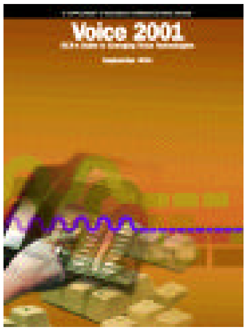

Protocol Wars Threaten VOIP Future

Larry Hettick

Instead of fighting over SIP and H.323, vendors should concentrate on making the two work together.

We all know that the public switched telephone network (PSTN) works. Most of us know that voice over IP (VOIP) works. But the convergence of the two is being delayed by a near-religious debate among industry insiders.

At the heart of the controversy is an either/or choice between the H.323 and SIP protocols. Some developers, vendors and service providers are evangelizing their preferences, but a flawed premise—that one protocol is superior to the other—underlies the debate. In fact, both standards bring positive elements to the table, and work is getting under way to make sure that both will operate in the networks of the future.

Fighting protocol wars at this stage of the game does nothing for either the H.323 or the SIP factions, and it casts doubt and uncertainty over customer perceptions of VOIP and converged services. In a market with billions of dollars at stake, equipment suppliers and service providers should be doing all they can to fully integrate both of these protocols, and to educate customers about the efficiency and new services that converged PSTN/VOIP networks can offer.

Who's Pushing What?

The battle between Session Initiation Protocol (SIP) and H.323 has been brewing for several years now. While SIP has been gaining ground since the IETF standardized it in 1999, H.323 systems, which were introduced in 1996, are more widely deployed (see "Protocol Primer: H.323, SIP and Megaco/H.248," p. 65 and also "Too Many VOIP Standards," in *BCR*, June 2000, pp. 64–69).

AT&T chose H.323 because it was the most mature protocol, according to Ann Machi, product manager for VOIP. "It does what we want to do for our retail and wholesale customers," she said, "especially for legacy integration with our existing network and the customer premises equipment at the edge."

At WorldCom, distinguished engineering member Dr. Henry Sinnreich couldn't disagree more. "H.323 has the wrong addressing, the wrong security, doesn't scale and has single points of failure," he said. "It is extremely complex and has a heavy footprint. With SIP, we're not just redesigning the PSTN, we're investing in new services with the promise of new revenues."

Level 3 Communications and GoBeam, an early converged service provider that resells Level 3 VOIP services, also weigh in for SIP. "H.323 is the PSTN, and SIP is the PSTN plus," said Level 3 senior manager Matt Johnson.

Equipment suppliers are more ambivalent, especially those who established their early VOIP business with H.323 gateways. Today, vendors like Cisco, Clarent, Commworks and Radvision support both H.323 and SIP.

According to Michelle Blank, marketing VP at Radvision, H.323's capabilities allow carriers to make money today. "If you're a large service provider with a need for hundreds of thousands of VOIP ports today, you must use H.323," added Ofer Shapiro, Radvision senior vice president for business development.

At Commworks, Houman Modarres, director for IP telephony, claims that the traffic mix users send across his company's equipment is split about 50/50 between H.323 and SIP, but he acknowledges that the company is "very high on SIP, because a majority of our service provider customers are beginning to deploy it." By comparison, Cisco is "strategically agnostic" relative to the protocols, said Alec Henderson, VOIP product marketing manager to service providers—"although our customers rarely are."

Why Choose Sides?

Service providers have good reasons to select either H.323 or SIP. The early VOIP adopters used H.323 gateways to bypass traditional circuit-switched paths and offer inexpensive international long-distance service. More recently, domestic providers have shown interest in VOIP, but they prefer SIP as a platform for new services, such as unified messaging, click to talk and others. As for the incumbent local exchange carriers, their lack of interest in VOIP

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is understandable: toll bypass efficiencies are not important in the local exchange, and new VOIP-enabled services are not important in a monopoly environment.

As a practical matter, service providers can't afford to fully equip their networks with gear for both VOIP protocols. They can successfully move from one technology base to another, as many did who started with analog cellular networks and then moved to digital, but the transition can be slow, complicated and costly.

Equipment suppliers, who want to sell those upgrades and alternatives, have every reason to remain agnostic. And since both SIP and H.323 are used today in different service provider networks, vendors who want to sell VOIP equipment will simply have to support both.

The major exception to that rule is traditional PBX vendors, who have an easier strategy. Their core business value comes from the feature set their PBX makes possible. Hence, incumbents like Nortel and Avaya have chosen to "IP-enable" their installed base of PBXs with H.323 transmission gateways, and left the traditional call control and feature sets in place. For these vendors, SIP remains a future option.

Newer PBX vendors, like Shoreline Communications and Cisco, who lack an installed base to upgrade, seem more focused on SIP. "While we started with a proprietary software load we call 'skinny,' we have since included SIP, a SIP call agent and MGCP software," said Hank Lambert, Cisco's enterprise VOIP marketing director.

"In our architecture," Lambert continued, "a call agent like Cisco's Call Manager can provide 'skinny,' SIP or MGCP as needed by the network, and can include updates as different standards are needed. The Call Manager can act as a proxy to translate between multiple standards as these standards evolve. H.323 is possible, but we haven't had sufficient demand to include it as an option."

Carriers can't fully equip their networks with both protocols

Protocol Primer: H.323, SIP and Megaco

The amount of traffic carried by IP-encapsulated "protocols" has now eclipsed the amount of circuit-based voice traffic by nearly two-to-one, according to Vertical Systems Group. In the 10 years since this shift was projected, PSTN network experts have been looking for ways to converge the PSTN with packet-based networks like the Internet.

The first such design emerged from the ITU-T's efforts to standardize videoconferencing in 1996. It was H.323, entitled "Visual Telephone Systems and Equipment for Local Area Networks that Do Not Provide Guaranteed Quality of Service." Subsequent additions, spurred by the fast-developing Internet, moved to include standards for packet telephony and wide area networks in the H.323 family.

"H.323 was originally created for multimedia, and then enhanced for voice" noted Radvision's Ofer Shapiro. "The enhancements—features like call hold, call transfer, call waiting—were driven by traditional telephone philosophy. The H.323 standards developers asked: How do we offer these PSTN capabilities on the Internet?"

H.323 specifies four elements designed to interwork with the PSTN. They are:

- Terminal, which provides a multimedia codec (coder/decoder) user interface to other H.323 system elements.
- Multipoint Control Unit (MCU), which controls sessions with three or more users.
- Gateway, which provides the interface between the circuit-switched and packet-switched network.
- Gatekeeper, an element that provides network management and call control functions.

The newer Session Initiation Protocol (SIP) was introduced by the Internet Engineering Task Force (IETF) in March, 1999. SIP was designed with many familiar characteristics borrowed from protocols like Hypertext Transfer Protocol (HTTP). SIP applications can be developed using HTTP, which increases the pool of development resources.

"SIP was designed by those who were big believers in the Internet and Internet technologies," explained Radvision's Shapiro. "Their first concern was to bring voice over the Internet, and then enhance for value-added features like multimedia."

Houman Modarres of CommWorks agrees. "The SIP model follows the Internet model for creativity. It is open and lightweight—a model that is created for adding features and applications in an open architecture."

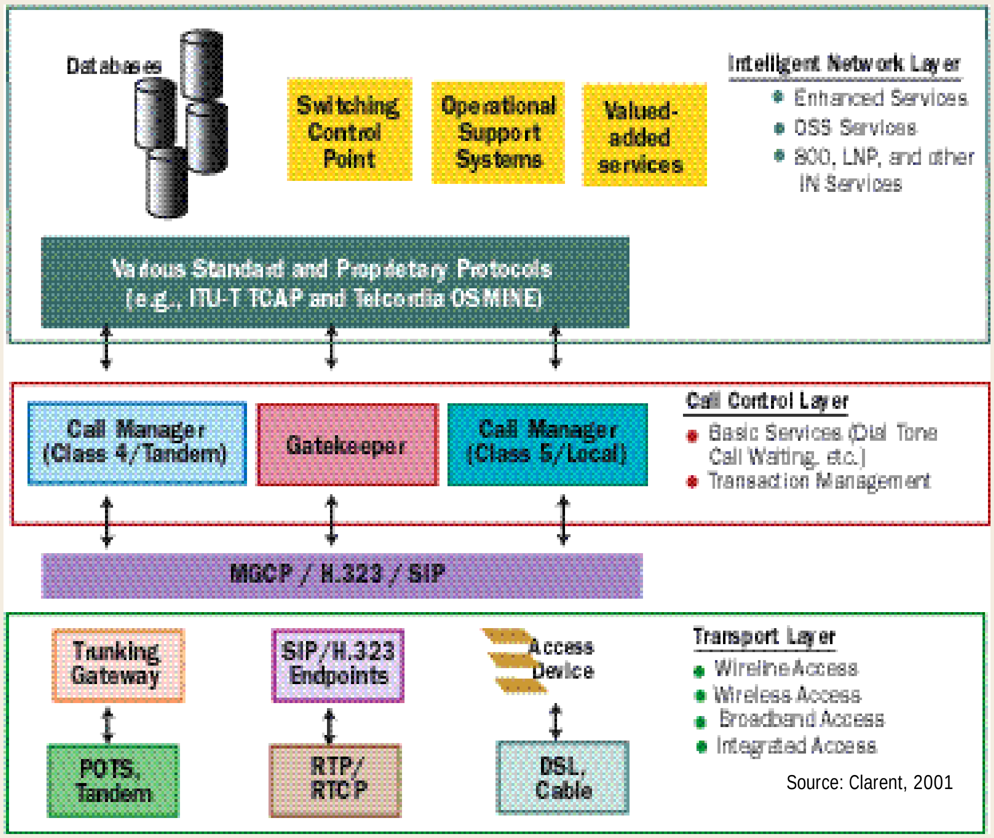
Like H.323, the SIP architecture also contains four main elements, but they don't exactly map one-for-one to the H.323 architecture. They include:

- User Agent, an endpoint that communicates either as a peer or in a client/server relationship with other agents.
- Proxy Server, which makes requests for User Agents.
- Redirect Server, which closely corresponds to the role played by an H.323 Gatekeeper.
- Registrar, which works with the Redirect Server to redirect calls to the current location(s) of users.

About the same time SIP was being developed, Media Gateway Control Protocol (MGCP) was also being developed for use with both SIP and H.323. MGCP separates call control (signaling) and media control, and initially was designed to function internally between the media gateway controllers (MGCs) and the media gateways (MGs) in "decomposed" VOIP network architectures. In the MGCP model, the MGC handles call processing between the PSTN and an IP signaling device—like an H.323 gatekeeper or a SIP server.

The standardized derivative of MGCP is the joint IETF/ITU's Megaco/H.248. This protocol set offers enhancements to MGCP, such that the media gateway controller can also control multimedia and multipoint conferencing enhanced services. Megaco/H.248 also offers improved syntax for more efficient message processing□

FIGURE 1 Functional Architecture



In other words, incumbent PBX vendors, like incumbent service providers, have largely elected to support H.323 as an efficient enhancement to their embedded base, with some new feature integration. And, like the newer service providers, new entrants to the PBX market have more broadly accepted SIP as the protocol for an expanded services focus.

Why Interoperability Is Key

H.323 gateways have made VOIP a useful, "save-money" application for international long distance providers and private network operators, while SIP-based PBXs are beginning to appear in customer premises. Yet a mass market for VOIP won't take off until vendors and service providers can get their implementations to interoperate—with each other and with the PSTN. Making H.323, SIP and the PSTN architectures work together is also the key to pushing VOIP beyond current service suites and into new services, and while progress has been made toward interoperability, much hard work remains.

The PSTN, H.323 and SIP have slightly different architectural approaches, but they all perform the same logical functions. Figure 1 shows the three basic, functional layers they share: transport, call control and intelligent services. Getting different vendor implementations of H.323 and SIP to interwork across these layers is a big challenge—interworking different vendors' equipment across architectural boundaries among the PSTN, H.323 and SIP is even more complex.

Because H.323 gateways have been deployed as "VOIP network islands"—i.e., matched pairs at either end of point-to-point links—they have operated basically at the transport layer. They don't need to interoperate with other VOIP islands, and they can rely on the PSTN or a PBX to provide call control and value-added features (e.g., stutter dial tone and call-waiting "beeps").

But without call control and service layer interoperability, these islands can't be connected, even though this represents an essential requirement. Jean-Francois Mule, Clarent's director for standards and architecture, noted, "Not only do large corporations need to connect to multiple carriers, but when corporations integrate multiple IT networks, they need to keep their existing [pre-integration] infrastructure. So the enterprise needs proven interoperability."

Mule views the progression of interworking in three phases. "First, you have publication of the standards. SIP and H.323 are both published standards. Next, you have profiling. The work for profiling these standards is currently in progress in the IETF. Then you have interoperability testing."

According to Mule, testing is under way between vendors on each protocol separately, and between vendors' H.323 and SIP implementations in several test labs, including some sponsored by interexchange carriers and by CableLabs. In addition, the International Softswitch Consortium, an industry group with more than 193 member companies, began interoperability testing in June.

Modarres of Commworks stated: "We've been in SIP and H.323 interoperability tests with companies like Cisco and Clarent. The basic features all work fine—features like call connectivity, Automatic Number Identification (ANI) and basic call treatment. Some vertical features like leaving voice mail also work. What gets more difficult are features like communications between messaging applications servers."

"More difficult" is a nice way to put it. The road to H.323, SIP and PSTN interoperability needs to be followed for each of the three layers. Although transport interoperability is largely complete and call control is making good progress, interoperability at the intelligent services layer has a long way to go. For example, the several varieties of instant messaging, all of which are Internet-based, work fine with SIP, but none work yet with H.323.

Can We Learn From The Past?

Still, it's important to keep developments in perspective. As Mike Gaines, marketing director at IP service control point (SCP) maker SS8 sees it, "The process is unavoidable and not entirely negative. Even SS7, a widely-used standard, has different implementations and variations. For example, there are no 800 [toll free] services available between the United States and Asia, yet I can roam with my cellular phone in both."

Gaines makes a good point. The cellular roaming he describes is only possible using multiple subscriber identification module (SIM) cards, or with "tri-mode" or "tri-band" cell phones. These phones can accommodate the multiple frequency ranges, analog and digital transmission and different signaling capabilities used in different cellular phone systems. And it is possible to call an American 800 number from Asia—but only if the caller first accesses a U.S.-based SS7 network (e.g., AT&T Direct). In other words, basic transmission and call control incompatibilities in the cellular and international SS7 networks are being solved today only with more intelligent phones or by user intervention.

So, one way to solve interoperability among the multiple VOIP protocols would be to manufacture phones and other "VOIP terminal adapters" that could either "roam" or provide a gateway between incompatible VOIP network protocols. But let's hope this is not the industry's "final answer." If we want to move VOIP forward, to give it the same ease of use as the PSTN, the service provider networks must provide VOIP protocol interworking just as they provide today's interworking between North American T1 and European E1 networks and between North American SS7 and European CCSS7.

There's more to the SS7 story that might interest service providers today. Recall that SS7 was initially deployed as an efficiency play. It saved interoffice trunking—the expense of establishing a circuit between two phones if the called party or intermediary trunks were busy—in much the same way that H.323 VOIP gateways have saved interoffice facilities costs in the last few years. But while the gateways created efficiency by packetizing and compressing voice calls at the transport layer, SS7 created efficiency with a new call control layer to better manage the transport.

An added benefit of SS7's call-control capabilities was the introduction of some $20 billion in new service and feature revenues, from offerings like Caller ID, 800 number portability and pre-paid calling. But these revenues couldn't be realized until the carriers interconnected their SS7 "island" networks.

Conclusion

By arguing that one protocol is better than another, service providers and IP telephony vendors introduce fear, uncertainty and doubt (FUD) about IP telephony. This could slow user adoption, denying everyone the efficiency and benefits that interoperable VOIP services could offer.

Customers don't want to have to make a choice between technologies, nor do they want the complexity of multiple systems. "End-users want to be able to use any device, from anywhere, at any time," asserted Radvision's Michelle Blank. "They don't care about the underlying protocol as long as the technology works." The good news is that multiprotocol and interworking gateways are emerging from companies like SS8 and CommWorks.

The PSTN, H.323 and SIP can work together as complementary and value-adding network technologies, and can serve the end users who will benefit from their integration. H.323 brings efficiencies to the PSTN, and SIP brings opportunities for new services. To best serve the enterprise, emerging VOIP technologies must combine transmission-efficient services to reduce the cost of telephony and create better communications services, regardless of protocol□

Companies Mentioned In This Article

AT&T (www.att.com)
Avaya (www.avaya.com)
Cisco (www.cisco.com)
Clarent (www.clarent.com)
CommWorks (www.commworks.com)
GoBeam (www.gobeam.com)
International Softswitch Consortium (www.softswitch.org)
Level 3 Communications (www.level3.com)
Nortel Networks (www.nortelnetworks.com)
Radvision (www.radvision.com)
Shoreline Communications (www.goshoreline.com)
SS8 Networks (www.ss8networks.com)
WorldCom (www.wcom.com)

Claiming either SIP or H.323 is better raises doubts about IP telephony