siptrunk™

eBook

Voice Modernization: The SIP Trunk Migration Playbook



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The voice modernization opportunity

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.

Why legacy PBX systems still dominate

Despite decades of technological advancement, legacy PBX systems remain widespread in organizations of all sizes. These systems work. They reliably handle call routing, voicemail, hunt groups, and auto-attendants. Staff members understand how to use them. The equipment has often been fully depreciated, meaning the ongoing cost of ownership is limited to maintenance and phone line charges.

The challenge is that these systems connect to the outside world through expensive, inflexible infrastructure. Traditional PRI lines require businesses to purchase capacity in fixed increments, typically 23 channels at a time. Organizations pay for this capacity regardless of actual usage. Long-distance and toll charges accumulate on every outbound call. Adding capacity requires scheduling technician visits and waiting for physical line installation.

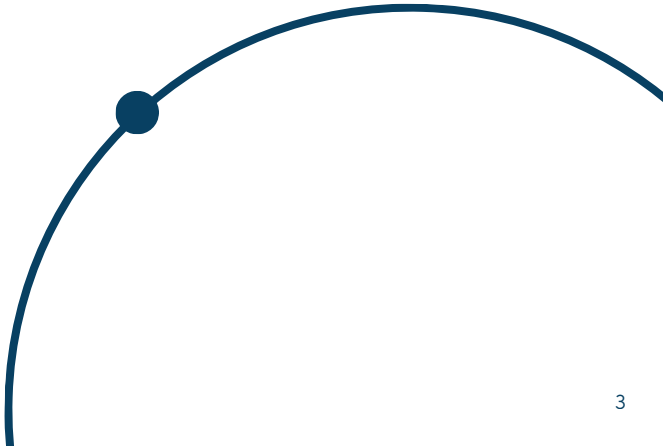
The role of SIP trunking in voice modernization

SIP trunking replaces these physical connections with virtual phone lines that operate over existing internet infrastructure. Voice signals convert to digital packets that travel across IP networks, eliminating the need for dedicated circuit-switched connections. Each SIP channel handles one concurrent call, and businesses can add or remove channels on demand without hardware changes.

Modernization occurs at the point where the PBX connects to the outside world. Internal operations remain unchanged. Employees continue using familiar phones and features while the organization gains access to lower calling rates, flexible capacity scaling, and advanced capabilities like geographic redundancy and disaster recovery routing.

The hybrid option

Hybrid telecom upgrades enable phased migration strategies that minimize disruption to daily operations and eliminate the need for employee retraining.



How SIP trunking integrates with legacy PBX infrastructure

The technical bridge between legacy PBX systems and SIP trunking services involves specific hardware and configuration components. Understanding these elements helps resellers explain the integration process to customers and set accurate expectations about implementation requirements.

Traditional PBX systems communicate using time-division multiplexing (TDM) protocols over T1/PRI lines. SIP trunking operates on session initiation protocol over IP networks. These two technologies speak different languages, requiring translation equipment at the point of connection.

VoIP gateways perform this translation function. These devices connect to the PBX using traditional interfaces (T1/PRI ports or analog FXO/FXS ports) on one side and to the internet via Ethernet on the other. The gateway converts TDM voice signals into SIP packets for outbound calls and reverses the process for inbound calls. From the PBX perspective, the gateway appears to be a standard phone line connection.

Session border controllers (SBCs) add security and interoperability functions to this connection. SBCs manage firewall traversal, encrypt voice traffic, and ensure compatibility between different SIP implementations. Hybrid SBCs combine gateway and security functions in a single device, simplifying deployments and providing a clear migration path when organizations eventually replace their legacy PBX with IP-native systems.

The integration process typically follows a straightforward sequence. The gateway or hybrid SBC connects to the existing PBX using available trunk ports. Network configuration establishes connectivity to the SIP trunking provider. Call routing rules direct traffic through the new connection while maintaining the option to fall back to existing lines during the transition period. Test calls verify audio quality and feature functionality before the organization commits to full migration.



Building the business case for SIP migration

The SIP trunking benefits that drive adoption fall into seven categories. Each one matters, but they compound when stacked, which is why SIP has become the default choice for modernizing business voice infrastructure.



Significant cost reduction

SIP eliminates per-line rental fees, long-distance charges in many markets, and most of the hardware overhead that comes with traditional telephony



Instant scalability

Adding channels takes minutes through a web portal (compared to PRI, which requires a technician visit, a circuit order, and a multi-week wait).



Geographic flexibility

Direct inward dialing (DID) numbers can be issued in markets where a business has no physical presence, supporting local customer experiences without local offices.



Built-in business continuity

Failover routing sends calls to backup numbers, mobile devices, or secondary locations during outages. Some platforms layer dual-network redundancy for additional resilience.



Unified communications integration

SIP trunks feed voice into UCaaS platforms, CRM systems, and contact center software, enabling click-to-dial, automated call logging, and analytics.



Future-proof infrastructure

Copper-based services are being phased out across major markets. SIP positions businesses ahead of those deadlines you aren't left scrambling to catch up.



Modern security and compliance

Encrypted signaling (TLS) and media (SRTP), combined with carrier fraud monitoring, support HIPAA, PCI, and other compliance frameworks.

For resellers, future-proofing carries particular weight. In the UK, for example, the legacy phone network is scheduled to shut down by January 31, 2027, and similar transitions are underway in other markets. In the US, most major carriers are retiring their POTS lines on a rolling, market-by-market schedule, with a general target to exit most copper operations by the end of 2029. Customers who delay this decision are shrinking the window they have to plan for it properly.

When to consider a hybrid approach

The financial case for which SIP trunking integration approach to use rests on measurable reductions in both fixed and variable telecom expenses. Organizations replacing PRI lines with SIP trunks consistently report substantial monthly savings, while the hybrid approach minimizes the up-front investment required to achieve these results.

Replace for lowest cost

Traditional PRI circuits carry high fixed costs regardless of usage patterns. Each PRI delivers 23 voice channels and generally costs several hundred dollars per month, depending on location and carrier, before factoring in long-distance charges and maintenance fees. Organizations needing more concurrent call capacity must purchase additional PRIs in full, even if they only need a few extra channels. Installation fees and service contracts add to the total expense.

SIP trunking restructures these economics. Channel-based pricing allows businesses to purchase exactly the capacity they need, with provider rates varying based on volume commitments and features. Metered options offer lower base costs for organizations with lighter call volumes, charging per minute rather than per channel. International rates drop compared to traditional carrier pricing.

The SIP trunking market reflects these advantages, driven by cost savings between 25% and 65% over legacy PRI lines.[1] Adopting SIP trunking is a straightforward financial decision for most organizations.

Go hybrid to accelerate ROI

The hybrid approach accelerates ROI by eliminating the need for PBX replacement. A VoIP gateway that connects a legacy system to SIP trunking services costs a fraction of the cost of new phone system hardware. Organizations preserve their investment in existing equipment while immediately capturing monthly savings on telecom charges.

Consider an organization currently spending $1,500 monthly on PRI circuits before long-distance charges. Replacing these with appropriately sized SIP trunk capacity can reduce the base connectivity cost, often by 40% or more. Eliminating per-minute long-distance fees pushes total savings even higher. The gateway hardware typically pays for itself within the first year of operation.

For resellers, this straightforward ROI calculation becomes a powerful sales tool. Customers can see exactly how the numbers work and understand that modernization doesn't require abandoning functional equipment. The conversation shifts from "replace your phone system" to "reduce your phone bill while adding capabilities."

[1] Mordor Intelligence, SIP Trunking Market Size & Share Analysis—Growth Trends and Forecast (2026 - 2031)

The migration process

A successful migration is less about technology and more about sequencing. Customers panic when voice service is at stake, so the resellers who win these deals present a calm, phased plan that minimizes risk and lets the customer test the waters before committing fully.

Step 1	Assess the current environment	Inventory existing circuits, contracts, PBX hardware, and call patterns. Identify contract end dates so the migration aligns with natural cancellation windows. Document peak concurrent call counts to right-size SIP channel capacity.
Step 2	Confirm network readiness	SIP voice quality depends on bandwidth and network configuration. Each call uses roughly 85 to 100 Kbps of bandwidth on standard codecs, and you should plan for about 20% overhead for call quality. Verify the customer's internet connection has the headroom, and recommend quality of service (QoS) configuration on the local network to prioritize voice traffic.
Step 3	Pilot with a single location or department	Start small. Port a test set of numbers to SIP, run parallel service for a few weeks, and validate call quality, failover behavior, and integration with existing systems. This step builds customer confidence and surfaces issues before they affect the whole business.
Step 4	Phased number porting	Port DIDs in batches rather than all at once. This separation reduces risk if any individual port hits a snag, and it lets the customer maintain operational continuity throughout the transition. Most carriers handle ports in 7 to 14 business days per batch.
Step 5	Decommission and document	Once all numbers are migrated and the new system has run cleanly for 30 to 60 days, cancel the PRI circuits and remove the legacy hardware. Document the final configuration, failover paths, and support procedures so the customer can confidently manage day-to-day operations.

Don’t stop at the cutover

Delivering ongoing support and account management

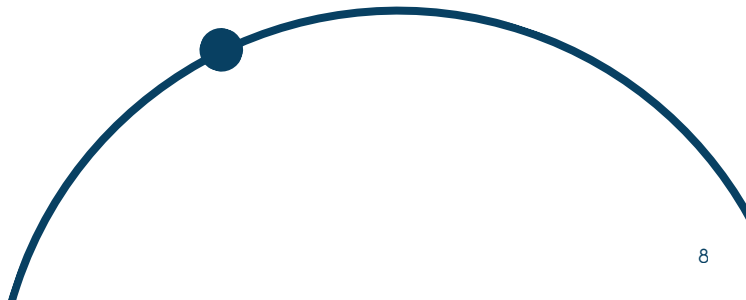
The initial setup is only part of the equation. Long-term success as a SIP trunk provider hinges on how well you support your customers after deployment. Many businesses expect a hands-off experience with their telecom services, but they still appreciate proactive communication, fast issue resolution, and strategic guidance as their needs evolve. Offering reliable post-sale support can help you reduce churn and turn one-time buyers into long-term partners.

Make use of your platform’s built-in analytics, such as call data records and usage trends, to help customers optimize their plans over time. Monitor account performance and flag opportunities for cost savings, service upgrades, or expansion. Even if your back-end provider handles the technical support and billing, your role as the reseller is to remain visible and engaged. Providing this layer of dedicated account management builds trust and drives the kind of loyalty that turns recurring revenue into long-term growth.

Expanding the customer's voice environment

Advanced DID features can transform how businesses manage their communications, making operations smoother and more efficient. Here some key features that can improve business communications:

- **Caller ID management:** Allows businesses to control the number displayed when making outbound calls. This feature is essential for maintaining a professional image and ensuring that calls are recognized and answered.
- **Inbound call routing:** Directs incoming calls to the appropriate department or individual. This ensures that calls are handled efficiently and reduces wait times for customers. When 80% of customers think customer service is just as important as the product or service, ensuring calls reach the right person faster becomes a priority.
- **Call analytics:** Provides insights into call patterns, durations, and volumes. This data can help businesses optimize their communication strategies and improve customer service.
- **Time-based routing:** Directs calls based on the time of day. This feature is particularly useful for businesses with different operating hours or those that offer after-hours support.
- **Geographic routing:** Directs calls based on the caller’s location. This can help businesses provide localized support and improve customer satisfaction.
- **Failover routing:** Ensures calls are redirected to a backup line in case of a failure. This feature is crucial for maintaining communication during outages or technical issues.
- **Number insight:** Provides detailed information about incoming calls, such as the caller’s location, carrier, and line type. This data can help businesses tailor their responses and improve customer interactions.



Looking ahead to the next stage of modernization

The telecommunications industry rarely stands still, but the pace of change in SIP trunking is only accelerating. Current VoIP trends indicate that organizations are embracing intelligent, integrated communication systems. This opens up concrete opportunities for SIP trunking resellers to expand service offerings, deepen customer relationships, and expand sustainable recurring revenue streams. The resellers who recognize these patterns early and adapt their strategies accordingly will find themselves well-positioned as trusted advisors.

The following developments represent the SIP technology future, creating opportunities and challenges for telecom resellers in the coming years:



AI-driven customer experience enhancement: Customers increasingly expect intelligent call handling, automated transcription, and sentiment analysis as standard features rather than premium add-ons. Resellers should evaluate their providers' AI roadmaps and be prepared to demonstrate these capabilities.



Proactive security and compliance positioning: Rather than treating security as a cost center, forward-thinking resellers are positioning compliance expertise as a value driver. The ability to navigate STIR/SHAKEN requirements, implement encryption, and document security measures for audits differentiates premium providers.



UCaaS platform partnerships: Developing expertise in Microsoft Teams Direct Routing, Zoom Phone, and other major platforms opens doors to organizations that have already made UCaaS commitments. These customers need SIP trunking partners who understand their chosen environment.



Flexible pricing models: The market is moving toward consumption-based pricing that aligns costs with actual usage. Resellers who can offer metered, unlimited, and hybrid options accommodate diverse customer needs and capture business that might otherwise go to more flexible competitors.



Disaster recovery and business continuity focus: Organizations are prioritizing communication resilience following disruptions that exposed vulnerabilities in traditional systems. SIP trunking's inherent flexibility enables automatic failover to backup numbers, geographic redundancy, and rapid recovery capabilities that resonate with risk-conscious buyers.

Choosing the right SIP trunking partner

Finding the right partner is a strategic move that can shape your business's long-term success—making it easier to scale, support customers, and build lasting revenue. Your SIP trunking partner should support your business with the right tools, infrastructure, and service model. Since your customers depend on you for reliable, modern communication capabilities, the platform you choose should deliver on performance while minimizing operational complexity.

- **Choose flexible SIP trunk pricing that works for you.** A good provider offers flexible rate plans, like pay-as-you-go and metered bundles, so you can meet a range of customer preferences without overcomplicating your offer. Adaptability is key when selling into diverse markets or accommodating customers with fluctuating needs.

- **Simplify onboarding with instant provisioning and porting.** SIP providers that offer real-time setup, number porting, and instant DID access help you move faster and reduce delays in deployment. Automated provisioning tools can save hours of manual effort and help you scale without a heavy lift.

- **Insist on a reliable, Tier-1 network backbone.** Network reliability is a non-negotiable. Look for providers operating on Tier-1 networks with built-in redundancy and real-time routing. These ensure high-quality calls and help safeguard your customers from outages or voice disruptions that can hurt your reputation.

- **Ensure compatibility with popular PBX systems.** Since your customers may use a variety of PBX systems, broad platform compatibility is essential. The ideal provider will support major IP PBX platforms and SIP trunk connections, allowing seamless integration across different setups.

- **Offload complex tasks like billing and telecom taxes.** With telecom billing and taxes often being a compliance headache, SIP trunk providers that handle this piece for you give you back time and reduce risk. Hands-off billing allows you to focus on growing your business without getting buried in admin work.

- **Find a partner that helps you scale.** Ultimately, the right partner should simplify, not complicate, your business. When a provider offers built-in automation, real-time control, and expert support, you're free to focus on delivering excellent service, expanding your reach, and growing a dependable monthly recurring revenue stream.

Why choose SIPTRUNK as your voice modernization partner

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.