Bandwidth.com

OUTBOUND-ONLY SIP TRUNK

Solution Overview & Interface Specification

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Section 1

Outbound-Only SIP Trunk Introduction

What is an Outbound-Only SIP Trunk?

Introduction to the solution's features, components, and underlying technology

What is an Outbound-Only SIP Trunk and what are the limitations?

Overview

Bandwidth.com's Outbound-Only SIP Trunk service delivers virtual voice channels over an Internet connection via SIP. Bandwidth.com provides outbound calling over these logical voice channels, serving as a gateway to the PSTN (Public Switched Telephone Network) – allowing a customer to send traffic to (origination) the PSTN or other peered IP networks.

Limitations

Outbound-Only SIP Trunks only provide outbound calling. A number of limitations are inherent in this service including:

- No support for any type of inbound call including Toll-free/800 inbound calls
- No 911 service
- No 411 or any operator services
- No outbound calls to any of the following types of numbers: 976, 900, 888, 877, 866, 800, 700, 1010xxx
- No local calling rates are available
- No DIDs or 800 numbers are available for use with Outbound-Only SIP Trunks
- Outbound caller location may be limited

! IMPORTANT !

Customer is responsible for clearly communicating to all end-users that Outbound-Only SIP Trunks do not provide any 911 service

.

What types of PBX's are supported?

Bandwidth.com's Outbound-Only SIP Trunk solution can be supported over both an IP (SIP) –based Private Branch Exchange (PBX) or a traditional Time Division Multiplexing (TDM) based PBX.

IP-Based PBX

- ➨ A basic requirement for any IP-PBX is that it must support SIP

> **! IMPORTANT !**
>
> **To determine if your IP-PBX is compliant with our service, you MUST refer to the "SUPPORTED SPECIFICATIONS" and "UNSUPPORTED SPECIFICATIONS" sections of this document.**
>
> **These sections will outline the specific features or RFCs (Request for Comments) that are supported and the manner in which they must be implemented on your IP-PBX.**

TDM-Based PBX

- ➨ Outbound-Only SIP Trunks can be utilized across a wide range of TDM-based PBXs
- ➨ To support these PBXs, Bandwidth.com will configure and ship an Integrated Access Device (IAD) to convert the SIP protocol signaling to traditional TDM signaling. This device may also serve as the edge router for your network and will connect directly to your PBX
- ➨ Bandwidth.com can deploy an IAD to support the following types of interfaces for your TDM-based PBX:
  - ☑ Analog
  - ☑ PRI

What is SIP?

Outbound-Only SIP Trunks leverages the Session Initiation Protocol (SIP) as the signaling standard for VoIP calls. SIP is an open industry standard built on top of established Internet standards such as TCP and DNS. SIP allows for customers to establish a single, pure IP connection to the wider VoIP and PSTN networks, with voice simply running as another application over a customer's IP/Internet connection.

All of the call control and signaling (e.g. call setup, call teardown) for your voice calls that traverse your SIP Trunks will be conducted using SIP.

The actual voice traffic, or media, will be transmitted via a separate protocol, Real-Time Transport Protocol (RTP).

Section
2
Component & Feature Overview

What components and features are provided?

Outlines the components that comprise an Outbound-Only SIP Trunk and the features provided with the solution

Components of the Service

Bandwidth.com's Outbound-Only SIP Trunk service is comprised of **2** basic components

1. **Virtual Trunks:** The essential component of the service is the trunk. Analogous to a channel in the traditional TDM world, SIP Trunks are simply a virtual voice channel that allows for a single, concurrent outbound voice call. For example, if a customer's application required the support of 12 concurrent calls, 12 SIP Trunks would be required.

   Trunks inherently provide Long Distance (LD) Termination only. As previously outlined, no inbound calling, 411 / Operator service or 911 service is available.

2. **Integrated Access Device (IAD):** For customers utilizing TDM-based PBXs, Bandwidth.com will provide an integrated Access Device which will:

   - Serve as the edge router for the customer's network, terminating their Dedicated Internet Access circuit(s) (optional)
   - Convert the VoIP, SIP-based traffic to TDM. Bandwidth.com will pre-configure the customer's IAD and ship the device in a plug-and-play fashion
   - Provide Quality of Service (QoS) to ensure voice traffic is appropriately prioritized above other, lower-priority traffic types
   - Provide a firewall service; if required
   - IADs can support the following types of PBX interfaces:
     - ☑ Analog
     - ☑ PRI

Calling Services & Features

Bandwidth.com's Outbound-Only SIP Trunks provides a broad set of dialing services. The suite of services provided allows for a voice solution that is limited to supporting only outbound flows. The specific calling services and features are outlined below.

1. **<u>Long Distance (LD) Termination (Domestic & International):</u>** LD Termination allows you to terminate calls to any location in the world. These rates apply to all allowed outbound calls and are subject to unique minute-of-use rates based on whether the call is interstate, intrastate or international. The rating of the call as intrastate, interstate or international LD is based on the originating ANI provided by the Customer. Details on specific intrastate rates are located in the "Intrastate Rates" file. The manner in which they are applied is outlined further in the 'Terms & Conditions' document. Both of these documents are located on http://www.bandwidth.com/content/legal.

Mapping of Features to Outbound-Only SIP Trunk Components

Following is summary of the potential dialing features required by a customer and the component required to deliver these features:

Feature	Required Component (s)
LD Termination	▪ **Virtual Trunk**

Section
3

Supported Specifications

What features and specs are supported?

Detail on supported features and how they are implemented with Bandwidth.com's Outbound-Only SIP Trunk service

Signaling & Routing

- **Supported Protocols**: SIP is the only supported signaling protocol. Bandwidth.com has implemented SIP per RFC 3261
- **SIP Request Methods**: The following SIP request methods are supported: INVITE, ACK, BYE, CANCEL, OPTIONS
- **SIP URI:** Customers send SIP URIs to Bandwidth.com. Bandwidth.com does not support "tel" URIs. E.164 numbers must be formatted in #.164 format with a "+" prefix by the customer. The "user=phone" parameter has no meaning in the Bandwidth.com system. Local numbers without E.164 equivalents are not supported
- **UDP Transport:** Customers must utilize UDP to transport SIP signaling. Bandwidth.com servers listen on port 5060
- **Full & Compact Headers:** Bandwidth.com sends full headers. Bandwidth.com will receive full and compact headers
- **Call Hold:** Call Hold using RFC 2543 methods (c=0.0.0.0) is supported
- **Session Timer Refreshes**: Customer initiated re-invites are supported, however it should be noted that RFC 4028 is not supported and thus Bandwidth.com will not take on the responsibility of initiating session timer refreshes
- **Re-invite Addressing**: Re-invites from the same or multiple addresses in SDP is supported

Media (SDP and RTP)

- **RFC 2727 SDP:** Bandwidth.com supports the Session Description Protocol defined in RFC2327.
- **CODECs:** The G.711ulaw (AVT payload 0) and G.729a (AVT payload 18) are the only compression standards supported.

- **DTMF:** Only in-band Dual Tone, Multi-Frequency (DTMF) is supported. SIP Info Method is not supported.

- **DTMF Named Events:** RFC 2833 DTMF Named Events to the same address/port as audio RTP is supported for both G.711u and G.729a. DTMF Named Events will not be present in the offer (outbound Invite) for PSTN to IP calls. For IP to PSTN calls, DTMF Named Events are honored in the initial offer (Invite) from the customer. DTMF Named Events are honored in re-Invites in all cases. RTP dynamic payload Type 101 will be used to offer DTMF Named Events.

- **Default p-time**: Media will be 20ms default p-time

- **P-Time Attributes:** Bandwidth.com will not send a ptime attribute in its offer (Invite) In the customer's invite the included CODEC should be limited to G.711u and G.729a. If the entire list of CODECs is provided, then a failure will occur. It should be noted that the default behavior for Asterisk is to send the entire list – this should be pared down to G.711u and G.729a.

Section 4

Unsupported Specifications

What features are not supported?

Overview of features, specifications and RFCs that are not supported

Signaling & Routing

The following types of LAN interfaces can be supported on a Bandwidth.com Managed Router or Switch:

- **Signaling Protocols:** MGCP and H323 are not supported
- **SIP-T**: Not supported
- **SIP Request Methods:** REGISTER and INFO are not supported
- **Security of Signaling:** S-MIME, SIPS/TLS and other application level authentication and encryption techniques are not supported
- **Unknown and Proprietary Headers:** Bandwidth.com obeys RFC3261 and ignores any headers it does not understand or are outside the specifications of this RFC
- **Session Timer Refreshes:** RFC 4028 which required Bandwidth.com to initiate session timer refreshes is not supported
- **Call Hold:** The “Send Only” SDP attribute used for call hold per RFC 3264 is not supported.
- **Forking:** Bandwidth.com will not fork SIP requests. Bandwidth.com does not support receiving multiple responses due to forking in a customer’s network architecture. We will act on the first non-100 SIP response receiving with a “To” tag and ignore any follow-on responses received with a different “To” tag.
- **Locating SIP servers using DNS:** RFC 3263. IPV4 is sent in the ‘Request-URI’ to the customer
- **Tel URI for Telephone Numbers:** RFC 3966 is not supported. Bandwidth.com will send a “400 Bad Request”
- **E.164 Number and DNS (ENUM):** RFC 2916 is not supported. Bandwidth.com will respond with a “400 Bad Request”

Media

- **TCP Transport:** Not supported, but will be considered on a case-by-case basis

- **CODEC:** Neither G.711a nor any CODEC other than G.711u or G.729a are supported
- **Fax Transmission:** Neither T.38 nor T.31 are supported
- **Silence Suppression**: Not supported
- **DTMF**: Out-of-band DTMF is not supported. SIP Info Method is not supported.

N11 Features

The following n11 features are NOT supported:

- **911**: 91 emergency calls are not supported.
- **I2 Emergency Calls**: I2 Emergency calls are not supported
- **411:** 411 is not supported.
- **211** – Community Referral Services
- **311** – Non-Emergency Government Services
- **511** – Travel Information Telephone Services
- **611** – Repair Services
- **811** – Business Office Services
- **976** – Pay Services

Additional Limitations

- **No Inbound calling services are available**
- **Inbound Caller ID & Location:** No inbound caller ID & location services are provided
- **White Page Listing:** Bandwithd.com does not provide white page listing for any DIDs provided.
- **No local calling rates are available.**
- **The following types of calls are not supported:**
  - 976
  - 900
  - 888, 877, 866, 800
  - 700
  - 1010xxx

Section 5

Customer Support Processes

What support can I expect from Bandwidth.com for my Outbound-Only SIP Trunk service?

View into customer Installation, Trouble and Change Management processes

Scope of Support

The scope of Bandwidth.com's support for Outbound-Only SIP Trunks is limited to Bandwidth.com's own network infrastructure and any Customer Premise Equipment (CPE) devices provided for use in conjunction with the service such as Integrated Access Devises (IADs) for customers with TDM-based PBXs.

When providing support, Bandwidth.com support engineers will utilize packet trace tools to determine if an issue is isolated to Bandwidth.com's infrastructure or a customer's premise equipment. Bandwidth.com is specifically not responsible for troubleshooting or resolving any issues related to the configuration of a customer's PBX or any CPE not specifically provided by Bandwidth.com. Similarly, Bandwidth.com will not take on any aspect of configuring a customer's PBX or any CPE not provided by Bandwidth.com.

Installation Process

Bandwidth.com has designed a simple, scalable process for provisioning Outbound-Only SIP Trunks, providing customers regular updates throughout the installation of their service. The specifics are outlined below:

Activity	Description
Intro Call & Collection of IP Address	▪ Within days of receiving your order, you will be contacted by your Bandwidth.com Installation Engineer to introduce themselves, verify your order and outline the process ▪ If not collected on your intro call, you will subsequently receive an email from your Bandwidth.com installation Engineer asking for you to provide the IP address of your PBX or IAD
Configuration & Shipment of IAD (if applicable)	▪ If the customer is utilizing a TDM-based PBX, Bandwidth.com will pre-configure the required IAD ▪ Once configured and passing an initial test, the device will be shipped to the customer's location
Provisioning of Your Outbound-Only SIP Trunk	▪ Once your service has been provisioned, you will receive an email outlining the specific TNs we have provisioned for you and the IP address you should use to direct all terminating traffic
Service Acceptance	▪ Your Bandwidth.com Installation Engineer will work with you to confirm all call flows are operational and conduct any necessary troubleshooting for issues relating to Bandwidth.com infrastructure

Trouble Management

Bandwidth.com provides 24x7x365 support for Outbound-Only SIP Trunk service. Should you experience an issue related to your SIP Trunking service, you should submit a trouble ticket to our Network Operations Center (NOC) using one of the following methods:

a. Select the “Create a Trouble Ticket” Link in your MyBandwidth.com portal

b. Call our Customer Support Center at **(800) 808-5150** and press “2”

Section 6

The Customer Commitment

For what will I, as the customer, be responsible?

Customer Responsibilities

In order to ensure a quick, efficient turn-up of your Outbound-Only SIP Trunk and to ensure a high quality of ongoing service, a number of mandatory specifications must be implemented within your SIP, SDP and RTP protocol stacks:

GENERAL:

- ☑ *INFORM ALL END-USERS THAT OUTBOUND-ONLY SIP TRUNKS DO NOT PROVIDE ANY INBOUND DIALING SERVICE NOR ANY LEVEL OF 911 SERVICE.*
- ☑ *IF YOU HAVE AN IP-PBX, REVIEW SECTIONS 3 AND 4 OF THIS DOCUMENT IN THEIR ENTIRETY TO ENSURE YOU UNDERSTAND THE FEATURES AND REQUIRED SPECIFICATIONS FOR SIP TRUNKING*
- ☑ *IMPLEMENT A SINGLE PLAIN OLD TELEPHONE SERVICE (POTS)/STANDARD TELEPHONE LINE AT EACH SITE TO SUPPORT ANY FAX, ALARM SYSTEM, MODEM COMMUNICATION, CREDIT CARD AUTHORIZATION MACHINE OR OTHER NON-VOIP COMPLIANT APPLICATION*
- ☑ *UNPACKING, POWERING UP AND WIRE CONNECTIONS FOR ANY INTEGRATED ACCESS DEVICES (IAD) AT EACH SITE*
- ☑ *CONFORMANCE TO BANDWIDTH.COM TROUBLE TICKETING SUBMISSION PROCESSES*
- ☑ *SECURE AVAILABILITY OF DEDICATED CONTACT TO PARTICIPATE IN SIP TRUNKING / TURN-UP CALLS*

SIP :

- ☑ *CUSTOMERS MUST COMMUNICATE SIP WITH BANDWIDTH.COM ON PORT 5060, AT THE IP ADDRESS PROVIDED TO THEM DURING THE PROVISIONING/TURN-UP PROCESS, AND LISTEN ON PORT 5060 FOR SIP MESSAGES FROM BANDWIDTH.COM*
- ☑ *CUSTOMER SHOULD SEND '100 TRYING' RESPONSE FOR CALL PRGRESSION FOR EACH INVITE RECEIVED*
- ☑ *DISPLAY NAMES SHALL NOT CONTAIN WHITE OR BLANK SPACE UNLESS IT IS ENCLOSED IN QUOTES. EXMAPLE: "Pat Patterson"<sip:+17208881000>*
- ☑ *CUSTOMERS MUST BE CAPABLE OF ACCEPTING FROM HEADERS WITH ALPHANUMERIC USER PARTS. EXAMPLE: "From: sip:restricted@bw.gw.net"*
- ☑ *UNRECOGNIZED URI PARAMETERS MUST BE IGNORED PER RFC 3261*
- ☑ *IP TO PSTN TOLL FREE (8XX) CALLS MUST INCLUDE A CALLING PARTY NUMBER ASSIGNED BY BANDWIDTH.COM DURING PROVISIONING IN THE 'SIP FROM' HEADER*

- ☑ *CUSTOMERS MUST PROVIDE A LEGITIMATE ORIGINATING ANI. IF BANDWIDTH.COM CANNOT RESOLVE THE ORIGINATING ANI (VIA LOOKUP TO MASTER NPA-NXX DIRECTORIES) AS LEGITIMATE, THE CALL WILL BE ASSUMED TO BE INTRASTATE LD AND THE CUSTOMER WILL BE BILLED AT THE PREVAILING INTRASTATE RATE, ASSUMING THE CALL IS NOT INTERNATIONAL IN JURISDICTION.*

- ☑ *E.164 MUST BE IMPLEMENTED FOR ALL CALLS. (+1 FOR DOMESTIC; +1&COUNTRY CODE FOR INTERNATIONAL) – (011 SHOULD BE SUPPRESSED FOR INTERNATIONAL CALLS)*

RTP:

- ☑ *A 2 SECOND TIMEOUT SHOULD BE ESTABLISHED FOR ALL CUSTOMER-SENT RTP*