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Building the Business Case for SIP Trunking

Whitepaper

Moving to VoIP and SIP trunking has the potential to reduce Telecommunication costs by 50 percent or more. Exactly how much will an organization save by moving to SIP trunks? This whitepaper reviews the opportunity and benefits of moving to SIP trunking for an enterprise based in the United States.

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Telecom costs represent 10 percent of the typical IT fixed expense budget of a Fortune 1000 company. Moving to VoIP and SIP trunking has the potential to reduce this cost by 20-50 percent. Exactly how much will an organization save by moving to SIP trunks? This whitepaper reviews the cost savings opportunities and benefits of moving to SIP trunking within the continental U.S.

Overview

Building a business case for SIP trunking is a six-step process:

1. Understand the business requirements for voice connectivity in terms of availability, capacity, quality, security, and features.
2. Gather existing voice communication costs by reviewing existing infrastructure and rates.
3. Issue RFPs to estimate the costs for migration to SIP trunks.
4. Determine the appropriate architectural model – Centralized vs. Distributed
5. Put together the business case with a cost-benefit analysis
6. Highlight other strategic reasons for implementing SIP trunks besides cost savings.

Implementing SIP trunking is usually the third and final phase in converting an organization's voice communication system to total Voice over Internet Protocol (VoIP) and using the Session Initiation Protocol (SIP) for communication signaling. The first phase in moving an organization to VoIP is moving to IP-Telephony by putting in IP-PBX and IP phones. Originally this was done using H.323 or a proprietary protocol, but SIP has evolved into the de facto industry standard. The second phase is using VoIP/SIP to interconnect auxiliary voice communication systems, such as voice messaging, call recording, and IVRs.

The focus of this whitepaper is the business case for SIP trunking. The moves to IP Telephony and VoIP adjuncts have their own business cases.

Introduction

Currently most organizations use traditional digital phone trunks when placing voice calls to callers who are at another location. In the U.S., phone trunks fall into three categories:

- **Local trunks** – Used for local phone calls within a LATA {1} that comes from a LEC {2}. Direct Inward Dial (DID) trunks are local trunks that the telephone company assigns standard phone numbers to. Local trunks are also used for outbound toll free dialing, which is most commonly used to access a 3rd party conferencing bridge.
- **Long distance trunks** – Used for calls outside of a LATA. Calls can be intra-state (within a state), inter-state (between different U.S. states), or International that come through an IXC {3}.
- **Toll-free trunks** – Used for callers to call an organization without incurring long distance charges; they start with 800, 888, 877, or 866.

These digital phone trunks typically ride across a T1 and use an ISDN {4} PRI D channel for their signaling. A T1/PRI will carry 23 voice channels and one D channel. An office will normally have two T1/PRIs for every 100 full-time people who work in it. Figure 1 shows an overview of traditional digital trunk connectivity.

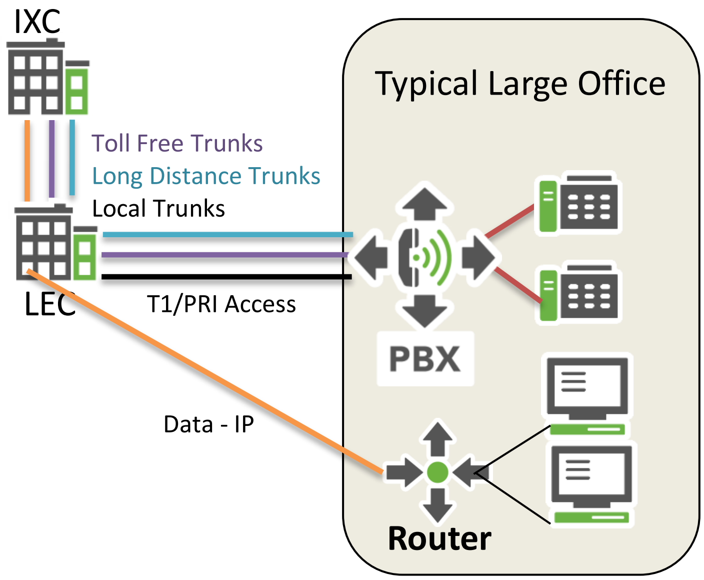

Figure 1 – Traditional Digital Trunking

As organizations moved to IP-Telephony, most kept their T1/PRI digital trunks to the carriers at each office but converted them to VoIP within a gateway. They also centralized their call processing and started running VoIP over the WAN {5} for voice messages and intra-company calling (illustrated in Figure 2).

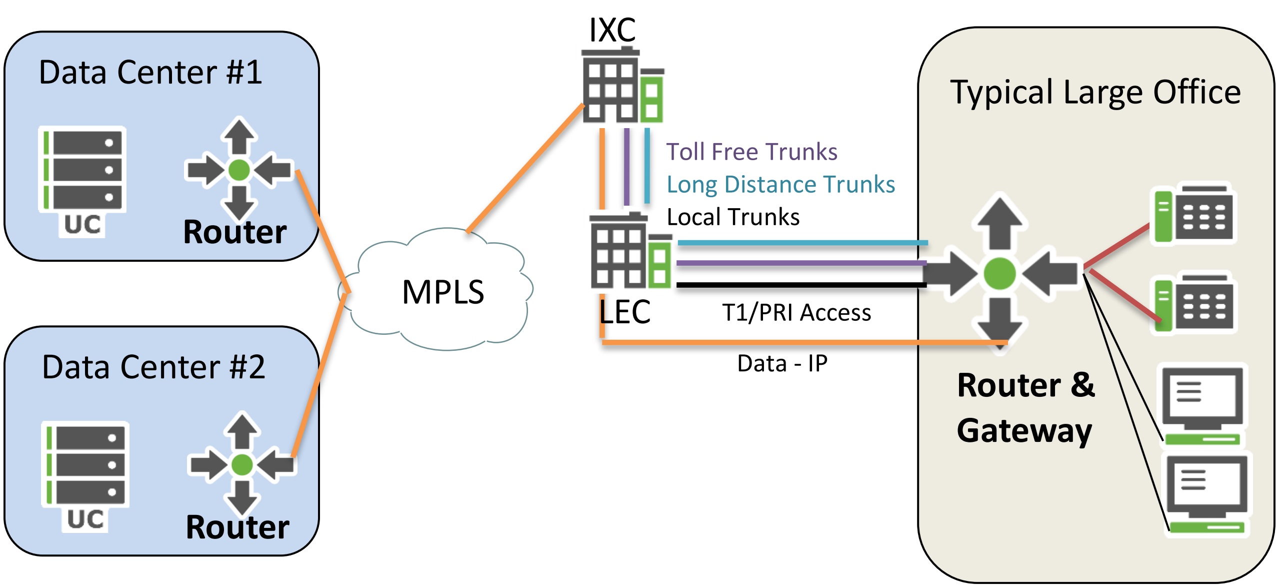

Figure 2 – IP Telephony With Digital Trunks

The final step in moving all voice communication to VoIP is implementing SIP trunking. SIP trunking is a service offered by Internet Telephony Service Providers (ITSP) that connects a company's telephony system to the existing telephone system infrastructure, PSTN {6} via the Internet using the SIP VoIP standard.

The most common architecture is to consolidate all trunks into the data centers to optimize costs and minimize the amount of technology. SIP trunks are provided by traditional carriers like AT&T, Verizon, XO, CenturyLink, or a long list of other service providers. A key component in moving to SIP trunking is the Enterprise Session Border Controller (E-SBC). The E-SBC provides SIP interoperability, security, management, monitoring, support, and reporting for SIP trunks. This is illustrated in Figure 3.

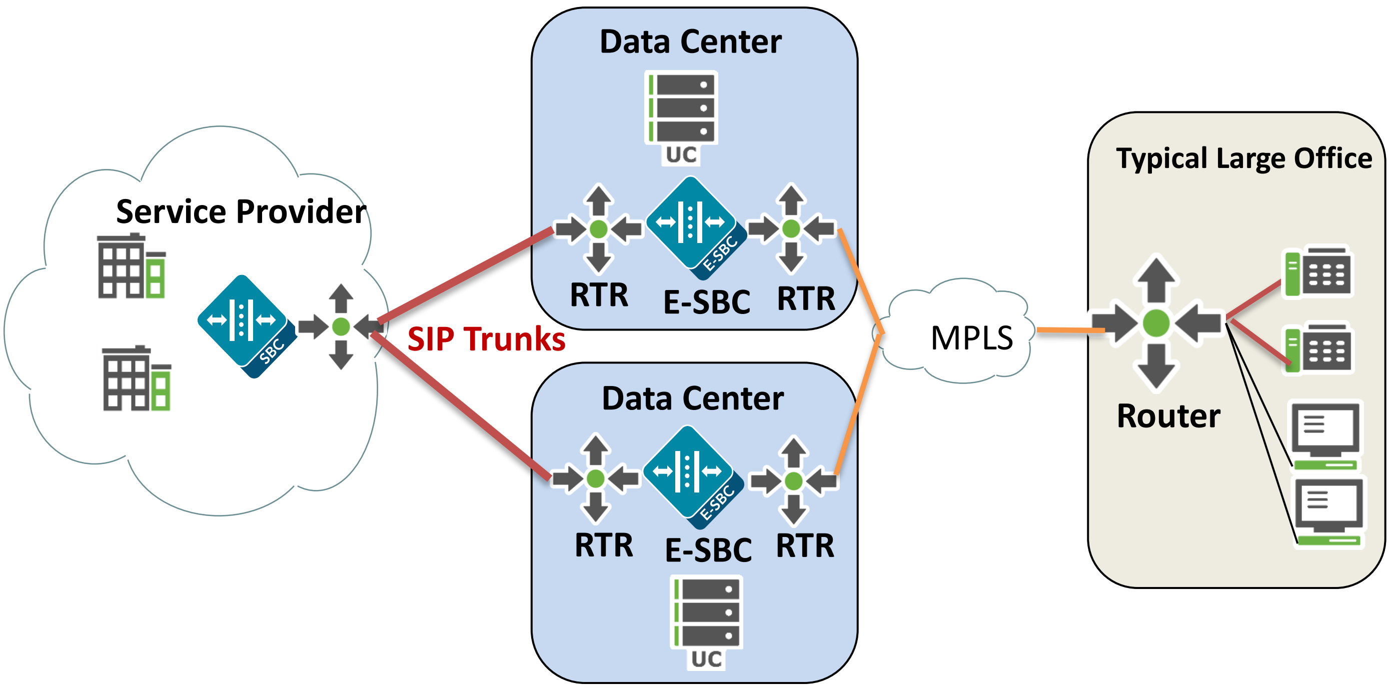

Figure 3 – IP Telephony With Centralized SIP Trunks

Building the Business Case - Step 1: Business Requirements

Every business has different expectations and requirements for their voice communication systems, including voice trunking. These requirements will have a large impact on the design and cost of moving to SIP trunks. The first step in the six-step process for building the business case for SIP trunking is understanding the business requirements for voice trunking in terms of availability, capacity, quality, security, and features.

Availability

Availability is defined in terms of how reliable the system is, or how often the system is not available over a period of time. What are the business' expectations with regards to how reliable voice connectivity is? Typically, expectations fall into one of three categories:

1. **Ultra-Reliable** – 99.999 percent - Voice connectivity is always available with the average annual down time of less than six minutes per year. Voice connectivity

should work even if there is a loss of power, if a telecom connection into the building is compromised, or if a piece of telecom equipment fails.
2. **Reliable** – 99.99 percent - Less than one hour of unplanned downtime per year. If the average time to repair a problem is four hours at a site, then only one out of four sites can have an outage per year.
3. **Standard** – 99.9 percent - Less than nine hours of unplanned downtime per year. If the average time to repair a problem is four hours, a site can have two outages per year.
4. **Best Effort** – 99.5 percent – 40hrs of unplanned downtime per year.

Voice connectivity used to be mission critical to most businesses, but this is changing. When all else fails, the phone system and its telecom connectivity should work to call for help and to keep the business operational. As cellular messaging technologies like email, chat, and SMS have taken off, voice applications now are on par with other critical business applications.

Typically, voice connectivity needs to be as reliable as a company's WAN and critical business applications. Most businesses rely on centralized data centers to run their applications. If the WAN is down, most business functions will not operate and employees cannot effectively handle most calls. Thus, voice applications are considered to be just as critical as other business applications, but not more so as they use to be.

Reliable to 99.99 percent is the typical requirement these days. The additional cost to provide ultra-reliable connectivity usually does not have a corresponding business benefit. This is especially true when voice connectivity can be routed to another site, so that the site can be down, but customer and other calls can still be answered by the enterprise. Standard connectivity is usually not good enough for an enterprise since it is too disruptive to the business. The size of the site is also a factor; larger sites have higher availability requirements.

Disaster Recovery (DR) is also a requirement that needs to be included in the above calculations. Long before a site actually fails, a back-up site should be determined; the back-up site(s) should have the capacity to handle the additional call volume.

Capacity

Capacity is defined as how many voice trunks are required. How many people will be on the phone at the same time, and what level of blocking (fast busy signal) is acceptable? Capacity is typically measured by percent of calls blocked during the busiest hour in a given week. P.01 means that one percent of call attempts will be blocked during the busiest hour in a given week. Call volume can also be seasonal with certain vertical markets getting twice the call volume during one month out of a year. For retail, this is around the Christmas holidays, for health insurance this is at the beginning of the year for new enrollment, and for tax preparation companies, it is around April 15.

Typically for most businesses, the busiest periods are at 10 a.m. and again at 2 p.m. During the few minutes surrounding these times, half the people in an office can be on the phone. Conference or person-to-person calls that are scheduled to go from 9 – 10 a.m., for example, may run over a few minutes while calls from 10 – 11 a.m. are just getting started. Over the last 20 years, the number of trunks into an office with the same number of employees has gone up due to more collaboration with people outside of the office.

Capacity requirements are broken down into three categories:

1. **Ultra-high** – P.001 – Less than one out of 1,000 calls are blocked during the busy hour for the year. Typically, this requires a 5:4 person-to-trunk ratio; thus, a 100-person office needs 80 voice trunks.
2. **High** – P.01 – One out of 100 calls are blocked during any given week, and if blocking does occur, a trunk should be available in less than a minute. Typically, this requires a 2:1 person-to-trunk ratio, depending on the business; thus, a 100-person office needs 50 voice trunks.
3. **Standard** – P.02 – One out of 50 calls are blocked during any given week. Typically this requires a 3:1 person-to-trunk ratio; thus, a 100-person office needs 33 voice trunks.
4. **Best Effort –** P.05 – One out of 20 calls are blocked during any given week. Typically this requires a 4:1 person to trunk ratio; thus a 100 person office needs 25 voice trunks.

Capacity requirements are also determined by type of phone trunks. In-bound toll-free trunks typically have the highest requirements, and outbound local calls have the lowest requirements. The default requirement is P.01 for in-bound calls (toll-free and local DID), and P.02 for outbound calls. A business wants to be available for in-bound customer calls, but maintaining extra trunks for occasional large conference calls is usually not worth it.

Again, with the proliferation of cell phones as an option, the need to have extra voice trunks to handle peak capacity may no longer be warranted. In many businesses about one-third of phone lines are used less than one hour per month.

Quality

Voice quality is subjective to the caller. Some people have grown accustomed to repeating themselves and working hard to listen to what a caller is saying. Others, who make their living by talking to people, require that voice quality be as good as if the communication were face-to-face.

One measure of voice quality is the Mean Opinion Score (MOS), which provides a numerical indication of perceived quality. MOS is expressed as a number from one to five, where one is the lowest perceived audio quality and five is the highest.

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible
3	Fair	Slightly Annoying
2	Poor	Annoying
1	Bad	Very Annoying

Table 1 – MOS Scores Measuring Voice Quality

Voice codecs and compression is the process of converting a human audio conversation to a digital bit stream and minimizing the data bit stream to save bandwidth and associated costs. There is usually a linear correlation between increasing voice compression and lower voice quality. What is determined as “good enough” is left up to an organization.

There are three general classes of voice compression:

1. **Wideband Audio** – 50 to 7,000 hertz. – High quality audio with an average MOS score of 4.8. Codecs such as G.722 use from 48 – 64Kbps of bandwidth.
2. **Standard Audio** – 300 to 4,000 hertz – Standard telephony quality audio with an average MOS score of 4.3. Codec is G.711 and is the industry standard, using 64Kbps of bandwidth.
3. **Compressed Audio** – 300 to 4,000 hertz – Compressed to 8Kbps of bandwidth with a MOS score of 3.9. The traditional standard codec is G.729.

When the first generation of VoIP systems came to market in the late 1990s, voice quality took a step back. The second, most recent generation of VoIP systems is capable of higher voice quality than traditional TDM systems. For contact center agents, sales people, and others who spend significant amounts of time on the phone every day, high voice quality is critical. Poor voice quality leads to greater mental fatigue, lower productivity, and a poor communication experience.

Because the industry is still in transition from TDM to VoIP, G.711 is the default standard audio that most organizations use with G.729 used over their WAN. One of the great benefits is that SIP allows the end points to negotiate the call quality.

Most organizations run G.729 with their VoIP systems trying to save every penny of WAN bandwidth costs. On the other hand, most external voice solutions, such as Skype, and new players to the game, such as Microsoft, offer proprietary wide-band codes that are adaptive to the network that they are on. These codecs use less bandwidth than a standard G.711 call, but sound better. Many users report that their home Skype call to their child in college sounds better than what they utilize at the office. As bandwidth costs continue to decline, organizations have the opportunity to improve voice quality.

Security

Different organizations have different security requirements for their voice communication. When voice was on its own digital infrastructure and not connected to the corporate data network, the risk of someone hacking the voice system was lower. Yes, hackers try to crack voice systems to get free long distance and eavesdrop for credit card information; however, the value of hacking into a voice system is a lot less than hacking into a web server, which typically stores numerous credit card numbers and access to products. When voice moves to IP, it is subject to all the security required for the organization’s IT applications. Security falls into the following three levels:

1. **Highly Secure** – Encrypt all media and associated SIP signaling and information
2. **Secure** – Encrypt all SIP signaling and information, but not the media
3. **Standard** – No encryption

Most organizations default to secure communication where all information about the caller is encrypted, but the media is not encrypted. SIP-TLS is the most common form of SIP signaling and information encryption with SIP trunking.

An Enterprise Session Border Controller (E-SBC) plays a critical role in providing SIP trunking security. The E-SBC acts as an SIP back-to-back user agent, or B2BUA {7}, and sits between an organization and any third party that it sends calls to. The E-SBC provides a demarcation point with the following security features:

- **Topology Hiding & Privacy** – Hides the IP network topology of an organization to prevent directed attacks and preserves confidentiality; masks user information for privacy and confidentiality; provides security isolation between networks; and monitors the media for lawful intercept and/or fraud prevention.
- **Access Control** – Permits only specific and known networks, devices, and applications to communicate with an organization's voice systems.
- **Denial of Service (DoS) Protection** – Protects an organization from malicious attacks and non-malicious overloads so the voice systems do not get overloaded.
- **Encryption** – Encrypting the SIP signaling and the media as required.
- **Virus & Worm Protection** – Protects SIP messages from malicious attachments and prevents malformed messages that may contain a virus, worm, or Trojan.
- **Logging, Monitoring & Reporting** – Monitors and reports on alarms for attacks and overloads, provides an audit trail in response to an attack or fraud investigation, and logs all configuration changes by support personnel.

An E-SBC is used to secure inter-company voice communication in the following cases:

1) **SIP trunking border** — connections to service provider SIP networks linking the enterprise to the outside world of PSTN and IP endpoints
2) **Internet border** — connections to small offices, users working from home and mobile employees over the public Internet
3) **Hosted services interconnect border** — private connections to service providers or Application Service Providers (ASP) that offer hosted IP-based audio and videoconferencing services, IP contact center services, IP Centrex to augment premise-based systems for certain sites, business groups or divisions and VoIP-enabled business applications such as salesforce.com.

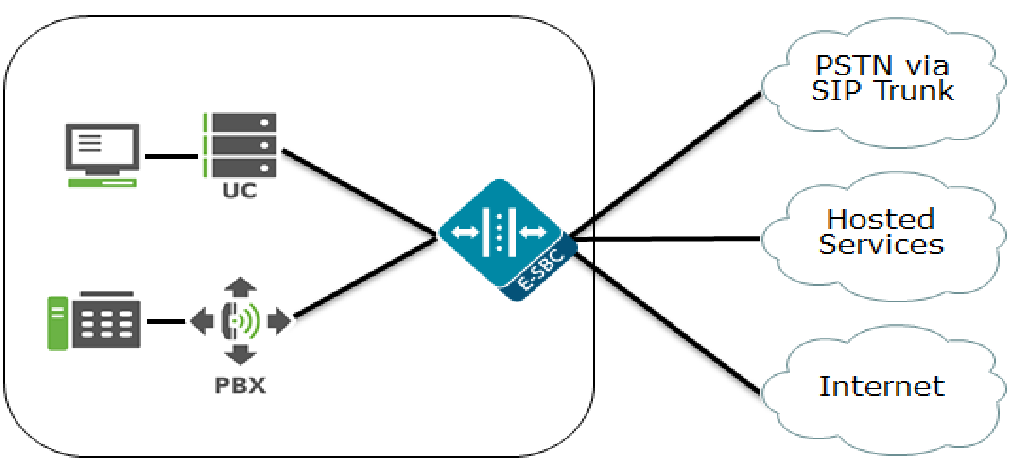

Figure 4 – Common Enterprise Session Border Control Points

VoIP/SIP has many unique security requirements that the traditional corporate firewall cannot provide, thus the reason for adding Enterprise Session Border Controllers as part of an SIP trunking solution.

Features

There are three categories of voice trunks as discussed in the introduction section: toll-free, long distance, and local. Each category of voice trunks has its own set of features that are broken down into standard characteristics for that type of trunk, as well as advanced features that can be customized to meet specific business requirements.

Basic features for each of the three trunking categories are described below. Advanced features cost extra, so understanding a business' desire for them is important. The standards and testing for basic feature sets have been worked out within the industry, while some of the advanced feature sets are still not offered by all SIP trunking Service Providers, or are done on a case-by-case basis. The features of each category of trunks are:

1. **Toll-free** – In-bound 800 trunks for customers, partners, and others to be able to call a business without getting charged for long distance.
    a. Basic Features – Toll-free numbers, DNIS, Trunk Groups, Call Transferring, and Trunk Allocation
    b. Advanced Features – Caller Entered Digits, UUI info, Automatic Call Rerouting
2. **Long Distance** – Calls between LATA's, states, or countries.
    a. Basic Features – Managed dial plans for on-net and off-net calling, private routing, international calling, operator assistance
    b. Advanced – ANI manipulation, International call authorization
3. **Local Trunks** – Calls within a LATA or given metro area and out bound toll-free
    a. Basic Features – DID numbers, Directory Assistance (411), Directory Listing, operator services, Emergency (911), outbound 8XX access
    b. Advanced – Other n11 services such as 711, 311

Historically, toll-free and long distance trunks come from an Inter-exchange carrier (IXC) such as AT&T and Verizon, while local trunks come from the Local Exchange Carrier (LEC). With all the mergers over the last decade, IXCs and LECs have become the same company in a lot of areas, but the tariffs and provisioning follows the traditional model. Most companies will provision each of the above trunks separately, each having their own capacity.

One important note is that outbound toll-free calls must go across a local trunk. Thus, if a business outsources its conference calling to a third party, it must add additional local trunks to support it. This can triple the number of local phone trunks that are required into an office site.

Summary

The criticality of voice trunks in terms of availability, capacity, quality, security, and features is different for every business. As cellular and other forms of communication become more predominant, it is important to delineate past versus future business requirements for voice services. The level of criticality plays a large role in current and future voice trunking costs.

Business Requirements	Ultra	High	Standard
Reliability	99.999%	99.99%	99.90%
Capacity	P.001	P.01	P.02
Quality	Wideband	Standard	Compressed
Security	Encrypt Media	Encrypt Info	No Encryption
Features	10	5	1

Table 2 - Business Requirements for Voice Trunks

For the sample business case and all examples within this whitepaper, the business requirements will be assumed as “high,” as noted in blue highlight above.

Building the Business Case - Step 2: Existing Voice Trunking Costs

To gather the existing telecommunication trunking costs, one must do an inventory of existing voice trunks at each office, get a copy of carrier contracts detailing current telecom rates, and review bills from the various LECs and IXCs. Using this information, telecom costs can be broken into five different categories:

1. **Access –** This is the transport from the business location to the phone company, sometimes referred to as the local loop. In most cases, this is a T1/PRI. Typically T1 access runs $200/month anywhere within the U.S. and an additional charge of $100/month for the D channel.
2. **Trunk –** Each voice channel is referred to as a trunk that can carry one voice call. Voice trunks are broken into three different categories: toll-free, long distance, and local. Typically a local trunk costs $35/month. Long distance and toll-free trunks are usually “free” since the cost is recouped in the per-minute usage charge.
3. **Usage –** The per-minute charge for the use of each trunk. Usage varies based on the call type and location of the calling parties, and is broken down into:
    - **a. Local –** Calls made within a LATA. These calls can either be free or have a per-minute rate based on how far away the caller is, based on bands (A, B, or C). Typically the cost is $0.01/minute
    - **b. Long Distance**
        - i. **Intra-State –** Calls outside of a LATA but within the same state. Typically these calls are $0.05/minute. This also applies to toll-free calls where a caller dials a toll-free number within the same state as the person who answers it.
        - ii. **Inter-State –** Calls between states within the U.S. Typical cost is $0.02/minute.
        - iii. **On-net calls** (intra-company calls that may be part of a private dial plan) and off-net calls (inter-company calls) are treated the same, even though on-net calls are typically 10 to 20 percent less than off-net calls.
        - iv. **International 1 –** Calls between the U.S. and another country.
        - v. **International 2 –** Call between two foreign countries.

   c. **Toll-free** – Calls made to a toll-free number registered by a business. Typical cost is $0.02/minute. Most external conference bridges charge both the toll free per minute charge of $0.02/minute along with a conference bridge charge of $0.03/minute, for each participant, for the duration of the call.

4. **Features** – The cost for advanced services above basic features that are included in the base trunk or usage charge. The most common advanced features charges include:
   a. Reserved Numbers – toll-free and DID
   b. Operator – Someone to assist with a call or 411 information services
   c. Directory Listing – Advertising of a number
   d. Transfer Costs – $0.25 per call transferred or $0.025 per call if all calls in a trunk group have this feature, whether the call is transferred or not
5. **Support** – One person for every 2000 people – The cost for staff to move, add, change, or delete (MACD) a trunk, along with finance staff who check the bill and who also may allocate costs to specific business groups. Assume a fully-loaded support person costs $8,000/month.
6. **Taxes & Fees** – Local, state, and national taxes, along with fees levied by various government agencies. The Universal Service Fund (USF) is an example of a Federal tax of 15.7% {9}. Taxes and fees can add up to 20-25% percent of the total phone bill.

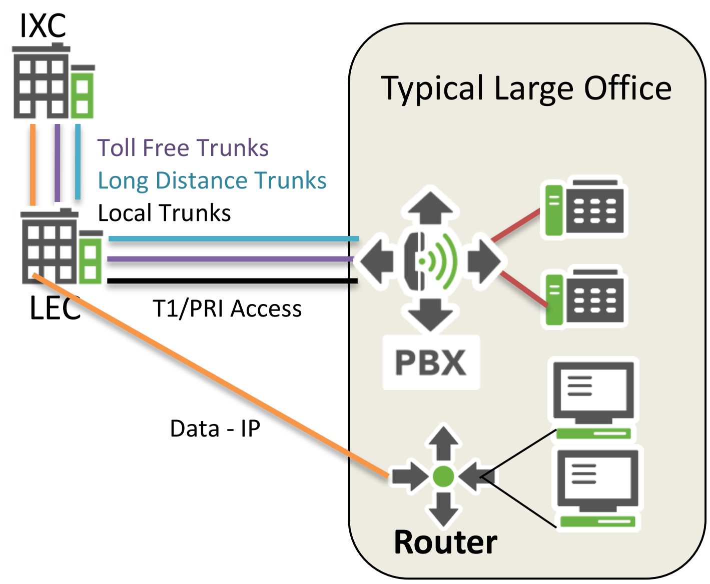

Figure 5 – Typical Business Voice Trunking Connectivity

Figure 5 illustrates the basic components of a typical business office as discussed above. Some of the inefficiencies of traditional trunking architecture are:

1. **T1 Access** – Voice circuits come in bundles of 24, so if an organization needs 60 circuits, they need three T1s to the office. If an organization only needs 15 voice trunks, it is still cheaper to get a T1 for access than run analog business lines with an average cost of $50 per line.
2. **Separate Services** – Local and long distance trunks are separate; thus, if one group of trunks fills up, blocking will occur while other trunks are available. PBXs

can be programmed to get around this, but due to additional complexity and cost, this is rarely done.

3. **Provisioning** – A majority of the MACD work requires manual human intervention.

After gathering the above information, the following spreadsheet should be used to calculate existing telecom trunking costs.

Trunk Type	Access T1/PRI	Trunks	Usage Inter-State	Usage Intra-State	Features Basic	Sub Total
Toll-free						
Default	*$300*	*0*	*.02/min*	*.05/min*	*$50/trunk*	
Long Distance						
Default	*$300*	*0*	*0.02/min*	*.05/min*	*$3/trunk*	
Local						
Default	*$300*	*$35*		*.01/min*	*$5/trunk*	
Support						
Taxes						
Total						

Table 3. Existing Monthly Telecom Costs Spreadsheet

In cases where the information is not known, the default can be used based on number of employees/contractors and industry average rates. The defaults are based on 99.99% availability, P.01 capacity, standard quality, and five basic features. The defaults are:

Toll-free

- Number of trunks = 1.5 times the total number of named (not concurrent) call center agents. Extra trunks are necessary for time spent in IVR and Queuing
- Access = number of trunks divided by 23 and rounded up
- Usage = total number of call center agents times 10,000 minutes per month
- Inter-state vs. Intra-state usage = Assume 15% of above minutes are intra-state
- Feature charges = $50 per trunk (most of this is for transfer charges)

Long Distance

- Number of trunks = Total number of employees divided by 10
- Usage = Number of employees times 1,000 minutes per month
- Inter-state vs. Intra-state usage = Assume 25% of minutes are intra-state
- On-net vs. Off-net usage – Intra-company calling vs. Inter-company calling. For the sake of simplicity, the difference between the two will not be accounted for.
- Feature charges = $3 per trunk

Local

- Number of trunks = Total number of employees divided by 5
- Usage = Number of employees times 500 minutes per month of local calls
- Feature charges = $5 per trunk

** Assumptions are that audio conferencing is an external service and employees dial a toll-free number to use it. Audio conferencing toll-free costs are separate, but access/trunk costs are included. Also, intra-company calling is still done over the PSTN vs. VoIP internally.*

Using the above assumptions, this would be the monthly cost for a Fortune 1000 company with 5,000 employees within the U.S., of which 500 are call center agents:

Trunk Type	Access T1/PRI	Trunks	Usage Inter-State	Usage Intra-State	Features Basic	Sub Total
Toll-free	33	750	4250000	750000	750	
Default	*$300*	*0*	*0.02*	*0.05*	*$50*	
Sub Total	*$9,783*	*$0*	*$85,000*	*$37,500*	*$37,500*	***$169,783***
Long Distance	22	500	3750000	1250000	500	
Default	*$300*	*0*	*0.02*	*0.05*	*$3*	
Sub Total	*$6,522*	*$0*	*$75,000*	*$62,500*	*$1,500*	***$145,522***
Local	43	1000	0	2500000	1000	
Default	*$300*	*$35*	*0*	*0.01*	*$5*	
Sub Total	*$13,043*	*$35,000*	*$0*	*$25,000*	*$5,000*	***$78,043***
Support					2.5 people	***$20,000***
Taxes					20%	**$78,670**
Monthly Total						**$492,018**
Yearly Total						**$5,904,211.20**

Table 4. Monthly Voice Trunking Costs for a 5,000-person company

The above numbers are an ideal case. Many companies are over-trunked due to the fact that over the years they have not deleted excess capacity as the business changed. Also, quite a few single analog phone lines may be in place for use with modems for systems such as external alarming, fax machines, and small and home offices. As much as businesses like to say we are in the digital age where all documentation is electronic, there are still a lot of modems and fax machines in place. Part of the business case for going to SIP trunking may be cleaning up the existing environment, but it is not used in this analysis.

One reason the usage on local trunks is significantly higher than long distance or toll free is the assumption that most organizations are communicating more via audio and web conferences versus a point to point call. The increase in conferencing usage has been driven by:

1) More meetings involve collaboration from people who are not in the same physical site and/or company
2) More people are mobile – working at home, working at a customer site, or out and about
3) People use conferencing as a way to avoid long distance charges or to keep from using up minutes are their mobile phones (depending on their call plan), esp. internationally. This typically costs an organization more money, but users are motivated by how easy conferencing is to use and do not want to incur an expense on their own home or mobile phone for a work call.

The $5.9 Million in annual cost is in line with an average cost of $1,000 per year per full time employee for voice telecom costs. This does not include cellular costs.

Summary

Calculating existing telecom costs can be time consuming. The examples used above are for estimating only and as a general rule, within 25 percent. To truly understand the costs, a detailed circuit inventory must be performed, with associated usage compared against negotiated rates and the actual bill.

Building the Business Case - Step 3: New SIP Trunking Costs

The first part in determining SIP trunking costs is to issue an RFP to multiple SIP trunking providers. This can include existing incumbent voice trunking carriers along with a few new SIP trunking service providers who are hungry for new business. Within the RFP, the business requirements, as discussed above, should be specified along with the current number of trunks, usage, and features.

In the response to the RFP, costs should be broken out into:

1. **Access** – Ethernet (10/100/1000) vs. T1 with IP connectivity as private or public.
    a. Private IP – VPLS or MPLS private IP connection between the enterprise and the carrier
    b. Public IP – Internet connectivity running a secure tunnel or other type of encryption
2. **Trunks** – Fixed cost per trunk with an assumption of an average number of concurrent voice trunks used during the busy hour of a normal week and the peak number of trunks that would ever be required. A fixed amount of long distance should be included in the monthly trunking cost. Concurrent call ports are defined by:
    a. Long distance only – Inter-company
    b. Long distance & local
    c. In-bound only – Typically for toll-free
3. **Usage** – Cost per minute for usage
    a. Unlimited
        i. Local only
        ii. Local and long distance
    b. Tiered Domestic long distance – X number of minutes per month bundled with each trunk
    c. Metered – Traditional model of a charge per minute for usage – Not recommended due to the common rule "40 percent of the phone bill *is* the phone bill."
4. **Features Costs**
    a. Bundled Basic Features Include – Fixed quantity of DID/Toll numbers, call transfers, ANI, dial plan and routing strategies, international authorization codes
    b. Additional Charges – Directory listings, Operator/411, 911 (possibly provided by a third party)

Based on the response to the RFP and an organization's business requirements, two important architectural decisions must be made that have a significant impact on the ROI of SIP trunking and the scope of the project:

1. **Centralized vs. decentralized trunking model.** A centralized trunking model has all SIP trunks coming into a few data centers and then riding a company's

WAN to all office sites. A decentralized trunking model has all voice trunks coming into each local office, which is the historical model.

2. **One vs. multiple SIP trunking providers**. A single provider is easier and initially cheaper but based on high availability requirements, coverage, and competition, multiple providers may be better over the long run.

The sample business case and all assumptions are based on a centralized trunking model with a single SIP trunking provider, per the graphic below.

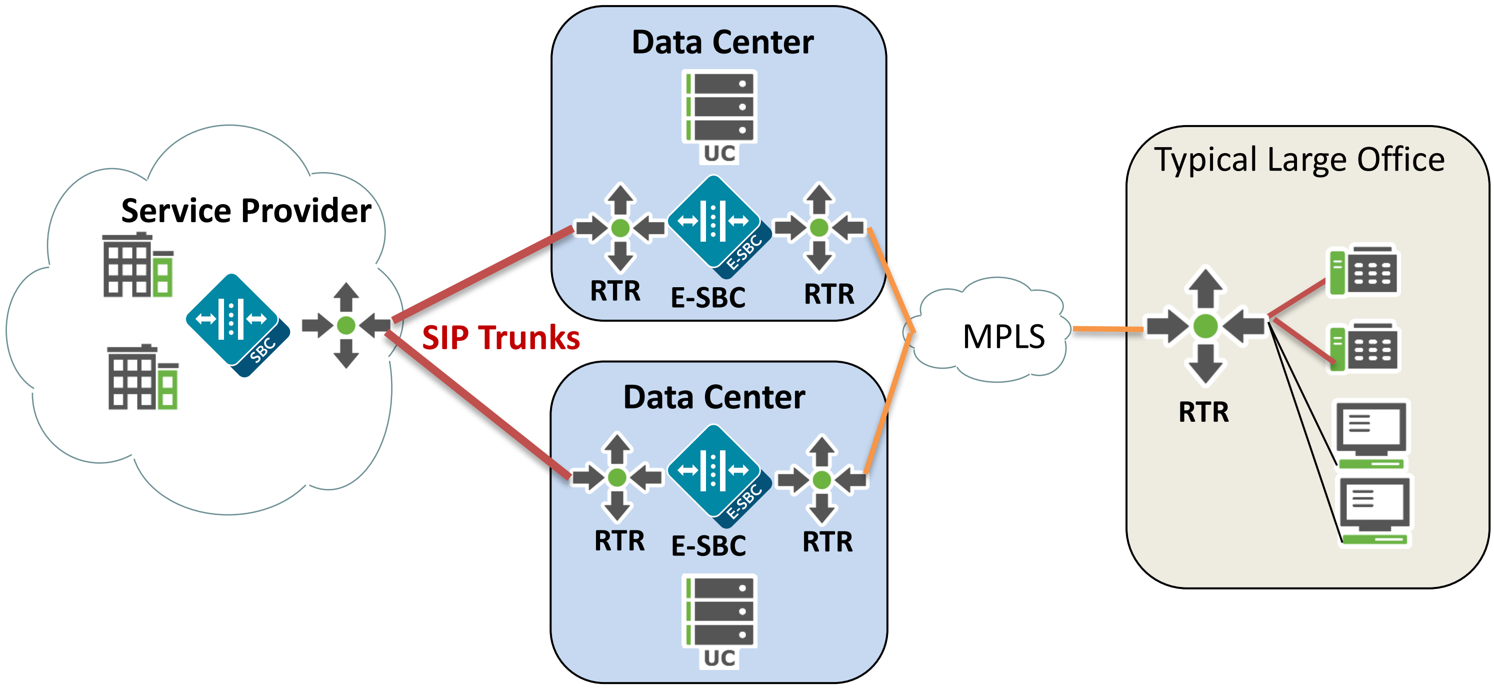

Figure 6. - Centralized SIP Trunking Architecture

SIP trunking is cheaper than traditional TDM trunking due to:

1. **Aggregation of Trunks**– By combining all voice trunks from many office sites into a few data centers, a business can reduce the number of voice trunks required across the entire enterprise by **30 to 50 percent,** while still maintaining capacity requirements for the business. This is because:
    a. **Busy Hour** – Due to different time zones, the maximum number of calls coming into an office varies; thus, one office receives a substantial number of calls while another office may not.
    b. **Sharing** – Local, long distance, and toll-free trunks can all share the same access
    c. **Seasonal Capacity** – Since adding extra voice trunks is just a software change, voice trunks are sized for average peak usage, not seasonal peak usage.
    d. **Erlang C Calculations** – If a business has 50 sites of 100 people and needs P.01 of service, it may need 60 voice trunks at each office for a total of 3,000 across the enterprise. To meet the same P.01 service requirement in a centralized model, only 2,200 trunks are required.

   e. **Exact Number** – Instead of having to buy trunks in increments of 24 on a T1 per site, like 72 in the above example for each office, an organization can get the exact number that they need with the option to burst above this normal concurrent volume.

2. **Aggregation of Access** – A few Ethernet access lines are cheaper than many T1s into each office. Since most companies already have redundant fiber connectivity from their data centers, adding additional Ethernet links for voice access is cheap and easy. Access cost can drop by as much as 80 percent.
3. **On-Net Calling** – All inter-company calls go over the WAN and no longer require carrier trunks. Most small and medium size organizations will utilize a 3rd party to provide their IVR and conferencing services, as a Software as a Service (SaaS) cloud model. Large organizations have an opportunity to either host IVR and conferencing in-house, or run private SIP trunks from their data centers to the external SaaS/Cloud provider and avoid the cost of service provider SIP trunks.
4. **Free Features** – With SIP trunks, many of the features that were charged separately are now bundled into the overall fixed monthly charge. This varies by SIP provider, but the general free features are:
   a. **D Channel** – The signaling and information pasted on the ISDN D channel is part of the SIP invite message. There is no longer a separate charge of $100 per D channel.
   b. **Transfers** – An SIP Refer will send the call to a different organization at no charge or just a one-time charge and at a lower rate, typically 1 cent per SIP Refer. Traditionally, this cost 2 cents for every call on a toll free number (whether used or not used) or 25 cents if used selectively.
   c. **Long Distance** – Typically, 1,000 minutes of long distance are included with a local trunk as part of the base charge.
   d. **DID Numbers** – 20 DID numbers per 100 voice trunks are included. Existing DID numbers can be kept in most cases and ported over to a new service provider.
5. **Tariffs** – With SIP trunks, many traditional tariffs in TDM telephony no longer apply. This includes:
   a. **Intra-state –** Since all calls go through data centers that are out-of-state (through appropriate routing), the high cost for intra-state long distance/toll-free goes away.
   b. **Local Usage** – Local calling within a LATA does not have a per-minute charge.
   c. **USF** – As of the writing of this whitepaper, USF charges are not being applied to VoIP/SIP trunking.
6. **Competition** – In a centralized model it is easier to have multiple SIP trunking providers and/or to switch SIP trunking service providers.
7. **Billing** – Forty percent of the phone bill *is* the phone bill. An extraordinary amount of effort goes toward tracking usage and getting the appropriate internal charge backs. Some Service Providers offer SIP trunking with fixed-rate billing, allowing voice bills to look like data networking bills. The costs are fixed and predictable, allowing a company to pay a flat rate for normal usage and a fee if they go above the negotiated threshold. Also, the current tax and fee structure for SIP trunks is lower than that of traditional TDM trunks.
8. **Support** – In a centralized model, the managing, administration, and support of voice trunks is simpler, easier, and cheaper.

Building the Business Case - Step 4: Architecture Options

Centralized vs. Decentralized

Most organizations, which have deployed IP Telephony, centralize all call control processing into the data center, but still keep the trunking local and have a backup call control processor at the site. This allows for calls to be made, even if the IP network to the data center is lost. While this sounds good in theory, the reality is that in most cases when IP network connectivity to the data center is lost, the phone connectivity is also lost, or of little value. The primary reason for IP network connectivity loss is a failure of local access, which in most cases impacts traditional voice trunks also. The second most common reason is loss of power, which results in the lack of lighting and the ability to use computers, making it generally impossible for people to work.

Centralizing voice trunks into data centers is the next step in the evolution of voice communication as it is integrated with data for Unified Communications and offered as a service, versus a point solution. Long term, the next generation of IT architecture is for all IT services to live in private or public "clouds" and to minimize the amount of technology within an office and an end device.

The <u>advantages</u> of a centralized voice trunking model are:

1. **Higher Availability –** Data centers have redundant local access, power, on-site support and HVAC. Using multiple data centers in an active-active model ensures the highest availability of voice services possible.
2. **Business Continuity –** Ability to close an office due to weather, pandemics, or other causes and still be able to handle all in-coming calls by rerouting to an alternate location.
3. **24/7/365 –** Ability to answer calls anytime if the business chooses without having to listen to messages or wait while being transferred around.
4. **100 percent Call Recording –** Recording and analyzing all calls for industry compliance, quality, and speech analytics to understand why customers call and how to improve sales and service. Call recording use is being expanded from not just in-bound toll-free calls, but to all customer contact. This includes both in-bound and outbound calls. Certain industries are required to record calls for compliance. And, by recording calls at the trunk interface, organizations are able to improve fidelity and capture the entire call, including IVR interactions.
5. **Improving Productivity –** Further integration of people, processes, and information with communication to improve the efficiency and effectiveness of a business.
6. **Better ROI –** Economies of scale as outlined above.
7. **Security –** Fewer points of entry to manage, control, and log.
8. **Flexibility –** Integration of voice with other systems, along with the adaptability to meet changing business needs.
9. **Conferencing –** Whether conferencing is done in-house, or at a 3rd party, funneling all calls through the data center enables better economies of scale and in the case of 3rd parties, the ability to provide an IP connection directly to the 3rd party versus paying carrier SIP trunking rates.
10. **Work at home –** Utilizing the internet to provide enterprise telephony services for work at home employees. The traditional model was to provide a phone line from the local PBX in the area to the home and paying for both a phone line on the PBX and an analog line into the home. By putting in a router and IP (hard and/or

soft) phone, an organization can get rid of the phone lines. Since most Internet connectivity for an enterprise is centralized, having the SIP trunks centralized minimizes any WAN back-haul.

The disadvantages of a centralized voice trunking model are:

1. **Bigger Project** – Upgrade of the WAN to handle additional voice traffic. The cost of WAN bandwidth continues to drop dramatically with Appendix 2 providing an example of current competitive pricing. If an organization is still utilizing T1s for data access to its small sites, the network will need to be upgraded using Ethernet or Cable/DSL access and with public or private IP network on top of it.
2. **Organizational –** In the past, voice has been its own island within IT. Moving to 100 percent VoIP/SIP in a centralized model folds voice into IT as another application that is managed within the environment and subject to standard IT governance. Some organizations still view voice as special, not just another critical IT application.

IP Telephony vendors have a vested interest in keeping telephony trunks decentralized, since they can sell more hardware/software with this model. In a recent survey of 600 organizations deploying SIP trunking, the majority of them did so in a centralized model. So the industry is gravitating to a centralized model where voice communication is an IT application/service. **1*

Single SIP Trunking Service Provider vs. Multiple Service Providers

A single service provider is the easiest and cheapest model, at least in the short-run. This is the model recommended for small and medium-sized organizations. Large organizations have the scale to potential justify multiple service providers.

A second service provider does not offer greater availability for in-bound DID numbers, since they are assigned by a specific service provider. The availability of in-bound calls is then tied to the carrier that a company has its DID numbers registered to.

A second service provider does offer greater availability for in-bound toll-free numbers, since they can be allocated across multiple service providers using a capability called RespOrg, which allows an organization to dictate how their toll free numbers are routed

RespOrg (a contraction for responsible organization) is a term that refers to the companies that have access to the Service Management System, the database that controls routing on all toll-free telephone numbers. RespOrgs were established in 1993 as part of the FCC order to institute toll free number portability. Every individual ten-digit toll free telephone number is managed by a RespOrg. A RespOrg can be a long distance company, reseller, end user or an independent that offers an outsourced service.

In the past decade, large Tier1 carriers have not been able to deliver all toll free calls within their network both regionally and nationally, primarily due to large call volumes events like 9/11, *American Idol* voting, first business day of the year, extremely large conference calls, natural disasters, and technical challenges within their networks. Property insurance companies for example should utilize multiple carriers, because in natural disasters, when communication is needed the most, a lot of times one carrier is available in a given area.

Another advantage of a second SIP trunking service provider is for outbound calling, both in terms of redundancy and costs. By peering {8} with multiple service providers, an organization can send outbound calls to the service provider that has the lowest costs for the called party. Again, this makes the most sense for very large organizations that do millions of minutes of outbound calls in a month. Companies with very large international calling requirements are adopting multiple carriers the quickest and utilizing the most cost effective provider for the type of call.

Building the Business Case - Step 5: the Business Case

With the knowledge of business requirements for voice trunking, existing voice trunking costs, SIP trunking costs from a Service Provider(s), and an architectural model, the business opportunity for moving to SIP trunking can be calculated. This calculation can be an estimate based on the number of employees within an organization and industry averages for voice trunking usage and costs, or it can be detailed based on gathering all pertinent information.

The case study below is an estimate based on the number of employees within an organization and industry averages for voice trunking usage and costs.

Case Study

Back to our Fortune 1000 company that has 5,000 employees, of which 500 are contact center agents. We know what their estimated current telecom spend is $5.9M/year. What would it be if they implemented SIP trunking in a centralized, single Service Provider model? Based on the following SIP RFP response assumptions, the table below provides an estimate.

SIP Trunking RFP Response Assumption Results:

- **Access – $10,000/month** - 100M Ethernet connectivity between Service Provider and Data Center for four, 100M Ethernet connections running VPLS with a full port speed, each capable of carrying 800 concurrent G.711 calls.
- **Trunk - $25/month** – Trunk includes basic features plus 1000 minutes of long distance.
- **Overflow Usage – $0.01 cents/minute** – Usage above standard monthly allocation.
- **Feature Charges** – Same as current costs for operator/411, directory listings, and other services not included in standard voice trunk costs.

SIP Trunking business requirements for our hypothetical Fortune 1000 company:

- **Reduce to 1,500 trunks** instead of the 2,250 currently used. Since intra-company calling and conferencing now ride over the WAN and trunk aggregation, 750 fewer trunks are required.
- **750,000 minutes of long distance** since intra-company calling and conferencing now ride over the WAN. Since all 1500 trunks have 1000

minutes of long distance included for a total of 1,500,000 minutes a month, no additional overflow usage charges should occur.

- **Feature charges will drop**, especially for toll-free, such as transfer connect. Let's assume they average $5 per trunk.
- **Less Support** – One employee can now support 4,000 callers instead of 2,000
- **Lower Taxes** – Assume that taxes and fees drop to 5 percent

Trunk Type	Access Ethernet	Trunks	Usage Inter-State	Usage Intra-State	Features Basic	Sub Total
SIP Trunks	4	1500	0	0	0	
Default	*$2,500*	*$25*	*0*	*0*	*$5*	
Sub Total	*$10,000*	*$37,500*	*$0*	*$0*	*$7,500*	***$55,000***
Support					1.25 people	***$10,000***
Taxes					5%	**$3,250**
Monthly Total						**$68,250**
Yearly Total						**$819,000.00**

Table 5 – Monthly Costs For SIP Trunking Example

The resulting telecom voice savings are $5 Million dollars a year!

There are some off-setting costs of moving to this model. They include:

1. Hardware - Data Center Routers, Enterprise Session Border Controllers, and associated hardware and space. Assume 500K to 1M
2. WAN - Upgrade of the WAN to carry all voice traffic back to the data center. For most large offices that already have DS-3 access and port speed, the associated gold-level commit rate upgrade for voice is nominal. For smaller offices, the cost can be higher unless an organization moves to next generation Ethernet access. Assume $100,000/month in additional WAN costs.
3. Implementation - A project this size takes 10,000 – 20,000 hours at $100 an hour, which equals $1 million to $2 million in costs.

Based on these numbers, this SIP trunking project would break even in 9 – 12 months. The Net Present Value and ROI would be over $15 million depending on the cost of capital and number of years that the project would be capitalized over.

One important cost that was excluded in the above calculation is the centralization of the PBX, IP-PBX, or IP Telephony infrastructure into the data center and upgrades to support VoIP/SIP. As mentioned earlier, IP Telephony has its own business case and should stand on its own, though the two projects are dependent on one another to achieve optimal savings.

Summary

There is significant cost savings to be had in moving from traditional telephony phone trunks to SIP trunking. Most organizations will see their overall monthly expense drop by 20-50 percent. Much of the savings potential is based on business requirements and if a centralized architectural model is adopted.

Building the Business Case - Step 6: Other Benefits of SIP Trunks

Besides lowering monthly telecom trunking costs, SIP trunking can also reduce costs by:

1. **Clean-up** – Most telecom environments have evolved over decades and a lot of trunks sit idle while they are still being paid for.
2. **Green IT** – SIP trunks use 50-75 percent less power than their TDM predecessors due to the power required to drive and terminate copper T1s versus using fiber optics and high-density servers.

SIP trunking is also strategic for an organization due to:

1. **One Network** – Getting all corporate applications on one IP network instead of having separate voice, video, and data networks. Maintaining parallel networks is expensive and slows down an organization from adapting to every changing business requirements. SIP trunks can be more rapidly provisioned than TDM trunks and services. As shown in Appendix 1, the price of bandwidth gets cheaper with bigger pipes.
2. **Multi-Channel Communication** – As cellular technology moves to 4G VoIP LTE and IP phone providers like Skype take off, integrating chat, voice, video, and the Web will become the norm. HTML 5 and WebRTC with voice codecs embedded in a standard Internet browser further enables this unification. Over time, the number of phone calls will continue to rise and it will be the adoption of multi-channel communication that will shorten the average duration of a call, which will enable a business to optimize employee productivity in terms of efficiency and effectiveness.
3. **Standards Based** – SIP is a standard that enables interoperability between software, hardware, and service provider vendors. While SIP is still fairly young, it has reached critical mass and is what the telecom industry has adopted. Like with all standards, SIP-based hardware and transport will become a commodity, and enhanced communication features through software will add top line value to businesses and their customers.
4. **International –** As domestic calling becomes flat rate and runs an average of one cent a minute, the majority of future telecommunication expense reduction will be on the International side. Appendix 3 has some illustrative pricing. Organizations will have two options internationally:
    a. **Best Effort** – Going with an Internet based service such as Skype or FaceTime that can be accessed via any IP phone including mobile phones on WiFi. This provides a very low cost solution, but the quality is not always there in terms of added delay and/or intermentant dropped packets. Overtime, these solutions are improving, but most will sacrifice delay or quality.
    b. **Quality** – Using a local phone provider to ensure quality while finding an International service provider to provide the lowest cost rate. These vary by country, so a call by call routing solution needs to be put in place.
5. **Mobile –** VoIP LTE – The next big step in moving to VoIP in the communication industry is the mobile/cellular carriers offer 4G voice utilizing the LTE network and VoIP. This will enable VoIP calls end to end, no matter if the other party is on a mobile, Internet, or IP/SIP trunking end point.
6. **Video –** Once mobile devices adopt VoIP, adding video will be as easy as pushing a button. Most enterprises do not allow external device/desktop video

into their internal employees. SIP trunking providers are testing offering video such has H.263/4 that will allow calling parties to easily add video to the communication.

7. **Conferencing –** Most organizations have relied on 3^{rd} party audio and web conferencing solutions. As collaboration suites take off, there is a lot of synergy to hosting conferencing internally so that it can be integrated with the directory that has details on all the participants, file shares for common documentation, and project management suites for tracking who is doing what. With centralized SIP trunking, a buy versus build decision can be evaluated without the traditional high telecom expenses associated with an in-house solution. Some enterprises use this exercise to leverage their existing conferencing providers to reduce their rates.

In order to keep the risk of migrating to SIP trunking to a minimum, an organization should:

- Test and pilot the technology, and in the process, educate engineering and support staff.
- Get the appropriate monitoring, alarming, reporting, and support tools in place.
- Start with long distance and conferencing trunks first, then toll-free, then local trunks last.

Outbound calls offer a quick ROI with minimal risk to an organization since the existing PSTN can be left in place as a backup while the lower-cost SIP trunks are being used. Also, outbound calling offers fewer security and implementation challenges.

A good E-SBC is a critical component to the success of a SIP trunking project. The E-SBC is the demarcation point, not only from a security perspective, but also from a support, monitoring, and reporting perspective.

Summary

SIP trunking is both a short-term solution for cutting telecom costs along with a long-term solution for multi-channel communication. All good IT infrastructure projects cut bottom line costs while enabling top line revenue growth. A good E-SBC is a critical component to the success of an SIP trunking implementation.

SIP Trunking Savings Using the SIP Trunking Calculator

How much can your organization save by moving to SIP trunking? Gather the information below and then use the SIP trunking calculator to estimate the potential business opportunity. The calculator estimates how much an organization can lower its monthly voice trunking costs to telecom carriers by moving to SIP trunking.

Current Business Telephony Environment

1. Number of Employees & Contractors
2. Number of Call Center Agents
3. Total Number of PRIs and trunks
    a. Toll-free
    b. Long distance

    c. Local
4. Monthly usage in minutes
    a. Toll-free
        i. Interstate
        ii. Intrastate
    b. Long distance
        i. Interstate
        ii. Intrastate
    c. Local
        i. Free
        ii. Charged

Information in light blue is all that is required for a high-level estimate.

<u>Current Telephony Trunking Costs</u>

1. T1 Access
2. D Channel
3. Long distance
    a. Inter-state
    b. Intra-state
4. Toll-free
    a. Inter-state
    b. Intra-state
5. Local
    a. Local usage charge rate
6. Feature Charges
    a. Toll-free
    b. Long distance
    c. Local
7. Support – Total number of telecom support people (technical, finance) associated with telecom trunking
8. Taxes & Fees such as Universal Service Fee (USF)

<u>New SIP Trunking Costs</u>

1. Access
2. Trunks
3. Excessive Usage
4. Features
5. Taxes

Using the SIP Trunking Calculator

Enter the above information into the appropriate fields. Use the calculator to run costs under multiple scenarios with different business requirements. Also utilize the calculator to compare the cost savings from various SIP trunking providers to see which one offers the best deal, both in the short term, and over 3 years.

Next Steps

The objective of this whitepaper is to show the business cost savings opportunity in moving from traditional voice trunking to SIP trunking. For most organizations, the potential cost savings is significant. By presenting the opportunity to senior management within an organization, the next step would then be to form a team to perform a detailed analysis, create a design, build a project plan, and produce a financial cost-benefit analysis.

Conclusion

SIP trunking offers an organization both significant cost savings potential along with further integration of voice communication into IT applications/services. An organization can cut average telecom costs per employee of $1,000/year by 50 percent or more while integrating voice with data to provide unified communication solutions.

Glossary

(1) **LATA** - Local Access and Transport Area is a term used in U.S. telecommunications regulation. It represents a geographical area of the United States under the terms of the Modification of Final Judgment (MFJ) that precipitated the breakup of the original AT&T into the "Baby Bells" or created since that time for wire line regulation. For more info: http://en.wikipedia.org/wiki/LATA
(2) **LEC** - Local Exchange Carrier is a regulatory term in telecommunications for the local telephone company. http://en.wikipedia.org/wiki/Local_exchange_carrier
(3) **IXC** – Inter-exchange Carrier is a U.S. legal and regulatory term for a telecommunications company, commonly called a long-distance telephone company, such as MCI (before its absorption by Verizon), Sprint and the former AT&T (before its merger with SBC in 2005) in the United States. It is defined as any carrier that provides inter-LATA communication.
(4) **ISDN** - Integrated Services Digital Network (ISDN) is a set of communications standards for digital transmission of voice over the traditional circuits of the public switched telephone network.
(5) **WAN** – Wide Area Network – A network that interconnects all of an organization's local data networks.
(6) **PSTN** – Public Switched Telephone Network - The network of the world's telephone networks.
(7) **B2BUA -** A Back-to-Back User Agent is a logical SIP network element. It resides between both end points of a phone call or communications session and divides the communication session into two call legs. It mediates all SIP signaling between both ends of the call, from call establishment to termination. Each call is tracked from beginning to end, allowing the operators of the B2BUA to offer value-added features to the call.
(8) **Peering** - Peering is a voluntary interconnection of administratively separate Internet networks for the purpose of exchanging traffic between the customers of each network.
(9) **USF** - http://en.wikipedia.org/wiki/Universal_Service_Fund

References

1. No Jitter Webinar – "Overcoming The Technical Obstacles Of SIP Trunking" – June 24, 2009. On-line participant survey.

Appendix 1 WAN Rates

The cost of Wide Area Network bandwidth has dropped dramatically in the past 5 years driven by:

- **Ethernet Access –** Moving to Ethernet access and away from DS1/3 and SONET OC-X speeds. Ethernet equipment is cheaper and allows for easier aggregation. A lot of providers will offer 10, 100, 1,000Mbps links to their customers and then provide 10Gbps up-links to their backbone.
- **VPLS –** Layer 2 Ethernet based services. For links that are providing bandwidth from a SIP trunking provider to an enterprise, a layer 2 solution is good enough. If the only traffic that is on the link is voice media and its associated SIP signaling, QoS does not offer much benefit. One thing to note is Ethernet adds about 20% bandwidth overhead to voice packets.

Example competitive carrier pricing is shown below. This pricing can be attained from both tier 1 and tier 2 providers and is the current mid-point. Access assumes that it is in a city with a population above 250,000 and that the provider can/will provide its own access to the site.

Mbps	Ethernet Access	VPLS	Total Monthly
10	500	200	**$700**
50	750	750	**$1,500**
100	1,000	1,500	**$2,500**
500	3,500	2,000	**$5,500**
1,000	3,500	4,000	**$7,500**

SLA's	
Jitter	< 3ms within U.S.
Delay	< 50ms within continental US
Dropped packets	< 0.01% (less than one out of 10,000 packets)
Availability	99.9% - Single access into the site -
Provisioning	90-120 business days

For calculation purposes, assume 100Kbps per a single G.711 voice trunk utilizing Ethernet and IP and that a connection can handle 80% utilization before instantaneous bandwidth contention may start becoming a factor. As an example, a 100M connection can support 800 concurrent calls.

Appendix 2 – Private SIP Trunking

Private SIP trunking is used when an enterprise wants to bypass Telecom Service Providers and provide direct voice connectivity to a 3rd party. This is most commonly done with SaaS/Cloud IVR and Conferencing vendors. It can also be done between organizations that have a lot of voice trafficking between them. Figure 1 illustrates typical 3rd party voice connectivity through the SIP Trunking Service Provider.

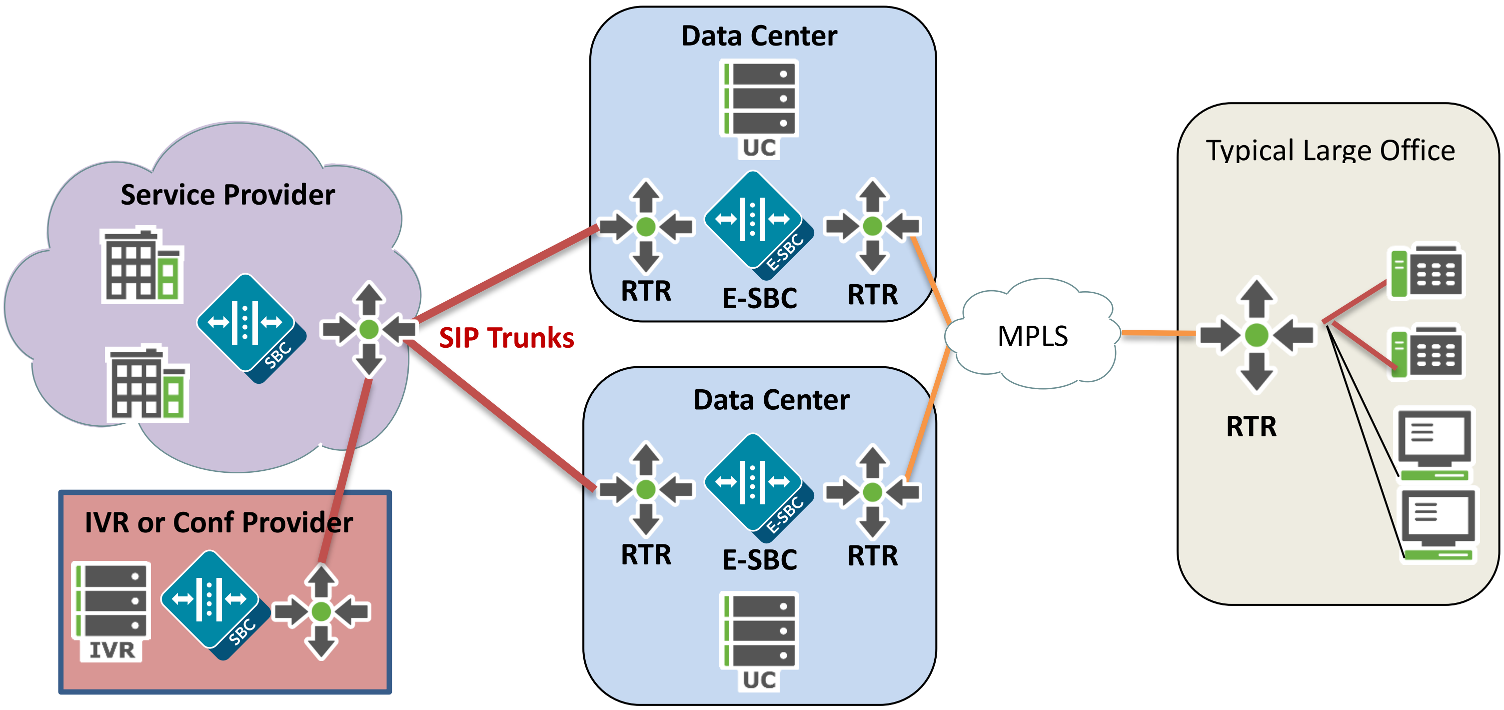

Figure 1 - Standard 3rd Party Voice Connectivity

Implementing private SIP trunks makes sense when there are ~400 or more concurrent trunks. The below table illustrates the cost saving opportunities when utilizing private SIP trunking to a 3rd party as shown in figure 2. The below columns represent:

- **Trunks** – The peak number of concurrent trunks
- **Carrier** Cost – The $25 per SIP trunk from the Service Provider
- **WAN** - The WAN costs from Appendix 1 assuming two network connections.
- **H/S, O&M** – The additional hardware and software licensing costs (capitalized over 4 years) plus operational support and maintenance

Trunks	Carrier Cost	WAN	H/S, O&M	**Monthly Savings**
100	$2,500	$1,400	$4,000	**-$2,900**
500	$12,500	$3,000	$6,000	**$3,500**
1,000	$25,000	$5,000	$9,000	**$11,000**
5,000	$125,000	$15,000	$12,000	**$98,000**

Table 1 – Private SIP Trunking Cost Savings

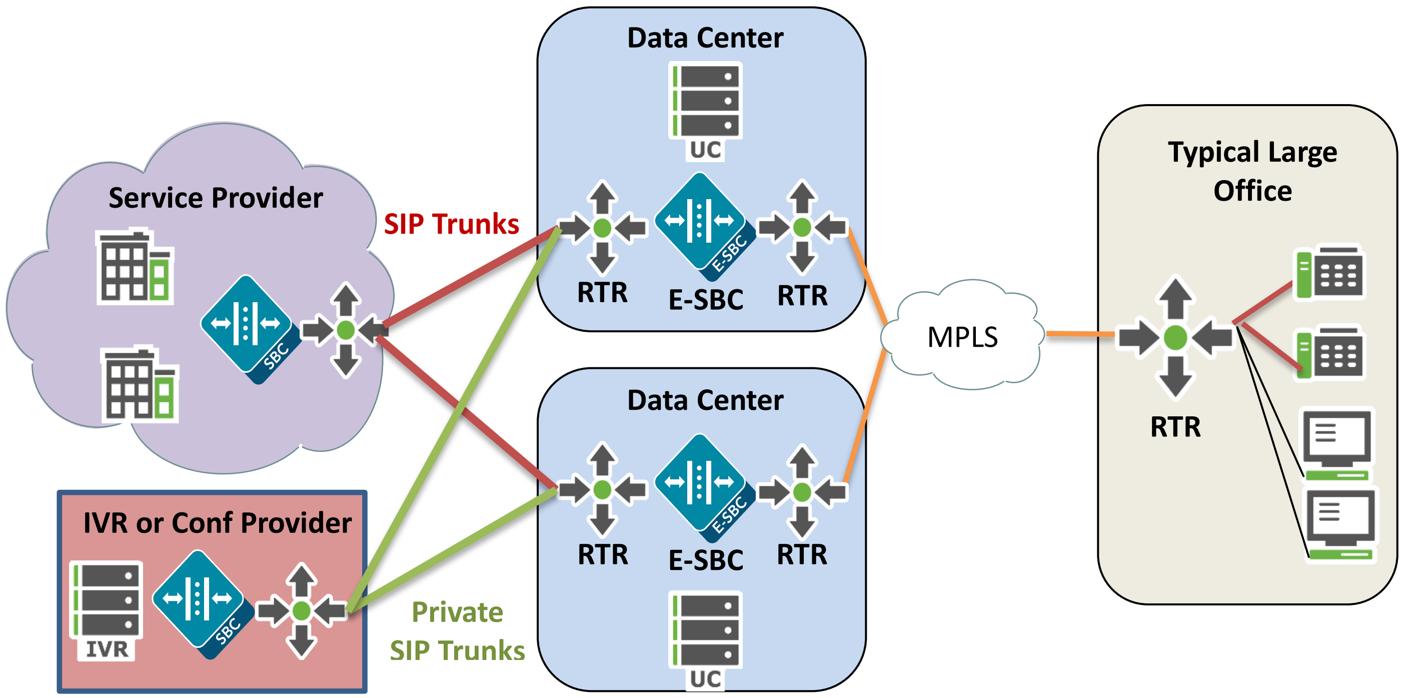

Figure 2 – Example of Private SIP Trunking to a 3rd Party

Appendix 3 – International Rates

Calling internationally is still expensive, especially if the international party is on a mobile phone. Below is a price comparison of what a standard Fortune 500 enterprise would pay between utilizing the traditional PSTN, Skype, and a SIP trunking provider. International prices and contracts vary widely, so please use the below as a general example. The assumptions on the pricing below are:

- **Skype –** A Skype client in the U.S. going a non-Skype phone overseas with an average of 500 minutes in that country a month
- **SIP Trunking Provider –** A tier 1 carrier that has international agreements with the local telephony providers and has direct IP/SIP cross-connects.

COUNTRY	COUNTRY CODE	PSTN Rate	Skype	SIP Trunking Provider
INDIA	91	$0.24	$0.08	$0.12
INDIA MOBILE	91	$0.65	$0.08	$0.31
UNITED KINGDOM	44	$0.04	$0.03	$0.04
UK MOBILE	44	$0.05	$0.03	$0.04
VIETNAM	84	$0.41	$0.13	$0.22
VIETNAM MOBILE	84	$1.20	$0.13	$0.74
SPAIN	34	$0.08	$0.05	$0.06
SPAIN MOBILE	34	$0.32	$0.05	$0.18
ISRAEL	972	$0.09	$0.05	$0.08
ISRAEL MOBILE	972	$0.35	$0.05	$0.20
CHINA	86	$0.15	$0.12	$0.15
CHINA MOBILE	86	$0.44	$0.12	$0.25
JAPAN	81	$0.06	$0.05	$0.06
JAPAN MOBILE	81	$0.38	$0.05	$0.16