Make SIP Trunking Work for Your Company

Reduce costs, optimize call quality, and streamline implementation with the right application of SIP trunks.

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From Analog to Digital Telephony

In less than 150 years since Alexander Graham Bell made the first telephone call, the capabilities of this communication method have transformed dramatically.

In the years following Bell's first call in 1876, the build out of copper-wire phone lines made it possible for individuals and businesses to make calls across the U.S., though international calls took several decades longer to become viable.

In fact, while transmission via radio made the first TransAtlantic telephone call possible in 1927, it wasn't until 1956 that the first telephone cable was laid across the Atlantic.

A few years later, in 1963, Bell System introduced the first touch-tone dial phone, but the infrastructure behind telephony in many ways remained largely the same, as it was still based on an analog system of physical cables.

The advent of the internet, however, opened up new possibilities for voice transmission, and in 1974, the first voice call occurred via an IP network.

Voice over IP (VoIP) technology still took a few decades to become more widely available, so during the 1970s, businesses primarily used Private Branch Exchanges (PBX) to be able to communicate internally. Using PBX trunks, they were able to physically connect to the public switched telephone network (PSTN), mainly over T1 lines, to call outside their own networks.

By the 1990s, the growth of the internet and faster speeds made VoIP more viable, and the Session Internet Protocol (SIP) was established in 1999 to transmit all forms of media, including voice, video, and instant messages.

From Analog to Digital Telephony

From there, SIP trunking emerged as a way to digitally connect businesses to the PSTN, rather than needing physical copper lines. Calls today made using SIP trunks are primarily transmitted as packets of data over the internet, which provides a whole host of benefits, which will be discussed in this e-book.

Today, businesses are increasingly modernizing their voice technology to take advantage of VoIP and SIP trunking technology. Doing so can lead to better call quality, lower costs and broader reach, among other benefits that will be covered here. You'll also learn the basics of SIP trunking, how to find the right provider, the basics of how to update your systems, and how to monitor the impact of your upgraded technology.

Key Terms Used in This Ebook

VoIP
The Voice over Internet Protocol (VoIP) enables users to make voice calls over the internet by sending the information as packets of data rather than electrical signals over physical wires.

SIP
The Session Initiation Protocol (SIP) is a set of rules for initiating, maintaining, and terminating calls over VoIP.

SIP Trunking
A virtual connection that allows business phone systems to make and receive calls via the Internet using SIP. In addition to voice, SIP trunks can transmit other media such as video and instant messages.

PSTN
The Public Switched Telephone Network (PSTN) comprises all circuit-switched telephone networks in the world. Think of the PSTN as the operations center linking callers to each other.

PBX
A Private Branch Exchange (PBX) is a private phone network that allows for both internal and external communication. Businesses use PBXs so employees can call one another internally as well as to place and receive calls outside the company. PBX is a rather old construct and most modern PBXs natively enable SIP.

Gateway
A telecommunications element that connects one network to another, e.g., a VoIP gateway enables an IP network to connect to the PSTN. Think of a gateway as the “way out” for a call.

Softswitch
A software switch (softswitch) is a software device that manages call control and signaling so that calls can be digitally linked from one line to another as needed.

Softphone
An application that uses VoIP to make and receive calls over a computer and IP network. Think of a softphone as simply a software version of a phone.

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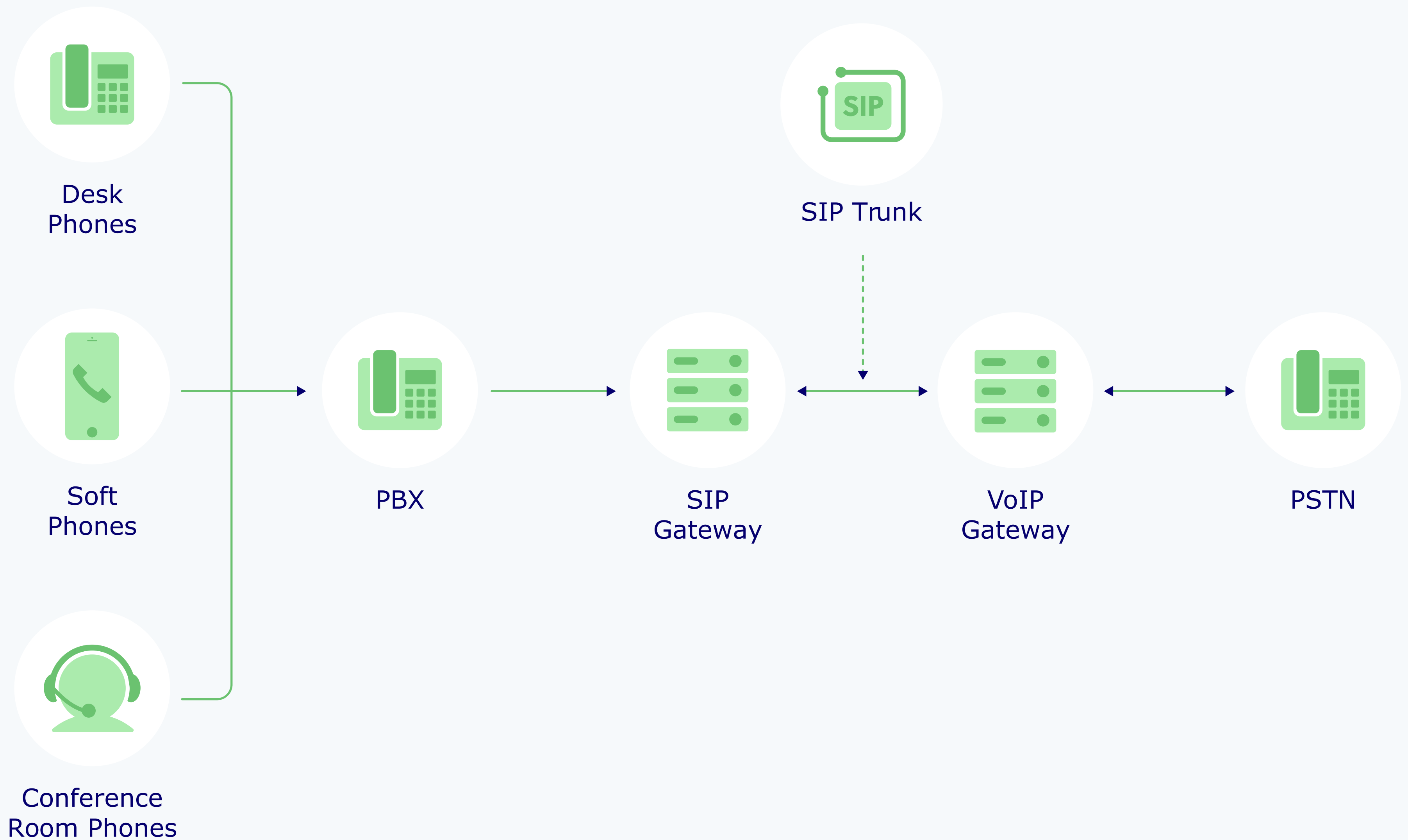
Your Telephony Domain



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STEP 1

Establish a VoIP Foundation

While VoIP and SIP can be confused as competing technologies, they are more so a set of the same standards. Both involve transmitting communications over the internet — VoIP refers to the standard and SIP relates to rules within VoIP.

In other words, VoIP is a set of technologies while SIP is a specific protocol enabling VoIP. SIP trunks are used to virtually connect a company's phone system to other phone networks.

In order to use a SIP trunking solution, you need a VoIP-enabled phone system, such as an IP-PBX (which transmits calls digitally rather than over physical connections like a standard PBX system does) or softswitches, which connect phone lines to each other via software. Most softswitches are IP-PBXs, but IP-PBXs can be hardware-based.

VoIP-enabled systems can either be on-premise solutions, where the hardware is managed by the user on site at their office, or a hosted solution, where a third party provider manages the system on their end and the company receiving the service has cloud-based access.

Both hardware and software offer certain benefits, but unless a company is large enough and has consistent enough phone demands to warrant the upfront price of on-premise technology, a hosted solution is generally cheaper and more flexible to use.

Either way, the phone system ultimately connects to the PSTN, either using legacy PBX trunks (i.e. groups of physical phone lines), or, increasingly, SIP trunks that transmit calls digitally.

STEP 2

Select the Right Provider - Business Considerations

To get the most value out of a SIP trunking solution, consider commercial factors such as:

1. ON DEMAND CAPACITY: Traditional providers typically require their customers to provision call capacity in advance. The challenge is that provisioning just for your current needs makes scaling difficult. The other challenge is over-provisioning, which means you may pay for capacity you never use. Modern, cloud-based SIP trunking providers eliminate the need for capacity planning. You can simply scale up and down seamlessly, depending on your call volumes.

2. GEOGRAPHIC COVERAGE: Some providers focus on narrow geographic regions and scaling becomes complex as it requires working with multiple providers across regions. If your business is global or expanding in that direction, be sure to look for a SIP trunking provider that has the international scale to meet your company's needs.

3. ANALYTICS: Modern SIP providers should offer robust, self-service monitoring tools, such as call detail records to help you analyze usage.

4. PRICING: Traditional providers typically charge for channels, setup fees, number porting as well as the actual calls. Look for "pay-as-you-go" pricing, free account set-up and technical support. It's also a good idea to check for variability in billing intervals across countries.

5. PRICE-QUALITY EQUILIBRIUM: Price-per-call is one attribute you'll want to assess, but beware of trading off cost at the expense of quality. Some providers offer least-cost routing (LCR), which connects calls over whatever carrier has the lowest cost. While that may sound appealing, LCR has some known drawbacks, including audio delays and dropped calls, which may outweigh the cost-savings benefits.

STEP 3

Select the Right Provider - Technical Considerations

To find a provider that appropriately balances cost and quality, consider these technical factors, including:

1. VOICE QUALITY: Look for a provider that works with in-country local voice carriers to consistently deliver high-quality voice transmission and ensure your calls get delivered over a premium network.

2. INTELLIGENT ROUTING: Some providers use an intelligent routing system to measure the quality of calls, then create a quality score to reorder the priority of carriers and connect calls through the highest-quality carrier available.

3. GLOBAL INFRASTRUCTURE: Find a SIP trunking provider that has at least one point of presence (PoP) in each of the regions you want to reach to ensure low latency, call clarity, and no out-of-region audio looping. Focusing on voice quality may not always mean that calls are as cheap as possible, but it’s still practical to have high quality with low costs.

4. RELIABILITY: SIP trunking providers should have multiple carrier connections and redundant infrastructure across regions to ensure that calls can still go through even if one area experiences an outage. To maximize connection reliability, look for carriers with a minimum of 99.95% uptime and interconnections with several Tier 1 carriers in each region.

5. INTEROPERABILITY: Look for a SIP trunking provider that has interoperability with standard softswitches and IP-PBXs, particularly if you’ve already invested in a VoIP-enabled phone system. That way, you don’t have to overhaul your technology to gain the benefits of SIP trunking.

Step 3: Select the Right Provider — Technical Considerations

6. SECURITY: Given that SIP trunking involves transmitting calls over the Internet, you'll want to select a provider that has solid security features baked into their offering. Look for protections like IP authentication and trunk encryption that uses Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP).

7. IMPLEMVVVENTATION: If you're working with the right provider, setting up SIP trunks should be simple and straightforward. Look for providers that offer instant provisioning of phone numbers and access to multiple phone number types such as toll-free, local and mobile.

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STEP 4

Integrate With Your Phone System

Once you've found a provider that has the right mix of commercial and technical attributes, you'll need to integrate their offering with your business phone system, such as by taking the following steps:

1. SET UP VOIP-ENABLED PHONE SYSTEM: If you haven't already done so, install an IP-PBX or softswitch on your operating system or programming environment.

2. REFER TO INTERCONNECTION GUIDES: Once you have your VoIP infrastructure established, check the provider's website for interconnection or interoperability guides such as this one for Plivo and 3XC integration.

3. CREATE AN OUTBOUND TRUNK: To connect your communications infrastructure to landlines and mobile phones, you'll need to create an authenticated outbound trunk that ensures only traffic coming from your infrastructure goes through that trunk. Doing so either involves setting up a username/password authentication or adding the specific IP addresses from which your traffic will come.

4. CREATE AN INBOUND TRUNK: To receive phone calls, simply add the primary Uniform Resource Identifier (URI) and fallback URI of your phone system (the main and backup fully qualified domain name (FQDN) or IP address of your VoIP infrastructure) to your SIP trunking solution. From there, you can assign a phone number to the trunk.

STEP 5

Monitor Usage in Real Time

Look for a provider that offers cloud-based dashboards and analytic tools that make it easy to monitor and optimize the following metrics:

1. ORIGINATION PHONE NUMBERS: Keep track of incoming calls to identify if certain regions need more support and to determine if a specific customer has previously called.

2. DESTINATION PHONE NUMBERS: Monitor destination phone numbers to confirm attempts to reach customers or to identify the volume of calls going to certain regions.

3. START TIME: Monitoring logs to see call start times can help companies identify busy periods of the day and confirm call originations.

4. DURATION: Call duration can also be used to help determine capacity needs, as companies can see the amount of time spent on the phone for different departments or regions.

5. COST: Lastly, monitoring the cost of calls with pay-as-you-go pricing can help you better understand your usage and see whether you're able to qualify for volume-based discounts.

You're On Your Way to SIP Trunking Success!

By following the steps laid out in this e-book, you could gain a significant upgrade over traditional analog systems or even existing VoIP systems. With the right SIP trunking provider, you can:

Save money by having more calls charged at local rates and by only paying for what you use.

Quickly scale your phone service up or down, whether you have seasonal flows or more permanent changes such as office expansions.

Improve call quality by having calls routed based on the best-available carrier, rather than simply the lowest-cost one.

These are just some of the many benefits that come with upgrading to SIP trunking, and it's an easy switch to make.

Zentrunk, by Plivo is a high-quality, global, SIP trunk solution that is simple to launch and easy to scale.

Zentrunk features instant provisioning, pay-as-you-go pricing, and unlimited concurrent calls. Zentrunk plugs easily into any call center software and is interoperable with most standard IP-PBX systems and softswitches.

With global reach and quality-based routing, Plivo meets the highest standards for delivery and scale.

Request a demo to learn more about how Zentrunk can help you save money and improve voice quality at scale.

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