A Guide to SIP Trunking

Since 2002, VoIP Supply has helped over 125,000 people just like you create, deploy and maintain a VoIP solution.
If at any point in this guide you get stuck, are unsure of what is being discussed, or just want to skip the details and have someone select the right open source PBX system for you, then please give us a call at 800-398-8647.

One of our experienced representatives will be more than happy to walk you through the process of selecting the SIP trunking solution system that is right for you.

Thank you for your time and enjoy the read.

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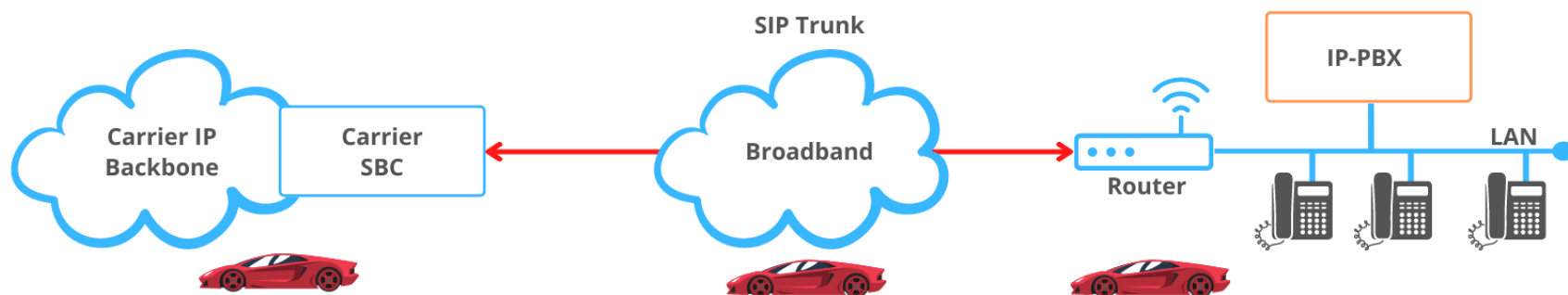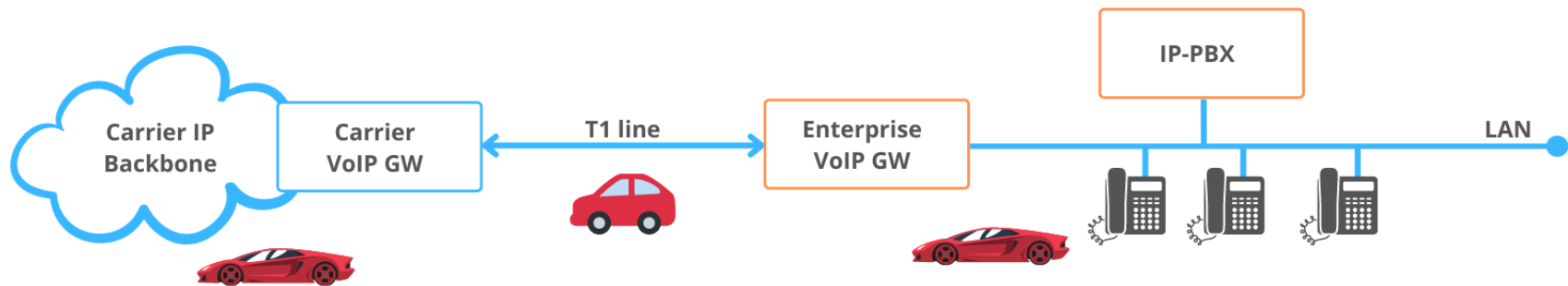
What is a SIP Trunk?

If you're just getting into VoIP, you probably have a lot of questions. We've put together a brief guide to help clarify some of the concepts of SIP trunking, and why it's a realistic option for your business.

A SIP trunk is a method of delivering telephone service through an IP connection. That differs from the traditional method, such as a leased line or analog pair, in that you aren't limited to providers in your local area. With SIP trunking, you are free to choose your provider as long as you can connect with them over the internet or with a private connection.

A SIP trunk refers to two things. SIP (session initiation protocol), which is the protocol used to start and end a call, and a trunk, which is a term adopted from traditional telecommunications. A trunk cable is a cable with a lot of copper pairs that are designed to facilitate many concurrent calls. In the world of VoIP, we use the term "trunk" in a logical sense. Of course, with a physical trunk cable, you have a finite number of concurrent calls, but with SIP trunking, that number is definable and is theoretically only limited to a customer's bandwidth.

In the same way, voltage travels through an analog telephone pair, RTP (Real-Time Transport Protocol), travels across a session in a SIP trunk. RTP is the audio portion of a call, and the quality of that audio has a lot to do with the network it's traversing. A reliable, low latency connection to the internet is critical to the success of a SIP trunk.

Carrier IP Backbone — **Carrier VoIP GW** — **T1 line** — **Enterprise VoIP GW** — **IP-PBX** — **LAN**

SIP Trunk

Carrier IP Backbone — **Carrier SBC** — **Broadband** — **Router** — **IP-PBX** — **LAN**

Why choose SIP Trunking?

If Netflix, Amazon Prime Video, and Google Play have made you question your need for traditional cable service, SIP trunking should make you question the same for your telecommunications needs. You finally have options, and it's a competitive market.

Cost is generally the biggest motivating factor when making a choice, and (mostly) gone are the days of paying per channel on a T1. The fact is the majority of T1 installations nowadays are simply a T1 handoff from a SIP connection to the carrier via a Cisco, or Adtran IAD (integrated access device). Paying by channel in such a scenario doesn't make sense when you could be paying a much lower monthly rate for a greater number of call paths with comparable audio quality with a high-speed, reliable connection.

The greatest benefit to SIP trunking is the level of control you gain. With the majority of services, you can add and remove telephone numbers at will, and make call routing changes on the fly. You can essentially become your own phone company which reduces time and effort when changes are needed.

What are some SIP trunk provider **features** to look for?

SIP trunking is more than just voice, and not all providers are created equally. When shopping around, look for features like these:

Number Portability
This gives you the ability to move telephone numbers to and from your SIP trunk provider.

E911 (Enhanced 911)
Just like the 911, you're used to, but you can define the address for a particular phone number that is automatically provided to the 911 dispatcher.

911 Test Number
Some providers will have a test number you can dial to verify your address without having to call 911.

Disaster Recovery Routing
Routing rules that apply in the event that you lose connectivity to your SIP trunk. You could route calls to another trunk, or another telephone number.

Fax Support (T.38)
Faxing and SIP aren't always in agreement, so you want to make sure a provider supports T.38.

Call Detail Reporting
This gives you insight into your call records on the provider side.

Call Debugging
If you are having issues, self-service debugging can be a huge help in conjunction with support.

Responsive Support
A responsive support team can make a huge difference to your bottom line if inbound or outbound calls are a major component of your business.

API Integration
Some providers will have an API that lets you integrate custom applications with your SIP trunk provider.

SMS Integration
SMS integration can be useful if you need to communicate with your customers without giving out private cell phone numbers.

Will a SIP trunk work with my phone system?

If you have a modern VoIP phone system, the chances are yes. Some systems may have an extra license for a SIP trunk feature, but the majority of systems offer this compatibility as standard. The only way you'll know for sure is to read your phone system's documentation, or contact their support.

If you are already using any of the following, chances are you can take advantage of SIP Trunking

FreePBX	Yeastar	3CX	
Grandstream UCM	Xorcom	Adtran	Sangoma Switchvox

In the event your system is older and doesn't support SIP trunking, it's still possible to take advantage of SIP trunking through a SIP gateway. A SIP gateway sits between the SIP trunk provider and your phone system.

Some recommended brands for a SIP gateway are:

Sangoma	Adtran	Patton	AudioCodes	Grandstream

SANGOMA SIPStation
Your VoIP Connection to the World!

SIPStation is Sangoma's SIP Trunking service providing Canadian and USA Small-to-Medium businesses (SMBs) and large enterprises with feature-rich telephony services using a standard internet connection.

SIPStation SIP Trunks gives you the ability to...

- Choose your own phone numbers and represent your business nationally remotely.
- Send and receive faxes using our integrated FAXStation solution for high latency business critical environments
- Share your voice services throughout multiple locations, reducing your monthly costs
- Add additional lines whenever wherever — no contracts are required! Choose a monthly service or a plan tailed to your business needs
- Create High Volume SIP trunks that are fax and SMS compatible
- Port your own phone numbers to us

KEY BENEFITS OF SIPStation

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INTEGRATED WITH FREEPBX AND PBXACT
The sign-up process is completely integrated with FreePBX and PBXact, which configures your PBX routing automatically

INTEGRATED FAX SERVICES
IP based faxing gives you the ability to send and receive faxes from a web interface or from your local fax machine

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Sangoma SIPStation DID - Regular

These are numbers that people will call to get through to your PBX or individual extensions. It is the cost per North America Inbound Number/DIDs per month.

Sangoma DID Toll Free

The Sangoma DID Toll are phone numbers for your customers to call you toll free.

High Volume Two-Way SIP Trunk 12 Months Plan

The High Volume Two-Way SIP Trunk 12 Months Plan contains 3000 minutes per month per trunk channel. Save more by signing up for the 12 Months Plan today!

High Volume Two-Way SIP Trunk 36 Months Plan

The High Volume Two-Way SIP Trunk 36 Months Plan contains 3000 minutes per trunk channel making it a cost effective solution you should implement for your business today!

High Volume Two-Way SIP Trunk Month-to-Month

The Sangoma High Volume Two-Way SIP Trunk Month-to-Month provides you a cost effective solution containing 3000 minutes per month per trunk channel for those who are not looking to make a commitment.

Thanks again for your time – the VoIP Supply team.