SIP TRUNKING 101:
A Primer

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KEY BENEFITS

- Cost reduction in the order of 30-50%
- Increased flexibility, as numbers are no longer tied to locations
- Cost effective disaster recovery capability
- Support for wideband audio and video
- In sync with developing requirements for unified communications and collaboration

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Enterprise communications have been rocked by two major developments over the past decade. On the network side, legacy network services like PRI and frame relay have given way to all-IP based alternatives. On the premises side, traditional circuit switched or TDM-based PBX systems have been overtaken by hybrid or full voice over IP (VoIP) solutions and the functionality of those systems is now being enhanced by unified communications (UC). Capitalizing on these trends, enterprise users are increasingly moving to SIP trunking services to reduce cost and increase reliability while opening the door to enhanced functionality.

According to a 2012 study of 500 enterprise communications buyers conducted by The SIP School, 55% of companies are now using SIP trunking services and the percentage of companies reporting they are testing the service jumped to 26.0% from 16.6% a year earlier. The rapid shift to SIP trunking has also created more than its share of confusion. Each service provider will typically portray its SIP trunking offering as "the definitive way" that the service is offered- the reality is much different. There are significant differences in the way SIP trunking services are deployed, how they are priced, and the features and functions they can deliver. Enterprise users will need to have a clear picture of how they see their networks evolving to choose the most effective partners for their SIP trunking initiatives.

To be sure, SIP trunking services can meet the core reliability requirements of the enterprise. Unlike early consumer-oriented VoIP services, SIP Trunking can provide a secure, reliable, enterprise grade voice service that can significantly reduce cost. Importantly, organizations need not wait until they upgrade their TDM PBX systems to take advantage of the cost savings and increased flexibility SIP Trunking can afford. Further, as those voice networks evolve to unified communications, SIP trunking provides exactly the type of flexible communications services that will be essential in that environment.

The purpose of this paper is to take a step back and look at the overall market for SIP trunking services, describe the major areas where they differ, and to provide a better understanding of the alternatives and how those elements should be factored into the buying decision. Make no mistake about it, the migration to SIP trunking is underway (and with good reason), but there are important differences that must be understood to ensure a good outcome.

TDM to VoIP

The migration of the long distance portion of the telephone network to IP began in the late-1990s, as carriers found that it was far more cost effective to carry voice calls over an IP backbone than through a traditional network of fixed sized, dedicated channels. The interface to the customer continued to use legacy TDM technologies like T-1 and Primary Rate Interface (PRI) that were supported on traditional digital PBX systems. However, once those calls hit the public network they were converted into packet form and forwarded over an IP core. When the call arrived at the last central office, it was again converted back to a T-1 or PRI format (or to a basic analog interface) and delivered to the recipient.

As enterprise customers began migrating their PBX systems to IP-based alternatives, the TDM-based network access connection became an anomaly in the path. What's more, it was also an inflexible choke point in what was otherwise an end-to-end IP-based voice connection. However, transitioning to an end-to-end VoIP environment would call for a more flexible and functional mechanism for setting up connections and activating features.

This is where the power of SIP can be seen. Rather than the "one-size-fits-all" approach of the traditional telephone network, SIP signaling allows any type of real time connection, with any required bit rate, to be established through a

flexible IP network. What's more, SIP has features whereby the end points can advertise and negotiate the parameters of the connection including the voice or video encoding based on the range of options each can support. So as enterprises move to the rich communications environment defined by UC, SIP provides a signaling mechanism that is developed specifically for that type of environment.

SIP Trunking Configurations

In its simplest form, SIP Trunking is an alternative to traditional T-1 or PRI network connections between an enterprise PBX and a service provider. Rather than taking a 1.544 Mbps channel and dividing it into 23 or 24-fixed size 64 Kbps "voice" channels, SIP trunking uses an IP-based link of whatever size to connect the enterprise's telephone system to the service provider's network. That access link can be dedicated to SIP trunking or it can be shared with other IP applications.

In deploying SIP trunks, there are a number of configuration options available. First, SIP trunks can be deployed at each company location much the way traditional T-1 or PRI trunks are used today. The other option would be to deploy all of the SIP trunks at a single data center, interconnect all of the sites over an IP backbone, and route all external calls through that data center for access to the public network. The advantage of that centralized trunking approach is that larger trunk groups are inherently more efficient and you can typically use fewer trunks to support the total volume of calls; if the locations are in different time zones, those savings can be even greater. Of course, centralizing all network access introduces a single point of failure, so many organizations deploy SIP trunks at two or more locations (possibly including a disaster recover site) to ensure continuous network access.

To set up a call (or more specifically a "session") a series of SIP signaling messages are exchanged between the PBX or UC system and the network. When the session is in place, the voice or video is forwarded in a series of packets addressed to the other party. Those packets are transported over the IP backbone and at the far end they can be delivered over another SIP trunk or passed through a gateway to be delivered over a traditional analog or TDM-based connection.

While that description would be true of all SIP trunking services, there are a number of important differences in the way the service is implemented and the features that can be offered. Those differences will have an impact on both the total cost of ownership (TCO) as well as the services and configurations that can be supported.

Internet Backbone Options

Internet providers offer two types of IP services: basic IP and Multi-Protocol Label Switching (MPLS). Basic IP provides a "best effort" transport service where packets may be delayed or dropped randomly if they encounter congestion in the network. MPLS is a higher priced IP service where the operator will provide service level agreements (SLAs) specifying worst-case performance for delay, packet loss, and jitter (i.e. the maximum variation in delay from packet to packet).

As VoIP uses the User Datagram Protocol (UDP) to transport voice packets, there is no recovery from dropped packets. The reason for this is that if the sender were to resend a voice packet (or any packet carrying real time information), it would arrive too late to have any relevance to the voice stream. If too many packets are lost, the quality of the voice transmission will be degraded.

The possibility of packet loss has led many SIP trunking providers to tout the necessity of using the more expensive MPLS service to ensure adequate voice quality. However, there is a growing body of evidence that suggests this is an unnecessary cost. When consumer VoIP services like Vonage were first launched, the IP service providers were struggling to keep pace with the exploding traffic growth in their networks. As a result, voice quality did suffer to some extent leading to a bad reputation for VoIP services as a whole.

Fast forward a few years and we see a completely different picture for basic Internet service. Prices for higher and higher capacity backbone routers have plummeted, and the move to wave division multiplexing (WDM) on fiber optic transmission systems has allowed the IP network operators to cost effectively increase the capacity of their networks. The result is that packet loss and delay performance continues to improve to the point that many question whether an MPLS-based service is worth the extra cost.

The other advantage of using broadband Internet service is that the SIP trunking service can be turned up much faster. MPLS services must be designed and provisioned for each customer individually leading to installation intervals that are measured in weeks or months. Virtually every company has basic Internet service installed for web access and other applications, and if there is capacity available over those links, SIP trunking services can be turned up in a matter of days. Many customers who have installed SIP trunks in that fashion on a trial basis have found the performance of basic Internet service to be more than adequate and continue to deploy SIP trunks over those services.

One other factor that has contributed to the move to basic Internet services is improved voice coding and packet-loss masking options. VoIP systems are designed in anticipation of the fact that packets will be lost. As a result, VoIP receivers incorporate buffers and techniques to mask the impact of lost packets. Additionally, the low (and decreasing) cost of bandwidth, coupled with predictable and infrequent Internet transit delays create a viable architecture and strong economic case for routing real-time media applications that still deliver high QoS over the public Internet. This avoids the need to deploy expensive, parallel connectivity over the pre-existing low cost broadband connection.

Premises Equipment Compatibility: Gateways and Session Border Controllers (SBCs)

Premise – based equipment is required to connect to a SIP trunking service. The specific requirements are based on the type of telephone system in use and the options are typically a gateway or a session border controller (SBC).

Connecting a traditional TDM-based PBX system to a SIP trunk will require some type of gateway device; the trunk interfaces on those systems are typically limited to analog, T-1, and PRI. It is important to note that you need not defer your transition to SIP trunking based on a legacy PBX. Gateways take one of those traditional interfaces and convert it into a VoIP/SIP interface. Gateways are available from companies like Adtran, AudioCodes, and Sonus (NET). IntelePeer can supply a list of gateway devices that are tested and certified to work with its services.

If you have already made the shift to an IP PBX or UC solution for voice, those systems can connect directly to a SIP trunk, however you will need to be cognizant of the compatibility issues. Like any data protocol, the SIP standards describe a wide variety of options, which means that not all SIP-capable devices and services can fully inter-operate. Fortunately, SIP trunking involves far fewer features than what would be found on a multi-button SIP handset installed behind an IP PBX. SIP handsets will need to support hold, conference, transfer, multiple line appearances, message waiting lights and the rest of the feature set typically found on a modern PBX system. Even though the feature set with a SIP trunk may be more limited, you would be wise to search out SIP trunking providers that have been certified to interoperate with all of the various PBX and UC systems you have deployed.

While an IP PBX or UC system may be directly compatible with the SIP trunking service, some customers choose to install a session border controller (SBC) between the telephone system and the SIP trunking provider. SBCs can provide a number of security functions like protection from denial-of-service (DoS) attacks, toll fraud via rogue media streams, network topology masking, and malformed packet protection. SBCs can also translate between different voice coding systems and incompatible SIP signaling implementations. SBCs are available from leading companies like AudioCodes, Avaya, Cisco Systems, Oracle (AcmePacket) and Sonus. IntelePeer can supply a list of SBCs that are fully tested and certified to work with its services.

911, E911 and Nomadic E911 Capability

The other important (and often overlooked) element in SIP trunking is compatibility with Enhanced 911 (E911) services. The 911 emergency call service operates on a special network that was designed to work in conjunction with the traditional circuit switched telephone network. When a user dials "911" the call is routed to the geographically appropriate Public Safety Answering Point (PSAP). Where basic 911 provides the telephone number of the calling party to the PSAP agent, E911 does a database dip and provides the location as well; the availability of E911 capability is dependent on the subscriber's local PSAP. Needless to say, the nature of IP-based services adds another level of complexity to that equation.

Mobile devices present a particular challenge in E911 services. UC clients running on tablets or laptops can be highly mobile as users may roam throughout an enterprise campus, transition between telework locations (e.g. a home office,

remote office, the local Starbucks, etc.), or may even be traveling around the world. In order to appropriately establish the location of these users at the time of an emergency call, a dynamic location identification service will be required.

In looking for potential SIP trunking providers it is essential to confirm that the provider can support 911 and E911 service at all locations as well as offer solutions for mobile or nomadic UC users. Further, you will need to confirm the specific addresses (including floor or building numbers) associated with each DID number that is passed to the PSAP. Finally, given the potential life-and-death nature of those calls, it will be important to understand the various levels of backup the provider has implemented to ensure that those calls are delivered in a timely fashion.

SIP Pricing Plans and Cost Advantages

There are a number of different pricing plans offered for SIP trunking services, which will directly impact TCO. Among the pricing elements and options you will find are:

- **Access Connection**: In virtually all cases the basic connection between your office and the SIP trunking provider's point of presence will be billed separately. As we are essentially talking about an Internet access connection, there are a number of different options. For small locations or telecommuters, DSL or cable modem services could be used. Larger locations can be served with multiple T-1 (1.544 Mbps) or T-3 (44.736 Mbps) dedicated links. For even higher capacity access, very cost effective metro Ethernet and wavelength services are available. Those access links can be used to support SIP trunking services exclusively or SIP trunks can share an access connection with web access and other data services.

  Sizing the access link appropriately will be critical to success. Again, it is important to include the IP overhead, so a 64 Kbps (G.711) channel requires about 90 Kbps and an 8 Kbps (G.729a) channel requires about 30 Kbps. If video calls are to be carried as well, you will need to ascertain the required capacity per channel from the video supplier.

- **Calling Charges**: As with traditional telephone services, there is often a cost per minute for calls placed over the SIP trunk. Cost per minute is typically lower than with traditional calling services due to the advantageous tariff and tax rates applied to IP calls. If you intend to be placing video calls as well, it is important to determine how those calls will be billed. IntelePeer offers its customers the option of choosing such a minute of use (MOU)-based billing model.

  SIP trunking providers are now adding features like those found in traditional T-1 and PRI services like included local calling and toll free service or the ability to transfer or reroute calls based on time of day or in the event of a disaster, facilitating the PRI to SIP trunk price comparison.

- **Simultaneous Calls** ("Call Paths", or "Ports"): Many SIP trunking providers base their pricing in part on the maximum number of simultaneous calls or ports the service will support. One flexibility advantage in SIP trunking is that you can purchase the exact number of ports you require. In leasing traditional T-1 or PRI network access, you are forced to purchase inflexible bundles of 23- or 24-voice trunks.

  One feature that can add to the cost effectiveness is port sharing, which allows idle ports at one location to be used for calls at other locations. For example, if you have 10 ports configured at each of your offices in New York, Atlanta, and Los Angeles, but no one is using the ports in Los Angeles (e.g. early in the morning), those 10 ports are available to support calls at the New York or Atlanta locations. That scenario assumes there is sufficient capacity on the access links in New York and Atlanta to support those additional calls. IntelePeer offers its customers the option of choosing such a port-based billing model.

- **Bundled Services**: Some SIP trunking providers are going to a simpler pricing structure based on the number of users. Rather than trying to determine the exact number of trunks required based on a voice traffic study (i.e. "Erlang study"), local and long distance service can be priced at a flat monthly rate per user. In some cases those bundles can be priced like a flat-rate SIP port that is shared by a number of users. Those bundled pricing arrangements will typically have a fixed premium to support video as well as voice calls. IntelePeer offers both user- and port-based pricing options that include unlimited local and long distance calling, E911, as well as an allocation of toll free and local DID phone numbers, port-ins and directory listings.

The ROI for SIP Trunking

In today's economic climate, there is a real need to establish a sound ROI for technology initiatives. A recent white paper published by Unified IT Systems illustrates the potential savings for a 5,000 employee organization migrating from TDM to SIP trunking and finds those savings to be in the order of 30% to 50%. The impact can be seen in a number of areas.

- **Trunk Aggregation**: The majority of enterprise customers deploy SIP trunks in conjunction with a strategy of centralizing all of their voice network access at one or more data centers. Distributed sites are typically interconnected with an MPLS virtual private network, and the cost of the additional capacity to carry the voice traffic to the data centers is minimal. One of the basic principles of traffic engineering is that one large group of trunks handles traffic more efficiently than several smaller groups, and that reduction in the number of trunks (or call paths) can be on the order of 25%.

  An additional efficiency from centralized trunking can be seen when locations are scattered across several time zones. The busiest hours for telephone usage generally occur around 10 to 11:00 AM and 2 to 3:00 PM local time. With locations in different time zones, those busy hours will occur at different times reducing the total number of call paths required to support busy hour traffic peaks.

- **Flexibility**: Where T-1 and PRI access arrangements require that trunks be purchased in bunches of 23 or 24 voice channels, with SIP trunking an organization can order exactly the number of trunks it requires. To estimate the savings, you can assume a monthly saving equivalent to half the cost of an additional PRI.

  Those savings can be further enhanced if the organization has highly seasonal calling patterns. Retailers have traffic peaks prior to holidays and tax preparation companies peak prior to April 15. SIP trunking services typically allow ports to be added or deleted quickly.

- **Aggregated Access**: For each T-1 or PRI access connection, the customer has to rent a 1.544 Mbps dedicated line to the carrier's serving point. With data access there are considerable economies of scale (i.e. the higher the access capacity the lower the cost per bit). Using Metro Ethernet or other high capacity access options, access costs can be cut by as much as 80%.

- **Network Features**: SIP trunking services often include services that are separately charged for on PRI services. Features like call transfer may cost $50 per trunk per month, making a SIP option far more advantageous.

- **Bill Processing and Administration**: Organizations spend considerable time and effort processing and auditing telephone bills, tracking usage, and setting up internal chargebacks. SIP trunking providers may offer flat rate bundled service plans effectively eliminating the administrative overhead of the traditional phone bill. Further, having all of the network access terminated at one or a small number of sites rather than scattered to all network locations also simplifies the administration and maintenance.

What's Next for SIP Trunking?

At the outset, SIP trunking is being seen as a lower cost, more flexible alternative to traditional T-1 or PRI network access to support basic voice telephony. However, to look at SIP trunking as an apples-to-apples swap for those legacy services would be missing a big point about SIP.

With the advent of UC, the standalone nature of voice service is changing. In a UC environment, voice is just one of a number of communications options that will be available to the user. Presence capability will allow colleagues to determine if a party is available for a communications session and what type (i.e. text, voice, video, web collaboration). The caller can then request the connection with a single click (or escalate the session from one communications modality to another) and their UC system will generate the required string of SIP messages to establish a point-to-point or conference connection.

The task of merging all of those various communications capabilities into a single shared interface goes far beyond the ability of traditional trunking interfaces and signaling protocols. The basic design of SIP envisioned this type of rich communications environment and incorporates the tools to establish a variety of sessions, support multiple media

types, and even negotiate the types of encoding that will be used. This will allow for such functions as upgrading a text conversation to a voice or video call, or to integrate calendar and conferencing capabilities so a user can join a conference by clicking a single button.

Probably the most important addition to SIP is SIMPLE or *SIP with IM and Presence Leveraging Extensions*; a similar set of capabilities exists with XMPP (the Extensible Messaging and Presence Protocol). As more organizations move to UC-based communications solutions, they will need to extend those rich communications capabilities to suppliers, partners, and other outside parties. UC federation, or the ability to link UC functionality between different platforms in different organizations, will be key, and tools like SIMPLE and XMPP will provide the essential signaling elements required to bring that about.

The other major development is cloud-based deployments, and we already have dozens if not hundreds of providers offering cloud-based IP PBX and UC solutions on virtually every platform available. As users move some or all of their users to these cloud-based offerings, SIP trunking will still be an element in the solution as the cloud providers have been among the first to make the change.

Conclusion

Enterprise communications has entered a new phase with the advent of unified communications. Users will now be able to intermix their real time, near real time, and asynchronous communications and access a wide array of collaboration tools in new ways that spur productivity and speed decision making. Whether we are looking at traditional office communications or the expanding range of options in the contact center, UC is having a major impact on how people will communicate and collaborate.

That type of rich communications will call for a new set of tools with the ability to grow and expand to support a continuously evolving set of capabilities. SIP trunking provides just such a capability, and one that can reduce TCO in the short term while providing the foundation on which these new capabilities can be deployed as time goes on.

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Michael Finneran is Principal at dBrn Associates, an independent industry analyst and consulting firm. Along with providing consulting assistance to carriers, equipment manufacturers and end-user organizations, Michael is a frequent speaker at industry conferences including InterOp, Enterprise Connect, and the Information Week 500. He has published over 300 articles as well as numerous white papers and market reports. His blogs appear regularly in Information Week, NoJitter.com, UCStrategies.com, and the Voice Report. Michael is a member of the Society of Telecommunications Consultants and he has a Masters Degree from the J. L. Kellogg Graduate School of Management at Northwestern University.

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