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SIP Trunking for Dummies

THE

Authoritative Guide

Contents

The Nerd's Guide to SIP Trunking.

All you need to know about SIP Trunking for small business

Okay, maybe a little more than you need to know....

Introduction to SIP Trunking.

The phrase "SIP trunking" or "SIP trunks", is increasingly common in the world of telecom and they are rapidly replacing traditional telephone lines. However if you are reading this, you are probably not in the telecom business and would like to know more.

Is SIP the Same as VoiP?

To understand SIP trunking you first need to understand VOIP. VoiP is short for Voice over Internet Protocol. It is the name given to any voice call transmitted over a data network. SIP trunking is a version of VOIP, specifically designed to make and deliver phone calls. Just as a point of interest, Hosted PBX, (a PBX in the cloud that is used as a service in place of a premise PBX) is another version of VoiP.

What is SIP?

SIP stands for "Session Initiation Protocol. Over the past few years, the telecommunications industry has standardized on SIP as the phone call protocol of choice. A SIP "session" could be a regular phone call between two participants, a multi-party conference call or even a video call.

What Is SIP Trunking?

Now that you better understand SIP, we can move on to the term "SIP Trunk". A SIP Trunk provides the same service you get from a traditional analog phone line or a channel (trunk) on a PRI. The difference is, instead of being a physical wire, a SIP Trunk is a "virtual" phone line which is provided by a SIP trunk provider. It uses your data circuit (T1, Cable modem, DSL, Ethernet over Copper, Fiber, etc.) to connect your phone system to the Internet.

The End of the Traditional Telephone Network.

A SIP Trunk costs providers considerably less than traditional telephone service. Customers are dumping their old service to save a ton of money.

Meanwhile the traditional phone companies are losing gobs of money on supporting the old network. They've lost over 30% of their customers to wireless and VoiP already and that number grows larger every day. If that's not enough, the FCC is mandating the end of PSTN to speed the acceptance of an all-IP network. Between customers saving money moving to VOIP, phone companies lobbying to stop losing money on old technology and the Federal Government's mandate to improve Internet speeds, traditional phone service has reached its end.

Early Fears about VoiP.

When VoiP was first being implemented, there were frequently issues over call quality. SIP was a new service that was not fully baked and under-engineered. Plus, Internet quality was many times substandard and the bandwidth inadequate. As a result, there were issues.

But that was then, and this is now. Providers like TeleDynamic offer best-in-breed voice and Internet solutions so call quality is clear and reliable. SIP also offers better fail-over options for businesses that have fears about their phone system crashing. SIP also provides a much easier path to move to a hosted PBX at a later date.

Who offers SIP Trunking services?

There are as many choices in SIP trunking as there are in traditional telecom. There are companies that specialize in SIP trunking. Nearly every data and voice carrier also offers the service.

TeleDynamic Communications is the leading SIP trunk provider here in the San Francisco Bay Area.

SIP Saves.

The cost of phone calls for VoiP compared to PRI and analog lines are considerably less, saving your business money on its communications. By putting your telephone system onto a data network you could realize savings up to 50% or more on your monthly phone bill.

Trunks, Call Paths & Lines.

Different parts of the industry use different names, thus the need for this clarification. For the purposes of this document, think of these as different names for the same thing – a single trunk, call path or line supports one phone call with the outside world.

Ready to Learn More?

The remainder of this document will go into substantial detail about SIP trunking. After reading this publication, you should have a very well-rounded knowledge of SIP trunking and how it works for business.

SIP Trunking Features.

Cost Efficiency - SIP Trunks are far more cost effective than traditional circuits. A typical business could save 50% or much more on their monthly communications bill.

Minutes No Longer Matter: Most SIP trunk providers offer unlimited inbound and outbound phone calls.

Sizing Increments: Traditional PRI circuits have 23 voice channels. SIP trunking comes in increments of one channel. Pay only for what you need.

Eliminates Hardware: Most SIP trunking implementations do away with traditional line cards. This lowers costs and eliminates future hardware failures.

Geographic Resilience: Traditional telephone lines (PRI and analog) are directly connected to a single Central Office location near your office. A major physical disaster such as a fire or earthquake can wipe out your communications. SIP trunks are typically designed so the service is redundant at two separate locations to avoid catastrophic failures.

Disaster Recovery Options: If your PBX fails, you have a major power outage or building damage, or your connection to the Central Office goes down, your SIP trunks will still work. You can simply re-route calls to any telephone number in the world.

Choose Phone Numbers from Anywhere: Numbers can be provided from almost any location in North America. This permits enterprises to establish a "virtual" presence with incoming calls ringing at a centralized facility.

Caller ID: Caller ID is a standard feature

DID: DID is a standard feature

Caller Name ID: Caller Name ID is an optional feature

Multi-Media: SIP trunking supports voice and video traffic

Adding & Removing Capacity: You can add or remove a SIP trunk in one business day and usually less. Traditional phone companies take weeks to make these changes.

Geographic Location Independence - Traditional voice is attached to a physical location, but the flexibility of SIP trunking allows call routing to any IP address worldwide, including home offices and soft phones.

What Are My Simultaneous Call Needs?

If you're going to switch to SIP Trunking, you need to determine how many trunks (call paths) you'll need to support your calling needs. Remember that one trunk supports one phone call to the outside world.

Calculating Your SIP Trunking Needs.

When trying to size a SIP trunking solution, many vendors just use an old rule of thumb of one trunk for every three employees.

Another simple method is to duplicate the number of trunks that you are presently using.

Using either of those methods can lead you to the wrong answer. Since PRI circuits come in increments of 23, duplicating that number of SIP trunks is just perpetuating your present configuration, with no consideration as to your true needs. And using the "one trunk per three employees" ratio is using a business average as your basis for calculations, rather than a well-reasoned decision that's right for your organization.

Fortunately, determining the proper quantity of SIP trunks for your business is a pretty straightforward process. First, if you have a receptionist, that person will have a very good idea of the maximum number of simultaneous calls your company experiences in the course of business. If you have a relatively small office, you can also ask the employees how many of them are on the phone at once.

A fact to consider is that most businesses are experiencing less phone calls than in the past, as employees and customers are communicating more on email and mobile phones. You need to ask yourself if that is the case for your business. If so, you are probably currently over-trunked.

Once you feel comfortable that you know the number of simultaneous calls at your peak times, you'll have a good baseline number. Now, just to be safe, increase it by 20% to provide a buffer. Remember, you are saving money by converting to SIP trunking, so the extra trunk costs won't kill you and you sure don't want busy signals.

SIP Trunk Data Circuit Considerations.

Next up, we tackle the question of the type of circuits that are suitable for SIP trunking. Although speed is certainly a consideration and will be given full attention here, there are many other elements that go into determining the suitability of a given Internet connection for SIP trunking.

Background: Old analog and PRI service was reliable, but also very expensive and has tinny voice quality. With VoiP, reliability and voice quality are variable, with a large factor being the type and quality of your Internet service. If you place an extremely high value on voice quality, VoiP service can be engineered to be superior to analog phone calls. That's right, VoiP can provide a clearer voice than the old service. This is possible due to a technology called "Wideband" or "HD Audio". On the other hand, if your business has a very low volume of calls and you are on a budget, an inexpensive data circuit with a bit lower quality may be acceptable.

Synchronous Vs Asynchronous:

Sheesh, you didn't know that there would be so much to know about a simple Internet connection, did you?

Synchronous circuits maintain the same speed in both directions, uploads as well as downloads. Synchronous circuits are typically "business quality" meaning that they have greater reliability, closer monitoring and guaranteed service level agreements (SLA's). Synchronous is best.

Asynchronous circuits have a higher downstream speed (from the Internet to your office), and a considerably lower upload speed. A typical asynchronous circuit has five times more download capacity than upload capacity. If the circuit is used primarily for browsing web sites and downloading information, this speed difference isn't really relevant. However, if the circuit is used for cloud applications, substantial file uploading and voice, the slower upload speed becomes a substantial factor.

Cable and DSL are the two most popular asynchronous circuits. The bottom line is asynchronous circuits have much higher bandwidth and are much less expensive. However, they are less reliable than synchronous circuits. A quality SIP trunk provider should work with you to provide multiple options and help you weigh the pros and cons of each type of circuit. Professional advice is highly recommended in this area as it can make or break your SIP trunking conversion.

Types of Circuits Suitable for SIP Trunking:

Fiber: Fiber has been around for a while, but what's new is that it is now being widely installed in business offices. AT&T alone is spending $6B by 2015 on fiber. Fiber circuits can handle vast amounts of bandwidth. However, fiber is not everywhere and it's expensive to lay new fiber cable. If you are lucky, your building is already "lit".

- Bandwidth: 10MB to over 1GB
- Type: Synchronous, bandwidth guaranteed
- Price*: $599 for 5MB, $1,799 for 100MB
- Lead time: 3 weeks to one year

Ethernet over Copper (EoC): This again is a newer technology. It uses old copper wires and "bonds" them (up to 24 pairs) to increase speed. It is the least expensive business-class synchronous data circuit. This service is not available to businesses located far from a Central Office.

- Bandwidth: 1.5MB to 50MB
- Type: Synchronous, bandwidth guaranteed
- Price*: $249 to $1,799
- Lead time: Three weeks to three months

Cable: Cable companies provide cheap and plentiful bandwidth, so their circuits provide plenty capacity for voice and data for smaller offices. However, cable companies offer their basic business service on a "best efforts" basis, meaning that they don't guarantee quality or reliability. For smaller companies, this "best efforts" approach has been adequate, but expectations need to be managed. It has been our experience that "your mileage may vary" but in general is good enough for voice for smaller offices (less than 15 users). It is recommended to establish QoS on any cable circuit to prioritize voice.

There is a caveat when working with Comcast in the Bay Area. They will sign contracts for locations for which they do not have service. They then attempt to get the landlord to pay the construction costs and try to sell other customers within the building. If they are not successful, they will refuse to deliver service or present the customer with the construction bill. The highest construction bill that we've seen was over $100,000. The customer promptly ripped up the contract.

- Bandwidth: 3MB upload/20MB download - 20MB upload/150MB download.
- Type: Asynchronous, bandwidth – best efforts
- Price*: $39 - $199
- Lead Time: One month to infinity

T1: A single T1 has 1.5MB of bandwidth. T1's are old technology but they are reliable. They can be bonded together to increase bandwidth to 6MB. T1's are rapidly being replaced by newer technology that is less expensive and provides much greater bandwidth. If you have a contract, you'll have to keep your T1 until the contract expires. Until then, you'll have a high quality/low bandwidth reliable connection for your SIP trunking.

- Bandwidth: 1.5MB – 6MB (four T1's bonded)
- Type: Synchronous, guaranteed bandwidth
- Price*: $240 - $970
- Lead time: 3-4 weeks

DSL: The bandwidth required for one uncompressed phone call is 80K. The slowest DSL connection provides 384K upload speed. Simple math would indicate that the circuit would support four simultaneous calls. However, that leaves no room for any data which is impractical. So, if your company has an older slow speed DSL circuit, VoiP will not work.

- Bandwidth: 384K/1.5MB – 1.5MB/6MB
- Type: Asynchronous, bandwidth – best efforts
- Price*: $29 to $99
- Lead time: 2-3 weeks

ADSL2+: There is a relatively new type of DSL called ADSL2+, and it is an improvement over traditional DSL. AT&T (UVerse Max product), Sonic.net and TelePacific are local providers who utilize this technology. ADSL2+ allows for several calls while leaving suitable bandwidth for Internet access. Our experience is that it works well in very small environments. Having said that, DSL is the least suitable choice for VoiP and a hardware QoS device is highly recommended.

- Bandwidth: 1MB/12MB – 3MB/40MB
- Type: Asynchronous, bandwidth – best efforts
- Price*: $29 to $99
- Lead time: 2-3 weeks

Wireless: Wireless Internet circuits are typically not recommended for SIP trunking applications. There are many different types of technologies and frequencies used for wireless data and each produces different results with VoiP. Some of them are susceptible to transmission failures due to inclement weather. With so many variables, it's hard to generically speak to the suitability of wireless Internet for SIP trunking. However, it is possible in some areas to find a wireless provide who has VoiP technology down.

Best Efforts Vs Guaranteed Business Circuits.

DSL and cable are both services that are provided on a "best efforts" basis by the carrier. This means that the carrier will not guarantee the speed or up time. Very rarely do these circuits deliver the speed they is advertised.

While both DSL and cable are improving every day, putting VoiP on a best-efforts circuit in turn gives you "best efforts phone quality". For some businesses that do not heavily rely on their phones to conduct business, DSL or cable is adequate. For critical telephone users, best-efforts probably isn't good enough.

PRICING: All pricing is based upon averages in the SF Bay area on July 2014.

Single Solution or Over the Top (OTT) Provider.

Next up is a discussion about the two different types of SIP trunk vendors and why it's important thatyou know the facts. This is an area where some salespeople put a heavy spin on their superior advantages, but the educated buyer (you, after reading this document!) will be able to "cut through the crap." First a bit in the way of definitions:

Single Solution Provider: A single solution SIP trunk provider delivers both the data circuit and the SIP trunks.

Over The Top (OTT): This is a phrase that got started with data carriers and companies like Netflix who stream movies over the Internet. The carrier provides access to the Internet and the content providerstreams the movie over that circuit. So, two different providers, providing two different services. In the SIP trunking world, an Over the Top (OTT) provider delivers SIP trunking over a separate data carrier's circuit.

So, is there a superior solution? No, just like most things in life, there are benefits and weaknesses to both models....otherwise in a competitive world they wouldn't co-exist. Once again, the right solutionwill be different depending upon your circumstances. Just beware that the single solution locks you down tighter, but oddly that sometimes comforts "C" level folks, especially when it's well advertised brand.

Single Solution Sales Points.

You will get one monthly bill: This is true for most providers but not all providers. Anybody who has AT&T service understands this.

No finger pointing: If it's a circuit or a SIP trunk issue, you only need to contact one company. The old (now) proverbial "one throat to choke".

Bundled pricing: Some providers provide a bundled discount if you buy both the circuit and SIP trunking from them.

Higher Quality: Since everything goes through the same company, VOIP quality is under their complete control with the result being superior quality, versus putting one company's service on top of another company's circuit.

OTT Solution Sales Points.

Best of Breed Providers: DedicatedData carriers do Internet service best. SIP trunk providers specialize in this particular service. No one vendor is superior with both. If they were, the competition wouldevaporate. It's a matter of specialization versus one-size-fits-all. The average phone company today provides MPLS, T1, fiber, email, cloud services, data backup, security and on – they are truly generalists.

Pricing Advantages: Particularly in the area of data circuits, prices and coverage varies widely and you can only find the best provider if you can shop the entire market. The same holds true for SIP trunking providers.

Changing Your Data Carrier: You will have to dump your existing carrier and replace them with a new single solution provider, unless of course, you purchase SIP trunking from your existing provider.

Vendor Lock In: What if you dislike one of the services? Well, you are stuck with them unless you want to make a complete change. In an OTT environment, you just boot the offending service provider, while keeping the service that is working well.

Finger Pointing Addressed: Theoretically working with a single solution provider means that you are working with one cohesive entity. The fact is that services are delivered by different divisions. They may work together seamlessly or they may be a bureaucratic nightmare. For the majority of SIP trunking service issues it is very apparent what service is not working correctly. Finally, all companies wish to provide quality service to retain their customers, it's not the exclusive domain of a company that provides multiple solutions.

Quality Issue Addressed: Single solution providers heartily tout their "total control of the entire network" advantage. And it is true....but only as long as the phone call remains on their network. Andthe last I checked there are hundreds of data providers in the U.S. The chance of you making an outside call that remains on a particular provider's network all the way to the other side is infinitely small.

Resiliency: This is an area where the ability to hand choose Internet providers is a big plus. An all-in-one provider can offer two separate circuits providing redundancy. However, their resiliency capabilities are pretty weak. The circuits are both from the same provider and most times on the sameinfrastructure. With an OTT solution, the ideal situation is to have two different providers and mostimportantly, two different media.

Oh boy, "media", just what you need is another new term. In the networking world, media refers to thephysical conduit down which your voice and data travels. Common media are copper, fiber, wireless and cable. These different media take different geographic paths and require separate cables to reach your office (wireless uses "air" as the media). For maximum resiliency, a company should deploy carriers using two different media. There is no single company that can provide a rock solid resilient networkwhile a multiple-carrier solution inherently provides it.

Summary: Single solution providers are quite appealing to smaller customers who want simplicity and want everything delivered by one company, while OTT solutions provide a more robust, resilient best-of-breed approach favored by larger organizations.

Redundant Voice/Data Circuits.

Redundant Internet Connection.

The old paradigm was to have a dedicated voice circuit and a dedicated data circuit. With the advent of VoiP, the dedicated voice circuit can be eliminated. While that saves money, it also makes a business more vulnerable in the case of a data circuit failure. And that vulnerability grows even greater with the shift to cloud-based applications.

So, this sounds like a bad situation, but there is a happy story in here. Due to the savings of SIP trunking, many customers have been able to install a second data circuit to increase reliability and Internet speed without additional cost. We consider this a triple win:

- Eliminate the costly traditional telephone bill
- Increase Internet bandwidth
- Improve data and voice reliability by installing redundant data circuits

And most times this can be accomplished with little or no increased monthly cost. Sure, the savings from SIP trunking is absorbed by the higher amount spent on data circuits but in one fell swoop, you've updated your entire network to support VoiP, cloud applications and increased Internet speed.

Redundant Circuit Option 1:

In addition to the usual method of running VoiP on your data circuit, there are two alternatives of delivering quality data connections that support VoiP. The first is the simple solution of having two circuits. One that is dedicated to voice and the other dedicated to data. This alleviates from having all critical services going through one circuit. So, much like the world before VoiP, if data fails, voice keeps working and visa versa. This solution allows for a cheap, fast connection such as a cable circuit for data, and a more reliable, albeit slower circuit for voice.

Redundant Circuit Option 2:

The second solution also uses two circuits but adds a device called a dual WAN router. This device will allow the two circuits to auto-failover. So, if your primary data circuit is interrupted, data traffic moves to the voice circuit. And the same backup capabilities exist if the voice circuit has problems. This is a beautiful solution as it actually makes VoiP more reliable than the old phone service it replaced.

In summary, Internet connection speed, while important, is only one of many factors that need to be considered when moving to your new VoiP service. It's a bit of upfront engineering and cost, but the result is lowered monthly costs, improved features and better voice quality. Welcome to the 21st century of telecommunications.

Phone Call Quality Measurement.

So, now that we have a good Internet circuit (or two), it's time to talk about the components that make up VoiP call quality.

"So clear, you can hear a pin drop!"

Are you old enough to remember the very successful tag line for Sprint Communications' long distance service back in the ancient 1990's?

"Can you hear me now?"

Or.... Did you grow up remembering the equally (if not more) successful, phrase that Verizon Wireless made part of our cultural reality in the new millennium?

Regardless of which slogan became part of your lexicon, the fact remains that since the advent of cell phones and VoiP calls, we now regard voice quality in a different light. Today, the most common method to use for judging call quality is to ask; "is it less than, equal to or better than an excellent cell phone call?" Most business users consider a call that is the equal of a good cell phone call acceptable for business purposes.

There is a more scientific method that is used when evaluating VoiP call quality using a scale.

Mean Opinion Scores (MOS).

MOS is a test that has been used for decades in telephony networks to measure the human user's experience of the quality of a phone call.

The MOS is assigned by a group of listeners using the following values:

- **5** - Excellent
- **4** - Good
- **3** - Fair
- **2** - Poor
- **1** – Bad

Codecs & Call Quality.

What's a Codec?

A codec, which stands for coder-decoder, converts an audio signal (your voice) into digital form for transmission (VoiP) and then back into an audio signal for replay. Codecs vary in the bandwidth required per phone call and the quality of the voice transmission.

Common VoiP Codec Protocols.

G.729: G.729 is a codec that substantially compresses voice from 64K to 8K, but still provides decent call quality. It has a MOS rating of 4.0 and is preferred in situations where bandwidth is precious, such as international calling.

G.711: G.711 is a codec that has no compression. It is the default for quality SIP trunk providers and is used anywhere good bandwidth available. G.711 has a MOS rating of 4.2. It is the most common used codec in domestic phone calls.

G.722: G.722 is a high bit rate codec which, because it is of even better quality than the traditional public switched telephone network (PSTN), it can be used for a variety of higher quality speech applications. This standard also requires an adequate amount of bandwidth and usually rates a 5.0 on the MOS scale. This codec is not widely available on SIP trunks as of this writing.

How to Decide?

The codecs that provide the best quality consume the most data bandwidth, thus there is a trade-off that you need to consider. The easiest way is to ascertain whether you want the voice conversation to be:

- Slightly less than the quality of an excellent cell phone call (G.729)
  In crease Internet bandwidth
- Equal to the quality of an analog land-line today (G.711)
- Better than the public switched telephone network for voice critical applications (G.722)

QoS & Call Quality.

QOS.

Equally if not more important than raw bandwidth is quality of service (QoS). QoS is the technology used to prioritize a specific type of data on a data network. In this case, we are talking about prioritizing voice over all other data traffic.

QoS is most valuable in situations where your data circuit is less than optimal or you have high a high volume of usage on your circuit. A good analogy is the modern freeway. If it's 3 AM and there is light traffic, every vehicle in every lane travels unimpeded, just as voice would travel unimpeded on a circuit with little traffic.

However, this all changes as traffic volume increases. QoS is the equivalent of the commuter lane where voice gets special priority. However, in rush hour traffic, even the fast lane slows. QoS is a great tool to ensure high voice quality on a good data circuit, but it doesn't fix a substandard or overworked data circuit.

Do I Need QoS?

QoS is recommended on all but the largest asynchronous circuits. QoS is not required for larger synchronous circuits that aren't filled to capacity. And for locations that have no or very little regular data traffic, there is no need to prioritize voice. You can't place a priority over something that doesn't exist.

How do I get QoS.

QoS can be provided in two ways:

Hardware QoS: QoS can be established by implementing a piece of hardware that prioritizes voice. There is an upfront cost if you don't already have the equipment that can be set up to provide QoS. QoS is typically established in the settings of a router or firewall.

Carrier QoS: The Internet carrier can prioritize voice in its network. A minority of carriers provide this at no charge, with most charging a monthly fee. Carrier-based QoS is somewhat superior to hardware-based QoS, but more importantly, any QoS is quite superior to having no QoS.

This is another area where a quality SIP trunk provider can provide consultation, recommendations and aid in implementing the proper solution.

SIP Trunking on Legacy Phone Systems.

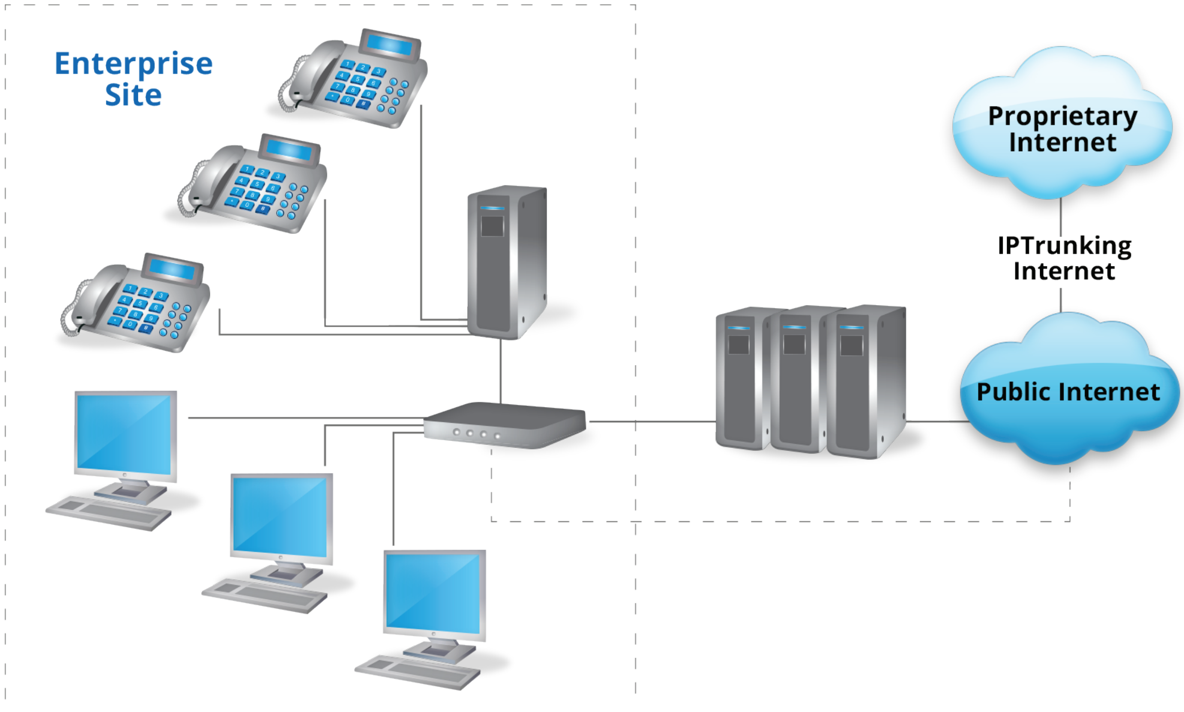
If you interested in the cost saving benefits of SIP Trunking but aren't ready to give up your old PBX just yet, you're in luck. Here's how you can do it.

SIP Trunking on Legacy PBX.

To utilize SIP trunking on a PBX connected to analog lines or a PRI, a gateway is needed to convert the SIP signal coming in from the Internet back into the old school PRI or analog connection.

From the diagram, you can see a voice gateway placed in-between your Internet service provider and your PBX. What's nice about the gateway is that when you're not using the phones, it manages your available band width to increase the speed for your regular data applications when employees aren't using the phones.

Figure 1.0.



3 Benefits to Keep Your Existing Phone System.

- No new training
- No new capital outlay (other than the SIP/Analog Gateway)
- No reprogramming of phones

3 Benefits to Keep Your Existing Phone System.

- Monthly savings of up to 50%
- Choose local phone numbers from anywhere in the US/Canada
- Allow easy migration to Hosted or Premise VoiP phone system when ready
- Easier to provision new lines

Plumbed for the Future.

One of the great benefits of installing SIP trunks at your business today is that you will be fully prepared for the future.

The phone companies, with the full support of the United States Federal Government, have anticipated the shutdown of the old analog communication system for 2020. While this is still a few years away, it shows the inevitable direction towards VoiP and the ultimate demise of analog and PRI circuits.

The phone companies are demonstrating their strategy by steadily raising the price of these circuits, sometimes up to twice per year. The fact is that the vast majority of businesses will have converted to VoiP well in advance of 2020. Why spend a premium to maintain your old analog and PRI circuits, when you could move to current technology now?

Alarm Lines, Fax & Point-of-Sales Devices.

What about my Other Analog Devices?

SIP trunking cannot be used to replace analog lines that are being used for your fax machine, alarm company connection and credit card machine. Fortunately, the Internet provides alternative solutions for these devices also.

Fax.

You have two options here:

#1. You can either replace your fax by using your computers to make and receive faxes. This is called e-fax.

#2. You can install an Ethernet adaptor which converts your faxes into Internet transmissions. You would retain your existing fax machine.

Point-Of-Sale.

Here too, you have options:

#1. Credit card processers now provide Internet-based credit card machines. Just swap out your analog unit for a newer device.

#2. You could consider using a service such as Square which turns your smart phone or tablet into a credit card reader.

#3. Lastly, you have the option of entering credit card information via your credit card processor's payment portal.

Alarm.

You guessed it, you have options here also:

#1. Modern alarm systems have Ethernet connectivity.

#2. Many alarm providers also have a cell phone connection.

Summary.

In addition to converting your existing voice lines to SIP trunking, you can also migrate your other devices away from analog connections. At approximately $40 per analog line per month, you can again save a considerable amount of money on a monthly basis by doing away with them.

SIP Trunking Setup Costs.

Set Up Fees.

This can vary by SIP Trunking provider. Some providers charge a setup fee while others recover their setup costs over time by charging higher monthly fees. If the provider charges a set-up fee, expect fees in the range of $50 to $250.

411 Directory Listing.

When you set up your phone numbers, you're required by law to notify all the phone directory services of the change. This is so that your white and yellow page information is accurate. Your provider should charge between $10 and $50.

Local Number Portability (LNP).

If you wish to keep your old phone numbers when you move to your new SIP trunking provider, you'll need to "LNP" your numbers. A charge per number moved is from $5 to $25, depending upon quantity and vendor. Some vendors will negotiate or waive the cost for large quantities of DID's.

New DIDs.

If you want to establish new DID (direct inward dial) service so everyone in your office has their own telephone, there is a one-time cost to establish those numbers. The typical cost is between .25 to $2.00 per number.

PBX Configuration.

The cost for implementing SIP trunking on your PBX has a very wide range. SIP trunking on a traditional legacy PBX (Nortel, Avaya, NEC, Mitel, etc.) might be expensive, as you'll need to purchase a gateway that converts SIP to PRI. If you have a newer IP PBX such as a Shoretel or Cisco, your cost will be the price of a SIP card or license and installation. If you already own a PBX that natively supports SIP, the cost to configure for SIP trunks is minimal and it will be easy for your vendor.

Ongoing Monthly SIP Trunking Costs.

SIP trunk providers typically offer two versions:

- Unlimited Usage: These trunks cost more, but include unlimited domestic inbound and outbound calls. The range is from $30 to $49 per trunk per month
- Metered Call Path: On a metered trunk you pay for your phone calls on a per minute basis, much as you did with your old phone service. The monthly cost per path ranges from free to $15 per month. The cost per minute is in the range of two cents to five cents per minute.

International Calls.

Rates for international calls are different for each country called. But typically, SIP trunk providers charge 10% to 30% less than a traditional carrier.

Toll Free Numbers.

Providers charge monthly for each toll free number you have. Prices range from $1 to $25 per number per month.

Toll Free Usage.

You will be charged for toll free usage. Rates range from 2-5 cents per minute typically.

911 Service.

Expect to pay about $2 per month for e911 service, plus a fee for each 911 call.

Surcharges, Taxes and Fees.

You can't avoid these surcharges, taxes and fees, even with SIP Trunking. But you can reduce them significantly with SIP trunking because they're based on your monthly charges. Lower monthly charges means less surcharges, taxes and fees.

SIP Trunking Implementation Process.

Local Number Porting (LNP).

Most customers wish to remain some or all of their telephone numbers as they move to SIP trunking. This process is called “porting”. Your new provider contacts the old provider and they work with a neutral third party (NPAC) to coordinate the change.

Telephone Numbers.

You must submit all of the telephone numbers you wish to port to your new carrier. You will only want to port numbers that are published. Unused DID’s or numbers that are part of a hunt group do not need to be ported. Remember, you are paying for every number you port.

Lead Time.

Converting from your old analog or PRI to SIP trunking is a process that takes from 10 days to two months. Moving a few numbers that are on one phone bill is relatively fast. Moving numbers from multiple phone bills or larger numbers of numbers stretches out the time required. Lastly, some losing carriers have put processes in place to purposefully slow the process down, thus keeping the revenue coming in for a bit longer.

Fax, Alarm & Point-of-Sale.

If you want to completely rid yourself of your old telephone company you’ll need to convert these devices to an alternative method of connection, preferably Internet-based. Most people find it quite rewarding to be free of old their telephone company.

Letter of Agency.

You will need to sign a carrier letter of agency that authorizes your new carrier to act on your behalf and place the necessary LNP order.

Copy of Your Phone Bill.

You will be asked to provide a copy of your current telephone bill(s) to your new carrier. That bill can be no more than 30 days old.

PBX Preparation.

On legacy PBXes, a gateway will need to be installed and tested in advance. For IP-based system, the requirements vary: Some will require a card, others an external device called a Session Border Controller (BGP) and some a simple SIP trunking license. For SIP-based PBXes such as the Digium Switchvox and Asterisk, no additional equipment or licensing is required.

Data Network Changes.

Your router and firewall must to be voice-friendly, meaning that they'll reliably pass phone traffic both directions. Depending upon your configuration, you or your data company may have to install a new router or firewall, set up QoS and define the proper routes for voice and data traffic.

If you are going to change circuits, add a new circuit dedicated to SIP trunking or set-up a redundant network, you'll have some technical work to do to establish this new cloud-friendly, VoiP capable network.

New Data Circuit (Optional).

You may need to order a new data circuit as per your SIP trunking provider's recommendations. The lead time for data circuits ranges from 10 business days (DSL) to several months (fiber). After the data circuit is delivered it must be tested. Finally, you data vendor or IT person may have to make some internal network changes.

Porting Process.

The actual porting of phone numbers on the due date is a delightfully quick and smooth process.

Cancelling Your Old Telephone Service.

While porting numbers to the new carrier, many circuits are automatically disconnected. However, most companies don't port all of their numbers over. Make sure you contact your old phone company immediately after the port to confirm that they have or will disconnect all of the unused numbers.

Congratulations!

Your business communications are now in the 21st century.

Summary.

You Completed a Deep Dive.

Congratulations for making it through this document. You probably know more about SIP trunking than 99.99% of the people on this planet.

Is SIP Trunking That Complex?

Like most things in life, what appears to be simple on the surface is in fact, complicated underneath. Yes, there is a lot of technology involved in SIP trunking. But that's no different than any other current technology. Your provider's job is to create a smooth transition.

Why Choosing the Right SIP Trunk Supplier is so important.

Only the exceedingly brave or very knowledgeable customer should tackle a SIP trunk conversion without professional assistance. A quality SIP trunk provider can make the conversion process seem simple and deliver a quality implementation with little or no negative impact upon your business.

The Challenges of National SIP Trunk Providers.

You can search the Internet and find dozens of SIP trunk providers in the U.S. There are also many providers who appear to be U.S.-based but in fact are foreign companies (using SIP trunks so that they have local US phone numbers).

Many of these organizations appear to be much larger and more sophisticated then they really are. Since the industry is relatively new, there is not a lot of background information available so due diligence can be challenging. A spiffy website and shiny marketing materials does not a good SIP trunking company make.

To add to the multitude of choices, every voice and data carrier in our great land presents themselves as SIP trunk providers. However, SIP trunking is just one service amongst the hundreds they offer. Frequently, their marketing and sales materials exceed their capabilities of delivering a quality service. They also require you to use their data circuits, which may not be in your best interest, or if you have any existing data circuit contract, impossible.

Why a Local SIP Trunk Provider is a Better Choice.

Selecting a local provider with a good reputation means that you aren't risking everything on an unknown entity. If the provider has been in the telecommunications business for a long time you can bet they aren't selling a substandard product to their loyal customers. They can't afford the hit to their reputation.

Local providers can also provide a complete turnkey solution, including recommendations and implementation of the best data circuit(s), working directly with your IT resources and they can be on your premise when necessary.

They will typically not be the lowest price vendor – for that you need to choose a web-based provider. But what they will deliver is a quality service at a fair price and as part of their process deliver a quality, cloud-ready, VoiP 21st century data network.

Thank you for taking the time to read this document to learn more about SIP trunking.
If you have any feedback or questions, please don't hesitate to contact me.

we get it

SAY HELLO!

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