Moving to VoIP and SIP trunking has the potential to reduce 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.

Sorell Slaymaker
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 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.

Building a business case for SIP trunking is a 6-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, rates, and bills from the carriers.
3. Issue RFPs to get the costs for 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 United States, these 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.
- **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 so callers can 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.
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).

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, Qwest, or a long list of other service providers. A key component in moving to SIP trunking is the Session Border Controller (SBC). The SBC provides security, management, monitoring, support, and reporting for SIP trunks. This is illustrated in Figure 3.
Building the Business Case for SIP Trunking

Help and to keep the business operational. As cellular messaging technologies like

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

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 problems 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
Building the Business Case for SIP Trunking

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.
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 inbound 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.

<table>
<thead>
<tr>
<th>MOS</th>
<th>Quality</th>
<th>Impairment</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>Imperceptible</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>Perceptible</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>Slightly Annoying</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>Annoying</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
<td>Very Annoying</td>
</tr>
</tbody>
</table>

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, poorer 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.

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 free product. When voice moves to IP, it is subject to all the security required for all of 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.

A session border controller (SBC) plays a critical role in providing SIP trunking security. The SBC acts as an SIP back-to-back user agent, or B2BUA, and sits between an organization and any third party that it sends calls to, especially the SIP trunking provider(s). The 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 that the voice system never has more calls than it can handle.
- **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 and usage by support personnel.

An SBC is used to secure inter-company voice communication in one of the following four cases:
1) **IP trunking border** — connections to service provider IP networks linking the enterprise to the outside world of PSTN and IP endpoints

2) **Private network border** — connections to internal employees located on the enterprise campus LAN and in remote offices connected via private WAN services such as MPLS VPNs

3) **Internet border** — connections to small offices, users working from home and mobile employees over the public Internet

4) **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.

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**Figure 4 – Four Common Session Border Control Points**

VoIP/SIP has many unique security requirements that the traditional corporate firewall cannot provide, thus the reason for adding Session Border Controllers as part 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.
   a. Basic Features – DID numbers, Directory Assistance (411), Directory Listing, operator services, Emergency (911)
   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.
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 IXC. 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, but this varies widely within U.S. metro areas.
   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.
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. This comes to at least 10 percent of the total phone bill.

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**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.
<table>
<thead>
<tr>
<th>Trunk Type</th>
<th>Access</th>
<th>Trunks</th>
<th>Usage Inter-State</th>
<th>Usage Intra-State</th>
<th>Features</th>
<th>Sub Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Toll-free</td>
<td>$300</td>
<td>0</td>
<td>.02/min</td>
<td>.05/min</td>
<td>Basic</td>
<td>$50/trunk</td>
</tr>
<tr>
<td>Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Long Distance</td>
<td>$300</td>
<td>0</td>
<td>.02/min</td>
<td>.05/min</td>
<td>Basic</td>
<td>$3/trunk</td>
</tr>
<tr>
<td>Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Local</td>
<td>$300</td>
<td>$35</td>
<td>.01/min</td>
<td></td>
<td>Basic</td>
<td>$5/trunk</td>
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<tr>
<td>Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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 percent availability, P.01 capacity, standard quality, SIP-TLS security, and five basic features. The defaults are:

**Toll-free**
- Number of trunks = 1.5 times the total number of 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 percent 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 percent of minutes are Intra-state
- On-net vs. Off-net usage – Intra-company calling vs. Inter-company calling. For the sake of simplicity, the 10 percent 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:

<table>
<thead>
<tr>
<th>Trunk Type</th>
<th>Access T1/PRI</th>
<th>Trunks</th>
<th>Usage Inter-State</th>
<th>Usage Intra-State</th>
<th>Features Basic</th>
<th>Sub Total</th>
</tr>
</thead>
<tbody>
<tr>
<td>Toll-free</td>
<td>33</td>
<td>750</td>
<td>4250000</td>
<td>750000</td>
<td>750</td>
<td>$9,783</td>
</tr>
<tr>
<td>Default</td>
<td>$300</td>
<td>0</td>
<td>0.02</td>
<td>0.05</td>
<td>$50</td>
<td>$169,783</td>
</tr>
<tr>
<td>Long Distance</td>
<td>22</td>
<td>500</td>
<td>3750000</td>
<td>1250000</td>
<td>500</td>
<td>$145,522</td>
</tr>
<tr>
<td>Default</td>
<td>$300</td>
<td>0</td>
<td>0.02</td>
<td>0.05</td>
<td>$3</td>
<td>$145,522</td>
</tr>
<tr>
<td>Local</td>
<td>43</td>
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<td>0</td>
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<td>1000</td>
<td>$13,043</td>
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<tr>
<td>Default</td>
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<td>$35</td>
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<td>0.01</td>
<td>$5</td>
<td>$78,043</td>
</tr>
<tr>
<td>Support</td>
<td>2.5 people</td>
<td>$13,043</td>
<td>$35,000</td>
<td>$0</td>
<td>$5,000</td>
<td>$20,000</td>
</tr>
<tr>
<td>Taxes</td>
<td>10%</td>
<td>$39,335</td>
<td></td>
<td></td>
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<td>$39,335</td>
</tr>
<tr>
<td>Monthly Total</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>$452,683</td>
</tr>
<tr>
<td>Yearly Total</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>$5,432,191</td>
</tr>
</tbody>
</table>

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, 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.

The $5,432,191 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** – DSL or Ethernet (10/100/1000) vs. T1 with IP connectivity as private or public.
   a. Private IP – MPLS or other 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. Inbound 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.

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. If audio conferencing and/or IVR services are outsourced to a third
party, the connection to the third party is through an IP/SIP connection versus telecom 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 Redirect will send the call to a different organization at no charge.
   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.

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
Building the Business Case for SIP Trunking

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 advantages 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 incoming calls.
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.
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.

The disadvantages of a centralized voice trunking model are:

1. **Bigger Project** – Upgrade of the WAN to handle additional voice traffic.
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.

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.

From a reliability perspective, a second service provider does not offer greater availability for inbound calls since DID and toll-free numbers are assigned by a specific service provider. The availability of inbound calls is then tied to the carrier that a company has its numbers registered to. In the past decade, large Tier1 carriers have not been able to deliver all calls within their network both regionally and nationally, primarily due to large call volume events like 9/11, American Idol voting, first business day of the year, and extremely large conference calls. These outages—where a significant number of calls are blocked—have occurred only for one day. It takes two or more days to port a number from one carrier to another.
The biggest advantage of a second SIP trunking service provider is for outbound calling, both in terms of redundancy and costs. By peering 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.

**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**

Now 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 $5.4M/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 – $20,000/month** - 100M Ethernet connectivity between Service Provider and Data Center for four, 100M Ethernet connections running MPLS with a full port speed and 90 percent high-queue commitment, each capable of carrying 900 concurrent G.711 calls.
- **Trunk - $25/month** – Trunk includes basic features plus 1000 minutes of long distance.
- **Overflow Usage – $0.015 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.**

- **1,500 trunks** instead of the current 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 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, as transfer connect fees go away. Let’s assume they average $5 per trunk.
- Support – One employee can now support 4,000 callers instead of 2,000
- Taxes – Assume that taxes and fees drop to 5 percent
Building the Business Case for SIP Trunking

Whitepaper

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| Support      | 1.25 people | $10,000 |
| Taxes        | 5% | $3,750 |
| Monthly Total|            | $78,750 |
| Yearly Total |            | $945,000.00 |

Table 5 – Monthly Costs For SIP Trunking Example

The resulting telecom voice savings are $4.4 Million dollars a year!

There are some offsetting costs of moving to this model. They include:

1. Hardware - Data Center Routers, Session Border Controllers, and associated hardware and space. Assume $1 million to $3 million.

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 access such as business grade DSL, Cable, or Ethernet. 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 12 – 18 months. The Net Present Value and ROI would be over $10 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 over 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 coppers 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.
2. **Multi-Channel Communication** – As cellular technology moves to 4G and IP phones like Skype take off, which are VoIP/SIP based, integrating chat, voice, video, and co-browsing will become the norm. 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.

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 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 back-up while the lower-cost SIP trunks are being used. Also, outbound calling offers fewer security and implementation challenges.

A good SBC is a critical component to the success of a SIP trunking project. The 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 SBC is a critical component to the success of an SIP trunking implementation.

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 detailed financial cost-benefit analysis.
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.

References