Speakeasy SIP Trunking

Service Guide

Cost-effective voice delivery with IP benefits

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Product Overview

WHAT IS SPEAKEASY SIP TRUNKING?

SIP Trunking is a dial tone service that is delivered to a premises-based IP PBX by utilizing VoIP technology. SIP (Session Initiation Protocol) is a signaling method for establishing multimedia sessions over an IP network. Compatible IP PBXs register directly against Speakeasy's voice platform via a provided username and password.

Voice traffic is transmitted to and from a customer's location over an existing broadband connection. Voice and data services share a single dynamic connection. Bandwidth utilization is optimized because bandwidth remains available for data when it is not being used for calls. No IAD is required because no conversions of the IP traffic are made between the IP PBX and Speakeasy.

The **key benefits** of Speakeasy SIP Trunking include:

- Substantial cost savings over traditional dedicated voice services
- Low long distance rates – choice of calling plans, rates as low as $0.019/minute
- Unlimited local calling, caller name and number included at no additional cost
- Optimal broadband utilization – when phones are idle, all bandwidth is available for data
- Standard features such as e911 emergency calling, DIDs, toll free numbers, operator services, directory listings and local number portability

Advanced hosted voice / IP-Centrex functionality, such as hunt groups, auto attendants, and audio conferencing, as well as powerful productivity and mobility features like Remote Office and Find Me / Follow Me for PBX users.

BASE SERVICES

Below are the basic service options for Speakeasy SIP Trunking. All services include unlimited local calling.

National and international shared long distance minute plans are available. Unlimited long distance packages, including Canada, are also available.

Product Name	Description
SIP Trunk	» Individual trunk for direct SIP delivery » Minimum 2 trunk purchase per location
Telephone Number (DID)	» Individual DIDs may be purchased in any quantity
Enhanced PBX User	» Specific PBX user with enhanced hosted features, such as Remote Office, Simultaneous Ring, and Voice Communication Toolbar

STANDARD FEATURES

The following standard features are included with any SIP Trunking delivery.

Features	Description
Basic Line Hunting	Series Hunting enabled as needed. Each line is attempted in a predefined order starting with the dialed number; calls do not loop. Call Waiting must be disabled.
Call Logs	Call logs are available through administrative portal and through some applications.
Call Waiting	Notifies call recipient of a second call while a call is already in progress. Allows switching between calls.
Inbound Caller Name Delivery	Caller Name will be sent to the IP PBX as provided to the Speakeasy network.
Inbound Caller Number Delivery	Caller Number will be sent to the IP PBX as provided to the Speakeasy network.
Outbound Caller Name Delivery	Caller Name will be sent from the Speakeasy network to the called number. The default value is the Company Name.
Outbound Caller Number Delivery	Caller Number will be sent from the Speakeasy network to the called number as provided by the IP PBX.

OPTIONAL ADD-ON FEATURES

Feature Name	Description
Account Codes	›› Tracking of calls by dialed code ›› Verified and non-verified account codes supported
Audio Conferencing	Company audio conferencing services
Auto Attendant	Automated receptionist for inbound call routing
Directory Listing	White Page Directory Listings for company name, address, and telephone number
Forwarding Numbers	Simple forwarding number service from one DID to another, on or off-net
Growth Reserved Number	›› Spare telephone numbers allocated for future growth ›› May be purchased in any quantity
Hunt Group	›› Advanced incoming call hunting ›› Sequential, Circular, Uniform and Simultaneous hunting policies supported ›› Call forwarding on busy or by time schedule ›› Optional hosted voicemail box included
Local Number Portability	LNP service is available for porting numbers from another provider to Speakeasy within the same local rate center.
Toll Free Numbers	›› Inbound-only toll free service for 800, 866, 877 and 888 numbers ›› Vanity toll free numbers are available
Voicemail Box	Hosted voicemail box, does not integrate with customer PBX
Voicemail Portal	›› Hosted voicemail system ›› Required if hosted voicemail boxes are needed

CALLING FEATURES

Calling Feature	Description
Directory Assistance	Local directory assistance service reachable by dialing 411 or (area code) 555-1212
Enhanced 911 (E911)	Speakeasy provides 911 routing to the appropriate local emergency dispatch center. 911 services with Speakeasy may operate differently than traditional 911 services. For additional details, please refer to the terms and conditions at http://www.speakeasy.net/tos/msa.php#B4.
Inbound	Inbound calling allows for receiving calls from the PSTN or other Speakeasy users. Calls must be directed to Speakeasy-provided number or numbers ported to the Speakeasy network.
Local Outbound	Outbound calling is included with each Business Line or Trunk to the customer's local calling area. Calls must originate from a Speakeasy-provided number or numbers ported to the Speakeasy network.
Long Distance (Domestic and International)	Long distance calls may be made to any destination in the world. Unlimited domestic calling packages are available, as well as domestic and international shared minute plans.
Operated Assisted Dialing	Customers may dial "0" to reach an operator for assistance with calling cards, credit card, third party and collect dialing
Toll Free Inbound	Toll Free Inbound service is available if toll free numbers are acquired from Speakeasy. Minute plans are available.

Supported Specifications

Please note: For more detail on supported specifications, please refer to the comprehensive Speakeasy SIP Specifications documentation.

RFC SUPPORT

Below is the list of RFCs that are supported at the Speakeasy SIP Trunking Interface.

RFC	Description
1889	RTP: A Transport Protocol for Real-Time Applications
2327	SDP: Session Description Protocol
2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
3261	SIP: Session Initiation Protocol
3262	Reliability of Provisional Responses in Session Initiation Protocol
3263	Session Initiation Protocol (SIP): Locating SIP Servers
3264	An Offer/Answer Model with Session Description Protocol
3326	The Reason Header Field for the Session Initiation Protocol
3824	Using E.164 numbers with the Session Initiation Protocol
3891	The Session Initiation Protocol (SIP) "Replaces" Header
3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
3986	Uniform Resource Identifier (URI): Generic Syntax
4028	Session Timers in the Session Initiation Protocol

SIGNALING AND ROUTING

Supported Protocols
SIP is the only supported signaling protocol per RFC 3261.

Transport Methods
Speakeasy only supports unencrypted UDP for transport of SIP.

SIP Methods
The following SIP Methods are supported.

SIP Method	RFC	Receive	Send
INVITE	3261	Y	Y
ACK	3261	Y	Y
CANCEL	3261	Y	Y

SIP Method (continued)	RFC	Receive	Send
BYE	3261	Y	Y
REGISTER	3261	Y	N
PRACK	3262	Y	Y
NOTIFY	3265	N	Y

SIP Headers
The following SIP Headers are supported.

Header	RFC	Receive	Send
Accept	3261	Y	Y
Allow	3261	Y	Y
Authorization	3261	Y	N
Call-ID	3261	Y	Y
Call-Info	3261	Y	N
Contact	3261	Y	Y
Content-Length	3261	Y	Y
Content-Type	3261	Y	Y
CSeq	3261	Y	Y
Diversion	draft-levy-sip-diversion-08	Y	Y
Expires	3261	Y	Y
From	3261	Y	Y
Max-Forwards	3261	Y	Y
Reason	3326	Y	Y
Record-Route	3261	Y	Y
Reply-To	3261	Y	N
Require	3261	Y	Y
Retry-After	3261	Y	Y
Request	3261	Y	Y
Route	3261	Y	Y
Server	3261	Y	N
Subject	3261	Y	N
Supported	3261	Y	Y
Timestamp	3261	Y	N
To	3261	Y	Y

Header (continued)	RFC	Receive	Send
Unsupported	3261	Y	Y
User-Agent	3261	Y	Y
Via	3261	Y	Y
Warning	3261	Y	N
WWW-Authenticate	3261	Y	N

SIP Response
The following SIP Responses are supported.

1xx Provisional Responses	**RFC**	**Receive**	**Send**
100 Trying	3261	Y	Y
180 Ringing	3261	Y	Y
183 Session Progress	3261	Y	Y
2xx Successful Responses	**RFC**	**Receive**	**Send**
200 OK	3261	Y	Y
4xx Request Failure Responses	**RFC**	**Receive**	**Send**
400 Bad Request	3261	Y	Y
401 Unauthorized	3261	N	Y
403 Forbidden	3261	Y	Y
404 Not Found	3261	Y	Y
408 Request Timeout	3261	Y	Y
480 Temporarily Unavailable	3261	Y	Y
482 Loop Detected	3261	Y	Y
486 Busy Here	3261	Y	Y
487 Request Terminated	3261	Y	Y
5xx Server Failure Responses	**RFC**	**Receive**	**Send**
500 Server Internal Error	3261	Y	Y
503 Service Unavailable	3261	Y	Y
6xx Global Failures 6xx	**RFC**	**Receive**	**Send**
600 Busy Everywhere	3261	Y	Y
603 Decline	3261	Y	Y
604 Does Not Exist Anywhere	3261	Y	Y
606 Not Acceptable	3261	Y	Y

MEDIA (SDP AND RTP)

Codecs

The following audio codecs are supported:

- G.711 (PCMU), SDP Payload type of 0, and ptime size of 20ms
- G.729a, SDP Payload type of 8

SDP Support

The following SDP attributes are supported:

SDP Attribute	Support
v= (protocol version)	Required (Sent/Received)
o= (owner/creator and session identifier)	Required (Sent/Received)
s= (session name)	Required (Sent/Received)
i=* (session information)	Optional (Received)
u=* (URI of description)	Optional (Received)
e=* (email address)	Optional (Received)
p=* (phone number)	Optional (Received)
c=* (connection information - not required if included in all media sections)	Optional (Received)
b=* (bandwidth information)	Optional (Received)
One or more time descriptions	Required (Sent/Received)
z=* (time zone adjustments)	Optional (Received)
k=* (encryption key)	Optional (Received)
a=* (zero or more session attribute lines)	Optional (Received)
One or More media descriptions (See Below)	Required (Sent/Received)
Time Description	
t= (time the session is active)	Required (Sent/Received)
r=* (zero or more repeat times)	Optional (Received)
Media Description	
m= (media name and transport address)	Required (Sent/Received)
i=* (media title)	Optional (Received)
c=* (connection information - optional if included at session-level)	Optional (Received)
b=* (zero or more bandwidth information lines)	Optional (Received)
k=* (encryption key)	Optional (Received)
a=* (zero or more media attribute lines)	Optional (Received)

DTMF
The customer IP PBX must support RFC 2833 style DTMF Tones (dynamic payload type of 101).

Fax/Modem
All faxes must be sent using G.711. If the call was set up using another codec, the customer IP PBX must send a reINVITE as described in RFC 3261.

RTP
The Speakeasy Media Interface only supports RTP in a symmetric unicast manner with packetization rate of 20ms. The media interface uses UDP ports in the range from 49152 to 65535 for receiving RTP and does not support encryption.

Customer Support Process

INSTALLATION SERVICES

Speakeasy will oversee the installation of your SIP Trunking services to ensure smooth transition and turn-up. An Installation Project Coordinator (IPC) will be assigned to your order to manage the schedule and address any concerns you may have. The guidelines below describe the expectations for each voice installation.

Speakeasy Responsibility
Speakeasy will provision the voice services and provide guidance to the customer on configuring their equipment. Reference material in the form of setup guides and SIP Specifications can be provided.

Customer Responsibility
The customer is responsible for all setup, installation, and configuration of the IP PBX and network equipment.

ONGOING CUSTOMER SUPPORT

If you should experience any issues with your SIP Trunking or Speakeasy-provided broadband services, Speakeasy Customer Support services are available around the clock to assist you. Speakeasy can help with any network-related issues, but cannot provide configuration and troubleshooting support for your IP PBX.

Speakeasy is committed to providing quality SIP messaging and media from our voice platform. The demarcation for the SIP Trunking service is Speakeasy's Session Border Controller. For Speakeasy-provided data services, the demarcation is the connectivity router.

If it is determined that a trouble issues resides beyond our demarcation point, it is your responsibility to provide a technician capable of troubleshooting your phone equipment.

To reach Speakeasy Customer Support:

- Select the "Create New Service Ticket" option from the Customer Service center on your MySpeakeasy portal
- OR, call the Customer Support Center at (800) 556-5829

LET US SHOW YOU MORE

Call Speakeasy at 800.556.5829 for complete analysis of your telephony and data needs.

Or learn more online at **www.speakeasy.net/business/iv**

Nationwide	**Partner Program**
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