# Level (3)®

COMMUNICATIONS

Connecting and Protecting
the Networked World℠

# WHITE PAPER

SIP TRUNKING: HOW TO GET IT
AND WHERE YOU CAN EXPECT TO SAVE

## Executive Summary

Voice over IP (or VoIP) telephony is rapidly capturing a major portion of the enterprise voice telephone business. VoIP has a number of advantages over traditional voice services (TDM), including the ability to route voice traffic over the same circuits used for enterprise data communications — dynamically allocating available capacity and adding a wide range of new calling features and capabilities that would be expensive or impossible to implement on legacy telephone networks.

### SIP trunks offer a direct connection between a carrier's network and a customer's devices, for significant cost savings and business process improvements.

Session Initiation Protocol (or SIP) is the primary mechanism used for communication within VoIP devices and networks. Carriers such as Level 3 that provide SIP trunks are offering a direct connection between a customer's devices and their network, allowing significant cost savings and business process improvements by supporting the native protocol (SIP) that controls the customer's VoIP systems. SIP trunks offer benefits in three significant areas:

- Cost savings, which will be explored in this paper, arising from many factors including reduced telecommunications network charges and streamlined operations.

- Unified communications, where voice, video, email, text and other messaging technologies are combined to provide greater flexibility for users by enabling new ways to transfer information and manage connectivity.

- Business Continuity and Disaster Recovery, where the right physical configuration in conjunction with intelligence in the network can be leveraged to provide uninterrupted communications and alternative means to stay connected for employees in the event of system bottlenecks or failures.

Through the use of direct SIP trunk connections to carriers, enterprises can capture these benefits by

ensuring that the advanced features supported by VoIP systems are not blocked at the enterprise/carrier interface. Services like the Level 3® SIP Voice Complete offering allow the benefits of SIP technology to extend beyond the borders of the enterprise to enable a rich set of features for flexible, reliable communications across a range of technologies and media.

### Upgrading an existing voice network to take advantage of SIP trunking technology does not require wholesale replacement of existing equipment.

Contrary to popular belief, upgrading an existing voice network to take advantage of SIP trunking technology does not require wholesale replacement of existing equipment. In many cases, existing PBX equipment already supports or can be upgraded in-place to provide SIP capabilities. In other cases, adapters can be deployed to enable older equipment to interface with SIP-based networks.

Considering the potential for cost savings, the improved business practices, and the choices that exist for upgrading an existing voice network, SIP trunking is certainly worthy of a closer look by CIOs and telecommunications managers.

## What is SIP trunking?

In its most basic form, a SIP trunk is a connection between SIP user premises equipment and a provider's SIP network. Much like a PRI (Primary Rate Interface) trunk on an ISDN system or a T1 trunk on a traditional digital telephone system, it provides a path for both communication (voice calls) and signaling (the dialing instructions and call supervision/control) between an enterprise's equipment and a carrier's equipment. One big difference between SIP trunks and PRI connections is that with SIP, the total bandwidth can be shared, giving much more flexibility in adapting the available bandwidth for changing amounts of voice and data traffic. SIP trunks can be delivered over many different types of network connectivity, including MPLS services and other managed IP transport networks.  A SIP trunk

can even share a connection with carrier data service, providing further cost savings.

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VoIP systems that use SIP can provide a number of benefits to enterprises. For example, SIP supports many types of communications that can be delivered over a common network, including videoconferencing and HD Voice (which offers a dramatic improvement in sound quality through the use of higher data rates for each voice signal). Data services such as e-mail and text messaging can also be delivered over the same system, allowing for interesting new applications, such as converting voice mails into e-mail attachments for delivery to mobile devices.

As delivered by Level 3, all SIP trunking services include both primary and secondary trunk connections to support Business Continuity and Disaster Recovery (BCDR). For enterprises that require an enhanced degree of survivability, these trunks can optionally be connected to geographically diverse server clusters. Automatic reroute paths for calls provide further protection that can be triggered by network/device failures or congestion. In addition, enterprises are able to configure fail-over behaviors for entire trunks or individual telephone numbers, allowing each enterprise to setup BCDR policies and procedures to deal with a wide variety of threats.

Not every service provider and user defines the term "SIP trunking" in the same way. For the sake of clarity, Level 3 always uses this term to mean a direct IP connection between a SIP-enabled system on the customer premises and a true SIP-compliant carrier network. Nowhere in this connection path is conversion equipment deployed to convert customer signals from SIP into other formats such as ISDN, which would interfere with the ability to handle some types of SIP functionality.

The descriptions in the following sections of this white paper are focused on routine enterprise usage of SIP trunking for standard local and long-distance telephony, along with associated services. Dedicated SIP trunks that are employed for call center applications are beyond the scope of this document, because different factors must be taken into account to perform the usage and cost savings analyses.

## Requirements for SIP Trunking

Businesses that want to use SIP trunking must have SIP-enabled equipment. This can take many forms, ranging from a fully SIP-compliant IP telephone system to a legacy phone system that has been equipped with a SIP adapter. Several different configurations are commonly used:

### IP Telephone Systems

Many currently-installed IP telephone systems offer native SIP trunk interfaces as part of their standard configuration. This is often enabled by a simple software configuration change or upgrade; but in some cases additional hardware is required. Once the SIP trunk function is enabled in the equipment, connection can typically be made directly to the carrier's SIP trunk service.

### Newer PBXs

Recent-vintage PBXs (sold within the past five years or so) may be upgradeable to provide a SIP trunk interface. The exact method of doing this upgrade varies significantly from manufacturer to manufacturer; in some cases it might be a simple software upgrade, in others, a major hardware and software change out. In either case, the result is a direct connection from the enterprise's equipment to the SIP trunk provider's network.

### Older PBXs

Older PBXs can be adapted to work with a SIP trunk through the use of an Integrated Access Device (IAD) or similar border element. The IAD features a SIP trunk

interface on the carrier interface side, and either a PRI, T1 or Business Line interface on the user side. As shown in Figure 1 (on the following page), the IAD sits between the legacy PBX and the SIP trunk, converting the voice signals and signaling protocols into the correct format for each interface. IADs are available from several commercial suppliers, and provide a good way to leverage installed equipment while still capturing the benefits of SIP trunking.
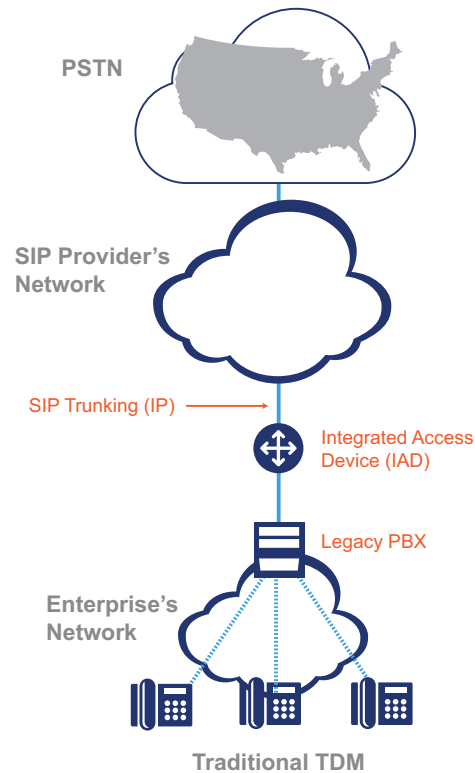
A common misconception about using SIP trunks is that enterprises must replace their existing hardware and telephone handsets with a SIP-capable PBX and SIP phones for every user. But in fact, many installations use preexisting handsets and wiring for the initial launch. Upgrades can be done in the future if the enterprise decides to deploy SIP-enabled phones that offer advanced capabilities.

## Session Border Controller

A Session Border Controller (or SBC) is an optional piece of equipment used on many SIP trunk implementations. Carriers like Level 3 provide them at the carrier-side interface for SIP trunks. Enterprises can also add them at their own premises, but these SBCs are not required. As shown in Figure 2, an SBC can be located at just one end of a SIP trunk or both ends, if desired.
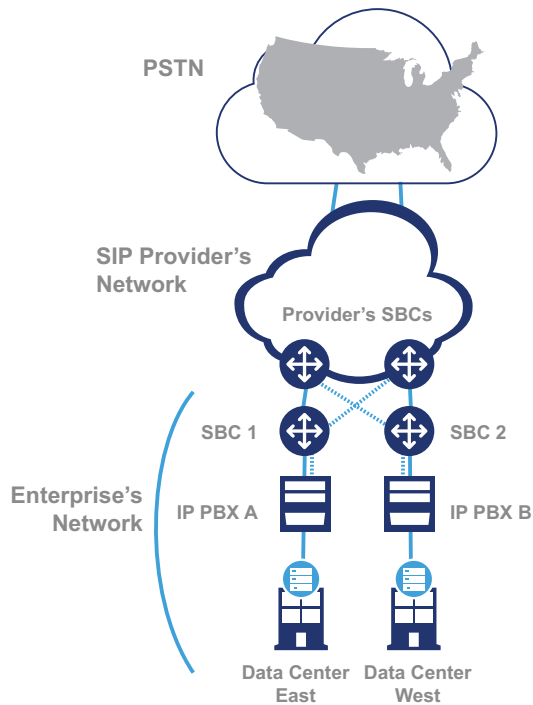
An SBC can serve multiple functions; two of the most important are increasing security and improving the ability for voice communication systems to recover from a disaster. Security can be improved in a variety of ways, from encrypting the calls (both the call setup information and the voice and data signals themselves) to providing a secure mechanism to process calls across a corporate firewall or Network Address Translation (NAT) device.

**Figure 1: Sample IAD Deployment**



PSTN

SIP Provider's Network

SIP Trunking (IP) →

Integrated Access Device (IAD)

Legacy PBX

Enterprise's Network

Traditional TDM

Disaster recovery can be enhanced by configuring the SBC to provide backup connections in the unlikely event that the primary SIP trunk connection is unreachable. (See Figure 2.)

**Figure 2: SBC Connection Example**



PSTN

SIP Provider's Network

Provider's SBCs

SBC 1

SBC 2

Enterprise's Network

IP PBX A

IP PBX B

Data Center East

Data Center West

## WAN Network Requirements

Another key element required for a SIP trunk deployment is a suitable corporate WAN network, which can drive success or failure for the entire SIP trunking project. First and foremost, the corporate WAN must have enough bandwidth to handle the existing data traffic, with enough available capacity to handle the additional load of the SIP traffic. It is also highly recommended that the installed WAN technology supports preferential treatment for SIP voice traffic, using differentiated quality of service (QoS). This keeps delay under control and helps shield SIP traffic from the bottlenecks that might be experienced by other forms of data traffic, which could cause a drop in call quality that would be noticed by users. And, since voice communications are crucial to many businesses, some type of redundancy or emergency backup system in the WAN network is recommended.

## Cost Savings

Enterprises typically enjoy substantial cost savings from installing SIP trunks. These savings are generated by three main mechanisms: converging voice and data facilities, pooling concurrent call paths, and inter-office calling via private WAN backbone.

> The cost savings that result from pooling CCPs into a central connection are due primarily to the fact that CCPs can be managed more efficiently, not from any difference in the carrier price for CCPs vis-à-vis ISDN or T-1 ports.

**Converging Voice and Data Facilities**

Since most enterprises already have a WAN operating among locations, using SIP telephony allows voice traffic to be transported across the same network as corporate data. By doing this, enterprises can reduce or eliminate the costs of leasing and managing a second network used exclusively for telephony by disconnecting the facilities that provide access to local carriers in each location. These savings can be substantial, when all the costs are considered. Most noticeable are the fees paid to carriers for leasing network access trunks. Other costs include the internal staff or outside contractors who are required to maintain the systems, including monitoring healthy systems and repairing failed equipment or connections. From both a labor and a management standpoint, it is more efficient to manage a single high-bandwidth network than it is to manage separate voice and data networks.

Sharing a common WAN for both telephony and data traffic also makes sense when typical traffic patterns are considered. For example, data traffic on many networks can intensify when nightly backups are performed or large files need to be moved between locations for archiving or for business continuity enhancement. If these data transfers can be scheduled when call volumes are typically low, or if they can be

set to a lower priority than VoIP traffic, then a common WAN can be shared by both applications with minimal need to add capacity to the network. Furthermore, since data traffic tends to be tolerant of small amounts of network latency, mechanisms to prioritize voice over data (such as QoS) can also diminish the need for additional WAN bandwidth. If additional WAN bandwidth must be added, the amounts needed for SIP traffic are not excessive; a 1 Megabit increase in network bandwidth is enough to simultaneously carry 10 high-quality voice signals without the use of any compression.

The actual cost savings that can be achieved by combining voice and data networks will be different for each enterprise, but some general guidelines apply. If most locations are already served by a WAN connection that has enough capacity to support VoIP traffic, then savings can be achieved by eliminating the costs of the local telco access circuits. Consolidating network operations and maintenance staff could generate additional savings, or lead to increases in productivity by freeing-up resources for other projects. But recognize that SIP systems will still require support for regular maintenance, such as handset adds, moves and changes.

**Pooling Concurrent Call Paths**

Carrier contracts for SIP trunks will normally specify a maximum number of Concurrent Call Paths (CCPs) that will be supported. One CCP is occupied during each active SIP call and then released back into the pool of available CCPs when the call ends. In this sense, they are similar to the 23 voice channels delivered on a standard PRI circuit — the total number of CCPs indicates the total number of simultaneous calls that are permitted on the SIP trunk.

In another sense, CCPs are very different from PRI or T1 ports, which represent channels on a physical circuit that is constructed to the enterprise premises. As such, to increase or decrease the number of voice ports available to a facility, an enterprise must add or remove an entire PRI or T1 circuit, thereby changing the port count in steps of 23. In contrast, the number of CCPs on a SIP trunk can be set to any desired value

— it can be configured for 5 or 17 or 27 simultaneous calls, or whatever number the specific application demands. When multiple sites are connected with SIP telephony over a corporate WAN, the CCPs from all the sites can be pooled into a single SIP trunk connection to the carrier. This standard feature from Level 3 (which is an optional extra from a few other carriers and not available from most others) is particularly beneficial for smaller company sites. Instead of requiring a PRI connection to a remote office that may-have only a dozen employees, and therefore a need to support perhaps three to seven concurrent calls, these callers can be grouped together with all of an enterprise's other sites, and share a common pool of CCPs. As queuing theory and telephone traffic studies show, fewer connections are required to provide a high quality of service to a single group of 100 telephones than the number of connections that would be required to provide that same quality of service to 10 groups of 10 telephones each.

The cost savings that result from pooling CCPs into a central connection are due primarily to the fact that CCPs can be managed more efficiently, not from any difference in the carrier price for CCPs vis-à-vis ISDN or T1 ports. In fact, even though individual CCPs frequently carry a somewhat higher price than the corresponding single channel within an ISDN PRI, the ability to deploy CCPs only as needed across the enterprise will save money as compared to buying multiple "lumps" of ISDN ports.

Additional voice traffic efficiencies can be gained when an enterprise spans multiple time zones. Employees in each time zone have natural variations in the amount of voice traffic they generate based on factors like the time they arrive at work, depart to go home, and take breaks for lunch. The effects of these offset cycles can be shown to have the net effect of reducing the aggregate peak level of voice traffic routed to the carrier, thereby reducing the number of CCPs that need to be leased for the overall enterprise.

Savings generated by pooling CCPs for an organization are generated through two mechanisms. The greatest savings are typically due to eliminating the costs of

unused trunk ports on PRI or T-1 interfaces to branch offices, where the last few channels of the 23 available connections may never be used even though they are included in the monthly charge. Additional savings are derived from the efficiencies gained by having one common pool of CCPs shared across the enterprise in place of multiple smaller pools that would otherwise be required at each location.

**Inter-office calling via private wan backbone**

When different enterprise locations are connected using SIP telephony over a WAN, cost savings are generated in two major ways. The first source of savings is due to the removal of carrier trunk connections at each remote site, which can substantially reduce the monthly costs paid for telephony connections.

The second source of savings is the elimination of carrier networks for calls within the enterprise. Costs from inter-office calls within the enterprise accrue in three places: the cost of the carrier trunk interface at the source of the call, the cost of the carrier trunk interface at the destination of the call, and any usage fees charged by the carrier(s) to transmit the call between the two locations. The total amount of these cost savings can be difficult to predict with any accuracy, because there are so many variables at play. The number of calls made between locations and the duration of these intra-enterprise calls varies depending on each team's role within the company (for example, HR could make more internal calls than sales). Call volumes also vary by industry; a lumber products company could have lower call volumes per-employee than a brokerage firm. Additional variation in call volumes can be driven by the company's organizational principles, driven by the amount of autonomy given to the remote operations versus the amount of centralized administrative control.

Unlike services from most other carriers, SIP Voice Complete services from Level 3 offer free calling among enterprise locations, even when the calls are routed over the Level 3 Network instead of the enterprise WAN. This allows customers to avoid the headache of implementing complicated call routing logic in their

equipment solely to prevent intra-company calls from traversing the carrier's network.

## Calculating the Cost Savings

To illustrate the cost savings that are possible through the use of SIP trunking, consider two examples, one based on a large, multi-state enterprise and one based on a smaller, regional enterprise.

Enterprise A has three main locations with 1,000 employees across them, and 100 branch locations with another 1,500 employees. 25 percent of the enterprise's employees are located in the Pacific time zone, 10 percent in the Mountain, 25 percent in the Central, and 40 percent in the Eastern time zone. Using industry average values for the costs of local access PRI circuits and ports, costs of PRI ports, per-minute rate for long distance calls, and intra-company calling rates, SIP trunking would generate monthly cost savings of $39,000 in telecom network expenses, a savings of 65 percent over an ISDN-based solution. These savings come from several sources:

- $16,100 (41 percent) by eliminating the T-1 access lines in most facilities
- $15,000 (39 percent) by eliminating unused ISDN ports (sharing a smaller number of CCPs)
- $4,300 (11 percent) from getting better long-distance calling rates
- $2,500 (6 percent) by taking advantage of the different time-zone peaks
- $1,100 (3 percent) by eliminating long-distance charges among locations
- Enterprise B has one main location with 100 employees and five locations with another 75 employees. All of the offices are in a single time zone. Again using industry standard values for the per-circuit and per-call rates, this enterprise could reduce monthly phone charges by 50 percent, for a total savings of $1,936 per month, broken down as follows:
- $708 (36 percent) by eliminating unused ISDN ports (reduced CCPs)

- $632 (34 percent) by eliminating the T-1 access lines in most facilities
- $476 (24 percent) from getting better long-distance calling rates
- $120 (6 percent) by eliminating long-distance charges among locations

Of course, the above numbers are simple estimates, and by necessity leave out some details that could have in impact on the overall amount of savings. In particular, the results would be altered if prices currently paid by an enterprise are significantly different from industry averages or if actual usage patterns differ.

Ideal Candidates for SIP Trunking

The business benefits and cost savings generated by upgrading to SIP trunking will vary among enterprises. How much a given enterprise will benefit from SIP trunking can be estimated through analysis of existing communications patterns within the enterprise. In any case, it is useful to describe some of the key attributes that would make up an ideal candidate enterprise for SIP trunking.

The ideal SIP candidate enterprise—one that would experience the largest cost savings—would be a medium to large organization with multiple facilities spread across different time zones. These facilities could range in size from small to large, with the smaller facilities typically generating the largest cost savings by eliminating excess capacity in the PRI or T-1 trunks used for connection to the local telephone carrier.

The ideal enterprise would have relatively new telephone equipment that could have SIP capabilities simply switched on or added via upgrades. Or, an enterprise that was planning to purchase new equipment could easily select SIP-capable equipment from a wide variety of manufacturers. Any enterprise that has underutilized or superfluous infrastructure, either in the form of excess capacity in local access trunks or in the form of available bandwidth in the WAN, would be well-suited for upgrading to SIP trunking.

From a business strategy standpoint, enterprises that have a strong motivation for unified communications would find SIP technology beneficial. This might include organizations with a highly mobile salesforce or a geographically dispersed workforce that would benefit from desktop and mobile videoconferencing. Enterprises that put a strong emphasis on business continuity or those that must be able to recover quickly from a natural or man-made disaster are also ideal candidates for SIP trunking.

## Conclusion

The market for SIP trunking is rapidly growing, and is set to explode in the year ahead. This is due to a number of factors, including the widespread availability of PBX and other equipment that supports SIP technology, growth in carrier offerings and, most importantly, the substantial cost savings. As more enterprises seek to adopt unified communications technologies to improve productivity, and as they are required by regulators and stakeholders to deploy more robust systems for business continuity and disaster recovery, the incentives to transition to SIP technologies will only increase.

SIP trunking is the best way to expand the benefits of SIP beyond the borders of the enterprise, and to capture all cost savings and performance benefits of this expanding technology.

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