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# SIP: myth versus reality

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Business  
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It's a new world. Modern packet-based communications are sweeping away traditional telephony standards and changing the way businesses stay connected. SIP, the latest IP enabler to hit the market, is often portrayed as the key to implementing more sophisticated, flexible voice and unified communications. But is that a fact, or merely well-publicized fiction? This paper explains SIP, as well as its background, capabilities and status and then looks beyond the hype and analyzes the relevancy of SIP to today's IP telephony deployments.

## what is SIP?

SIP, also known as session initiation protocol, is a signaling protocol for managing real-time communications sessions with two or more participants. Because it is transport and device independent, it can work across voice, video, messaging and collaboration sessions linking phones, PDAs, mobile devices, instant messaging clients, etc.

As a "lightweight" protocol, SIP requires only three messages to set up a session, making it much less complex than H.323, its tried-and-true predecessor. SIP's text-based foundation (modeled after HTTP) gives it low overhead for a communications protocol. Importantly, SIP does not define telephony features; rather, it focuses on call set-up and signaling. The telephony features must be built into network elements and end points.

Notably, SIP was built from the ground up by the IP community, i.e., the Internet Engineering Task Force (IETF), whereas the International Telecommunication Union (ITU) developed H.323 as an outgrowth of ISDN Q.931. SIP's IP roots make it a favorite of the Internet community, and the protocol works well with other web technologies, such as SIMPLE (SIP for Instant Messaging and Presence Leveraging Extensions), SOAP (Simple Object Access Protocol) and XML (eXtensible Markup Language). This flexibility is useful for integrating SIP with enterprise applications and for services like text-to-speech, enabling the kind of customized integration of telephony and applications not possible in the public switched telephone network (PSTN) world.

# SIP's many applications

SIP is designed for use throughout the network, whether it's an enterprise or service provider network, fixed or mobile, time division multiplexing (TDM) or IP. It works on a softphone, desk phone, mobile phone, Wi-Fi device or even a videoconferencing unit. Each user is assigned an address or "number," such as "sip:firstname.lastname@company.com," called a Universal Resource Identifier (URI), which leads to all of the user's voice and data devices and accounts.

SIP uses information about user preferences, availability and location to route communications sessions. For example, users change their preferences based on where they are and how to best reach them so colleagues can contact them on their first try, rather than leaving multiple voicemails and emails. Another example is enabling users to allow specific callers to reach them but appear unavailable to everyone else. Still other features include the ability to change the parameters of a voice call to those of a video session after a call has begun and moving the cursor over the author's name when reading a document to determine if that person is available to answer a question in real time.

# vendor implementation and interoperability

Given the above, you may be surprised to learn that according to the IETF Standards Process, SIP has not been ratified as a “Standard;” rather, it is still in the “Proposed Standard” phase. This is not to say that SIP is not developed enough to be useful while it continues to mature. The initial SIP draft was published in 1999, and it reached Proposed Standard in 2005, offering 16 basic features (RFC 3216). The SIPPING committee worked to extend the basic feature set with additional Requests for Comment (RFCs) to define another 30 to 40 features. All told, this now represents 95% of the features employed by the majority of users today.

SIP has gained support from industry heavyweights and all leading vendors, including Alcatel-Lucent, Avaya, Cisco, Genesys, Microsoft, Mitel, NEC, Nortel and Siemens, now support it. There are SIP-only platforms, such as Microsoft Live Communications Server and Office Communications Server, Nortel Multimedia Communications Server, BroadSoft, etc. These platforms only offer native SIP. Other vendors, such as Alcatel-Lucent and Cisco, use a “dual stack” approach that runs SIP alongside their own proprietary protocols (e.g., Cisco SCCP or “Skinny”). Hybrids represent yet a third type of implementation, in which vendors use SIP subsystems, such as a SIP proxy server, to front-end their telephony software and call servers; Avaya and Nortel fall into this category.

However, here is where the SIP vendor implementation needs to be clarified and the myth separated from the reality. For example, SIP champions herald the protocol as “standards based” and “open.” They claim that because they support SIP, their IP PBXs are also standards based and open. Some people extol SIP as a means for achieving interoperability between vendors’ or equipment makers’ products. In fact, SIPit (SIP interoperability test) events are gatherings of SIP implementers to test and showcase the interoperability of their products. For example, will vendor X’s IP PBX inter-work with vendor Y’s IP PBX, or can vendor Y’s phones be used on vendor X’s IP PBX?

These claims of “openness” and interoperability are not exactly true, though. If the past is any predictor of the future, vendors have a vested interest in selling their complete systems and not “opening” them, except to tangent third-party products to fill gaps in their own portfolios. A good example is Q.SIG, the PBX industry’s answer to interoperability in the 1990s. Q.SIG provided the ability to pass PBX-specific features, like call transfer, abbreviated dialing, call forwarding

and ring back the phone when free, across the network. Theoretically, it should have worked across different vendors' PBXs, if they had implemented Q.SIG in a standard fashion; but most vendors did not, and Q.SIG features could only be assured if you used the protocol between the same vendor's products. The same rule now applies to SIP.

As mentioned earlier, SIP RFC 3216 offers 16 basic features that, if implemented according to the standard, should work between different vendor IP PBXs. However, once you go beyond basic SIP, interoperability is not guaranteed. Vendors can either interpret the SIP RFCs in their own ways or apply vendor-specific extensions to the SIP features. This allows them to differentiate the SIP features on their systems from the SIP features of their competition through a proprietary twist. The need to differentiate their products often supersedes their motivation to assure interoperability, leaving many desirable features to work only in a single vendor's implementation.

The need for differentiation among vendors renders the promise of feature transparency between multi-vendor systems largely unfulfilled. As a case-in-point, the Cisco Unified Communications Manager implementation of SIP boasts 176 features. Twenty-five of these features are based on the SIP standard; the rest represent a SIP extension to a Cisco proprietary feature. This is not to say that proprietary is "bad," especially when the vendor tackles integration and ensures that the feature works well within its own system. The proprietary extension may even be an improvement on the proposed standard or a better way to deliver the feature.

Another myth held by customers and somewhat perpetuated by the market involves third-party phones. People believe that they can buy a leading vendor's IP PBX system but substitute inexpensive SIP-based phones from third parties or low-cost manufacturers. This is misleading because vendors charge a licensing fee for each phone or end point managed by the IP PBX, regardless of who manufactures the phone. In addition, as mentioned above, vendors' SIP implementations may require proprietary extensions to access the full features/functionality of their systems, which relegates third-party phones to basic features only. This makes phone substitution possible only with IP centrex services when the phone is not central to the IP centrex vendor's business model (e.g., BroadSoft and Comverse). As you examine a vendor's IP telephony proposal, you will discover that the phone makes up 30% to 50% of the total cost of the purchase price; hence, the motive to include as much of the vendor's own product (phones) in the sale as possible.

While dispelling myths concerning SIP, another fertile myth revolves around SIP trunking. Some marketing people and service providers would have you believe that SIP enables IP trunking, a new cost-saving innovation. Service providers claim that their SIP support allows an enterprise to connect to their service in native IP using a SIP trunk. This allows the enterprise to remove its

circuit-switched voice trunks (ISDN basic or primary rate interfaces – BRI or PRI – or dedicated access lines) and eliminate a PSTN gateway, thereby lowering cost and improving call quality.

The benefits of this trunking solution are accurate; however, the myth is that either you need SIP to achieve IP trunking, or SIP uniquely enables IP trunking. The reality is that Orange Business Services has offered IP trunking based on H.323 – or MGCP (media gateway control protocol) – since 2002.

When Orange first introduced IP trunking, it involved native IP connectivity directly to IP end points. Circuit-switched voice trunks were removed as we converged voice and data over a single broadband connection. Today with Business Talk local service, which includes routing of direct inward dialing (DID) calls, customers no longer need gateways to connect to the PSTN to make or receive phone calls. We enable DID with local calling to our customers' locations and can completely bypass PSTN access. So, the advent of SIP did not make this value proposition possible; it just offered another flavor (protocol) with which to enable IP trunking. IP trunking actually has less to do with the protocol used than the ability to make or receive calls using only IP access.

# the Orange Business Services offerings

You can take advantage of IP trunking today along with all of the cost-saving benefits that it affords. To get started, you don't need to wait to figure out when, where and how you will implement SIP. Orange can help with those decisions and future-proof your way forward.

Our Business Talk service offers:

- IP trunking – signaling and call flow in native IP format from IP end points through an IP PBX or an IP-enabled PBX to our IP VPN network
- interoperability – full inter-working between equipment vendors' IP telephony systems and our intelligent network consisting of session border controllers for multi-layer security and NAT proxy, carrier-class gatekeeper and global gateways for secure, low-cost on and off-net calling
- VoIP VPN with network-based features, such as private dial plan, virtual on-net, forced on-net, black list to block calls to certain locations or numbers, white list and call forwarding (unconditional or based on ring no answer or busy signal)
- advanced routing based on time-of-day, day-of-week and holidays with the ability to reroute calls based on network conditions
- highly secure IP voice network service
- support for the leading IP telephony equipment from Alcatel-Lucent, Avaya, Cisco and Nortel
- lower cost, reduced risk and smooth migration to a converged IP infrastructure for IP telephony

# the future

In addition to offering your business tried-and-true IP trunking services today, Orange Business Services fully supports SIP with a robust service development roadmap incorporating the latest releases in hardware, software, features and applications, including:

- Cisco Unified Communications Manager 7.0 and 6.1, which both offer full support of SIP-line side features and SIP running on Cisco IP phones.
- Cisco Unified Presence Server, which collects information about a user's availability and communications capabilities, including whether or not the person is using a communications device, such as a phone, at a particular time. This application can improve productivity by helping colleagues connect more efficiently by determining the most effective avenue for collaborative communication.
- Alcatel-Lucent OmniPXC Enterprise 9.0, which is SIP enabled.
- OmniTouch Unified Communications, a suite of enterprise unified communications applications, that allows users to tailor, control and manage calls, messages, directories, collaborative work tools and information from any location using any device and any software interface.
- Avaya Communication Manager 5.0
- Nortel Communications Server 1000

All vendors' releases and unified communications applications have gone through rigorous testing with our Business Talk IP trunking service. Our optimization program reduces your risk when deploying the latest IP telephony system and paves the way to a successful implementation or upgrade.

# conclusion

Going forward, there are multiple ways to implement SIP. SIP holds the promise of unifying communications, including real-time and non-real-time, along with support for presence to quickly find a colleague and determine his/her availability to communicate, even before starting a session. Ultimately, it will improve the way organizations work by removing human latency and speeding business processes.

But why wait? While the market sorts out which applications will yield benefits and produce a positive business case, you can implement a robust IP voice service and IP trunking now. Our telephony solution consultants can help you sort through the SIP versus H.323 debate, considering your business objectives and helping you separate myth from reality. Our advanced IP voice network interoperates with the leading IP telephony systems, offering an optimized design that mitigates migration risk while lowering overall cost – and it is available today. Then, as SIP and unified communications applications develop, we'll be there to support its integration and your smooth migration.

Leverage the investments that we've made in our carrier-class gateways, gatekeepers and security to reduce your capital and operational costs and realize the business benefits of this emerging technology. Contact your account manager today to discuss SIP and H.323 for your IP telephony deployments. Together, we'll find a solution that's just right for your business.

for more information, visit  
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