



Mitel Discovery Series: SIP Trunking – more than just cost savings

Questions and Answers

July 2009



The purpose of this document is to address the questions submitted during the Mitel® Discovery Series webinar that was held on June 30th with guest speaker noted industry analyst, Jon Arnold. Jon is the Principal of J Arnold and Associates, an independent analyst and marketing consultancy with a focus on IP communications.

Q. This appears to be an argument for IP communication. No problem there, but why not just call it that?

A. SIP is a subset of IP communication, and is relevant as it offers the opportunity to standardize communication connections over this transport.

Q. Besides more bandwidth, how is SIP different than MPLS and IP networking? This seems to be just the Internet and what you can do using it. How is this different? How is this different from ordinary broadband?

A. SIP is a protocol that supports voice (and other media) transmission and interconnection over IP networking, the internet, MPLS or broadband. It is conceptually just like transmitting POP3 email over the IP network, except in this case it is for multi-media communications, like voice.

Q. Technology wise, how is SIP better than other protocols like H323 etc.?

A. This is very debatable, especially amongst the developer community (kind of like the Betamax vs VHS war), however certainly SIP has evolved to become the defacto standard of choice for IP communications. One important advantage of SIP is its ability to handle multi-media.

Q. How are SIP Trunks delivered from the LEC to the Customer Prem? Broadband?

A. SIP is delivered via broadband connection.

Q. How universally available is SIP Trunking, and is the SIP protocol implemented the same way by different carriers?

A. Today, in most regions you will be able to find a local / regional service providers that provide SIP Trunking. The availability of SIP will continue to grow over the coming years. Best efforts are made by the carriers and vendors to comply with the standard, however to ensure quality and integrity of service, interoperability testing is recommended, which the vendors conduct.

Q. Are there other countries using SIP or further ahead than the US in implementing SIP?

A. Yes, SIP is a global standard and growing in all regions of the world. US and Europe are quite advanced in Service Provider availability of SIP.

Q. Will SIP Trunking allow a client to broker services across multiple service providers swapping traffic for best value services?

A. Yes. You could independently subscribe to service from two different SIP carriers, and internally route calls on the Mitel 3300 ICP (for example) as appropriate to take advantage of best value services.

Q. Can you outpulse the local number in another market with SIP?

A. Calling Number is transmitted via SIP and will appear through another carrier's service, even on TDM circuit-switched facilities. Also, for example, DTMF over SIP is supported to allow outpulsing of digits while on a call to allow access to IVRs, voice mail systems and other automated service applications across SIP / TDM domains and between carriers.

Q. Can we safely replace PRIs with SIP Trunks and not lose functionality?

A. As a blanket statement, you will not lose functionality. However, pay specific attention to the list of services and features your Service Provider delivers with SIP, compared to your PRI provider.

Q. Can I get an internal SIP phone number from any country?

A. This does not necessarily have anything to do with SIP, and is dependant on the reach of the Service Provider, country specific regulations and whether this type of service is offered.

Q. Can you have both name and number with caller ID?

A. Yes, as long as the Service Provider supports this.

Q. Deeply concerned about proprietary SIP, interoperability issues.

A. SIP is an industry standard that has a defined structure and protocol. This is managed by the IETF, under a series of standards documents known as RFCs. The standard is continually evolving and being updated. All vendors attempt to comply with the standard in good faith – it is in their best interest to do so. However, our experience is that implementations often differ slightly, so interoperability testing is important, in fact mandatory. This is why Mitel has established the SIP Center of Excellence for interop testing. Rest assured, if vendors have cross certified interop, the interworking is very solid. SIP also allows for proprietary functional extensions (feature enhancements) to be added and accommodated, which many vendors do incorporate for features or unique innovations not covered by the standard – sometimes these are shared between vendors for specific interoperability. In general, we are seeing proprietary extensions to the standard is on the decline, and as the standard and vendor implementations continue to mature, interop will also become more solid and increasingly easier.

Q. Does Mitel have a list of SIP trunk service providers? Which SIP providers are "Approved" with Mitel?

A. Yes. Please see the Mitel SIP Center of Excellence Interop Guide.

Q. Is anyone doing SIP certification like the certification done by the WiMAX Forum?

A. There are SIP interoperability forums, such as SIPconnect and SIPIT, although we have seen a trend more toward direct vendor to vendor relationships for interop.

Q. Verizon is using standardization as a way to disadvantage products they don't sell. How might vendors successfully push this issue?

A. Service Providers have an obligation to ensure interoperability of their services and public facilities is as widespread as possible. Based on demand, vendors and Service Providers will prioritize interop testing in this regard, such as for SIP Trunking. As an alternative, if a particular product is not certified for connection to the SIP service, a certified gateway can be used to interconnect between these.

Q. Can you talk about the scalability of SIP?

A. SIP is scalable from a small business to large enterprise very easily. Instead of provisioning T1s (or some subset of partial T1s), SIP can be provisioned and purchased on a single channel basis. It is also provisioned (licensed) in the Mitel 3300 ICP in the same manner. Therefore you only need to provision as many Trunking channels as you need, and there is less constraint on configuration rules (like hardware capacities in the PBX for example). The Mitel 3300 ICP can natively support up to 2,000 SIP Trunking channels out of a single controller, so this is an excellent example of the scalability of SIP. There is a consideration for the bandwidth (and router) needed to support the amount of SIP based traffic.

Q. Can you give scenarios of UC with and without SIP Trunking?

A. You can use pretty much all the basic features of UC without SIP Trunking, but having it enables a better quality experience. This matters most for conferencing, where it's much easier to support wideband codecs. So, if you want to use HD voice or video, SIP Trunking is the way to go. The difference is quite noticeable, which can really help for sales presentations or customer support. SIP Trunking also makes UC more effective when being used across multiple sites. This ensures that all locations have end-to-end IP, which means everyone can have the same HD experience when using UC to collaborate.

Q. Besides business use, where do you see a single end users (home use) benefit from SIP Trunking today (other than say a Skype type app)? Why does SIP Trunking allow for fewer PRI's?

A. On a single site basis the number of channels required would be the same for both PRI or SIP Trunking – no real saving here, except for the monthly tariff. SIP does offer more efficient use of bandwidth than PRI, and the opportunity to aggregate this bandwidth with broadband / internet access for the business – an efficiency saving. The real saving comes when you can aggregate Trunking for multiple sites together at a single location. For example if ten sites each on their own require four channels of trunking service, aggregating this at one location may allow for only 30 channels to be subscribed to instead of 40. This is because local number presence for each location can be aggregated anywhere within the service area of your Service Provider, and you will utilize the power of averaging traffic patterns for each site in this centralized model (if four trunks on each site are required for peak traffic, but most of the time they are idle, it makes logical sense that you will not be required to provide 40 channels if these are all aggregated and terminated at a single location).

Q. How does SIP Trunking reduce the need for toll free numbers?

A. SIP Trunking provides end-to-end IP, which gives you more capabilities for integrating telephony with Web-based applications such as click-to-call. Now you can give customers a choice to call you on a toll free number or just click to call from their PC. The latter can be easier for some people, so this should cut down on the need to use the TFN. Furthermore, with SIP Trunking, that click-to-call session can quickly be escalated into a more collaborative experience where you have video, whiteboarding, ad hoc conferencing, etc. This make for a more engaging experience where you can get a better result for the customer and hopefully generate more revenues.

Q. If I have VoIP how does SIP Trunking further reduce LD cost?

A. In terms of your LD cost with an existing legacy carrier, there may not be much further savings, especially if your business is fairly small. However, if you have many branch offices and you connect them all over SIP Trunking, you will be able to divert more LD traffic off the PSTN, which would lower your LD cost a bit. There is a second way you may be able to gain some LD cost savings. One of the incentives a SIP Trunking provider will offer to move you away from your incumbent telco is lower international calling rates. Today, domestic LD is a minor expense if at all in North America, but toll charges still apply to international calls.

Q. If LD, legacy TDM time and materials, site to site dialing are not a factor, how do you get an ROI with SIP in an enterprise that has 40+ locations across the US?

A. If those elements are not a factor, the tariff for SIP on a per channel basis may be more favorable, and the opportunity to reduce the overall quantity of trunking channels to service 40 sites through aggregation will provide a more attractive ROI than TDM Trunking.

Q. Would you propose dedicating internet bandwidth to SIP Trunking vs. bundling all in one circuit?

A. Bundling is the most common approach, and managing the bandwidth with QoS. However customers that implement SIP (in some case large bulk quantities of channels) will separate the circuits and manage the SIP trunks via a Session Border Controller.

Q. With SIP what security vulnerabilities should we be prepared for? Explain security and SIP.

A. SIP over the public internet or broadband is not necessarily secure, and varies depending on the carrier or service provider. Encryption of media and signaling is still maturing in the standard and in deployment from vendors. Within a private environment / network SIP traffic can be secured.

Q. How is E-911 handled? What are the E-911 requirements for SIP? Please comment on how SIP complicates E-911.

A. Calls originating from a specific location need to be routed to the local emergency response center. If a caller in Atlanta routes through its IP-PBX via SIP Trunking to the public network in Orlando, location information is transmitted with the call identifying its source as the location in Atlanta and thereby routed by the Service Provider to the 911 center in Atlanta, and obviously not Orlando. CESID, which is the protocol to support transmission of a location identifier with the 911 call is supported over SIP Trunking.

Q. For E-911, a lot of businesses maintain a separate POTS line.

A. True.

Q. How would E-911 work in the hospitality industry, since I'm a vendor supplying guestroom phones to hotels, 911 calls are of importance, and there still seems to be an issue as to how 911 will be addressed. Would you suggest having analog lines that would capture the 911 calls from the guestrooms?

A. The issue of correctly transmitting a guest room identifier with a 911 call is not the problem of the trunking service employed (analog, T1, ISDN or SIP). It is an issue of ensuring room location information is transmitted with the 911 call at origination. That is per E-911 (CESID) standards and registered with the local PSAP database (Public Safety Answer Point).

- Q.** We know there are challenges with fax over SIP. Is there an expected timeframe for carriers to work out the issues with the delivery with the carrier or the receiving with Mitel? How are CLECs addressing fax over SIP? Why are there T.38 Fax and Video problems in getting many providers approved?
- A.** T.38 is the protocol to support fax over SIP Trunking, and therefore must also be supported by the service provider. Mitel supports T.38 for fax over IP and / or SIP. In my experience, most Service Providers are supporting T.38 for fax.
- Q.** What does the competition have that compares to this product?
- A.** Almost all IP PBX vendors have some support or implementation of SIP.
- Q.** Is there a prerequisite version of Mitel O/S for SIP?
- A.** Mitel answer: the minimum software version of the 3300 ICP supporting SIP is Rel 7.0 UR2, and for the Mitel 5000 is Rel 3.1. Due to the evolving protocol support for SIP within these products, it is encouraged to maintain the latest software revision.
- Q.** I think SIP Trunking would be good for smaller, or branch offices, and you may not be able to answer it but why is SIP Trunking not supported on the Mitel 200 ICP which is designed for Smaller / Medium Businesses?
- A.** Mitel answer: SIP is not yet supported on the 200 ICP, unless a TDM to SIP gateway is used. Any application requiring SIP is encouraged to deploy the Mitel 3300 ICP, including migration of existing 200 ICP sites to the 3300 ICP utilizing the available migration program.
- Q.** With SIP technology can we use any device to act as a phone rather than proprietary devices from Mitel?
- A.** Yes. As long as the device and Mitel call control (SIP station side support from the 3300 ICP) are interoperable. Mitel certifies many devices as interoperable, however customers can also implement non-certified devices, and many will work perfectly fine. However Mitel or the device vendor cannot guarantee functional performance.
- Q.** What's FMC?
- A.** Fixed Mobile Convergence – the ability to make / receive private network "PBX" office calls on a public mobile device, generally that is dual mode – e.g. Wi-Fi / GSM.
- Q.** What's Mashups?
- A.** Mashups is really a Web 2.0-style term that refers to "mashing up" two or more communications applications to create new services. The world of Web 2.0 is very different from telecom, and here, most applications are open, and built on standards-based protocols, which makes it easy to mix and match across various modalities. An example would be to speech-enable your calendar to make an appointment, then have a reminder automatically sent to each invitee along with a link to Google™ Maps showing where the meeting is. Mashups are important for businesses because they can be customized to address specific needs and problems, and have value by making business processes more efficient.

Q. What do MACs have to do with PSTN?

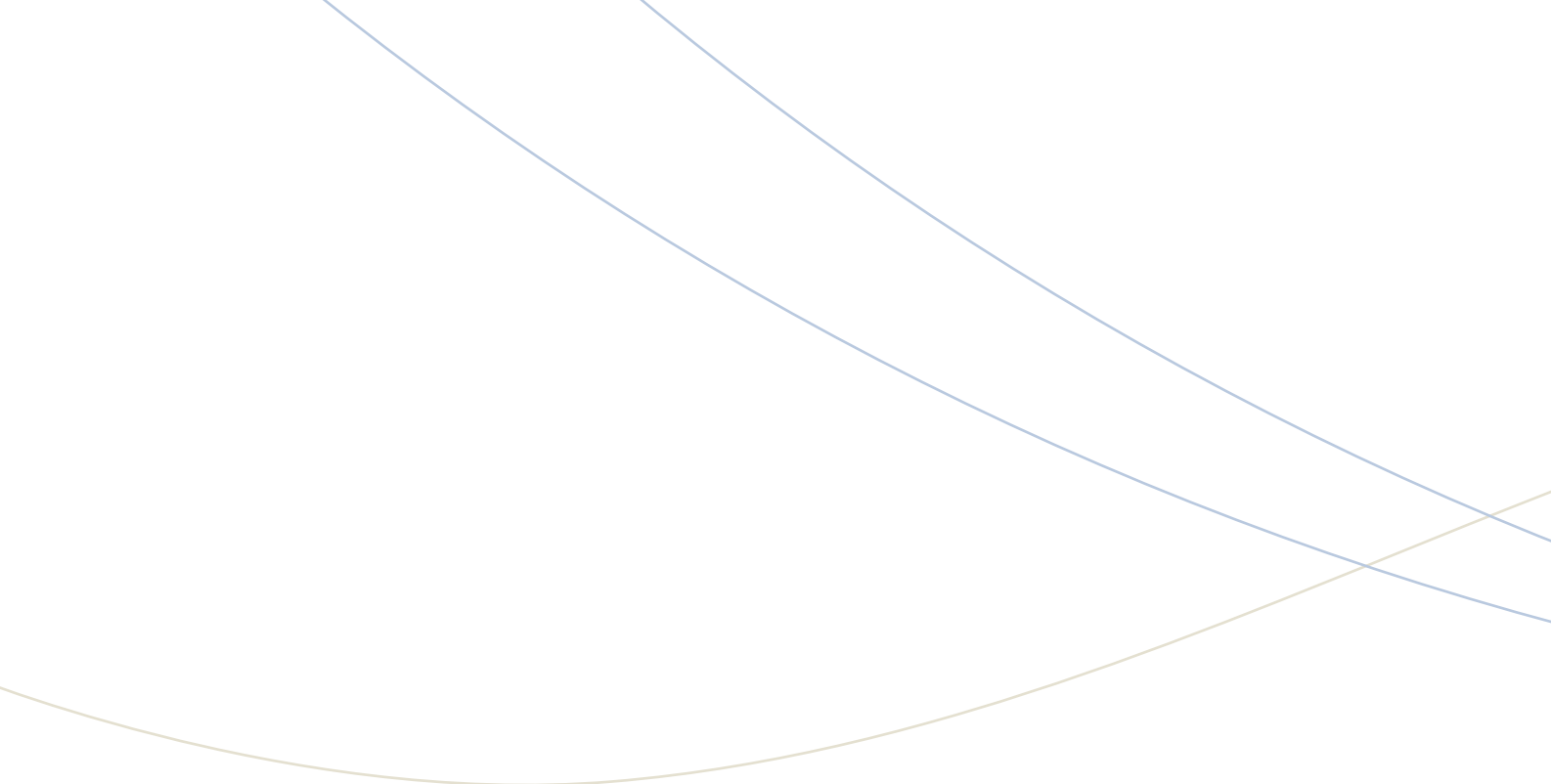
A. MAC means Moves, Adds and Changes. In the PSTN world, the service provider controls the network and the overall experience, right up to the deskphone. Whenever you need to move the physical location of the phone to another office, or add new lines, or make changes to the services / features, the carrier has to do it, and you pay for these changes. It's long been a lucrative revenue source for them, but IP telephony changes all that. With IP, you don't need the carrier to do these things – MACs can be self-provisioned, and no truck roll is required. Changing locations is a simple matter, so long as the phone jack you're moving to is connected to the LAN. The network will detect the IP address of the device from any jack in the system. The only time you'll need the PSTN now – your carrier, actually – is when you need to physically install new jacks. They're happy to do that and charge you, but they'd rather you kept using them from MACs since they make so much money doing that.

Q. How do you define business size (small, medium, large)?

A. Small business is generally defined as sub-500 users. Medium is sub-2000 users and large above that.

Q. Only the younger people will be savvy at the coming technology? Why is that when people today and now are creating the new ways that we will be using in communications?

A. This is a general observation and example only, and you are quite correct in this view as well.



Global Headquarters	U.S.	EMEA	CALA	Asia Pacific
Tel: +1(613) 592-2122 Fax: +1(613) 592-4784	Tel: +1(480) 961-9000 Fax: +1(480) 961-1370	Tel: +44(0)1291-430000 Fax: +44(0)1291-430400	Tel: +1(613) 592-2122 Fax: +1(613) 592-7825	Tel: +852 2508 9780 Fax: +852 2508 9232

www.mitel.com

For more information on our worldwide office locations, visit our website at www.mitel.com/offices

THIS DOCUMENT IS PROVIDED TO YOU FOR INFORMATIONAL PURPOSES ONLY. The information furnished in this document, believed by Mitel to be accurate as of the date of its publication, is subject to change without notice. Mitel assumes no responsibility for any errors or omissions in this document and shall have no obligation to you as a result of having made this document available to you or based upon the information it contains.

M MITEL (design) is a registered trademark of Mitel Networks Corporation. All other products and services are the registered trademarks of their respective holders.

© Copyright 2009, Mitel Networks Corporation. All Rights Reserved.

GD 176_4295

