Trunking for Business: Past, Present and Future

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Telephony has evolved substantially in the last 50 years. Today's telephone would be unrecognizable to Alexander Graham Bell, as voice services have morphed from analog to digital and now to SIP (Session Initiation Protocol). Thanks to SIP Trunking, the modern business can enjoy more options than ever to set up and expand its communications network. Voice over IP (VoIP) and SIP have seriously boosted the capabilities of communications networks, giving organizations multiple options beyond voice. Now it's possible to use the same network circuits for video, messaging and presence applications.

As new technologies such as Software Defined Networking (SDN), Software Defined Wide Area Networks (SD-WAN) and Network Feature Virtualization (NFV) find their way into the market, SIP Trunking is about to deliver even more capabilities. Organizations looking to optimize network performance and add new services now will have deeper insights to determine exactly what changes they must make to meet the needs of employees, customers and guests. With this in mind, companies considering an upgrade of their communications systems should take a close look at the SIP Trunk option.

The Transition to Digital

Over the past 30 years, Trunking from a premise-based Private Branch Exchange (PBX) to the Public Switched Telephone Network (PSTN) has changed significantly – first with Integrated Services Digital Network Primary Rate Interface (ISDN PRI) in 1988 and then with the design (1996) and standardization (1999) of Session Initiation Protocol (SIP). SIP increased flexibility and scalability, surpassing even what ISDN PRI provides today. Prior to 1988, analog services such as Plain Old Telephone Services (POTS) and Analog Trunks connected to Key Systems PBXs were the norm.

ISDN PRI allowed businesses to consolidate networks and increase efficiency and productivity, but it could not overcome some limitations of the underlying network infrastructure (Time Division Multiplexing or TDM). The introduction of Voice over Internet Protocol (VoIP), and specifically SIP Trunking, brought businesses the much-needed increased flexibility, scalability and affordability of Internet Protocol (IP). SIP gives organizations greater network design flexibility, media options besides voice, and scalability to grow based on business needs. It also delivers data insights to help manage the network and the business continuity solutions that ensure inbound calls can always be answered.

Now SIP Trunking is poised to deliver even more functionality, thanks to the introduction of SD-WAN and Session Border Controller as a Service (SBCaaS). SIP will remain a transformative protocol, allowing businesses to leverage video and instant messaging & presence (IM&P) in addition to voice. Organizations will gain even greater insights and control of applications, including voice over their MPLS or broadband networks. SBCs and Voice Gateways will become virtualized and offered "as a service" so customers can pay for what they use rather than having to make capital investments based on peak periods.

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The Best Customer Experiences Start with SIP Trunking

More and more businesses understand delivering a **positive customer experience is the most important factor** in customer satisfaction. Why? Because customers don't want to wait. They expect instant gratification when contacting Sales or Customer Care via a toll-free numbers and/or online chat bots. Customers don't care how you provide these resources, as long as you make everything accessible to them via their preferred methods, such as mobile device, desktop or the web.

The problem is many companies are **struggling to keep pace** with their customers' ever-higher expectations. Upgrading your telecommunications solutions is the best way to address this challenge. That's why businesses are looking at current communications infrastructure and evaluating what new technologies they can quickly implement to ensure long-term viability. However, it's not always easy to understand exactly how or where to start.

Most new communication solutions operate at the software application (cloud) level, rather than the physical hardware (PBX) level. This is compelling companies to turn to SIP Trunking so they can employ advanced communications to better serve their customers. In the following pages, we explain **what** SIP is, **how** it differs from VoIP, and **when** to use traditional PRI lines instead of SIP.

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What Is SIP Trunking and Why Is It Important?

SIP is an application-layer communications protocol for signaling and controlling multimedia communication sessions. SIP Trunking uses a packet switched model to establish connections between ports, and internet connections to link to IP-based phone systems. This is important because SIP delivers multiple modern features beyond POTS that can be an ideal next step for businesses looking to make their voice communications more digital, flexible and smart.

PRI uses a switched circuit to connect to analog or digital phones, but SIP Trunking allows digital packets delivering **digital IP voice**, **video**, **over the internet**, rather than through a TDM line.

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How does SIP Trunking Work?

SIP doesn't provide any services; it works with other protocols, such as RTP (Real Time Transport) to deliver applications like unified communications. SIP Trunking uses distinct fields in order to deliver specific packets to different ports and provides the instructions to manage those digital packets.

Most SIP Trunk service providers set up Quality of Service (QoS) routing using either ports, IP or other applications, which allows them to dependably run high-priority applications and traffic with limited network capacity. QoS technologies provide differentiated handling and capacity allocation to specific flows in network traffic. This enables the network administrator to assign the order in which packets are handled.

For example, perhaps **voice** is the most important application to your organization because conference calls make it possible for a distributed workforce to collaborate. As part of SIP Trunking, your service provider will configure those packets as a priority to ensure audible voice communications even with increased internet traffic. Each application or packet will then be provided an adequate amount of bandwidth.

Working with a SIP Trunk service provider, you also can configure a solution for more advanced SIP services, such as routing calls appropriately during an outage to avoid missed calls. Legacy trunking services have limited business continuity capabilities and may not be able to complete calls during disasters like a power outage. Service providers that deliver PRI over IP vs. TDM, can provide business continuity solutions to reroute the calls. SIP Trunks provide multiple rerouting options to **another endpoint without sacrificing call quality or integrity**.

VoIP vs. SIP: What's the Difference?

VoIP lets you **make and/or receive phone calls over internet protocol (IP)**. SIP Trunking enables calling based on providing instructions through other various protocol communications using VoIP. SIP can transmit information between two or more endpoints, so it's designed for video conferencing, voice calling, and IP-based applications such as unified communications. SIP is also standardized by the Internet Engineering Task Force (IETF), which makes configuring new solutions and endpoints easier.

Bandwidth is a key component when determining the amount of traffic a business expects to traverse their SIP service. As a result, companies that implement a SIP Trunk solution must consider **how they will use their internet capacity**. Organizations also need to anticipate how much data, video, and other real-time applications they will use so they can obtain enough bandwidth and throughput for a positive employee and customer experience. SIP service providers typically allow their customers to scale based on their business requirements.

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What Does SIP Trunking Require?

For SIP Trunking to work, four elements must be in place:

Sufficient Bandwidth

This is determined by your number of concurrent call sessions (CCS), number of total users using the infrastructure, and bandwidth needs for other applications. Since data, video, voice, and unified communications applications may share bandwidth, most businesses need to increase their bandwidth to handle the load. To ensure all packets are delivered, businesses need to audit their bandwidth use.

SIP Protocol Compatible Endpoints

Most modern PBX systems can handle the SIP protocol simply by adding a SIP card. For older PBX boxes incompatible with VoIP, you can implement a VoIP gateway. However, setting up a VoIP gateway can be expensive, so many businesses choose to upgrade their PBX. Most companies are evaluating using headsets or headphones as a softphone to plug into their computers, rather than deploying IP-based desk phones.

Security Configuration

Firewalls must be configured to work with SIP Trunking in order to secure a LAN or WAN environment. In addition, Enterprise Session Border Controllers (eSBCs) provide businesses with additional security for their voice traffic via high performance firewalls which includes features such as:

- Enhanced SIP Traffic Control
- Inherent URL Filtering
- Stateful packet inspection
- Protection against Denial of Service (DOS) attacks

Emergency Calling System

SIP Trunks must be set up with Enhanced E911 for the emergency calling system to work. This means setting up a VoIP emergency caller with a physical location and a designated caller's phone number. Service providers typically offer a 911 solution with automatic location information, which designates a phone number not only to a physical building but also a physical office location.

SIP Trunking Isn't Perfect

If you live in an area with limited Internet or bandwidth options, SIP Trunking may not meet your business needs. Packet loss as a result of QoS routing may negatively impact applications such as HD video.

Some applications experience latency, jitter and packet loss or network availability issues if the network isn't configured correctly. Companies investing in SIP Trunking should not only invest in the PBX but also in desktop devices such as softphones or IP handsets.

Maximize Your SIP Trunking Benefits

Configuration is key to maximizing the benefits of a SIP Trunking solution. Centralized, decentralized, single-point, and hybrid SIP Trunking solutions are available. Different providers may recommend different solutions based on your geography and unique company needs. Leverage your service provider's sales engineers for the best configuration for your business needs.

Enterprise businesses may need inbound, outbound, two-way, single, or multiple trunk groups. With the right trunk group, enterprises can activate business continuity features so they avoid missing a call. Here are major SIP Trunking benefits:

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1. Direct Phone Numbers for All Employees

Direct Outward Dialing (DOD) is an attractive feature for sales teams that make outbound calls, and Direct Inward Dialing (DID) is perfect for receiving inbound customer calls. Both allow employees to have a specific number that integrates their systems into a CRM solution. This helps track the activity of a caller.

2. Caller ID

SIP Trunks enable Caller ID so your users can readily identify who is calling before picking up the phone. SIP can also help you forward calls to a mobile workforce.

3. Event-Based Call Volume

Each industry has cyclical and event-driven call volumes. You need enough capacity to handle peak times but you shouldn't have to pay for additional unnecessary bandwidth the rest of the year. An advanced SIP Trunking solution allows you to burst up to 20 percent over your call capacity, and help you manage unexpected events that drive substantially higher call volumes.

4. Increased Business Continuity Solutions/Failover Capacity

Don't let weather get you down. Many PRI lines lose their direct dial during electrical outages. While a traditional PRI line can only be attributed to one phone number, a SIP Trunk can be rolled over to different sites until the call is picked up so your customers can get through.

5. Total Cost of Ownership

SIP Trunking helps manage costs. You have an opportunity to reduce TCO through subscribing to the appropriate bandwidth, the number of CCS, and consolidating your infrastructure to reduce your business' maintenance and management.

6. Scalable in Small Increments

SIP Trunks don't require you to buy 23 voice paths at once. You can purchase smaller increments to scale as needed. Once your initial configuration is established, adding more trunks is quick and easy.

What Do You Need From Your SIP Trunking Service Provider?

When looking at a service provider, consider working with one that delivers an integrated suite of services to meet your business needs – Internet, SD-WAN, SIP, security, to name a few. This approach can deliver short-term and long-term benefits and help to drive satisfaction among end customers with a long-range communications strategy.

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Specifically, when testing a SIP Trunking solution, work with the provider that operates the last mile connection to your premises. That provider has the best insight into the configuration. The provider can help you forecast future bandwidth needs to support call quality and clarity.

Also consider the following:

- If keeping your existing DID numbers is important to your business, your provider typically can port your current block of phone numbers.
- Map out the network configuration option before installing a SIP Trunk into your production business network to ensure you can easily add more services.
- Don't forget to discuss SIP Trunk remote status monitoring and troubleshooting methods with your service provider.
- Learn what type of advanced and basic services the provider delivers. Customers and employees expect their technology to work trouble-free.

SIP Trunking may not be for every business but failure to look into it could limit your business's future flexibility.

Future of SIP

Although adoption has been slower than expected, there is little doubt SIP Trunking will replace ISDN PRI worldwide. Even as PBX shipments are shrinking worldwide, IP PBX shipments continue to grow. Communication will continue to evolve to embed more and more collaboration-like applications over SIP via WebRTC (Real Time Communication). WebRTC-enabled browsers can be viewed as another endpoint for a SIP Trunking Solution, broadening the communication structure.

As previously mentioned, on-premise solutions will change dramatically with the introduction of SD-WAN and NFVs. Virtualized environments will replace hardware solutions. Leading SBC providers ate taking steps to provide virtualized Enterprise-class SBCs today that can further lower the total cost of ownership of trunking from ISDN to SIP.

It's safe to say the future is now for SIP Trunking. Consulting with a Service Provider will help you determine how to best prepare your business for its journey toward infrastructure and network modernization as a key part of your digital transformation.

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