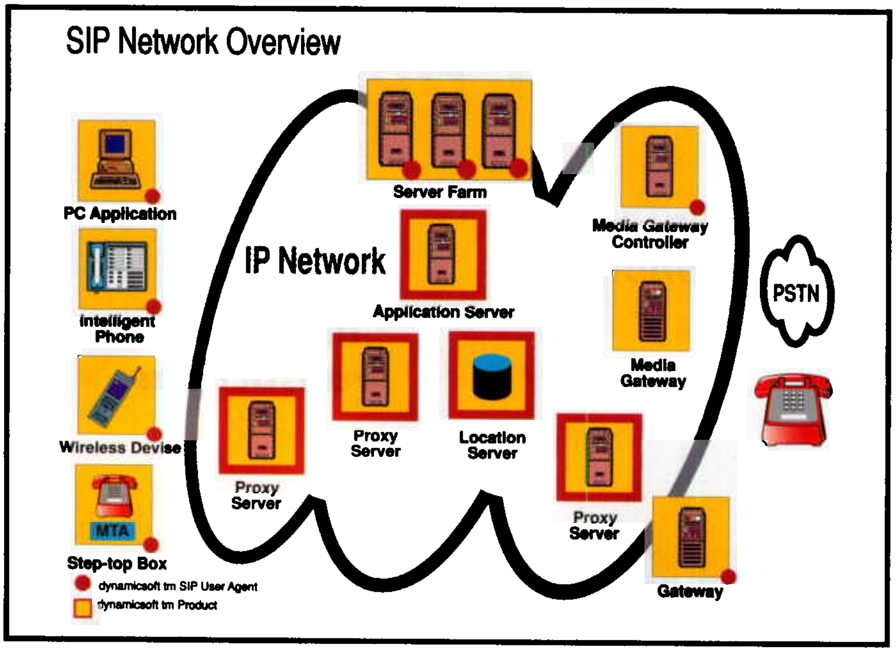


Dynamicsoft makes SIP software for services and users agents. The latter can be embedded in any variety of client devices. The former serve as a basis for a distributed network of apps and services.

BROADSOFT

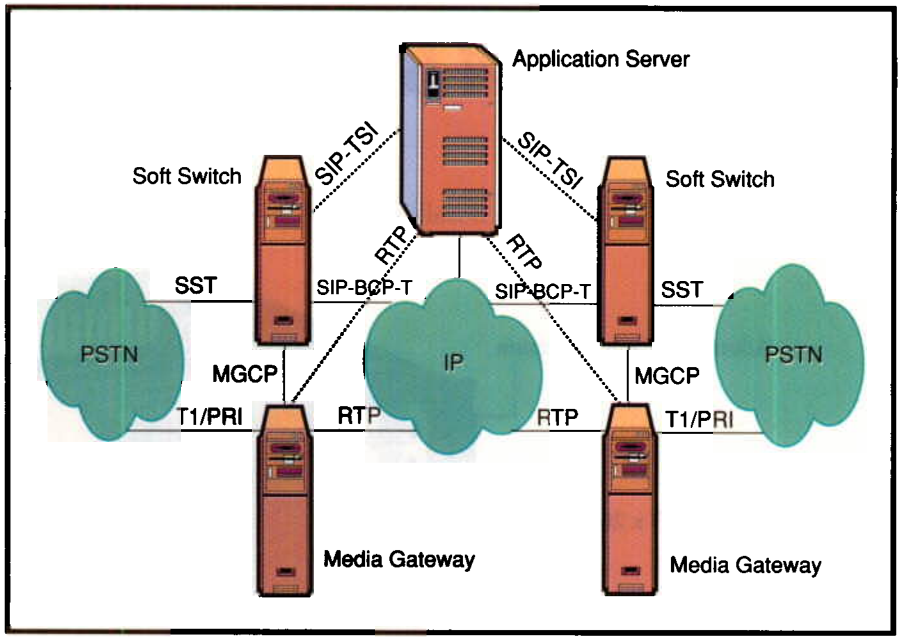
Broadsoft (Gaithersburg, MD — 301-977-9440, www.broadsoft.com) makes an application server and service creation environment that was recently made available to the market. **BroadWorks**, as the system is known, replaces class 5 features in a converged network, and adds enhanced services like messaging, auto attendant, and conferencing. The scope of applications can even extend to intelligent call routing, as well as personal communications services like find-me/follow-me. The system as a whole includes an integrated media server (for functions like IVR and conference setup), along with the software framework for creating and delivering applications. In addition, it incorporates a web server that can be used to give subscribers browser-based access to their service profiles and accounts.

Architecturally, BroadWorks could sit behind a series of network gateways interfacing to the PSTN and SS7, or behind a softswitch and media gateway connecting to the packet network. The idea is that users (most likely small- to medium-sized businesses) would connect to the data network either with an IP phone or behind a gateway at the customer premise, and access network services hosted by their CLEC. While all of BroadWorks services can be delivered in a "legacy" environment as well as a packet network, client endpoints that are connected through IP can provide a much more elegant way of delivering web-integrated services, particularly if they can communicate using SIP.

Beyond supporting SIP as a call setup interface to end devices, however, Broadsoft is also pursuing the possibility of using the protocol as a standard for interfacing between a softswitch and an application server. To this end, the company is working on the SIP-TSI initiative with IPeria, as well as participating in SIP bake-offs to promote interoperability. So far, Broadsoft reports they've found interoperability a much more readily obtainable goal with SIP than with H.323.

DYNAMICSOFT

There's no questioning dynamicsoft's (East Hanover, NJ — 973-736-6580, www.dy-



SIP-TSI is a proposed standard interface for connecting softswitches to application servers in the network. The specification is currently being submitted to the International Softswitch Consortium by a group of companies led by IPeria. **(See page 72)**

namicsoft.com) commitment to, or innovation with, SIP. Jonathan Rosenberg, one of the protocol's co-authors, serves as the company's chief scientist, and Eric Sumner, a former CTO and colleague of Rosenberg at Lucent, was recently named president and CEO.

Although the company is young, it is moving forward extremely quickly with developing a line of SIP-based network servers, forming multiple strategic partnerships, and even securing its first customers. (The day I visited dynamicsoft in March, they were moving from their first offices to a facility more than ten times as large). dynamicsoft's product line, grouped under the name **eConvergence**, includes a SIP proxy server, location server, user agent, and application server. The latter, of course, can be seen as the most valuable, least generic element in the mix, though at the moment dynamicsoft's other products are equally if not more crucial in defining a next-gen IP architecture.

The proxy server is essentially responsible for call routing and session management. It also performs redirect functions, routing to media gateways, and user authentication. While there seems to be some functional overlap between a SIP proxy server and a softswitch, dynamicsoft takes care to distinguish between the two, and sees the softswitch as necessary for interfacing to the PSTN, while its own product is firmly rooted in IP. Before routing a session to its addressee, the proxy must query either a domain name server or a SIP location server. The purpose of the latter is to maintain user profiles and provide subscriber registration. The registration function is one of the more unique aspects of the SIP model. It effectively provides a mechanism whereby a user registers with the network each time she comes online, and can access individual profiles (contained within the same server) that specify information for routing based on a number of different criteria. On the other side of the proxy server lies the SIP user agent, which is actually a piece of software that gets embedded in a range of other boxes — from a softswitch to an IP-PBX to an IP phone to a gateway. The user agent initiates and manages the basic connection between two endpoints. It is available as either a C++ or Java program, and incorporates an extensive API and development toolkit. The user agent also supports SIP BCP-T (for softswitch-to-softswitch communication), as well as the SIP control processing language (CPL) and the common gateway interface (CGI).

The application server, which will become available this summer, will provide a horizontal platform for interfacing to and creating new applications to run on the SIP-based network. Like the user agent, the application server will support SIP-specific interfaces in addition to general Internet protocols, languages, and APIs.

dynamicsoft's most recent news includes being chosen to build out an end-to-end SIP network by Level 3 Communications, as well as working with smaller providers like I-Link and eStara to SIP-enable their networks. The company also recently announced it would partner with Aravox to develop a SIP-controlled firewall for the carrier market.

IPERIA

IPeria (Burlington, MA — 781-993-3500, www.iperia.com) makes an enhanced services platform, **IPeria Service Node**, that combines elements of a media server and an IP applications server, but remains independent of network transport. While IPeria's product, as even the company's name would indicate, is clearly aimed at converging carriers, it can deliver its core services, like unified messaging and voice-activated e-mail, over the PSTN as well as over a packet network. The guiding idea is to let users access any type of communication (voice, fax, e-mail) from any device (phone, wireless, web, Palm, etc.).

One result of IPeria's hybrid approach is a proposed application of SIP that uses the protocol in interesting ways to merge legacy and next-gen infrastructures. Specifically, IPeria, under the direction of CTO Greg Girard, is drafting a spec called SIP-Telephony Service Interface (SIP-TSI) that seeks to define a standard way of communicating between a softswitch and an application server. The main purpose of SIP-TSI is to allow the application server, through the mediation of the softswitch, access core switching and DSP resources located in a media gateway, in order that these resources do not have to be replicated locally in the app server itself (as would generally be the case in a PSTN/IN implementation). DSPs located in the media gateway are used to transmit and receive fax, and for signal detection and signal transformation of voice and DTMF tones. While SIP itself does not define commands for mid-stream control of media transmissions (as is necessary, e.g., for collecting DTMF tones), an adjunct called the SIP "INFO Method" lets mid-session control messages be passed between two SIP user agents (which in this case would be contained respectively within the softswitch and the application server). SIP-TSI, which is being submitted by IPeria and other vendors to the International Softswitch Consortium for approval, is one example of how the protocol can be extended in the network, and how elements such as a "user agent" are not necessarily bound to specific hardware or software devices.

IPeria's Service Node platform recently became generally available, along with unified messaging and voice-activated e-mail (incorporating speech rec and text-to-speech) as the first two apps. Other personal communications services will be released over time.

IPVERSE

ipVerse (Sunnyvale, CA — 408-830-3200, www.ipverse.com) makes a softswitch that leverages SIP in a number of ways. The product, **ControlSwitch**, has a relatively clearly defined role in the network: It resides between the layers of media gateway/trunking equipment on one end, supplying call control and addressing services to these devices, and service delivery platforms, providing APIs and open interfaces, on the other.

One thing that ipVerse prides itself on is its openness in communicating with other systems, whether on the level of signaling or of APIs. Not only does the control switch support the full range of next-gen and legacy signaling protocols — SIP, MGCP, H.323, IPDC, Q.931, SS7 — but it boasts the ability to interface between any of them seamlessly, allowing communication between devices which on their own would not talk to one another. (This includes communication between the ControlSwitch and other vendors' softswitches via SIP-T). In terms of APIs, JAIN, Parlay, XML-based interfaces, and a host of others make it easier to connect the control-switch into multi-vendor application or policy servers.

Another key differentiator for ipVerse is its scalability: The ControlSwitch has been shown to handle up to one million simultaneous calls.

Beyond the intrinsic values of such openness and scalability, ipVerse's emphasis on these attributes raises an important point — namely, that a softswitch on the market today should be designed with a mutability of function in mind. Because network architectures are currently so much in a state of transition and flux, a softswitch should be capable of easily adapting to the somewhat disparate roles of a signaling gateway in cases of simple modem offload, an enabler of enhanced service delivery in a centralized next-gen network, or a proxy/address server in a network where intelligence is fully distributed to the edges (as may well be one of the effects of SIP). The ipVerse platform takes all of these possible scenarios into account.

LUCENT/ELEMEDIA

Lucent Technologies (Murray Hill, NJ — 888-4-LUCENT, www.lucent.com) is using SIP on two distinct but interrelated fronts. The first concerns Lucent's Communications Software Group, a division that focuses on IP-enabling intelligent network applications. The most recent releases to come out of this group are the **PacketIN** application server and a complementary Java/XML-based service creation environment. PacketIN works in both a circuit switched environment, where it would use SS7 and TCAP to deliver services to the switch, as well as in next-gen or hybrid architectures (which are its primary focus), where it would build on a softswitch (which Lucent also supplies) and a media gateway. In the softswitch scenario, Lucent has developed some PacketIN applications that leverage SIP in interesting ways.

One example is Lucent's Online Communications Center (unofficially referred to as Internet Call Waiting on Steroids), a piece of software which Lucent had previously released but has now enabled for use with the softswitch. In addition to standard Internet call waiting (which gives you a screen pop with caller ID info), OCC lets you accept incoming PSTN calls over the Internet, dynamically re-route calls to another phone number, and gives you other desktop call management capabilities. What's particularly cool, however, is that this service could be fully integrated with circuit-switched IN: Using a TCP/IP interface it developed for its own Service Control Points (SCP), Lucent can trigger a SIP message directly from the SS7 network to the PacketIN server, which in turn responds with instructions about what to do with the call. While this level of fairly involved internetworking may seem a bit complex for an Internet call waiting app, it raises potentially intriguing possibilities for things like network-based contact center routing — another service Lucent is offering in conjunction with PacketIN. (Of course, running these services in a fully next-gen environment, i.e. over IP, makes for a much more elegant scenario, and lets you program everything using standard APIs — JAIN, Parlay, etc.).

The second major implementation of SIP within Lucent comes through its **elemedia** (Holmdel, NJ — 888-elemedia, www.elemedia.com) subsidiary, which developed a SIP Server on the basis of the Lucent softwswitch. The server performs authentication, registration, and redirect, in addition to some of the same type of features that could run on a softswitch (personal mobility, time-of-day routing, etc.). The SIP server is a newer addition to elemedia's product line, which also includes an H.323 protocol stack and gatekeeper software. The project was begun last year, under the direction of Jonathan Rosenberg (co-author of SIP) and Eric Sumner, both of whom have since moved to dynamicsoft.

NETERGY

Netergy, formerly 8x8, (Santa Clara, CA — 888-843-9898, www.netenergynet.com) recently announced the availability of its **Netergy Advanced Telephony** system, a hosted IP-PBX for service providers that makes interesting use of both MGCP and SIP. Architecturally, the product centers on a piece of Java-based software, the iPBX, that runs in a distributed fashion across a cluster of Sun Netra servers. The iPBX software manages call setup and teardown, and implements standard features like transfer, forward, and hold. At the customer premise, 8x8 offers a number of options. Currently, Netergy would use 8x8's Symphony Media Hub, a two- or four-port gateway that lets you hook up analog phones and fax machines, and connects to a router via Ethernet. In this case, MGCP is used to communicate between the hub and the "switch" (which really should be thought of more like a media gateway controller). 8x8 also offers a larger, 24-port gateway, and will interoperate with similar CPE products from Cisco. A much sexier solution, of course, would involve putting IP phones at the customer premise, and 8x8 has approached this on a couple of fronts. One is to let customers use IP phones from third-party vendors, and interoperability trials with Cisco and PingTel are underway. Another is using IP phones built around 8x8's Audacity T2 Processor (controlled by MGCP). 8x8 also offers a browser-based soft client that performs third-party call control.

On the back-end, the iPBX gives you a number of options in terms of network interfaces. The most conventional would be to an H.323 gateway/gatekeeper, and out to the PSTN. But you could also connect the servers to a SIP-compliant softswitch and use IP throughout the network.

While a lot of the interfaces for both end devices and network elements depend on products that are still being developed by others, 8x8 is remaining very open and supportive of most available protocols.

PINGTEL

After several months of quiet buzz, Pingtel (Woburn, MA — 781-938-5306, www.pingtel.com) has unveiled its SIP-based IP phones to the public. Having now seen and heard about the product, called **xpressa**, we think it could very well be the first to deliver on the intrinsic, but as yet elusive, promises of IP telephony.

Simply put, Pingtel built its product from an Internet mindset. CEO Jay Batson is very clear about what this means: Intelligence in the endpoints. Batson explains that in an IP architecture, "the phone should be an open, extensible platform for delivering new software features. Ostensibly it's a piece of hardware, but really our product is a software platform for running Java applications." And when Pingtel talks about openness, ex-

Pingtel's xpressa looks and feels (mostly) like a phone, but extensive SIP support and an embedded Java Virtual Machine make it more like an Internet appliance.

tensibility, and Java applications, they aren't just paying lip service. The phones have their own, embedded JVM, a Sun developers' toolkit, JTAPI interfaces, and Pingtel is going a long way to provide resources like style guides and online communities to any potential Java developers.

The architectural model behind these initiatives is basically that some core telephony features (generally transparent to the user) will remain located at central points in the carrier's network, while enhanced services will be thoroughly distributed, and accessible through the phone client as through a web browser. In fact, because these "services" will themselves be Java applets or HTTP-based content, any logical distinction between the web and an IP-based network of intelligent phone clients begins to break down altogether. This, of course, is a completely different way of thinking of "enhanced services" than what we know today. And the difference lies partly in the fact that the "service provider" as such (ISP, CLEC, etc.) is no longer the sole supplier of applications to a user (a proposition about which carriers, of course, have some mixed feelings). Rather, new software could now be downloaded to a phone exactly as one downloads a new app to a Palm Pilot.

In addition to leveraging a Java framework and significant internal processing power, a key enabler for Pingtel has been its early and extensive incorporation of SIP. According to Batson, the company was attracted to SIP from the beginning for its natural extensibility and for the power it puts in the hands of intelligent endpoints (i.e., SIP's peer-to-peer negotiation model for setting up a session). Batson takes care to note that, like HTTP, "SIP as a protocol is simply a vehicle for delivering applications": That is to say, the protocol itself should not necessarily become involved in defining those applications. In this he echoes SIP's authors, and goes on to explain that with xpressa, any number of Internet protocols (HTTP, LDAP, MIME, etc.) might get involved in the execution of phone applications. SIP's inherent ability to integrate with these other protocols only makes it a more natural choice for the platform.

Pingtel expects its phones will be primarily distributed through service providers, in the context of IP Centrex or virtual PBX applications. They also plan to form OEM relationships with IP-PBX manufacturers. Beta tests are just getting underway.

SIEMENS

Siemens (Santa Clara, CA — 408-492-2000, www.siemens.com) has been doing some interesting and impressive work with SIP in recent months, mainly in the area of IP telephones. The **LP5100**, Siemens' first IP phone design, was also one of the first to market to use a standards-based architecture (with H.323 in its original iteration). Now Siemens is offering the same phone model, using all the same hardware as in the original version, but with the option of a SIP protocol stack incorporated instead of H.323. While the company will continue to offer both protocols as software options (and is looking at adding MGCP), they do report finding certain advantages to SIP. According to Siemens' Joan Vandermate, SIP has shown better performance than H.323, with faster call setup time and increased resource efficiency, using the same CPU and other hardware components. The direct implication of this, Vandermate points out, is that a richer feature set could be added to a SIP phone without necessarily requiring any hardware upgrade. And this is in line with Siemens' overall approach to the IP phone market, a crucial

Siemens is making a strong push to become a mass-market vendor of standards-based IP telephones. Support for SIP has recently been added to its product line.

tenet of which is keeping the cost of the device down as far as possible without sacrificing quality or availability of features.

Siemens plans to market its SIP-based version of its IP phone as an endpoint for IP Centrex applications, and will aggressively pursue service providers and carriers as potential customers for the product in the coming months. Initially the phones will offer all the basic functionality necessary to do Centrex features, but Siemens remains open to developing enhanced feature support as it is called for by the market. Likely what will emerge will be a diverse line of IP phones (Siemens is already about to release a scaled down, less expensive version called the LP2100) that address everyone from consumers to high-end business users. Having an already robust manufacturing and distribution plant in place (through which Siemens' Gigaset wireless phones are currently produced in mass quantities) will provide a strong foundation for entering the market.

SYNDEO

While softswitch development to this point seems to have followed a progression from infrastructure (signaling gateway, call management functions) to enhanced services, Syndeo (Cupertino, CA — 408-861-9800, www.syndeocorp.com) is taking a slightly more accelerated approach with a system intended to provide class 5 features from the outset. Nevertheless, the **Syndeo Broadband Services** system does much more than simply replicate class 5 functionality.

While the product contains a component that calls itself a softswitch, it recognizes that the softswitch function is very loosely defined, and highly dependent on the environment in which it is deployed. The Syndeo Softswitch, therefore, is built to be able to act as a call agent in an MGCP scenario, a proxy server in a SIP network, a gatekeeper in H.323, or an SSP in SS7. Essentially what this means is that the product supports all of the above as embedded interfaces and can employ them as necessary.

The other unique aspect of the Syndeo system is a web-based management and control offering, Communication Manager, that subscribers can use to access their service profiles. Not only does this let users configure features and options via the web, but it can also be adapted to run as a persistent JAVA applet to serve mobility services. For instance, a user could run the applet on her laptop, and use it to assign her local profile (phone number, features) to a remote phone. Communication Managers also integrate directly with PDA and contact management applications.

With regard to SIP, Syndeo has already formed partnerships with 3Com, Pingtel, and AudioTalk, to support their SIP-based clients. For service creation on top of its platforms, Syndeo supports the JAIN API, along with other open standards.

TELEPHONY EXPERTS/NUERA

Telephony Experts (Los Angeles, CA — 310-445-1822, www.telephonyexperts.com) and Nuera Communications (San Diego, CA — 858-625-2400, www.nuera.com) have recently partnered on an offering that joins the former's **Talking IP** product with the latter's VoIP gateways in a SIP-based environment. Telephony Experts' Talking IP is a software-only enhanced services and billing platform. Nuera makes high-volume trunking gateways for service provider networks. Both companies recently announced that they would be supporting SIP in the latest versions of their products. The way the combined scenario works begins with a call entering the Nuera media gateway and being passed to a media gateway controller or softswitch (which Nuera also produces) to receive routing instructions (here the means of communication is MGCP). If the MGC detects a request for enhanced services (e.g. prepaid calling card, messaging, etc.), a SIP session is established (via a re-invite command) between the media gateway controller and the Talking IP server.

Talking IP uses a SQL database, and includes its own service creation environment. Nuera's products support multiple standards, including SIP-T for softswitch-to-softswitch communication. The companies' relationship is non-exclusive.

TI/TELOGY

Texas Instruments (Dallas, TX — 800-477-8924, www.ti.com) subsidiary Telogy Networks (Germantown, MD — 301-515-8580, www.telogy.com) has recently come out with SIP support in a number of implementations of its **Golden Gateway** software. Golden Gateway is voice-over-packet processing and conversion software that runs on TI's programmable DSPs and RISC/CISC microprocessors. At the DSP level, Telogy's software performs tone detection, echo cancellation, and a number of other functions related to voice/fax/modem signal processing. This software is also responsible for processing telephony signals, such as FXO/FXS, BRI/PRI, and QSig. Once a call has been packetized and stamped with RTP headers (all of which takes place in the DSP), the packets are sent to the microprocessor (typically a part of the same chipset), which runs a separate but related part of the Telogy software to implement an appropriate packet signaling protocol. Telogy's first deployments used H.323 only for instances of voice-over-IP, but the company has since added SIP and MGCP to its repertoire.

One notable instantiation of SIP is a reference design for a voice-enabled cable modem, announced last December. Since then, Telogy has also incorporated SIP into its integrated chipset for IP phone manufacturers — a product that combines TI DSPs, a RISC processor, an embedded Ethernet switch, real-time operating system, and Golden Gateway software. According to Nancy Goguen, Telogy's VP of marketing, implementing SIP on Telogy's existing product was a relatively easy project, and one that could be carried out in-house on the basis of the published specs, as opposed to its original implementation of H.323, for which Telogy chose to license the RADVision protocol stack. Other benefits of SIP which Goguen cites include its decentralized nature, its scalability, and its backward compatibility. Still, she notes, uptake from customers is still mixed ("it's about 50/50 in the IP phone world"), and she stresses the importance of multi-protocol support from TI's standpoint for at least the forseeable future.

TRILLIUM

Trillium Digital Systems (Los Angeles, CA — 310-442-9222, www.trillium.com) writes source code for a variety of protocol stacks within IP, ATM, SS7, and others. Trillium software is typically embedded within a microprocessor, but the company emphasizes what it refers to as "portability" — being able to abstract the core software itself from the individual processor, OS, or compiler, and thus give equipment manufacturers a greater degree of flexibility. A set of common interfaces are essentially all that needs to be tweaked in order to move a piece of Trillium software across different underlying systems.

About a year ago, Trillium released its first H.323 stack and has since licensed the product to gateway manufacturers and client software developers. More recently, with regard to VoIP, the company has been focussing on MGCP (which it demoed at Spring VON), as well as multi-protocol label switching (MPLS), for which it plans to release source code in the near future. SIP software is a bit further out, with a release expected later this year. While individual protocols are being added incrementally, Trillium's CEO Jeff Lawrence makes it clear that the company has a long-term vision of how the new network architecture is shaping up. Although Trillium will continue to support a wide range of (sometimes competing) protocols, Lawrence is pretty confident that "SIP will eventually become the signaling protocol that replaces SS7." Of course, Lawrence does not expect that this shift will occur overnight, or even necessarily over the next couple of years, but he does feel that SIP's simplicity, scalability, and recent momentum in terms of industry support will propel it forward as a successor to H.323.

Lawrence also makes the important point that SIP will need to interwork with other protocols in the network to supply functions like quality of service, security, transport, and gateway control, and points to MPLS, IP Sec, SCTP, and MGCP/MEGACO as crucial parts of the network architecture. Although most of these protocols are only now emerging, and commercially available software to support them is still being developed, Trillium says it plans to supply customers with frameworks that allow for interworking as standards are adopted.

UBIQUITY

Ubiquity Software Corporation (Kanata, ON, Canada — 613-271-2027, www.ubiquity.net) makes an application server and service creation environment they call **Helmsman**. Ubiquity's position is essentially as an enabler of applications, rather than a developer of them. While the company will offer certain core features on top of its platform, and can do customization for individual service providers, the idea is basically to make it as easy as possible for a third-party (either an outside developer or even the service provider himself) to deploy whatever applications are desired. In this way, Ubiquity's product is like a series of connectors or hooks that interface, on the one hand, to endpoint devices (phones, PCs, PDAs, etc.) and on the other to network elements (SIP servers, gatekeepers, call agents, etc.). In the case of the latter, Ubiquity is supplying its own SIP server, to perform proxy, redirect, location, and registration services. Nevertheless, the company still considers the "service enabling" layer as its sweet spot, and will partner with other vendors of network equipment to supply carriers with the components necessary for call control and packet transmission.

One significant customer win for Ubiquity has been **I-Link** (Draper, UT — 801-576-5000, www.i-link.com), a next-gen service provider building out a network to host distributed communications applications. I-Link is using a combination of the Helmsman server and a Helmsman Desktop client that Ubiquity designed to integrate with Windows PCs. The client app essentially lends call control and VoIP capabilities to software like Outlook or Goldmine, and can also be integrated with web servers and used in e-commerce applications. Helmsman Desktop incorporates a SIP user agent to communicate with elements in the network like a softswitch. In addition to SIP, Helmsman supports H.323, MGCP/MEGACO, and TAPI.

UNISPHERE

Unisphere Solutions' (Chelmsford, MA — 978-848-0300, www.unispheresolutions.com), who offers a line of routers, switches, and gateways to the carrier market, has been devoting its most recent energy to developing its high-density softswitch, the **SRX-3000**, and its service creation environment (a software platform to create and deploy applications in the network). At this point, Unisphere says that its main goals — and most pressing concerns in terms of securing customers — are to ensure interoperability and any-to-any switching, regardless of the underlying protocol. Nevertheless, the company has found session iniation protocol to be an asset for performing particular tasks like softswitch-to-softswitch communication (using SIP BCP-T, a more formalized version of SIP+). Even more recently, Unisphere, like others, has been looking at SIP as a way of interfacing to its service creation nodes, which would likely speed the deployment of features. Unisphere likes the fact that SIP is lightweight, relatively easy to implement, and is more of a native protocol for public networks than H.323 is.

Unisphere's "any-to-any" switching philosophy is carried out by way of a mechanism within the softswitch that normalizes every incoming interface and implements common call handling procedures within the server. Because this was a goal from the beginning, they've constructed the system in such a way that it's easy to add a new protocol or variant without fundamentally altering the call control scripts. In its initial release, scheduled to have gone into limited availability at the end of March, Unisphere will offer interworking between SS7, SIP, MGCP, and H.323, along with limited class 5 features. The product will become generally available in the third quarter. ✲