PRODUCT DATA SHEET

SIP TRUNKS

How we work with SIP Trunks

CCXP

Executive Summary

One important concept to understand in telecom today is SIP Trunking.

The traditional public telephone network - PSTN - with its copper network and lines has slowly been replaced with data network-based VoIP and SIP trunking.

SIP is a common connection protocol you can use for VoIP communications. It provides a way to convert voice signals into data that can be decoded by devices used by a caller.

SIP Trunking is where two parties exchange connection information so they can route phone calls between each other.

SIP (Session Initiation Protocol) negotiates connections for data and voice transfers over the internet. A trunk often refers to a bundle of phone lines shared between users. Modern trunking builds on that and refers to a virtual link between a communication device (such as PBX) and the PSTN using an internet connection.

If you are changing from an analogue PBX, you have a choice to make between an on-prem IP PBX or a fully cloud-hosted PBX system.

Evolution of ISDN

An IP PBX needs a way to connect their exchange to the wider PSTN. Traditionally, this was done using circuits (such as T1 or ISDN), otherwise known as Analog Trunking.

Analog Trunking referred to a physical circuit between the phone company's central office and a business. A trunk was composed of individual phone lines (wires), allowing multiple concurrent calls. Calls were made over these phone lines using analog signals.

The introduction of VoIP replaced analog signalling over physical circuits with digital messages, known as **SIP messages**, generally routed over the internet. SIP Trunking is no longer a physical trunk, rather two parties configuring their software or hardware to exchange SIP messages, generally over the internet.

Using this method saves you cost by only paying for only what you use, allows geographic flexibility, with the easy to scale up or down, depending on usage.

Establishing a SIP trunk

To establish a SIP Trunk, the two parties need to exchange Endpoint information, such as the following:

- IP address from which SIP messages will originate, if a static IP address will be used.
    - This is provided so each party can whitelist the IP address of the other party in their firewall
- Username and password credentials (optional)
    - This is generally only needed if messages will be sent from a dynamic IP address.
- IP address and port to which SIP messages should be sent.
    - This is often the same IP address from which messages are sent.
    - If a dynamic IP address will be used, SIP registration is generally required.
    - SIP registration establishes a persistent connection between the parties over which messages can be exchanged.

There are a few decision points to make during this process:

- Which protocol to use - UDP or TCP If a secure (encrypted) connection - TLS/SRTP - will be used
- If health checks, known as SIP OPTIONS, should be sent
- Which CODEC to use for audio encoding - in the U.S., this is often G.711U

Each party enters the information in their software/hardware and that is it! A virtual SIP Trunk is now configured.

Establishing a SIP Trunk does not generally produce constant communication between the two parties, unless SIP registration or health checks are used. It simply means the two parties can send and receive messages.

Differences between SIP Trunks and VoIP phones

VoIP makes and receives phone calls over the internet or internal networks. SIP is an application layer protocol used to make, manage and terminate multimedia sessions, including voice, video and messaging. While SIP is an initiation in its own right, it is mainly used to support VoIP calls.

The same information to configure a SIP Trunk is used to configure a VoIP phone, since a VoIP phone also sends and receives SIP messages. Each VoIP phone can be thought of as establishing a SIP Trunk.

Although configuring a VoIP phone is similar, a SIP Trunk usually implies connecting two platforms/services that will support multiple concurrent calls, rather than a desktop phone. Surprisingly, establishing a connection to a physical VoIP phone requires more information and configuration than establishing a SIP Trunk between two servers.

Testing a SIP Trunk

After the SIP Trunk is configured, we will want to make a phone call to see if the SIP messages make it through firewalls. To test this, we will dial a phone number and have the other party answer.

Each party will speak and verify the other party can hear them. Once confirmed, we will hang-up and verify the other side hangs up automatically. This test is repeated with the other party initiating the call and hanging up.

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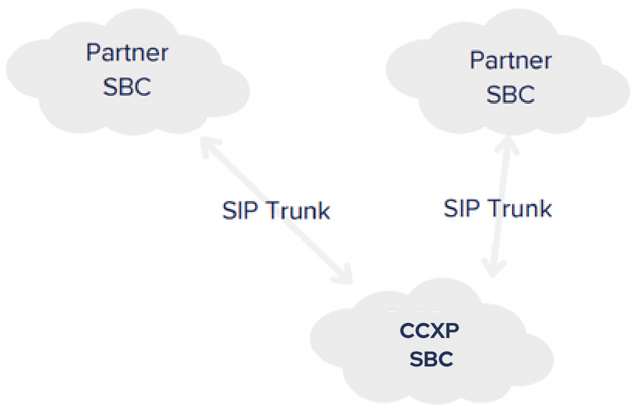
SIP Connectivity

A normal inbound/origination flow will look similar to the following:



Inbound SIP Trunks:

- Multiple Partner SBCs optional SIP OPTIONS may be used to heartbeat
- Available CODECs: PCMU, PCMA, OPUS, G722, G729, H264, VP8, VP9

SBC:

- Single public IP address
- Redundant internally

A normal outbound/termination flow will look the same as the above.

Outbound SIP Trunks:

- Multiple Partner SBCs optional
    - Round Robin Primary - Secondary (secondary used on timeout/primary error)
    - SIP OPTIONS may be used to heartbeat
- Available CODECs: PCMU, PCMA, OPUS, G722, G729, H264, VP8, VP9

SBC:

- All CCXP SIP Traffic will originate from single public IP address
- Redundant internally

Multiple Trunks

If you'd like to setup multiple SIP Trunks, then we can configure those as a Trunk Group. We can send calls to your preferred location first, then send to the second location in the case of a failure.

If you have multiple purpose-based trunks, such as multiple carrier connections or existing premise equipment, the platform must determine which trunk to use when sending SIP messages during outbound calls. Although one trunk is selected by default, an alternate trunk can be selected by matching a dialing pattern.

An example of this would be if 4-digit numbers for extension dialing is the premise trunk, while 10-digit domestic dialing selects one carrier trunk and international dialing selects another carrier trunk.

Inbound calls from all trunks are always accepted in these scenerios, as dialing patterns and trunk groups are outbound dialing concepts.

Getting started

For more information on setting up a SIP trunk, reach out to your sales representative or account manager to get started.