WHITEPAPER

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SIP Trunking and SMEs

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Executive Summary

As businesses modernize their communications infrastructure, it is becoming more common to transition to an IP telephony-based system. To maximize the return on their investment, many businesses are also choosing to invest in SIP trunking, which has been shown to reduce costs related to voice and data connectivity by up to 50%. Additional savings are often also realized as a result of lower long-distance charges due to the increased utility of VoIP services when paired with SIP trunking; businesses also report increased productivity through the leveraging of new communication and collaboration functionality.

IPT

As telecommunications infrastructure ages, it usually requires increasing amounts of attention in the form of repairs, upgrades, and even replacements. In recent years, more businesses are coming to the conclusion that rather than continuing to throw time and money into maintaining their increasingly irrelevant TDM (time-division multiplexing) technology, it makes more business sense to effect a transition to the more modern and more functional IPT (Internet Protocol telephony) framework.

At its most basic level, IPT is the use of general-purpose data networks to carry telephony services such as voice and video. This is in contrast to traditional telephony systems that use special-purpose networks to do the same thing. The special-purpose nature of traditional TDM networks means that, in general, they are less flexible and less easily adaptable than IPT networks; as a result, TDM networks are often more expensive to set up and to maintain. Special-purpose networks also require special-purpose hardware, which requires a significant capital investment in equipment that performs only one function.

It is that significant investment in legacy technology that is primarily responsible for the slow initial adoption of IPT technology in corporate spaces. Companies that had invested heavily in traditional communications infrastructure were tied to that technology until they had managed to recoup that initial investment. This is why most businesses make a more gradual shift from TDM to a hybrid TDM/IPT system--with a pure IPT system planned for the eventual future; although, to be sure, some companies have transitioned directly to a pure IPT framework. Those companies that gradually transition give themselves the opportunity to continue making the most of their heavy investment, while also taking advantage of the cost savings and advanced functionality afforded by the newer IPT technologies.

Although most companies do experience lower set-up and maintenance costs with IPT infrastructure, it's really the high ROI (return on investment), lower management overhead, and increased functionality of many IPT applications that provide the greatest cost savings. Applications such as

Voice over Internet Protocol (VoIP), Unified Communications (UC), and Session Initiation Protocol trunking (SIP trunking) allow for previously unattainable levels of efficiency, functionality, and ease of deployment.

VoIP

Voice over Internet Protocol is, as the name implies, the transmission of voice communications over a data network using standards-compliant Internet protocols. The VoIP software converts audio to data, which is then transmitted over a general-purpose data network before being converted back into sound on the other end. The process works on private, internal networks as well as on the public Internet; this means that with only minor alterations--or, at times, with none at all--it often can be provisioned using a company's existing data network.

Although VoIP was once strictly the domain of very large enterprises, the basic technology now is mature enough that it is available at the consumer level on an individual basis, and it is commonly used by small and medium-sized businesses. In addition to a lower total cost of ownership, VoIP also provides enhanced security, improved reliability, and lower local and long-distance calling costs. Most of the major telecom providers actually use VoIP technology to carry their own long-distance traffic.

UC

Unified Communications (UC) is a catch-all phrase that describes a collection of technologies that together serve the purpose of combining all of a company's communications mediums into one interface. UC packages commonly incorporate VoIP, videoconferencing, Remote Desktop functionality, and virtual whiteboards.

In addition to making better use of a business's already-existing data network, UC cuts costs and reduces management overhead by consolidating several services; it also provides vastly superior ease of deployment, geographical independence, and additional functionality when compared to traditional telephony services. By handling both voice and video as data, UC frameworks offer advanced messaging capabilities, as well as the means to easily store audio and video communications so that virtual meetings can be replayed at a later date.

Ease of deployment and flexibility of use are the result of UC systems being software routed rather than hardware-based. Provided that users are connected to the network, regardless of where they are connected from, the system treats them the same way. This means that a single number can be used for each employee--regardless of where he or she is working--and that each employee is always available, even if that employee is mobile.

SIP Trunking

While VoIP and UC tend to capture most of the limelight, it's SIP trunking that can really provide the lion's share of the cost savings when it comes to IP telephony solutions. SIP stands for Session Initiation Protocol and is a way for two devices on a network to establish and maintain real-time, multimedia communication. SIP trunking uses that protocol as a way to share multiple communication streams over one line.

What SIP trunking effectively does is this: It extends the utility of VoIP and UC solutions to the world outside of the enterprise. Instead of connecting directly to the public switched telephone network (PSTN) through a traditional private branch exchange (PBX), the company connects to an Internet telephony service provider (ITSP) via an IP PBX over a broadband connection. The ITSP can then route traffic over the data network or the PSTN, as appropriate, resulting in greater efficiencies, increased reliability, and cost savings. This document will examine how SIP trunking works and what benefits it holds for modern businesses.

Defining SIP and SIP trunking

SIP

Session Initiation Protocol is an open-standard signaling protocol maintained by the Internet Engineering Task Force (IETF). It is used to control real-time, two-way streaming of multimedia data, most often in applications such as video conferencing, VoIP, telepresence, and messaging. The session initiation protocol is what tells all endpoints of the communication what type of data to expect and in what format. It can be used for communication between two parties, as with traditional phone calls, or it can be used for communication among multiple parties, as with videoconferencing.

SIP is very similar to HTTP in that it is text-based and, therefore, directly readable by humans; in addition, it is a relatively simple protocol that can be used to develop complex systems. SIP is not responsible for the actual transfer of data over the network; instead, it establishes the connection used to transfer the data, manages that connection while it is in use, and closes the connection once it is no longer needed. Managing the connection includes tasks such as routing and authentication as well.

Because it is an open standard, implementation can vary widely from vendor to vendor. This is why, when using an SIP-based communications system, it is important to consider certification and to perform interoperability testing. This is also why, in most cases, two SIP-enabled devices will communicate by way of an intermediary called a SIP proxy, which can act as a sort of interpreter between the two systems.

One of the key functions of SIP as it pertains to use in the enterprise space is its role in determining user location. SIP functions for IP telephony in much the same way as a DNS server does for IP addresses on the public Internet, converting users' names on the network to their current location, which is what allows a connection to be made with that user regardless of where they are. This is a particularly important function of SIP in business environments that have highly mobile employees or employees who work from multiple locations on a regular or semi-regular basis.

Another important function of SIP is that, in addition to providing a mechanism for participants to agree on which features will be used in a given communication session, it also allows for those features and even the participants to be modified while that communication session is in progress. This provides an easy way for users to be added to or removed from videoconferencing sessions and virtual meetings, for example. It also allows for users to switch from a voice call to a video call on the fly, or for the addition of a virtual whiteboard or remote desktop functionality without needing to end and then re-initiate the communication session should that additional functionality be required suddenly.

SIP systems are composed of two types of user agents: clients and servers. While it's possible for one user agent to be acting as both client and server, generally clients are comprised of the individuals using the network, or the software they're using, and servers are the parts of the network that process the requests sent by those clients.

There are four general types of servers in a SIP network. A proxy server functions in much the same way as a router on a data network. Requests are generally sent from a client to a proxy server on the client's network, which then forwards the request to a proxy server on the recipient's network. The proxy server on each network is responsible for determining the location of their respective client and delivering the request properly.

A redirect server is responsible for returning the request to the client in a situation where the recipient's location is unknown or when the recipient cannot be located. This is the IP telephony version of HTML's 404 error.

The location server and the registrar work together to keep an up-to-date record of all client locations on the network. From time to time, client applications will report their location to the registrar, which then records that location in the location server. The records in the location server are then used by the proxy server in order to successfully deliver requests from other client applications--either from within the network or from outside of it.

There are only six commands in the main (RFC 3261) SIP specification, but, using them, it is possible to construct an extremely complex system that mirrors the functionality of a traditional PSTN endpoint. Those commands are INVITE, ACK, BYE, CANCEL, OPTIONS, and REGISTER.

Although there are other protocols that perform similar functions in the IP telephony industry, SIP is by far the most commonly used. Its simplicity, flexibility, and power make it the obvious choice for most IPT applications.

SIP Trunking

In the telephony world, trunking refers to sharing one line or frequency among multiple users. SIP trunking is a method of using the session initiation protocol to share a network connection intelligently among many users--often an entire company.

Before the advent of IPT technology, companies would have to provision enough hard lines from their telephone carrier to accommodate their peak usage--not their average or normal usage. This invariably resulted in significant under-utilization of very expensive infrastructure, because most businesses, most of the time, use a great deal less than their peak usage bandwidth. Under-utilization, especially of such a costly resource, unfortunately is just another way of saying that they were overspending.

SIP trunking frees businesses of the need for multiple, expensive hard lines and the overspending that goes with them, while at the same time allowing them to make full use of their already existing broadband data connection.

Rather than provisioning the network for peak usage even when peak usage capacity is unnecessary, SIP trunking allows a business to adjust capacity on-the-fly by making changes to specifications such as the type of encoding used. During times of minimal load on the system, codecs such as G .711 and G .722 can be enabled, providing better sound quality at the expense of higher bandwidth requirements.

During peak usage times when the system is under heavier load, codecs such as G .729 can be switched in to preserve bandwidth at the expense of a small amount of call clarity. It is also possible to assign specific codecs to individual phones so that those who require higher-quality audio have it and those who do not are less of a drain on the system, which results in a more balanced network overall.

SIP trunking connects the IPT system to an Internet telephony service provider who, in the case of VoIP calls, will route the call over the data network or over to the PSTN as necessary. This means that calls to endpoints outside of the company appear as if made on the traditional telephone network, while calls made to different branch offices around the world stay on the data network, avoiding long-distance toll charges.

Although SIP trunking is relatively new, trunking itself is not. Very large businesses with high amounts of telephone traffic have been using trunking for years with their old TDM systems, but doing so requires expensive, specialized equipment that more often than not is unreliable and uses a proprietary interface. If any one thing is responsible for the recent increase in the rate of IPT adoption and VoiP adoption in particular by small- and medium-sized businesses, it's the standardization of SIP trunking.

SIP Trunking Requirements

Although not recommended for any but the smallest businesses, even the slowest of modern broadband connections can result in acceptable VoIP performance if configured properly. The most commonly used encoding algorithms for VoIP applications produce equivalent call quality and call clarity to a traditional telephone connection while using approximately half of the bandwidth. This makes telecommuting from home a very real possibility for many employees, as the majority of home broadband connections are more than adequate to the task of maintaining a VoIP connection to the workplace.

If an employee is frequently working via a home connection, one thing to consider is that most home broadband connections are asynchronous, meaning that the download speed is significantly faster than the upload speed. This is perfectly acceptable for most home Internet users, as most users pull more information from the Internet then they push to the Internet. For VoIP applications, however, there is usually a relatively equal amount of data traveling to and from each user, and, while the download speeds of home broadband are sufficient, the upload speeds may not be. For frequent home users of VoIP applications, a synchronous connection (one with equal upload and download speeds) is recommended.

For businesses with more than two or three employees using IPT technology, the connection between their network and the ITSP should absolutely be synchronous as opposed to asynchronous. If the business uses more bandwidth-intensive IPT services, such as videoconferencing across multiple locations, it is important to perform load testing to ensure that the bandwidth available is adequate to the job.

Once a business is certain that its connection provides a suitable amount of bandwidth for its needs, it is important to negotiate a service-level agreement with the provider that guarantees an acceptable level of uptime and a minimum amount of bandwidth. This will help to ensure a functional connection to the ITSP at all times.

As mentioned, even a slow broadband connection can be adequate for light VoIP use--if that connection is configured properly. Even with faster connections, configuration is a key factor in maintaining acceptable levels of performance. All modern network equipment includes quality of service (QoS) functionality that allows an administrator to prioritize certain types of traffic on the network. This should be used to prioritize voice traffic over data traffic on a shared voice and data network so that, regardless of the load on the network, call quality does not suffer.

New Applications

SIP trunking is instrumental in enabling a variety of new, exciting, and empowering communications tools. Communication is often a significant determining factor in how successful a business is. By combining voice and data networks, businesses can not only improve the effectiveness and efficiency with which they communicate, but they also can enable new ways of communicating and, thereby, new ways of doing business.

With IPT-based systems, employees can access the company's communications infrastructure from anywhere at any time. This can help avoid a loss of productivity for employees who must travel or who would otherwise be temporarily unavailable.

Equally empowering, if not more so, is the one-number provisioning made possible by Internet protocol telephony. Because the routing is done primarily through software, the network does not differentiate by location but, rather, by user. This means that no matter where an employee connects to the network from, the network knows how to route that employee's calls. Each employee can always be reached by the same number, no matter where that employee is physically located. This is, in large part, what makes UC possible.

Provisioning new users and relocating current users is, like most things, much easier with IPT systems than with traditional TDM systems. With the old TDM system, each phone number in the network is tied to a specific line--a specific phone. Moving a number to a different location usually involved complicated reprogramming of physical switches. With IPT systems, because the routing is done primarily in software, adding a new user or moving a current user to a new location can be accomplished with a few clicks of a mouse.

In addition to providing a less expensive and more efficient way of performing traditional communications tasks, IPT allows for communications applications that simply are not possible on a traditional TDM network. Because an IPT network is centrally managed, and because it routes to the user, it is possible for each user of the network to have one mailbox for all of their messages regardless of medium. This allows employees to check their email, SMS text messages, voicemails, and video messages all in one place, and they can respond to those messages from within that same interface using the same or a different medium to answer--responding to a voicemail via text message, for example.

Presence management systems are also a welcome addition to most enterprises. They allow users to quickly and easily determine the location, availability, and preferred contact method for other users on the network, and all that information can be updated automatically and on-the-fly, ensuring that it is always accurate and that there are almost no times when an employee is unavailable.

One of the most interesting aspects of IPT is the possibility of developing APIs (Application Programming Interfaces) to tie already-existing programs and processes into the communications infrastructure. These APIs provide an almost infinite number of ways that an IPT system can be expanded to provide employees with new and powerful forms of collaboration and communication.

The one common thread that ties all of these applications together is that they would not be possible or at least would not be feasible without SIP trunking. By enabling the combination of voice and data networks, and by providing a method for that combined network to connect with the outside world, SIP trunking is driving a host of new innovations in the ways businesses can and do communicate.

Benefits of SIP Trunking

Although there are many benefits to IP telephony in general, SIP trunking can increase the impact of those benefits by a significant margin, and it can provide advantages that are unavailable without it.

By outsourcing PSTN connectivity to a third-party with multiple, geographically separated locations, enterprises can save significant amounts on long-distance charges. Additionally, one employee can have local numbers in multiple locations without incurring long-distance tolls, allowing customers or clients to have easier access.

Expanding the network as a business grows is also much less costly with an IPT system than with a TDM system, and the ability to maximize and customize utilization of existing bandwidth on-the-fly means expanding the network is less frequently necessary. With a TDM system, once an organization has fully used their currently provisioned lines, the addition of even one more line requires the purchase of additional T1 (23 lines) or E1 (30 lines) connection. This also necessitates the purchase of additional equipment to make use of the new lines. With an IPT network, all of the equipment necessary to operate the system is present from the beginning, so the network can scale more or less infinitely without additional, expensive equipment.

In some organizations, the consolidation of multiple services and billing cycles, as well as the reduced management overhead involved in maintaining one combined connection as opposed to two or more separate connections, can result in significant cost savings. The software-based nature of IPT networks removes the need for buying and maintaining expensive, proprietary equipment on premises. Initial investment for IPT equipment is significantly lower than that for TDM equipment, there is less equipment involved, and maintenance of that equipment is simpler and less expensive. The total cost of ownership for an IPT network is much lower than that for a TDM network.

Conclusion

As existing communications infrastructures age out, enterprises are faced with the decision to either double-down on their old technological base or to transition to a new framework that is more flexible, more cost-effective, and more functional. With the widespread availability of standards-compliant SIP trunking, modern IPT networks are now able to duplicate the functionality of traditional TDM networks and to do so more reliably and at a lower cost. They are also able to offer functionality that would be downright impossible with TDM-- such as UC and free or low-cost, long-distance calling.

Given such a stark contrast in cost and functionality, it's no wonder that there are more IP PBXs currently being used in the marketplace than there are TDM-based PBXs. The combination of lower initial investment, lower cost of operation, and increased functionality and productivity mean that enterprises who make the investment in SIP trunking can often recoup their initial investment in less than a year.