SIP trunks

Deployment note

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Introduction

SIP trunking has become a byword in recent times and it surely is a bellwether service – not merely an indicator of future trends; rather a leading trend in its own right.

SIP trunking – via so-called SIP trunks – is a service offered by carriers and Internet telephony service providers (ITSPs) that allows businesses to use voice over the Internet protocol (VoIP) across their Internet connection. It is used to extend VoIP or IP telephony use beyond the enterprise network. Essentially, it provides direct IP connectivity to an IP-PBX or call/communications manager and permits VoIP-enabling an enterprise's legacy PABX, via a gateway. It may also be used outside the enterprise network to provide functionality for home workers or other remotely located staff.

For businesses with brand spanking new IP-PBXs, SIP trunks satisfy the fundamental need to communicate externally as well as internally over IP. This allows them to make full use of their IP-based capital expenditure (CAPEX) purchase, without the need for a separate enterprise gateway to connect to the traditional public switched telephone network (PSTN).

Whereas some vendors suggest that SIP trunks mean the end to bundles of physical wires delivered from a service provider, we shouldn't forget that those very same 'bundles of wire' are very often still used for the physical Internet connection – the SIP trunk from the ITSP.

A SIP trunk allows a company to achieve PSTN connectivity indirectly via its service provider or an ITSP. This eliminates the need to own and maintain PSTN gateway hardware for an IP-PBX. And importantly, SIP trunks offer significant cost savings compared with the cost of traditional fixed PSTN lines; ISDN basic rate or primary rate interfaces (BRI/PRI).

This application note looks at SIP trunks in a little more detail…

What are SIP trunks?

SIP trunking provides direct access to and from a service provider or ITSP, with the ITSP providing the gateway connection to the PSTN. SIP trunks are seen as a viable alternative to traditional ISDN as businesses use SIP trunking to provide cost-effective communication services to their offices through an IP-PBX. See diagram 1.

Whilst VoIP can mean different things – H.323, Skype, IAX, IP or Internet telephony, Asterisk, etc. – practically all ITSPs and carriers use the session initiation protocol (SIP). SIP is the signalling protocol that controls setting up and tearing down a media session (think of it as a phone call in this context). The voice path (or media) is controlled via the real time protocol (RTP) and delivered as normal IP packets. For those concerned with security, transport layer security (TLS) and SecureRTP (SRTP) can be employed on SIP calls.

The term ‘SIP trunk’ has a different interpretation in some markets. It can be used to refer to individual SIP sessions or the capacity to handle multiple concurrent sessions. In practice, businesses will buy a number of SIP trunks depending upon the nature of their business and the volume of calls, much in the same way they procured a number of BRI or PRI lines from a PSTN service provider. Ideally, the number of ‘trunks’ or ‘session’ capacity should be based on the anticipated peak volume of calls. In the PSTN world, such provisioning was based on ‘busy hour’ analysis.

SIP trunks can be based on the traditional trunk bundles sold within a particular market. Certainly, in the United Kingdom, SIP trunks are sold in a similar fashion to ISDN30e lines. This means that 8, 15 or 30 SIP trunks will allow you to have 8, 15 or 30 simultaneous calls, respectively. Whereas an E1 trunk would typically allow up to 30 simultaneous calls, you would need 30 SIP trunks to achieve the same thing.

In other parts of the World, the term ‘SIP trunk’ is given to a single connection, regardless of bandwidth or delivery technology. One SIP trunk in this case enables a number of concurrent SIP sessions (calls). However, for the equivalent bandwidth, using the same codec, the simultaneous call capacity is the same.

These definitions can bring quality and other issues to light for an enterprise. Some areas to watch are discussed below.

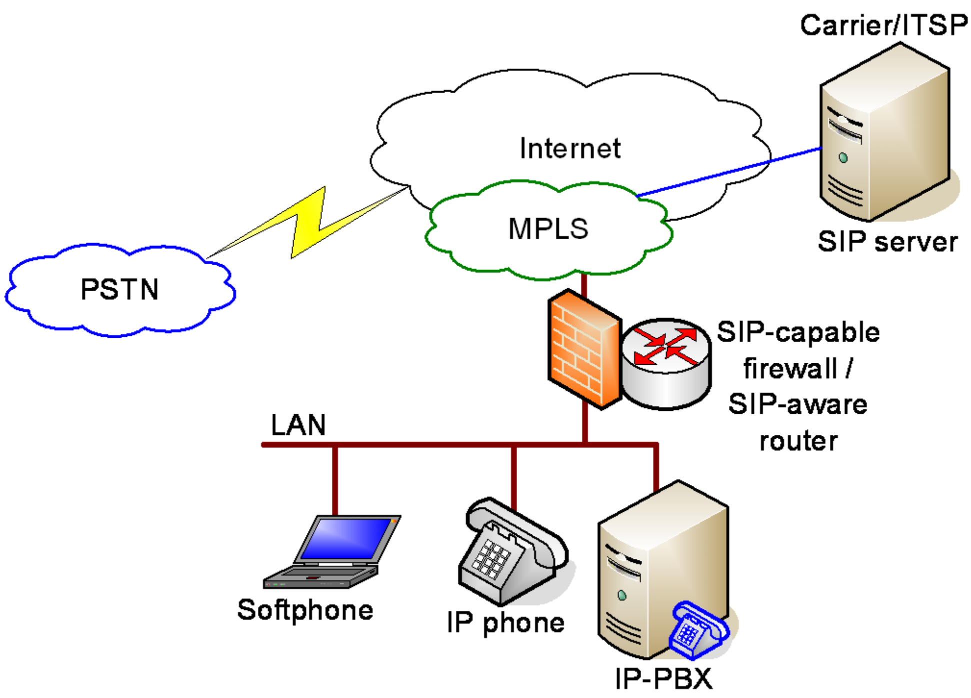

Diagram 1 – SIP trunking from an ITSP

SIP trunks presentation

SIP trunks connect to an IP-PBX just as ISDN PRI lines (traditional E1s or T1s) connect to a legacy PBX, and they can offer capacities equivalent to BRI, PRI or fractional PRI lines. Such ‘PRI equivalence’ is ideal for small-to-medium businesses (SME/SMBs). In other ways, they are not much different from any broadband or Internet connection, where bandwidth is a key metric.

The physical connection can vary according to the enterprise’s needs. It can be as small as a single DSL line or reuse a carrier’s existing E1 or T1, or grow to DS3, Gigabit Ethernet and more. A SIP trunk can be provisioned as a separate line or provided via an existing Internet connection. As an ITSP’s service also provides access to the Internet, an advantage stems from being able to tap into unused bandwidth and gain savings through lower operating expenses (OPEX). Additional benefits stem from the inherent scalability and the ease of adding extra capacity as SIP trunks can be added one-by-one as opposed to bundles of 23 (or 8, 15 or 30).

Beneficially, businesses can also retain their existing number ranges using SIP trunks. As the ITSP will use the electronic numbering (ENUM) protocol to convert E.164 telephone numbers – inward direct dial (DDI/DNIS) numbers – into IP addresses (and vice versa), they can be resolved by the Internet's DNS system like traditional website domains.

The essential physical connection components for voice communications are an IP Internet bearer circuit and a host telephone line. Those copper cables are still there (or it may be a co-axial connection from a cable operator), nevertheless the ongoing costs will undoubtedly be less than for the equivalent ISDN line. As a SIP trunk is largely software and IP-based, it is much easier to manage remotely and, therefore, cheaper for the service provider to maintain than traditional ISDN connections.

The line is terminated by a SIP aware router or an enterprise border element. This border element can be a firewall, with complete support for SIP, or an edge device connected to the firewall. These devices handle the traversal of SIP traffic to and from the enterprise using network address translation (NAT). The IP-PBX is typically connected to the firewall via an enterprise LAN in the same way as other data equipment.

Taking a balanced approach

Two key factors must be considered when sourcing SIP trunks from any ITSP or service provider. Reliable access to bandwidth is essential for successful SIP trunking operation. Voice traffic contention with data traffic is the other key factor. Issues such as jitter, latency and packet loss are related to bandwidth restrictions and each has an impact on contention for bandwidth amongst routine Internet traffic. Letting the packets fight it out is no way to operate a voice service that should be as reliable and dependable as traditional PSTN services.

Consequently, it is good advice to ensure that the telco or ITSP provides the required bandwidth. When you order 10 SIP trunks – or one SIP trunk with 10 concurrent sessions – you need enough bandwidth to make 10 simultaneous calls.

Regardless of the type of business, both upload and download speeds can be important, which is why a low contention symmetric digital subscriber line (SDSL) is offered by some providers. Note that in some cases, if the distance to the local exchange is too far for an asymmetric digital subscriber line (ADSL) connection, SDSL will often be the default.

Notwithstanding the fact that most voice conversations offer balanced traffic to the network (both parties speak about half the time), the reasoning for SDSL is simple. When the same speeds for both upload and download are available, the maximum number of full duplex calls is not limited to the lowest common denominator. With an unbalanced line, the maximum number of calls is limited to the lower (upstream) rate.

As a rule of thumb, around 80-90kbits/s of bandwidth is needed for a single VoIP call using G.711 encoding. With an ADSL connection having an upstream rate of 640kbits/s, the maximum number of full duplex calls is 7 or 8, regardless of the downstream rate. Conversely, using a 1.5Mbits/s SDSL link, equivalent to the downstream rate of a typical ADSL link, the maximum number of calls is between 16 and 18.

Balancing the load

One of the advantages of SIP trunking is the ability to pool phone and Internet connectivity over a single connection. ITSPs achieve the demarcation between traffic types by prioritising phone calls over data transfers using quality of service (QoS) methods or private multi-protocol label switching (MPLS) networks. This can be viewed as a ‘pipe within a pipe’ method in order to avoid contention between voice and data traffic.

From a QoS perspective, ITSPs manage different kinds of data streams based on priority and differentiated service plan. So for customers that subscribe to SIP trunks, they can see minimal latency and packet loss. Service providers use label edge and switching routers in their MPLS networks to divert and route traffic based on data stream type and Internet access priorities.

As an example, a representative ITSP product for SME/SMB scale deployments might consist of a 3Mbits/s, 1:1 (non-contended) SDSL circuit. This could be offered to the customer as a SIP trunk guaranteeing up to 15 concurrent, high quality, full duplex, G.711 SIP calls, with the remainder of the ‘pipe’ available for Internet data. This number of SIP calls is equivalent to a fractional, 15 circuit ISDN line. However, it brings significant cost benefits as ongoing monthly OPEX savings can be between 30% and 70% compared to traditional PSTN rates.

Interoperability

Another significant issue to consider is interoperability. Once again, it is important that the SIP service provider has tested its circuits with the relevant system or equipment in use. The majority of carriers are keen on promoting interoperability and testing. This is particularly true in Europe and the United States, where SIP trunks are becoming increasingly popular. The SIP Forum's SIPconnect programme exists to promote interoperability through its SIPconnect Technical Recommendation.

The majority of carriers and ITSPs offer a fully integrated voice and data service, and this is surely the prudent route to follow. Quite apart from the OPEX benefits, if SIP trunks are procured as part of a solution that also includes the underlying Internet connection, an enterprise is likely to get a more solid, reliable and dependable service. Direct connectivity to an ITSP, as opposed to wholesale termination, means calls travel over a dedicated, non-congested connection. This serves to limit traffic congestion and maintain call quality.

A hosted alternative

SIP trunks are also used to provide managed or hosted IP telephony services, where there is no IP-PBX at the enterprise premises, only SIP phones. A tier 1 or tier 2 carrier, service provider or ITSP, commonly an LEC, ILEC or CLEC in the United States, will supply a dedicated, fully managed connection from their point of presence (PoP) to the enterprise site. This is a popular OPEX alternative for many SME/SMBs that are not prepared for the capital investment (CAPEX) needed to procure an IP-PBX.

For developers

Aculab has a range of readily available enabling technology products that provide application developers or service delivery platform vendors with the tools needed to flourish in this SIP trunking environment. Aculab's integrated SIP stack is common to both Prosody X hardware and its Prosody S software alternative, meaning developers can take an either/or approach to their application deployments. Changing between one platform and the other is as easy as making the choice.

For built in, 'in the skins' TDM-to-SIP gateway functionality for a traditional automatic call distribution (ACD) platform, Aculab's Prosody X hardware range provides the answer. This is equally so if an IP media processing resource board is needed for a new, IP only contact centre platform.

For enterprise voice connectivity Aculab is able to offer a 'best of both worlds' solution. The all-IP customer with a SIP-based software application server can connect to a SIP trunking carrier via Prosody X. And the legacy customer's PBX or TDM-based application server can connect to the SIP trunk via Prosody X configured as a gateway.

Additionally, in an IP only SIP trunking environment, Aculab's Prosody S host media processing software is also suitable, both for systems deployed in the enterprise and for hosted, SIP-based application services.

The SIP standard is comprised of a large number of recommendations – the SIP eco-system – and most vendors do not implement all of them. SIPconnect is a specific subset of these recommendations, which defines SIP trunking. Many SIP trunking services right now only support basic voice traffic and features. Requirements, such as IP (T.38) faxing and other additional or supplementary functions are not yet supported on many SIP trunking services. This is also true for call progress analysis (CPA).

In terms of interoperability, Aculab's SIP stack offers developers access to the essential functionality needed for compatibility with the SIPconnect Technical Recommendation. Operation can be largely autonomous, through the 'below hood' operation of the SIP stack itself, or via extended, low level API access. Using the Aculab API, developers can implement the functionality needed by a SIP application server (SAS) or customer premise equipment (CPE), such as a SIP-based IP-PBX or equivalent.

Functionality such as interacting with SIP Proxy and REGISTRAR servers for registration and authentication is inherent in the Aculab SIP stack. Relaying cause codes between TDM protocols and SIP as per IETF RFC 3398 (something which real world feedback suggests there is a lack of strict adherence to), can be dealt with at the API level, enabling such codes to be appropriately translated.

Call transfer is a common interoperability problem as some ITSPs and providers do not support this feature, and support is also lacking in some SIP user agents. An end user system built on a foundation of Aculab's products has no such issues as call hold and transfer are supported via the SIP stack with both Prosody X and Prosody S. Third party call control is also inherently available with Aculab's SIP stack, enabling cost-efficient utilisation of media resources whilst retaining full signalling control of all calls, much like a back-to-back user agent (B2BUA).

An exciting aspect of Aculab's offering, which further reinforces the concept that architecting high availability systems is easier in a SIP trunking environment, is also available. Dual redundant SIP, discussed in a recent white paper from Aculab, can be very effectively used with multiple SIP trunks. Utilised at either or both the ITSP and enterprise ends of the connection, dual system redundancy for the SIP signalling can help to ensure no compromise, carrier-grade ('high-nines') reliability and resilience.

Conclusion

A SIP trunking service provides a simple, cost-effective solution for business customers that want a VoIP telephone service for use with their SIP enabled VoIP phone system. SIP trunks also provide a competitive solution for extra telephone lines or for adding extra DDI/DNIS capability, or back up of traditional telephone line solutions.

However, SIP trunking by itself is not the whole connectivity story and an enterprise must be aware of other potential issues. Bandwidth, reliability, and interoperability must all be considered when making a decision regarding how a business connects to its customers.

Aculab's Prosody line of media processing products provides developers and service providers with the tools to create SIP trunking products and solutions, which meet the rigid performance demands of this technology. The Aculab SIP stack supports numerous capabilities beyond the baseline set defined by SIPConnect. The feature set and flexible API help facilitate interoperability of products developed on the Aculab platform with the broadest set of ITSPs and vendors in the marketplace.

About Aculab

Aculab is an innovative, market leading company that provides world class IP and media processing boards and software to the global communications market. With many years of experience in helping to drive our customers' success, our enabling technology provides the essential components required to deliver multimodal voice, data, fax and video solutions for use within IP, PSTN and mobile networks – with performance levels that are second to none.

Aculab serves the evolving needs of developers, integrators, service providers and equipment manufacturers with cost-effective, deployment proven, high performance products. Companies worldwide have adopted our technology for a wide variety of business critical services and solutions.

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