What is SIP Trunking?

SIP merges your existing voice and data networks into a single network, delivering high-quality communications, usually with substantial savings. Adding SIP is a great option for businesses currently using non-IP or digital PBX systems because it leverages your existing infrastructure. Moving all – or just some – of your expensive analog lines over to SIP trunks can immediately save as much as 40-50% off your monthly business phone bills.

Why Businesses Choose SIP Trunking

- **Cost savings:** reduce your monthly phone expenses while getting more functionality from your solution; establishing a predictable, pay-as-you-go model that turns CapEx into OpEx.
- **Scalability:** purchase Trunks individually, so that you only buy what you need. Additionally, you can add more Trunks as your business grows. If required, you can also have more DID lines than SIP Trunks.
- **Simplicity:** manage a single, converged network for voice and data. Cut back on the number of providers required to support your business networks.
- **Flexibility:** pave the way for the next generation of voice technology and easily expand into cloud (hosted) or hybrid-cloud services.
- **Continuity:** built-in redundancy provides reliability and high-quality consistent service keeps you connected in the event of an outage, providing you peace of mind.
- **Advanced Functionality and Easy Management:** SIP Trunking offers key features not found in analog systems. For example, you have the option to make unlimited local and long-distance calls. SIP Trunking also supports bursting to manage peak call volumes. Additionally, you can add DID and caller name and number ID to efficiently handle calls.
- **Easily Expand and Transition:** SIP adoption allows for transitioning to next generation technology. It can be simply expanded to hybrid or hosted PBX services. SIP Trunking also makes it easy to scale your line requirements gradually to insure you are not paying for more than your business needs.

Simple Implementation Process

We promise to make it easy to communicate, so getting started is simple. We'll be with you every step of the way, offering support so that you'll never feel like you're on your own.

Once Intuity has a signed order you will be assigned a Project Coordinator (PC) for implementation. The PC will be your contact for implementation questions, training, and information gathering. They will contact you and walk you through the process and make sure you know what is happening along the way.

What Can You Expect?

1. **Configuration Phase**: We start by building your SIP network to your business specifications. Our Project Coordinator will work with you to define how you want your service to be set up. Items like how to set up your DID's, call routing, call queues and call control.

2. **Testing Phase:** Our engineering operations team will access your current network to download a latent voice analysis program to conduct an in-house QoS study (usually 48-72 hours). Once our team has confirmed that your network is SIP compliant, we will then proceed to the implementation process.

3. **Implementation Phase:** Once testing has been performed and accepted, our team will conduct on-site test calls from your network prior to porting numbers to ensure the quality of every call made or received. Our goal is to be 100% certain that your new SIP phone service meets and exceeds your quality expectations.

4. **Support Phase:** As you begin to use your new SIP service often there are questions on features or requests for call routing or programming changes. The Intuity Technical Support team is ready to take your call and help support your business. We conduct a quarterly review of your SIP network and make recommendations related to call quality and traffic usage.

Business Determinations

1. **Capacity:** Capacity is the amount of voice trunks required. You must ask yourself how many people will be on the phone at once, and what amount of blocking (fast busy signal) is permissible? Capacity is usually defined by the percentage of blocked calls during the busiest hour in a given week. P.01 means that one percent of attempted call will be blocked during the busiest hour in a given week. Call volume can also be seasonal with some markets receiving double the call volume during one month out of a year. For retail, this is around the Christmas holidays, for tax preparation businesses, it is around April 15, and for health insurance this begins at the start of the year for new enrollment.

   In order to determine capacity requirements, you have to take into account the type of phone trunks. Inbound toll-free trunks typically have the most requirements, and outbound local calls have the least requirements. The default requirement is P.01 for inbound calls (toll-free and local DID), and P.02 for outbound calls.

2. **Quality:** Voice quality usually depends on each individual caller. For people who have grown used to repeating themselves, it is not as important. On the other hand, if you earn your living by talking to people, you probably require a higher level voice. For those particular people, they probably need the voice quality to be as good as if they were speaking face-to-face.

   Voice codecs and compression is the method of converting human audio to a digital bit stream. This process also minimizes the data bit stream in order to conserve bandwidth and associated costs. There is almost always a correlation between increasing voice compression and lower voice quality. It is up to each particular company to determine how much or how little they compress audio in order to save costs. There are three main types of voice compression:

   - 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.
   - 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.
   - 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.

3. **Security:** In the past, voice networks were often on its own digital infrastructure, not connected to the corporate data network. Therefore, the risk of someone hacking into the voice system was much less than it is today. Unfortunately, today hackers attempt to get into voice systems to eavesdrop for credit card information and to get free long distance. With that said, the value of hacking into a voice system is a lot less when compared to hacking into a web server; typically, web servers have numerous credit card numbers and access to free product. When voice migrates to IP, it is susceptible to all the security required for all of the business's IT applications. Security falls into the following three levels:

   - Highly Secure – Encrypt all media and associated SIP signaling and information
   - Secure – Encrypt all SIP signaling and information, but not the media
   - Standard – No encryption

     Most business' default to secure communication where all of the details about the caller is encrypted except for the media. SIP-TLS is the most common form of SIP signaling and information encryption with SIP Trunking

4. **Features:** Primarily, there are three main types of voice trunks: toll-free, long distance, and local. Each type of voice trunks has its specific set of features that are defined by standard characteristics for that type of trunk, as well as advanced features that can be tailored to meet a business' specific necessities.

Basic features for each of the three Trunking categories are outlined below. Advanced features cost extra, so understanding a business' desire for them is vital. The features of each category of trunks are:

- **Toll-free** – In-bound 800 trunks for customers, partners, and others to call into a business without getting charged long distance fees.
  1. Basic Features – Toll-free numbers, Trunk Groups, DNIS, Call Transferring, and Trunk Allocation
  2. Advanced Features – Caller Entered Digits, Automatic Call Rerouting, UUI info

- **Long Distance** – Calls between LATA's, States, or Countries.
  1. Basic Features – Managed dial plans for on-net and off-net calling, private routing, operator assistance, international calling
  2. Advanced – ANI manipulation, International call authorization

- **Local Trunks** – Calls within a LATA or given metro area.
  1. Basic Features – DID numbers, Directory Listing, Directory Assistance (411), operator services, Emergency (911)
  2. Advanced – Other n11 services such as 711, 311