siptrunk™

eBook

The Complete SIP Trunking Handbook: A Primer for Business and Technical Leaders



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Why SIP trunking matters now

The copper sunset

We are in the era of copper infrastructure retirement—the so-called "copper sunset." In its "Voice Telephone Services: Status as of June 30, 2025" report, the FCC counted roughly 15 million switched access lines still in service as of mid-2025, against 63 million interconnected VoIP subscriptions. Switched lines are declining at a compound annual rate near 18%, pushing a large pool of businesses toward IP voice on a timeline they don't control.

Customer migration is underway, and most businesses haven't finished it, creating a strong entry point. You aren't creating demand from nothing—you're intercepting a transition carriers set in motion that many smaller businesses keep deferring.

Voice modernization

SIP trunking offers far more than a forced migration that you can support—it offers a range of real business benefits. Voice system modernization through SIP trunking involves replacing traditional analog phone lines or PRI circuits with internet-based connections while retaining the PBX hardware that routes internal calls. This upgrade enhances how a business connects to the public telephone network without requiring changes to desk phones, internal extensions, or established call-handling procedures. SIP trunks deliver everything a traditional phone line does at a fraction of the cost with vastly more flexibility. That single value proposition unlocks a cascade of secondary benefits, from remote work enablement to unified communications integration.

SIP trunking offers telecom agents, dealers, VARs, distributors, and IT consultants an opportunity to add a recurring-revenue service to their portfolio of offerings. If you're new to SIP trunking, this handbook is your place to start. It covers the basics of SIP trunking—how it works and the value it offers—for business and technical leaders.

SIP trunking 101: What it is and why it exists

What is SIP trunking?

SIP, which stands for session initiation protocol, is a standards-based protocol first created in 1996 and standardized in 1999, that is used for voice and video communications across the internet. It is the default protocol used by the majority of businesses using VoIP.

SIP trunking refers to the packetized SIP media stream that moves across the network transporting your voice call between the PBX and end call destination.

Before SIP, voice calls were carried over the public switched telephone network (PSTN), the copper phone lines we traditionally associate with the "telephone." The PSTN is a physical, circuit switch network that requires a physical connection between two points in order for a call to take place. SIP trunks, on the other hand, are virtual phone lines that allow you to make and receive phone calls over the internet to anyone in the world with a phone number whether they are using VoIP or the PSTN. SIP trunks use a packet switch network,

where your voice calls are broken down into digital packets and routed across a network to its final destination.

SIP isn't just about voice calls. It's the backbone for:

- ·Voice calls over the internet, making it easy to chat without traditional phone lines.
- ·Video conferencing, allowing face-to-face meetings with people around the world.
- ·Multimedia sessions, which can include anything from sharing your screen during a call to playing an online game with friends.

SIP vs. VoIP: What's the difference?

In digital communication, you'll hear two terms: SIP and VoIP. Both are crucial for internet-based calls, but they serve different roles. VoIP is a broad term for the technology that lets you make calls without needing a phone company's traditional lines. With VoIP, when you make a phone call, instead of using a traditional phone line, you're talking through the internet. VoIP is like sending your voice as data over the web. SIP is the manager of VoIP calls. It's a set of rules (protocol) for starting, managing, and ending these internet-based calls. But SIP doesn't stop at voice calls; it also manages video calls, messaging, and other types of multimedia communication. It's like the director of a play, making sure everyone enters and exits the stage at the right time. In short, VoIP is the method of making voice calls over the internet, and SIP is a protocol that can initiate these calls and much more.

The business case for SIP trunking

Cost savings

Traditional phone lines can be expensive, with high local and long-distance fees. SIP trunks eliminate these costs by consolidating voice and data traffic over a single internet connection. Businesses only pay for what they use, making it a more economical choice.

Scalability

With SIP trunking, businesses can easily scale their communication needs. Adding or removing lines is straightforward, unlike traditional phone systems that require significant pre-planning. This flexibility is especially beneficial for seasonal businesses or those experiencing rapid growth.

Flexibility

SIP trunking allows for on-demand adjustments. Companies can add channels, trunks, or minutes as needed, without long-term commitments. This adaptability makes it easier to respond to changing business needs.

Remote and hybrid support

SIP trunking enables employees to make and receive business calls from anywhere with an internet connection, supporting the modern remote workforce while maintaining professional communication capabilities.

Resilience

One of SIP trunking's strongest selling points is its built-in support for business continuity. Automatic call rerouting, whether to mobile devices, remote teams, or backup systems, can keep communication flowing even during outages or network failures.

Unified communications

By combining voice and data networks, SIP trunking supports unified communications features such as instant messaging, video conferencing, presence, screen sharing, and more. This integration streamlines operations and enhances collaboration within the organization.

The SIP trunking architecture

SIP channels

Your communication pipelines: Think of SIP channels as individual lines within a larger pipe. Each channel represents a single conversation, be it a call, video chat, or multimedia message. Multiple SIP channels allow for several conversations to happen at once. This flexibility means that as your business grows, you can add more channels to accommodate more simultaneous calls without laying down new physical lines.

SIP trunks

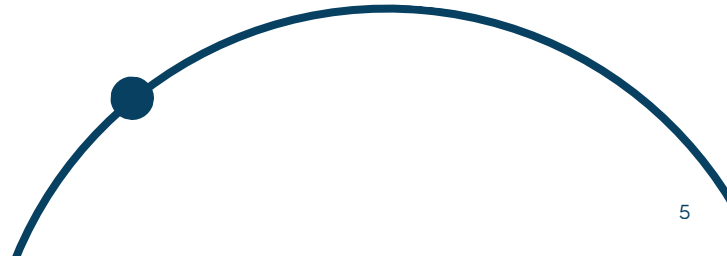
The main pipe: Think of SIP trunks as the main pipe that houses all the individual channels; a single SIP trunk can carry many SIP channels. This is where the scalability of SIP trunking shines. You're not restricted by the number of physical phone lines coming into your building; instead, your limit is based on your internet bandwidth

SIP protocol

The rules of engagement: The SIP protocol is the set of rules that governs how sessions are initiated, conducted, and terminated over the internet. It's like the traffic laws that manage how data "vehicles" move on the "roads" of your internet connection. Whether it's a voice call, video conference, or instant messaging, the SIP protocol ensures these communications are seamlessly set up, conducted, and concluded.

Internet connection

The backbone: None of this would be possible without a stable internet connection; this is the backbone of the architecture. The quality of your SIP service is directly tied to the quality of your internet. Ensuring you have sufficient bandwidth is crucial to maintaining clear, uninterrupted calls and a smooth communication experience.



What happens when you make a SIP call?

High-level

Setup →
SIP starts by sending a message to the person you want to call, asking if they're available.

Bell ringing →
If they are available, their device sends a message back saying "yes," and the call can start.

Talking time →
SIP manages the flow of data, making sure everyone can hear and see each other.

Saying goodbye
When the call is over, SIP sends a message to end the session, and everyone hangs up.

Technical deep-dive

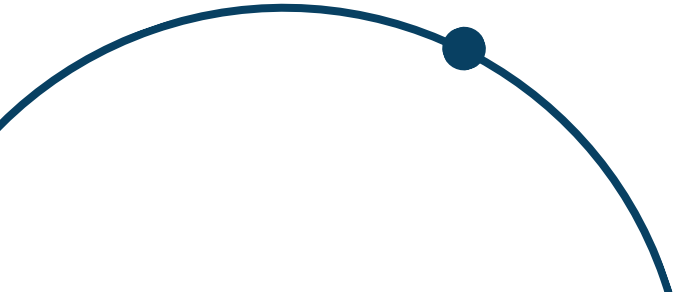
Call signaling sets up the call

When you dial a number, your phone system sends a SIP packet to your carrier. That SIP packet contains all the data necessary to create the call to the person you want to reach. The SIP packet, which is responsible for creating the call, is known as the INVITE. Your carrier uses the INVITE as notification of an intended call and performs a quick LRN (location routing number) lookup, if applicable, to find the number, which you request in the "Request" portion of the SIP packet. Within about half a second your carrier has figured out where your call needs to go and has sent your SIP packet, INVITE and all, to the number you dialed.

When your INVITE reaches the destination carrier, that carrier typically sends back a provisional response (which in SIP is "1xx") which means, "I'm looking for that number on my network, and while I do you won't be billed." When your INVITE arrives at its destination, a few lines of information riding along with your SIP packet, called session description protocol (SDP), make the introduction in terms of how the media or "meat" of the call should be set up, including media ports to use and audio codecs the sender is partial to.

The call is answered

When your call is finally answered, whether by a person or a voicemail or IVR robot, a "200" response is sent back to your system indicating that your call has been received. The response also carries additional SDP parameters that state, "Here's how I'm willing to talk" to finalize negotiations of the call. The calling party sends back an acknowledgement ("ACK") that they've received the "200," to ensure you didn't fall off mid call negotiation.



Media takes over

Once signaling does its job, media flows between the established ports as a series of digital packets. And now it's a question of quality. Phone calls are a real-time communication medium. So, unlike other internet-based connections that also transport information in a series of bite-sized packets, if an audio packet gets dropped because of a temporarily dropped connection or lag, it can't be resent. Resending packets would deliver them out of order, scrambling your syllables. SIP continues to check in every once in a while to make sure the call is still happening. These check-ins are called session timers and they are important because they make sure calls are ended in the event your network experiences any hiccups or if your phone becomes unresponsive causing it to not correctly end calls. They do this by sending a REINVITE at preprogrammed intervals. If the call is still on, the other end sends back "100," to which the caller responds with "200," to which the callee responds with "ACK." If any of those legs don't happen, the SIP tells your phone system the transaction has failed, at which point, your system must send a "BYE" to officially end the call dialogue.

SIP trunk features and capabilities

Numbers and call management

- **DIDs:** Direct inward dialing is a service where a telecom provider assigns a block of unique phone numbers to a business and connects them to the company's phone system, allowing callers to reach specific employees or departments directly. This allows a business to purchase one phone line, but multiple direct numbers could still be routed to specific individuals or voicemails within their organization.
- **Local numbers:** Similar to DID options, SIP trunking allows company employees to work from anywhere in the world and still have a phone number that shows as local to the business's primary address. This feature is beneficial if being perceived as local is a big priority for the business.
- **Toll-free numbers:** SIP trunking supports 800 and other toll-free numbers. This is very helpful for businesses with many customer support needs.
- **Caller ID:** In addition to inbound caller ID, you get a highly customizable outbound caller ID that lets you alter the business ID or individual ID however you want.

Operational and management capabilities

- **CDRs:** Call data records can help businesses understand the results and insights of their communications and shed light on their callers' requests. With SIP, you can get precise, reliable data on the callers and review call history as far back as a year prior.
- **Provisioning and setup:** Modern SIP trunking platforms enable resellers to provision services, port existing numbers, and configure routing rules through intuitive self-service portals rather than waiting for manual carrier processes.
- **QoS optimization:** QoS (quality of service) configuration ensures voice packets receive network priority over less time-sensitive data traffic. When a large file download competes with an active phone call for available bandwidth, proper QoS settings guarantee the call maintains clarity while the download simply takes longer to complete.

- **Interoperability:** Since SIP is now considered an industry standard for voice communications, it's supported by the vast majority of VoIP software and equipment, such as desk phones, IP PBX systems, and gateways for VoIP. With the flexibility to mix and match the software and hardware, you're free to try out what is suitable for your company.

Emergency calling, security, and regulatory protection

- **e911:** e911, or enhanced 911, means that when a user calls out to 911 from a phone that may not have a local number, the calls will by default be routed to an emergency center that is associated with their physical address. This potentially life-saving feature is required by federal law in the US for telecommunication carriers, internet phone services, and businesses that use office phone systems.
- **SBC:** A session border controller is a dedicated hardware device or software application that functions as a specialized firewall for SIP traffic, providing security features such as encryption, topology hiding, and anomaly detection.
- **TLS and SRTP encryption:** These two complementary security protocols work together to fully encrypt a SIP trunk, protecting both call setup details and the actual audio conversation.
- **Toll fraud protection:** Internet toll fraud prevention is something that is critical for SIP trunking services. In essence, this feature is designed to spot suspicious calling patterns and can even suspend callers or international calls, so you avoid fraudulent attacks.
- **STIR/SHAKEN compliance:** STIR/SHAKEN, a security framework designed to stop illegal caller ID spoofing and reduce fraudulent robocalls, is essential as carriers implement caller ID authentication requirements.

Glossary of SIP trunking terms

CDR (call data record): A data log created by a phone system or network switch every time a user makes a call, sends a text, or uses mobile data that tracks metadata like time, duration, and numbers

Copper: The traditional, legacy metal wire infrastructure used to transmit voice and data signals (see also Copper sunset)

Copper sunset: The industry-wide phase-out and retirement of legacy copper wire telephone networks; carriers are shutting down their POTS lines and replacing them with modern fiber, VoIP, and cellular wireless technologies

DID (direct inward dialing): A service where a telecom provider assigns a block of unique phone numbers to a business and connects them to the company's phone system, allowing callers to reach specific employees or departments directly

E911 (enhanced 911): A public safety system, required by law, that automatically sends a caller's physical address or location coordinates to emergency dispatchers when they dial 911

FCC (Federal Communications Commission): The independent US government agency that regulates interstate and international communications

IVR (interactive voice response): An automated phone technology that can greet callers, play pre-recorded menus, let users interact using voice commands or phone keypad clicks, route calls to the correct department, and provide automated self-service information

LRN (location routing number): A unique 10-digit number formatted like a standard phone number that identifies a specific telephone switch

PBX (private branch exchange): A private telephone network used inside a company or organization

POTS (plain old telephone service): The traditional analog voice transmission telephone system that runs over physical copper twisted-pair wires

PRI (primary rate interface): A standard telecommunications technology used to connect a business phone system to the public network

PSTN (public switched telephone network): The global collection of traditional, circuit-switched telephone lines, copper wires, fiber optics, and switching centers that handle landline and voice calls

QoS (quality of service): A set of technologies and mechanisms used to manage and prioritize network traffic; instead of treating all data equally, QoS ensures that critical, high-priority services—such as voice calls and video conferencing—receive guaranteed bandwidth and lower latency during network congestion

RTP (real-time transport protocol): A standard network protocol used to deliver audio and video over IP networks to provides low-latency data transfer for real-time applications like voice calls, video meetings, live streaming, etc.

SBC (session border controller): A dedicated hardware device or software application deployed at the edge of a network to manage, secure, and regulate real-time IP communications like VoIP calls and video, acting as a gatekeeper between internal private systems and external public networks

SDP (session description protocol): A text-based format that describes multimedia communication sessions—such as IP addresses, port numbers, and codecs—so two endpoints know how to exchange audio or video

SIP (session initiation protocol): A standards-based protocol that is used for voice and video communications across the internet; SIP is a part of VoIP and is a specific helper tool that starts, changes, and stops those internet calls

SIP session timers: A keepalive mechanism that periodically refreshes active VoIP calls to ensure network resources are freed and hung calls are terminated if an endpoint crashes or a BYE message is lost

SIP termination: The delivery of an outbound call from the phone system to the public telephone network

SIP trunk: The packetized SIP media stream that moves across the network, transporting a voice call between the PBX and end call destination

SRTP (secure real-time transport protocol) encryption: A specialized security profile for the RTP that adds encryption, message authentication, and replay attack protection to real-time audio and video transmissions

STIR/SHAKEN (Secure Telephony Identity Revisited/ Signature-based Handling of Asserted information using toKENs): A digital signature system for phone calls traveling across VoIP networks to stop illegal caller ID spoofing and reduce fraudulent robocalls

TLS (transport layer security) encryption: A security protocol that encrypts data sent over a computer network to keep information private and safe between a user's device and a web server.

VoIP (voice over IP): A technology that converts voice into digital data packets and sends them through the internet

Why resellers choose SIPTRUNK

The SIPTRUNK platform is designed for businesses that want to build a SIP trunking practice and recurring revenue stream selling SIP trunking services, quickly and painlessly. With SIPTRUNK you'll get access to a simple-to-use SIP trunk solution that makes selling and managing your customers a breeze.

SIPTRUNK offers:

Platform compatibility
SIPTRUNK works with all major IP PBX, IP gateways and open-source PBX projects. No matter the manufacturer you represent or the system deployed, SIPTRUNK is compatible.

No contracts
With SIPTRUNK, neither you nor your customers will need to enter into any long-term contracts or make large up-front investments. Everyone pays for what they need, when they need it.

Expert support
Our industry-leading support center provides access to telco veterans ready to help you win opportunities or resolve technical issues. You and your customers are in the best hands.

Hands-off billing and taxes
We take care of end-customer billing and taxation so you don't need to worry about monthly invoicing or handling complex state, federal, and other phone service taxes.

Automated provisioning
SIPTRUNK features fully automated account creation and service provisioning. It takes less than 10 minutes—start to finish—to have SIP service ready for use.

Transparent commissions
Receive transparent monthly reporting and automated commission payouts directly into your bank account. Every invoice has a detailed breakdown of earnings.

Bring your own bandwidth
Unlike many other SIP trunking providers, with SIPTRUNK you and your clients can use any internet provider you choose. This gives you the flexibility to select the best possible internet service offering.

Toll fraud protection
SIPTRUNK systems are constantly monitoring for unauthorized calling and are set-up to kill any suspicious activity immediately, helping to protect from potential fraud.

Widespread DID coverage
SIPTRUNK has a large database of telephone numbers across rate centers in the US and Canada, plus 40 other countries. This means most DID requests get provisioned immediately.

Quick path to revenue
SIPTRUNK makes it easy to get started. You can sign up, complete your agreements, and be ready to sell without ever handling a piece of paper or writing a check. You'll be earning commissions in no time.

To get started building a profitable SIP trunking practice, visit SIPTRUNK.com/get-started/.

About SIPTRUNK

SIPTRUNK was founded by telecommunications experts with over 50 years of combined SIP trunking experience. We've called on this experience to bring you a platform that will propel you toward rapid success.

With SIPTRUNK you don't have to worry about the major up-front expense of purchasing your own softswitch or the monthly costs associated with hiring customer service and technical professionals. SIPTRUNK provides all of this for you. And, unlike other providers, we take care of end-customer billing and taxation for you, so you don't need to worry about monthly invoicing or handling complex state, federal, and other phone service taxes. Learn more at SIPTRUNK.com.