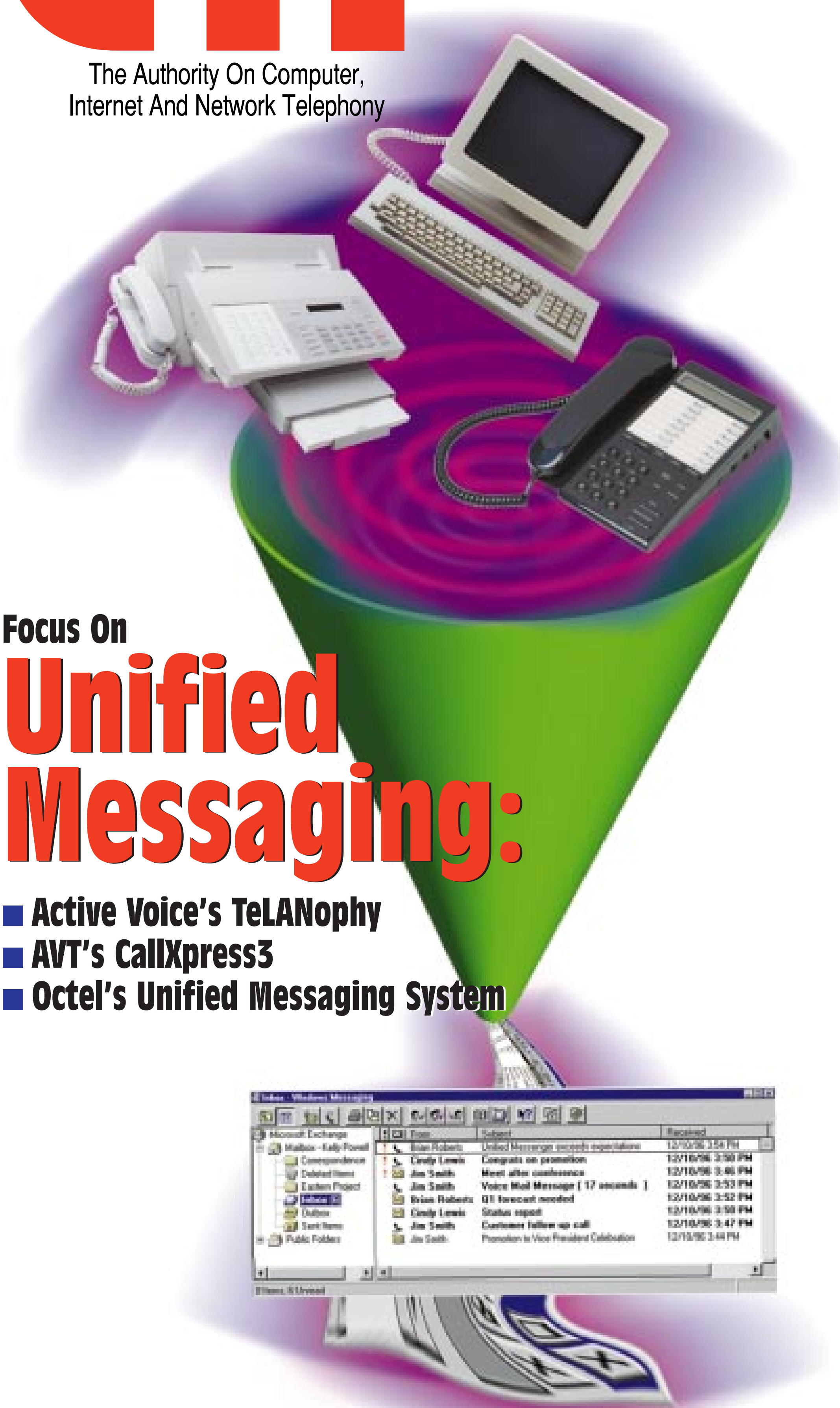

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CTI™ is a bimonthly publication dedicated to serving the information needs of MIS Directors, Telecom Managers, and Systems Integrators — the people ultimately responsible for making CTI work in any given implementation. Our editorial, which includes tutorial pieces, application stories, reviews, and product and service information, is designed to facilitate the work of these professionals and to further the development, acquisition, implementation, and use of CTI technology.

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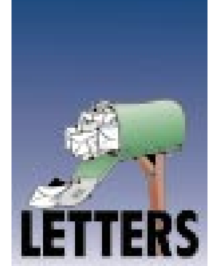

It's funny how things evolve in the high-tech industry. Every so often, we stumble across a "new technology" that is proclaimed to change the world as we know it. Immediately, a research group does a study on the usability of this technology and forecasts how much money it will bring to the industry.

Several years ago, research groups reported that the CTI industry was a multibillion dollar untapped market. The industry exploded. So-called CTI experts appeared overnight. Applications were created to accomplish relatively simple tasks — dialing and answering the phone. Everyone expected the money to come rolling in. Unfortunately, things did not go quite that smoothly. Some companies did make money, but the billions of dollars in revenue predicted by the research groups never materialized.

The reality of the CTI industry is that this new technology is really not all that new. While most of the new applications now have an updated Windows look, the processes being automated — dialing the phone or using the phone's key pad as a data entry tool — are much the same as they were ten years ago. And despite what many research groups say, most CTI applications are solutions looking for problems. Selling the solutions will generate some revenue, but in order for CTI to reach its full potential, the problems must not only exist, but must also be recognized.

I often hear and read complaints about CTI. Most say that it is too expensive and too complex. I like to compare CTI to a good air-powered nail gun. Once you use one, you never want to swing a hammer again. But air-powered nail guns usually aren't cheap. If you have a great deal of nailing to do, then the technology is well worth the investment. However, if you only have a few small nails to place, then the nail gun, coupled with the air compressor and special nails, will certainly be overkill, not to mention costly and hard to justify to your spouse. Plus, a nail gun demands special handling. Use it incorrectly, and you'll end up nailing yourself to the wall.

Such is the state of the CTI industry. We don't want to manually dial or answer the phone anymore, because we've seen the power of a great CTI application. Unfortunately, CTI applications aren't cheap. They require skilled installation and special hardware, and most applications are very complex. CTI pundits shouldn't complain about the cost and complexity, because that is the current reality of CTI applications. But in order for us to justify the cost of using CTI, we must first have a problem that can be solved by this wonderful, yet expensive and complex solution.

Tim Davis
President,
Davis Software Engineering

I'd like to take issue with Tom Keating's suggestion that we start using the acronym IHR (Interactive Hypertext Response). This acronym is unlikely to catch on. For one thing, IWR (Interactive Web Response) is already gaining currency. For another, interactions via the Web won't be limited to clicking on hypertext links. Multimedia is coming, as you must be aware, and formats such as MPEG will arise to stand alongside HTML. So, as far as the IHR suggestion is concerned, what we have is a case of too little too late.

James P. Carroll
Jakarta, Indonesia

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ROLM

SIEMEN'S ROLM

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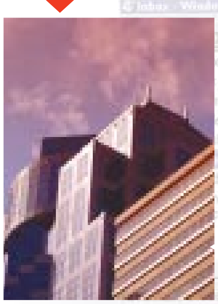

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PUBLISHER'S OUTLOOK

THE EQUALIZER: CTI IN THE SOHO ARENA

Fortune-class companies take note: smaller companies are taking advantage of CTI to move in on your territory! Many people have observed that smaller companies tend to be nimbler than large companies, and this tendency is being reinforced by CTI, a technology that gives everyone (not just corporate giants) flexible and powerful communications abilities.

This news may not be welcome to large corporations, but there is a silver lining. Large companies can also participate in the SOHO (small office/home office) phenomenon. For example, we will soon have technology that will allow any remote PC and/or phone to become an extension of a corporate PBX. SOHO is driving profound changes in the way we work, and individuals, small companies, and large companies are all reevaluating their options.

WHY IS SOHO ASSUMING SUCH IMPORTANCE?

Technology alone can't account for the growth of SOHO. Rather, technology is responding to a social demand, and this demand can be attributed to various factors:

• ***The need to conserve energy and reduce pollution.*** Anyone who commutes to work knows that traffic is murder. It's not unusual to hear people complain that they spend two, three, or even four hours each day getting to and from work. Moreover, we are poisoning the very air we breathe, and we are depleting our precious fossil fuel reserves. One answer to these pressing problems is telecommuting.

Of course, many people regard telecommuting with suspicion. How can someone be a serious and productive member of society if they work in their pajamas all day long? Well, we'll simply have to get over such prejudices. Already, legislation has been passed that promotes telecommuting. For better or worse, we will be using technology that allows us to work from our homes.

• ***The rise of two-income families.*** Declining living standards have made two-income families more the rule than the exception. And it's no surprise that such families are under a lot of strain, especially those with children. For such families, time saved from unnecessary commuting can be devoted to running errands, helping a child with homework, reducing daycare and babysitting needs, etc. Now *that's* "family values."

• ***The trend toward downsizing/outsourcing.*** From a social responsibility point of view, the merits of this trend are debatable, but no one can deny it's happening. Many "downsized" employees have ventured forth on their own, and some even enjoy the autonomy they gain. Regardless, everyone who takes this route should appreciate what SOHO CTI applications can do for them. SOHO CTI can enable independent workers to assume tasks that

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CC:

BY TOM KEATING

File Edit View Insert Format Tools Compose Help

To... CTI Subscribers

Cc... Active Voice; Altigen; Applied Voice; Automatic Answer; Cobotyx; Microsoft; Lucent; MediaMail; Octel; RealAudio; Others

Subject: Unified Messaging's Impending Triumph

Unified messaging will be huge. But how soon? That depends on at least two factors:

1) The speed with which remaining integration difficulties are resolved.

2) The time it takes potential users to appreciate that unified messaging is no "bleeding edge" technology. It is, in many instances, ready to go right now.

These issues are already being addressed. And, as we will see, unified messaging is bound to draw additional interest with the appearance of Internet applications that have unified messaging capabilities.

SUCCESS IS INEVITABLE

Unified messaging can't miss because it addresses an urgent need. People are overwhelmed by the quantity and variety of messages they receive. Indeed, as office workers struggle to respond to all their messages, they start to look like the plate-spinning acrobats who appeared on the Ed Sullivan show. (This comparison can be unfair. After all, office workers, unlike the acrobats, frequently let the plates fall.)

Disparate messaging systems present another problem: they waste time. Add up all the time people spend logging onto their e-mail software to retrieve e-mail, then logging on to their voice mail systems to retrieve voice messages, and then trudging off to the fax machine to retrieve their faxes. The waste is staggering and, more to the point, completely unnecessary.

Implementing unified messaging delivers a dose of sanity to the harried world of business (see the table entitled *How Unified Messaging Can Simplify Your Life*). Unified messaging allows users to send and receive voice, fax, and e-mail messages from one program. Voice mail, for example, can be encoded in several ways, such as in .WAV files, in a unified mailbox. (.WAV files are particularly convenient because any computer with a sound card can play them.) If your unified messaging platform conforms to e-mail standards, you can even forward voice mail and fax to someone who doesn't have unified messaging.

With unified messaging, users will be relieved of the burden of keeping track of disparate messages;

furthermore, they'll be free to concentrate on productive work.

WHY HESITATE?

It's amazing to me, given the state of technology we have now, that more businesses have not implemented unified messaging solutions. But that will soon change, as technical difficulties are resolved and unified messaging's benefits become widely appreciated.

Resolving Integration Issues

Part of the problem involves legacy systems, or large investments in fax servers, voice mail systems, or e-mail packages which don't integrate well with each other. That's starting to change now, but there still are some integration issues. (For a description of a typical integration problem, see our review of Active Voice's TeLANophy in the CTI Labs section of this issue. We had more than a little difficulty hooking up TeLANophy to our Comdial DXP switch.) Once the hardware vendors start to make hardware that is less proprietary and that conforms to industry standards, these problems will happen less frequently.

Getting Attention

Despite its obvious benefits, unified messaging is not in the spotlight. No, at present, everyone is paying attention to applications that are related to the Internet. Everything from IVR-style Internet applications (which I refer to as IHR, or interactive hypertext response), to Internet telephony, to retrieving voice mail over the Web has all been getting tons of press in the CTI industry. Ironically, this fascination with things related to the Internet may start to work in unified messaging's favor. Why? We are now seeing the emergence of unified messaging applications that are based on Web browsers.

UNIFIED MESSAGING MEETS THE INTERNET

Unified messaging, by itself, is capable of transforming how you send and receive messages. But, in combination with the Internet, unified messaging will be unstoppable. Here are a few areas in which CTI-enhanced message handling and the Internet are combining their strengths:

PC-Based PBXs

Some of the new PC-based PBX vendors have demonstrated incredible inte-

How Unified Messaging Can Simplify Your Life

	Disparate Messaging Systems	Unified Messaging
Logon	Voice mail, e-mail, and fax systems are all separate, which means it takes extra time to logon to several systems. You might also have to remember multiple passwords.	You no longer need to logon to multiple systems to retrieve different types of messages. You logon once, with one password.
Interface	You're obliged to use a primitive telephone keypad for functions such as fast forward, rewind, save message, etc.	One word: GUI. Having all your messages accessible from a simple-to-use GUI makes unified messaging easier than using a keypad to retrieve voice mail.
Search Capabilities	Existing voice mail systems lack search capabilities: You must scroll through saved messages one at a time. Even if you have a fast forward or skip-to-next-message feature, the keypad is an inefficient interface if you're looking for a particular message.	You can visually scan all your voice messages, which can be ordered by date, time, or even a caller's name (which can be displayed in your inbox via a caller ID and database lookup).
Notification	Usually a blinking red light on your phone indicates new voice mail, but your phone will not tell you when the message was delivered or who sent it until you logon.	Immediate notification of voice messages and faxes, all with time and date stamps. Some systems can indicate the name and/or number of the caller.
Organization	Messages of different types are scattered across multiple systems. Even the most organized person has trouble keeping track of disparate faxes, e-mails, and voice mail messages. Prioritizing is difficult.	All voice mail, fax, and e-mail messages are in one central location. The ability to organize messages as new, old, or saved allows for better prioritization.

> **GUI interfaces giving you access to all your messages may become as ubiquitous as the phone, and managing all your messages will be as simple as handling your e-mail.**

gration between the Internet and their respective platforms. Altigen Communications (www.altigen.com) has come out with Altiserv, a system based on Microsoft Windows NT. Being based on NT allows the server to perform multiple server duties, including those of an Internet server. The coupling of these once disparate systems into one computer allows us to exploit new paradigms in how we handle our telephony system. For example, we can use a Web browser to configure the telephone system and make parameter modifications on-the-fly from remote locations. This PC PBX supports unified messaging over the Internet out of the box. No additional equipment is needed.

Forwarding Voice Mail

We take our ability to forward e-mail for granted. (That's because international standards, such as SMTP, POP, and MIME, impose some uniformity over how e-mail is sent, received, and encoded.) But forwarding a voice mail message outside your own office or into your office is another story.

Octel (www.octel.com) has tackled this problem with OcteLink, a product that allows disparate voice mail systems to send and receive voice mail from one another. Those of us that do not have a unified post office at the moment can use OcteLink to transfer messages to each other.

Exploiting RealAudio

RealAudio, already well known to diehard Net surfers, is beginning to enjoy business use. Early versions of RealAudio, a compact audio format, allowed you to listen to sound played in real-time over the Internet. (You can download a free RealAudio player from www.realaudio.com.)

Listeners used this form of RealAudio to tune into distant radio stations and play an untold number of records. But now we're seeing business applications such as MediaMail's Internet Message Center (www.mediamail.com). This application features a mixed-media Web browser interface that incorporates e-mail, voice mail, and fax. Since this product is browser-based, it can work with any standard supported by the latest browser technology.

CONCLUSION

When businesses realize how much money they can save by implementing unified messaging, we'll see applications of this type on every desktop. No doubt many of these applications will be Internet-based.

Consider the latest rave in the computer market: network computers (NCs). These are designed to provide low-cost access to the Internet and basic processing functionality. Web TV devices also promise to deliver low-cost Internet access. As these devices become more pervasive, it will be possible to retrieve all your messages from anywhere in the world. Easy Internet access can be set up on a pay-per-use basis in hotel lobbies, airports … wherever. GUI interfaces giving you access to all your messages may become as ubiquitous as the phone, and managing all your messages will be as simple as handling your e-mail.

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UNIFIED MESSAGING: THE NEXT KILLER APP?

BY ROBERT H. FRITZINGER, VOICE TECHNOLOGIES GROUP, INC.

A good friend of mine once supplied me with the best working definition for "killer app" I have ever heard. Since my friend is known to turn a colorful phrase on occasion, I cannot reproduce his comment unedited, but it goes something like this: "I'll know it when I see it."

My own experience is entirely consistent with my friend's view. I am old enough to have a closet full of old computers — mostly old Apple IIs, Macs, and early PCs. One in particular, the old IBM PC AT, is my favorite. It still has two programs installed on it that I consider to be prototypical killer applications: SideKick and 1-2-3. Both products proliferated very quickly after their initial release; both enjoyed critical acclaim and commercial success. Rapid commercial impact contributed to their cachet, but for me one thing that distinguished these products from others is how long I remained fascinated by their design and implementation and how long I continued to find new uses for them.

A lot has been written about the lack of killer applications in recent years. The initial appeal of software like SideKick was that it thoroughly exposed a capability of the PC that most were only dimly aware of, the Terminate and Stay Resident (TSR) program. The capabilities of our equipment and their operating systems are now so well documented that such surprises are no longer possible. Technical gimmicks no longer impress. It now takes sheer genius in design and the evolution of a whole new class of product to allow the "killer app" label to applied. Until recently, I harbored suspicions we would never see another again.

Until we installed unified messaging at our company, that is.

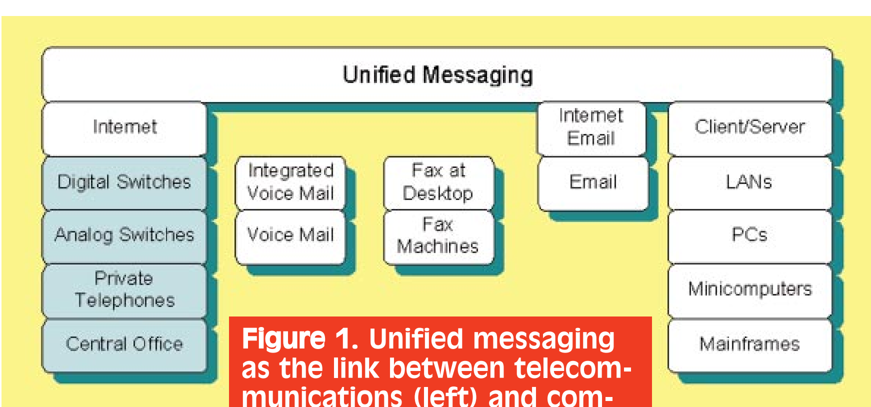

Figure 1. Unified messaging as the link between telecommunications (left) and computers (right).

HISTORICAL BACKGROUND

There is a temptation to call unified messaging the next generation of voice mail. But unified messaging is much more than that. In fact, the advent of unified messaging signals a basic change in the development and implementation of communications

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technology. Thus, if we are to understand what unified messaging is really all about, we need to place it in historical perspective.

Most companies own or lease infrastructure for two different networking capabilities: telecommunications and data communications. For millions of small companies, this is a key telephone system and a collection of personal computers on a local area network (LAN) sharing a printer resource. For some large businesses, sophisticated private phone networks and data Intranets on wide area networks (WANs) are required. For most businesses, these networks remain largely separate and completely heterogeneous even though they may be carried on common transport.

Today's telecommunications industry (Figure 1, left side of diagram) derives from a heritage of regulated monopolistic industry practice. The resulting mainframe design and closed architecture philosophy of PBXs are not surprising given this background. Nor is the relative lack of value-added capability.

The computer industry (Figure 1, right side of diagram) evolved more rapidly, but in a hotly competitive and unregulated business environment. It is no surprise, therefore, that technologies like mainframes, minicomputers, PCs, LANs, and client/server have evolved into successful marketplace solutions more rapidly than the modest generational changes made by the telephone systems manufacturers. It is also no surprise that advances in adjunct equipment for telephone networks, such as fax and voice mail, actually track technology advances in the computer industry.

With the widespread availability of the Internet and its rapid commercialization via the World Wide Web, the stage is set for an application capable of building a viable and workable bridge between both networks and their technologies. Unified messaging is the first application to assume this role, to serve as the impetus for network convergence.

THE NEXT KILLER APP

Unified messaging is unlike traditional voice mail in four distinct ways. For the purposes of this article, these points of distinction are designated

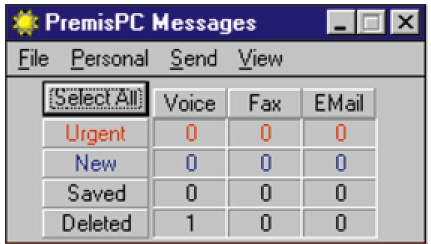

Figure 2. An example of how a unified messaging application can organize your messages.

"anything," "anytime," "anywhere," and "anyway" functionality. These differences don't just set unified messaging apart, they recommend it as the next killer app.

Anything

I receive a constant inflow of information, the majority of which comes to me via e-mail, fax, and voice mail. Infrastructure changes, with their associated hard and soft costs, have been required to insure reliable delivery of all types of communications to me. In particular, fax can consume considerable soft costs in a growing company since it must really be manipulated in much the same way as traditional mail, but with an added time imperative.

The obvious impact of unified messaging is that I, as a subscriber, can manage all of these communications consistently and continuously using a single software tool. For me, unified messaging paid for itself when I realized I no longer needed to trudge off to the photocopier to "forward" a fax message!

Don't view unified messaging simply as a convenience application for subscribers, however. View it as an enabler for your client base. We did not install unified messaging to make my life simpler. It was installed to permit our clients and vendors to communicate with us on their terms, in their own way, and according to their own needs, with no loss of responsiveness in our company. Remember that blinking light on your phone? It's a thing of the past. Screen-based displays, like the one in Figure 2, are the future.

Anytime

Traditional voice mail and e-mail have always permitted the subscriber the ability to retrieve and compose messages at any time. They have also permitted the caller to leave a message at any time. Fax has been problematic, but with its integration into the unified message store, that problem is now solved. You should look to your new unified messaging system for more, however.

Ours includes the obvious paging and message notification and delivery capabilities. Whenever a message is labeled urgent by the sender, the system undertakes a set of preprogrammed actions to deliver an alert to the appropriate recipient. Our system, PremisMail, also permits me to attach these notification capabilities to any message from a particular individual. You may be too modest to label your message as urgent, but I may want to be informed instantly!

This theme is important. Traditional messaging systems have left the sender substantially in control of the transaction. While the recipient has had tools to manage the message, he/she has had little power to manage the transaction itself. Unified messaging changes the rules because of the power of the PC interface and the ability the subscriber has to program his/her behavior. The caller is still king, but the subscriber now has a hand in managing the transaction to fulfillment.

Anywhere

In the past it took me three phone calls from a hotel room to check all of my messages: a call to voice mail, a call to my secretary, and a RAS session to check e-

Unified messaging is the first application to drive the convergence of telecommunications and data communications networks.

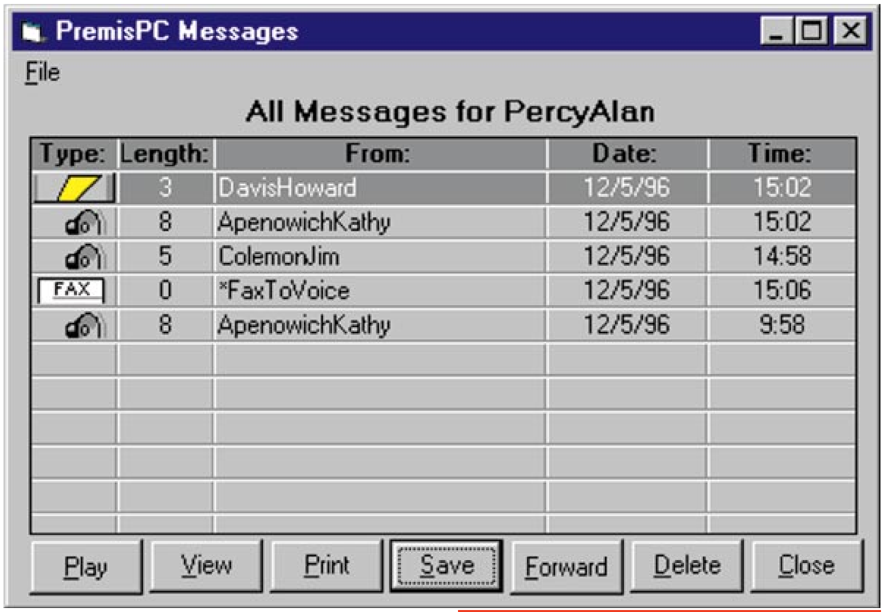

Figure 3. An example of how a unified messaging application can display information about your messages.

mail. It now takes one call, and that call can be from either a phone or a PC.

Much to my wife's dismay, PremisMail also includes MVIP-based switching capabilities and a sophisticated "Find Me" capability. I have the ability to log into the system and inform it of my location either via a touch-tone interaction or a remote access session. If Find Me is activated, the system will switch an incoming call to my temporary location, making it much less a messaging system and more a personal assistant. You may be willing to leave me a message, but I may want to talk with you!

Anyway

Much has been written about unified messaging's ability to make voice mail, e-mail, and fax available through either a telephone or PC transaction. With our new system, I can access my entire message store via telephone. Fax OCR and text-to-speech conversion technologies are used as required, and I am able to listen to my e-mail and my fax, including the entire body of a fax message. With a multimedia PC, I am able to view e-mail and fax images and listen to my voice mail. I can also cut text from a fax and paste it into a document for further processing. So, not only will the annoying blinking red light disappear, the greeting that says "you have one new message" will be replaced with a display like the one shown in Figure 3.

PC and modem manufacturers take notice: Encourage the development of this technology. Given the sophistication and capabilities of the PC, it quickly becomes the preferred terminal from which you process all messages. I have heard stories about unified messaging systems being installed simply because the PC interface permits nonsequential access to voice messages!

FUTURE OF MESSAGING

The future of unified messaging is bright. There are no significant hardware, software, or cost barriers prohibiting wide deployment of the application. Evolving standards for video, voice over Internet, and fax on Internet permit the inclusion of other media as well as the immediate use of the Internet as a vendor-independent "gateway" between unified messaging systems. Know my Internet address? Forward *any* content to me, and I will respond to it.

The evolution of single-line, board-level TAPI-based systems promises the delivery of a wave of small office/home office (SOHO) unified systems that will be tightly integrated with office systems. Imagine when your home PC can answer using the prompts of your office machine and take messages for you. When you RAS into the office, log into your workgroup software and synchronize message stores. All prompts will be updated, and all message folders will be updated, including the delivery of messages composed for you on the office machine.

Because unified messaging is based on standards such as MAPI, there is no barrier to the creation of multimedia messages. This opens up the exciting prospects: creating a spreadsheet with embedded video and audio clips, transmitting it to a community of users with common interests, and receiving feedback in various mediums.

Unified messaging is clearly the first in a new wave of applications that will be LAN-centric in design, philosophy, and implementation. It will not be a simple telephony system adjunct. The implications are numerous. Widespread support for this type of application by operating system vendors with technology like TAPI, TSAPI, and Java insures its quick development and adherence to commercially viable standards. Emerging trends forcing the convergence of networks into a single, packet-switched customer network supporting all data, voice, and video communications insures that this application will eventually reduce itself to software only.

There is a unified messaging system in your future. As cost constraints and technology barriers fall away, the ability to communicate, manage communications, and serve clients — while taking advantage of anything, anytime, anywhere, anyway functionality — will become the norm rather than the exception.

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Industry Leaders Band Together In IT Rapid Deployment Drive

VocalTec, Ltd. announced its founding membership in the Voice Over IP (VoIP) Forum, a cooperative coalition of Internet, telephony, hardware, software, and networking industry leaders, which is intended to create and drive rapid deployment of a set of standards for the compatibility of Internet telephony (IT) products. In addition to VocalTec, founding participants include Cisco Systems, Dialogic Corp., and other industry leaders. The 40 initial members have joined forces based on their shared interest in accelerating the growth of telephony applications over the Internet, corporate Intranets, and bridging to the regular or public switched network.

The VoIP Forum will work towards an implementation agreement that incorporates and extends existing standards and elements of the International Telecommunications Union H.323 interoperability standard, industry-accepted codecs, call management, and other issues. For more information, contact Simona Wilrich (VocalTec) at 201-768-9400.

Alcatel Telecom Allies With Data Logic For CTI Software

Alcatel Telecom has announced a strategic alliance with Data Logic Ltd., wherein Alcatel will supply and support Data Logic's Easyphone integrated call management software with its own 4400 PABX platform.

"The strategic alliance with Data Logic will allow Alcatel Telecom to offer customer applications, which take telephony from being a passive part of the business, to being at the very core," said Perry Burns, Director – Large Voice – of Alcatel Telecom.

Easyphone is a single integrated call management software product which provides solutions for inbound and outbound call handling, call blending, and campaign management.

Continued...

Dialogic Offers Computer Telephony Training And Certification Program For VARs

Dialogic Corp. announced what it is calling the first computer telephony training and certification program for value-added resellers (VARs). The program is open both to VARs already offering computer telephony solutions and to network-based/PC resellers looking to expand into new markets.

"A comprehensive VAR training program is an essential step toward accelerating adoption of computer telephony solutions," said Sam Liss, vice president of corporate marketing at Dialogic.

Dialogic CT training and certification programs will include several types of endorsements and levels of certification to meet the varied needs of VARs and their technical staff. The courses will combine formal lectures with hands-on lab exercises to provide the most up-to-date information about the market, products and services, CT application solution design, and implementation and deployment strategies. The classes will be taught by seasoned professionals with extensive industry experience.

For course registration and information, as well as an overview of the new certification program visit http://www.dialogic.com. A complete 1997 catalog will be available at the Web site. Alternatively, call Dialogic at 1-800-553-8004. For more information, contact Rosabel Tao at 201-993-3000 x6320.

Circle No. 500 on Reader Service Card

Alpha Technologies Introduces New Wall-Mount UPS System

Alpha Technologies has announced it is introducing Nexsys – a wall-mount uninterruptable power supply (UPS) designed for today's telecommunications applications. The product is designed to protect key systems and other mission-critical communications equipment from damage due to line noise, surges, spikes, and brownouts. Nexsys provides up to two hours of internal runtime, in the event of complete utility loss, and more than eight hours of backup from optional wall-mount battery packs. Thanks to the wall-mount enclosure, the UPS and battery packs can be installed above the floor and near the equipment that is to be protected. A front-cover design similar to that on many key systems provides visible compatibility, allowing Nexsys UPS systems to be installed in open plan areas.

Nexsys is a maintenance-free, 600VA UPS system with automatic self-testing and an intelligent RS-232 interface for standard UPS alarm communication. A test panel on the front of the unit provides access to system information such as battery condition, input voltage levels, and overload alarm information.

Nexsys supplies a stable 60 ± .1 Hz true sine wave output. The Nexsys system is designed to ensure complete blackout protection. It provides power conditioning with output regulation of ± 5 percent over an input voltage range of 92 to 132VAC (+12 to –23 percent). The design employs a high-performance filter to guard against common noise mode while meeting the requirements of IEEE 62.41. For more information, contact Eric Wentz at 360-647-2360.

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It is CTI enabled to take advantage of the Alcatel 4400's PBX/CSTA technology, and its client/server architecture employs industry standard open platforms to allow integration into existing corporate networks. The Alcatel 4400 is a multifunctional corporate PBX designed for organizations with requirements for 30 or more extensions. It also serves as a hub, multiplexer, wireless communications front end, multimedia communications integrator, and an application platform. For more information, contact Data Logic at bpatel@datlog.co.uk.

Apple And FORE Collaborate On Common API For High-Speed ATM

Apple Computer, Inc. announced a licensing agreement with FORE Systems to develop a common API for high-speed ATM technology. This agreement will enable FORE to develop an implementation of an ATM API, allowing applications to take advantage of native ATM services on both Macintosh and UNIX platforms.

The collaboration with FORE complements Apple's recent announcement of the Apple ATM Middleware, which is intended to help accelerate ATM solutions development for both adapter and software engineers. The ATM API is intended to be written to the X-Open XTI specification and would provide all the necessary interfaces, network, and OS integration that will be an integral part of the ATM Middleware.

There is an increasing need in the entertainment and broadcast industry to have access to high-speed, multiplatform communications for the digital studio market. Apple believes a new class of applications and peripherals could take advantage of the features of ATM, including speed, quality of service guarantees, and scalability without sacrificing connectivity to legacy networks, the Internet, and existing applications. For more information, contact Maureen O'Connell (Apple) at 408-862-6689.

Continued...

PairGain Rolls Out New Megabit Access Products

PairGain Technologies, Inc., announced additions to its megabit access product lines. The company believes this will offer telephone companies and other providers even more options to deliver high-bandwidth services to their customers over ordinary copper lines. Included among these additions is a line of Megabit Modem products.

The Megabit Modem 768 delivers up to 768kbps in both directions (symmetrically) and simultaneously, over a single copper telephone line. The Megabit Modem 768 Plus includes the same functionality, as well as a front-panel alphanumeric LCD display and control buttons. The display shows line margin and other status and modem parameters. The Megabit Modem NIC (network interface card) is an internal modem, which also delivers up to 768 kbps. According to PairGain, it is compatible with PCI-based personal computers, with drivers available for Windows 95, Windows NT, and many versions of UNIX. For more information, contact PairGain on the Internet at http://www.pairgain.com.

Circle No. 502 on Reader Service Card

Positron Launches SONET Transmultiplexer

Positron Fiber Systems announced its new SONET transmultiplexer, Positron MIST. According to Positron, this is the first product that provides a single, easy-to-use platform to connect SONET and asynchronous networks while maintaining full SONET end-to-end management. "MIST completely changes the cost dynamic for inter-carrier connections, by allowing carriers to significantly reduce the number of broadband cross-connects in their network, while enhancing end-to-end SONET visibility," said Andrew Knott, vice president of marketing and sales at Positron.

The company says that MIST allows any VT1.5 granular EC1 connection from a SONET network to be presented as an M13-compatible DS3 interface. For customer access applications, MIST can demultiplex an EC1 tributary to provide 28 fully protected DS1 terminations. According to Positron, list price for MIST starts at $11,000. The product is to be available for delivery in the second quarter of 1997. MIST has two packaging options: compact standalone or high-density, rack-mount enclosures. For more information, contact Andrew Knott at 609-222-1288 or Ilene Wiener (PR@Vantage for Positron) at 201-461-1400.

Circle No. 503 on Reader Service Card

MITEL Personal Assistant To Support USB

Mitel Corp. introduced the MITEL Personal Assistant, a telephony peripheral that supports the Universal Serial Bus (USB). "Mitel is dedicated to bringing telecommunications and computer technology together to solve desktop communications roadblocks," said Howard Tweddle, Head of DeskTop Interfaces, Mitel Corp. "With support for USB, MITEL Personal Assistant is the first computer peripheral to deliver the full power of CTI to users in the home and small office." USB is suited for computer telephony integration because of its support for isochronous data communications, which is necessary for support of real-time synchronous applications such as live voice and video transmission.

Mitel believes that by supporting USB, the MITEL Personal Assistant delivers true telecommunications plug and play. Jim Pappas, USB program manager at Intel Corp. says, "USB makes PC peripherals easier to use than ever before by delivering 'outside the box' plug-and-play capabilities. Mitel is an avid supporter of the USB specification, and as the first USB-enabled telephony peripheral, MITEL Personal Assistant is just the beginning for USB products of all kinds." For more information, contact Ann Hatchell (media relations) or Ian Chadsey (investor relations) at 613-592-2122.

Circle No. 504 on Reader Service Card

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Genesys And NCR Sign Global VAR Agreement

Genesys Telecommunications Laboratories, Inc., and NCR announced that they have entered into a global VAR relationship. Under the terms of the agreement, NCR will offer the full suite of Genesys Computer Telephony Integration products throughout the world. The agreement also establishes NCR as the first distributor of Genesys software in Japan. NCR will market Genesys solutions to financial, communications, and retail industries, where both companies have an established presence. In addition, NCR is planning to market Genesys solutions to its data warehouse customer base.

"Because NCR is committed to helping customers in all computing environments, as is Genesys, working together is a natural fit. Under this agreement, we are bringing a powerful solution set to the financial, retail, and telecommunications marketplace, offering total integrated solutions to customers around the world," said George Geros, vice president, channel sales for Genesys. "Genesys looks forward to a long-term relationship with NCR."

For more information, contact Karine Hagen (Genesys) at 415-437-1163 or Susana Thompson (NCR) at 506-526-1601.

IBM And Nabnasset To Offer Integration Of Voice And Data Systems

IBM announced that Nabnasset Corporation has become an IBM Midrange Business Partner, Industry Remarketer Affiliate. As a result of the partnership, IBM and Nabnasset will deliver integrated call center applications supporting a wide range of PBX and ACD switch environments. Nabnasset will offer its CTI software, VESP (Voice Enhanced Services Platform), on IBM's CallPath server, thus increasing the number of companies that can implement the VESP software.

CallPath is a voice and data solution which integrates telephone systems and computer systems to facili-

Continued...

Dialogic And SCO Team Up To Announce CT-Connect Open CTI Server For UNIX Systems

Dialogic Corporation and SCO announced that they are working together to develop, market, and support Dialogic's computer-telephony server software, CT-Connect, on SCO UNIX servers. According to the announcement, the software will be the first open standards-based CTI server for the UNIX operating system. "CT-Connect has been around for a bit, but now there's a fully native UNIX version being built, available second quarter. This is the very first fully UNIX CTI capability, and both Dialogic and SCO are proud to bring that to the ISV community," said Carl Strathmeyer, Director of Marketing, Computer Telephone Division at Dialogic.

Although Dialogic and SCO have been working together in the CTI marketplace for some time, this is the first formal agreement between the two companies.

The SCO UNIX system-based version will support the Microsoft Telephony API (TAPI) and Dynamic Data Exchange (DDE) interfaces at initial release, and subsequently will support the Novell Telephony Services API (TSAPI) interface. CT-Connect for SCO UNIX will support a wide range of telephone systems, including the proprietary link protocols used by the Lucent Technologies DEFINITY PBX and the Nortel Meridian 1. The software will likewise support the international standard CSTA link protocol used by Alcatel, BBS Telecom, Bosch Telenorma, Cortelco, Ericsson, Intecom, Mitel, Rockwell, Siemens Business Communication Systems, Tadiran, and many other switches.

SCO's UNIX system operating environments are being widely utilized in a vareity of CTI solutions. In addition, there are many UNIX system-based applications, that can be enhanced by adding CTI capabilities.

Among the enhancements offered by CT-Connect is call control. This is the ability to direct calls with computer control or to find out more information about arriving calls, and act on them accordingly.

For example, a person makes a call to a bank to find out about their balance and is not satisfied with the automated voice-response system and chooses to speak with an operator. Rather than giving all the information to the voice-response system, then switching to a human operator and recounting all the information that was given to the automated system, call control will be able to forward all the pertinent information to that operator, and on down the line if it becomes necessary to switch between representatives. Mr. Strathmeyer likens it to a "third party, outside looking in on a call — the telephone operator's perspective on a call."

CT-Connect consists of a central server module and a set of distributable client modules. While Dialogic has offered CT-Connect client modules for a range of operating systems in the past, a UNIX system-based server has not been previously available. The SCO UNIX system-based CT-Connect server will be available in four configurations:

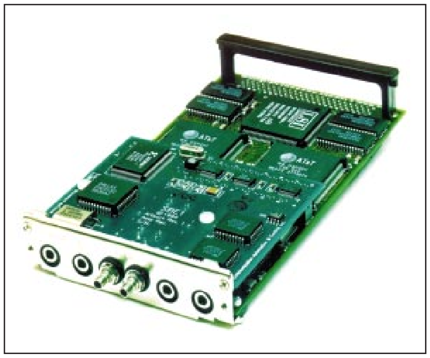
1) a Full configuration, supporting all types of client modules and an unlimited number of client systems; a Desktop configuration, supporting an unlimited number of client systems but restricted to desktop applications; Desktop Lite configuration, supporting up to 36 client systems running desktop applications; and an Evaluation configuration, licensed free of charge, supporting up to 16 client systems running desktop applications.

CT-Connect has no usage-based license charges, and all client modules are included with each CT-Connect server at no additional charge. For more information, contact Bev Rindfleisch (Dialogic) at 408-432-3346 or Monika Laud (SCO) at 408-427-7421.

Circle No. 505 on Reader Service Card

CAC Introduces 75-MFLOPS S-Bus Telephony Board

Communication Automation & Control, Inc. announced the SB32C, an S-Bus telephony board available with either a T1 or E1 interface. The board, equipped with two Lucent Technologies DSP32C digital signal processors, provides four CD-quality audio input/output channels and up to 1 MB of zero-wait-state SRAM. The SB32C interfaces directly with T1 and E1 lines and can transmit and receive in full duplex mode. Control for the framer chip is mapped into the DSP's memory space. T1 and E1 data are transmitted via the DSP's serial port.



The SB32C is available with either one or two DSP32C processors. The single-processor version features a peak performance of 37 MFLOPS, provides one stereo codec, and comes with 512 KB of private SRAM. The dual-processor version features a peak performance of 74 MFLOPS, provides two stereo codecs, and comes with 1 MB of zero-wait-state private SRAM (512 KB per DSP). The two DSPs communicate via an FIFO mapped into each DSP's memory space. One SB32C with a single 50-MHz DSP32C and a T1 interface costs $2,750 and is available now. For more information, contact Jim Bridges at 800-367-6735.

Circle No. 506 on Reader Service Card

Commetrex And Xionics To Join MultiFax And IPS

Commetrex Corp. and Xionics Document Technologies, Inc. announced that Commetrex Group 3 fax technology, MultiFax, will be incorporated into Xionics' Intelligent Peripheral System (IPS). IPS is a combination of peripheral controller software and silicon, which supports fax, copy, scan, and print functions. IPS is designed to reduce time-to-market and development costs and to provide a scalable solution that enables OEMs to amortize costs across multiple product lines.

"Until now, MultiFax was used exclusively by computer telephony OEMs," said Mike Coffee, President of Commetrex. "Xionics' adoption of Commetrex's MultiFax technology marks its acceptance as a key element in mainstream computer peripherals and office equipment destined for worldwide distribution." For more information, contact Martin S. Lippman (Commetrex) at 770-449-7775 x370 or Suzanne Dumaresq (Xionics) at 617-229-4142.

Circle No. 507 on Reader Service Card

Commetrex Cuts Multiline Fax Price In Half

Commetrex Corporation announced a major price break in the cost of multiline fax boards. According to Commetrex, the company is now offering an integrated fax board with per-port pricing roughly half the price of competitive boards. The 8-line board with loop-start line interfaces is $3,100; the integrated T1 board is $8,400. Both boards offer an MVIP connector and may be interconnected with other MVIP-compatible PC add-in boards. Fax systems of up to 96 ports per PC can be created.

"It has always been our belief that the use of fax in computer telephony systems was limited by the high per-port prices of multiline fax system resources, and that Commetrex could dramatically lower the price of multiline fax systems, while providing developers with state-of-the-art technology and comprehensive tools for them to build sophisticated, reliable, and reasonably priced products for their customers," said Mike Coffee, President of Commetrex. "The introduction of these boards is the first step in that direction. We intend to continue this initiative with additional products for voice, fax, data, and other media in the future." For more information, contact Martin S. Lippman at 770-449-7775 x370.

Circle No. 508 on Reader Service Card

Coresoft Introduces CenterPoint Communications Manager

Coresoft Technologies, Inc. introduced CenterPoint, a software product that is being billed by the company as a Windows-based, integrated client/server communications manager. The product was designed to integrate and manage devices, services, and systems such as voice mail, e-mail, fax, Internet access, cellular phones, pagers, and contact managers, and combine them into one user-friendly environment. According to Coresoft, the CenterPoint software:

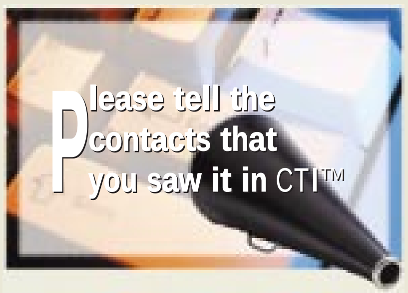
- simplifies conference calling by providing an easy click, drag, and drop option for conferencing or transferring calls;
- provides drag and drop broadcast fax or e-mail capabilities;
- stores all messages (voice, fax, e-mail) in one location, which can be accessed by computer or any telephone;
- offers callers choices, in addition to whom they would like to contact, such as accessing prerecorded informational messages and fax-back documents.

At the core of the CenterPoint software lies what the company calls the products most outstanding features: an Event Broker with Rules Engine and an IVR Builder. These features will allow VARs to customize the interactions among all of the end user's messaging components, thus providing the flexibility to configure systems to the specific needs of the end users.

Coresoft says CenterPoint can be integrated with many common business tools currently in use, such as MAPI-based e-mail systems, Internet browsers (Netscape Navigator or Microsoft Internet Explorer), contact managers (Symantec Act!), or voice mail systems (CallWare). In a separate announcement, Coresoft made public its alliance with Dialogic Corp. The partnership will focus on joint product development, integrating CenterPoint with Dialogic hardware, and educating VAR channels about unified messaging and CT opportunities. For more information, contact Brad Thatcher (Coltrin & Associates for Coresoft) at 212-221-1616.

Circle No. 509 on Reader Service Card

tate the exchange of information between PBX or ACD products and the VESP system. Currently, more than 20 telephone switches can integrate with the VESP/CallPath solution. These include: Alcatel 4400, Aspect via CSA Gateway, AT&T DEFINITY, Ericsson MD-110 via ApplicationLink, Northern Telecom DMS-100, Northern Telecom Meridian 1 Communications System, Philips Soppho and Octopus, ROLM 9751 Computerized Branch Exchange (CBX), and Siemens Hicom-300. For more information, contact Larry Edwards (IBM) at 919-254-7773 or Susan Cohen (Nabnasset) at 508-787-2851.



Ingram Micro Agreements With Active Voice And Siemens

Ingram Micro, Inc. announced that it has signed a letter of intent with Active Voice Corp. to distribute its CTI and voice-processing hardware and software products throughout the United States. "We are very excited that Active Voice, a pioneer in CTI and voice processing, will join Ingram Micro's roster of CTI solution providers," said Laura Skinner, vice president and general manager of Ingram Micro's Telecom Integration Division.

Under the terms of the prospective agreement, Ingram Micro will distribute Active Voice's entire line of CTI products: the TeLANophy suite of desktop unified messaging and call management applications, including ViewMail, ViewFax, ViewCall, and E-Mail Integration, as well as the Repartee, Replay, and Replay Plus voice processing systems.

Ingram Micro, together with Siemens Business Communications, Inc., further announced a reseller

Continued...

Pacific Telephony Design Unveils Search Engine To Locate Used Telecomm Equipment and Parts

Pacific Telephony Design unveiled a new Internet search engine, which enables visitors to easily locate used telecommunications equipment and accessories. The search engine enables buyers and sellers to place free online advertisements for used telecommunications gear and accessories such as PBX and voice mail systems. Visitors can search for equipment by category, keyword, and price range. Internet advertisements can be placed free of charge. Visitors can browse and place advertisements by going to the URL (http://www.phonezone.com/swapmeet). For more information, contact Brian McConnell at 415-647-9911 x201.

Circle No. 510 on Reader Service Card

Ariel Announces High-Density, All-Digital Modem Boards

Ariel Corp. announced two products in a line of high-density, all-digital modem boards, T1-Modem and CTI-Modem. Featuring up to 24 ports at a cost of less than $500 per port, Ariel claims the two boards pack two to four times the functional density of their nearest competitors, at a significantly lower price per port.

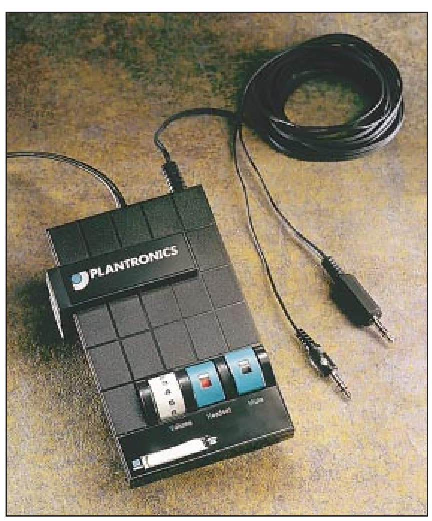
T1-Modem is a high-density modem board that provides either twelve or twenty-four 28.8 kbit/sec v.34 modems in a single ISA bus slot. CTI-Modem is a soft (programmable) DSP-based modem board providing up to sixteen v.32ter modems in a single slot. Both boards are equipped with MVIP and SCSA computer telephony interfaces, which enable them to be connected to T1, ISDN, and analog line cards. The products target data-intensive applications requiring large numbers of modems. Some of these include transaction processing (credit card purchases, order entry, record retrieval), on-line services (travel agencies, home shopping), Internet service providers, PBX adjuncts, and Teletext.

T1-Modem costs $13,500 in quantity, and the cost of CTI-Modem is $10,000 in quantity, according to Ariel. For more information, contact Steve Curtin at 609-860-2900.

Circle No. 511 on Reader Service Card

Plantronics Offers Integrated Headset Adapter

Plantronics, Inc., has introduced the Headset Switcher adapter (Switcher), which is an integrated computer and telephone headset adapter. According to Plantronics, with the Switcher, users can utilize a single Plantronics headset to easily transition from a telephone call to listening to or recording computer voice annotations, performing speech recognition applications, listening to music, or any number of other computer and audio applications.



The Switcher is compatible with many desktop telephones. In computer mode, it supplies a self-powered microphone (to provide compatibility with PCs and sound cards) and a cleaner signal (for speech-recognition applications). The Switcher connects to many audio devices with 3.5-mm ports for microphone and headphone connections to support personal conferencing, Internet telephony, speech recognition, and other applications. The product will have a suggested retail price of $149.95, and Plantronics believes it will be available in early 1997. For more information, contact Joyce Shimizu at 408-458-4484.

Circle No. 512 on Reader Service Card

TTC Offers TIMS-45 Analog Option For FIRE-BERD Test Instruments

Telecommunications Techniques Corp. announced the addition of a TIMS-45 analog option to the FIREBERD 6000A and the FIREBERD 4000. The TIMS-45 option enables the user to implement Transmission Impairment Measurement Set (TIMS) tests on both Voice Frequency (VF) and Wideband analog lines. Using new E and F filters, the user is able to qualify High Bit-Rate Digital Subscriber Line (HDSL), Digital Data Services (DDS), and Integrated Services Digital Network (ISDN) circuits.

The TIMS-45 mounts on the FIRE-BERD. In addition, TRT is offering a lid option, which contains a screen to view test set-up and results at a glance as well as a keypad. The option includes an external power adapter, which allows the lid to be operated separately from the FIREBERD 6000A/4000. The TIMS-45 Analog Lid option costs $3995. For more information, contact Rob Clark at 301-353-1560 x2184.

Circle No. 513 on Reader Service Card

Dialogic Ships PureSpeech Speech-Recognition Software

Dialogic announced it is shipping ReCite! speech recognition software from PureSpeech. The package runs on the Antares open digital signal processor (DSP) platform, under the SCO UNIX operating system. ReCite! offers advanced speech recognition, and relieves end-user customers of the burden created by limited vocabularies or awkward pauses, allowing them to speak naturally.

The ReCite!/Antares combination allows developers to create call processing, automatic speech recognition (ASR)-based speech applications for telephony systems. The product enables continuous speech, large vocabulary applications (up to 2,000 words), speaker-independent recognition, and real-time vocabulary generation. The PureSpeech ReCite! SCO UNIX version is currently available, and Dialogic believes that a Windows NT version will be available in early 1997. For more information, contact Amy Limb (PureSpeech) at 617-441-0000.

Circle No. 514 on Reader Service Card

AVA Systems To Support AMIS Networking And Overhead Paging

AVA Technology announced the latest release of software for its AVA-100, AVA-200, and AVA-200 Plus Voice/Fax Processing Systems. Version 4.30 comes with several new features, including support for AMIS (Audio Messaging Interchange Specification) Networking, Overhead Paging, and a Quick Install/Setup Utility.

AMIS support allows AVA users at remote office locations with completely separate AVA systems to communicate with each other via their own mailbox. AMIS support also allows users to exchange messages with other companies that have an AMIS-compatible voice mail system. Overhead Paging provides callers with the option of paging the user at the requested extension after unsuccessful transfers. The Quick Setup Utility is aimed at installers and technicians. When an AVA system is powered up for the first time, it will alert the installer that it has not been programmed. Based on a series of simple questions and responses, the system will automatically program itself for the phone system and integration method required. For more information, contact Karen Clough at 1-800-488-8840.

Circle No. 515 on Reader Service Card

TsDesign Introduces Milborne

TsDesign, Inc. announced Milborne, which converts SS#7 (ANSI/ETSI/CCITT/Country Variants) to TCP/IP or, conversely, TCP/IP to SS#7. TsDesign states that not only does Milborne support Call Handling (ISUP/TUP/BTNUP), but supports the Transaction Parts (SCCP and TCAP) as well. The Milborne can also be purchased to support MAP SMS (Mobile Application Part Short Message Service), which the company says will provide access to the GSM (Global System for Mobile Communications) network. TsDesign provides Milborne as either a turnkey solution or as cards and software. For more information, contact Dawn Billet at 404-238-0528.

Circle No. 536 on Reader Service Card

Toshiba Unveils 155Mbps SONET/SDH Framer IC

Toshiba America Electronic Components, Inc. announced a 155-Mbps framer chip for asynchronous transfer mode (ATM) systems, with volume shipments to begin in the first quarter of 1997. The 155-Mbps framer, designated the TC35821F, was developed for network interface card, switch, and router applications. According to Toshiba, the device supports SONET standards STS-3C, STS-1, STS-1 half, and SDH at STM-1.

The single 155-Mbps SONET/SDH framer is housed in a 144-pin plastic quad flat pack. Samples of the TC35821F are available now. Toshiba says the framer is priced at $35 each, in quantities of 1,000 pieces. For more information, contact Annette Birkett at 800-879-4963.

Circle No. 537 on Reader Service Card

Versatility Offers Direct Banking Call Center Solutions

Versatility, Inc. demonstrated its direct banking call center software solutions at Retail Delivery Systems ’96. The Versatility Series is a suite

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distribution agreement under which Ingram Micro will distribute Siemens' OfficePoint system to VARs throughout the United States and Canada. OfficePoint is an ISDN system designed for small offices.

For more information, contact Kirsten Frosh (Ingram Micro) at 714-566-1000 x2727 or Anita Giani (Siemens) at 408-777-4546.

Unimax And Supply Technology Announce Partnership

Unimax Systems Corp. and Supply Technology, Inc. have announced a distribution agreement which will add Unimax's configuration management software to the suite of products Supply Technology supplies to its customers.

According to Unimax CEO Fred Sweeny, "Our partnership will provide Supply Technology customers with software tools to save time and money in the administration of their PBX and voice mail systems, through an easy-to-use GUI on a PC."

Supply Technology principal Beth Morley adds, "As national remarketers of telecommunication equipment, we add value by providing our customers with products that help them control their total telecom costs. The Unimax products allow us to give them a powerful new tool to eliminate programming hours and travel time on one of their most costly departmental activities — moves, adds, and changes."

Unimax configuration management products also support fault management and provide updating of multiple telephony applications, such as call accounting, corporate directory, and E911 systems. For more information, contact Jan Carron (Unimax) at 800-886-0390 or Beth Morley (Supply Technology) at 714-646-7292.

Lucent Extends PassageWay Telephony Services, Ships Windows NT Version

Lucent Technologies announced the general availability of its PassageWay Telephony Services for Microsoft Windows NT, in response to customer requests. PassageWay Telephony Services, available on Novell's NetWare since 1994, provides customers a common open platform based on the TSAPI interface, for building computer telephony solutions.

Using the TSAPI interface supported by many PBX vendors, CTI applications written for PassageWay Telephony Services for NetWare run unchanged on the NT version, allowing customers to grow their corporate networks with both platforms. PassageWay Telephony Services supports a wide variety of clients: Microsoft Windows 3.x, Windows 95, Windows NT, OS/2, Macintosh, UnixWare, and HP-UX client workstations running TCP/IP to either a NetWare or Windows NT server. For more information, contact Ryerson Schwark at 908-953-7528.

Circle No. 516 on Reader Service Card

National Semiconductor Ships First Microsoft Windows TAPI Service Provider Software Targeted For Local Area Networks

National Semiconductor Corp. announced that it is now shipping a Telephony Application Programming Interface (TAPI) Service Provider software for Local Area Network connected PCs. The software is integrated with the company's isoEthernet ISA adapter cards.

According to National Semiconductor, the software enables real-time interactive multimedia applications and provides a platform for all current and future TAPI-enabled applications for Windows 3.1, Windows for Workgroups, and Windows 95 systems. Furthermore, the product establishes "phone call connections" for real-time voice, video, data conferencing, and telephony processing applications from the LAN-connected PC. This call control or "signaling" software runs entirely on the PC's processor and interfaces to the TAPI service provider interface. This allows existing TAPI applications to interface to networking hardware such as isoEthernet and other isochronous communications protocols such as ISDN and ATM. For more information, contact National's Customer Response Group at 1-800-272-9959.

Circle No. 517 on Reader Service Card

NEC And Cintech Release JAZZ2000 ACD Software For Call Centers

NEC America and Cintech Tele-Management Systems, Inc. announced the commercial availability of JAZZ2000, a PC software application designed by Cintech exclusively for NEC's NEAX 2000 IVS PBX and offered only through the NEC America authorized distribution channel. The JAZZ2000 ACD product customizes the system to achieve a variety of sophistication levels.

"The development of JAZZ2000 is significant because it allows us to deliver an advanced ACD application for the exploding call center market, while complementing the power and versatility of our NEAX 2000 IVS PBX system for the under 500 phone lines market," said Bob Talty, marketing vice president for NEC America Corporate Networks Group.

System variables include decisions on the number of phone lines needed, in addition to announcement, overflow, reporting, and management requirements. The application will handle a maximum of 80 active agents, and 24 agent groups, as well as the NEAX 2000 IVS system, offering expandability of up to 512 ports. For more information, contact Jim Dundon (NEC) at 972-518-4945 or Julie Hopkins (Cintech) at 513-731-6000.

Circle No. 518 on Reader Service Card

Natural MicroSystems Expands Wireless Telephony Offerings

To further integrate the core SS7 technology gained from its recent acquisition of TEKnique, Inc., Natural MicroSystems has released an enhanced version of its communications processor, the TX2000. This latest version of the TX2000 adds capabilities including a dual-T1 interface, which enables the passing of both voice and data over a T1 line. The new enhancement provides a digital connection for SS7 and voice.

Allen Carney, vice president of Marketing and Business Development at Natural MicroSystems said, "We have moved very quickly to integrate TEKnique's core technology into our product line. The combination of NMS's exceptional trunking, programmable switching, and voice encoding/decoding technology with our new SS7 technology offers developers an excellent solution as they build systems for the IN and wireless networks." The TX2000 has a suggested list price of $2,495 in a basic configuration, with interface options and software development kits priced separately. The TX2000 and the new T1 interface are available immediately. For more information, contact Patrick Fetterman at 508-650-1372.

Circle No. 519 on Reader Service Card

Nortel Introduces Norstar-PLUS Compact ICS

Northern Telecom unveiled its new Norstar-PLUS digital key system that brings high-end business communication capabilities to businesses that need up to 8 outside lines and 24 telephones. The Integrated Communications System (ICS) offers over 100 features, including an integrated automated attendant that allows calls to be answered and routed to the right person or group more efficiently.

According to Johnnie Holt, Nortel's assistant vice president, Norstar, "Many Norstar customers with smaller capacity requirements need many of the most sophisticated applications to compete in today's business climate. Based on the reaction we have received from this market segment, the Norstar-PLUS Compact ICS will meet those needs."

Applications that can be added to the Norstar-PLUS Compact ICS include voice mail and fax messaging capabilities. The system will soon support Microsoft's TAPI and Novell's TSAPI interfaces. The Norstar-PLUS Compact ICS is currently available from Nortel's Authorized Norstar Distributors. For more information, contact Brian D. Murphy at 214-684-8589.

Circle No. 520 on Reader Service Card

RADVision Unveils Standards-Compliant H.323/H.320 Gateway

RADVision announced the L2W-323, a fully standards-compliant H.323/H.320 gateway, which will enable H.323 users to communicate with any remote station, by networking LAN or Intranet-based H.323 terminals with H.320 terminals connected to ISDN or other switched digital networks.

The L2W-323 is a stand alone unit that connects up to four concurrent calls between IP-based H.323 codecs and remote users connected to the switched digital networks, at speeds up to 384 kbps per call. Multiple gateways can be installed in parallel to support higher call volume. The WAN ports of the L2W-323 can be configured as ISDN/BRI or V.35 for higher speeds, while the LAN ports include Ethernet and Token Ring. According to RADVision, other network types will be added.

Current users of RADVision's L2W-20 LAN/WAN, which is used to connect H.320 systems across local and wide-area networks, will be able to migrate to the L2W-323 by downloading a new software version. RADVision expects to release the L2W-323 in the first quarter of 1997. Prices will start at $5,950. For more information, contact Ilene Wiener at 201-461-1400.

Circle No. 521 on Reader Service Card

of CTI-enabled, open, client/server-based applications that the company says can support all aspects of customer interaction transactions. "Versatility understands the customer relationship management goals of the retail banking industry, and has developed software solutions that leverage the Windows NT operating environment," said Matt Capoccia, VP of products and strategy at Versatility. "This allows banks to provide their customers the flexibility and convenience of banking by telephone any time from anywhere." Versatility supports Microsoft Windows 3.1, Windows 95, and Windows NT. For more information, contact Lana Sansur at 703-934-7631.

Circle No. 538 on Reader Service Card

Madge Networks Shows Multiservice Network Technologies

Madge Networks demonstrated its latest networking technologies and products at Networld+Interop, Paris. The demonstrated products and technologies are all components of the MadgeOne architecture. These included video networking, voice and data integration, ATM, Ethernet and Token Ring switching, layer 3 switching, and ATM switched network management.

Michael Beadsmoore, Madge's VP of Marketing for EMEA said, "The MadgeOne architecture introduces the concept of multiservice networking, allowing applications to combine voice, video, and data across a single networking infrastructure." For more information contact Pascal Ozanne at +33 (1) 41-92-52-20 or via e-mail at pozanne@madge.com.

Circle No. 539 on Reader Service Card

Securicor Unveils I3000 At Voice 96

Securicor Telecoms Ltd. showed its Interconnect 3000 (I3000) PBX at Voice 96 together with Novell Authorized Reseller du Pre Telecoms. According to Securicor, the I3000 is the world's first PBX to be awarded the Novell Yes certification. The company says the certification,

Continued on page 29

US ONE Communications Allies With IEX

US ONE Communications announced that it has teamed up with IEX to implement an enhanced services platform for its nationwide telecommunications network. US ONE has equipped its network with IEX Nexus SCPs, intelligent network platforms that act as database servers to provide network control functions and enhanced services to its carrier customers. The Nexus SCP platforms will allow US ONE to offer a broad menu of services, including long-distance dialing for businesses, account/billing code authorization, 800 services, and call reorigination.

John H. Jacquay, president of US ONE's Carrier Services Division, points out, "The swift implementation of state-of-the-art technology and on-site maintenance and support facilitated by IEX positions US ONE to provide an unsurpassed variety of advanced services to our customers. This strategy gives US ONE carrier customers a true, high-tech advantage in the marketplace." Gary Crockett, president of IEX Corp., says, "We are pleased to forge a strategic relationship with a leading firm like US ONE who is poised to take advantage of telecommunications deregulation by providing the sophisticated technology needed to launch major, nationwide telecommunications networks."

For more information, contact Stan Jasinski (IEX) at 972-301-1300 or Christine Forbes McDermott (US ONE) at 602-816-1004. ■

Ariel Releases RASsoft Remote Access Software

Ariel Corp. announced RASsoft, a turnkey software solution for building remote access servers. According to Ariel, RASsoft gives systems integrators all of the software components needed to build open architecture servers providing remote access to the Internet, corporate Intranets, and online services using off-the-shelf hardware components. Servers utilizing RASsoft can support up to 240 sessions per shelf, and over 1,000 sessions in a single rack.

The new software solution supports both V.34+ (33.6kbps) and ISDN subscribers, and provides all the call control, Point-to-Point Protocol (PPP), RADIUS authentication and billing, SNMP agent, and administration features needed for a complete remote access server offering. RASsoft is based on open UnixWare architecture. RASsoft's HTML-based administration software enables system administrators to configure and monitor the system with a Web browser such as Netscape Navigator or Microsoft Internet Explorer. The graphical interface provides easy-to-use screens for both configuration and administration. For more information, contact Steve Curtin (Ariel) at 609-860-2900.

Circle No. 522 on Reader Service Card

Copia International Introduces FFWEB

FFWEB, a new application from Copia International, retrieves Web documents and translates them to a faxable format. Documents on a Web site can now be downloaded to fax machines. When used in conjunction with FaxFacts Fax-on-Demand system, FFWEB makes Web pages available to customers with fax machines, but no Internet access.

A search engine retrieves Web pages from the Internet and converts them to a faxable format. FFWEB automatically builds a database of document locations that can be cross-referenced to document numbers keyed at the telephone keypad by callers. When a caller chooses a document number, FFWEB cross-references it to a URL site, converts the document to fax format, and delivers it to the keyed fax number. Documents are automatically paginated, and colors are replaced with grays and black. If a list of document numbers is offered as a menu choice on the Fax-on-Demand system, callers can easily locate the documents they need. For more information, contact Dorothy Gaden-Flanagan at 630-682-8898.

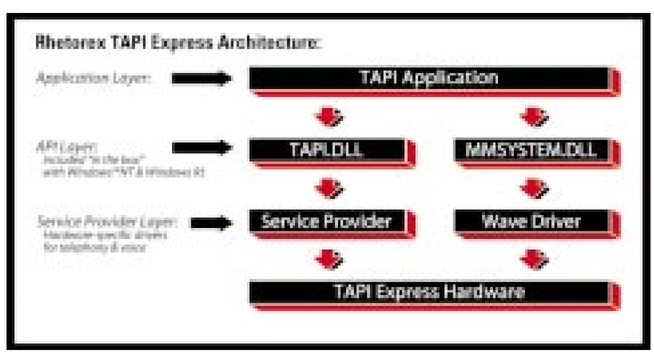
Circle No. 523 on Reader Service Card

Rhetorex Unveils TAPI Express For Client/Server Unified Messaging

Rhetorex, Inc. announced the TAPI Express computer telephony (CT) board, specifically designed for TAPI-based client/server unified messaging applications. Built on the Rhetorex RealCT platform, the TAPI Express board resides in the server with no requirements for additional TAPI hardware in client PCs.

"With TAPI Express, we are taking a leadership position in the delivery of unified messaging solutions today," said Kenneth Lavine, president and CEO of Rhetorex. "Rhetorex has made it easier for VARs and SIs to implement robust client/server unified messaging applications by providing a complete hardware solution that is open, easy to install, and cost-effective."

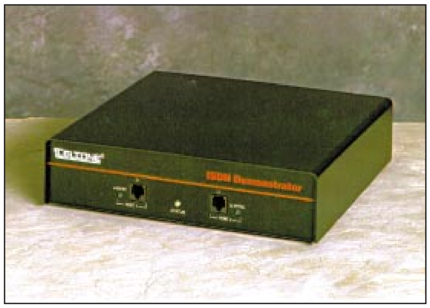
According to the company, TAPI Express for TAPI 2.0 will be available in the first quarter of 1997. A version for TAPI 1.4 (Windows 95) is available now. Pricing starts at $995 for a single board.



For more information, contact Stephen Weekly at 408-370-0881 x1110.

Circle No. 524 on Reader Service Card

Teltone Introduces ISDN Demonstrator



Teltone Corporation has announced the ISDN Demonstrator, which offers an easy and inexpensive way to demonstrate ISDN CPE equipment, without requiring an ISDN service connection. The product is well-suited for trade shows or on-site customer presentations.

Primary features of the ISDN Demonstrator include: two ISDN-BRI (2B+D) lines for end-to-end simulation; single U (2B1Q) interface at each line; speech; support for National ISDN-1 switch provisioning (AT&T 5ESS is optional); Windows-based configuration software; 3.1 kHz audio, 7 kHz audio, circuit mode data, and D channel packet mode data; point-to-multipoint configurations; in-band tones, and primary line power to six terminating devices. The ISDN Demonstrator uses 115 VAC 50/60 Hz. Teltone offers the product at a list price of $1,495. For more information, contact Tom Trexler or Mike Balch at 800-426-3926 or 206-487-1515.

Circle No. 525 on Reader Service Card

TRT Launches CommPoint For Windows NT 4.0

Telephone Response Technologies, Inc. announced their next-generation client/server CT product, CommPoint. TRT notes the product is designed to take advantage of the many benefits of Windows NT 4.0. The first of CommPoint's modules to be released includes CP/Mail, a voice mail/auto-attendant module; CP/Script, a customization script language; and CP/Query, an SQL database access module. Another module, CP/Toolkit, will provide a low-level API for Microsoft Visual C++ development of 32-bit Windows NT-based CT applications, and TRT expects that module to ship in early 1997.

The first release of CommPoint will support digital T-1, ISDN PRI, and analog lines. TRT is offering CommPoint as software only, software bundled with hardware cards, and as fully integrated turnkey systems. For more information, contact Chris Bajorek at 916-784-7777 x123.

Circle No. 526 on Reader Service Card

Genesys Offers Intelligent Routing Software For Enabling Video Call Centers

Genesys Telecommunication Laboratories announced a new software product: VideoACD. According to Genesys, the software is capable of intelligently routing inbound video calls through multimedia call centers and organizations. VideoACD will run on either POTS or ISDN lines.

"As waves of businesses and consumers begin to use the video capabilities of multimedia computers and the World Wide Web, video-enabled call centers will increase in importance," said John Bara, director of Product Marketing at Genesys. Scott Darling, director of Marketing for Intel's Internet and Communications Group, added, "Videoconferencing is now a critical capability that businesses must have as part of their telecommunications capabilities."

In a video-enabled call center, operators can see and hear their callers in real time, enabling them to more easily share information or demonstrate products. Genesys believes the first adopters of VideoACD will be the telecommunications, financial, retail, and healthcare industries.

The company says VideoACD will be available sometime in the first quarter of 1997, in systems solutions intended for enterprise and premise-based call center environments. Genesys believes that such a call center could be implemented for less than $4,000 per seat. For more information, contact Karine Hagen at 415-437-1163.

Circle No. 527 on Reader Service Card

awarded in August 1996, means that the I3000 is fully compatible with CTI applications running on Novell LANs using Novell's TSAPI. I3000 connects to Novell LANs using Securicor's Network Loadable Module (NLM) PBX Driver installed on the customer's server. Via the NLM, the I3000 interfaces directly with NetWare Telephony Services, putting all information on telephone traffic and transactions on the network. For more information, contact Michael Clark at 01-61-654-9990.

Circle No. 540 on Reader Service Card

Teloquent Announces Internet/Intranet Products

Teloquent Communications Corp. announced the first two products in the company's Internet/Intranet strategy, Open@gent and Web Call Center. These new products are intended to complement Teloquent's Distributed Call Center switchless ACD. Open@gent, a new family of desktop software products, is designed to leverage corporate Intranets as the backbone of the call center. Web Call Center was created to broaden a customer's zone of contact by allowing them to reach a company's call center agents while browsing information on the World Wide Web. For more information, contact Bob McGrail at 508-663-7570.

Circle No. 541 on Reader Service Card

New Releases From Black Ice

Black Ice Software, Inc. announced the release of two new products: Image SDK Plus/OCX version 3.0 and Fax C++OCX. The Fax C++ OCX includes support for Class 1, Class 2, and Class 2.0 fax modems. Combined with other Black Ice products such as the Generic Print Driver and Imaging libraries, the company says that software developers can develop color and monochrome fax applications or fax-enable existing applications. The Image SDK Plus/OCX toolkit allows programmers to add state-of-the-art image-processing functionality to their applications. For more information, contact Carolyn Slater-Doherty at 603-673-1019.

Circle No. 542 on Reader Service Card

HP Introduces PowerWise Assistance For The Web

Hewlett-Packard Company announced PowerWise assistance for the Web, a Web-based solution for managing networked power-protection systems. Available at no cost from HP, Web assistance makes it possible for network managers to access, monitor, and configure the UPS from any remote location by clicking on a Web browser. The browser gives network administrators an up-to-date view of real-time power problems, performance, and status, so that they can manage power protection. Similar to other HP PowerWise units, Web assistance is part of a single box solution — complete with hardware unit, cables, and CD-ROM software package, including power-management software and SNMP agent.

John Page, UPS program manager for HP's New Jersey Division said, "Web-enabled power management is a huge timesaver for network administrators. It makes it easy for them to administer their UPS from any client, in the simple and convenient Web framework."

Once installed, Web assistance runs on a network server along with the Power Management software and the SNMP software agent. Web assistance is management-platform independent and supports several platforms ranging from UNIX to Windows NT systems. For more information, contact Julie Lydon or Pat Arcand of Copithorne & Bellows for HP at 617-450-4300.

Circle No. 528 on Reader Service Card

TYAN Unleashes Netscaliber Soft Modem/ISDN Board

TYAN computer announced Netscaliber, a combination soft modem/ISDN board. The Netscaliber is a combination V.34bis modem and a 128Kbps ISDN modem, and these modes can either run independently or in concert. It is designed for the user who needs the ISDN modem now, as well as for the user who needs a high-speed analog modem now, but plans to move to ISDN in the future. Users don't have to concern themselves over whether the other party is using an analog or ISDN modem.

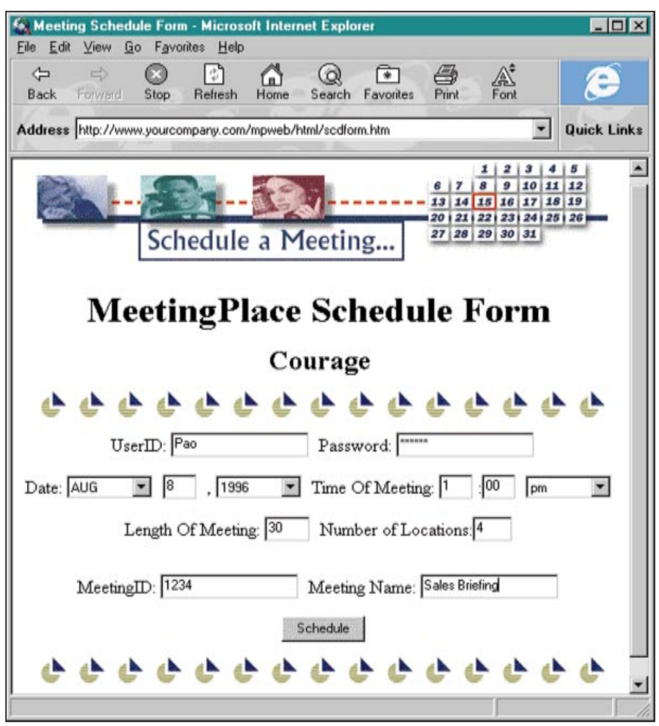
TYAN allows its modem to utilize host-based technology, which uses the main CPU's unused processing cycles for communications processing. Netscaliber also supports the AT command set. TYAN asserts this feature is important because many communications programs only use AT commands. Support for the AT command set ensures compatibility with communications programs such as Procomm for Windows and PC Anywhere. For more information, contact Lee Shaeffer or Daryl Pipkin at 408-956-8000.

Circle No. 529 on Reader Service Card

MeetingPlace WebPublisher Software From Latitude

Latitude Communications announced MeetingPlace WebPublisher, which is described as a new software product that integrates the MeetingPlace conference server with a company's existing Intranet or Internet Web sites. MeetingPlace WebPublisher allows browser users to schedule conferences, access meeting materials, and listen to recordings using the phone or a RealAudio-enabled Web site.

MeetingPlace WebPublisher provides a form from which users can set up voice conferences. From their Web browsers, users can schedule the date, time, length, password, and number of locations for the meeting, as well as specify recording options, arrival and departure announcements, and passwords. This Web-based scheduling capability extends the suite of conference scheduling interfaces provided by MeetingPlace, which also provides touch-tone telephone, Windows, Macintosh, and e-mail interfaces for conference scheduling. MeetingPlace WebPublisher is available immediately for Windows NT for $4,995 as a software component to the MeetingPlace conference server. Other options, such as MeetingNotes, gateways to e-mail and fax systems, and client software are also available. For more information, contact Shirley Macbeth or Jennifer Knapp (Schwartz Communications, Inc., for Latitude) at 415-512-0770.



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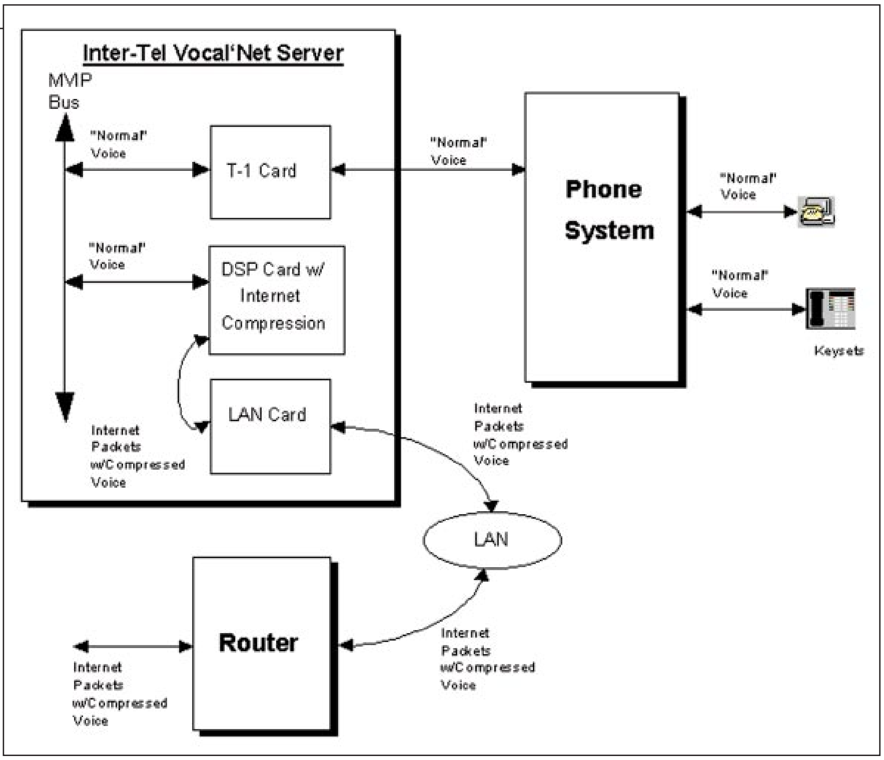


Inter-Tel Debuts Vocal'Net Server

Inter-Tel announced the Vocal'Net Server, its Internet communications solution. According to Inter-Tel, the Vocal'Net Server enables individuals in an organization to make calls over the Internet and private computer networks, or Intranets, using a desktop telephone. No specialized software or card is required in each desktop PC, and custom telephone sets are not required. The Vocal'Net Server is designed to create a bridge between the public telephone network and a TCP/IP network, and allow two-way voice communications.

Supporting up to 24 simultaneous calls, the Vocal'Net Server was designed for business applications, and since it does not rely on the server's CPU for compression and packetization, the server will remain free to handle other functions as well. Inter-Tel says that in the initial release, scheduled for March 1997, Vocal'Net Server will support calls placed from telephone to telephone. Later releases of the product are planned to support communications from telephone to computer, computer to telephone, computer to computer, and fax to fax. For more information, contact Steven Mihaylo, chairman/CEO, or Kurt Kneip, vice president/CFO, at 602-302-8900.

Circle No. 531 on Reader Service Card

Syntellect Releases VocalPoint And Bank Works

Syntellect, Inc. has introduced VocalPoint, a new Interactive Web Response (IWR) platform, and Bank Works, a predefined set of VocalPoint IWR home-banking applications that are customizable to match the individual bank's corporate identity. IWR is described as an enterprise-wide solution for high-volume processing of business transactions via the Internet. The Bank Works application suite can extend a bank's home page to provide a fully interactive solution set for its customers.

"IVR revolutionized the way businesses interacted with their customers in the 80's and early 90's, now IWR is taking customer self-service even further by facilitating such complex transactions as home banking, insurance benefits enrollment, and order processing," said Roger Reece, vice president of Marketing and Communications at Syntellect.

Tricia Lester, vice president of Marketing for Syntellect's Inbound Call Center System explains, "With the Bank Works IWR system, a bank has hands-on control of their applications. And if the bank expects a high volume of transactions, the system will cost-justify itself quickly."

For more information, contact Ann D. Conrad at 800-347-9907 x7313 or Roger Reece at x7567.

Circle No. 532 on Reader Service Card

CTI BRIEFS

Waterhouse Securities Sets Record With Periphonics IVR

Waterhouse Securities, Inc. recently took in what it calls a record 92,000 telephone calls in a single day with its TradeDirect service. TradeDirect is an automated brokerage system which has allowed Waterhouse to increase its trade volume a reported 2.5 times without increasing the number of available brokers. The firm uses interactive voice response (IVR) technology to support its TradeDirect service. Waterhouse recently replaced their IVR system due to a lack of capacity. According to Waterhouse, Periphonics was chosen for the replacement project because of their reputation for providing innovative and reliable IVR systems to the financial industry. For more information, contact Anne M. Strauss (Public Relations) at 516-468-9591 or Karen L. Ferraro (Marketing) at 516-468-9276.

Circle No. 543 on Reader Service Card

Cobotyx Aims Unified Messaging At Small To Medium-Sized Operations

Cobotyx announced the Cobot/LAN Unified Messaging System. Intended for small to medium-sized operations, the product is designed to bring full unified messaging capability (voice, fax, e-mail, video mail, and the Internet) to desktop PCs in a networked environment. Jack Antonich, general manager of Cobotyx, states, "With computer telephony identified as the next major growth area in the telecommunications industry, this timely introduction of Cobot/LAN signifies our continuing strategy to be a major factor in all areas that fall under the CT heading." Cobotyx also announced the immediate availability of turnkey and modular systems. For more information, contact Jack Antonich at 800-288-6342 x271. ■

Circle No. 544 on Reader Service Card

Product Enhancements From Pronexus

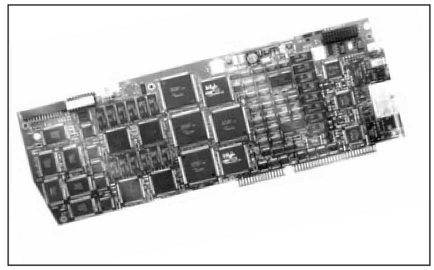
Pronexus announced several product enhancements, including: VBFax, Win32 edition (32-bit fax); expanded VBVoice voice card compatibility (including Dialogic, Pika, and Aculab); VBVoice, Win32 edition (high-density digital connectivity — up to 96 lines, T1, E1, ISDN); and speech recognition (SR) and text-to-speech (TTS) functionality.

VBFax, Win32 edition is a component-based software development toolkit that enables creation of applications for fax-on-demand, fax broadcasting, network fax server, fax storage and forwarding, e-mail fax gateway, and imaging systems. VBVoice, Win32 edition now offers greater compatibility. Pronexus has integrated Pika and Dialogic voice-driver support into its 32-bit OCX for commercial telephony applications on Windows NT. VBVoice Win32 Edition is now capable of supporting 96-line configurations.

Pronexus supports analog and digital T1-line interfaces on the Rhetorex RDSP/RTNI platform and an E1/ISDN-line interface with Aculab drivers. The company plans to introduce support for the Dialogic 300SC-E1 voice card. Pronexus is now distributing the Watson Advanced Speech Recognition (ASR) platform from AT&T with its VBVoice, Win32 edition application generator for Windows 95. Voice recognition is enabled through an interface control that processes speaker-independent, continuous speech phrases within a user-defined vocabulary. TTS functionality is provided through a system phrase property that reads data from any source and converts it into words and phrases. For more information, contact Marc Wolvin at 613-839-0033 x525.

Circle No. 533 on Reader Service Card

Dialogic Ships DualSpan



Dialogic Corp. announced it is now shipping DualSpan, a series of voice and call processing boards designed for large-scale PC-based systems. The DualSpan D/480SC-2T1 is a 48-port voice processing card which includes two digital network spans in a single PC slot. According to Dialogic, developers interested in building large voice processing systems can now comfortably fit a 336-port voice processing system in a single PC. The DualSpan D/240SC-2T1 has two ports of network interface for every port of voice processing. This makes it a candidate for applications such as debit card and international call back, which do not require fully populated voice processing boards. The D/240SC-2T1 also fits into a single slot.

The DualSpan series supports application development under Windows NT, UNIX, and DOS operating systems. An application written for Dialogic's SCbus products can now work on boards ranging from the 4-port D/41ESC to the 48-port D/480SC-2T1. For more information, contact Rosabel Tao at 201-993-3000 x6320.

Circle No. 534 on Reader Service Card

Artisoft InfoFast Fax-On-Demand/Audiotext Solution Bundled With Dialogic Hardware Now Shipping

Artisoft announced that it is shipping Artisoft InfoFast v1.0 fax-on-demand/audiotext software, the first in a series of computer telephony (CT) solutions jointly promoted with Dialogic Corp. Billed as an affordable, professional business solution that provides 24-hour access to fax documents, Web documents, and voice recordings via fax or phone, the new software comes bundled with Dialogic's ProLine/2V voice processing component and CPi/100 fax processing component, and is priced at $1995 MSRP. The bundled product will support two voice lines and one fax line, and is scalable to eight voice lines and four fax lines. Artisoft expects that additional capabilities, including interactive voice response, will be released in the first half of 1997.

InfoFast is designed for phone-intensive small and medium-sized businesses that handle repetitive customer requests for information. With InfoFast v1.0, customers can call any time and receive items such as company brochures, directions, hours of operation, order forms, and the like, via fax or voice recording. The software provides ease-of-use features such as a drag and drop graphical user interface; a fully automated catalog or index of faxable documents; a recording studio utility that allows customization of all voice prompts in one sitting; and one-button HTML document synchronization that translates Web pages into faxable images for fax-on-demand.

The software is designed to run on a standalone PC or across Windows 95, Windows NT, NetWare, and LANtastic networks. Additional options include a number restriction feature, which would allow users to lock out certain fax numbers (e.g., 900, 976, 411); the ability to create customized document headers, covers, and end pages; a capability which would allow users to generate fax files from any Windows-based application using a built-in Windows printer driver; and a feature that would allow users to configure their systems for one-call faxing, two-call faxing, or a combination of both, so callers can request documents from a phone or fax handset. InfoFast also offers such features as: built-in error checking for consistent information fulfillment; transmission log for monitoring and record-keeping; restriction for the number of documents faxed per caller; intelligent retry strategy for faxing documents; full multimedia support for voice-prompt creation; and automated start and stop dates for menu choices. For more information, call Debra Deininger at 520-670-7300.

Circle No. 535 on Reader Service Card

COMDIAL
4/C

SatisFAXtion-On-Demand

WildCard Technologies, Inc.
180 West Beaver Creek Road
Richmond Hill, Ontario,
Canada L4B 1B4
Ph: 905-731-6444; Fx: 905-731-7017
Technical Support: 905-731-4533
BBS: 905-731-4679
E-mail: support@wildcardtech.com, sales@wildcardtech.com
Web Site: www.wildcardtech.com

Wildcard's SatisFAXtion-On-Demand system is a comprehensive fax-on-demand product. You can integrate it with your existing fax server over the network, or you can use it as a standalone fax-on-demand server.

Wildcard provides three different flavors for this product, SatisFAXtion-On-Demand Silver, Gold, and Platinum. The Silver version includes 2 voice ports and integrates with your existing fax server to provide fax-on-demand capabilities. The Gold version includes 2 voice ports and 1 fax port, which will allow you to fax and perform voice processing tasks from the same server. The Platinum version includes 4 voice ports plus 2 fax ports. Each version includes the software, and each version can be upgraded.

With SatisFAXtion-On-Demand, you can make documents accessible to customers and field staff 24 hours a day. (Such documents might include catalogues, product specifications, order forms, credit applications, press releases, position papers, and tax and legal forms.) You can also use SatisFAXtion-On-Demand to reduce the cost of technical support by automating access to FAQs (frequently asked questions).

The product can handle international call dialing, and multiple language support is built-in, allowing customers to listen to prompts in their own language. In addition, the product allows you to collect a caller's fax number, which you can then use to fax broadcast your marketing materials.

INSTALLATION

The hardware installations went smoothly. Neither of the cards, which included a Rhetorex voice processing board and a WildCard SatisFAXtion 2000 board, required any jumper settings. We had no memory address or IRQ conflicts.

The software installation was a bit lengthy and slightly complex. Specifically, we had to install several sets of disks in a particular sequence. (We installed the Platinum version for our testing purposes.)

When we installed the CommandFax software, we got to the very last disk (Disk 7) when we got an error message: "Cannot find 301.msg. Skip, Abort, or Retry." We removed the disk and performed a surface scan on it (on another PC). The scan was negative. We put the disk back into the first PC and clicked on "Skip" after several futile "Retry" attempts.

We took this snag as a variation on Murphy's Law. In our case, it stipulates the probability the installation will crash becomes greater with each disk, and exponentially greater once you insert the last disk! Well, after clicking on "Skip," we got the same error message with the very next file on the disk. We thought the disk might be bad, but when we

SatisFAXtion-On Demand Specifications

The technical specifications here pertain to the SatisFAXtion 2000, which was included in the product version we tested (SatisFAXtion-On-Demand Platinum).

Type: Two-line co-processed 16-bit ISA fax data modem. Manages T.30 fax protocol, image conversion and I/O to the host PC on the card.

Number of lines: Two per card. Available in the following configurations: 2000 (2 analog lines plus LocalFax) and 2000DID (1 analog line and 1 DID line).

DAA Type: Analog.

Installation: Automatic. No switches or jumpers.

Telco connectors: 2 RJ11 for telephone line connection; 1 microDIN for LocalFax.

Card processors: 32-bit Motorola 68000 with Rockwell fax modems.

Maximum lines per PC: 32.

Regulatory: UL, CSA, CSA/NRTL, RIR/SOR 475 Class A, FCC Part 15 Class A, FCC Part 68 Class A, DOC CS-03.

Fax speed: Up to 14,400 Bps.

DOS and Windows 3.x APIs: Extended multichannel CAS 1.3, PDXCAS extended CAS 1.3.

Windows 95 32-bit APIs: CAS32(TM), PDPORT port level.

Windows NT 32-bit APIs: CAS32(TM), PDPORT port level.

Compression: MH, MR.

Modes: V.17, V.29, V.27ter, V.21 channel 2.

Image file formats: PCX, DCX, ASCII, TIF supported with CAS 1.3 drivers.

DOS Printer Emulations: Epson FX and HP LaserJet.

Routing support: T.30 Subaddressing, DTMF, distinctive ring and line.

Environmental: Storage temperature, –40 to 70°C (–40 to 158°F); operating temperature, 5 to 50°C (40 to 120°F); humidity, 10 to 90% non-condensing.

Warranty: 3 years.

List price: $1399 for standard version; $1499 for DID version. ■

INTERNET WORLD 4/C

brought it over to another PC, we were able to view and copy the 301.msg and all the other files as well. We had to choose "Skip" about 30 times and then copy the files to the correct path after the installation was complete.

We had similar problems in the old days of Windows 3.1 when we used PC Kwik cache, which could be balky when you worked with floppy disks. With PC Kwik cache running, many installations would accept disk after disk only to stop reading after the last disk was inserted. (One such installation problem arose with Microsoft Word, after about 11 disks.) In our review installation, however, we used DOS's Smartdrv caching program. We doubt it was the source of the problem. Still, the caching program could be at fault, or even a bad floppy drive, even though we've never had problems with this floppy drive in the past (nor are we currently).

We also installed Rhetorex drivers, which were contained on separate disks, and the fax drivers for the SatisFAXtion board. These performed flawlessly. Our only complaint: we would have liked to have seen a prompt asking us if we wished to have the Rhetorex and the fax drivers added to the AUTOEXEC.BAT file. The installation omits mention of what modifications are necessary. However, the documentation indicates that you must modify the AUTOEXEC by hand with certain entries to auto-load the necessary drivers.

Lastly, we installed the CommandFax for Windows and the Print-2-Fax (Windows) software programs. Both of these installations went very smoothly. Overall, we gave the installation an average rating (3), which was due more to some documentation flaws (hard-to-follow installation instructions) than anything else.

DOCUMENTATION

Overall, the documentation was excellent. The table of contents was well organized and studded with helpful subheadings. The index was just as good. We were able to look up all the basic keywords we could think of, as well as a few not-so-obvious ones. Step-by-step instructions along with many accompanying screenshots are contained throughout the documentation.

All the same, the installation instructions were a bit convoluted. We found ourselves skipping ahead a chapter and then back a few chapters to install the different components (the Rhetorex drivers, fax drivers, print drivers, and the CommandFax software). We would have liked to have a READ.ME or a quick start guide/addendum containing an overview of the installation steps on one sheet or page, including page numbers and disk labels (Insert Disk 1-Rhetorex Drivers, Insert Disk 1-CommandFax) for reference. There is a quick start page in the documentation, but it is much too high-level.

During the installation of the Rhetorex drivers, we found that a couple of the screenshots in the manual (pages 62 – 63) differed slightly from what we saw on our screen. So, we did some guesswork, which is never a good thing. We suspect that the Rhetorex drivers were updated since the last reprint of the documentation. Fortunately, we guessed well, and the system worked (as you will see in the Operational Testing section).

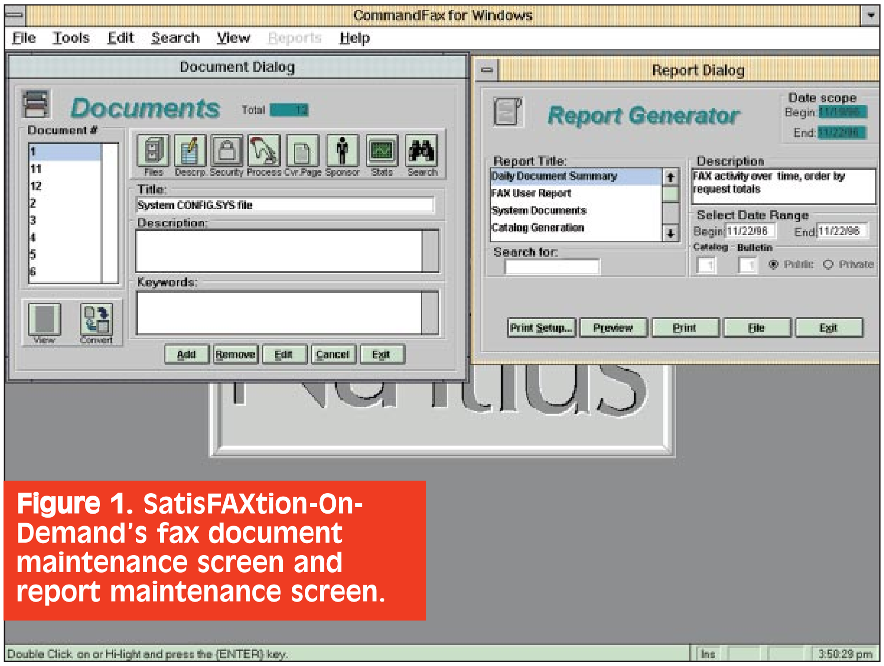


Figure 1. SatisFAXtion-On-Demand's fax document maintenance screen and report maintenance screen.

FEATURES

The bulk of SatisFAXtion-On-Demand's features are listed in the sidebar. However, it should be emphasized that the SatisFAXtion board uses an on-board co-processor. This approach ensures the highest fax performance for multiline servers by managing fax transmissions with dedicated processors. Class 1 and 2 modems, on the other hand, rely on the host PC's processor for fax conversion, compression, and transmission. This impacts performance in standalone and server environments and generally limits processing speed.

For high-volume fax environments, SatisFAXtion scales up to 32 channels per chassis and provides fault tolerance through the use of Auto-Reload and LocalFAX features. Under an error condition, the "WatchDog" timer detects when the host or LAN stops responding and automatically resets the board to re-establish communications. And with LocalFAX your network server can be backed up by a locally attached fax machine, allowing you to continue to send and receive paper faxes, even when your server is down.

OPERATIONAL TESTING

We broke down testing into four areas. These include "hookup" to the phone system; database management; using the fax-on-demand system; and using the reporting features.

Hookup To The Phone System

Hooking up the system is as simple as drawing four analog phone lines from four analog ports on your phone system and plugging them into the Rhetorex voice processing board. Incoming calls sent to the Rhetorex board are picked up after a preset number of rings. Then, an initial greeting is played.

After you choose four ports with phone numbers you wish to reserve for the fax-on-demand system, you can add this information to your company's auto-attendant (that is, "Press '1' to access the fax-on-demand system"). You can also tell your customers or clients the direct number to access the system, or you can choose to keep the direct number unpublished and instead transfer calls from your main number to the fax-on-demand system.

We plugged in the four lines for incoming calls and got the system to work the first time we tried it. We also hooked up two analog lines for outgoing faxes. This, too, was a simple setup. Overall, hardware setup was very easy. We encountered no problems whatsoever.

Database Management

SatisFAXtion-On-Demand, which uses the DBF (Dbase) standard for its database, shields you from some of the details of database management. For example, the product provides a DOS program, CFAXEDIT, which allows you to manage fax documents. In addition, the CFAXEDIT program takes care of writing to the DBF file.

You could probably use your favorite database to edit any DBF file. However, this procedure could affect the Microsoft FoxPro CDX indexes, so we wouldn't recommend trying it without backing up the file first.

The screens for maintaining, modifying, and adding fax documents are fairly user-friendly. We were able to familiarize ourselves with the interface without consulting the documentation. In addition, the product's CommandFax program, which runs under Windows, has a very user-friendly GUI for modifying and adding fax documents (Figure 1).

Figure 3. SatisFAXtion-On-Demand's screen for modifying document security and expiration settings.

Overall, the most impressive feature is the ease with which you can manage the database to add new fax documents and attach either default cover pages or specific custom pages. The database files, including their locations, can be modified via a very simple setup screen (Figure 2).

The database is able to store cover logos and cover text to each individual fax document for customized cover pages. The system stores passwords and institutes security levels for individual fax documents (Figure 3). Also, CommandFax can be configured to support 900 numbers, and the software allows you to observe the pertinent FCC regulations.

CommandFax will also process credit card purchases of faxes that have a cost associated with them within the database. It performs the Modulo 10 algorithm to determine if the number is a true credit card number, but it does not check to see if it is an invalid card (say, stolen or expired). It doesn't actually perform credit verification or payment deductions, but it creates a log file of purchases which can be submitted to a bank for credit deductions.

Another database management feature is the ability to provide documents with expiration dates. This is useful for subscription information, show brochures, product announcements, etc. Finally, it has a database of area codes and local exchanges which will automatically "parse" out any unnecessary information (extraneous area codes or the long-distance prefix) depending on the phone number entered, so your faxes will always be directed to the right number (assuming, of course, the caller entered the phone number correctly).

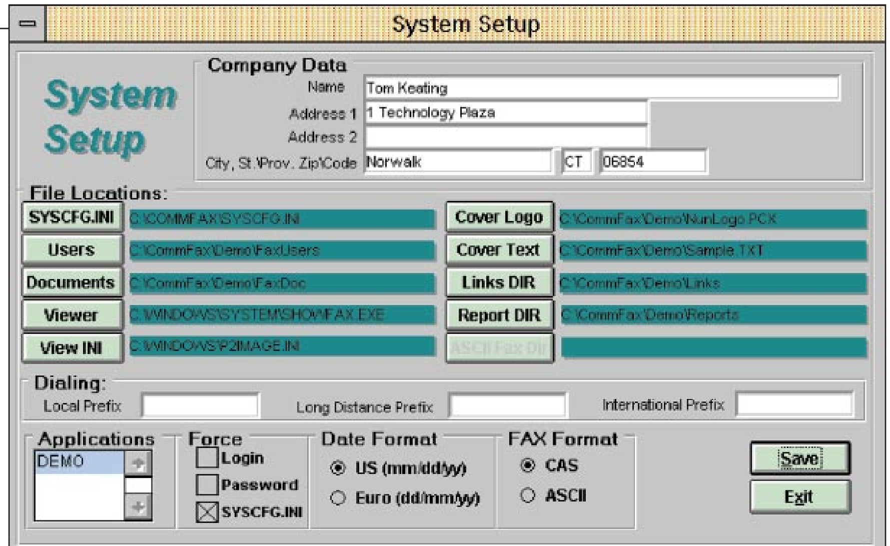


Figure 2. SatisFAXtion-On-Demand's system configuration options.

Using The Fax-On-Demand System

Our tests included mixing text with graphics files, calling in as a super-user and recording voice prompts remotely, requesting documents and bulletins, and leaving voice mail. We also tried the product's print utility, CommandFax Print-2-Image. This utility allowed us to print from any Windows application to create a fax image file (which can then be faxed).

The fax-on-demand software allows you to store up to 81 simple audiotext messages, which you can use to make brief announcements, provide additional instructions to the caller, or relay other information. CommandFax, the main piece of software, comes with a set of pre-recorded demonstration voice messages which work with its demonstration system.

To customize your own recordings, you can easily record them over existing recordings by using a telephone connected to the voice card and by going into sysop mode. You can rerecord the digits in your own voice as well, by entering sysop mode, and then simply pressing the digit on the phone you wish to record, followed by the pound (#) sign. A complete script of CommandFax's default recordings were included in a file called SCRIPT.TXT, which you can view with an ASCII editor. The manual also contains a printout of the script filenames and corresponding descriptions, which is useful for recording your own fax-on-demand voice prompts.

We were able to call into the demonstration system and request multiple documents and bulletins quickly and easily. We added a few of our own custom documents and voice prompts, and the system handled these without a hitch. We also included faxes with both text and DCX format files. The system worked flawlessly.

During our test runs we could watch the fax-on-demand server's screen display and see what was going on, such as "playing Digits: 2" or "playing MSG 300." Figure 4 shows a run-time example of the fax-on-demand system playing back the fax number the user entered. The status of each port, total calls today, error calls, and other statistics are shown in Figure 5.

Reporting Features

The reporting features were excellent (Figure 1). One of the reports included was a document catalog — a simple listing of all documents in the system. A more detailed listing, including all the fields of all the documents, is also available.

RATINGS (1-5)

Installation: 3
Documentation: 4.25
Features: 4.5
GUI: 4.25

Also included is a document diagnostics report. This useful feature checks whether all the PCX/DCX and TIF files are valid. You can also create a report of what documents were requested by day or series of days. User activity reports and other reports are also included. Reports can be output to an LPT port or to an ASCII text file, which you can then open in a Windows word-processor application for formatting and printing.

The only improvement to the reporting we would suggest concerns report output. We would like to be able to output these reports to quote-comma-delimited or other standard database file formats. Such a feature would have allowed us to perform our own data manipulation, such as sorting within our favorite database program. Of course, you could use the FAXDOC.DBF or FAXLOG.DBF database files (and write your own report using these files), but it would be nice to use the built-in reports, which already parse out extraneous fields.

ROOM FOR IMPROVEMENT

Although you could say it is the

SatisFAXtion On-Demand Features

General Features

Voice

- Customizable voice menus; ability to implement changes remotely.
- Support for up to nine languages at a time (ideal for international applications).
- Individually configurable voice lines for different voice menus, cover and back pages (ideal for service providers).
- Audiotext document descriptions.

Telephony

- Configurable outdial modes (to permit local calls, long-distance calls to specified area codes, calls to all area codes, international calls, any or all of the preceding modes).
- Configurable line modes (callback and simultaneous inbound voice and outbound).
- Support for shared lines (except on the server version — customer must call from the handset of their fax machine).
- Lock out applicable to specific numbers (such as 900, 976, 411, etc.).
- Operates in areas with multiple local area codes (localities with overlay area codes).

Facsimile

- Transmission of mixed file types; ability to combine text (.TXT) and graphics (PCX/DCX/TIF).
- Optional cover page and end page on all fax transmissions.
- High-volume broadcast fax from any DBF database (built-in).
- Choice of low-resolution (100 × 200) and high-resolution (200 × 200) fax transmission.

Document Management

- Document databases in .DBF file format; indexes in Microsoft FoxPro.CDX format.
- Separate cover logo and text for users or documents (or no cover page).
- Configurable up to 81 catalogs/9 bulletins and a nearly unlimited number of documents.
- Expiration, password protection, and embargo dates for documents and users.
- Up to three levels of password protection for documents and users.

Management And Reporting

- Keeps track of account balance restrictions.
- Collects and validates credit card numbers.
- Includes standard reports. (For custom reports, use comma-delimited or .DBF transaction logs.)
- Provides transaction logs for each call. (Records date, time, duration, fax and voice numbers, document(s) ordered.)
- Defines up to three levels of security/access. (Password protection on documents and system configuration.)

Other

- Built-in broadcast fax. (Use your system simultaneously to broadcast faxes and service fax-on-demand requests.)
- Ability to interface to most voice mail systems. (The product can also store voice messages from callers within the system.)

SatisFAXtion-On-Demand Gold

Capable of handling up to 10 calls per hour and of sending up to 80 fax pages per hour, this entry-level standalone information-on-demand server includes a 2 channel Rhetorex Duet voice card, a SatisFAXtion 400 intelligent fax card, and Nuntius CommandFax Information on Demand application software. This system upgrades easily if you require additional fax or voice lines. Handles 30+ calls/hour. Retail price is $1,999.

responsibility of the voice card and fax drivers to modify the CONFIG.SYS and AUTOEXEC.BAT, we would like the CommandFax installation to detect which cards are being used and then make the appropriate changes to the AUTOEXEC.BAT and CONFIG.SYS files automatically. This is particularly true if you are using the standard hardware, which ships with the SatisFAXtion-On-Demand product. The automatic modifications should happen after prompting the user with the changes that are to be made to these files. This would make for a complete, tightly integrated fax-on-demand solution with minimal installation hassles.

We'd like to see support for 32-bit fax and voice processing drivers, which use less conventional memory, and which generally load faster. Also, we would like to see the ability to "call/run" the fax management program (CFAXEDIT) from within the run-time CommandFax (CFAX) software. Of course, no outgoing faxes must be in queue or presently transmitting for any changes to the DBF database to take place. This is probably a security precaution, since it's possible to corrupt the database if you modify it while it is in use.

Figure 4. SatisFAXtion-On-Demand: Run-time example of voice prompts being played. In this case, the fax number entered by the caller is being read back to the caller.

CONCLUSION

We really liked the user-friendliness of SatisFAXtion-On-Demand. We were a bit surprised at the DOS interface's ease of use for fax management and monitoring. The Windows version of CommandFax was also very easy to use and maintain.

The feature set for this system is very good. If you are interested in seeing a demonstration, you can call Wildcard's demonstration system at 800-801-4549, which is available 24 hours a day, 7 days a week.

SatisFAXtion-On-Demand Platinum
Capable of handling up to 30 calls per hour and of sending up to 200 fax pages per hour, this mid-range package includes a 4 channel Rhetorex Quartet voice card, a 2 channel SatisFAXtion 2000 intelligent fax card, and Nuntius CommandFax Information on Demand software. This system also upgrades easily if you require additional fax or voice lines. Retail price is $3339.

SatisFAXtion-On-Demand For Fax Servers
Designed to handle up to 30+ calls per hour, this package is appropriate if you want to connect your Information on Demand system to your existing fax server. The package includes a 4 channel Rhetorex Quartet voice card and Nuntius CommandFax Information on Demand software. Retail price is $1,699. This system can be connected as a user to the following popular fax servers:
•FACSys 3.40a from Optus Software
•FaxPress from Castelle
•FaxServe from Cheyenne Communications
•LanFAX Redirector from Alcom
•NetSatisFAXtion from Delrina or Intel
•ObjectFax from Traffic Software
•RightFax from RightFax ■

Voice E-Mail 3.0 For Exchange

Bonzi Software
5839 Brookline Lane
San Luis Obispo, CA 93401
Fx: 805-238-5798
E-mail: info@bonzi.com
Web Site: www.bonzi.com

Bonzi's Voice E-Mail, which allows you to combine e-mail and voice mail capabilities, isn't necessarily a means of replacing your existing voice mail system. Rather, by introducing

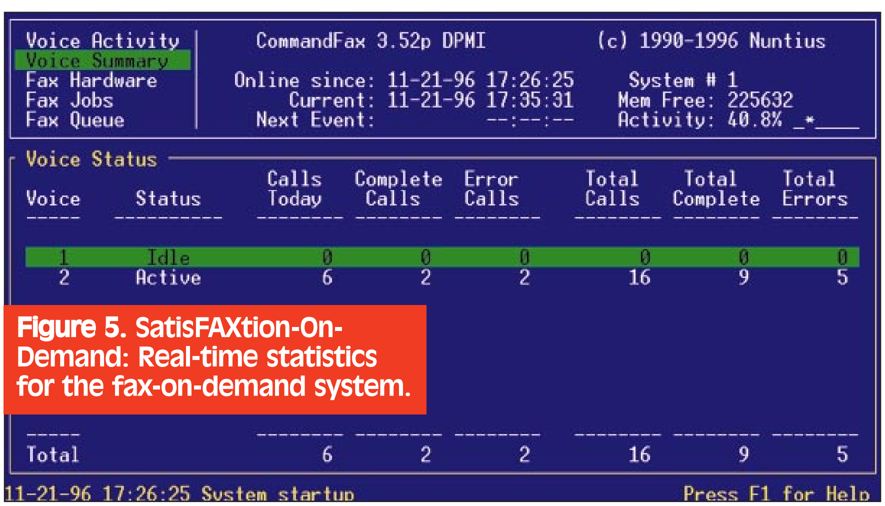


Figure 5. SatisFAXtion-On-Demand: Real-time statistics for the fax-on-demand system.

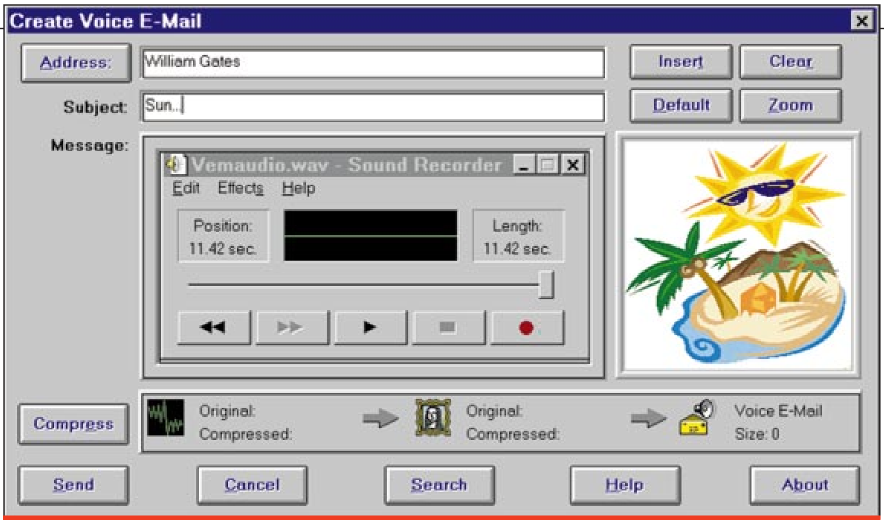


Figure 6. Voice E-Mail: Main screen for creating voice e-mail, including recording a message and inserting a picture.

voice e-mail, the product acts as an "enhancement" to your existing voice mail system. But, in any case, why would you consider Voice E-mail? Two reasons: 1) Adding the ability to clearly convey emotion and/or attitude with your text messages. 2) Increasing productivity by adding content to your messages as quickly as you can speak it.

Voice E-Mail, by giving users the ability to add emotion to messages (whether over internal e-mail or over the Internet), satisfies a basic need for people who are attempting to communicate via text messages. If you doubt this need exists, we invite you to consider why else "emoticons" would have become so popular.

As for the productivity issue, we would like to point out that while most computer users type 20–30 words per minute, human speech averages 200–250 words per minute. Thus, the use of voice over written text increases how much content can be packed into a message in a given period of time, and thus increases productivity.

FEATURES

Voice E-Mail supports the inclusion of pictures and photographs in your messages. Thus, a person's photograph can be accompanied by a voice sample from that person.

The Voice E-Mail Viewer supports all of the following image file formats: Bitmap Files (*.BMP), GIF Files (*.GIF), JPEG Files (*.JPG), TIFF Files (*.TIF), PCX Files (*.PCX), WMF Files (*.WMF), Targa Files (*.TGA), EPS Files (*.EPS), Kodak PhotoCD (*.PCD), MacPaint Files (*.MAC), MS Paint Files (*.MSP), OS/2 Bitmap (*.BMP), JFIF Files (*.JPG), CMP Files (*.CMP), JTIF Files (*.JPG), SUN Raster (*.RAS), CALS Raster (*.CAL), TIFF LZW Files (*.TIF), GEM Image (*.IMG), WordPerfect (*.WPG), Macintosh PICT (*.PCT), and CCITT Files (*.TIF).

Voice E-Mail 3.0 uses a new "lossless" compression technology specifically optimized for digital audio. Most audio compression algorithms today are "lossy," which means they lose digital waveform data during compression. The more information is lost, the more the quality suffers. And there is no way to get the information back once it's gone, which means an uncompressed recording will never again sound exactly like the original.

"Lossless" compression, on the other hand, doesn't lose any of the original data. The uncompressed audio sounds *exactly* like the original recording because *it is* the original recording. No information is lost. The uncompressed audio and the original recording are identical in size and quality.

To conserve disk space, all of your Voice E-Mail files are stored in compressed format on your hard disk. However, you can save any original .WAV file out of its compressed .VEM file by clicking the Save As button on the Voice E-Mail Player, and then giving it a name other than VEMAUDIO.WAV.

RATINGS (1-5)

Installation: 4.5
Features: 4.5
Documentation (online help): 4
GUI: 4.25

Figure 7. Voice E-Mail: The error message we received when we inserted a fairly large TIF file (5 meg).

SYSTEM REQUIREMENTS

To use Voice E-Mail, you need:

- a sound card and a microphone.
- Windows 3.1, Windows for Workgroups 3.11, Windows 95, or Windows NT.
- 4 MB of RAM and 2 MB of free disk space.

Supported platforms include WinCIM, America Online, Microsoft Mail, Microsoft Exchange, Eudora, and Netscape

OPERATIONAL TESTING

We tested Voice E-Mail 3.0 using the Microsoft Exchange version on a machine with multimedia support. We refer readers to Figure 6 to see the main interface for creating a voice e-mail.

To insert an image, graphic, or picture, you simply click on the Insert button on the "Create Voice E-Mail" screen and then select the file you desire. All images, graphics, and pictures are re-sized to fit into this thumbnail. To expand the thumbnail into the full viewer, you just click on the Zoom button, and the picture will zoom out to its full size. To remove an image, graphic, or picture, there is a Clear button. After you click on this button, the thumbnail viewer displays "No Picture." If you find yourself repeatedly inserting the same picture, you can click on the Default button. Then, every time you create a new Voice E-Mail, the Viewer will default to that picture.

Bonzi Software's Voice E-mail 3.0 crashed (Figure 7) when we attached a moderately sized .TIF file to the picture box. The crash would occur when Voice E-Mail was trying to compress the voice. Apparently, inadequate memory caused the error, which then crashed the voice e-mail program. We were forced to redo the voice e-mail. We rebooted and retried a similarly sized .TIF (about 1 meg), and the program crashed again. When we limited

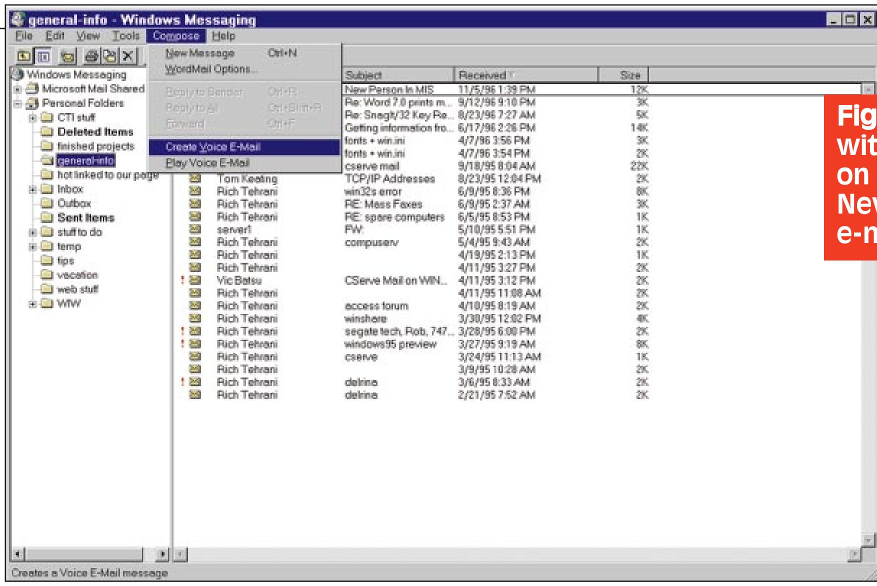


Figure 8. Voice E-Mail: Integration with Microsoft Exchange. You click on Compose and then on Create New Voice E-Mail to create a voice e-mail message.

our testing to smaller graphics, the program seemed to work fine.

To play a voice e-mail, you must have Bonzi's player. Thus, you won't be able to send voice e-mails to just anyone. (Before you send anyone a voice e-mail, you must make sure they have the player.) Fortunately, the player can be downloaded for free at Bonzi's Web site. Actually, if you do decide to send a voice e-mail to someone, and you are unsure whether this person has the player, you don't need to worry. Voice E-Mail 3.0 "tags" a message to the bottom of every voice e-mail. The message reads, "This is a Voice E-Mail (.VEM) file, created using Voice E-Mail 3.0 for Microsoft Mail/Exchange. For complete information, go to http://www.bonzi.com."

Thus, anyone who receives this message is alerted to the availability of the Web site, where they can learn about the free download option. Of course, the recipient will need Web access and a bit of computer expertise. Also, the free player permits only the viewing and playing of voice e-mails. To create your own voice e-mails, you will need to purchase the full version.

ROOM FOR IMPROVEMENT

We would like the product to make it easier for users to opt for having the voice e-mail automatically play the voice portion (auto-play) of a message. Ideally, this functionality should be available whenever a user double-clicks on a voice e-mail. At present, you have to click on the play button each time you want to listen to the voice portion of the message.

In the Exchange inbox itself, it would help if Voice E-Mail distinguished voice e-mail messages from regular e-mail messages. Color-coded icons, for example, would let you know at a glance what message types you had.

We were surprised that this product lacked a button on Exchange's toolbar to create a new "voice e-mail." Instead, you must click on Compose and then on Create New Voice E-Mail, which was a bit tedious (Figure 8). Also, in those instances where we already started a new message, we would like to have a button on the new message's toolbar to create a voice e-mail.

We would have also liked to have had the option of omitting the text at the bottom of every message directing the message's recipient to Bonzi's Web site for more information. Finally, we would have liked to have been able to attach other types of files other than picture files. If this feature were added, users would no longer be obliged to send separate e-mails for file attachments.

TeLANophy

Active Voice Corporation
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Seattle, WA 98121
Ph: 206-441-4700; Fx: 206-441-4784
Web Site: www.activevoice.com

TeLANophy, a computer-telephony and unified messaging software application, brings total call handling and mail management to the computer desktop. The product uses Microsoft Windows, and works in conjunction with Active Voice's Repartee voice mail/call processor system. The Repartee system has voice mail, automated attendant, audiotext, and facsimile functions, all of which are fully integrated.

TeLANophy, when integrated on your local area network with Repartee's voice processing system, provides fully integrated features, including voice mail, e-mail, faxes, integration with your database, as well as caller-ID and automatic number identification (ANI). TeLANophy offers total call control, including dial out, take an incoming call, request hold, transfer, and other advanced features.

INSTALLATION

A technician from Active Voice visited CTI Labs and installed the Repartee voice mail system and its accompanying software. On the client side, our installation of the TeLANophy modules went without a hitch. Getting the client to connect to the Repartee system was a little tricky, though.

We had difficulty connecting to the Repartee server because the subnet mask was entered incorrectly during the installation on the Repartee server. (It was supposed to be 255.255.255.0, but it was entered 255.0.0.0 instead.) We couldn't find the configuration program to change the subnet mask, so we ended up re-installing over 10 disks.

It did seem odd that the OS/2 server lacked an intuitive means of changing the TCP/IP settings. (Then again, the TCP/IP protocol is a recent addition to the Repartee system.) After we fixed the TCP/IP setting, we were able to have the client software on a Windows 95 machine logon to the Repartee server.

We also had some difficulty hooking up the Repartee server to our Comdial DXP switch. (This problem, however, was in no way the fault of the TeLANophy product.) At first, we were unable to access any phone functionality. After several hours, we decided to call it a day. We figured we would try and call a Comdial technician to aid us in the integration.

A few days later, our team of CTI engineers resumed its installation efforts. We disconnected a couple of our voice mail ports and hooked them up to the Repartee system. After we customized a few settings in the Comdial DXP software, we saw some functionality between the Repartee system and the Comdial switch. We were able to make an outgoing call from ViewCall (one of TeLANophy's components), which automatically dialed our extension. After we picked up the handset (which was ringing), the Repartee system started sending DTMF digits of the phone number, which we could hear on the receiver.

Unfortunately, there was no dial tone. The system could not dial out. So, we played with some different configuration settings on our Comdial DXP switch, but no sooner would we activate one feature than find we had deactivated another. We consulted a Comdial technician, who told us that unlike intercom ports, the voice mail ports are not call-conference-enabled by design; therefore, they cannot be used to connect two parties. The technician told us that using Comdial's wideopen.office would allow Active Voice's TeLANophy product to be able to perform dial out, transfers, and other call functionality.

We were a little disappointed that Active Voice lacked this information. So, we were all the more grateful that the Comdial representative took so much time to help us figure out that the Comdial DXP didn't support what we wanted to do with the Active Voice product.

[*Technology Editor's Note:* Complications such as these are common when you attempt to integrate products from different vendors. Certainly, Active Voice is not alone in having less than perfect knowledge about how its products integrate with products from other vendors. We only mention complications such as these to help make people in the industry aware of the need for continuous efforts to facilitate integration. Discussions of such problems can usually be generalized to the industry as a whole, and how it stands on interoperability issues. We expect that emphasizing the benefits of standardization and conducting open, objective discussions of integration problems will contribute to the industry's efforts to facilitate interoperability.]

Because the integration problem encountered in this installation wasn't fully resolved, our testing of some of the call functionality was limited to looking at how the GUI interface worked and how it would facilitate call handling. Some of our comments are based on a call functionality demonstration by Active Voice. However, we were able to conduct thorough tests of the product's voice mail features.

Adding Unified Messaging To Your Phone System: The Alternatives And What They Mean To You

BY SUSAN WARREN
RHETOREX MARKETING EVANGELIST

One of the best things about unified messaging is that it replaces the time-consuming, sequential interface of your old voice mail system ("Press 5 to listen to the next message…") with an intuitive, visual interface. That's great, but how do the voice messages get there in the first place? And, for that matter, how do they get back out again?

Chances are that your unified messaging server will be connected to your company's telephone system — not directly to the public telephone network. The connection between the messaging server and your company's phone system (PBX or key system) lets callers record a voice message for you when you don't answer your phone and allows users to call from the road and listen to their messages (voice, e-mail, and fax). It likewise enables a very convenient way for desk jockeys to listen to voice messages without a sound board — by playing them over their office phone.

In effect, the connection between the phone system and messaging server is a gateway for voice traffic between the data network and the public telephone network. In addition to the voice signal, other information about the call (caller ID, etc.) must pass through this gateway.

How is this critical connection made? Therein lies the problem. Unlike the IT (Internet telephony) world, where standards define most hardware and much software, telephone systems are almost always proprietary in design. Most telephone systems sold today provide methods to interconnect with PCs, but at a price. And while many of the methods provided adhere to a standard, no one method is pervasive. Some methods pass only the call information or only the voice signal, so two methods must be used together to provide a complete connection. Worse yet, about half of the legacy telephone systems already installed in North America support proprietary connections only.

TSAPI

One of the standard methods you hear a lot about these days is Novell's Telephony Services Application Programming Interface (TSAPI). TSAPI provides the kind of connection needed for unified messaging (and much more), but at a steep price. First, you must pay the telephone system manufacturer for a TSAPI NLM (NetWare Loadable Module), which can range from $10,000 to $40,000. Then there is the cost of the Novell per-seat license: a Netware Telephony Services 2.1 license is priced from $140 to $260 per user. These numbers assume your current telephone system can use TSAPI at all (manufacturers tend to offer it only on their newest models). Even if you don't buy a new phone system, the price tag for TSAPI can add up to big bucks. Unless your business plans to use its powerful call-control features for a call center, TSAPI is overkill (both technically and financially) for unified messaging.

DOCUMENTATION

The TeLANophy documentation briefly and straightforwardly covered the various TeLANophy modules and described the various icons. Though sketchy, the documentation proved adequate, mostly because the product's modules (ViewCall, ViewMail, ViewFax, and others) had easy-to-understand graphical user interfaces. In any case, more detailed instructions are available via TeLANophy's online help, which is quite good.

Actually using TeLANophy didn't occasion any difficulties as challenging as the product's configuration and setup, particularly with respect to accounts and other information on the Repartee system. Although we are not reviewing the Repartee system itself, the documentation for the Repartee voice mail system was very good (despite being divided among a potentially confusing array of separate manuals). Overall, we gave the documentation a 4.0 rating.

COMPONENTS

TeLANophy is a multi-component application. These components include ViewCall, ViewCall Plus, ViewMail, ViewFax, and E-Mail Notify/Delivery. Each of these is discussed below.

[*Technology Editor's note:* Active Voice also is shipping a third-party software called PhoneMax. It's very similar to the ViewCall Plus product, in that it is a standalone TAPI/TSAPI software program. Connect it to a TSAPI- or TAPI-compatible system, and you have a powerful first-party call control software program right from your desktop. With this setup, you don't need a Repartee server. Some of the features of the PhoneMax product include drag-and-drop conference calls, a call log, and a contact PIM.]

ViewCall

Most people can only handle one incoming call at a time. But with ViewCall, you can manage multiple incoming calls and assist important callers immediately — even when you're already using the telephone. By controlling many calls at once, ViewCall manages callers and messages in ways that would ordinarily require an assistant's attention.

With ViewCall, you can selectively transfer callers to your voice mailbox or another extension, all with the click of a mouse. Whether on or off the phone, you can play a prerecorded greeting which politely tells callers you're unavailable, and then gives them the option of transferring or leaving a message. A neat feature lets you listen to messages as they're

SMDI

Another standard method is SMDI (Simplified Message Desk Interface). With SMDI, call information is passed over a low-speed serial link from the PBX to the messaging server. The voice signal is passed using the analog station emulation method described below. It's up to the unified messaging software to match up the call information with the appropriate call. SMDI is usually cheap or free, and it provides exactly the information the messaging server requires. Since information for all calls is passed over a single link, there can be some delay while the messaging server sorts out the call, especially if the server is answering many calls at once. SMDI is a good choice for unified messaging, but not many telephone systems support it.

ANALOG STATION EMULATION

The method supported by more telephone systems than any other is analog station emulation. In telephony-speak, a station is an office phone or extension attached to a telephone system. Virtually all telephone systems work only with stations from the same manufacturer with only one exception: plain old analog phones, like the ones you use at home. Most telephone systems in use today can support analog phones as stations. Using the analog station emulation method, the messaging server connects through analog voice processing boards to emulate a group of analog stations on the telephone system. The cost of analog station emulation varies greatly between telephone systems, but is typically only a fraction of the cost of TSAPI. Since this method passes the voice signal only, it must be used together with SMDI or in-band signaling to provide the sort of connection needed for unified messaging.

The in-band signaling method passes the call information over the analog line before connecting the voice signal. Like SMDI, this method is cheap or free, and it provides exactly the information the messaging server requires. Unlike SMDI, the call information arrives with the call, reducing the chance for server delays.

For the large percentage of telephone systems that are completely proprietary, only two connection options are available. Analog station emulation can be used but, without SMDI or in-band signaling, the utility of the connection is severely limited.

PROPRIETARY STATION EMULATION

An alternative to analog station emulation is to connect using voice processing cards that emulate the proprietary stations for the telephone system. This approach, known as proprietary station emulation, costs about the same as analog station emulation, and it passes both the voice signal and call information, as required for unified messaging. Cards that emulate proprietary stations are available only for those telephone systems with large market share; nonetheless, such cards can be an attractive option when you need a unified messaging solution.

Rhetorex, Inc. designs, develops, manufactures, and markets microcomputer-based digital signal processing (DSP) hardware and software. The company, a wholly owned subsidiary of Octel Communications Corporation, is located in the heart of Silicon Valley with additional offices in New Jersey, Georgia, Texas, Illinois, and the United Kingdom. Rhetorex offers a wide range of voice processing products from the simplest voice mail applications to the largest and most sophisticated computer-telephony systems. For more information, contact Rhetorex at 408-370-0881 or visit Rhetorex's Web site at www.rhetorex.com. ■

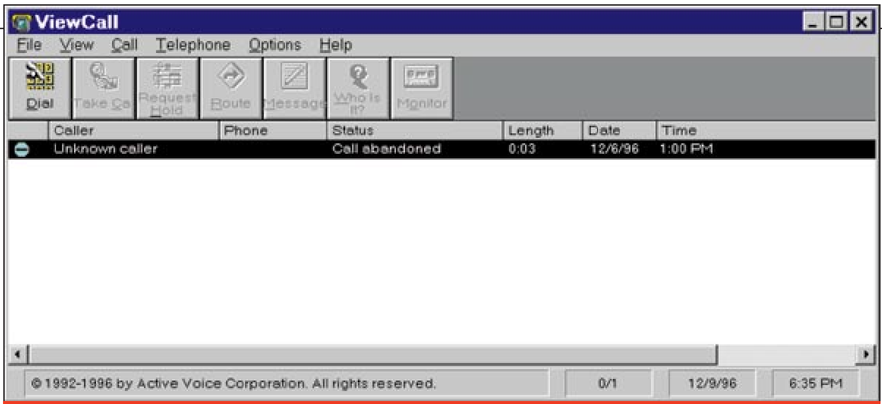


Figure 9. TeLANophy: The main screen for handling your calls, including answering, transferring, sending to voice mail, dialing, and other features.

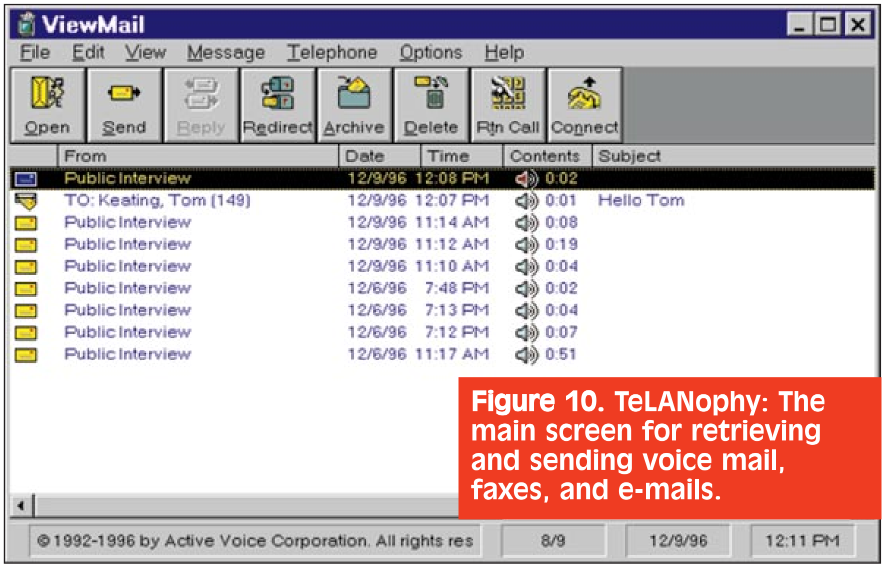


Figure 10. TeLANophy: The main screen for retrieving and sending voice mail, faxes, and e-mails.

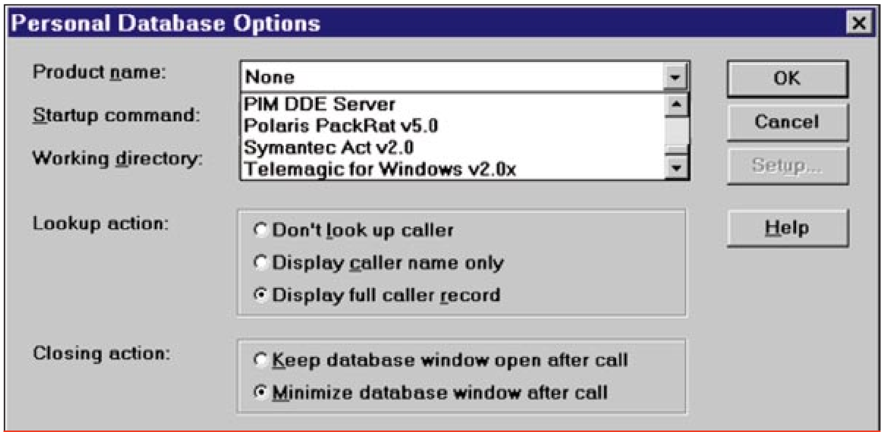


Figure 11. TeLANophy: Screenshot showing some of the built-in contact management (PIM) supports, as well as some other options.

being recorded, and also lets you pull callers out of voice mail.

ViewCall displays data about each caller from the moment Repartee routes them to your extension (Figure 9). Using caller identification or caller-entered account numbers, ViewCall retrieves records from PIMs (Personal Information Managers) and pops them on-screen automatically. Live, important data is at your fingertips before you pick up the telephone, helping you to reduce time spent on each call and to improve service for every caller.

ViewCall Plus

ViewCall Plus has all the capabilities of ViewCall and more. It controls all calls — both inbound and outbound — from your PC using the power of TAPI and TSAPI telephone systems. Basically, ViewCall Plus lets you manage calls on the PC instead of the telephone.

ViewCall Plus manages several incoming and outgoing calls simultaneously so you can handle more calls in less time with less effort. Three integrated windows control every aspect of your telephone communications. The Telephone Control window shows the flow of calls to and from your extension; the Call Log window records all call activity; and the Contact List manages data about each caller.

You can dial a number by selecting a name from the Call Log or the Contact List and clicking on the dial button. Or, you can “drag and drop” the name to the Telephone Control window. In addition, records from the Contact List or your personal database (when available) pop up automatically to give you important information for each call.

ViewMail

An intuitive Windows interface shows you the sender’s name, the subject, and the date and time messages were sent. This information allows you to quickly prioritize and respond to your messages (Figure 10).

Messages are managed with a few mouse clicks. Intuitive buttons let you hear, leave, reply, redirect, archive, and delete messages, and you can rewind, pause, and fast-forward them during playback. On multimedia PCs, ViewMail lets you use a sound device to play and record messages, so you can control them without picking up the telephone. Your hands are free for other tasks, and the telephone is available for inbound and outbound calls.

ViewFax

ViewFax makes manually sending and receiving faxes obsolete. No more walking to the fax machine, waiting in line, dialing numbers, feeding in pages, or checking every few minutes for important documents. It manages both inbound and outbound faxes right from your desktop PC, making fax communications faster, easier, and totally confidential.

Incoming faxes can be sent directly to your universal mailbox using DID, so you can access them via computer or telephone. They can be forwarded to e-mail or sent to a nearby fax machine

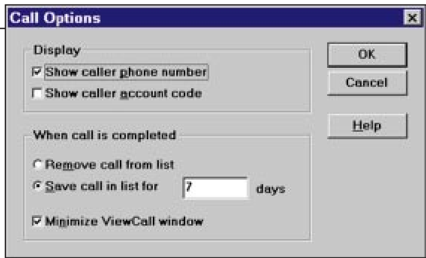


Figure 12. TeLANophy: Screenshot showing display options for calls, as well as feature for setting how long a call may be saved in a list.

Figure 13. TeLANophy: Different notification methods, as well as options for customizing sound files depending on certain events (incoming call, expiring hold, etc.).

while you're traveling, or they can be redirected to other Repartee users.

ViewFax also makes it easy to send outbound faxes right from your desktop with Print-to-Fax. Print-to-Fax integrates with many popular personal database applications, so you can deliver a fax to any contact in just a few seconds.

E-Mail Notify/Delivery

The E-mail Integration package gives you 24-hour, two-way access to e-mail messages, without a laptop or modem connection. No matter where you are or what time it is, your e-mail is as close as the nearest telephone or fax machine.

When you check your universal mailbox, the E-Mail Notify/Delivery module includes e-mail with voice and fax messages. Information is provided about each message so you can quickly prioritize them, skip long ones, and listen to attachments when necessary. You also have the option to forward e-mail to the nearest fax machine. With the E-Mail Reader module, you can listen to any e-mail message using text-to-speech conversion. Once you've heard the message, you can record a reply that's sent as a voice mail message or an e-mail with a .WAV file attachment.

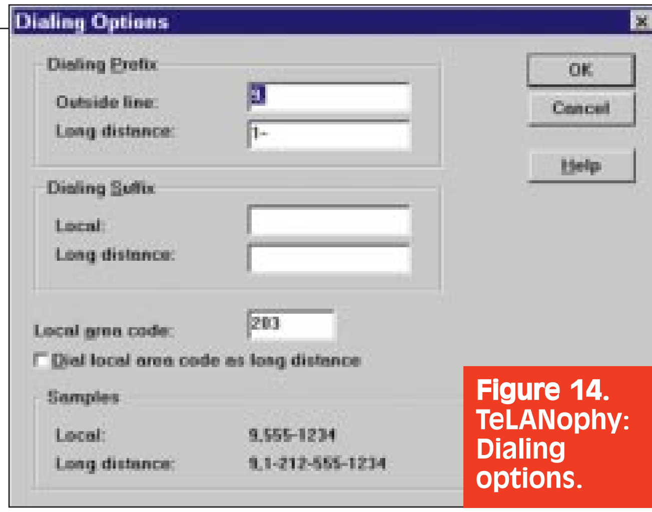


Figure 14. TeLANophy: Dialing options.

FEATURES

We will briefly discuss TeLANophy's features in the areas of call control, security, and unified messaging.

Call Control

TeLANophy gives you complete control over incoming calls, right from your computer. Calls which are answered by Repartee's automated attendant are displayed on your computer screen within the ViewCall program the instant they are routed to your extension.

Identifying The Caller: TeLANophy provides you with three ways to identify callers:

• *Caller ID or ANI information:* Whenever caller ID or ANI information is available, the system in conjunction with your database will identify a caller by name, number, and other pertinent information. Thus, you will know who is calling before you answer the call.

TeLANophy supports any caller identification information supplied by the public telephone network. For example, TeLANophy automatically displays an incoming caller's telephone number.

• *Account number or telephone number (captured as DTMF digits):* TeLANophy can ask a caller to enter a telephone number, account number, or other identifying number using touch-tones. Once this number is collected, TeLANophy displays this information automatically.

Using this information, you can have TeLANophy search your personal database of telephone contacts for a matching record. If TeLANophy finds a match, it will show the caller's name, along with the telephone number. It can even open your database and display the caller's complete record. Some of the databases and PIMS with built-in support include TeleMagic, Act!, Commence 3.0, GoldMine, and Microsoft Access; the product also supports DDE (Figure 11).

• *Recording (and playback) of information spoken by the caller:* In cases where a caller's telephone number is unknown, TeLANophy records the caller's response to a query (usually a request for the caller's name and/or affiliation). TeLANophy immediately plays the recording back to you at your workstation (if a sound device is present) or over the telephone.

TeLANophy's ViewCall interface has large, easy-to-read icons which allow you to take a call, ask a caller to hold or leave a message, or transfer a call to a different extension with a single mouse click. Some options which are available within ViewCall include setting how many days to keep a call in a save list, showing callers' phone numbers (caller ID), and keeping ViewCall minimized (Figure 12).

Notification Features: On an incoming call, you can flash the ViewCall icon (minimized) or open the ViewCall window, as well as play a specified .WAV file. You can also be informed of when a caller is approaching maximum hold time, is abandoning hold, or is leaving a message. All of these actions are indicated by user-specified .WAV sound files. Finally, you can request that all incoming calls automatically play the screened caller's name as well as have ViewCall automatically monitor the caller's message (similar to call-screening). All of these notification options are shown in Figure 13.

Dialing Options: These include specifying any dial-out prefixes (such as "9") and dialing suffixes. You can also designate your local area code and whether or not a particular area code should be considered long-distance (Figure 14).

This product could be even more powerful if it had more "smart" dialing features, such as those available with Wildcard's SatisFAXtion-On-Demand. As mentioned in our review of this product (see page 34), SatisFAXtion-On-

Demand has a database of area codes and local exchanges which automatically "parse" out any extraneous area codes or the long-distance prefix, depending on the phone number entered.

We would like Active Voice's dialing mechanism to be able to call up its own database table containing all the local prefixes/exchanges for your area. This table could be user-configurable to add just your local exchanges, and then the Active Voice ViewCall program could call this table, modify the phone number as needed, and then dial the phone number. The table shouldn't be that large, so it should be fairly easy to implement. Otherwise, you might try to use the Return Call button. Then your system might try to dial a local phone number or a phone number within your state with a "1" prefix, even if using the

AVT's CallXpress3

Applied Voice Technology (AVT) announced that it is releasing version 3.0 of its Desktop Message Manager, a client application that provides a graphical interface for managing voice and fax messages. Version 3.0 incorporates some new functionality and such capabilities as caller ID, ANI (automatic number identification), and DNIS (dialed number identification service). The product is based on AVT's CallXpress3 voice- and call-processing platform, a multi-application CTI product designed to support from 4 to 64 ports while enabling CTI applications, such as unified messaging and interactive voice response (IVR).

AVT asserts there is no single solution for something as complex as unified messaging, particularly considering that:

1) the base of e-mail clients has grown so large.

2) multiple e-mail vendors have established themselves in the market.

Thus, AVT decided it wouldn't just add e-mail to its Desktop product. Instead, AVT chose to make its product's voice and fax messages accessible to the e-mail client.

With the availability of APIs and other mechanisms established by an open computer industry, it became feasible to merge multiple mailbox storage sites into one unified graphical environment. In a recent interview with Joe Staples, vice president of marketing of Applied Voice Technologies, we discussed the unified messaging features of CallXpress3, the integration options that will become available to users of Novell's GroupWise5, and other specifics of the new product.

CTI: What e-mail platforms do you support?

AVT: We do integrations, and we have different integrations for the TUI (the telephone user interface) and the GUI (the graphical user interface). At the graphical interface, we are shipping an integration with Microsoft Exchange Inbox. So, from the Exchange Inbox you can see a single list and manage e-mail, fax mail, and voice mail messages. We're bringing that same type of integration to Novell's GroupWise5.

CTI: So you support a variety of operating systems?

AVT: Today, CallXpress3 runs on OS/2 as an operating system. As a client, we are Windows-centric, so we support Windows 95, NT client, or Windows 3.1. For e-mail integration from the telephone interface, we have a product, called E-mail Access, that allows you to have your e-mail messages read to you over the telephone. For that we support integrations with cc:Mail, Microsoft Mail, and Notes. From the telephone I can get access to voice messages, fax messages, and e-mail. The commands using the telephone are all the same, so if someone sent me a voice mail message and I wanted to forward it or rewind it or reply to it, I would press all the same digits on the telephone keypad regardless of whether I was working with an e-mail message or a voice mail message.

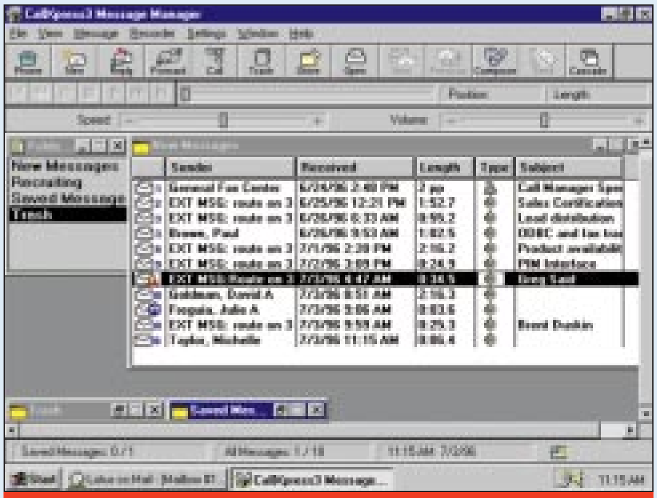
CTI: As far as the text-to-speech goes, is that a standard feature, or is that an option?

AVT: It's optional. It's sold as a package. The actual product name is E-Mail Access.

CTI: What makes CallXpress3 unique?

AVT: Live Reply certainly does. It gives you the option of being able to provide responses in real-time mode to whoever left a message (Do you want to respond with a message, or do you want to go out and actually call that person?) This capability is available either internally or externally, via the telephone or the graphical interface.

The product's incorporation of caller ID, ANI, and DNIS information is certainly on the leading edge. (There are more products that lack



AVT's Call Xpress3: User interface showing new voice and fax messages. The user can listen to any voice message by selecting it and clicking the forward arrow. If the caller is difficult to hear or understand, the user can adjust the speed and volume controls with the mouse.

prefix was inappropriate. This problem could arise if you use caller ID, since this feature always provides the area code in the phone number even if the phone number is considered local.

Thus, it would be great if you could always enter the *entire* phone number, including area codes, and yet have the system automatically correct the phone number without user intervention. Just as an example, our company's location has at least six local exchanges where the prefix 1 and the area code are unnecessary. It also happens that we have several contacts within this dialing area. If we were to use caller ID and Active Voice's "return call" feature, ViewCall would return the number "1-203-838-5555" instead of the correct "838-5555" number ("838" is a local exchange in our area).

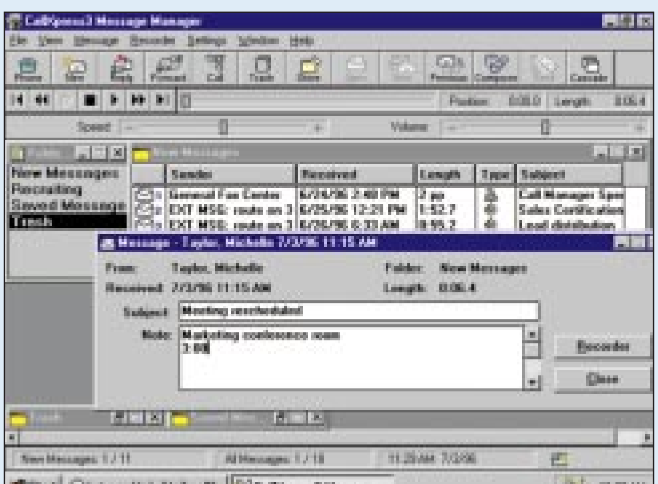
these features than include them.) Other important features are incorporated into the graphical environment of Desktop Message Manager. For example, speed and volume control, return receipt, future delivery — these are all accessible through the graphical interface.

Our integration with the Exchange Inbox is unique. We did not create a custom form as most of our competitors did. We used the Exchange Inbox look and feel and user interface. We think that's the right way to do it because the majority of people who are going to take our product, integrate it with Exchange, and use it, are going to be people who are familiar with the Exchange look and feel. They will want consistency whether they're listening to a voice mail message or viewing an e-mail message.

Our product ships with a CD-ROM called Coach. This addition, which is included at no extra charge, provides multimedia training on the use of our product, both from the graphical interface as well as the telephone interface. If you examine the deployment of unified messaging solutions, one of the big costs associated with these solutions is training all the users. So, we put together this multimedia tutorial that can be loaded either locally on a person's computer or put out on the network.

On the telephone user interface, through E-mail Access, we definitely have the broadest range of e-mail integrations available in the marketplace. From a telephone standpoint, we integrate with cc:Mail, Microsoft Mail, Notes, and (very soon) GroupWise. Many of our competitors integrate with a single e-mail package.

CTI: Does the software run on a server? That is, does it work on your own voice mail server, or can you integrate it with another vendor's voice-mail system?



AVT's CallXpress3: Screenshot showing the ability of the user to type notes about the call or fax. With this feature, the user knows what the message is about without having to listen to the message or view the fax again.

AVT: CallExpress3 provides the actual voice services. It is the voice server, so that's a required component.

CTI: Are there general hardware specifications or minimum requirements that people should be aware of?

AVT: There's really nothing out of the ordinary. It's pretty standard, available hardware. You can buy a turnkey package, where the server is bundled with the software, or you can buy the software and then do your own integration.

CTI: What kind of boards do you use?

AVT: We use Dialogic cards.

CTI: Does the product have auto-attendant features?

AVT: Yes. In fact, you can set up multiple branches of trees and unlimited call processors to carry out various functions. CallXpress3 is really a multi-application platform. It offers messaging, call processing (from the standpoint of auto-attendant), interactive voice response (where we tie to a whole series of different databases), and call management (where a front-end application is used to manage live-call traffic). So, with incoming calls, you can determine who's calling and why they're calling, and you can route the calls based on different information.

CTI: Does the product do any switching?

AVT: No. We provide an application that allows the user to more intuitively control the switching capability of a PBX or key system.

CTI: How is the product installed? By a reseller? An interconnect?

AVT: We have about 250 resellers, most of whom are interconnects. All must complete a technical certification program before they are able to sell, service, and install our products.

Applied Voice Technology, Inc. (AVT) develops, manufactures, markets, and supports a broad line of open systems-based computer-telephony products, including advanced CTI software, basic call answering systems, and voice messaging systems. For more information, contact AVT at 206-820-6000 or visit their Web site at www.appliedvoice.com. ■

Other Call Control Features: These include setting the default route (transfer) extension and the maximum number of minutes for holding on request. If this maximum time is reached, you have the option of having ViewCall automatically route the call to your voice mail or whatever extension you specify (Figure 15). If a caller has already been sent to your voice mail personal greeting, TeLANophy allows you to listen in as the caller leaves a message, as well as the ability to pull the caller out of the greeting and take the call — a remarkable and impressive feature.

Security

TeLANophy features a personal ID and password for security. You can also be away from your desk and logon using your personal ID from another computer connected to your LAN. Thus, you can arrange to have your system permit remote access to your voice mail, fax, and e-mail messages, and yet maintain security.

Unified Messaging

In brief, TeLANophy lets you check all voice mail and fax messages on-screen in one easy-to-use window. This screen-based approach allows you to access messages in any order. In addition, you can access messages much more quickly and easily than on the telephone.

To be more specific, you can use TeLANophy to:

• Check your e-mail messages from any touch-tone telephone and listen to them via text-to-speech conversion.

• Play voice messages over your handset or over your multimedia PC speakers.

Octel's Unified Messaging System

Octel Communications Corporation has indicated that it will soon release a unified messaging application called the Unified Messaging system. The product is designed to integrate voice, fax, and e-mail messaging into an easily accessed, centralized mailbox.

Octel's group marketing manager, Don Nanneman, spoke with us briefly about the new product. The following transcript is a result of that conversation.

CTI: What platforms does Unified Messaging run on?

Octel: The product is a voice messaging system that's built on top of an e-mail system, so it actually utilizes the underlying architecture of the e-mail system to store the messages and serve as a directory. The e-mail system that we're supporting at the initial launch is Microsoft Exchange server. We are also working on an implementation for Lotus Notes server (version 4.0). We're evaluating other e-mail platforms, such as GroupWise and the new open I-MAP architecture.

CTI: Does the product use MAPI calls?

Octel: It uses a software API interface to talk to the Exchange server.

CTI: Are you using an open architecture, or do you provide a turnkey system?

Octel: Unified Messaging is a complete open-architecture product. The software, which was written on NT Server, provides the telephony-user interface (TUI) and the integrations to the PBX. Unified Messaging is also the platform where the text-to-speech software operates.

The text-to-speech is a standard part of the product. Unified Messaging provides three major functions: call answering, voice messaging (where you can send voice messages to other subscribers), and auto-attendant.

CTI: What is the maximum number of ports?

Octel: It's really an infinite number, because you simply add additional servers to the network, with more ports. It's a scaleable architecture since it's built on NT, so it's not confined to how much you can put on one physical server. If you need more capacity than a single server can provide, you just add more servers to the network.

CTI: What steps are you taking to distribute and install the Unified Messaging system?

Octel: We have set up a distribution channel for Unified Messaging that is made up of existing Microsoft solution providers who are certified to implement NT and Exchange. We provide them with training and certification on the Octel components, so the customer will be getting this from the same organization that would have provided their Exchange server or NT server.

CTI: Are there any administration tools?

Octel: The interesting thing about the architecture is that it's built on top of Microsoft Exchange server. All messages are stored in the subscriber's mailbox on Exchange. There is no voice mail system storing messages. There is only one mailbox — the subscriber's mailbox. So, the administration environment at the system level is very elegant. Basically, the system administrator works with Microsoft Exchange server and uses the tab page that we've added. This tab page provides for the administration of voice functions.

CTI: Describe some of the functionality.

Octel: You can reserve extension numbers for different purposes. These include extensions at which you receive messages and what we call the zero-out extensions. You would use a zero-out extension to have a call transferred to another person when you're unavailable.

We emphasize "call class of service," which is a nice way of saying that you can establish templates for the kinds of services that people can access. Then, you can very easily implement a given class of service for a certain group of people, say, all the people in the sales organization or all of the senior executives. They might have different levels of access to the services.

CTI: What is your target audience?

Octel: Because the product is built on Microsoft Exchange, the target audience is anyone who has Microsoft Exchange Server. The target buyers would include IS managers and e-mail system managers.

• Reply to e-mail messages with a voice mail or e-mail message.

• Send your e-mail to any fax machine.

• Find out about e-mail attachments and listen to text or .WAV files associated with any e-mail message.

• View faxes on-screen via a fully featured fax message viewer, which includes zoom, rotate, and printing capabilities.

• Fax any Windows document to many people at once right from your PC. Send documents from your fax-on-

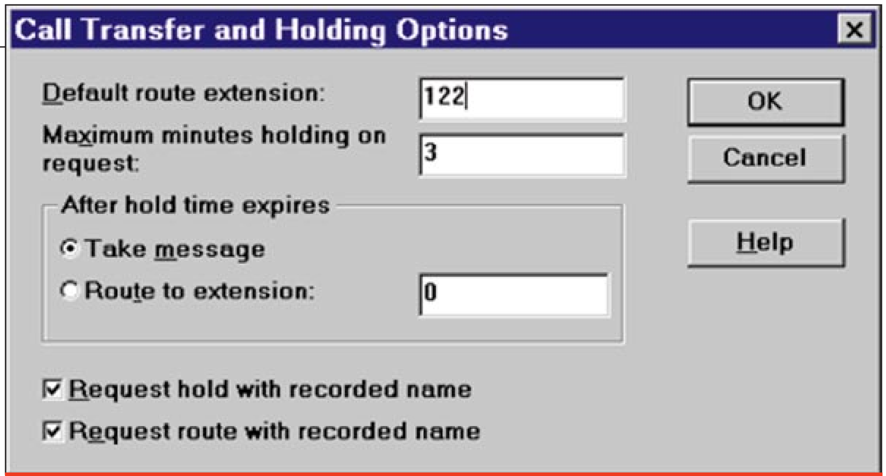
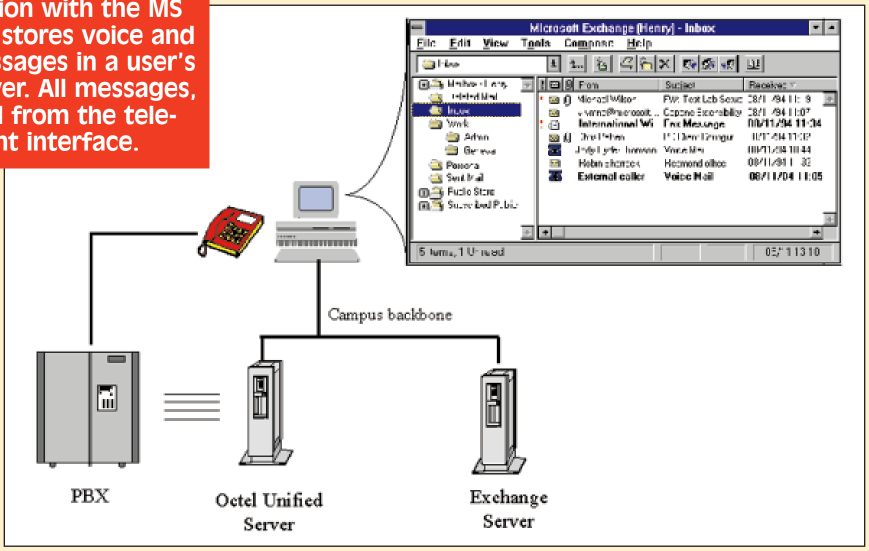


Figure 15. TeLANophy: Screenshot showing some of the default call handling options for a transfer or hold.

Octel's Unified Architecture: The Unified Messaging system operates as an application running on an NT-based server on the LAN in conjunction with the MS Exchange server. Unified Messaging stores voice and fax messages along with e-mail messages in a user's single mailbox on the Exchange server. All messages, regardless of type, may be accessed from the telephone or through the Exchange client interface.



CTI: Since it's a unified messaging product, how do you handle faxing?

Octel: For the past four years, we have had a compound mailbox on our voice mail systems that is capable of holding fax. We're taking that core technology and implementing it on the Unified Messaging system. The new product will determine whether incoming calls are voice or fax calls and handle them accordingly. Ultimately, they will end up in the subscriber's mailbox.

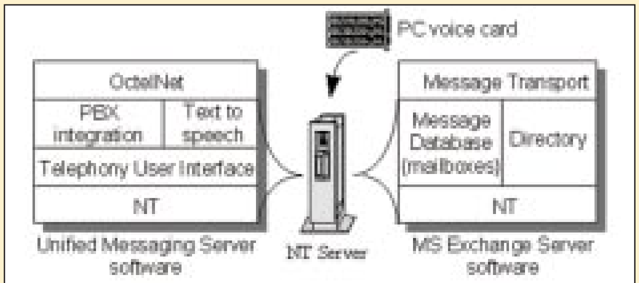
All of your messages, regardless of media type, will end up in your mailbox. So, if you're looking at your Exchange inbox, you will see an envelope icon for text, a telephone icon for voice, and a document icon for a fax image. You will be able to access any of these at the desktop.

CTI: What if you're doing any internal voice mail, say, leaving voice mail for someone in your company? Can you can just double-click an e-mail and reply using your sound card or your phone or …

Octel: Yes, the user interface for our Unified Messaging product is very similar to a form that is displayed right within Microsoft Exchange, except it allows you to listen to voice messages or to create voice messages, and you can even mix media. If someone sent you a text message, and you wanted to reply to it with voice, you can do that. Or if someone sent you a voice message (say they asked you a question about the budget for the next year), well, rather than rattle off all that information as a voice message, it might make more sense to send them a text message. A message is a message is a message — it doesn't matter what medium created it.

CTI: In terms of the proprietary nature of your product, is it necessary to purchase the voice mail player and have it installed? Could you send a voice mail message across the Internet to someone who's not using your software?

Octel: You can send a message to anyone because messages are stored in the e-mail system. The only thing you need to play our voice messages is what's called an ACM, which is a



Octel's Unified Server: In this example, both the MS Exchange server software and the unified messaging server software are operating on the same NT-based PC server. MS Exchange server acts as the repository for all subscriber messages and provides the single unified directory. The unified messaging server provides the telephony user interface heard by the caller as well as the PC cards, which provide the interface to the PBX.

demand library to any recipient directly from your PC.

• Sort messages in any order you choose.

• Specify groups (such as employees by department) for group e-mails, faxes, or voice messages.

• Take advantage of message properties such as: archiving a message for a specified number of days; indicating whether a message is private or urgent; requesting a return receipt; and scheduling future delivery of a message.

In addition, the product's E-Mail Reader lets you hear and respond to e-mail right over the telephone using text-to-speech conversion. E-mail Reader "reads" the text portion of any e-mail message and will play any attached .WAV files.

The product supports complete integration with several e-mail platforms. These include Microsoft Exchange (Windows Messaging), Novell's GroupWise, and Lotus' Cc: Mail.

TeLANophy offers an extensive range of features. Although we have a few suggestions for improvements (see Room For Improvement), we gave the product an excellent rating (4.9) for its feature set.

OPERATIONAL TESTING

TeLANophy has simple, powerful controls which let you manage messages within a graphical user interface. Intuitive commands let you hear, leave, reply, redirect, archive, and delete messages with one or two mouse clicks. You can hear your messages in any order by simply pointing and clicking to play back any message.

Call Management

How well you handle your telephone traffic is critical to your success, regardless of the size of your company. TeLANophy's call management applications direct this traffic by managing calls on-screen and overcoming the limitations of the typical desktop telephone. With ViewCall and ViewCall Plus, you can control several calls at the same time and identify callers before picking up the telephone. This does more than enhance productivity. It actually decreases the average length of each call and improves service where it counts most — on customer contacts. To be honest, we were absolutely in awe at what this system could do, even if we couldn't get it to work completely with our switch. We did, however, see a demo of some of the call handling of ViewCall at various shows, so we know it works!

filter or a sound interpreter. For instance, if you're using a Microsoft-based machine that's using multimedia hardware, there's an ACM for .WAV files, and those come free with the PC when you set it up.

CTI: So you're using .WAV file format?

Octel: It's actually a form of a .WAV file, and our ACM can be downloaded from the Internet. In fact, it's actually being bundled with the Microsoft software that's coming up, all the Office '97 stuff. Anyone who has the current Microsoft voice player that's part of the standard Microsoft Windows set could play our voice messages.

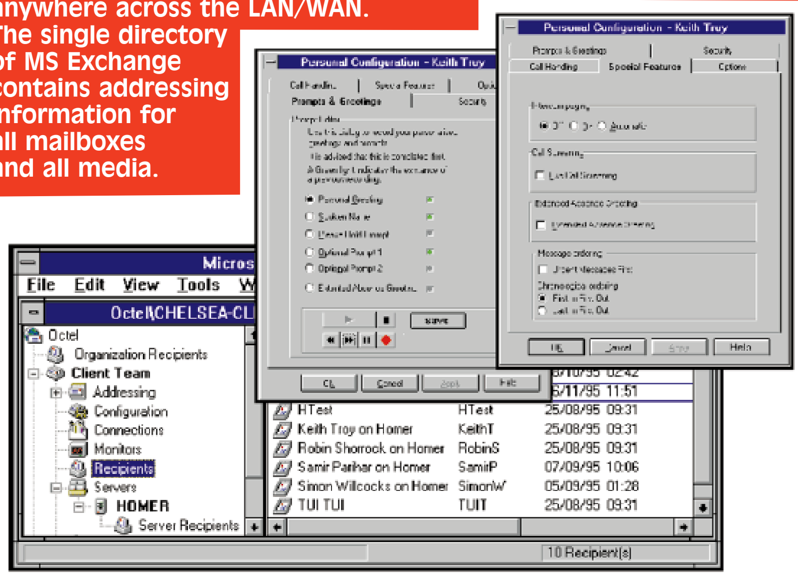
CTI: In other words, the roundabout way would be to use the sound recorder to record it, then attach it, but that's awfully tedious. It's nice that the product essentially does this for you.

Octel: Yes. The beauty, of course, is that if someone sent you a message from an external multimedia-based machine, but you happen to be at a machine that doesn't have multimedia, you can still play it back through the telephone.

CTI: How is that done?

Octel: When using the Unified Messaging client, you tell it whether you want your playback to be through multimedia, or whether you want your

Octel's Unified Administration: Here, the MS Exchange server administrative environment is used to accomplish system administration of mailbox features and the management of adds, moves, and changes. Administration may be performed anywhere across the LAN/WAN. The single directory of MS Exchange contains addressing information for all mailboxes and all media.



playback to be through your desktop telephone. And in the office, most people prefer to use the telephone, it's faster. It can be more confidential. Also, many people don't have multimedia yet.

CTI: Now, say the user wants the playback on the phone system. How are you integrating to the handset? Do you have a link between the computer and the phone, or is everything based on the server and the server system?

Octel: It's all done through the server and the PBX.

CTI: How does it inform the handset to send the data to a particular line?

Octel: The server, which has the connections to the PBX, has lines much like a typical voice mail system has lines from the PBX coming to it. We just take one of those lines and place a call to the phone at your desk through the PBX.

CTI: So you can handle digital lines then. You don't have to work with just analog.

Because TeLANophy allows you to selectively handle incoming calls, you are able to work without distraction and make more effective use of your time. And because you have information about, and control over, incoming calls, you are better prepared to answer calls and can give callers better service. With TeLANophy, you can handle your calls efficiently and quickly in ways that would ordinarily take a full-time secretary.

We liked being able to be on the phone and see calls coming into the ViewCall program. The ability to send them to voice mail, tell them to hold, or pull them out of voice mail were particularly useful features. Also, being able to select from eight different messages to play to the caller was a real plus. Another nice feature is having the option to have the caller's name automatically spoken to you. If you opt to leave this feature off, you can always click on the "Who is it?" icon on the toolbar when you want the caller's name spoken to you (Figure 9).

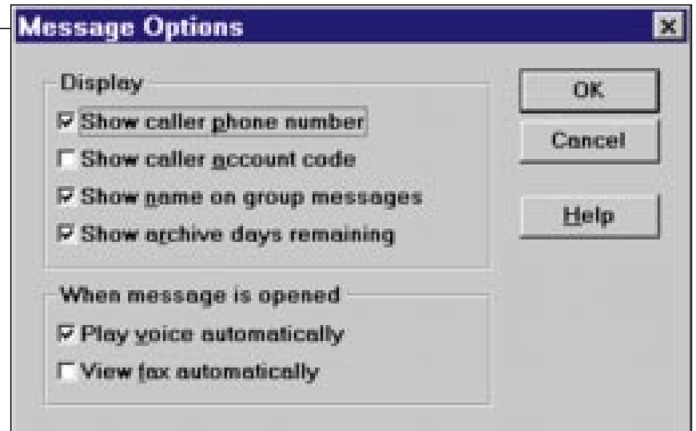


Figure 16. TeLANophy: Message options.

Alternative Access

As previously mentioned, you can use a telephone to access your unified messages, but there is another access method as well. Since TeLANophy can use the TCP/IP protocol (IPX also), we imagine it's possible you can access the TeLANophy modules using Windows 95 Dial-Up networking, Trumpet Winsock dialer, etc. Then, you would just dial into your LAN to access your messages, using RAS or a secure Internet/Intranet dial-up.

Dealing With "Locks"

While testing the ViewCall and ViewMail modules, one of our testing

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Octel: Actually, it doesn't matter. You don't have to have a brand-new PBX that supports TAPI. The Unified Messaging system will work with any type of PBX, even the old stuff with analog lines. We just pick up a line and place a call.

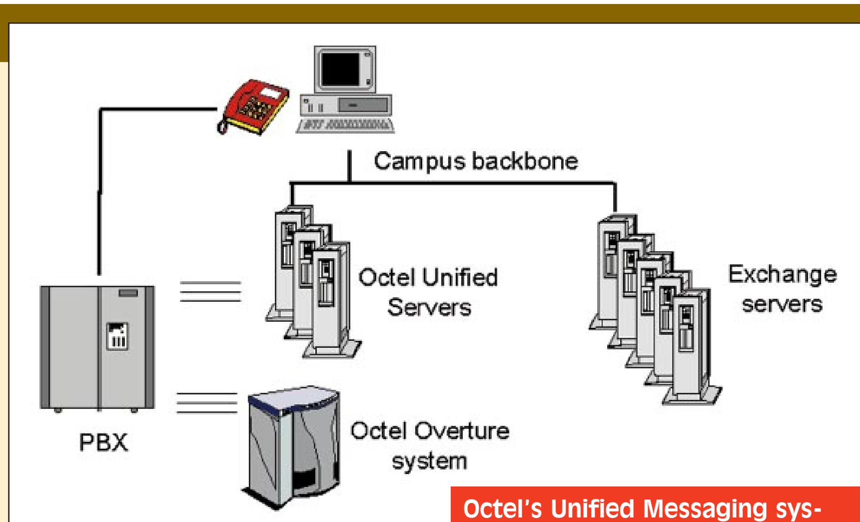
CTI: Do you support the major PBX types?

Octel: We support 128 different PBX types and models, and that's really one of the core competencies that we have here at Octel. Over the past 15 years, we've built our business on PBX integration. It's such a critical part of having a voice messaging system.

CTI: Is there any sort of tie-in with the Internet?

Octel: Yes. First, we facilitate use of the Internet as a transport. All our messages are stored in the e-mail system as messages, so you can route your messages anywhere you want on the Internet, whether it's from one corporate site to another or off the corporate site to anyone else. We can do this because in the Unified Messaging system, voice is just another message in the e-mail system.

Second, we're taking advantage of the technology that Microsoft is implementing with Exchange server. As regards the upcoming release of Exchange server 4.5, Microsoft has been talking about the Web browser client that will allow users, through a Web browser, to access their Exchange mailbox right over the Internet. So, we really don't do anything in addition to make that work, but we take advantage of all of that underlying technology. The beauty of using an open platform like this is that we can concentrate on the things that we're really good at, and we can take advantage of the things that other people do that they're really good at. We don't have to go out and build some sort of convoluted Web environment. We can take advantage of what the underlying system provides for us.



Octel's Unified Messaging system interoperating with an existing Octel voice mail system. Users are located where they prefer, and migration can occur at the customer's pace. Multiple voice/exchange server architecture eliminates "single point of failure." Voice applications remain on the Overture system as messaging migrates to the unified system.

Octel Communications Corporation offers a wide array of products and services for the delivery, access, and exchange of voice and fax messages over worldwide communications networks. Octel's customer base includes businesses of all sizes, governments, educational institutions, telephone companies, and cellular service providers. For more information, contact the company at 408-321-2000 or visit Octel's Web site at www.octel.com. ■

computers hung, although we don't believe any of the TeLANophy modules was the cause. In any event, after rebooting the computer, we couldn't get back into ViewMail. Apparently, the Repartee voice mail system thought we were still logged on. (It gave an error message indicating that that particular user ID was "in use.")

We thought we might have to reboot the Repartee system. We called technical support to see if there was an alternate solution, since bringing down a company's voice mail system can be inconvenient, to say the least. Also, since everyone's computer locks once in awhile, it would not be a smart thing to have to reboot the voice mail system anytime a computer crashed and "locked" their user ID account.

Technical support at Active Voice told us we could log onto ViewMail as a system administrator, start a "new message," click on the "locked" user name, and then click on the LOGOFF button. We think it would have been better if the ability to log off users could have been accessed from a more convenient position within the software. Resorting to a New Message screen didn't seem very intuitive. We also think that each individual computer user should have the ability to "unlock" their own account without having to ask an administrator to unlock their account for them.

A Minor Mystery

When we went into ViewMail, we had some messages, each of which was called "Public Interview." We thought they were some test voice messages that were included with TeLANophy, but when we played them, we heard the voices of several of our staff people. Some of these recordings were two minutes long. Strangely, no one remembered making these recordings. We had a mystery on our hands…

Each time we tested the ViewCall program, making an outbound call, we

Ten Four's TFS Electronic Mail Gateway

TenFour's e-mail gateway software connects disparate e-mail platforms. It does so by combining:

• A modular approach.

• Support for the most popular LAN-based e-mail systems.

• A Windows administration interface.

Is the TFS Electronic Gateway appropriate for your company? It may be, especially if your company has acquired new sites that use different e-mail systems. The TFS Gateway could also help you if your company includes departments or sites that deviate from corporate norms in their choice of e-mail solutions. In any case, the TFS Gateway offers a wide range of features for the corporate user.

MODULARITY

E-mail gateways are usually installed between two different e-mail platforms. Such gateways are used to accomplish bi-directional translation between the two different systems. The TFS Gateway, however, first changes the specific format of an e-mail message into a general TFS file format and then into the format of the destination e-mail. It is this approach that accounts for the TSF product's modularity.

The TFS Gateway is actually a suite of Windows-based modules. You will need one of these modules for every different e-mail system you wish to connect. Thus, expanding your system and enhancing its flexibility is simply a matter of adding modules. (TFS Gateway modules are sold separately.)

CONNECTIVITY

LAN connect software modules support Novell GroupWise, Microsoft Mail, Microsoft Exchange, Lotus cc:Mail, Lotus Notes, and FirstClass. Global connect software modules support Internet mail (bundled UUCP and SMTP) and MCI Mail.

ADMINISTRATION

The Gateway comes with the TFS Administrator application, which is the nerve center of the Gateway. Most functions of the Gateway, including the e-mail modules, can be controlled from the Administrator.

FEATURES

The TFS Gateway includes features such as:

• Support for multiple Internet protocols (selectable options include both UUCP and SMTP).

• Directory synchronization (allows auto enrollment of users).

• MIME compatibility (supports the Internet standard for transmitting file attachments including text, binaries, sounds, and graphics).

• Virus scanning (capable of launching anti-virus utility of your choice to scan attached files).

• Mail forwarding (can route incoming messages to a remote address).

• Support for multiple file attachments (within a single message).

• Automatic detection of character set and attachment type (MIME or uuencode) and automatic decoding.

Additional features include mailback, signature files, automatic reply/receipt, usage statistics, group mailings, and conversion selection.

MINIMUM REQUIREMENTS

System requirements will vary depending on traffic load. However, you will need — at the least — a dedicated 486 machine, 8 MB of RAM, and 10 MB of hard drive space. You will also need to use Windows 3.X or Windows 95 or a Windows NT server or workstation.

IMPROVEMENTS

According to TenFour, the latest version of TFS Gateway (2.23) offers several enhancements. These include dial-up networking support, improved scheduler and directory synchronization, and a better virus scanning interface.

TenFour has announced it will

noticed another new ViewMail message in our inbox. It even had the current date and time, which meant that it was just recorded. At first, we thought it was the microphone hooked up to one of the machines running ViewCall, but when we tested this theory, we determined the microphone wasn't the source of these recordings. We joked that maybe the office was "bugged," and that we were somehow picking up the transmissions.

Then, one of our engineers picked up the handset to a phone hooked up to an AS-4 line simulator, which was also hooked into the Repartee server. He could hear the Repartee's auto-attendant asking us to leave a message. At last we found the source of these strange recordings. But now we were curious as to how the Repartee server was able to figure out that we had a handset connected on line 3.

We watched the Repartee server to see what was happening. It started to display "LAN client" on each of the four ports, one at a time, starting with port 4 and working backwards. Then port 3, where we had the simulator, said "OPENING BOX." When we picked up the handset, the auto-attendant started to speak. Apparently, the Repartee system was able to determine where there was a port hooked up.

RATINGS (1-5)

Installation (TeLANophy software only): 4.7
Documentation: 4.0
Features: 4.9
GUI: 4.5

Note: The installation rating excludes hardware considerations because we had difficulty integrating the product with our Comdial switch. (These problems were in no way the fault of Active Voice.) Thus, we concentrated on software issues in the installation portion of the review.

If no DTMF digits are entered in response to the auto-attendant (that is, if there is a time-out), the system, by default, sends the caller to a general voice mail, which Active Voice calls a "Public Interview." Thus, we were being recorded even though the handset to the phone was on-hook. This explained the variable length messages that we got. The system would record longer messages when two people were talking louder than normal or closer to the handset, and shorter messages if the people present were further away. (The Repartee system uses voice-activated recording, so if it doesn't hear anything, it stops recording.)

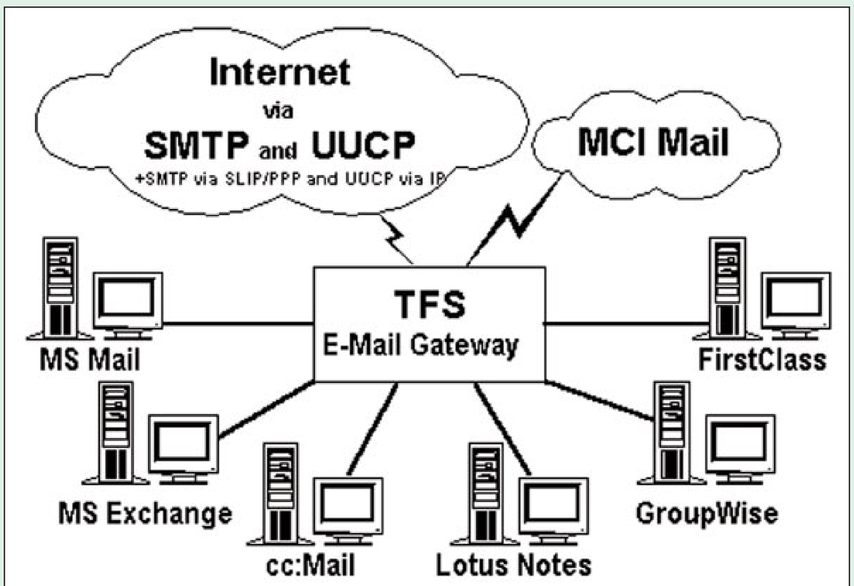
Small, But Welcome Conveniences

We were pleased the product gave us the option of automatically playing a voice mail message (the message would play immediately after we opened it with a double-click). This saved us the trouble of clicking on a Play button. This may seem a minor point, but we've seen many products that oblige you to double-click to open a message and then click once again, on the Play button, to listen to a voice/sound file. TeLANophy spared us this petty annoyance (Figure 16).

Another nice feature is a program called "ViewApps," which automatically loads both ViewCall and ViewMail. This saves you the trouble of having to double-click on two separate icons. You can also put the ViewApps program in your StartUp group. This way, when you reboot your computer, ViewApps will automatically start these applications.

ROOM FOR IMPROVEMENT

Let's start by recapping a couple of points made earlier, and finish with a



TenFour's TFS Electronic Mail Gateway.

soon offer a short message service (SMS) module. This enhancement will allow TFS Gateway users to send e-mail messages to digital mobile phones (provided these phones support the SMS standard) as well as alphanumeric pagers.

We would like to recommend a few additional improvements. To start with, future releases should be enhanced to support the addition or removal of gateway modules from the adminstrator. Adding context-sensitive help to setup dialog boxes would make it easier for users to understand some of the more arcane settings "on-the-fly." We would also like to see 32-bit modules across the board (at present, most modules are 16-bit). Finally, TenFour should add POP e-mail support to the Internet module.

TenFour, a Swedish software development company headquartered in Stockholm, has been working with the integration of electronic mail since 1991. Today, TFS Gateway is used worldwide to connect LAN-based e-mail systems to the Internet, to MCI Mail, and to each other. For more information, contact TenFour at 800-837-0046 or visit the company's Web site at www.tenfour.com. ■

couple of additional suggestions. As mentioned previously, we believe that each individual computer should have the ability to "unlock" its own account without having to rely on an administrator. Limiting unlocking capabilities in this way is inherently risky if you have only one or two administrators. A lockout could easily occur while an administrator is unavailable, which means that employees could find that they have no way of accessing their messages.

Another point made earlier concerns the ViewCall program. Basically, we would like the program to interface with a table containing local exchanges. The program should "fix" the phone number to make sure it is always dialed correctly.

As for new suggestions, we'd like Active Voice to make it even more convenient to select a voice message. Currently, the product allows you to hit a message button on the toolbar, which pops up a window. In this window, you can then click on one of eight customizable messages for playback to the caller. However, we'd like to be able to customize the toolbar and add our voice messages (four or so should suffice) as icons to the toolbar. Playing these messages could be accomplished with a single mouse click.

For example, we'd like to be able to customize and add icons/buttons to the toolbar that would correspond to: "Please hold. I'm almost done with this call," "I will call you right back when I am finished with this call," "Call my assistant at extension 100, and she will be able to help you," "I am transferring you to my assistant who will be able to assist you," and others. We'd like to designate these buttons MSG1, MSG2, etc. We would also like to use "bubble help" to select the appropriate button. A customizable default message button on the toolbar would also be useful.

One last thing we would like to see is an addition to Active Voice's Web site: a page containing information on how to integrate TeLANophy with products from the popular switch vendors (Nortel, AT&T, Siemens Business Communications, Comdial, and others). This page should at least list supported switches. Such information could aid prospective buyers. (Of course, generally speaking, the Repartee system and TeLANophy are sold through resellers, who normally take care of these issues and install the system for you.)

CONCLUSION

Integration problems aside, TeLANophy is a great product with a large feature set. Between its unified messaging and its call control features, this product is making some waves within the CTI industry. We look forward to seeing (and reviewing) the Microsoft Exchange (now called Windows Messaging) version of TeLANophy, which will be available by the time you read this review.

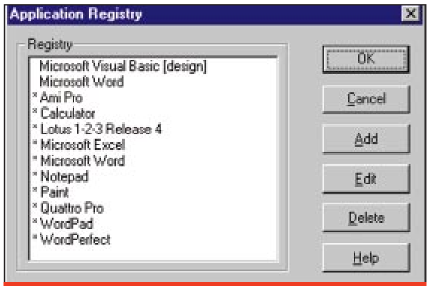


Figure 17. UPStart's Electronic Workbook: Screen for adding applications to the registry.

UPStart

SL Waber
520 Fellowship Road, Suite 306C
Mt. Laurel, NJ 08054
Ph: 800-634-1485
Web Site: www.waber.com

We've all heard the adage that nothing in life is certain, but for death and taxes. Nowadays there is another certainty to add to that list — computer system crashes.

If your computer crashes, chances are the cause is some kind of power disruption (a surge and/or outage). Depending on the state of your computer (for example, writing to the hard disk), a power spike, if sufficiently intense, can severely damage expensive hardware, such as memory, hard disks, processors, or other motherboard components. Of course, you'd like to avoid "frying" sensitive computer components no matter what you're doing with your computer. If you're using desktop CTI applications, however, you have additional risk. That is, you could destroy expensive (and hard to replace) voice processing boards, fax boards, and other computer-telephony hardware.

Running CTI applications raises the stakes in another way. With CTI, power outages, and the resultant data losses, can be particularly troublesome. Currently, many companies implement some sort of backup solution (tapes or other media). These backups, however, do not contain the most recent versions of current documents, or any new documents or files. This weakness is particularly important with respect to desktop CTI, where you might be transferring, conferencing, etc. It isn't good business to have your clients' or customers' calls disrupted. Thus, having a single uninterruptible power supply (UPS) on the PBX, IVR system, fax-

Specifications For The SL Waber UPStart

- Maximum ambient temperature rating: 40 °C
- Dual 250VA power outputs
- Reacts to power failure in less than 4 milliseconds
- Reacts to surges and spikes in less than 1 billionth of a second (The DataGard circuitry is designed to clamp surges and spikes before they can damage computer equipment, even when the master switch is off.)
- Recharge time: 2 hours
- RJ11 phone connector
- Battery life: 3–6 years
- Input and output voltage: 120V
- Input and output frequency: 60Hz (+/– 5 percent)
- Input and output current: 12 Amps

back server, or other server, is no longer adequate. Users of desktop CTI need to have a UPS on each desktop so that live conversations are not interrupted by a power outage.

If you do have UPS at your desktop, then, during a computer-based call control command, if some type of call transfer or other CTI-based transaction needs to be accomplished, you will have enough time to at least inform the other party of the power loss. Then, you can politely end the conversation. You can call the other party back when power is restored.

Typically, a UPS is a battery backup system with a limited power supply. If you are not at your desk to shut your computer down and save your files, it is possible the UPS could exhaust its power supply. If this happens, your documents will not be saved. Hence, a UPS that can act on your behalf is preferable. One such UPS, SL Waber's UPStart, is designed to keep your computer up and running, close active programs, save any unsaved files automatically, and shut down the computer at a preset interval after a power outage.

SL Waber has promoted UPStart as the world's first five-in-one UPS. It allows you to plug in a computer, monitor, modem, and printer, as well as another auxiliary peripheral to the unit. Communication to the computer is accomplished through the serial port, which allows UPStart to tell the computer when to save files and when to shut down if power is disrupted.

DOCUMENTATION AND INSTALLATION

The documentation was fair. One cavil: the technical support contact phone numbers weren't where you would expect to find them, in the contents or the index. Instead, they were located on the back of the manual.

Installation of UPStart was straightforward. Hardware installation consisted of plugging in the power cords to the clearly marked connections in the back of the unit. The software installation was very easy. The software consisted of one disk, and didn't have (or need) any customizable options other than the install-to directory.

During the installation of the Electronic Bookmark software, we noticed that the "splash" screen displayed the version number 1.0; actually,

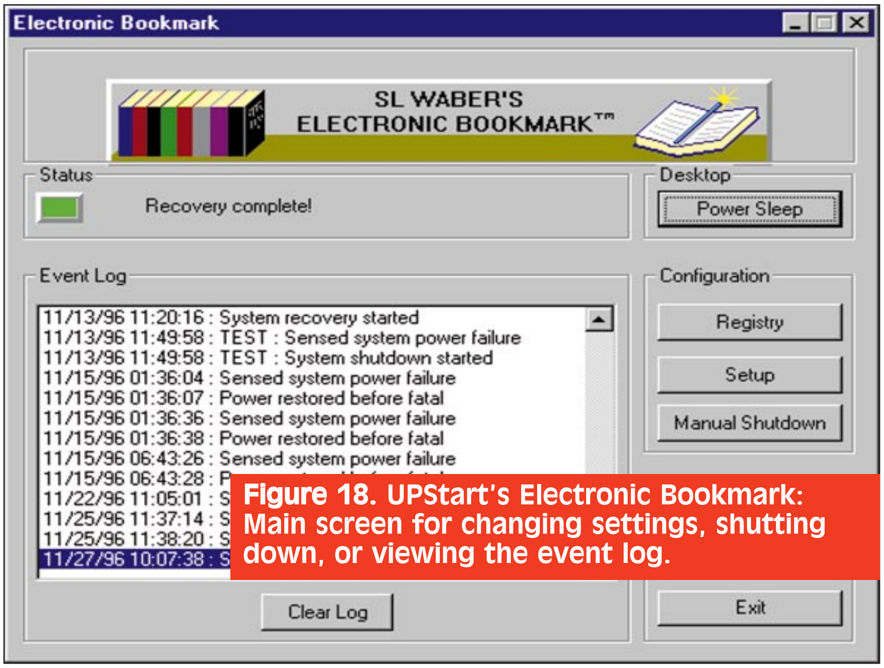


Figure 18. UPStart's Electronic Bookmark: Main screen for changing settings, shutting down, or viewing the event log.

it was version 1.2. The installation also brought up a typical "after-you've-installed, read-me" file, which actually had some pertinent information, which is unusual in these types of read-me files.

FEATURES

SL Waber's UPStart consists of a Power Center, which allows you to power up equipment individually or all at once with a master switch. Switched outlets control computer, monitor, printer, and two other peripherals or accessories. Two auxiliary outlets, which will accept transformers, are also included on the unit. The unit fits inside a flat model casing, thus allowing monitors to be placed on top.

UPStart's status LEDs (light-emitting diodes) and audible alarm indicate when power is interrupted. The unit offers up to five minutes of backup power to several connected devices.

Other features include protection from electromagnetic interference (EMI) and radio frequency interference (RFI). Also, an electrostatic discharge plate safely transfers and dissipates electrostatic charges to ground. Finally, UPStart provides solid-state protection to prevent spikes and surges. SL Waber offers a $25,000 guarantee that your equipment will not be damaged by a power surge.

OPERATIONAL TESTING

When we first turned on the UPStart, we were a bit puzzled. The computer, monitor, and the external modem all came on, but only for half a second. We pressed the master switch again, but again the equipment would turn on for only an instant, and then shut off. Then we pressed the master switch a little harder and held it there for half a second, and we heard a click. It worked. Apparently, there is a relay mechanism or trip switch in the master switch that requires you to hold the switch in for just a moment. It took only three engineers to figure out how to turn the thing on!

In order for the UPStart to correctly perform its Save feature, each application must have been entered into the UPStart's registry (not the Windows Registry) prior to operation (Figure 17). Actually, the UPStart has its registry stored and maintained by the Electronic Bookmark software (Figure 18), which works in conjunction with the UPStart.

Several applications are preset in the registry to handle automatic saves, including AmiPro, Lotus 1-2-3, Microsoft Excel, Notepad, Paintbrush, Quattro Pro, WordPerfect, Write, and Microsoft Word (Figure 17). However, we did have a problem with the Word entry in the registry. Although it was able to automatically save Word files, it didn't support long filenames at all. All the long filenames got truncated. We went into the Bookmark software and checked the option box for long filename support, but the Electronic

Bookmark software wouldn't allow us to change the settings for the Word entry. It notified us of an error stating "You are not allowed to modify a system entry!" Apparently the preset registry entries *cannot* be modified.

Ultimately, we had to create a new entry, call it Microsoft Word (same name as the system entry), and check the long filename support check box (Figure 19). Even then, the long filename support was limited. For instance, any long filename with a space character in it would be truncated. Thus, "this is a test of longfilenames.doc" would be truncated to "this.doc." But long filenames without spaces, such as "thisisatestoflongfilenames.doc," would work.

Several setup options are available. You a have a choice of serial ports, and you can set the interval between power loss and shutdown. Some of the options available in the Electronic Bookmark software are shown in Figure 20.

ROOM FOR IMPROVEMENT

The Electronic Bookmark could be improved to support long filenames. Also, the path where the software saves and restores is different from the original file. It saves the files under "C:\EB32\PF#," where # is the power failure number. (Accordingly, files saved after the first power failure would be under C:\EB32\PF1.) Thus, after a power failure (and a successful restore by the Electronic Bookmark software), you will have to perform a "Save As" to your original path.

This setup is probably intended to ensure that you have at least two copies of your files, in case of file corruption. We would have preferred the software to use the same path and filename as the original file(s). Alternatively, the software could have allowed us to decide how files are saved.

It is also possible that the UPS battery could die in the middle of saving, which would corrupt your file(s). You could surpass the battery's capacity if the battery is a few years old, or if you have large files open that take more than five

RATINGS (1-5)

Documentation: 3.5
Installation: 4.5
Features: 4
GUI: 4.5

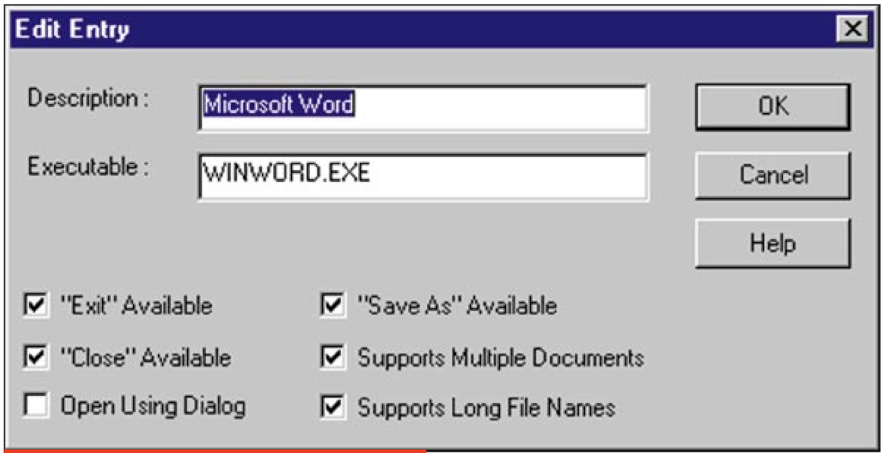


Figure 19. UPStart's Electronic Workbook: Registry settings. This screen displays our entries for Microsoft Word so we could enable long filenames.

minutes to save. This could happen if you are running graphics programs such as Adobe Photoshop or CorelDraw.

According to the documentation, Electronic Bookmark doesn't save unnamed files. Thus, working with files such as "Untitled-1" or "Document1" is unwise. Still, there are always computer users who will step away from their desks without saving their work. So, we decided to test unnamed files anyway. When we "pulled the plug" on unnamed Microsoft Word files, UPStart actually did save them, giving filenames such as "Document1," "Document2," etc. We suppose the Electronic Bookmark programmers designed the software better than they thought!

We could not get the Electronic Bookmark to work with Visual Basic 4.0, a popular CTI development software program. Since this program has multiple save options — Save File, Save Project, Save Project As, and others — you cannot use the generic "Save" call. We were awaiting a call from technical support for a solution that would allow us to work with Visual Basic, but we did not get a reply by the time this article went to print.

The Electronic Bookmark registry could have included more applications. We would like the preset list to include QuarkXpress, Visual Basic, C++, PowerBuilder, Visual Java, Visual C++, ACT, GoldMine, TeleMagic, and Adobe Photoshop, just to mention a few.

One final issue was the volume of the audible alarm. It wasn't as loud as we would have liked, but this was probably part of the design to conserve power. We did like how successive beeps sounded in closer succession as the battery drew close to losing all power. Still, a volume control on the unit would be a nice feature.

CONCLUSION

You can't afford to overlook power protection if you're responsible for maintaining CTI systems. CTI hardware, if damaged because of power spikes, can take a few days or more to replace. And, in mission-critical environments, downtime translates into significant financial losses. Of course, acquiring replacement parts is costly in itself. You could also spend a lot of time troubleshooting a computer that has been "spiked."

If you need a UPS for computer desktops, the SL Waber UPStart is a good choice, since it is a flat model, allowing you to place a monitor on top. For larger systems, which may include servers, PBXs, or equipment with huge

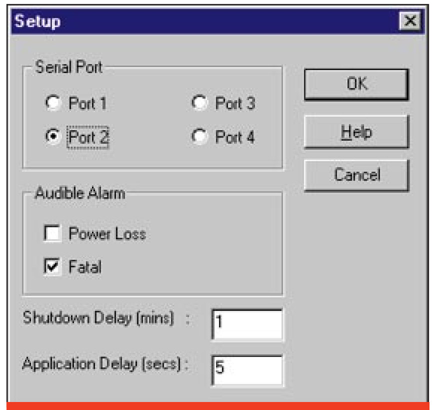


Figure 20. UPStart's Electronic Workbook: Setup screen.

wattage or power requirements, a larger UPS system may be better. But if you're looking for an economical, feature-rich UPS system made for the desktop, this product may be just what you need.

Users who would benefit from UPStart include computer-telephony application developers and programmers, as well as people who cannot afford to lose documents or other files. The UPStart will work on all platforms, but the Electronic Workforce software requires a Windows environment. It is now available in computer stores and retails for $199.

MasterConsole

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With multiple servers becoming commonplace, monitoring them all can become complicated. It isn't unusual for a company to have a fax server, fax-back system, IVR server, backup server, database server, Web server, and CTI development server together in one room, or even scattered across multiple rooms. (Although some companies will risk putting multiple functions on one server, most avoid putting all their eggs in one basket.)

Having a different monitor, keyboard, and mouse for each system is impractical. In any case, many companies like to perform remote monitoring, especially if they are running time-critical applications, such as computer-telephony applications.

If you have to deal with the many management and monitoring issues that arise in systems with multiple servers, you may be interested in Raritan Computer's MasterConsole, a compact computer-switching solution. MasterConsole controls any mix of 2 to 64 PCs. (You could, for example, control Macintoshes, Suns, and PCs all from one terminal.) MasterConsole can be used to manage network file servers, bridges and gateways, Internet servers, burn-in testing for PC production, hardware and software stress testing, systems integration, and access to applications running on multiple workstations, in addition to the other applications already mentioned.

Master Console can control computers running any operating system and any application software by simply plugging cables into the keyboard, mouse, and video port of each computer. Then, using one keyboard, one mouse, and one monitor hooked up to the MasterConsole, you can control all the computers.

DOCUMENTATION

The documentation consisted of several cabling diagrams showing connection schemes for various setups, such as connecting two computers, or daisy-chaining multiple MasterConsoles. The drawings were clear and easy to follow. The cables, which ship with the MasterConsole, are clearly labeled, and the labels can be matched against the cable diagrams to help you through installation. The other piece of important documentation, a sticker on the MasterConsole itself, denotes the hotkey definitions for performing channel switching and other tasks from the keyboard. This was a very handy addition to the unit.

The only drawback to the documentation was that it didn't have the hotkey definitions and instructions on how to use them. We looked at the MasterConsole sticker, which instructed us to press Scroll Lock twice to enter hotkey mode. When we tried this, nothing happened. After several frustrating tries, we discovered that we were not pressing the Scroll Lock key fast enough. We thought the documentation should have warned us to press the Scroll Lock key twice in a fairly rapid succession. The documentation earned an average rating of 3.

INSTALLATION

Installation was fairly easy. MasterConsole uses a PS/2 mouse connector and a PS/2 keyboard connection. Since we were using PC clones, we required adapters to connect our serial mouse and DB-9 keyboard to the MasterConsole. These adapters can be purchased directly from Raritan Computer. Using the adapters was a bit more complicated than just using a PS/2 type connector, but was still fairly easy. Overall, the installation earned a 4.5 rating.

FEATURES

MasterConsole comes in either a desktop or a rack-mount model that connects 2, 4, 8, or 16 machines. The unit can be daisy-chained to connect up to 64 computers and configured to add optional dual access up to 150 feet away, allowing you to monitor computers that are not in the same room. The ability to expand capacity up to 1,024 computers is available by cascading multiple MasterConsoles. Each connected computer can be remotely accessed from MasterConsole and its front panel or from the keyboard using hotkey commands.

The MasterConsole is "smarter" than a standard A-D keyboard/monitor switch. We have a couple A-D keyboard/monitor switches, but they often cause certain types of computers to crash. The reason is that these types of switches do not keep the keyboard signal continuous to each computer. When you switch from A to B, for example, the keyboard signal is cut from A and supplied to B. This is where some computers "freeze." With MasterConsole, you do not have this problem. You can be switched to any computer, and MasterConsole will provide a continuous keyboard and mouse signal to all the computers.

The continuous keyboard signal is important during reboots. In most instances, a computer BIOS will give you a keyboard error (such as "Keyboard Error. Keyboard Not Present") and fail to reboot until you plug a keyboard in. With "dumb" A-D switches, you must have the switch set to the computer you are rebooting for the keyboard signal to be present, or you will get the keyboard error. MasterConsole does not have this limitation.

The Auto Scan feature allows you to automatically monitor all connected computers at any time interval from 1 to 16 seconds. Auto Skip allows you to automatically bypass inactive ports.

MasterConsole ships with double-shielded coaxial cable for VGA, SVGA, and XGA video. Raritan Computer provides a generous supply of numbered stickers to help you label each of the cables.

Remote monitoring can be performed from the MasterConsole unit or from the keyboard itself. Tasks such as channel switching, bank switching, enabling Auto Scan, setting the delay time for Auto Scan, enabling Auto Skip, and others can all be carried out from the keyboard as well as from the MasterConsole unit.

OPERATIONAL TESTING

We tested 640 × 480 (VGA resolution) as well as 1024 × 768, and both worked flawlessly. The hotkeys, which allowed you to access the computers from the keyboard, worked very well. Some of the functionality from the keyboard included changing channels, changing banks, starting Auto Scan, and selecting channel by number. We liked the "bank number" and "channel number" illuminated indicators (large and easy to read) and the buttons to switch channels and banks from the MasterConsole unit (large and easy to manipulate). We also tested changing the hotkey to activate the MasterConsole and the time interval for the Auto Scan feature. Both performed flawlessly.

To test the Auto Scan feature, we scrolled through all the channel numbers, using MasterConsole's default setting, which let us stay on each channel for about a second. We found that this 1-second interval caused a problem for

RATINGS (1-5)

Installation: 4.75
Documentation: 3
Features: 4.75

monitors that have power-save modes. If you are not using one of the channels, or if the computer is off, then there is no video signal sent to the monitor, which causes a power-save monitor to shut itself down when the MasterConsole switches to this unused/computer-is-off channel. Then, when the MasterConsole switches to another channel, the signal to the monitor is cut off for a brief second, and the monitor has to power up again. By the time the monitor begins to power up, the screen is beginning to display, and then the next channel gets switched. Thus, you are left with a blank screen on every channel.

To get around this, you can increase the delay time. Or, you can turn on the Auto Skip feature, which will prevent the monitor from going into power-save mode. When Auto Skip is enabled, MasterConsole will always switch to a channel with an active computer (video) signal, and channels are changed in such a short interval (a millisecond) that monitors don't have time to go into power-save mode. We particularly liked the Auto Skip feature, since we didn't have to scroll through "blank" channels to get to the channels actually being used.

In most cases, you will probably want to leave Auto Skip on. The only exception would be if there is a computer failure, such as a power loss to the computer. If you leave Auto Skip off, you will at least see the monitor screen go blank, indicating a problem with one of the machines. Of course, you will need to be using all of MasterConsole's channels to use this method effectively.

Using the keyboard to change channels was pretty straightforward. You press the Scroll Lock key twice in quick succession. This puts the MasterConsole unit into a special hotkey mode. All the keyboard signals are sent through to the MasterConsole before they are passed onto the computer. Putting MasterConsole in this mode allows you to capture and interpret your keyboard responses to perform remote monitoring functions. You can also use the up and down arrow keys to switch channels, and the left and right arrow keys to switch banks.

Other features accessible from the keyboard include: setting the delay between Auto Scan switching, changing the activation key from Scroll Lock to Num Lock or Caps Lock, and selecting channel and bank numbers by entering the appropriate number.

MPG's Technical Support Helps Us Overcome Snag With NotifyMe's Audible Alarm

When we clicked on the button to change the sounds applied to each priority, the menu screen refused to pop up. So, we sent an e-mail about our problem to MPG's technical support. Here's how they responded:

When NotifyMe was written it was designed toward Windows 3.x (the standard at the time). When changing Sounds we called the sound module from the Control Panel instead of writing our own interface. We've seen this work on some Windows 95 machines and not work on others. We are rewriting that portion of the code since Windows 95 has taken over as the standard. Here are two options in the meantime:

1) If you have a PC with Windows 3.x on it copy control.exe from the Windows directory and snd.cpl from the Windows\System directory to your Windows 95 PC. This should allow NotifyMe to find the program to modify the .WAV files.

2) If you can't do 1) you can edit the WIN.INI file manually to change the sounds. In the [sounds] section you should find the three keywords below followed by the path to the .WAV file you want played. Change the .WAV file to the one you'd like played for each priority message.

For example:
[sounds]
NotifyMe1=c:\windows\chimes.wav, Notify Me (High Priority)
NotifyMe2=c:\windows\ring.wav,Notify Me (Normal Priority)
NotifyMe3=c:\windows\ding.wav,Notify Me (Low Priority)

We chose the second of the two options, and it worked fine. The original settings in the WIN.INI had blank entries where the sound files were supposed to be. So, NotifyMe1 had an entry equal to "NotifyMe1 = Notify Me (High Priority)." It was simple enough to manually add our own .WAV files to these parameters. ■

ROOM FOR IMPROVEMENT

There really is not much to be improved upon with this product. The documentation, however, should specify that you must press the Scroll Lock key very quickly to activate the remote switching from the keyboard. Also, the documentation should tell you that if you use the Auto Scan feature with a power-save monitor, you may need to enable the Auto Skip feature. Auto Skip will prevent the monitor from going into power-save mode, since it will always switch to a channel with an active computer (video) signal. We looked for a method of increasing the switching delay in the documentation, but none of the hotkeys were included. We did, however, find how to increase the delay from a sticker on the MasterConsole unit, which also contained all of the hotkeys used to control MasterConsole.

CONCLUSION

We found MasterConsole to be a great help in our CTI testing lab. We consolidated several monitors, and were able to use just one monitor in one section of our lab. This increased our work area, reduced our power consumption, and made testing and monitoring much easier. For testing computer-telephony equipment or for monitoring multiple servers, using a single console to monitor computer equipment is a must. Raritan Computer's MasterConsole is certainly a good choice. Prices range from $375 for a two-computer unit to $2,070 for a model which connects 16 computers.

NotifyMe

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NotifyMe, a Windows program which emulates an electronic display board, is designed to distribute messages to users across a network. It unobtrusively displays a scrolling marquee at the top or bottom of users' screens and provides a real-time communications link, which is particularly useful in call center environments, where up-to-the-minute statistics and messages are crucial. This functionality applies to other CTI applications as well, such as caller ID information, total hours logged by a particular agent, total hours for all agents, or current wait time. ACD queuing information is also handled by this program. Every operator can be aware of the number of calls in queue and of the average hold time for those calls. The program can even link to a user's voice mail system and display a message notifying the user that a voice mail message has been received.

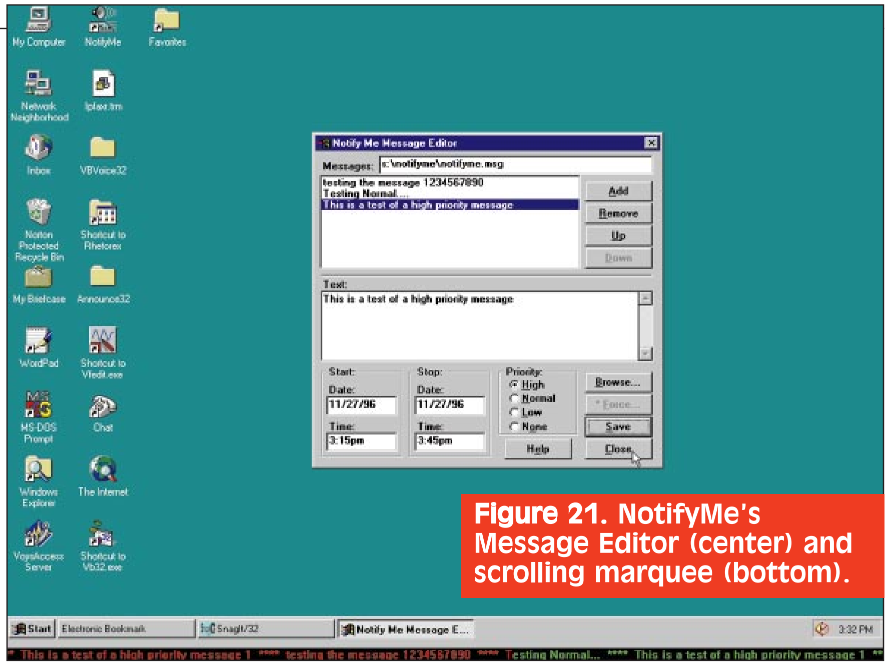

Figure 21. NotifyMe's Message Editor (center) and scrolling marquee (bottom).

While most companies encourage employees to communicate via e-mail, this medium does have its limits. Many e-mail packages audibly notify the user only once, when new e-mail arrives. Hence, a notify message is easy to miss. It is also easy to ignore. For example, a user may opt to turn off the notification pop-up message (often, this pop-up message interrupts work in progress). Also, a user may miss messages after logging out of an e-mail program and neglecting to log back on. Finally, a user may be flooded with a multitude of messages, which can cause a user to overlook a message that demands an immediate reply. Wouldn't it be great to announce that a meeting will commence in 15 minutes and have everyone see the message right away? NotifyMe can make it happen.

FEATURES

NotifyMe is network independent and will run on Windows 3.x., Windows 95, and Windows NT. It automatically notifies users when new messages arrive at their desktops. Users need not continually monitor a display board for additional information.

Messages can be prioritized, so users can determine whether they should address the message immediately or wait until they finish the call they are on. A priority of "None" will update the message but will not notify the user with an audible .WAV file. This feature is especially useful for displaying statistics or other frequently changing messages. Users are notified by two methods: the text of the message changes color (for example, red for high priority), and an audible .WAV file is played. The color scheme, including foreground and background for each priority level, is customizable.

NotifyMe allows you to distinguish between "obligatory" messages and optional messages.

For example, users may be obliged to receive such message types as "hot news," "system availability," and "statistics." In addition, users could opt to receive specific section and team messages, as well as birthday, trivia, or other noncritical messages.

Some other important features include:

• Customizing the audio .WAV file for each priority type.

• Specifying dates and times to start and stop messages.

• Setting the scroll speed of the marquee.

• Setting whether the message window is always on top of other application windows.

• Locating the message window on top or bottom of the screen, and whether it runs minimized and is only opened when new messages arrive.

• Selecting separator characters between each message, such as "****."

• Setting the number of times the audible notification is played.

• Setting whether the message windows will "jump" in front of the current application when a new message arrives, based on the priority of the message.

• Deciding which users should be able to receive messages, but not send them.

OPERATIONAL TESTING

NotifyMe stores its data in ASCII text files on a network server. Thus, all users with access to the network drive containing the NotifyMe message files can receive messages. In a LAN or WAN environment, any user can view any message files on the company server, based on the network file security.

NotifyMe is gratifyingly unobtrusive. Its notification system doesn't use a screen pop; instead, it uses a scrolling marquee at the bottom (or top) of the screen (Figure 21). NotifyMe does not take control of the user's system when a new message is received, even if it is set to pop to the front of any application the user is working on. Other network messaging systems tend to pop up a message and take control of the system until the user reads the message or clicks the OK button. With NotifyMe, the user can continue working until they are ready to read the message.

We liked seeing the messages change color after we clicked on them. We kept the default color scheme, which displayed all "read" messages in gray. We could tell at a glance if there were any new messages. Unread messages appeared in red (high priority), yellow (normal priority), or green (low priority). This scheme allowed us to gauge the importance of a message instantly.

Users can create messages through the message editor (Figure 21). As mentioned earlier, the receipt of certain messages can be made mandatory; others can be made optional. Mandatory files typically pertain to system outages, critical real-time information, payroll due reminders, or meeting reminders, to name a few. Other files (that is, subscription files) typically include team messages, department messages, individual messages, birthday wishes, morale boosters, trivia questions, and the like.

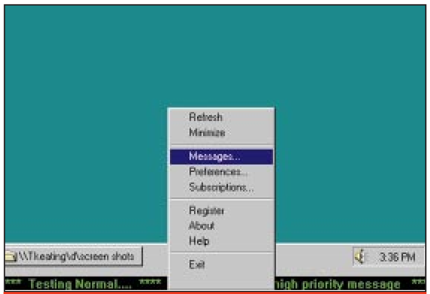

Figure 22. NotifyMe's right-click pop-up menu.

If you left-click on a scrolling message, you basically notify the program that you have read the new message. The program then changes the text color and stops playing any associated .WAV file. To avoid seeing old messages, you can set the foreground and background to the same color, rendering the old messages invisible. Holding down the left mouse button stops the message from scrolling until the button is released. A double-click on the scrolling message minimizes the window to an icon, and if a new message comes into the system, the marquee will either "flash" or "jump" in front of any current application, depending on the setup. A right click on the scrolling message will display NotifyMe's pop-up menu, where all the configuration and functionality of NotifyMe can be accessed (Figure 22).

To set the preferences, you right-click on the message window and choose "Preferences." You can then set text foreground and background color, size and type of font, speed of scroll, and location of scrolling message. Other options pertain to the refresh rate, whether messages should be "always on top," and the audible limit. You can also set notification types (have a window brought up front and/or have an icon flash) according to message priority (Figure 23).

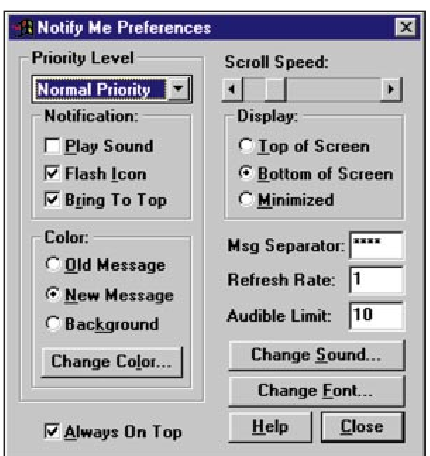

Figure 23. NotifyMe's dialog box for setting preferences.

This right-click context menu also lets you access the "Subscribe" option and define individual user subscriptions to message files. Users can subscribe to see messages from any NotifyMe message file, restricted only by the extent of their network access privileges. Files might include status.msg, callerid.msg, hotnews.msg, outages.msg, payroll.msg, and the like.

NotifyMe can also be used as a "reminder" or scheduling program. In lieu of a scheduling program, users can employ NotifyMe's calendar and messaging ability to set up appointments with reminders on a message file local to the user's hard disk. Then, users simply subscribe to this local file.

Some message files (for example, automated messages) needn't be created through the Message Editor. The NotifyMe message files are plain ASCII text files with several control characters to define the priority and time the message is to run. According to MPG Software, some companies set up automated tasks from their ACD and/or other systems to write a file out to the LAN every "x" number of minutes with phone or other statistics, which are then displayed as messages in NotifyMe. Automated messages, which are updated frequently, should be labeled with a priority of "None." This way, the user is not notified when the message is updated. Otherwise, the notification would soon lose its effectiveness, since the user would be notified continually of a new message every time the automated message was updated.

The NotifyMe program gives the user the option of displaying the marquee message at the top or the bottom of the screen. Opting to place the marquee at the bottom works fine with Windows 3.x. However, with Windows 95 (and NT 4.0 or later), the Start button and its accompanying task bar can get in the way. That is, if the task bar is set to "Always on top," which is the default in Windows 95, the marquee message will be hidden underneath the

Windows 95 task bar. If the "Always on top" setting for the task bar is turned off, the marquee may obscure the running programs (icons) on the task bar.

This display conflict can be overcome in several ways:

• Make the task bar a little larger, since the marquee has a minimal height. Then, programs will be visible on the task bar.

• Move the task bar to the top or side of the screen, leaving the marquee by itself on the bottom. (Most users prefer the task bar on the bottom, however.)

• Move the marquee to the top. (If the "Always on Top" option was turned on within NotifyMe, this solution can interfere with your ability to minimize and maximize programs. If this option is turned off, however, having the marquee on top is a good choice.)

• Turn on "Always on top" on both the task bar and within the NotifyMe program, and leave both the task bar and the marquee at the bottom of the screen. The marquee will hide underneath the task bar and only pop up in front of the task bar if there is a new message. There will be enough of the task bar showing to click on it to get back to the toolbar. (The user, however, will be unable to create new messages until the task bar is moved, since it is not possible to right-click on the marquee, which is hidden underneath the task bar.)

ROOM FOR IMPROVEMENT

If you neglect to fill in all the time and date fields, NotifyMe will make an announcement that, in effect, never expires. Such an announcement will become "read" when double-clicked, and will then appear in the color you use for read messages. We would have preferred that if the time and date fields were left blank, the program would send out a message with the current date and current time, and an end time of 1 minute later, so the message would expire right away. Such an arrangement would be good for quick announcements or messages, and it would save users the trouble of having to fill in every field.

Other messages are not meant to expire. For these, it would be nice to have the option of having the program automatically insert the current date and time into the date and time fields, which the user could then modify. Such an option could be implemented by clicking on a Refresh Date/Time button, or it could be implemented automatically, every 30 seconds or so.

RATINGS (1-5)

Documentation: 3.5 (online help)
Installation: 4.75
Features: 4.5
GUI: 4.75

Computers often lose time, or have their time set a few minutes apart from what an employee's watch or clock says. Thus, it's possible to send a message that has already expired, that is, a message stamped with a time that has already passed according to the computer (but not according to the user's watch or desktop/wall clock). Of course, such discrepancies aren't a problem if you run a program that displays the time kept by the computer. Nevertheless, we would like the next release to include a button that would allow users to insert the computer's current date and time into the corresponding fields.

CONCLUSION

This is a great product for retrieving and viewing real-time messages. NotifyMe is perfect for call centers, but can be used to advantage in other work environments as well. The open database structure allows for extensive customization options, including tie-ins to phone systems, CTI applications and products, and company databases. The unobtrusive messaging interface is probably the greatest asset of NotifyMe. This product is certainly worth a look.

NotifyMe will run on any Windows platform (3.x, 95, NT). Memory usage is 47K. Pricing is as follows: $19.95, individual user license; $299.95, single server license; $799.95, site license; $1499.95, corporate license. For evaluation copies, contact MPG software.

Heat 3.0

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Companies can now extend excellent customer service to any corner of the globe, and do so at any time, with Bendata's release of Heat for Windows Professional Edition 3.0 (Figure 24). Heat's "follow the sun" functionality is attributable to a new call ticket transfer feature. With the new feature, users needn't put a customer's problem on hold at the end of the business day. Instead, a service organization can transfer call tickets via e-mail to an open help desk in another time zone.

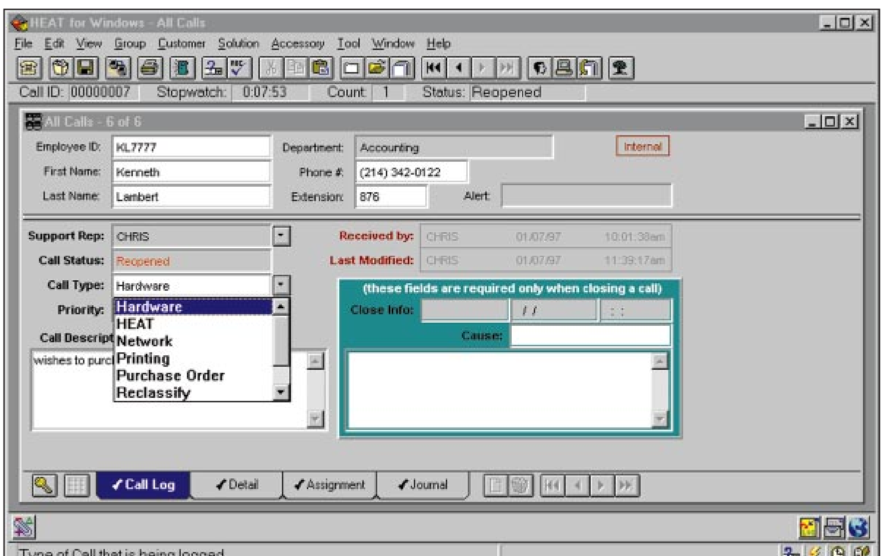

Figure 24. Heat's main screen and call log screen. Note that the call log shows information on a particular customer, as well as support codes in a drop-down box

Thus, the customer's problem can be addressed without delay. Heat 3.0 makes service even more immediate by dispatching service requests to field technicians via mobile communications.

The client/server software delivers true end-to-end operations, from the initial call through problem resolution. In addition, Heat grows more capable with use because it continually (and automatically) adds new problem-solving information to its online knowledge base.

DOCUMENTATION

The manual had separator tabs that broke the text into main sections. An index, however, was conspicuously absent. There was no proper table of contents. Nevertheless, the quality of the documentation was good, and there were plenty of screenshots throughout. Overall, the documentation earned a 3.95 rating.

INSTALLATION

Although most of the installation was easy, we did have a problem with the Crystal Reports module. The documentation recommended installing Crystal Reports under the Heat directory, which in our case was the C:\HFW\CRW directory. Thus, we clicked (in sequence) on "Set Location," the C:\HFW\CRW directory, the "OK" button, and the "Continue" button. As a result, we got a general protection fault in DSHELL in module _MSTEST.EXE at 0004:000033c6. We rebooted, ran SCANDISK, and retried the installation from the beginning. And again we got a general protection fault (GPF).

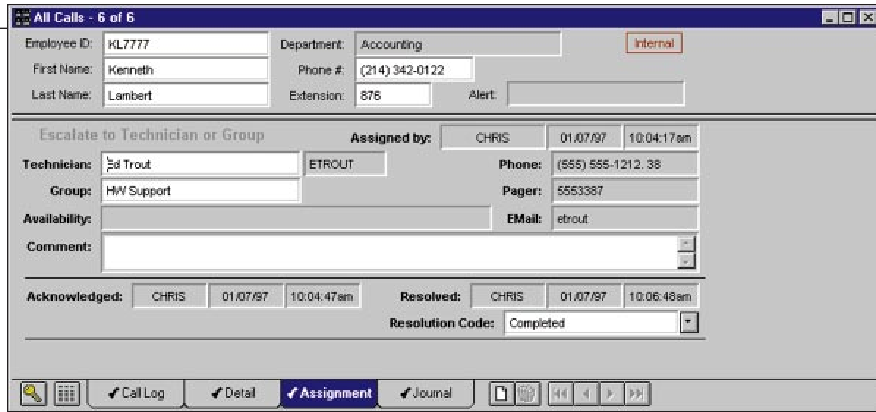

Figure 25. Heat's assignment screen. From here, you can assign technicians and groups and apply other relevant information.

On a whim, we typed the path in the text box instead of browsing for the directory. Then, we clicked on "Continue." The installation continued without giving us the GPF error. (This was one of the strangest bugs we had seen in a while. You could type the directory in, but you couldn't "browse" for it.)

The installation proceeded smoothly until we reached the 19th disk (out of 22) on the CD-ROM. The computer froze. We decided the crash was due to a faulty CD-ROM and a buggy 16-bit CD-ROM driver. After four retries, we were able to complete the installation.

Overall, we gave the installation a 3.90 rating. This rating doesn't reflect our problem with the CD-ROM; we cannot fault the software for some system hardware problems. Instead, the rating is due to the bug that caused the system crash, which led us to reboot and reinstall twice.

FEATURES

Remote Access

Mobile Heat, the software's new remote access capability, enables field service technicians to communicate with the main help desk via laptop computers and modems. With Mobile Heat, technicians can download call tickets from the main help desk, work on the call ticket offline, and upload changes to the main system later. The database is updated automatically. Thus, Heat's remote access capability streamlines the dispatch system, ensures more accurate call tracking, and enables field technicians to resolve problems more quickly.

Database Capabilities

Heat 3.0 has expanded its online knowledge base capabilities by integrating with CasePoint, Inference's case-based reasoning (CBR) product. In addition, the software's built-in knowledge base engine, called First Level Support, is now enhanced with auto-find and auto-populate capabilities. These technologies bring more knowledge to the desktop, enabling service personnel at all levels to easily find answers to customer problems.

Inference's CBR significantly reduces problem resolution time by automatically initiating a solution search based on key words as the call is being logged. When a solution is found in the knowledge base, Heat instantly notifies the user. Clicking on the notification icon pulls up a series of questions the user can ask the customer to find the cause of the problem. This enables even new customer service technicians to find answers quickly, reducing the amount of training needed.

Heat 3.0's auto-find and auto-populate capabilities make it easier to resolve problems. Auto-find searches the knowledge tree for possible solutions while a call is being logged. Auto-populate automatically adds new solu-

HeatLink To The Internet

Bendata's new communications module for Heat, HeatLink to the Internet, provides an interface between Heat and your Windows NT server. Your customers will be able to open, close, and update call tickets and check on the status of a ticket via the Internet. The product promises to reduce the number of phone calls to your help desk, shorten customer on-hold time, and cut down phone bills.

If you deploy this module, your help desk customers will be able to:

- Access their own call tickets. (All other tickets remain secure.)
- View all of their open calls.
- View any updates added by the help desk technicians.
- Enjoy the speed and convenience of opening calls over the Internet.
- Add pertinent information to call tickets by entering a journal entry at their convenience.
- Close calls over the Internet, if they resolve the problems on their own. ■

Figure 26. Heat's screen for implementing user preferences.

tions identified during a call to the knowledge tree, providing on-the-fly knowledge creation.

In addition to the internally created knowledge base, Heat 3.0 can search external knowledge base sources from Microsoft's TechNet, Novell's Network Support Encyclopedia, Micro House's Technical Library, ServiceWare's Knowledge Plus, and Lotus's Knowledge Base. Heat's open architecture and ODBC compliance allow compatibility with SQL databases (such as Oracle, SQL Server, and Watcom) and file/server databases (such as Access). Heat 3.0 ships with Crystal Reports Professional 5.0, a popular reporting package, which includes at least 100 pre-formatted reports.

CTI Features

Bendata is working to integrate its help desk software with Aurora's FastCall and Q.Sys's Phoneware. These enhancements will give Heat several CTI capabilities, including caller ID callup of a customer's call history. If the caller ID doesn't match any record, Q.SyS prompts the user to enter a customer number or call ticket number. Thus, you get the best of both worlds: caller ID record matching and caller-entered identification. If the caller has neither an account number nor a call-ticket number, the system automatically starts recording a new call history.

Heat allows you to monitor hold time. In addition, you can track hold time with Crystal Reports. Monitoring and tracking can be performed in such a way to confirm that service agreements are being met. With Q.SyS's software functionality in addition to Bendata's Heat, you can monitor extra items, such as number of rings before a call is answered, adding further tracking of performance and improving customer service as well.

Some of these CTI features are available now, and others will be available before the end of the first quarter of 1997. Still other CTI enhancements are planned for Heat 4.0, which will be released in early 1998.

Management And Reporting

Heat provides the Support Center Manager with tools to better track, manage and forecast support center performance and productivity. Heat's Statistics Monitor gives you an up-to-the-minute snapshot of your help desk statistics, such as the total number of open calls and calls not modified within service level agreement requirements, so you can better manage your staffing levels. As previously mentioned, Crystal Reports (the support center industry's leading report writing software) is also included, giving management instant access to vital statistics.

Other Features

With Call Locking, users can indicate to each other that work on a ticket is already in progress. This feature prevents users from making duplicate calls to customers who are already getting help (Figure 25).

With File Attachments, users can provide more information about a call without resorting to cutting and pasting information. Users simply attach documents, sound files, pictures, etc. to call tickets.

With the Calls On Hold feature, a help desk user can place a call in progress, but not ready to be saved, in a "Calls On Hold" group, storing it locally on the user's disk drive.

OPERATIONAL TESTING

Customizing Heat is easy: you simply enter your preferences via the user-friendly GUI. The menu tabs were particularly convenient (Figure 26). To help you get a head start configuring your system, Heat includes several built-in screen templates compiled from actual help desks and support center systems. These sample templates cover a variety of industries, including finance, software, and education (Figure 27).

We found that Heat sets up an intuitive call flow and makes accessing a customer's profile simple. If your customer's problem requires hands-on intervention, you can easily assign the call to an appropriate technician. You can print the call ticket, send a Heat message to another Heat user, send an e-mail to any VIM- or MAPI-compliant e-mail system, or page a technician directly from Heat. To ensure that your Help Desk is responding within your customer's service level agreement requirements, Auto-Escalation will automatically escalate a call to the next level if necessary.

The Auto-Find feature searches the knowledge bases in the background. Once you have found a solution, you can post it to the call ticket with a push of a button. Also, with Heat's journal features, you can record time- and date-stamped information for each customer

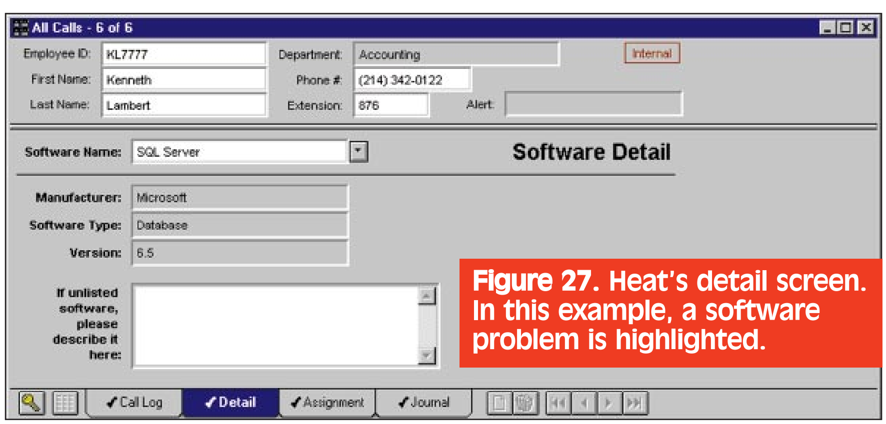

Figure 27. Heat's detail screen. In this example, a software problem is highlighted.

(Figure 28). You can automatically add new solutions into your knowledge base with Auto-Populate.

ROOM FOR IMPROVEMENT

We puzzled over why we had to manually type in the path during installation. We suppose a bug in the software is to blame. If so, we'd like to see that fixed in the next release.

Another suggestion we have is to add a column to the Open Group screen. This column could display the record count for each group, giving you a quick idea of the status of each group.

We liked Heat's support for external knowledge bases, but we were a little surprised the product didn't include one of them. Help desk software packages usually include one free knowledge base.

RATINGS (1-5)

GUI: 4.30
Documentation: 3.95
Installation: 3.90
Features: 4.75

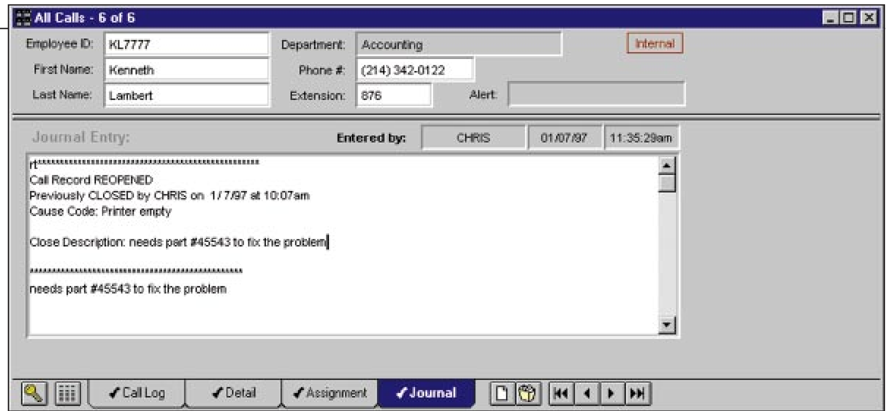

Figure 28. Heat's journal screen, which is used for attaching notes and other important information.

A general limitation is that Heat 3.0 is a 16-bit application and uses 16-bit ODBC drivers. The Crystal Reports software is also 16-bit. Until a 32-bit version of Heat becomes available, performance won't be everything it can be for users with Windows 95 machines.

Finally, a future release might include some of the following features:

- A means of collecting and managing customers' technical support e-mails. (These could be attached to the journal.)
- Support for Web-based forms customers could use to enter and submit a preliminary problem description. (Data gathered from these forms would then be incorporated into a Heat call ticket, which a help desk agent would use when contacting the customer.)
- Integration with Internet telephony, which would allow a help desk agent to call the customer back over the Internet, using the e-mail address entered by the customer as a means to establish the Internet phone connection. (Note: Most Internet telephony packages now support e-mail addresses to establish an Internet call.)
- Complete tutorials to familiarize new users with the product.

CONCLUSION

Heat's main strength is its openness with respect to external databases; that is, Heat doesn't limit you to a specific or proprietary database. The product's open approach is evident in another respect: the software is shipping with the industry-standard Crystal Reports.

Bendata is currently working or partnering with a few CTI vendors to add or enhance computer-telephony features in the Heat 3.0 product. At present, Heat's ODBC-compliant database helps ease the integration process with CTI products, since several CTI products are also ODBC-compliant.

The ease with which users can change and customize screens, in addition to the product's ability to integrate with popular knowledge bases, make this help desk software a good choice. To use Heat, you need a Microsoft Windows NT server (which needn't be the primary server) and an Internet information server (supplied by Microsoft Windows NT Server version 4.0). A browser that supports tables is recommended. Pricing ranges from $3,000 for a single user to $50,000 for an unlimited number of users. ■

ARIEL
4/C

On-Site Briefing

Resource Architecture

Dialogic Corporation
1515 Route Ten
Parsippany, NJ 07054
Ph: 201-993-3000; Fx: 201-993-3093
Web site: www.dialogic.com

Dialogic Corporation has developed a new computer-telephony (CT) resource architecture. According to Dialogic, the architecture exemplifies openness in its overall approach, and has the flexibility and performance to enable the rapid integration of multiple resources from multiple vendors.

The impending architecture is designated DM3. We first heard about it when representatives from Dialogic visited the offices of CTI Magazine for an on-site briefing. This article is the result of that meeting. Our thanks go to John Landau (vice president, Strategic Marketing and general manager, Dialogic Architecture Lab) and Marc Felsen (product marketing manager) for taking the time to inform us of this initiative from Dialogic.

OVERVIEW

Years ago, the computer industry was based on expensive, limited, proprietary systems built upon proprietary computing resources. Open systems developments in computing and data resources eventually enabled diverse systems built on open, multiple-vendor, best-of-breed resources. The CT industry of today is undergoing similar changes; in fact, a transition is underway to move away from the limited, proprietary systems toward open CT systems, which will also offer best-of-breed multivendor resources and telephone protocols.

Dialogic is hoping to provide a new level of reusable and extensible CT resource capability aimed at integration within CT systems. To that end, Dialogic has developed the DM3 resource architecture for creating board-level CT resource solutions. This resource architecture is designed to provide for independent, firmware-based network protocol and media-processing resources that will operate and integrate together upon compatible hardware platforms. The hardware platforms will range in scale and will reside within multiple types of open computing systems.

According to Dialogic, the DM3 architecture is a result of the single largest research and development undertaking in the company's history. With the new resource architecture, Dialogic hopes to provide competitively priced, scaleable, high-performance products to open-component original equipment manufacturers (OEMs), systems integrators (SIs), and value-added resellers (VARs). The development kits are designed to enable resource technology developers to rapidly create and integrate DSP- and RISC-based firmware resources for deployment on a common set of hardware boards or platforms. DM3 development kits will also offer an open development environment to proprietary technology OEMs, who can then create solutions integrating their own value-added resources with complementary commercial resource components.

FLEXIBILITY AND PERFORMANCE

DM3's modular multiprocessing software architecture is designed to create hardware flexibility by enabling firmware resources to be independent of the underlying hardware. The intent is to permit modular upgrades and modifications in DM3 hardware to support newer processors and memory devices, and to do so without requiring changes to existing applications or disruptions to firmware resources on other parts of the

DM3 Terminology

The DM3 resource *architecture* enables technology resource portability across multiple DM3 platform implementations.

DM3 *platforms* are modular, scalable, single processor and multiprocessor hardware implementations of the architecture and include high-density PCI, CompactPCI, and VME platforms. DM3 platforms are integrated with DM3 firmware resources to create *product bundles*.

DM3 *firmware resources* are implementations of network protocol and media-processing resources that use one or more of the DM3 platform processors, integrating together through the use of the DM3 software kernel.

hardware platform. Furthermore, since the firmware resources are designed to be portable across the DM3's hardware implementations, all DM3 resources should be available across all supported platforms, including PCI, CompactPCI, VME, as well as others that Dialogic plans to add in the future.

According to Dialogic, technology resource developers can create or port algorithms onto the DM3 architecture to build technology resources, while solution developers can create host applications that use the DM3 technology resources. The technology resources are embedded firmware algorithms, such as voice, fax, and automatic speech recognition (ASR), that are written so as to leverage the DM3 software kernel for interprocessor management and resource cooperation. (The DM3 kernel is embedded software that is common to all processors on the DM3 architecture.) DM3 supports multiple technology resources from multiple developers, and lets solution developers build interoperable solutions based on Signal Computing System Architecture (SCSA) SCbus hardware and SCSA software APIs or other APIs. Dialogic says DM3 cards will interoperate with other standards-based CT resource cards.

DM3's host software interfaces begin with a single low-level "direct interface," and layers above this interface will provide compatibility for multivendor/shared platform ECTF (Enterprise Computer Telephony Forum) S.100-based applications as well as some proprietary platform applications. Above this direct interface, DM3 is backwards compatible with applications developed on Dialogic's GlobalCall and 4.x voice APIs.

According to Dialogic, the initial DM3 architecture implementations will be high-performance PCI, CompactPCI, and VME system bus boards targeted toward complex and high-density applications. Real-time resource bus connection between multivendor network interface and media-processing resources is provided by the ANSI/VITA6-1994-compatible SCbus. While there will be a variety of specific products, the company anticipates the basic multiprocessor board will be configured to support 0, 1, 2, or 4 T1/E1 digital trunk interfaces (with ISDN PRI), from 0 to 1,800 fixed-point DSP MIPS computing capacity (up to eighteen Motorola 56030x processor units), and from 0 to 1,280 SPECint 92/SPECfp92 specmarks of RISC computing capacity (up to eight Motorola PowerPC 603e processor units), supported by a variety of memory configurations. Furthermore, Dialogic expects the DM3 firmware resources to be implemented on these DM3 hardware platforms to include store-and-forward voice processing (for voice messaging and IVR), fax, speech recognition, digital and analog network interfaces, real-time voice over IP, and SS7. Dialogic will support UNIX (initially Solaris) and Windows NT host operating systems, and a porting kit is expected to become available to enable porting to other operating systems and host computers.

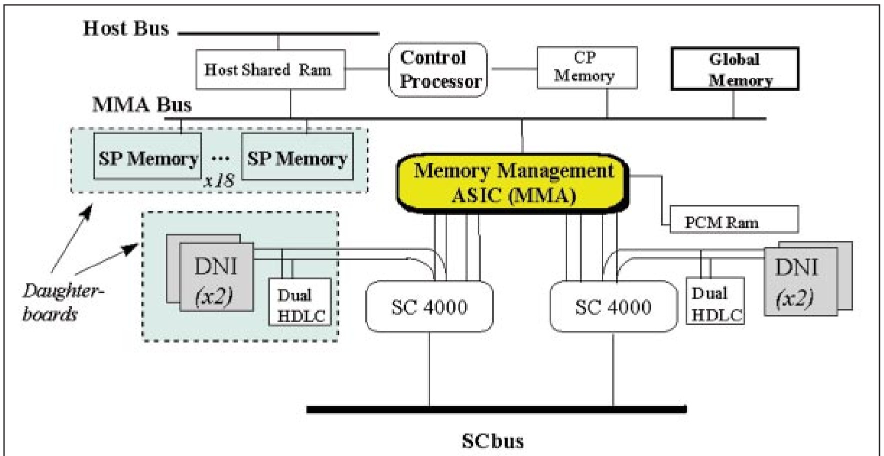

Figure 1. DM3 high-density hardware configuration.

HIGH-DENSITY HARDWARE PLATFORMS

The DM3 high-density platforms consist of three baseboard types: PCI, CompactPCI, and VME. These baseboard types support daughterboards for signal processing and computing, communications for development (Ethernet and RS-232), and additional network interfaces (T1, E1, or ISDN). As illustrated in Figure 1, the DM3 baseboard contains an Intel i960CF control processor (CP), associated RAM and flash memory, two complete network interfaces, two SC4000 ASICs configurable global memory, host RAM for communication between the CP and the host, and a memory-management ASIC (MMA). The MMA transfers all bulk data between the host RAM, CP, SPs, and SC4000 ASICs via a 32-bit DMA (direct memory access) bus to and from the DM3 global memory. (The global memory is installed on SIMM or SO DIMM modules for ease of upgradeability and is available in different sizes up to 16MB.) The MMA also compacts, expands, and moves PCM data between the SC4000 ASICs and the various processors' memories via the PCM buffer.

The single CP on the DM3 baseboard manages SCbus access and controls the SC message bus via the MMA and the SC4000 ASICs. Data on the SCbus is available in multiples of 64-Kbps TDM channels (bundles), in linear format or encoded in A-law or μ-law format. The two SC4000 ASICs can access up to 256 of the 2,048 SCbus time slots, allowing up to four T1, E1, or ISDN trunks with echo canceled data that can be exported to other technology resources, such as ASR.

Each SP daughterboard is configured with up to six Motorola 5630x DSPs or up to four PowerPC 603e processors. Dialogic says that other signal processors can be used on SP daughterboards, and daughterboards with different processors can be combined for maximum flexibility. The same SP daughterboards can be used on PCI, CompactPCI, and VME baseboards. The communication daughterboard contains 10 Base-T Ethernet and RS-232 serial port interfaces, which can be used for debugging or direct access to the board during development. The digital network interface (DNI) daughterboard provides two additional T1, E1, or ISDN network interfaces to supplement the two network interfaces on the baseboard. The SP daughterboards are stacked apart form the other daughterboards so as to permit up to three SP daughterboards, a communication daughterboard, and a DNI daughterboard to be stacked on a single baseboard. A baseboard with a single SP daughterboard requires only one PCI, compactPCI, or VME slot. A baseboard with two or three stacked SP daughterboards requires the space of two slots.

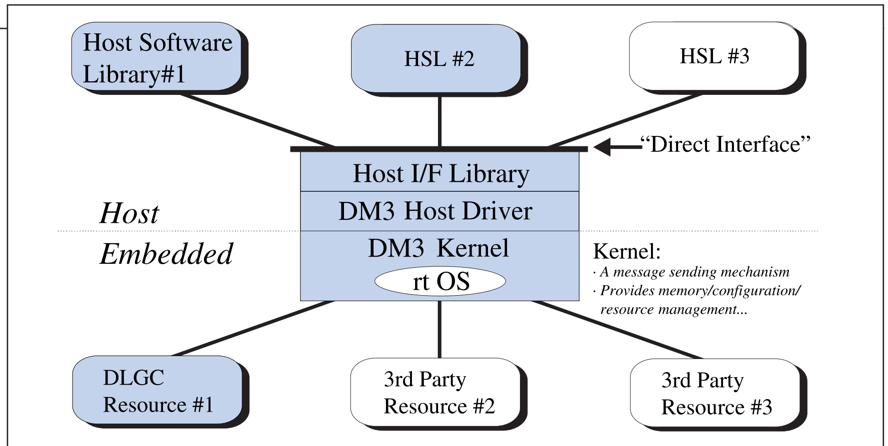

Figure 2. Overview of DM3 software architecture showing layered and modular design.

SOFTWARE ARCHITECTURE

The DM3 software model (Figure 2) consists of host software that runs directly on a PCI, CompactPCI, or VME target system and embedded software that runs directly on the DM3 platform. This layered and modular software design includes a host device driver, a host interface library, the DM3 kernel, underlying real-time operating systems (RTOSs), and technology resources. The RTOSs used beneath the kernel include SPOX (for DSPs, such as the Motorola 5630x) and VxWorks (for RISC, such as the i960 control processor and PPC603e signal processors). According to Dialogic, using industry-standard operating systems and tools will help reduce the time it takes to port technology resources and develop solutions. The DM3 kernel offers runtime services within and among the CP and SPs, and maintains a communication path between the DM3 technology resources and the host software. The kernel provides a message-sending mechanism and timer services, plus resource, configuration, and memory-management services.

DM3 technology resources are embedded firmware algorithms that run on the hardware using the kernel services. Resources can be divided into processor-specific applications, such as a player resource containing various types of decoder components. Components are assigned unique DM3 addresses and communicate with each other via messages that are routed by the kernel.

On the host software side, DM3 driver support is a technology-independent transport layer linked to a DM3 host interface library. The device driver passes messages and data from the host to the resources on the appropriate DM3 board using the functionality of the kernel. Host software libraries, such as a voice library, communicate with technology resources through the DM3 host interface library. Host software libraries are shielded from the device driver by the DM3 host interface library, providing protection from differences in host operating systems. To ease communication between the host and technology resources, the host interface library enables driver configuration, component management, message transfer, and data read/write operations.

COMPATIBILITY AND AVAILABILITY

DM3 is a Dialogic initiative to complement and improve the company's current product lines. DM3 products from Dialogic and resource development partners will be introduced in stages. DM3 will address a host of CTI issues:

- Quad digital telephone interface
- Quad voice with quad DTI
- Quad DTI with SS7
- Fax
- Real-time voice over IP
- ASR
- Text-to-speech (TTS)

Initial availability of the DM3 resource architecture and related products will begin in the first quarter of 1997.

For information about free subscriptions, call our customer service department at 800-243-6002 (toll free) or 203-852-6800, or visit our Web site at www.ctimag.com. Contact the publisher, Richard Tehrani, or the editor, Kevin M. Mayer, with questions or comments about *CTI* ™. E-mail (addressed to rtehrani@tmcnet.com or kmayer@tmcnet.com) is always welcome. ■

DM3 Compatible Resource Technologies

- Voice mail/voice messaging
- Audiotext
- Advanced intelligent network (AIN) applications on scalable IPs
- Call processing, including:
  - Call progress
  - Voice record and play
  - DTMF detection
  - Tone generation
  - Speed and volume control
  - Noise tolerance
  - Cut through
  - Talk off plus other SpringWare features
- Signal analysis for tone detection and call progress
- Digital network interfaces (T1, E1, or ISDN)
- Fax
- Automatic speech recognition (ASR)
- Text-to-speech (TTS)
- Audioconferencing
- SS7 protocols
- Real-time voice over IP

CTI™ Buyer's Guide Update

The information in this update was supplied to CTI magazine after the publication of the 1996 CTI Magazine Buyer's Guide (volume 1, number 2). Please keep a copy of this update with your Buyer's Guide issue.

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Fax: 512-928-1575
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CT EXPO PREVIEW

Computer Telephony Expo '97
March 4–6, 1997
LA Convention Center,
Los Angeles, CA
Call 212-691-8215 x2
Visit www.ctexpo.com

CT Expo '97, the CTI industry's biggest tradeshow, is fast approaching. If you haven't already registered to attend the show, we urge you to consider doing so. Nowhere will you find so many CTI vendors and products in one place.

To give you a sense of what the show's exhibitors will have on display, we've prepared this brief editorial. Below, we present information collected from a selection of vendors who will have a strong presence at the show. In addition, we present information from vendors who approached us with announcements about what they planned to emphasize at CT Expo '97.

Brooktrout Technology, Inc.
410 First Avenue
Needham, MA 02194
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Ph: 617-449-4100
hjm@brooktrout.com
www.brooktrout.com

Brooktrout Technology, a provider of electronic messaging products, will display and demonstrate its complete line of products at CT Expo '97. Among the products displayed will be Brooktrout's Show N Tel, a development environment for enterprisewide, client/server voice, fax, and computer-telephony applications. Show N Tel is designed as an open platform to allow integration of applications in heterogeneous telephony

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and data environments. Other showcased products will include the IP/FaxRouter, an embedded fax system for routing fax over IP networks such as the Internet; QuadraFax, a fax-on-demand and fax broadcast system; and the TR Series and TruFax fax and voice processing boards.

Brooktrout will also host a theater where its ISV partners will showcase their latest products. Brooktrout plans to introduce new products in the application development, platform, and peripheral markets as well.

Comdial Corporation
1180 Seminole Trail
Charlottesville, VA 22906
Contact: April Owens
Ph: 804-978-2287
www.comdial.com

Comdial is expected to showcase a variety of CTI solutions at CT Expo '97. Among the products that Comdial offers are:

• Quick Q ACD, an automatic call distribution (ACD) product for small to medium-sized call centers;

• Easy Track, a descendant of Comdial's Tracker, which is an on-site integrated paging system developed in cooperation with Motorola;

• Comdial's InnTouch Concierge Hospitality Package, which allows users to visually control their voice mail using a PC; and

• Comdial's CTI solutions wideopen.group and wideopen.office.

The wideopen.office product is a telephony server that allows TSAPI applications to run in most popular LANs. Together with wideopen.group, a CTI groupware application designed for collaborative call processing among workgroups, these products are compatible with a full range of operating systems including OS/2, Windows NT, and Windows 95.

Comdial is committed to standards,

including those being set by Novell (TSAPI) and Microsoft (TAPI).

Commetrex Corporation
6400 Atlantic Boulevard
Suite 190
Norcross, GA 30071
Contact: Martin Lippman
Ph: 770-449-7775 x370
mlippman@commetrex.com

At CT Expo '97, Commetrex will feature its first hardware product, the Media Stream Processor (MSP). In 1993, Commetrex introduced MultiFax, a third-party voice board add-in to bring software-based multiline fax to the computer-telephony OEM. The company is still emphasizing the benefits of what it calls media integration.

Commetrex believes that companies with different media-processing competencies developing stream-processing products for CT resource boards with standardized board-level environments will usher in the next architectural phase of computer telephony. The MSP is intended to be the first such media-neutral CT hardware resource. According to Commetrex, the MSP will support voice, fax, high-speed data, speech recognition, text-to-speech, and video, but Commetrex only intends to develop fax and data media-processing products for the board. Commetrex will make the board's software environment available to other developers in order to develop support for the other media.

Computer Telephony Expo '97
March 4–6, 1997
LA Convention Center,
Los Angeles, CA
Call 212-691-8215 x2
Visit www.ctexpo.com

Dialogic Corporation
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Contact: Tracy Sarkees
201-993-3000
www.dialogic.com

At CT Expo '97, Dialogic Corporation, an industry-leading manufacturer of high-performance, standards-based computer-telephony components, will feature leading-edge products, services, programs, and application solutions focusing on AIN/wireless and Internet-based voice and fax solutions. In addition, Dialogic will showcase enterprise solutions, which feature large switching applications and response systems, as well as highlight unified messaging and information-on-demand solutions. Dialogic's customer base includes value-added resellers (VARs), original equipment manufacturers (OEMs), service providers, and applications developers, ranging from small systems integrators to global telecommunications companies.

Ericsson, Inc., Business Systems
7001 Development Drive
PO Box 13969
Research Triangle Park, NC 27709
800-431-2345
www.ericsson.com

Ericsson, one of the world's largest suppliers of telephony networks, provides enterprisewide CTI solutions that integrate Ericsson's Consono MD110 PBX into IP networks and information systems creating comprehensive productivity increases and cost savings. Consono is Ericsson's combined offering of integrated communications systems for voice, data, and multimedia in private networks. Consono is a combination of systems, products, and services bringing all types of information together in an end-to-end network based on open standards, aimed at working together with unified network-management abilities and smooth migration into future technologies. The Consono MD110 PBX software supports applications such as a common directory for end users on the desktop, as well as ISDN-based multimedia such as Proshare multiparty videoconferencing. The core platform for solutions such as voice communications, personal applications, call center, and personal mobility, the Consono MD110 PBX, is based on separate modules for switching functions and more advanced services that can be customized to suit customer needs. Applications (call centers, screen-based telephony, directory, mobility, system and network management, messaging, and videoconferencing) are all integrated through the use of various interfaces (TAPI, TSAPI, NT-TAPI, and CSTA).

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Kanata, Ontario
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K2K 1X3
800-MITEL-SX
www.mitel.com

This year at CT Expo Mitel will highlight a pair of Developer Programs: developers@work and the Mitel MediaPath's Development Environment. These programs offer tools and support to design CTI applications for Mitel's SX-2000 LIGHT and SX-200 LIGHT PBXs and client/server telecom platforms. Mitel will continue in its commitment to support the computer-telephony developers by providing open standards-based application-developing environments.

Mitel Corporation designs, manufactures, and markets systems, subsys-

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tems, and microelectronic components for sale to world markets in the telephony, computer-telephony integration (CTI), and communications industries. The company's products include voice communications systems; public switching systems; network-enhancement and gateway products; CTI systems and applications; client/server telecommunications products; custom silicon wafers, integrated and hybrid circuits, and optoelectronic devices. Mitel's leadership strategy is centered on advancing people-to-people communications in an open, distributed, and standards-based environment.

Natural MicroSystems
8 Erie Drive
Natick, MA 01760
Contact: Vicky Stoneback
NMS Sales Hotline: 800-533-6120, Menu Option 2
www.nmss.com

Natural MicroSystems will host two exhibition booths at CT Expo '97. In one booth, the company will demonstrate its CT Access development platform, a series of APIs and telephony extensions for Windows NT designed to reduce time-to-market and increase functionality of applications built on the company's Alliance Generation family of DSP boards. Natural MicroSystems will also showcase a new benchmark for capacity and function in a DSP board: the Alliance Generation Quad T1 provides four complete T1 interfaces and 96 ports of integrated voice, fax, and speech recognition in a single PC slot.

Natural MicroSystems has invited 22 strategic partners to share their second exhibit booth. Here the partners will demonstrate their services and products, which add value to the Natural MicroSystems product line.

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Some partners expected to demonstrate their products include:

• Atlas Telecom – A U.S. integrated voice mail system developer;

• Big Sky Technologies – A U.S. unified messaging systems developer using text-to-speech, showing their Internet telephony mailbox;

• Group W – A U.S. application-generator (app-gen) tools vendor;

• Kemtec – A Singapore-based app-gen tools vendor;

• Mastermind Technologies -– A U.S. app-gen tools vendor; and

• Open Access – An Australian Systems Integrator.

NEC America, Inc.
Corporate Networks Group
1555 Walnut Hill Lane
Irving, TX 75038
800-TEAM-NEC
www.nec.com

In addition to its NEAX family of voice platforms (PBX) and ELECTRA family of key telephone systems, NEC will demonstrate a variety of CTI and other integrated applications, including switched video and integrated ATM voice applications at the CT Expo. ACD techniques (including screen pops) and IVR integration, as well as several individual user applications, such as unified messaging, will be demonstrated with availability for hands-on usage by attendees at the conference. These are all components of NEC's FUSION integrated environment, which is designed to integrate diverse hardware and software elements for CTI functionality.

As well as designing and manufacturing PBX systems and software, NEC America Corporate Networks Group develops audioconferencing, videoconferencing, key telephone systems, multimedia applications, data communications, and other products.

Parity Software
One Harbor Drive, Suite 110
Sausalito, CA 94965
Contact: Sales Department
415-332-5656
www.paritysw.com
sales@paritysw.com

Parity Software, a leading supplier of software tools and hardware components to computer-telephony systems integrators, VARs, consultants, and developers, will demonstrate its telephony power tools for building computer-telephony applications (for example, voice mail, fax-on-demand, international callback, and the like). These tools include VOS, a telephony applications language for Windows NT, UNIX, Windows 95, Windows 3.x, and DOS, and CallSuite, a family of ActiveX controls for building applications in Windows NT and Windows 95 in popular visual development environments such as Visual Basic, Visual C++, Delphi, PowerBuilder and other VBX/ActiveX hosts. Parity will also host a pavilion where Parity VARs will showcase applications powered by Parity's tools.

Parity Software's primary business is to create and deliver full-featured, industrial-strength software development tools. In addition, Parity Software is an Authorized Dialogic Toolkit vendor offering Dialogic boards and all major Dialogic-compatible hardware products. This relationship with Dialogic gives Parity access to new hardware products early in the engineering cycle, which offers Parity the opportunity to add support for new features to software tools in time for hardware product releases.

Computer Telephony Expo '97
March 4–6, 1997
LA Convention Center,
Los Angeles, CA
Call 212-691-8215 x2
Visit www.ctexpo.com

Rhetorex, Inc.
151 Albright Way
Los Gatos, CA 95030
Contact: Stephen Weekley
Ph: 408-370-0881 X1110
stephen.weekley@octel.com
www.rhetorex.com

Rhetorex, Inc., will host a partners pavilion at CT Expo'97, where companies such as TTM and Associates and Vcomm will showcase real-life solutions using RealCT. RealCT is a high-level computer-telephony API (application programming interface) for PC-based telephony components from Rhetorex. RealCT is implemented as a Dynamic Link Library (DLL) in Windows NT. The configurable API shields developers from the complexities of telephone connections and CT resource switching. The product is compatible with a number of other Windows application objects, such as Microsoft Foundation Classes and Active X.

The inclusion of a built-in MVIP resource manager enables the use of RealCT applications in a variety of analog and digital telephony environments. A companion software utility, the RealCT Resource Wizard, enables developers to configure applications for specific telephony environments. Applications based on RealCT can be ported from analog telephony boards to digital boards without the need for any source code changes.

A wholly owned subsidiary of Octel Communications Corporation, Rhetorex manufactures components

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(hardware and software) to connect PCs and LANs with telephone networks.

Sun Microsystems
2550 Garcia Avenue
Mountain View, CA 94043
Contact: Karen Richards
Ph: 415-336-2814
www.sun.com

Sun will highlight Java-based customer management and CTI solutions at CT Expo '97. In addition to showcasing its SunXTL Teleservice platform, which provides multi-client and multi-device support, Sun will provide an update on the Java Telephony Application Programming Interface (JavaTel API). Bringing Java's "Write Once, Run Anywhere" platform independence to a heterogeneous environment, JavaTel runs on top of the SunXTL architecture.

As SunXTL and JavaTel support multiple clients and multiple devices, Sun will be demonstrating the JavaStation, Sun's network computer, as a front-end desktop for call centers and customer management systems. Sun's embodiment of the thin-client concept, the JavaStation, is aimed at reducing administration and maintenance fees over traditional "fat-client" PCs, while providing comparable functionality.

For information about free subscriptions, call our customer service department at 800-243-6002 (toll free) or 203-852-6800, or visit our Web site at www.ctimag.com. Contact the publisher, Richard Tehrani, or the editor, Kevin M. Mayer, with questions or comments about *CTI*™. E-mail (addressed to rtehrani@tmcnet.com or kmayer@tmcnet.com) is always welcome. ■

INTERNET TELEPHONY:

What happens when you mix all the hype about the Internet with the rapidly growing market for CTI? The answer: Internet telephony. Slogans like "Talk forever for free" have caused some to predict the doom of the long-distance carriers. Others suggest that Internet telephony's future is contingent on a mere regulatory aberration, that is, cheap local access.

ANDREW SEARS,
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Others approach the issue from a more technical perspective. For example, Vint Cerf, vice president of MCI and "father" of the Internet, has predicted that packet switching may overtake circuit switching — a concept that has been termed "Cerf's Inversion."

So, many people, including people in the CTI industry, are left wondering how to separate reality from all the hype. How significant is the Internet, really? This article will explore the possibilities for the near term (easy to hype) and the long term (easy to overlook).

WHAT IS INTERNET TELEPHONY?

Internet telephony (IT) in its current, limited form is typified by one Internet user with a computer calling another user, with both parties using special software that allows for voice transmission through the Internet. Configuring a computer for calls can be difficult, and sound quality is unpredictable. For calls placed in the United States, quality is often equivalent to that obtainable in a cellular phone call, but with slightly higher delay. Quality for international calls varies widely.

If Internet telephony were to stop there, it would be easy to dismiss the nascent technology. But recent developments indicate there may be more to Internet telephony than first meets the eye. For example, several companies (VocalTec, Netspeak, Analogics, Lucent) are promoting "phone gateway" products that use CTI to allow the placement and receipt of Internet

Separating Facts From Hype

calls over regular telephones. Initiatives such as these highlight what is really interesting about Internet telephony, particularly with respect to CTI. Put simply, CTI is the interface between the Internet and the PSTN.

CTI has the potential to connect hundreds of millions of telephones on the PSTN and private networks to tens of millions of computers on the Internet. Will this potential be realized? There are at least four points of view:

1. The Hobbyist View: Until recently, hobbyists were the primary market for Internet telephony, largely because none but the enthusiast would be willing to put up with the poor sound quality and difficulty of use. A few people predict that IT will remain in this domain, and will be seen as a passing fad like CB and HAM radio. (Examples: early versions of Internet Phone, Webphone.)

2. The "Talk For Free" View: Here,

Internet Telephony Interoperability Consortium

The Massachusetts Institute of Technology has organized a group of researchers in academia and the computer and telecommunications industries to provide a basic understanding of the technical, policy, and economic implications of Internet telephony — the use of the Internet for real-time fax, voice, and video transmissions. The think tank, based at MIT, is called the Internet Telephony Interoperability Consortium, or ITI.

"No one is really sure right now how this technology will be regulated or what financial model will be used," said Lee McKnight, who is principal investigator for the Consortium and Associate Director for Research and Management at MIT's Research Program on Communications Policy (RPCP). "So the idea behind ITI is to bring people from different industries and backgrounds together to remove the barriers to growth of Internet telephony and multimedia businesses by developing unbiased and supportable information and technology."

The growing list of Consortium members includes such diverse companies as Sprint, Nokia, U.S. Robotics, Lucent Technologies, Mediatrix, VocalTec, Telia, Netspeak, Natural MicroSystems, Telecom Italia, and American Communication Services, Inc., (ACSI). The Consortium incorporates work on a range of issues presented by Internet telephony, including pricing of reliable services, regulatory issues, and business effects. The core concept of the Consortium is interoperability, the ability for applications and services to work between systems.

Interoperability will be best understood, McKnight said, by researching the intersection of telecommunications services and enhanced Internet services. The Consortium also plans to address the critical technical and politico-economic factors affecting infrastructure investment related to the deployment of voice services over the Internet, innovation strategies, and load management.

Specifically, the research focus of the project is divided into five areas:

• To work with research partners to enable interoperability among Internet telephony applications.

• To collaborate with member companies to explore the opportunities for increasing interoperability between the Internet and the public switched telephone network (PSTN).

• To perform economic analyses of the deployment of voice services over the Internet including: cost evaluation, pricing, and yield management models for the traffic originating from Internet telephony and related services.

• To perform studies of Internet telephony regulatory issues.

• To use this information to develop new business models and assess adaptations of old business models (this analysis includes assessing information to aid decisions for infrastructure development).

To learn more about the ITI, visit the ITI Web site at http://itel.mit.edu/. This site includes meeting overheads, articles, and links to networking standards, phone software, and news.

Housed within MIT's Center for Technology, Policy, and Industrial Development, the RPCP has its foundation in communications policy, from its work with high-definition television to research on Internet economics. The Program comprises researchers from such diverse backgrounds as management, engineering, and economics. Along with the formation of the ITI, a recent success for the program has been the development of the MIT-Polaroid high-definition PTC-9000 camera. ■

the focus is on the ability of the Internet to deliver "cheap" long-distance. Advocates of this view promote "phone gateways" which allow Internet telephony calls to be placed and received with regular telephones. The primary market is for international long-distance, which is likely to see some type of regulation. (Service examples: IDT's Net-to-phone, GXC, Delta3, and AlphaNet; hardware/software examples: telephony gateways.)

3. The Value-Added For Computers View: This view focuses on the multimedia potential of Internet telephony with a computer as the user interface (with telephone) providing audio/data/video conferencing, whiteboards, and application sharing. Advocates emphasize that the Internet provides options for functionality that are unavailable through telephony alone. In this camp, functionality overshadows issues such as cost or portability. (Examples: Microsoft's NetMeeting, VocalTec's Conference, Netscape's CoolTalk.)

4. The Value-Added For Telephony View: This is similar to the previous view, except that it focuses on the telephone as the user interface because of its ubiquity, portability, and low terminal cost. Services might include Internet-enabled voice mail, linking the Web and e-mail to telephones, and possibly providing speech recognition through the Internet. Future services might include server-side support for personal digital assistants. (Examples: VocalTec's Gateway, Netspeak's gateway, Web-to-Phone, e-mail readers, etc.)

WHERE IS THE OPPORTUNITY?

Now that different views of the market are clear, the next question becomes "Who can make money and where?" While this question has no easy answer, a few generalizations can be made. Clearly, the "hobbyist" view is a limited opportunity, and with Microsoft and Netscape giving away their Internet telephony clients, this market does not have much appeal.

While many have ignored the talk-for-free view, it is likely to have potential in some areas. My own research at MIT has shown that the average price in the United States for an Internet telephony call is $.03 cents per minute, as compared with over $1 per minute through the PSTN. For international calls, Internet telephony can offer customers considerable savings. While some argue that this is simply an arbitrage opportunity, it can represent a legitimate business plan in some cases.

Probably the key resource needed to enter the service market for consumers will be regulatory finesse. The business market may not have the regulatory problems, but it is questionable whether the quality of Internet telephony will be adequate for many businesses (except those on private Intranets). For years, Micom has been providing businesses with private networks the ability to combine their voice and data networks to cut costs. IT only makes this easier, while also adding advantages of interoperability with the Internet.

For many, the value-added-for-computers view has the most appeal. It has all the ingredients of a high-potential marketplace: the opportunity to provide obvious value, a clear business focus, the potential for high margins, etc. It seems to be the perfect market — except for one major problem: Microsoft has clearly targeted multimedia conferencing as a priority. With the release of its free client, NetMeeting 2.0, Microsoft will have the lead in functionality. Netscape, IBM/Lotus, and Intel all are also focused on this market, and there are already over two dozen other clients currently available. The conclusion: while this market may have the most potential, major players have already recognized the opportunity and are likely to capture the value, which suggests other players should focus on secondary products (compatible with those of market leaders) or on providing services.

The final area — providing value-added telephony — might hold the most potential for the CTI industry. Players in this market will need an understanding of the telephony, computer, and networking/Internet industries. Those in CTI already understand telephony and computers, which is likely to give them an advantage over other entrants. One key obstacle is that some CTI professionals still view CTI in terms of call centers and voice response applications, which is a limited view of the potential that the Internet brings.

HOW DOES IT DIFFER FROM CTI?

Another way of phrasing this question is, "What does the Internet enable that is not possible with computer telephony alone?" One obvious answer is that CTI applications on different computers across the world can talk to each other. This makes possible a global voice mail network similar to the global e-mail system. Another technology enabled by the Internet is multimedia conferencing and all the applications it brings. A third difference is a concept I developed in my research on Internet telephony: the Internet can *decouple* CTI software from hardware, meaning that software running anywhere on the Internet could provide CTI services.

DEVELOPMENT OF IT

One may wonder why there has been so much hype about Internet telephony. To look past the hype and put Internet telephony in perspective, it helps to remember that people often overestimate the potential for change over the short term and underestimate its potential over the long term. While Internet telephony may not be much now, in the long term it has the potential to revolutionize both CTI and the PSTN. All the same, Internet telephony still requires major developments before it can begin to reach its potential. Some of issues faced by IT include:

1. Regulation. America's Carriers Telecommunications Association (ACTA), a lobbying group for long-distance resellers, filed a petition requesting the Federal Communications Commission (FCC) to ban IT software. While the FCC chairman has given strong signals that the FCC will not act on the petition, other regulatory issues still exist, such as access charges. Computer-to-computer IT calls are the least likely to be regulated, while phone-to-phone IT calls might see regulation, especially if they use the PSTN for the local loop. The main deterrent to IT development is the prospect that future regulation will be unfavorable to Internet telephony. The uncertainty is discouraging investment.

2. Interoperability. While it appears that there is industry consensus on H.323, interoperability remains an issue. H.323 leaves many points unresolved, so Internet telephony software providers have formed a group called VoIP to resolve short-term issues. The Internet Telephony Consortium at MIT will do research on long-term interoperability issues, and input that research into the IETF.

3. Reliable QoS On The Internet. Routers enabled for a new protocol called RSVP are currently in production, and should be commercially available soon. These routers will provide a mechanism for reliable QoS on the Internet, but it will take some time until they are widely deployed. It is likely that they will be used in private Intranets before they are used in the public Internet. Researchers in MIT's Internet Telephony Consortium were key in the development of the RSVP standard.

4. Internet Growth. If the Internet is to handle a large volume of telephony traffic, then it will need to grow significantly in capacity. As capacity grows, so will reliability. It is likely that international lines will represent a key bottleneck.

5. New Pricing For Reliable QoS. Before reliable quality of service (QoS) protocols like RSVP can be deployed, a new pricing mechanism needs to be developed. It is important to distinguish this issue from the debate over flat-rate versus usage base pricing over existing protocols. Research in this area is one of the key goals of MIT's Internet Telephony Consortium.

USING IT WITH CTI

So, should you wait for the definitive resolution of all of these developments before you integrate the Internet into your CTI system? No! None of the above issues needs to be resolved for many Internet telephony applications, especially those which do not require real-time QoS.

As matters now stand, with these issues unresolved, the business market will likely find reliable QoS limited to private Intranets — at least for the next couple of years. Nonetheless, the long-term effects of Internet telephony could bring dramatic changes to the CTI industry. Thus, keeping current on developments in Internet telephony will be an abiding concern to CTI.

Andrew Sears is a researcher at MIT's Internet Telephony Consortium (http://itel.mit.edu), and helped initiate the consortium. He can be reached at asears@mit.edu or (617) 225-7129.

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ISDN GETS DOWN

BY VICKI FULLER,
SIEMENS BUSINESS
COMMUNICATION
SYSTEMS, INC.

Increasingly, small businesses and home-office "solo entrepreneurs" are turning to ISDN (Integrated Subscriber Digital Network) to satisfy their office telecommunications needs without resorting to patchwork systems of multiple (and incompatible) phones, fax machines, answering machines, and Internet-access vehicles, and the associated tangled mess of cabling and wires. These businesses, in addition to taking advantage of the obvious speed benefits of ISDN, are enjoying the benefits of digitally integrated video, voice, and data. The digitization of voice and video is contributing to the proliferation of ISDN, and has allowed ISDN to begin delivering on its promise of unifying data streams in a hassle-free, cost-efficient environment.

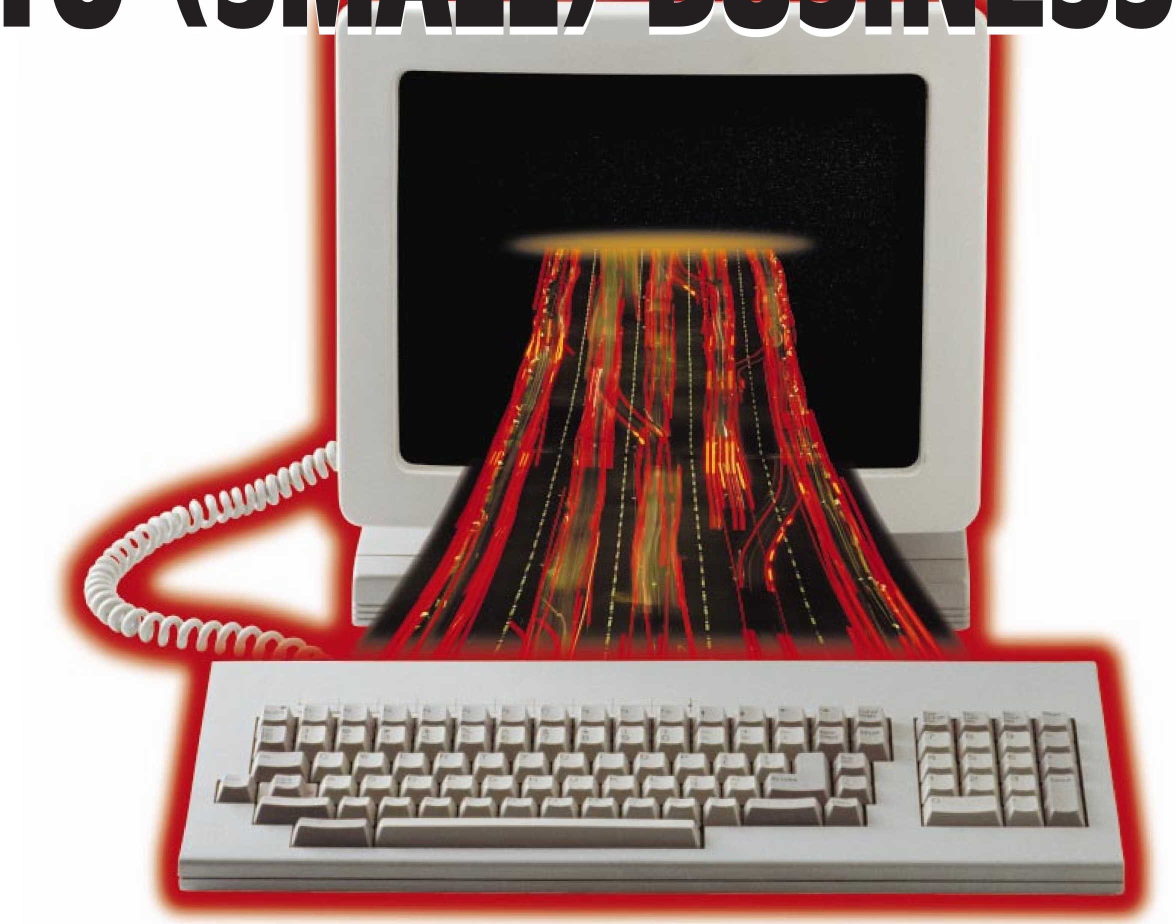
A Small Office Approach From Siemens

Siemens OfficePoint is an ISDN BRI NI1 system designed for 2 to 16 lines. Now compatible with all central office switches in the metropolitan United States, OfficePoint connects up to four ISDN devices, such as PC cards, network bridges/routers, Group IV fax machines, and videoconferencing applications without separate NT1 adapters. Its automatic bandwidth allocation allows a small office to use instant-connecting digital voice, high-speed data, and analog applications simultaneously. Users save on monthly line charges by reducing the number of central office lines needed; a flexible numbering system allows up to 24 individual numbers on one system. Four analog interfaces in the OfficePoint system handle Group III fax machines, answering machines, and cordless phones. ■

Smaller businesses usually opt for the Basic Rate Interface (BRI) form of ISDN. A single BRI line has two 64-kbps bearer (B) channels and a 16-kbps signaling (D) channel that can transfer digitized information at speeds up to 128 kbps. An ISDN-equipped small office provides Internet or Intranet file transfer that's 5 to 10 times faster than the current 28.8-kbps dial-up modem.

What costs might a business encounter with BRI? A single BRI connection that accommodates two standard analog telephone lines as well as the digital link to the Internet is billed (by Pacific Telephone, for example) at very little more than two standard analog lines. A relatively inexpensive ($300–400) terminal adapter (TA) connects computer and analog lines to the

TO (SMALL) BUSINESS



ISDN line. One phone line is typically dedicated to a fax setup and the other to a voice phone and answering machine.

THE DIGITAL SMALL OFFICE

In any ISDN setup, a TA is required to take the analog output of office equipment and digitize it for high-speed transmission across the ISDN line coming from the telephone company. This device, generally called an ISDN modem, together with a network-termination device (NT1) to terminate the ISDN line safely, are the only new telecommunications equipment normally required to connect to an ISDN circuit. If the line is dedicated to a single function or if there is only one person working in the office, a simple device with TA and NT1 in a box with a couple of analog plug-ins for a fax and an answering machine should suffice.

But in a group where 5 to 20 or more people work together (such as in a medical office or the local office of a larger enterprise), ISDN can provide a wide range of services when it arrives at a low-cost switch integrated with the TA and NT1. It becomes most cost effective when three BRI lines are connected to the switch. There are now systems on the market offered by major U.S. manufacturers that help small-business users achieve the kind of speed and flexibility usually associated with a corporate headquarters, with its PBX and mainframe database. With digital telephone sets on up to sixteen desks, four fax lines, and four lines dedicated to computer–data connections, this sort of system enables powerful, cost-effective ISDN applications.

By plugging a high-speed ISDN data line into a computer, the small business office can implement commonly used computer-telephony applications, including screen pop, where the incoming automatic-number identification (ANI) triggers a database lookup and brings up a caller's record during the first ring, and computer-assisted dial-out.

But when the ISDN modem is coupled with a PBX switch, a new set of options arrive. Familiar PBX features, such as call forwarding, conferencing, camping on, following, and distinctive ringing are available at every desk that has one of the system's digital telephones. Videoconferencing can be installed and used by anyone.

The key benefits of ISDN for small business office telephony almost all relate to speed and convenience. On a TA + NT1 switching system every telephone can have its own number. There's no need for an operator; everyone answers their own phone or forwards their own calls (or has their calls sent to a voice mail server). Direct inward dialing to each station is programmable by the master phone set. Connection setup is immediate. Caller ID transmission is automatic. The digital phones have a point-and-click display panel for choosing functions that are available during

The Personal PBX

DAVID FRANKEL, JETSTREAM COMMUNICATIONS, INC.

When it comes to telecommunications, home-office professionals no longer have to operate at a disadvantage to their corporate-office counterparts. A new category of products are now available that leverage the full potential of ISDN Basic Rate Interface (BRI) telephone service to provide home-office professionals with a set of communications capabilities equal (or even superior) to those found in most corporate offices. These products allow a home-based professional to be as productive, and appear as professional, as their corporate-based colleagues and competitors.

THE MARKET

Until now, home-office professionals have been forced to assemble cumbersome telecommunications environments using a number of products and services. A typical home office includes two or more telephone lines, multiple telephones, fax machines, modems, answering devices, and more. In addition, these products are often not designed to work together, and are frequently in conflict with one another.

To make matters worse, home-office professionals have no assistants or colleagues on site to help them manage their communications when they're already on the phone or out of the office. For example, no one is available to forward a critical fax to a hotel when they are traveling. Despite their best attempts, home-office professionals still face significant communications challenges, including:

- Professionally managing multiple callers.
- Managing the priority of inbound calls.
- Being selectively reachable when out of the office.
- Receiving prompt notification of waiting messages.
- Accessing faxes remotely.
- Having fast data links for accessing the Internet or remote corporate LAN.
- Ensuring reliable and simple operation.
- Keeping communications charges in control.

These challenges are becoming more mainstream as the number of home-office professionals increases at an accelerating pace. Governmental mandates for clean air and reduced traffic congestion, corporate initiatives aimed at employee retention and reducing costs, and initiatives on the part of individuals toward greater personal freedom are all driving this trend.

WHEN MARKET NEED AND TECHNOLOGY MEET

Just as the market for home-office products is reaching critical mass, so is the acceptance of ISDN BRI residential telephone service. Although ISDN telephone service has been around for many years, it is only recently that price reductions and increased availability have made ISDN service a viable technology upon which to base a home-office product. Demand for faster Internet access has been a key driver in the acceptance of ISDN.

A few companies have recognized this opportunity and have created a new category of ISDN-based telecommunications products designed specifically for this market: the "personal PBX." Unlike most ISDN-based products, which are designed to meet only the data-communications needs of the home-office market, these new personal PBXs leverage the full potential of ISDN BRI service to also provide advanced voice and fax communications capabilities.

Many people are familiar with the sophisticated data communications capabilities of ISDN — fast, easy data communications to the Internet or a remote corporate network. However, few are aware of ISDN's advanced voice and fax call-management capabilities. Nevertheless, these once overlooked features of ISDN are now enabling the development of personal PBXs.

THE PERSONAL PBX

A personal PBX acts as a communications hub for all the voice, fax, and data communications traffic in a home office. Because ISDN supports multiple, simultaneous voice, fax, and/or data calls on a single line, a personal PBX eliminates the need for multiple phone lines. With a Personal PBX, a user can be on the telephone and still receive a second call, send a fax, or log onto the Internet.

Typically, a personal PBX supports a single ISDN BRI interface to the telephone company, two or more analog POTS (plain old telephone service) ports for connecting the user's existing telephone and fax machine, and a serial communications port for connecting a computer. (See the accompanying graphic, which illustrates the typical home office before and after the introduction of a personal PBX.)

Some of the other unique capabilities of ISDN that personal PBXs leverage include:

• ***Multiple Telephone Numbers:*** Unlike analog telephone lines, a single ISDN line can support up to 64 telephone numbers. Now, a home-office professional can cost-effectively have separate telephone numbers for the various types of calls received, such as personal, business, fax, and priority. Having multiple telephone numbers allows a home-office professional to more effectively manage inbound calls and eliminates the need for workaround devices such as analog voice/fax line detectors. Based on the telephone number dialed, a personal PBX can route inbound calls to different locations. For example, calls to a or before a call, and this same panel displays the caller's station number. Also, the phones have several bays built in that accept optional analog modules for private lines and faxes.

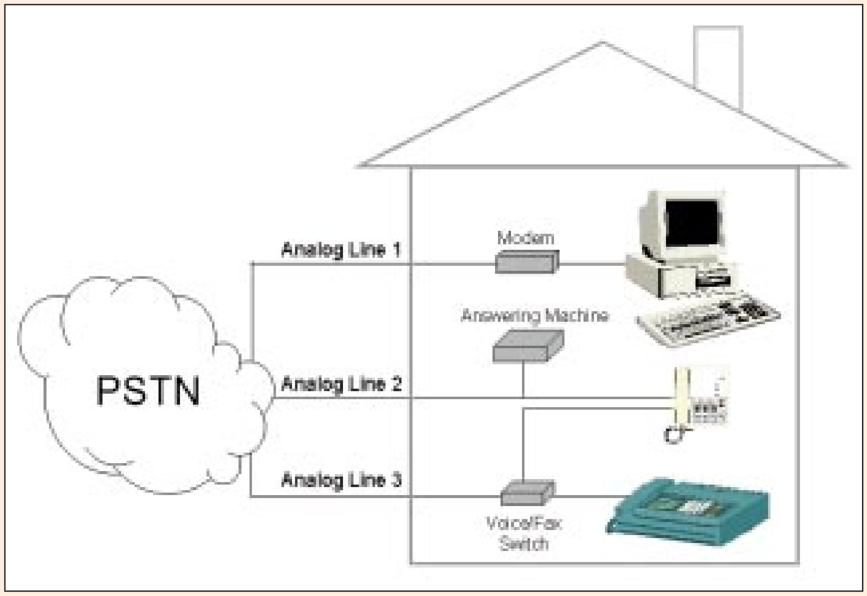
But the advantages of using an ISDN modem/switch for the small office are most obvious in comparisons between the costs of going digital and the costs of getting the same general level of performance from 56-kbps leased lines for Internet and Intranet connections, individual phone lines to each desk, videoconferencing circuits, and the software and training required.

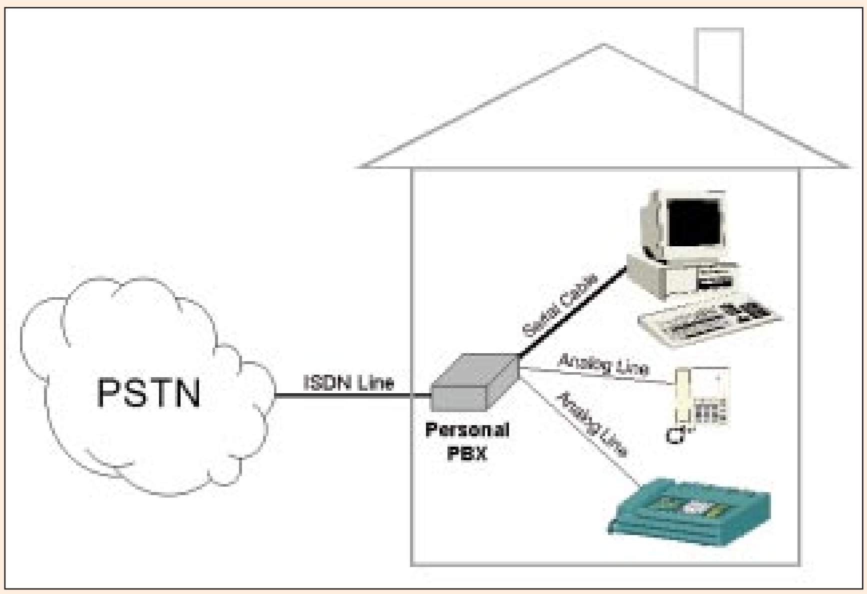
user's fax number are sent to the fax machine while calls to the business number are directed to the telephone, and if not answered, then to voice mail.

• ***Advanced Call Handling:*** Personal PBXs typically take advantage of the advanced call handling features inherent in ISDN. These features include call transfer, six-party conference calling, and 16-caller hold. Depending upon how the personal PBX presents these call handling features to the user, they can be more easily utilized than from many expensive digital display phones available on a corporate PBX. Personal PBXs can provide easy access to features that let a home-office professional manage calls more effectively.

• ***Caller ID:*** To further assist a home-office professional in managing in-bound calls, personal PBXs can take advantage of caller ID information made available through ISDN. A personal PBX can be instructed to route calls differently based on the caller ID information provided. For example, some personal PBXs can automatically forward a priority caller to the user's cell phone when they are out of the office, while all other callers are immediately forwarded to voice mail.

• ***Deterministic Signaling:*** Because ISDN uses the D channel for signaling between a personal PBX and the telephone company's central office, it allows for very efficient call processing of advanced features, such as call transfer and conference calling, as well as more sophisticated applications, such as "Find Me." The Find Me feature of some personal PBXs will automatically transfer certain callers to other telephone numbers (such as the user's cellular number) when the user is out of the office. Because ISDN uses deterministic signaling, the personal PBX knows the exact outcome of each attempted call transfer, such as busy or ring no answer, and handles the call accordingly. This type of service would be slow and unreliable if conducted with analog service where the personal PBX would have to listen to the line and try to interrupt the outcome of the call by what it hears (for example, ringing, busy tone, talking).





CONCLUSION

By leveraging ISDN for more than just high-speed data communications, these personal PBXs make ISDN service a cost-effective solution. For home-office professionals who rely on telephone, fax, and data communications, ISDN BRI service with a personal PBX can turn communications into a true strategic advantage.

David Frankel is president and founder of Jetstream Communications, Inc., which makes sophisticated communications products for the home office. Jetstream's products bring "corporate-level" communications to the home office. For more information, contact Jetstream at 408-777-4333 or visit the company's Web site at www.jetstream.com. ■

THE DIGITAL ARCHITECTURE

There is a simple distinction between using an ISDN modem only for Internet connectivity and mating it with a digital switch. In the first instance, the office remains essentially analog throughout, with a high-speed capability for uploading and downloading files over the Internet. In the second, the entire office becomes digital. Voice transmissions are digitized as they enter the handset, rather than being transmitted as analog signals to the central office, to be digitized later.

Once voice or fax is digitized, it can be switched, parked, packed, treated, and moved around with no degradation through electronic gates instead of being sent through mechanical switches. There are no sounds or pictures — only digits, a speeding stream of ones and zeros. This transition from analog to digital is the key to bringing point-and-click convenience to the small office.

ISDN GETS DOWN TO (SMALL) BUSINESS

In this section, we will briefly review a couple of real-life ISDN implementations. Both reflect the opportunities that are opening up for small business as ISDN becomes more accessible.

ISDN HELP FOR H.E.L.P.

Kim Martin manages a five-person office for H.E.L.P Suffolk in Yaphank, New York. Ms. Martin and her fellow counselors help the unemployed find work and the homeless find shelter in this relatively rural area of Long Island. Besides counseling clients, her operation offers practical help, including a place for them to work and phones for them to use for job and housing searches. Staff and clients also share an office fax machine, which seems to be busy most of the time.

Ms. Martin recently asked Interworks, a New York ISDN installer, to upgrade the office phone system. H.E.L.P. now has four lines, three of them ISDN circuits, and new digital handsets sitting behind a TA/switch combination. By using one 3-BRI connection and an analog line for the fax, Ms. Martin is now able to handle incoming calls at her desk or program the phone to send them to one of her associates when she's busy. At the same time, the system handles calls for the 15–20 clients she serves on an average day now. H.E.L.P. has 10 office desks set aside for people looking for work and housing. All are equipped with high-quality ISDN digital phones that Ms. Martin says are very clear and free from static — she and her four associates can now easily conference on important calls, speed dial, forward calls, and, "Most importantly," she added, "put people on hold nicely." According to Ms. Martin, installation took only a day — once Nynex was able to get ISDN to her location.

ISDN FOR AN ISDN DEVELOPER

Syrinex Corporation, founded by Dave Straitiff in Rochester, New York, is a new company that has seven people and great plans for rapid growth. Mr. Straitiff, who said he wanted "high-end features without the cost" in his phone system, chose a TA/switch system that could integrate three BRI lines for his new company. Paying $35–40 per month per BRI, he added a fourth circuit dedicated to the Internet; the other three he shares among seven people on nine digital phonesets, including one full duplex speakerphone in the conference room. The phone system switch gives Syrinex full direct inward dialing, integrated voice mail, and remote ISDN access. The conference room is also fully equipped for videoconferencing. "These capabilities make us look as big as we plan to become," said Straitiff.

Syrinex' mission is to create products for telecommuters. According to Mr. Straitiff, "We want to make it really simple for someone working at a home office or very small branch office to sit down at their PC and get the feeling they are totally integrated with their company, its databases, and all the other people in their cubicles."

Practicing what he preaches, Mr. Straitiff has brought an ISDN link to his home. When his phone rings at the office, his business phone at home rings at the same time; it doesn't matter where he is, his calls get answered as if he were in the office. He is connected to the office LAN through a remote-access feature of his office switch, so even though commutes in his area are no particular hassle, he can take advantage of good ideas when they're ready to pop, rather than waiting until he gets to the office.

On the office ISDN line dedicated to the Internet, Syrinex takes advantage of the quick-connect feature of digital switching by installing a dial-on-demand Cisco router at his office. No need to stay off-hook for long periods of time. Likewise, his Internet service provider (ISP) has installed the same kind of router that connects to www.syrinex.com on demand; the typical Web browser never notices any dial-up delay, and it significantly reduces connect-time costs.

CONCLUSION

ISDN is just beginning to show its promise. Impediments to the progress of ISDN remain to be overcome. For example, some people hesitate to commit to ISDN because they hear it suggested that cable modems, wireless digital phones and modems, and ADSL (asymmetrical digital subscriber line) could make ISDN a temporary stop along the road to broadband nirvana.

In addition, a historical technology split between NI1 and AT&T Custom, competing standards for central office ISDN switches, slowed the deployment of ISDN during a critical marketing decade. Only now is it possible to request ISDN from virtually any Regional Bell Operating Company (RBOC) and have some degree of confidence regarding compatible connections. On the positive side, ISDN user groups, such as the National ISDN Users Forum (NIUF), are lobbying to get all the phone companies on the same page and to set up universal standards, such as a 14-digit Service Profile Identifier (SPID).

Pitfalls associated with acquiring ISDN remain, however. There are no standards relating to tariffs, and ISDN-service packaging varies from one extreme to another. It is possible, however, to find some bargains in the marketplace. With a bit of foresight and planning, customers are able to find affordable ISDN service.

Vicki Fuller is manager of Business and Channel Development for Siemens Business Communication Systems, Inc. The company is responsible for OfficePoint, an ISDN BRI telephone system for small offices. For more information, contact Siemens Business Communication Systems at 408-492-2000 or visit the company's Web site at www.siemenscom.com.

For information about free subscriptions, call our customer service department at 800-243-6002 (toll free) or 203-852-6800, or visit our Web site at www.ctimag.com. Contact the publisher, Richard Tehrani, or the editor, Kevin M. Mayer, with questions or comments about *CTI™*. E-mail (addressed to rtehrani@tmcnet.com or kmayer@tmcnet.com) is always welcome. ■

1997—THE YEAR I

BY PETER GEIER,
EICON TECHNOLOGY
CORPORATION

Increased demand for fast, high-quality connections between PCs and the Internet and corporate Intranets raises a key question: What technology or technologies will emerge to meet this demand? This article will focus on the most familiar candidate — ISDN.

What About Competing Technologies?

With the increased demand for faster access to corporate Intranets and the Internet, many are speculating about alternatives to ISDN. Many industry pundits are euphoric about cable and 56-Kbps modems and ADSL. The question is, will these technologies surpass ISDN over the next year? Let's take a look at each and compare:

Cable Modems

Cable modems are a relatively easy technology to understand — users connect a cable modem to their existing cable television wiring, and then the user's PC is connected to the cable modem. Cable modem testing has been underway for over a year now, and the results are fairly positive. Households with cable modems can download off the Internet at speeds of up to 6 Mbps, nearly four times as fast as a T1 connection. Cable modems give users quick access to the Internet and video-on-demand.

So why haven't cable modems taken off yet? There are a number of complex standards and infrastructure issues that need to be resolved before this technology hits the mainstream. Experts anticipate that it will take a few years before cable modem standards are in place. While there is an infrastructure in place for residences, most offices are not currently wired for cable, but rather have an Ethernet (or some other) LAN infrastructure. Also, most of the current cable infrastructure is not designed for the two-way communications that cable modem connectivity will require cable companies to spend millions of dollars to complete.

By the end of the century, standards and end-user infrastructures will be more secure for the cable modem market. At this time, when these issues have reached resolution, cable modems will be a good alternative for home connectivity to the Internet, but not much more. Since the household market is primarily interested in Internet access, the current cable modem infrastructure can support the small amount of upstream data. Corporations, on the other hand, need a strong, two-way infrastructure to transmit large files. These companies may be reluctant to install these networks since they will not have control over them.

An additional concern for many users is the fact that they have to rely on their cable company to provide this service. Cable companies sometimes have a difficult time keeping their current cable service customers happy. The question is, will users be willing to take the next leap of faith with their local cable company?

56-Kbps Modems

If last Fall's Comdex serves as any indication, people will continue to buzz about 56K in 1997. 56K technology takes advantage of digital connections between many telephone companies and their large corporate customers or Internet Service Providers (ISPs). Major players in this market are making a lot of noise in this arena. Rockwell has started supplying chips for 56K recently, and it is expected that Lucent, Texas Instruments, and a

R FOR ISDN

To say ISDN is familiar is to acknowledge that ISDN is not a new technology. This lack of novelty is advantageous in several respects: ISDN works with current applications and is backwards-compatible with analog modems. In addition, ISDN is unlikely to present any unpleasant surprises. In the United States, well-known problems such as lack of widespread availability and difficulty of use are already being addressed.

few others will not be far behind.

Throughput over copper telephone lines between either the ISP or central site and the end-user connection site is currently a limitation for 56K connections. Business travelers (who typically connect from a laptop in a hotel room) often find that they are unable to connect at 20K. Usually, they end up connecting at closer to 10K. The problem is that throughout the United States, copper wires used in our current telephony system cannot handle high-speed connectivity. Additionally, current wiring within hotels does not allow 56K modems to talk to each other without special digital equipment on the server side (for example, a T1 connection).

High-speed analog connections will not work until service providers upgrade their equipment to provide 56K connections. In order for 56K speeds to be reached, the ISP or central site must have a digital connection (such as a T1 or ISDN PRI connection) to the carrier. Since the current phone lines have noise filters which limit bandwidth, many think that 56K over copper lines will never be reached. Also, since there are a number of different implementations emerging, it will take time for standards to be reached.

ADSL

Trials for this high-speed alternative are starting to emerge in North America. ADSL, which stands for the Asynchronous Digital Subscriber Line specification, uses the common telephone wiring found in today's offices and residences. ADSL modems connect directly to a network, serving as a viable solution to direct all of a LAN's TCP/IP traffic to the Internet. The speeds are also rather impressive, and this "ISDN on steroids" has been clocked at nearly 9Mbps for downloads over a distance of 12,000 feet of 24-gauge telephone cable.

Similar to cable modems, standards issues will prevent ADSL from making it onto the market in full force for a few years. And similar to 56K, the quality of copper telephony lines in the United States will prevent ADSL from reaching its full speed potential. ADSL will require expensive improvements for the current telephony cabling system. Keep in mind that this technology has distance limitations, and it will involve infrastructure changes to solve those problems. While there is a lot of potential for this technology, it won't be ready for some time.

WHY ISDN?

ISDN is four to five times faster than the fastest current analog modem and always provides users with full speed without speed-negotiation and fallback. There is instantaneous connection within milliseconds, unlike the near minute required for analog connections. ISDN is the only technology that already has worldwide approval and deployment.

Applications exist today that are ideal for ISDN, and there are many more on the horizon. Only ISDN has backwards compatibility to the analog world, meaning ISDN-connected devices can communicate with analog modems and devices. With the implementation of EZ-ISDN and the recent initiatives taken by long-distance carriers, ISDN is a technology which is about to take off. Additionally, initiatives from the NIUF (National ISDN User Forum) and VIA (Vendors ISDN Association) will significantly impact the acceptance of ISDN in the United States. ■

To get a picture of ISDN's future, it helps to look abroad. In Germany, ISDN lines are already cheaper than analog phone lines, and almost all switches are ISDN-compatible. In Japan, ISDN is available at public telephone booths, allowing users to simply plug in their laptop to connect to ISDN service.

In this article, we make the case that ISDN will soon enjoy wide-scale implementation in North America. Already, prices are falling and availability is growing. The growth of new applications — distributed networks, Java applets, linked and embedded objects, and multimedia — will further drive widespread adoption. Indeed, we will be so bold as to claim that 1997 will be the year for ISDN. Here are our predictions:

PREDICTION 1: RBOCs AND CARRIERS WILL EASE IMPLEMENTATION OF ISDN

AT&T, MCI, and Sprint are now fully committed to ISDN service, and local phone companies are starting to aggressively roll out the service to meet a wider end-user base. Before the Telecom Reform Act of 1996, the Regional Bell Operating Companies (RBOCs) were unable to provide service outside of their regions. This segmented model hindered the creation of a nationwide ISDN service. While individuals and small organizations could sometimes get ISDN from the RBOCs, large companies that had to deal with multiple phone companies to set up and maintain ISDN service were left eagerly awaiting developments that would enable a nationwide implementation of ISDN.

The long-distance carriers are aggressively pursuing the ISDN market. AT&T, for example, recently announced that they have implemented a new service that allows users to bundle ISDN charges into their regular telephone bills. AT&T can now allow ISDN charges to be included with ser-

Primary Rate Network Access

BY JOANNE SKERRY, ACULAB

Primary rate access (PRI) — the foundation of fast, clean telephony systems for large organizations — is capable of delivering bandwidth-on-demand and multipoint termination, and of mixing voice and data on a single call. PRI also offers business-critical features such as caller line identification and direct inward dialing information. So, why would anyone hesitate to take advantage of PRI?

Network access via PRI can be daunting. ISDN developed in a piecemeal fashion and accumulated its share of bewildering jargon. As David Angell, the author of *ISDN For Dummies* writes, "We're a long way from plug and play." Still, the road ahead for ISDN looks bright. Soon, developers will have access to all the elements that a mature industry offers: high-density, highly functional components; third-party compatibility; and far-reaching standards.

T1 AND E1 INTERFACES

Perhaps the first thing to understand about ISDN, is the worldwide separation between E1 and T1 digital interface variants. E1 is the standard for most of the world. T1 is used only in Japan, Hong Kong, and the United States.

T1 is the older connection. At the time of its inception (the 1960s), engineers could only get 23 channels of voice and data. A later creation, E1, delivers 30 channels with signaling information carried in packet switch mode on two additional channels. (T1 carries the same information on a single channel.) A variant of T1 — robbed bit — is able to deliver a full 24 channels, because signaling information is spread across all the channels. E1 is often referred to as 30B + 2D, and T1 as 23B + D. The letter B indicates a bearer channel, while D indicates the transmission of signaling data.

ISDN BOARD VENDORS

Once, only a handful of companies manufactured these boards. Today, as vendors move toward selling all the components needed to assemble complete telephony solutions (all that is needed to benefit from ISDN is a network-access board and a computer), the digital-access market is becoming crowded and competitive.

Last year, the list of vendors offering native E1 and T1 products included Aculab, Dialogic, Natural MicroSystems, Pika, and Xircom. Prices dropped throughout 1996, but competition has, interestingly enough, spawned digital-access devices with much higher density per board. Natural MicroSystems and Aculab sell a 2-port E1/T1 board offering developers up to 60 channels of ISDN in a single, space-saving PC slot. Dialogic's acquisition of Dianatel gave them access to one of the first 4-port E1/T1 boards.

Vendors are also combining speech-processing functions with their core access devices to offer complete off-the-shelf computer-telephony (CT) components. At present, it is possible to obtain platforms enabling pulse detection, replay control, and call-progress analysis. In addition, some platforms offer speech recognition, pulse and tone detection, record/replay, and matrix conferencing. Vendors plan to release further DSP features — promising integrated functionality that would have been unimaginable just two years ago.

KEY FEATURES

• ***Support For Multiple ISDN Protocols***: A core consideration when choosing network-access products is the number of ISDN protocols that the manufacturer supports. Yet another reason why ISDN can be a thorny vices to link telephone systems at multiple corporate sites. With this new service offering, AT&T predicts that customers can anticipate a savings of up to 30 percent on their ISDN service.

With the passage of the Telecom Reform Act, the local phone company will no longer be the only provider of ISDN in town. Long-distance providers and local phone companies will be competing for customers.

Like other services, ISDN will benefit from the competitive environment that is now developing across the country. Businesses now have a choice in selecting a provider for their ISDN service, which will, in turn, make it easier for users to get the service.

Previously, fees, connect charges, and interface issues prevented the growth of the ISDN market. The industry is working to develop ways to work around these issues. For example, in 1996, Microsoft facilitated ISDN setup by introducing two initiatives:

1. The Get ISDN Program, which allows a user's Windows 95 machine to properly configure the ISDN line.
2. The Accelerator Pack, which significantly improves ease-of-use for Internet access over ISDN for Windows 95 users.

By simplifying driver development for ISDN hardware manufacturers, Microsoft has moved most ISDN protocol support from hardware to the underlying Windows operating system, thereby improving interoperability between ISDN cards.

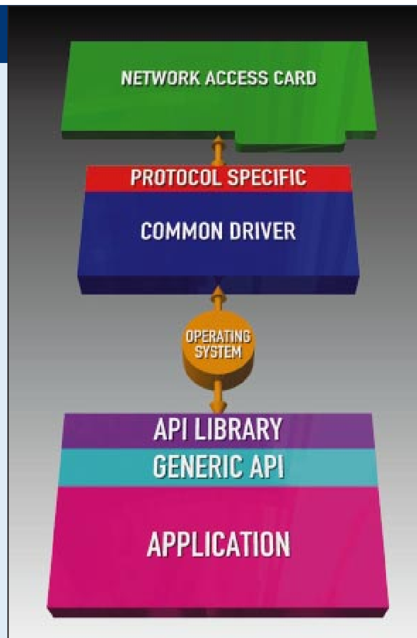
Bellcore has implemented a certification program called *EZ*-ISDN. This program allows both voice and data on an ISDN line plus a range of advanced telephony features, such as call waiting, for ISDN. In addition to simplifying the ordering process, *EZ*-ISDN gives users and RBOCs a bit more flexibility about how to set up ISDN lines as compared to inter-office channel (IOC) compatibility. Designed for the mass market, *EZ*-ISDN is already supported by sev-

subject: there are more than 40 different worldwide ISDN protocols, most of which are in a state of flux. To develop CT applications that sell into a global market, it is essential that developers choose a network-access manufacturer that has diligently researched, applied, and tested the protocols of a wide range of countries.

• ***Availability Of Generic APIs***: Another feature that allows international application developers to sell faster is a generic application programming interface (API), which shields application developers from very complex, low-level signaling tasks. Put simply, all that is necessary is to program to the API to ensure that the application is readily portable across national and international signaling systems (see figure).

• ***Compliance With Telephony Bus Standards***: For maximum market application, it makes sense to create applications on network-access products that do not lock developers into one type of bus. (A bus is needed to convey call information throughout the PC — to speech boards or pulse dialing boards, for example.)

There are three bus standards: MVIP (Multi-Vendor Integration Protocol), the SCbus, and PEB (PCM Expansion Bus). MVIP was initiated in 1990 by Natural MicroSystems to aid the fusion of computers and telephony. Today, there are more than 175



Although ISDN cannot yet be considered an off-the-shelf solution, use of common drivers and generic APIs can help to bridge the technical gaps associated with implementing ISDN.

MVIP-compatible components or board-level products for resale worldwide. The SCbus was developed by Dialogic and is now part of the SCSA architecture, which replaces the older PEB standard.

ISDN'S FUTURE

What of the future? Primary rate network access is (like computer telephony as a whole) a young industry. Vendors are desperately struggling to control standards and to be seen as connecting in an open manner with third-party products. The next 18 months will test these claims. Proposals for a new super bus could end the war between MVIP and the SCbus, while switch manufacturers are attempting to simplify the ISDN jigsaw by promoting Euro ISDN as the de facto E1 standard.

The PTT's (Post, Telephone, and Telegraph administration) commitment to push out additional ISDN supplementary services will continue to promote ISDN and reinforce the business benefits of a digital service. These changes won't make network access plug-and-play, but they will confirm it as the basis of a thriving computer-telephony industry.

Aculab Ltd. was formed in 1978 to undertake custom design and manufacturing in the professional audio field. Today, Aculab designs and produces intelligent E1/ISDN cards supporting national and international signaling systems. During 1996, Aculab introduced an enhanced E1 card with SCbus support, with T1 versions for North America, Hong Kong, and Japan. An API-compatible 4-port Basic Rate (BRI) card was also released, providing both T interface (central office exchange facing) and S interface (terminal facing) capabilities. ■

eral RBOCs, including Ameritech, Bell South, and NYNEX.

PREDICTION 2: THE COSTS FOR ISDN WILL DECLINE DRAMATICALLY

This isn't much of a prediction, for we are already seeing lower costs. In early 1996, prices for an ISDN interface card averaged around $400; today, they're under $200. These prices will continue to fall. In Germany, for example, simple ISDN interface cards start at around $100.

ISDN will be less expensive to set up, the equipment will be less expensive to own, and the effort to connect will be transparent to the user. Many people are asking if the lower costs will increase adoption in the United States. A similar situation arose when Philips was developing CDs. Back then, when the equipment was nearly $2000, hardly anyone believed the medium would ever take off. Now, CD players are everywhere, and the cost is under $200. Lower costs of equipment increased adoption of the technology. Decreasing prices for ISDN can affect the connectivity market the same way

> **ISDN will be less expensive to set up, the equipment will be less expensive to own, and the effort to connect will be transparent to the user.**

ISDN: Hate To Get It, But Love To Have It...

BY STEVEN OSSANDON,
INTERWORKS SYSTEMS

ISDN has received so much bad press that readers may be unaware of how strong ISDN has become. While the technology can still be difficult to implement, ISDN has matured. It is already delivering reliable applications and increased ease of use.

Many articles on the subject of ISDN have pointed out the frustration associated with ordering it and setting it up for practical use. Nonetheless, these negative appraisals are often accompanied by admissions that nothing else is available that can compete with ISDN, and that ISDN will probably remain unchallenged for some time to come. Since these admissions are often made grudgingly, just before the articles end, I fear that casual readers miss them, and get the impression that even the experts are confused by and lack faith in ISDN.

I have dealt with ISDN for several years and have witnessed dramatic changes. Not long ago, obtaining ISDN was a burden. Nonetheless, ISDN continues to proliferate, and the phone companies are making a noticeable effort to simplify the process. New standards are being implemented (NI-2, NI-3) that make ISDN implementation more akin to installing POTS (plain old telephone service).

The Internet, without question, is the catalyst pushing ISDN manufacturers and telephone companies to advance this technology. For surfing the Internet, ISDN is currently the best switched service available.

I recently had a yard sale for the pile of modems I had accumulated over the years in a vain search for one that did not cause extreme frustration. I have yet to see a true 28.8-kbps connection. It is always something less. And, for some reason, disconnects that occur halfway through file transfers are considered normal. ISDN always connects quickly and stays connected until I am ready to hang up the line.

Although the Internet has been a big driver for ISDN deployment, it represents only a small piece of what ISDN has to offer. LAN bridging, voice services, videoconferencing, and X.25 messaging really exploit the value of ISDN. This article will provide the reader with some real-world applications and tips on installing and using ISDN as regards the home office, telecommuting, and corporate voice and data networks.

INSTALLING ISDN

Terminal Adapters

By now there are well over a hundred manufacturers of terminal adapters, with the price of their products ranging from $168 to over $500. If your requirements are for connecting to the Internet, for about $400 you can pick up a popular product from Motorola. Since it was one of the first products of its kind to hit the street, many of the phone companies are familiar with configuring the lines — so it works on the first try. That is a good indicator of things to come. The day is fast approaching when all ISDN lines work on the first try!

Many companies now offer internal cards that sell for under $200. Some are real Windows 95 plug-and-play, using the Win95 dial-up adapter. (Check the Microsoft Web site for Win95-tested adapters.) I use one that includes fax software. The almost instantaneous connection that ISDN provides is a vast improvement over analog dialing. However, increased speed has its associated costs. For those of you thinking how much faster you can go using all 128K, remember, decreasing costs of CD equipment affected the audio market.

PREDICTION 3: APPLICATIONS WILL DRIVE THE SHIFT TO ISDN

With the shift from analog to digital communications, we are witnessing the growth of next-generation applications, and the demands these applications make on existing networks will demand next-generation connectivity. The ultimate success of these applications will be driven by high-speed digital communications.

Multimedia

The integration of graphics, data, text, voice, and video gives rise to speed demands that only digital communications, such as ISDN, can satisfy. In terms of productivity, using analog connections with multimedia is the equivalent of "throwing sand in the gears." Additionally, with decreasing costs for ISDN, you can expect that the cost of equipment for high-quality multimedia to the desktop will decline. Some analysts predict that multimedia ISDN cards will range in price from $400 to $500 by the end of 1997.

Next-Generation Applications

Besides multimedia, next-generation applications such as object linking and embedding (OLE), distributed Java applets, and the use of the network computer will put additional strain on the current communications infrastructure. Many companies have neither the time nor the patience to wait for ATM to come to the desktop. By the end of 1997, lower costs and new technologies will allow ISDN to be fully multimedia-enabled.

Videoconferencing

The next big application for notebook and desktop computing — integrated, affordable desktop videoconferencing — may come sooner than you think. Microsoft, with the functionality

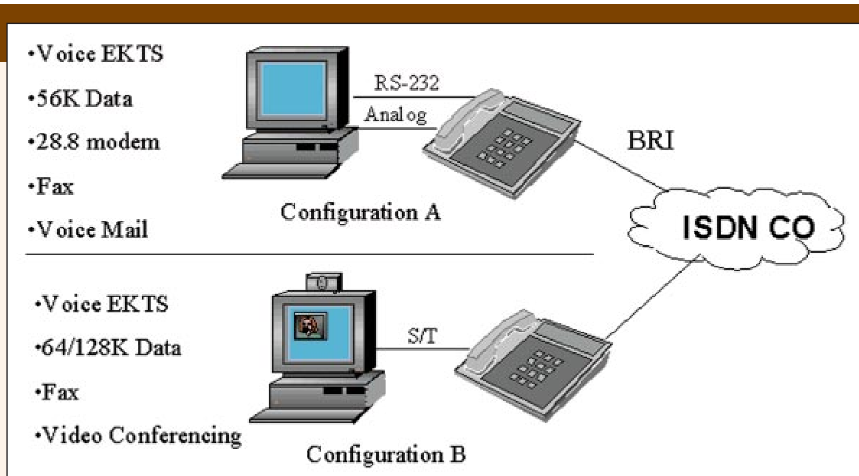


Figure 1. Two ISDN workstation configurations suitable for telecommuting and the home office.

the call costs twice as much with many phone companies. Also, your Internet service provider or the very site you are accessing can slow you down because of traffic congestion.

I suggest trying 56K/64K (asynchronous/synchronous). This arrangement should prove satisfactory, especially if you are coming from the world of modems. If you plan to transfer files on a regular basis, especially video or graphics, you will want to use a product that is capable of compressing data. Just be sure the host is using the same compression algorithm.

The more expensive adapters (some of them are just that — more expensive!), are suited for videoconferencing, and some give you low- and high-speed packet-switched data capabilities (X.25 on the D or B channel). For X.25 ISDN applications development, you can purchase a Visual Basic add-on to create messaging systems over the D channel without using B channels.

Remote call center agents and credit-card authorization are among the most popular applications. In addition, ISDN is suitable for many applications that are not bandwidth-hungry, or are not time-critical. Order processing, group schedule synchronization, message-waiting indicators, mail, security, and safety systems are a few examples.

My company has several full-time telecommuters who create software applications and provide support. They all use Siemens ISDN workstations that are telephones with 56K asynchronous data adapters that connect to communication ports on their PCs. It sets up easily, using Win95 dial-up adapters for Internet access and PC Anywhere for remote control of customers' PCs. The real benefit of this type of device is the voice-handling capability provided by ISDN. The employees can handle three calls simultaneously and can transfer back and forth to the office as well to each other on a single B-channel, all while maintaining a 56K data connection. There are many terminal adapters that include an outlet for a telephone, but they do not necessarily offer ISDN voice-handling features. Some manufacturers are planning adapters that incorporate those features, but I have yet to see any.

BRI Bridges/Routers

An ideal solution for small businesses, branch offices, or even home offices, BRI bridges and routers come in a variety of packages. They can produce cost savings over terminal adapters when connecting two or more workstations. Provisioning the ISDN for these devices is similar to that for terminal devices, and will not require configuration changes when you migrate from terminal adapters. The price tag will set you back $600 to $2,000, depending on the configuration and sophistication required.

Continued on page 92

in ActiveX and NetMeeting, has made implementing desktop videoconferencing as simple as buying a low-cost video camera.

Increased demand for high-quality, low-cost videoconferencing will reinforce the shift to ISDN. Currently, the installed base of desktop videoconferencing units numbers nearly 200,000. With the H.323 standard bringing high-quality video to a reasonable price point, the market for desktop videoconferencing could exceed 10 million units before the end of 1998. Some analysts predict that ISDN will connect at least 1.5 million of these units.

Remote Access/Telecommuting

With the growing popularity of remote computing, there is increased demand for faster, simplified access to mission-critical data. Companies are looking for new and easier ways to extend services from their central site to remote locations. Thus, telecommuting and remote computing also support the movement toward ISDN communication.

Utilizing two ISDN B channels, remote users and telecommuters will get a sense of being in the office when connecting to the corporate network. Since ISDN requires less time to establish a connection than analog modems do, users need only be connected when data is being transmitted. This advantage can translate to lower connect-time charges for ISDN users.

Other Applications

Widespread availability of ISDN will also generate support for several new applications. Although these applications probably won't drive the initial shift to ISDN, they will help ISDN flourish once it becomes widely available. As an example, companies in the publishing industry stand to achieve big savings by utilizing ISDN to transmit large pre-print files to print shops. Another example: ISDN will enable high-resolution, high-bandwidth applications such as telemedicine.

The retail and banking industries can use ISDN for instantaneous credit approvals. Today, when customers initiate a credit card purchase, the approval process takes nearly a

Continued from page 91

Once again, the cost of a particular product does not necessarily mean you are getting the functionality to match. This market is still emerging, and pricing for equivalent products varies greatly.

One of the most common applications to date has been to replace the 56K circuits used for remote bridges. ISDN lowers line costs and provides greater bandwidth without having to resort to an expensive T1. Many of these routers are installed as backups to T1 lines. They automatically reroute the traffic if the T1 connection fails.

We have installed routers in enterprise networks for disaster-recovery configurations designed to direct traffic around central communications centers. When the host site totally fails, a backup site goes online, and the ISDN routers dial it up without making undue demands of network administrators. Only a switched-data service can provide that type of advantage.

Features such as spoofing are very popular with NetWare users, since spoofing eliminates protocol noise that eats up bandwidth. The routers learn which network broadcasts need to go over the ISDN line, keep all IPX (Internet packet exchange) overhead local, and thereby reduce the cost of line usage.

USING ISDN

Telecommuting

With telecommuting on the rise, many managers are looking for ways to manage line usage and ensure that remote workers are being productive. Time of day and filtering are two features that address these concerns. Time of day restricts users' access to the line to working hours only, eliminating personal use during nonworking hours or use by unsupervised family members. Some products will let you program the router to dial a different number at a different time of day. Companies sometimes want to block user access to the Internet through the company connection during off hours, but will allow telecommuters to use the PC and router to access an ISP for personal use. Filtering can restrict specific application ports, such as allowing access to e-mail and to FTP to corporate servers, while restricting Web access. These features, in essence, allow the ISDN router to perform as a firewall.

Not only can you use ISDN for data services, you can use it to bundle voice services traditionally found in PBXs (call transfer, multiple call handling, conferencing, messaging, etc.).

Two typical configurations for ISDN workstations are shown in Figure 1. The configuration in the top panel uses a voice/fax modem (now included in most new PCs) with an ISDN workstation providing an analog port and an RS-232 communications cable to a PC com port. This is the easiest interface to set up and use. The configuration in the bottom panel uses an internal terminal adapter that connects to an ISDN workstation over an S/T interface. The terminal adapter supports 128K, fax, and video. This configuration requires an additional SPID, which is not yet available if you are in a Nortel DMS central office switch.

Small Office Home Office

The SOHO market, while offering integrators a large customer base, also raises its own concerns. Small companies cannot afford big-dollar service agreements, but require as much support as larger companies, if not more. It is not financially feasible to send a technician on every call. It becomes important to select products that allow remote managing of the customer's network. Since most providers are selling remote-access products anyhow, they can presumably include remote-management capabilities.

Our solution is a BRI router from ISDN Net with 4 or 8 SNMP (simple network management protocol)-manminute (most of this time is to establish the analog connection). A single ISDN B channel could support 300–500 credit card transactions per minute. This equates to 300–500 modems for establishing credit approval using analog communication techniques.

The Vendor's ISDN Association (VIA) is working on new ways to use ISDN technology. One proposed use is in the home-security industry. With current home-security systems, once a telephone line is cut, there is no way for the security company to monitor that home or business. This problem can be solved with an ISDN line. By using an ISDN D channel, which constantly polls for data, the security company will know immediately when a phone line is cut.

CONCLUSION

In 1997, both capital and operational costs for ISDN will continue to decline. With the proliferation of fast, reliable ISDN connections, we will finally see the maturation of the concept of network-centric computing. Specifically, ISDN will allow high-quality, affordable, desktop-networked multimedia and videoconferencing. ISDN will make telecommuting and the "virtual office" more practical. ISDN will enable next-generation applications.

Peter Geier is ISDN marketing manager at Eicon Technology, a worldwide provider of connectivity solutions for personal computers. The company develops, markets, and supports hardware and software products for connecting PC-based servers, desktop PCs, and notebook PCs to corporate networks, IBM host computers, and the Internet. For more information, contact Eicon at 514-745-5500 or visit the company's Web site at www.eicon.com.

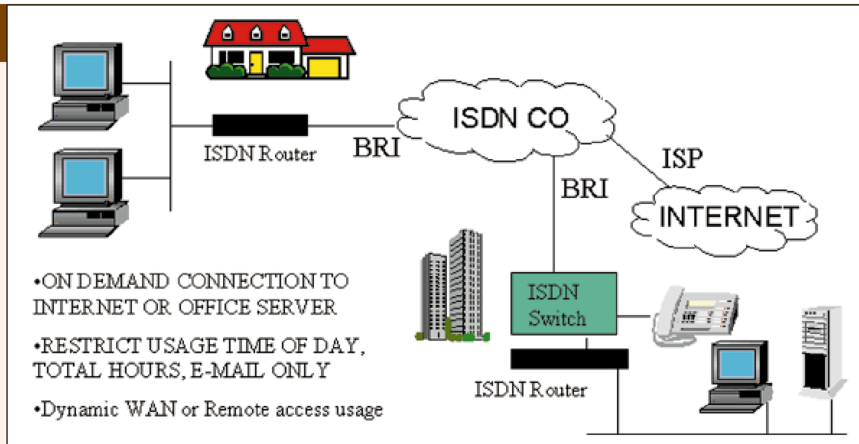


Figure 2. ISDN BRI routers with a built-in managed Ethernet hub make connecting to the office server or the Internet simple. The ports can be used for inbound network access or multiple users connecting outward to the Internet simultaneously.

aged Ethernet ports. Combined with PC Anywhere, the router allows us to fully administer the customer's network, take control of a workstation, check to see that data backup is ongoing, or troubleshoot the servers. If the network has problems, it will let us know, often before the customer calls for service.

Another feature allows customers to use the products during the day for WAN services and during the evening for remote access from home (Figure 2). They can be cascaded together to get more Ethernet ports and more ISDN ports for increased bandwidth to the same host or to connect to different hosts simultaneously. When this feature is used for Internet access, you pay for only one account, but you have support for multiple concurrent users.

Corporate Voice And Data Networks

By using ISDN systems that integrate voice, data, and video services into one package, you create a simplified office solution. By minimizing the number of components used, you reduce the number of points of potential failure. These types of systems are beginning to hit the market, and only ISDN can support these integrated services over a common transport. Small businesses can emerge from the dark ages of telephony and access many features traditionally offered by large PBX systems. Lower-cost direct inward dialing, PC dialing, screen pops, caller ID, intelligent call routing, and off-site transfer are just a few of the features available with ISDN. Connect your computer network to it for WAN and remote access along with videoconferencing software and you have an advanced business communication system. These are systems you can have today. They are reliable, fast, and easy to use.

CONCLUSION

There are other technologies on the horizon (how far off, no one is quite sure), such as cable modems and ADSL (asymmetrical digital subscriber line), and although they are faster for browsing the Internet, they are significantly different from ISDN. ADSL is not a switched service. It is not intended for videoconferencing or voice calls. ISDN will not be replaced by ADSL. They will coexist and complement each other. ISDN is maturing. With all the benefits it has to offer, ISDN is well on its way to replacing POTS and taking its place in the corporate workplace and the home office alike.

Steve Ossandon is president of Interworks Systems, a systems integration company located in Melville, NY. Interworks has extensive experience with ISDN. If you have any questions or comments, you can contact Steve Ossandon via e-mail at ossandon@iworksys.com.

VOICE OVER ATM TO THE DESKTOP: THE LAN AS PBX

BY BYRON BROOKS AND ERNST RIEMANN, SPHERE COMMUNICATIONS

Over the past two years, ATM (asynchronous transfer mode) has endured the usual ups and downs encountered by new technologies in the marketplace. While the pundits have debated the merits, success, or even premature demise of the technology, ATM has quietly become an indispensable component in corporate backbones, in the WAN, and on high-performance desktops. The remaining debate centers on a few key issues: whether ATM will ever be affordable enough to deploy on every corporate desktop; the uncertainty of integrating ATM into existing LAN Ethernet and Token Ring infrastructures; and whether LANs really need the bandwidth and complexity of ATM.

A more significant issue today may be how ATM stacks up against existing LANs to efficiently handle voice and multimedia to the desktop. After all, wasn't ATM created to carry simultaneous voice, data, and video? Is it possible that ATM is the answer to the inherent limitations of Ethernet technology for the transport of real-time multimedia? What exactly is multimedia anyway? Is it just the one-way playback of sound effects on a PC video game, or the playback of stored video? No. There is another, far more demanding kind of multimedia — real-time, interactive, two-way communication by voice and/or video. Here, high-quality performance is much harder (or even impossible) to achieve.

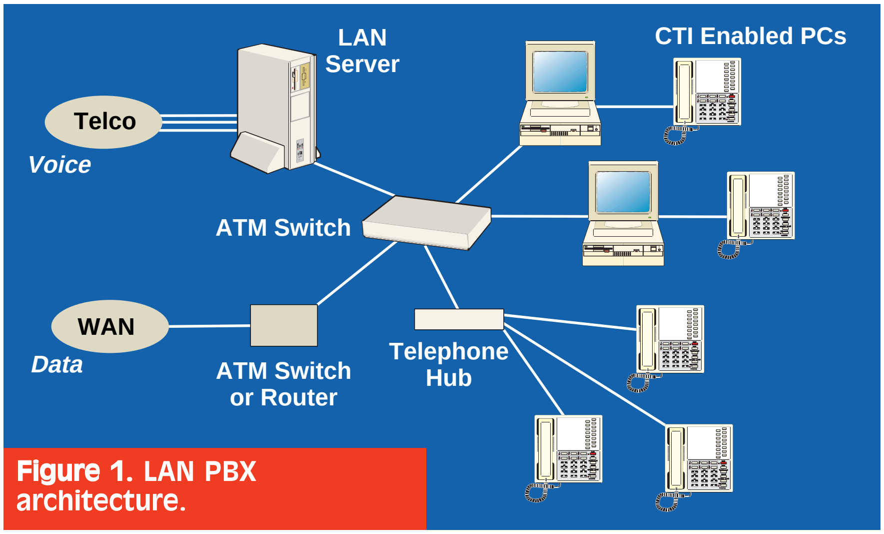


Figure 1. LAN PBX architecture.

WHY ATM?

ATM technology was developed to carry voice, data, and video traffic. It can simultaneously provide appropriate levels of service to these differing data types, recognizing that all bits are not created equally. For example, data traffic (database reads and writes, e-mail, and network file transfers) is transmitted across the network in bursts. Most applications are tolerant of significant delays in the transmission of data packets. Voice or video, however, are isochronous (time-dependent) forms of data that require special handling for effective transmission on a network, particularly when bidirectional communication is required.

While the bandwidth needed for each voice connection may be small (64,000 bits per second), the connection must capable of continuously transmitting

small voice data packets with very little delay to assure a high-quality voice connection. In normal LAN data transfers, delays of 100 milliseconds or more are common while the network sets up to begin the transfer. For voice, delays as short as 35 milliseconds are audible. Delays in the range of 100 to 300 milliseconds or more are readily noticeable (and, toward the top of this range, increasingly annoying).

Although such delays are fine for one-way voice or video playback, they can make a dynamic, interactive conversation between two parties frustrating and difficult. Most of us have heard the delays inherent in intercontinental satellite voice transmissions, and are familiar with the ambiguity caused by one speaker not being able to tell for hundreds of milliseconds that the other has begun speaking. We tolerate what amounts to half-duplex voice transmission over long distances when fiber-optic cable is unavailable, but experiencing the same delays when talking to the next cubicle over the LAN would be unacceptable.

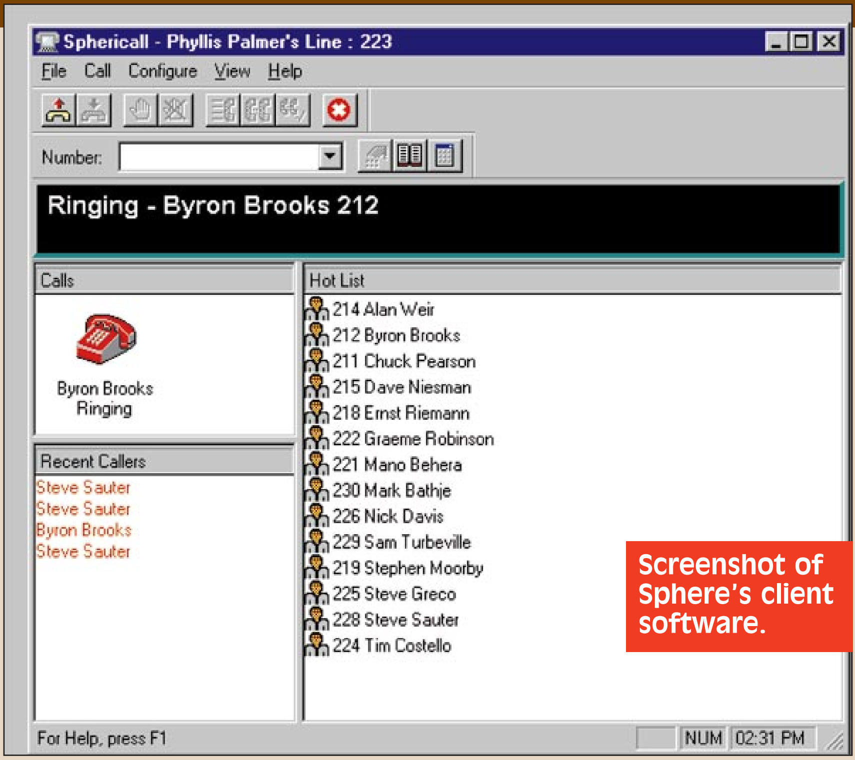
ATM connections are created with end-to-end guarantees that deliver the requested bandwidth in cells, at a specified rate within very small delay tolerances. This is ATM Quality of Service (QoS), which guarantees that each vir-

Sphere Communications' Sphericall

Sphericall is a full-featured PBX with CTI that runs as a client/server application under Windows NT Server 4.0 on a standard ATM-25 LAN. This architecture represents a radical departure from all other PBX implementations in two significant ways. First, voice processing and call control functions are implemented with software rather than hardware. Second, a single set of UTP-3 wires carries both data and voice to Windows NT or 95 client computers, where standard analog telephones are connected. Sphericall does not have the fixed hardware limitations (such as limited number of lines and voice mail ports) associated with hardware PBXs. Features include unified messaging with Microsoft Exchange, enhanced telephone operation through intuitive Windows-style GUIs, and easy setup and administration.

Sphere Communications has developed a technology called ATM Direct which processes real-time analog voice directly into ATM cells and reassembles the ATM cells back into analog signals on the receiving end. ATM Direct technology is used on the Sphericall server PCI bus card, which accepts up to eight incoming analog POTS lines, and on the Sphericall ATM client NIC, an ATM-25 PCI card, which is voice-enabled with the ATM Direct technology and a standard RJ-11 jack for standard phones.

A special Sphericall eight-phone wiring hub is available, which allows standard keyset phone service for areas that don't have computers. These hubs can be stacked for added capacity. ■



Screenshot of Sphere's client software.

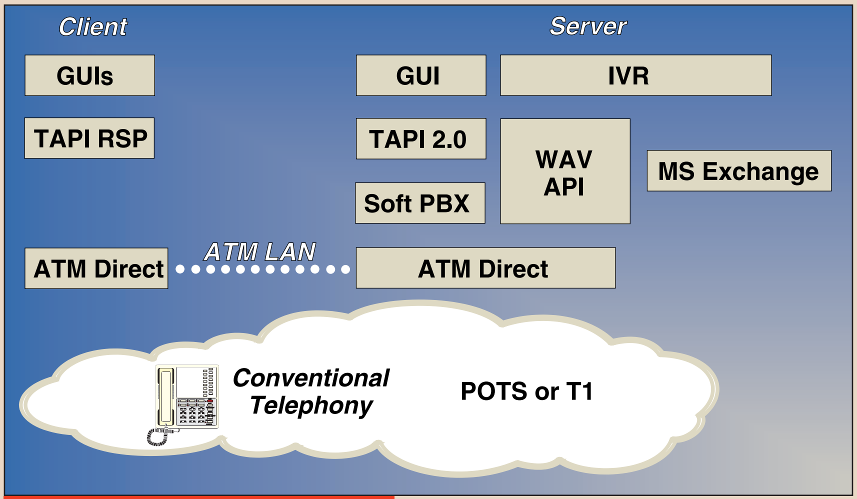


Diagram of ATM Direct.

tual circuit carrying voice over the ATM line receives its own priority and reserved bandwidth. Cell length is fixed at only 53 bytes so that data, voice, and video cells can be readily intermixed, with each getting bandwidth on a timely basis as needed by the source, not as it becomes available on the network. This contrasts with packet-based networks, which transport packets of varying length, and where it may be necessary to wait for a long packet before the next transmission can begin.

ATM's QoS makes ATM an attractive medium for dynamically establishing multiple voice or video connections. While it has become increasingly prevalent in WANs and corporate backbones (see the sidebar entitled *Voice Over ATM WANs: You're Probably Already Using ATM*) the older technologies, in particular Ethernet, remain the networking topology of choice on the desktop. New applications using ATM's spectacular real-time multimedia capabilities, coupled with a continuing decline in prices, should dramatically accelerate the demand of ATM at the desktop over the next few years.

Voice Over ATM WANs: You're Probably Already Using ATM

Several private and public networks already employ ATM in backbone and WAN environments. MCI, for example, has deployed ATM throughout its system for carrying voice via high-speed links to various points of presence. Private businesses that need to distribute voice traffic across many regions are installing or leasing ATM connections. Many of these ATM implementations run T3 or OC3, providing very high bandwidth services. Alternatives, where such great bandwidth is not needed, include linking ATM WANs to fractional leased lines. Companies have found the use of ATM to be a very flexible way to mix data, voice, and other multimedia streams across their geographically dispersed enterprise without setting arbitrary limits and channels for each. ATM enables the bandwidth to be used dynamically.

The ATM Forum is currently working on standards for connecting ATMs and standard PBXs. Two schemes currently exist — circuit emulation service (CES) and switched virtual circuits (SVC). Each has its advantages. In a CES scenario, constant ATM channels are implemented which emulate current T1 PBX trunk specifications. The PBX thinks it's talking to a T1, when in fact it's hooked to ATM. This implementation is especially attractive to companies that have massive investments in PBX hardware.

Switched virtual circuits, on the other hand, dynamically allocate point-to-point circuits from PBX to PBX, creating and dissolving connections on-the-fly. Physical connections to the PBX are reduced but require a higher level of integration.

While the ATM Forum is working to finalize these enterprise and other LAN ATM specifications, ATM voice is a reality in high-bandwidth, high-speed environments today. As with many standards we now take for granted, real-world implementation often leads the way. ■

INERTIA ON THE LAN

The speed figures associated with ATM and other technologies can be confusing. To make sense of them, it helps to remember that ATM is a broadband-based transport with QoS supporting multiple simultaneous virtual circuits over a single physical connection. Therefore, it accommodates simultaneous data, voice, and video streams, while other transports must stack all the data types up in a single high-speed pipe. So while gigabit Ethernet (for example) might sound impressive, it's only a very fast one-lane road. ATM, in contrast, is a multilane highway, enabling multiple data types to coexist in their own dedicated lanes, or virtual circuits.

ATM's QoS and broadband architecture make a compelling technology case. But technology alone isn't reason enough for a business to make the switch, regardless of bandwidth concerns. Because ATM is not widely deployed, it is still relatively expensive. Typical ATM-25 cost per seat is in the $300–600 price range — far higher than the cost of Ethernet.

WHO NEEDS THE BANDWIDTH?

Part of the reliance on Ethernet is the sheer inertia of "use what you know and what already works." However, some corporate networks are beginning to outgrow the bandwidth of even 100-Mbit Ethernet and need higher-speed alternatives. Power users, large files, graphics, and multimedia applications are driving certain LAN segments to the saturation point in terms of available bandwidth. These are the same constraints that pushed Arcnet out of early LANs and paved the way for Ethernet. While gigabit Ethernet promises faster speeds, the basic structure is identical: a single pipe, though much faster.

Many companies at the bandwidth-saturation crossroads find they must choose between going with the tried-and-true despite its architectural flaws (and working around them) or deploying a whole new technology (which could solve bandwidth problems now and in the future).

But even businesses experiencing these bandwidth bottlenecks are not hitting the wall uniformly. Perhaps only 10 percent of their traffic is bandwidth bound, while the remaining 90 percent of the normal LAN traffic is doing just fine.

THE LAN AS PBX

What many companies fail to recognize is that their LAN is not the only network on their premises. It may be the most visible, but their telephone system is yet another set of wires, switches, circuits, and data running in parallel to the data LAN. The PBX's data just happens to be voice. Maintaining both of these networks is expensive. In some small to medium-sized businesses, for example, a full PBX and digital phone system may actually be more expensive than the current data LAN. But because the telephone system is so ubiquitous, yet nearly invisible, it's costs and maintenance are not seen in the same terms as the cost and maintenance of computer networks.

As we've seen from the above technology comparisons, ATM was designed to carry multiple data types. It's already implemented throughout large-scale private and public telecommunications networks for its QoS and broad bandwidth characteristics.

In addition, ATM's point-to-point

switching characteristics provide an environment for virtual PBX call switching. PBX call control and features, such as ACD, auto attendant, and voice mail, can be moved from vendor-specific hardware to open, more cost-effective LAN platforms with ATM serving as the voice-delivery and switching medium. Current PBXs have software content locked into proprietary hardware platforms. By eliminating the proprietary hardware and converting almost all PBX functionality to software, businesses can realize significant cost savings.

A PBX built as a LAN does not encounter the traditional hardware-defined constraints (Figure 1). System capability can grow just like a LAN: client by client, and server by server, as capacity requirements increase.

ENABLING TECHNOLOGIES

The ability to economically combine data and voice networks and to convert PBX functionality to software has only become possible within the last few years. Several factors have converged to make this possible:

• ***Powerful PCs.*** Pentium-class PC servers are enormously powerful. They have everything from RAID (redundant array of inexpensive disks) to fully redundant power. They provide relatively inexpensive 24 x 7 availability.

• ***Microsoft NT 4.0.*** Microsoft's latest release of its networking software is not only a robust and reliable network operating system to base a voice network on, its core services incorporate many features which are telephony-rich via TAPI 2.0. With this new version of TAPI (Telephony Applications Programming Interface), Microsoft has made it possible to provide server-based telephony services to client computers across the LAN. A standard interface, it will encourage software developers to write new platform-independent telephony applications.

• 25 Mbit ATM And Winsock 2. While still more expensive than Ethernet, ATM25 can become more

Multiservice Networking

RON NASH,
MADGE NETWORKS

Desktops in large enterprises today are equipped with two kinds of communications terminals — the PC and the telephone. These devices are supported by two distinct single-service networks — the LAN and the PBX. Getting these two networks to talk to one another is no easy feat. But even the minimal level of integration achievable with today's CTI servers has such obvious benefits that it's natural to wonder what it would take to achieve a more complete integration of data and voice.

THE MULTISERVICE NETWORK

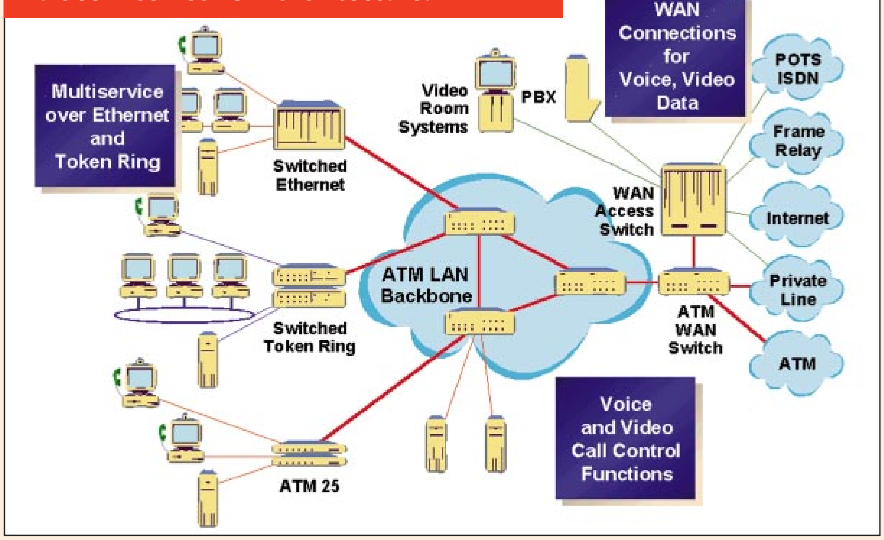
The answer to that question is now emerging under the concept of a multiservice network. Instead of ingeniously (and expensively) trying to connect single-service telephony and data networks, a multiservice network provides a single infrastructure for combining voice and data, as well as video. Such a network will enable new forms of telephony integration, where voice and data can interact at multiple points in the network, not just at the CTI server. CTI capabilities will be available throughout the enterprise, not just in the call center. Ultimately, voice and video become just types of data, to be stored, filed, retrieved, and integrated with graphics and text in multimedia documents.

Recent advances in LAN technology and new architectures for telephony and video services are making multiservice networking a reality. This article briefly describes some of the components of a multiservice network.

HIGH-SPEED SWITCHING WITH QUALITY-OF-SERVICE GUARANTEES

ATM switching is now being widely adopted in the LAN backbone for scaleable bandwidth. Though ATM today is used mostly for data, its connection-oriented approach is ideally suited for voice. Most importantly, ATM delivers the Quality of Service (QoS) necessary for real-time applications — the ability to reserve bandwidth and guarantee minimal delay. But that doesn't necessarily mean tearing out your existing Ethernet network and replacing it with ATM. ATM will be needed in the backbone, but switched Ethernet or Token Ring to the desktop will soon offer QoS. The Resource Reservation Protocol (RSVP), a new standard from the Internet Engineering Task Force, will give TCP/IP applications running on legacy desktops a standard way to request QoS, and the ATM backbone will take care of fulfilling those requests. For new networks, ATM-25 at the desktop provides the cleanest solution, at increasingly affordable prices.

Multiservice network architecture.



cost-effective when data and telephony are combined on the LAN. The Winsock 2 API, from Microsoft, makes it possible for programmers to utilize the unique virtual network characteristics and QoS of ATM for real-time multimedia under software control.

VOICE OVER ATM: BOON TO CTI

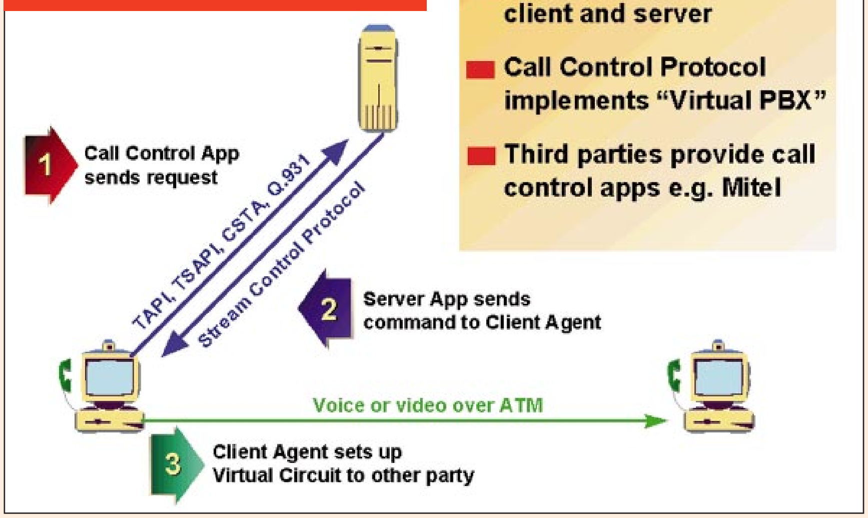
Once voice moves from older, vendor-specific PBX platforms to become a LAN application, all the rules change. Rather than integrating two basically incompatible networks — LANs and PBXs — developers can redirect their energy to provide enhanced software usability, features, and power.

FUTURE OF VOICE OVER THE LAN

Adding high-quality voice traffic to an existing Ethernet LAN is not feasible with current technology. Ten or even 100Mbit Ethernet has no QoS guarantee, and it cannot effectively handle the demands of isochronous voice, data, and video. Today, ATM is the only networking technology capable of combining simultaneous high-quality voice transmission with video and data on the same set of wires. In the future, another new technology, not yet on the horizon, may step up to replace ATM as the multimedia transport of choice. But the compelling economic and productivity arguments for combining voice and data over LAN infrastructures will remain.

Founded in 1994, Sphere Communications' strategy is to move telephony from the rigid customer premises equipment (CPE) model to a PC LAN application. For more information, contact Sphere Communications by phone at 847-247-8200 or visit the company's Web site at www.spherecom.com.

Client/server telephony call control architecture.



CALL CONTROL SERVICES

Since the switched network is capable of switching voice calls, the PBX is no longer needed to provide that function. But we still need call control — the intelligence that handles call setup, call transfer, call hold, etc. This year we have seen the first demonstrations of client/server architectures for call control, where the call control function is handled by an application running on a standard PC server, using established protocols such as TAPI and TSAPI to communicate with clients. The call control application functions as a directory server — mapping names and phone numbers to network addresses — and monitors calls, so that unanswered calls can be transferred appropriately. The actual call setup, however, is handled by the PC clients at the call control server's instruction. Working together, these components form a virtual PBX. Since neither a conventional PBX nor a CTI server are required, the cost savings of this approach are considerable.

GATEWAY TO PUBLIC NETWORKS

Switching voice calls in the LAN is of limited value if we can't connect calls to public networks. We need new kinds of circuit-switching gateways with LAN interfaces and telephony trunk connections to public networks and PBXs. A gateway must know how to translate 64K and Nx64K bit/second bitstreams into IP packets or ATM cells. Such a gateway can also connect H.323 video in the LAN with traditional videoconferencing systems that speak the H.320 protocol over ISDN. Interaction between videoconferencing stations and telephones will also be possible — very useful if you want to conference in a party that doesn't have video access, or send an unanswered video call to voice mail.

APPLICATION SERVERS

The client/server call control architecture lends itself naturally to a modular approach to application services. Voice mail, IVR, and ACD will be implemented as software running on standard PC servers rather than on proprietary circuit-switching platforms. This trend is already underway, but open interfaces to these application servers will finally make it possible to mix and match applications from different vendors.

Making multiservice networking a reality will require LAN equipment suppliers and traditional telephony vendors to unite in support of an open architecture. The CTI industry, with a foot in both camps, is uniquely positioned to help drive this trend forward.

Ron Nash is director of ATM Market Development for Madge Networks, which supplies end-to-end switched networking solutions for large enterprises, with a special focus on multiservice networking (that is, networks which will be able to support integrated data, voice, and video). In addition to ATM, Ethernet, ISDN, and Token Ring technology, Madge offers a range of products from desktop connections through LAN and WAN switches, and enterprise network management software. For more information, contact Madge Networks at 800-876-2343 or visit the company's Web site at www.madge.com. ■

CTI FEATURE

CTI DATA SERVERS

BY JOHN LYNCH, ARIEL CORPORATION

CTI is broadening its focus beyond voice and fax applications to embrace remote-access applications. Key developments driving this trend include the growing acceptance of CTI standards and the power and flexibility of hardware based on programmable digital signal processors (DSPs). The open CTI platforms that can now be built are not limited to supporting applications such as call processing, conferencing, fax broadcast, and fax-on-demand. CTI servers are also eminently suitable for providing remote data access for applications like Internet/Intranet access, online services, and transaction processing.

S PROVIDE REMOTE ACCESS

Internet service providers (ISPs) are already beginning to exploit CTI platforms to give many dial-up users access to the Internet. With more than 30 million remote-access subscribers (and more coming), the Internet offers tremendous opportunities for suppliers of data servers based on open CTI technology. Corporate environments also provide tremendous opportunities for suppliers of CTI-based remote-access equipment. With more employees traveling, and with more people working at home, many companies are adding remote-access equipment that can give mobile professionals remote access to both the Internet and corporate Intranet. With CTI, this capability can be integrated directly into existing voice and fax servers.

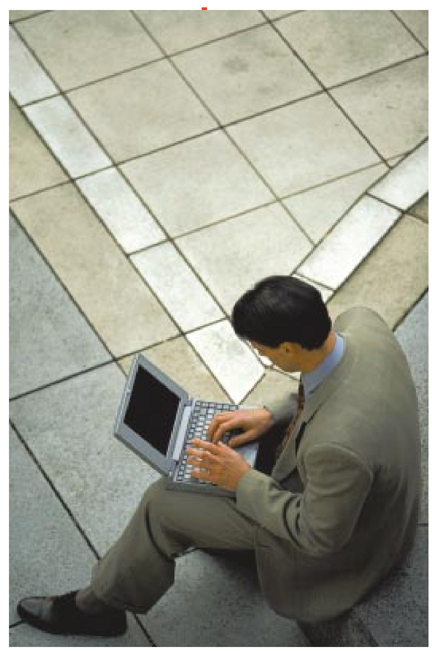
Transaction processing, though a more mature market, also offers exciting possibilities for suppliers of CTI-based remote-access equipment. Far-flung retail merchants, parts suppliers, insurance agents, real-estate brokers, travel agents, and other professionals all require remote dial-up access to central databases, and open CTI servers provide a flexible, cost-effective gateway. Together, transaction processing, Internet/Intranet, online services, and BBSs are driving the development of data technology that will soon take its place alongside voice and fax in CTI equipment that supports unified messaging.

WHY CTI?

Most of the servers used for today's remote-access applications are implemented as a medley of closed-box components that are strung together with proprietary network and software interfaces. As a result, upgrading these servers (adding new functions and integrating the server with other system components) can be difficult.

With the open CTI approach, diverse remote-access functions such as modem pools, WAN/LAN interfaces, routers, and computer servers can be integrated in the same PC enclosure utilizing common hardware and software interfaces. The open CTI approach also makes it easy to integrate remote access with existing voice and fax functions.

Open CTI platforms are especially attractive to small Internet and Intranet Points of Presence (POPs). These entities are striving to provide value-added services that bring them closer to their customers and differentiate them from larger competitors. Typical services



include filtering software (like Net Nanny, which screens undesirable programming), local search engines, agents (services that find information), and caching facilities that boost access speed by storing frequently accessed pages (such as the Yahoo index). A CTI platform's open hardware and software interfaces make it easy for integrators to acquire or develop the technologies needed to add these services.

SERVER ARCHITECTURE

The CTI servers used for remote access are architecturally similar to those used for traditional CTI voice and fax applications. In fact, the data modem function may coexist with other media-processing elements (such as fax, voice processing, and switching) in an integrated unified-messaging server.

Many CTI servers are PC based. Client terminals interact with the server through a variety of LAN and WAN interfaces. Within a facility, desktop PCs, phones, and fax machines may be linked to the server via LANs, Basic Rate ISDN, or plain old telephone service (POTS). The server, in turn, may communicate with the outside world via the public switched telephone network (PSTN), either through a PBX, or directly via a T1, Primary Rate ISDN, or POTS interface (see Figure 1).

Within a CTI server, network-interface and media-processing components (that is, fax, modem, speech recognition, and text-to-speech) communicate with the host via the ISA/EISA bus. The communications between CTI boards and the transport of voice, fax, and data traffic are offloaded to an auxiliary time division multiplexed (TDM) bus such as MVIP (Multi-Vendor Integration Protocol) or SCSA (Signal Controlling System Architecture).

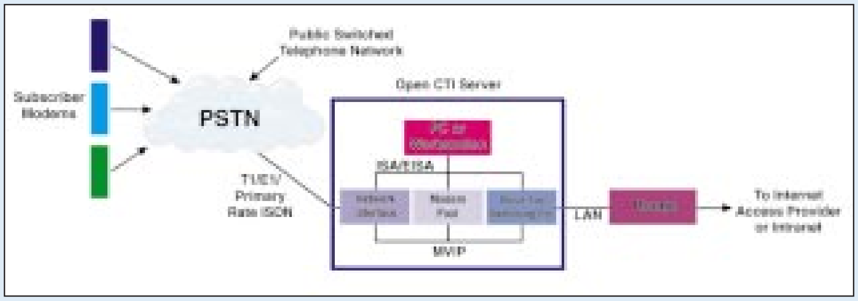
On the software side, standard APIs, such as TAPI and TSAPI, and bus standards, such as SCSA and MVIP connection control, are available to help simplify the development of hardware-independent computer telephony applications. To request computer telephony services, application developers make calls to the APIs from within their C programs (that is, play, record, send/receive). These calls, in turn, are translated by drivers (supplied by the board vendor) into board-specific function calls.

Standard CTI APIs exist for single-line desktop client systems, multiline SOHO systems (Small Office/Home Office), and large-scale networked servers. TAPI (telephony API), a Windows API that provides telephony control functions such as call progress, set up, and tear down, is tailor made for single-line desktop systems. Novell's TSAPI provides a comparable set of telephony functions, adding PBX control and networking functions that make it better suited for SOHOs and servers that are used in a LAN environ-

Ariel's Approach To Programmable DSP Hardware

Ariel has developed a 16-port ISA bus modem pool, CTI-Modem, to target the transaction-processing market. Though configured at the factory as a multiport modem, CTI-Modem's programmable DSP architecture also makes it suitable as a general-purpose CTI computer resource. To support designers who want to extend the functionality of CTI-Modem, the board is available with a library of CTI software that includes telephony signaling functions (such as caller ID, DTMF, and call progress). The product also supports data/fax modems; speech coding, recognition, and speech synthesis; MPEG and JPEG audio and image coding; FM/wavetable music synthesis and MIDI support; echo cancellation; speakerphone; and sample rate conversion.

To help manage the execution of these functions, as well as host/DSP communications, CTI-Modem also comes with a multimedia real-time operating system known as VCOS (Visible Caching Operating System).



VCOS offers frame-based multitasking and visible caching mechanisms that enable it to concurrently manage the execution of several media-processing functions in real-time. Its resource manager enables designers to execute functions that are distributed across multiple DSPs, and its ability to dynamically load host-based code modules makes it possible for the host to reconfigure CTI-Modem on the fly.

According to Ariel, such features make CTI-Modem suitable for SOHOs and CTI Servers that provide multiple telephony functions. Ariel also notes that the ability to support multiple functions on the same platform makes VCOS an option as a low-cost unified-messaging platform. For suppliers of traditional voice and fax servers, VCOS makes it possible to incorporate data capabilities such as dial-up remote access into their systems. At the same time, the VCOS/DSP tandem enables suppliers of dial-up data POPs to differentiate their products through value-added features such as voice, fax, and even video. For a closer look at an open CTI approach POP with unified-messaging capability, see the accompanying figure.

ment. SCSA and MVIP, which target CTI servers, build on the telephony functions provided by TAPI and TSAPI, adding media-processing facilities such as fax, speech coding, speech recognition, and text-to-speech.

INTERNET/INTRANET ACCESS

High-density modems equipped with standard PC host and computer telephony interfaces, such as SCSA and MVIP, provide the core technology for integrating data into CTI servers. High density is especially significant for ISPs and online service providers, who need to accommodate thousands of dial-up users and provide room for future expansion.

In these applications, the remote-access equipment and the real estate needed to store that equipment are often the dominant start-up costs. The higher the modem density, the more compact the equipment, and the less floor space that has to be leased. The modem pools used in dial-up servers for Internet and online services are typically based on standard 28.8-kbit/sec v.34 technology — the same technology used in desktop modems.

TRANSACTION-PROCESSING SERVERS

High modem density is also crucial for the large servers used to provide remote data access in transaction-processing systems. The dial-up transaction-processing market is dominated by players such as Verifone and Hypercom who supply the terminals used by merchants and the modem pools used by central hubs to access the network. Unlike the standard v.34 modems used in Internet/Intranet servers, the modems used in transaction-processing servers are based on non-standard ITU modulations. Because the amount of data transferred is small (for example, 100 bytes for a yes/no answer on a credit card transaction), these custom modems are optimized for small data packet transfers and quick response time. Typically, these modems run at 2400-bps data rates. Response times average just a few seconds, versus 30 seconds for a v.34 desktop modem.

To better serve the transaction-processing market, an approach utilizing programmable DSPs can be utilized to implement the modem function. The added flexibility afforded by programmable DSPs is important for the modems used in transaction-processing applications, which utilize a variety of non-standard ITU modulations. A DSP's programmability enables it to be customized for a broad range of transaction-processing environments (for example, to enhance start-up signaling or optimize data transfers for short or long packets). The flexibility afforded by programmable DSPs also makes it easier to upgrade the modem to handle more advanced modulations and higher data rates.

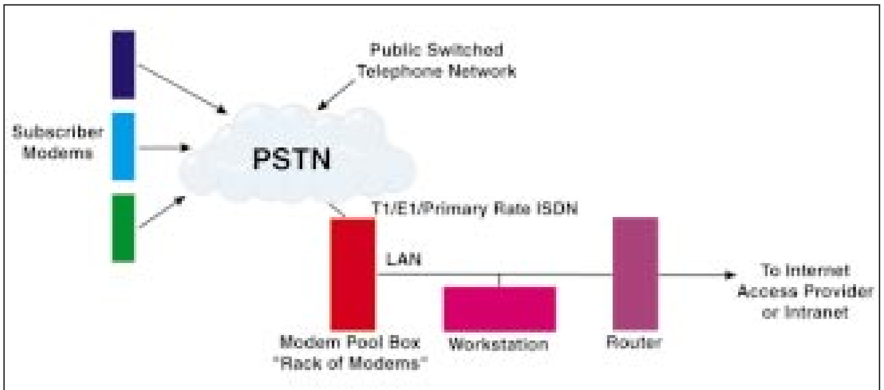


Figure 1. A traditional closed box approach to Internet/Intranet POPs.

In addition to maximizing flexibility, DSPs also afford significant opportunity for cost reduction.

In addition to maximizing flexibility, DSPs also afford significant opportunity for cost reduction. Custom implementations may be more cost effective for a single-port desktop implementation, but the ability to incorporate multiple slower-speed modems on a single device makes general-purpose DSPs more cost-effective for server environments. The flexibility offered by programmable DSPs extends well beyond the ability to quickly implement custom modem configurations. In fact, a homogeneous network of add-in cards based on programmable general-purpose DSPs provides an ideal computer resource for executing a broad range of multimedia and computer telephony functions, including fax, text-to-speech, and speech recognition.

Depending on the number of channels and the amount of computer power required for each function, each DSP may be dedicated to a given function (or functions), a given channel (or channels), or any combination. A DSP's programmability also makes it possible for the host to download new functions and change the system configuration on-the-fly, which is particularly useful for CTI servers. In such an architecture, adding a new function to the system is as easy as loading software modules for that function, and increasing the number of lines is as easy as adding more DSP cards to the network.

John Lynch is the vice president of Ariel's newly formed Computer Telephony Business Unit. Prior to joining Ariel, John was Director of Modem and Multimedia R&D at AT&T Bell Laboratories. Ariel provides high-density data modem solutions plus fax and voice solutions. For more information, call Ariel at 609-860-2900 or visit the company's Web site at www.ariel.com.

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zealously downsized (and hence understaffed) corporations can no longer perform for themselves.

WHO'S USING SOHO-STYLE CTI?

SOHO-type CTI solutions aren't just for small companies. We're also seeing interest from individuals and large companies.

• ***Individuals.*** Millions of SOHO/multimedia PCs have been purchased by families across America. These computers, coupled with powerful telephony-enabled software such as GoldMine, ACT, and TeleMagic, have shown users that a computer can dial telephone numbers for you and deliver screen pops when you get an incoming call. These users sometimes express frustration that they can't enjoy the same features at the office!

• ***Small Companies.*** With the growing availability of ISDN technology, a small company (or even a home office) can have the equivalent of PBX functionality at a tiny fraction of what a PBX would cost. (See the articles on this topic in this issue.)

Your local telco already has an expensive switch in its central office. If you use the new technology, you can take advantage of this switch by purchasing CENTREX features on an as-needed basis. Central office types of products have come a long way, allowing any small or home-based office to seem like a large corporation. SOHO products like the Creative Labs' PhoneBlaster allow a small company to have advanced telephony features (voice mail, fax broadcasting, fax on demand, pager notification of incoming messages, etc.) for a few hundred dollars.

• ***Large Companies.*** Large corporations are already experimenting with telecommuting solutions that allow workers at home to interact with workers in the office. Groupware and Intranets are key technologies here, but they are only the tip of the iceberg. Soon, any home telephone will be able to serve as an extension of a corporate PBX. For example, an electronic translator sitting between the PBX and the local SOHO phone can seamlessly deliver the full range of features available on the PBX.

CONCLUSION

CTI is a broad term that encompasses telecommuting, videoconferencing, collaborative computing, and other disparate technologies. By allowing individuals, small companies, and large companies to take advantage of these technologies, CTI makes us much more flexible in how we manage our work lives. (Just one example: companies can now hire people that were once out of reach due to geographic or other limitations.) I am proud to be part of an industry that helps people become more efficient, and thus gives them the freedom to accomplish more at work, in their families, and in their communities.

Sincerely,

Rich Tehrani (rtehrani@tmcnet.com)
Publisher, CTI magazine

Comments, TIPS, Suggestions?

We at **CTI**™ magazine invite our readers to contact us with comments, questions, or suggestions. If you have any Feedback, we encourage you to send e-mail (**editorial@tmcnet.com**), or snail mail (CTI, 1 Technology Plaza, Norwalk, CT 06854).

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