VOICE-DATA CONVERGENCE

SIP State Of Affairs

Steven Guthrie

SIP has come a long way in eight years. But its disruption of the enterprise voice market is just beginning.

Henning Schulzrinne had no idea his 1996 Internet Draft (ID), "Simple Conference Invitation Protocol" would have such a lasting impact on the world of enterprise communications. Nor did Mark Handley and Eve Schooler from Caltech, who simultaneously offered the ID, "Session Invitation Protocol." It was 1999 before the IETF truly recognized the trio's work, by then universally known as Session Initiation Protocol (SIP), and spun the effort out of its original place with the MMUSIC working group to establish the SIP Working Group.

While Handley and Schooler have far less visible roles in the world of telephony today, Schulzrinne is still very active in defining and developing SIP and other Internet techniques as a technology enabler for new enterprise communication solutions. He described SIP as "a small fledgling effort [that] slowly grew into a whole industry—It's the largest systems project the IETF has taken on in its history."

While the ITU-backed H.323 and MGCP (a compromise IETF/ITU signaling protocol) initially commanded most of the attention of voice over IP (VOIP) carriers, SIP caught on with many small niche players. They were trying to build enhanced application servers for both the carriers and the enterprise. Companies like BroadSoft, dynamicsoft, Hotsip, Ingate Systems, Jasomi Networks, Pingtel, Snom, Ubiquity Software, Xten Networks and Zultys Technologies were well known among passionate SIP technologists, but they were hardly mainstream brands to enterprise customers.

These SIP pioneers, and others, are still small, but they are surviving—a minor miracle given the multiyear telecom nuclear winter—and SIP itself is no longer confined to startup vendors. Thanks to Microsoft, IBM and MCI, among others, SIP has become a telecom term with widespread recognition. Recently Avaya, Cisco, Lucent Technologies, NEC and others have announced significant SIP products.

SIP is now central to all the latest solutions for telephony, ranging from hosted services to vendor-proprietary platforms, from handsets and softphones to open source software. This article highlights these and other recent SIP-related advances.

Will Hosted SIP Services Eclipse Centrex?

Over their 30-plus-year history, Centrex services have never accounted for more than 15 percent of the business lines in the U.S., no matter how hard the service providers tried. SIP could break that barrier, given the number of providers that have built SIP-facilitated networks and are offering hosted IP voice services.

MCI (nee WorldCom) deserves a lot of credit for promoting this use of SIP. With the affable Henry Sinnreich omnipresent at industry conferences and his technical counterpart Alan B. Johnston helping define the protocol, WorldCom was the first and most visible carrier to offer a hosted SIP service, as far back as 1998. Other interexchange carriers (IXCs), local exchange carriers (LECs) and competitive LECs (CLECs) dabbled in H.323 and MGCP trials, but WorldCom inspired and pushed the SIP community to grow and to meet its requirements. MCI Advantage, the company's current hosted SIP service, is now available in 95 metropolitan markets across the U.S. (For more about where H.323 is still holding on, see "When Will IP Videoconferencing Mean SIP Rather Than H.323?" p. 32)

Today, Qwest, Verizon, SBC, AT&T, Bell Canada and BT are among the giants offering SIP-based hosted IP voice services, according to Jim Hourihan, VP of marketing and product management for Acme Packet.

Hourihan maintains the hosted market has been held back, however, "due to the difficulty of traversing NATs [network address translation] and firewalls." Acme Packet, Jasomi Networks, Netrake and others have responded with session

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border controllers that solve these problems and also allow IP voice service providers to peer with one another. Acme Packet has deployments with Fox Communications, PointOne, voiceglo and Z-Tel Communications, which Hourihan regards as evidence the hosted market is ready to challenge the traditional 15-percent Centrex barrier.

Enterprise buyers also might consider hosted SIP services for more locations than traditional Centrex, which was often constrained by geographic limitations and onerous long-term contracts. And there may be other new uses for hosted SIP, said Christine Hartman, vice president, voice over packet networks, Probe Group, "such as adding a few more lines instead of replacing a maxed-out PBX, or for a short-term project or convention."

SIP, Interoperability And The Enterprise

On the systems front, early enterprise VOIP solutions didn't feature SIP. Systems from 3Com (nee NBX), AltiGen, Artisoft, Cisco, Interactive Intelligence, Mitel and Shoreline were packet- and IP-based, but otherwise as proprietary as traditional TDM platforms. The VOIP business model was like TDM, too, with each vendor supplying its own platform, handset, gateway, call processing and end-user feature set.

Despite SIP's momentum, major platform vendors continue to rely on their proprietary protocols

Within a few years, Cisco had combined its data networking role and IT relationships with its massive marketing and sales prowess to challenge incumbents Avaya and Nortel for enterprise communications market leadership. In those days, convergence meant two things: Converge voice onto the data network, and sell telephony to the IT buyer. To this day, the toughest challenge, especially for traditional voice vendors, is learning to sell what data people buy.

It was pressure from the data customer, accustomed to standards and interoperability, that turned the VOIP conversation in the direction of standards and interoperability, and to SIP. "Customers did not say 'We want SIP.' Customers said 'We want interoperability,'" said Greg Zweig, product manager for voice solutions, 3Com. "Then the development community got together and defined SIP as the interoperability solution."

Vendors hadn't made their products interoperable before because of competitive, not technical issues, according to Jim Davies, chief technical officer, Mitel Networks. "Technically, there was nothing stopping us from interoperability before [SIP]," he said. "Earlier attempts in the PBX market, such as ISDN and Q.SIG, were stymied by business pressures, not technical ones. SIP allows us to mix and match the service and business models."

Today, vendors can test the interoperability of their SIP implementations with other vendors three times a year at the SIP Forum SIPit events. At the 14th SIPit event this past February, 55 companies participated. Moreover, SIP is core to the VOIP strategy of 3Com, Avaya, Alcatel, NEC, Nortel, Mitel, Siemens and a host of second-, third- and fourth-tier players.

In fact, "2004 is probably the year of SIP, when SIP becomes real and support for SIP turns into real products," said Jim Su, senior manager, product and solutions marketing at Avaya. Product announcements from February illustrate the surge of SIP support and offerings for SIP-based enterprise communications:

- Avaya announced the Converged Communications Server, a SIP server that connects to Avaya's Communications Manager converged TDM-IP platform; a new SIP-enabled softphone; and a new SIP load for the Avaya 4602 IP Telephone.
- Cisco included SIP support on the "network side" of Call Manager 4.0 as part of an announcement that was more focused on video telephony than SIP, but which represents "a step in the SIP direction," said Hank Lambert, director of product marketing, enterprise call control solutions. In spite of Cisco's long-term role in the development of SIP and its many products that support SIP, Call Manager still supports only the proprietary Station Client Control Protocol (SCCP) or "Skinny," for handsets, protecting the handset business for Cisco's massive VOIP strategy.
- Lucent Technologies extended its portfolio of enterprise network solutions with a new enterprise telephony offering, based on the BroadSoft BroadWorks SIP-based network communication solution and a homegrown SIP-based softphone. The solution is an extension of an earlier agreement with BroadSoft for a carrier-based application, and represents a clear indication of Lucent's return to the market for premises-based enterprise communications.
- NEC announced the SV7000S, a SIP server that interoperates with traditional NEC phones based on its proprietary IP PROTIMS handset protocol. At the same time, NEC announced a roadmap for a variety of SIP-based handsets, including softphones and wireless handsets, scheduled for release late in 2004 (see *BCR*, March 2004, pp. 60–61).

Clearly, the major platform vendors continue to rely on proprietary protocols for both their platforms and their handset features, as demonstrated by these recent announcements. This indicates that their support for SIP, although growing, still

At the current pace, it will take 4–6 years to finish the SIP standards work

When Will IP Videoconferencing Mean SIP Rather Than H.323?

E. Brent Kelly

For millions of PC users running the Windows XP operating system, SIP-based video is already the *de facto* video standard, since it is being used (under the hood) in Microsoft's Messenger for Windows XP. H.323 has been the traditional choice of Polycom, VCON, and others for their desktop videoconferencing systems, but their sales of roughly 20,000 units per quarter are dwarfed by the millions of copies of Windows XP sold during the same time frame.

Microsoft's selection of SIP rather than H.323 for Messenger legitimized SIP for desktop video, but interoperability remains a major challenge. First, the details of Microsoft's SIP-based video implementation remain proprietary, and Microsoft will only divulge these details to selected partners. Some of these partners, like RADVision and First Virtual Communications (FVC), have embedded Microsoft's SIP technology in their own products. These video bridging devices handle Microsoft's flavor of SIP video and provide video gateway connections between that and H.323 video endpoints.

Another basic challenge facing SIP in the video market is the lack of an established standard for transmitting video over SIP. Unlike H.323, where everything is specified, including the audio and video compression algorithms used, SIP has no video specifications. Consequently, when a manufacturer states it uses SIP for video, that only means it uses SIP for signaling: That does not imply interoperability with other SIP video clients.

A case in point is Wave Three software, manufacturer of an excellent SIP-based video product. Wave Three uses a proprietary wavelet compression algorithm that is not compatible with Microsoft's H.263 video codec. Consequently, while both Wave Three and Microsoft have SIP-compliant video endpoints, they cannot communicate with each other.

Where do the big group videoconferencing players stand *vis a vis* SIP? Thus far, no announcements have been made by the Big Five (Polycom, Tandberg, Sony, VCON and VTEL) about supporting SIP. In part, this may be because the group videoconferencing market must still support the many legacy ISDN endpoints that use the H.320 protocol, H.323's PSTN cousin. For example, in Europe more than 90 percent of the group videoconferencing market still uses H.320.

To its credit, VCON does support SIP video in its hybrid Media Xchange Manager (MXM)

stops short of complete openness and interoperability.

But there have been very public endorsements of SIP: By Dave Morgan, VP of architecture and planning, Fidelity Investments, at the VoiceCon 2003 conference; by IBM for both its internal use as well as its Lotus Sametime collaboration software; and by the cellular phone community for its 3G handsets and networks; they make SIP the definitive direction for both wireline and wireless. The big enterprise voice vendors will have to come around—eventually.

Explanations Or Excuses?

In fairness, we can't blame the big vendors alone for SIP's slow progress to interoperability. The IETF still has a significant amount of work to do to replicate, in SIP, the hundreds of TDM PBX calling features.

"In the past, there were just a couple RFCs deployed in every user agent," said Schulzrinne in reference to the core element in an endpoint's ability to speak SIP. "We've now moved into work that is more narrowly tailored for fairly limited services that aren't applicable to all products and services."

These "fairly limited services" will give SIP the ability to address the hundreds of features found in the standard PBX and basic desk phone. Admits Schulzrinne, "From the end user perspective, we're really at the beginning of what we can do to replace existing technology."

"The level of activity simply indicates SIP is more than a protocol—we're trying to build a whole system, which is much more diverse and rich than the traditional IETF efforts," continued Schulzrinne. He recently calculated that, at the IETF's current pace, it would take four to six more years of hard authoring and reviewing Internet Drafts and creating RFCs to finish the job.

"We must accelerate the process," he added.

The IETF's deliberate, democratic pace is convenient for the big vendors, as they want to protect their proprietary systems, However, the choices will be limited for enterprise customers who want to move to an interoperable, all-SIP solution. And product, but the company does not have a SIP-based endpoint. We hear rumblings that Polycom will soon support SIP in its MGC video-bridging product, but again, no formal announcement has been made.

To complicate matters further, Cisco recently announced a desktop video product that is tightly integrated with its IP telephony solution. This desktop video product uses Cisco's proprietary Skinny Call Control Protocol (SCCP), derived from H.323. Cisco has the capability to connect to both SIP and H.323 endpoints through a gateway, but, like other large manufacturers playing in the group video world, no SIP endpoint is currently offered.

Both SIP and H.323 have very attractive features, as well as some flaws, along with deeply polarized and vocal proponents. SIP is lightweight and flexible, but not deep enough to specify all the mechanisms for a full video communications system, whereas H.323 contains everything necessary for a very robust video communications system and is inherently interoperable with H.320. The downside of H.323's rigid specification is its complexity, particularly the difficulty integrating H.323 video with other IP-based applications.

Because these standards are so philosophically different, it is likely that only the market will determine if there is a winner, a loser or a draw. While the market is deciding, good opportunities may exist for gateways, such as those provided in the RADVision and FVC products mentioned above.

No IETF/ITU Cooperation Yet

Some have wondered if there are any behind-the-scenes efforts toward collaboration between the standards body for SIP (IETF) and the standards body for H.323 (ITU). Word from both bodies is that this will never happen with regard to SIP and H.323. The standoff is likely more political than technical.

The ITU is a United Nations-structured and sometimes bureaucratic organization that has been around for a very long time, maintaining strong ties with governments and carriers worldwide. The IETF, on the other hand, is newer, grassroots-oriented, and more dynamic. The IETF has significant momentum because it is more akin to the developing, Internet-centric product and services markets.

In the near term, desktop video will continue to be dominated by SIP-based endpoints, while the traditional group videoconferencing market will remain H.323-based; they will meet through SIP-to-H.323 gateways as needed. Enterprises implementing either group or desktop video systems need to be aware of these competing standards and choose vendors that plan to support both seamlessly □

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The platform vendors risk losing their fat margins on desk sets

there will be more delays in what some perceive as the ultimate SIP dream: low-cost, feature-rich handsets from third-party phone vendors.

SIP Phones Are Lining Up

Some see handsets as the second area of major market disruption (after long distance) in the march of voice products and services from TDM to IP. "VOIP initially had the most disruptive effect on the carriers' transmission, and helped create a triangle of competition among the handset, the platform and the network," said Neal Shact, president of telephony solutions reseller CommuniTech. "The battle for the next two years will be among these three industries as each attempts to commoditize the other two."

Platform vendors like Avaya, Nortel, and lately Cisco, have enjoyed the most customer control and have the most to lose in this battle. According to one reseller, handsets can constitute 50 to 65 percent of the total PBX system price.

Action for the SIP handset is now fast and furious, with traditional "hard" phones based on SIP available from platform vendors and from many independents. The problem for platform vendors, if they fully embrace SIP, is that any SIP-compliant phone would work with their platforms. "The challenge they have going forward is replacing the margin they used to get off handsets," said Probe Group's Hartman.

Adding to the threat, SIP phones have matured significantly over the past year. For example, Syracuse University Real-World Labs tested six of them for an August, 2003, technical review published in Network Computing magazine. The tester and author, Peter Morrissey, concluded: "After testing six SIP phones, we feel it's safe to say that SIP for VOIP phones is mature and solid. Those vendors claiming there isn't a viable standard for VOIP phones have just run out of excuses."

With interoperability and functionality seemingly resolved, the next major hurdle is price. Long criticized for being too expensive at $250 and up for each phone, the price hurdle is now just $149, with the introduction of the new Swissvoice

SV-IP10S SIP hard phone.

3Com's Zweig thinks that SIP might disrupt the handset business itself, by blurring the distinction between desk phones and mobiles. "What SIP could do is change all the players," he said. "Nokia could become the handset for everyone."

Nokia? Don't be surprised: a SIP wireless handset could have a major play in the enterprise. "SIP crosses so many boundaries—mobile, PC, PDA, desktop phone, softphone," added Jeff Liebl, vice president of marketing with wireless and wireline SIP developer Ubiquity Software.

Mobility: Not Just For Cell Phones Anymore

With SIP-based softphones now showing up in the enterprise, outside the call center and on wireless PCs and PDAs, mobile telephony is much more than just cell phones. And like the cell phone—which users have accepted despite lower quality, dropped calls and a tiny form factor—the softphone is gaining acceptance despite its ergonomic challenges.

In the early days of VOIP for long distance, customers accepted softphones to save money, even though they had balked previously at the PC-based softphones mandated by computer telephony integration (CTI). Wearing a headset was the least of their complaints: Relying on the stability of the PC for telephony seemed foolhardy. Getting out of a document or spreadsheet application just to answer the phone seemed clumsy. Fielding calls when the PC was turned off, well, it just wasn't possible. Finally, it was difficult to match soundcards and find a good headset or handset.

Although they offered softphones, traditional platform vendors weren't really anxious to convert enterprise phone users from margin-rich handsets to low-cost softphones. "Some people saw it as an alternative to the desk phone, and that raised fear of cannibalizing sales of hardphones," said CommuniTech's Shact.

Today, Shact maintains that new factors are "putting the afterburners on" for softphone usage. "Instant messaging and presence is much better on a PC," he said, "and with the improvements and wider availability of softphones, it's all coming together on the desktop."

The next step will be video, which Shact said, "everybody has either announced or is about to announce." All of which is good for Shact's business as the global distributor of the Clarisys USB handset. This device sits on the desk and replaces the headset, rings audibly, incorporates a speakerphone, and as a USB device, avoids the PC's soundcard.

Platform vendors may be seeing new opportunities for softphones too. A case in point: Avaya's major SIP announcement in February included a softphone that incorporates SIP-based presence capabilities across both voice and instant messaging, plus secure SIP-based IM behind the corporate firewall, IM archiving, "click-to-talk" voice functionality, speech-to-text functionality, and specialty modes for the road warrior and telecommuter. In road warrior mode, the Avaya softphone transports signaling and media over the IP network. In telecommuter mode, the signaling is transported over IP while the media goes over either the PSTN or cell network.

What If It's Free?

While platform vendors may fret about softphones displacing their high-dollar hard phones, a more worrisome thought may be losing market share to Microsoft, or to some open-source startup. After all, softphones—and PBX platforms too—are, essentially, nothing more (or less) than software applications.

The latest development: Open-source PBXs

"For the vendors, SIP still raises the question of 'What does the adoption of SIP mean?'" said CommuniTech's Shact. "And: Is the open protocol simply a stalking horse for Microsoft to take over the voice desktop much the way it dominates the data side of the desktop today?" (For more on Microsoft's forays into IP voice, see *BCR*, October 2003, pp. 18–22.)

Then there's open source. Asterisk is an open source PBX with a global following, said Mark Spencer, who developed the application because he couldn't afford a telephone system for his small Linux support services firm. Asterisk PBX is free to download, although a sister company Digium derives operating revenue through sales of trunking gateway cards, support and development services and OEM licenses outside the GPL.

Asterisk cites a number of enterprise companies in the 500- to 700-user range as well as an enterprise connecting five locations, and another user with 38 locations. A review of the Asterisk online community gives one the sense of both the vibrancy of the users as well as the missing polish that some enterprise customers could find too risky for their organization. Indeed, a lack of documentation, such as a list of interoperable components, led Asterisk adherents to develop a web-based community at www.voip-info.org/asterisk where people can easily collaborate and post documents to share with others.

If it were up to Asterisk, selecting an enterprise communications solution would come down to price. Spencer cites the example of how a 500-user PBX and PSTN gateway capable of 96 simultaneous calls can be put in place for as little $2,300 (phones not included) with an $800 Dell computer and a $1,495 Digium quad-port gateway card. The Digium card, said Spencer, is comparable to a Dialogic (now Intel) component that's typically priced $6,000–$8,000.

Cost was the very reason why Discovery Research Group, a U.S. market research group, replaced four Avaya, Comdial and Panasonic systems with Asterisk. "It has done everything we expected it to do," said Brandon Patten of the Asterisk-based telephony system that serves the four locations and 600 phone positions.

Other free and open-sourced SIP products for the enterprise include Pingtel's instant xpressa softphone and its SIPxchange IP PBX, Xten's X-Lite softphone, Interactive Intelligence's SIP Proxy, Vovida's Vocal SIP Proxy, and free, closed-network calling services including Free World Dialup, Epygi Technologies and others.

Conclusion

According to SIP Forum chairman and longtime SIP supporter Jay Batson, the Forum and the IETF's SIP protocol process make it possible for so many new SIP-based products to continue entering the market.

"The democratic peer-review process in the IETF prevents special interest influence on the protocol specification, and the SIP Forum successfully facilitates extensive product interoperability testing to promote high standards-compliance," said Batson. "And now, with quality open-source SIP software becoming available, the number of SIP-enabled products will grow even faster."

SIP won't just affect telephony

"The concepts behind SIP will be more disruptive," predicts Avaya's Su. "What SIP represents is much more dramatic than VOIP. VOIP was just telephony. SIP is not just telephony, it means all sorts of applications like presence that open doors of capabilities that weren't possible in the TDM world."□

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