What is SIP trunking, and how should an enterprise deploy it? **JON ARNOLD,** Principal, J. Arnold & Associates, delves into SIP trunking specifics with an expert panel: **STEVEN JOHNSON,** President of Ingate Systems, **ALAN PERCY,** Director of Market Development of AudioCodes, **MICHAEL TIMAR,** Director of Product and Solution Development for Panasonic, and **ALAN KLEIN,** Consulting Systems Engineer with Avaya

MAKE THE CALL

JON ARNOLD **What is Session Initiation Protocol (SIP) trunking?**

STEVEN JOHNSON SIP trunking is a method by which companies are able to use their broadband connection to communicate generally via voice. SIP trunking replaces other methods of delivery of telephony services, which have mainly been delivered over a private, separate copper line. With SIP trunking, you get the benefit of using what might be underutilized broadband capacity to connect to an Internet telephony service provider that is placing your phone calls onto the public switch telephone network (PSTN). So it gives you the opportunity to better utilize your bandwidth, and it gives you the benefit of not having to manage your own PSTN gateway to a service provider. Also it can reduce long-distance costs, and in many cases, it gives you the benefit of being able to centralize your SIP trunking to a single location, and yet service all of your branch offices around the country or world.

The net result is that there is an ROI benefit, which has been measured by several of our customers to pay back the investments in new equipment within a year or less, which provides a tremendous incentive for people to move to SIP trunking. SIP trunking is a very important step forward; it gives companies a great ROI and quality calls, and also positions them for new ways of communicating using, for example, video, instant messaging and new applications using SIP that we are just beginning to see emerge on the market.

JA **What does SIP trunking mean, and what does it bring to the enterprise?**

ALAN KLEIN Previously, there were a lot of lab trials with SIP trunking, and now many of our enterprise customers are deploying. A lot of the necessary equipment is out there for customers to begin experimenting with SIP trunking. There are a lot of SIP based IP telephony services available, and at Avaya, we have a formal interoperability program, Avaya DevConnect, where we do interoperability testing with each provider, because they may have different underlying infrastructures. We're interfacing with those providers, and by bringing them into Avaya DevConnect, we're able to develop application notes that our customers can use. These show you how to configure the Avaya infrastructure equipment to interoperate with each service provider, and detail which screens they have to go through and any particular issues that were encountered during interoperability testing. That's one of the things we recommend, but it is not a requirement. We also have a generic application node that customers can use to do integration testing if the service provider hasn't been tested.

(SIP trunking) is removing the need to have a lot of extra infrastructure on the customer site, and places that over to the service provider. So a lot of the calls, instead of being handled from the IP layer and then converted to the PSTN, are actually being sent to the service provider, and it's up to them to figure out how to route the call and also how to change the call from IP to TDM. One of the major benefits is that the call between the enterprise and the service is all IP, leveraging the SIP protocol.

Some of the newer SIP trunking integrations that we're seeing and testing from the larger providers are actually expanding upon the basic SIP trunking functionality with a lot more interactions between the SIP protocol, and how messages are redirected between the enterprise customer and service provider and what information is being conveyed between the two parties. It's getting a lot more advanced in some of the call flows that are happening out there.

JON ARNOLD
PRINCIPAL
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Mr. Jon Arnold is Principal of J Arnold & Associates, an independent telecom analysts and marketing consultancy with a focus on IP communications. Previously, he was the VoIP Program Leader at Frost & Sullivan, responsible for managing their subscription service for Global VoIP Equipment Markets. He is a regular speaker at numerous VoIP and telecom conferences.

STEVEN JOHNSON
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Mr. Johnson has 15 years of senior management experience with Sanders Associates, Telco Systems and Lockheed Martin. He served as CEO of Abrena, Inc, a business development consulting firm which assists European and North American firms' expansion in the US. Formerly, he was President and CEO of Transcept Inc, a telecommunications infrastructure company. Mr. Johnson is on the Board of Directors for the SIP Forum.

ALAN PERCY
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Mr. Percy is Director of Market Development at AudioCodes, a leading provider of voice-over IP telephony products and enabling technology. In this role, Mr. Percy is responsible for identifying market trends and building relationships to foster new business opportunities. Mr. Percy joined AudioCodes in 2001 and brings over two decades of experience in the telecommunications, networking and wireless equipment industries. Mr. Percy is a frequent industry speaker and contributes to a number of industry journals and blogs.

MICHAEL TIMAR
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Panasonic Corporation is one of the largest electronic product manufacturers in the world, comprised of over 556 companies. It manufactures and markets a wide range of products under the Panasonic brand to enhance and enrich lifestyles all around the globe.

ALAN KLEIN
CONSULTING SYSTEMS ENGINEER
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Mr. Klein has 12 years experience with Avaya. At Avaya, he is an expert on pre-sales subject matter on converged security session initiation protocol (SIP) and wireless LAN technologies. He presents sales presentations to Fortune 500 companies and government entities. He focuses on informing and educating customers and partners about Avaya's products and associated new technologies, while assisting sales teams with product evaluations and presentations.

JA What does SIP trunking look like, and where does it fit into the network environment?

ALAN PERCY At a very high level, SIP trunking moves the voice traffic that previously was using dedicated time division multiplexing (TDM) circuits to IP infrastructure. Your voice traffic is being transported over IP infrastructure into your Internet telephony service provider (ITSP). At that point, the service provider will send it through a large media gateway, which is inside their infrastructure, to be delivered into the PSTN. With SIP trunking, the role of the media gateway moves from each enterprise having their own media gateway on premise, to the service provider having a common media gateway that all their subscribers are sharing.

Another challenge is getting the applications, whether it is a TDM-PBX, IP-BPX, Unified Communication system or other SIP-based application, to be compatible with SIP trunks. Lately, SIP Interoperability is an area where there has been a lot of work. Other areas that we're still addressing are some of the security issues with having an IP connection come from a SIP trunking service provider directly into an IP-PBX.

JA Why is SIP trunking gaining traction now; what has the technology done today to make it attractive?

MICHAEL TIMAR There are quite a few things happening here. The adoption in SIP trunking is definitely accelerating, and talking with the service providers, it's also starting to lead into more enhanced SIP trunking features. When I talk to end-users, the enterprise customers, a lot of what I'm hearing is about the cost savings leveraging the existing infrastructure, which is what you'd typically expect. What I also hear is, when you're talking about a long-term voice strategy for the company, SIP trunks actually become a starting point for a longer-term strategy of how to manage communications costs and really broaden what's covered within communications. As we communicate in more and different ways, the SIP trunk is just the entry point to much more advanced communications-management methods. This is all controlled by the IT manager, and they're looking for single points of contact for single points of redundancy within their network, and the SIP trunks definitely lend themselves to a much more robust infrastructure.

JA Why should enterprises think about SIP trunking today as opposed to a year ago?

SJ There are a number of factors coming together. We hear from customers that they're looking at this is as a cost saver, and even inside some of the most troubled industries today, companies are looking at SIP trunking to save money. However, it is not the only reason. We are now seeing many more service providers offering SIP trunking services, so previously it was ITSP start-ups that were typically out evangelizing SIP trunking and initiating the market. Today, we are seeing in the RBOC space AT&T, Qwest and Verizon, all having a SIP trunking service available. Many of the cable TV companies are either bringing SIP trunking to market today or planning to in the near future; so that's another element. There's much more capability in the service provider space and therefore many more opportunities to obtain service from both the incumbent and startup service providers.

The third factor is SIPConnect, and the SIP Forum is doing a great deal of work to define the best practices for connecting SIP trunks between service providers and customers. That too is a very valuable move forward: to have a simple, single way that is adopted by the industry to enable SIP trunking between your premise and your service provider. Now that Ingate, among others, has created the equipment that can sit between an IP-PBX and a service provider and ensure interoperability, »

there are less headaches for the company, for PBX vendors, and for the service providers to do all of the work necessary to maintain interoperability. So interoperability is now better, we have demonstrated cost benefits, and we have a greater field of service providers who are in this space. All three of those elements are contributing to the now accelerating adoption of SIP trunking by enterprises.

JA What do you think is the stronger force here: the enterprises looking to add SIP trunking or the service providers trying to keep their customers and find new ways to add value?

AP When SIP trunking first took off, most of the industry thought it was a way for IP-PBX customers to reduce some capital equipment costs of their new system. However, it has turned out to be a very valuable tool for businesses that are trying to save operational costs, in a few different areas: First, SIP trunking reduces the cost of their long-distance or local access calls. Second, it has enabled distributed enterprises to eliminate many of the separate analog or TDM trunks that they had servicing branch offices and consolidate that voice traffic to a centralized site, eliminating the costs of the separate lines. However, to achieve these savings, a lot of enterprises faced an expensive upgrade to their PBX infrastructure, which in many cases meant entirely replacing their infrastructure with a new IP-based solution that would be disruptive to the business to install and re-train the users. Frankly, we've found that the bigger opportunity is the much larger installed base of legacy TDM-PBX or Key Systems that are already servicing customers, leaving them in place and using an AudioCodes media gateway to connect these legacy systems to a SIP Trunk, thus reduce operating costs with a very quick return on investment.

JA What should enterprise decision-makers know and understand about SIP trunking?

MT There are two ways we could look at that. From the Panasonic perspective, with a recent addition we've made, virtually every PBX we've sold in the last five years can be upgraded with a very modest cost to support SIP trunks directly. That has been a very popular option with our existing customers. However, the other side of it is you can also just use a PRI gateway and actually run the SIP trunks into the existing PBX using PRI channels, with virtually no change in the existing infrastructure. You lose some of the features and functionality by doing that, but it definitely is a very modest cost. From an existing infrastructure standpoint, the biggest challenge that we see from the IT managers' perspective is how they handle the SIP trunk input through their firewall into their PBX, into their network. Do they want to run it through their existing network using a session border controller, or do they want to actually bring in a separate data channel to support those SIP trunks?

AK One of the things Avaya recently announced is a product called Avaya Aura Session Manager: It is a common core component for a lot of different SIP based systems to be tied into. Some of those newer systems may be tied in directly via SIP trunks, if they're SIP capable; some may be tied in by AudioCodes gateways to handle the TDM to SIP conversion. By doing that, they all traverse this common centralized dial plan, and are able to utilize this central point to connect out to various different public SIP trunking services. It basically ties all of those disparate PBXs that an enterprise customer may have and creates a common logical routing platform. A lot of the direction, particularly with Avaya, is certainly around SIP — everything being SIP to the core. This is just one example of leveraging technology like public SIP trunking and allowing the whole enterprise, even though there might not be a common system, to be able to act like a common system via the centralized Session Manager.

JA The media gateway has an important and evolving role here. What does SIP trunking mean for the gateway?

AP We see this as a first step in an evolutionary transition to a pure SIP environment. We often see accounts that have installed TDM infrastructure and are trying to integrate those systems with new SIP-based applications and services like SIP trunking. The objective of our SIP strategy allows businesses to upgrade piece by piece, so they don't have to completely replace all of their equipment at once. As SIP applications are becoming more popular, we're seeing greater integration challenges, with businesses trying to replace portions of their infrastructure but retain other parts that are working perfectly fine. Businesses want to be able to use SIP trunks because of the costs savings, but without disrupting their business. So we see businesses using our media gateways with TDM on one side going towards the legacy systems and SIP on the other side going to the SIP trunking service providers. The next step is, as they replace each system with SIP-based solutions, the media gateway actually takes a different role: providing interoperability and security. In this scenario, the gateway becomes a SIP-to-SIP gateway, meaning it plays an intermediary between different SIP systems or services, solving many of the interoperability and security issues with multi-vendor SIP deployments.

The result is, an enterprise will be able to move system by system, with the media gateway playing the role of intermediary doing conversion from one format to another. In some cases, the gateway may eventually end up with no TDM connections to it at all. In these situations, it's purely "SIP in and SIP out."

JA What does it mean for a firewall to be "SIP aware"?

SJ Firewalls have been one of the things that have stood in the way of adoption by enterprises, because an enterprise has protected its network, rightly so, from the Internet by placing a firewall at its edge. SIP is an application layer protocol, and firewalls, especially network address translation (NAT), existing at the transport layer means that NAT breaks SIP; and there is no way for that firewall, unless it is fully SIP aware, to be able to re-route calls from the outside off to a SIP-PBX or SIP phone on the inside, because the private IP addressing space is hidden from the outside world, by design.

Being fully SIP aware means, first and foremost, to be able to get past that NAT space and be able to rewrite the header information so that the calls can be directed toward the inside of the network effectively. It is having a device, like what Ingate produces, that is both capable of resolving this NAT traversal issue, but is also capable of making sure that what is allowed into the network is in fact valid based on rules established by the company that owns the equipment, and secondly based on the proper formation of the SIP signaling information being delivered. That's what Ingate does with a very strict parser and with a very firewall-like set of rules around SIP traffic to give the company maximum control.

The security issue here is that a SIP-PBX is not unlike any other server on your network and must be protected like other servers — such as Web servers, financial servers and data servers — from the outside world with a firewall. The existing firewall is probably not optimized for SIP, and it is important to have an SIP-optimized firewall in place so that the ability to make phone calls, which is as much the life blood of that company as any of the other servers are, cannot be compromised. Security, NAT traversal and all of the activities around controlling who is permitted into the network are essential for a proper implementation of SIP or SIP trunking into an enterprise environment.

JA What do customers need to know in terms of setting expectations about what SIP trunking can or cannot do?

AK Something that seems to come up frequently is how fax is handled by the different service providers, which can be a big issue depending on the customer. There is a standards-based way of handling faxes across IP networks called T.38, although »

each service provider may handle fax differently.

Other things that people may not be fully aware of, because they may not be familiar with SIP, is that SIP is just a signaling protocol. It has nothing to do with the media; it's just used to set up, modify and tear down the communications session, and the media could be different things, like voice and video, taking independent paths through the network. If it's real-time media, they're going to require that the customer's data network is capable of supporting voice. So depending on the customer's current WAN infrastructure, it may require some modification. Also, working with their service provider to make sure the proper QOS classes are in place, so that the voice and video receives priority over the other types of communications that are going back and forth across the WAN.

Finally is the need to become knowledgeable of SIP itself. It's interesting that the people that are looking at SIP trunks are more on the business side and they don't initially work that closely with other people, primarily security. At some point, the security people will get involved, so it's good to bring them in as early as possible. A lot of those challenges can be talked about at the beginning, so as to not hold up the project when it comes time to implement the solution — you need to make both groups happy about what is going to be put in place.

MT The security issue is probably the biggest issue the IT manager faces when he's making the decision.

AP I just wanted to add that some of the SIP trunking service providers do support T-38. Others don't; instead, they try to do fax over G-711, while others who don't support fax at all. That would be, unfortunately, quite a surprise for someone who has gone through a migration to find out that their faxes don't work. Another area that we've been working with some partners on is the choices of voice coders and necessary broadband connection hardening. We've seen some installations go in, and if they haven't done sufficient planning on their broadband connection or they've made some assumptions about the broadband and the traffic of their voice being added, they may have some "fits and starts" as they get into it. It's important to do proper planning from a broadband connectivity standpoint, ensuring that the right voice coders are being used so that the business best utilizes the bandwidth on their broadband connection.

SJ It is key to plan for the implementation to ensure your network is prepared for it, including quality of service and codec selection. Selecting the appropriate and best service provider for you and your area and your calling patterns is crucial. The standard T-38 for faxing is a critical one and is something that needs to be well understood before you adopt SIP. Then it is important to make sure you have optimized your installation and taken care of the NAT traversal and security issues. However, with all of that, SIP trunking is ready for primetime and is something that companies should be looking at very seriously, as they evaluate their cost structures and future directions of telephony and communications in general.

JA Where is SIP trunking evolving, and how are the interoperability programs making this an easier task than before?

AK Here is an example: Some of the solutions are leveraging portions of SIP that have since been updated to standards. There is a group that's been creating all of the RFCs for SIP called the Internet Engineering Task Force, and as part of the development or evolution of SIP there's been quite a number of drafts, and sometimes these drafts become RFCs. A lot of times, given that initially there was rapid development around SIP standards from trying to recreate some of the things that were happening in traditional telephony, some of those early drafts that may have been incorporated into the infrastructure have since been replaced by RFC standards. It takes a while for some of those expired drafts to get "out" of the equipment. There may be some infrastructure that is using things that aren't part of the SIP standard now, but they're in place, so we have to interoperate with them. We are strongly behind the standardization of SIP, but at times in order to do what's best for the customer, you have to support things that never made it to a standard. For example, the SIP diversion header, which is pretty heavily used even though it has since been replaced by history info. It forces us to support both, even though, technically, you would like to just support the standard.

One of the things that SIPconnect specifications does is try to look at all of those different RFCs and figure out which pieces would be the most useful, and have the vendors and service providers supporting the same common parts. So the interoperability aspect is a lot more streamlined. For example, equipment supporting the SIP diversion header can cause other features to fail; if a service provider is looking for something with a diversion header and a vendor doesn't support that header, certain call flows may break, while other things may work perfectly fine. The whole idea is trying to get that subset agreed upon by everyone, and if everyone is supporting that, there is a very high likelihood that there is going to be interoperability and a low likelihood that some of these fringe features are going to have problems.

JA What can enterprise expect from SIP trunking over the next year or so?

AP The security and interoperability effort around SIP trunking will remain a significant challenge. I think one of the common mistakes that people make with SIP trunking is thinking: "I buy a brand new IP-PBX, I take my broadband connection, plug it in, and 'poof' I'm set." The reality is that there will be a device or point of demarcation that is installed at the customer premise — doing security and interoperability.

SIP is still an evolving specification. However, the service providers aren't necessarily going to be able to go back in and upgrade all of their equipment just because one of the IETF drafts changed. In fact, many have told us they will not be doing that; they are going to need to have some stability in their network. Service providers may end up providing interoperability equipment; it may be included with their service, or it may be something the enterprise purchases. Right now, we're seeing the smaller, more agile start-up service providers who adopted SIP trunking aggressively. Now you're starting to see larger, more traditional service providers enter the market with SIP-based services. There is definitely a trend of growth, interoperability solutions and security solutions.

JA How would you describe the ideal SIP trunking deployment scenario?

SJ The ideal implementation solution, from our perspective, is not complicated. It is one where the company has either upgraded to an IP-PBX or has used a media gateway, like those offered by Dialogic and AudioCodes to name two, to SIP enable a legacy PBX, which is supporting SIP on the trunk side at least and is situated on their LAN behind their firewall for security purposes. That is then fronted by an enterprise session border control, and as Alan Percy pointed out, that might be delivered by the service provider, or it might be delivered by the PBX vendor, or it might be acquired directly by the company. It is a very key element in the installation of SIP, in that it is the device that is the demarcation point between the networks; it does perform the function of interoperability normalization and it does provide the security fabric for SIP.

Ideally, before the company has embarked on this, they will have done some work to test their network to make sure there are no bottlenecks and make the appropriate upgrades if there are. Once that is accomplished, the most important thing is to find the appropriate service provider that provides the kind of number portability that you're looking for and can centralize SIP trunks into a single location. The provider should be willing to work with you on finding the most effective way for you to adopt SIP trunking — and there are many of them around. The ideal situation is well-planned, well-executed and well-secured within an enterprise session border controller. **BTQ**