

The background features a stylized, light gray illustration of several network cables and connectors. The cables are shown in various orientations, some with RJ45 connectors visible. The overall aesthetic is clean and technical.

BROADVOICE™

SIP Trunking User Manual

Table of Contents



1. Introduction

2. Connection to Broadvoice

3. Technical Contacts

4. Product Guidelines

5. Technical Guidelines

6. Suggested Interoperability Testing

1. Introduction

Congratulations on your new Broadvoice SIP trunk connection. This manual will cover the basic setup and call flows as well as provide some useful information on your connection going forward.

2. Connection to Broadvoice

All SIP trunk connections to Broadvoice are done over the public internet. Broadvoice does not currently offer any dedicated circuit connections or BGP peering.

Connection Details

Peering IP: 208.64.8.13 or 206.15.130.13
Protocol: SIP
Port: 5060
Protocol: UDP
Codecs: G711u, G729a
FAX: T.38, G711u pass through
DTMF: RFC2833, G711u in-band

3. Technical Contacts

Technical Support 24x7

(Call Center 5am to 8pm PST Monday - Friday, 6am to 5pm Weekends, On Call after hours):

Email: Support@Broadvoice.com

Phone: 800-991-5015

Live chat via website at www.Broadvoice.com

4. Product Guidelines

A SIP trunk is a session between one or more SIP endpoints on your network and Broadvoice's border elements. Trunks may be Termination only or Bi-directional (Origination and Termination).

Trunk ID

When the trunk is configured you will be assigned a trunk ID. This will be a 10 digit domestic telephone number, and may be a number you ported in or, if no numbers are ported, Broadvoice will assign this number. This 10 digit TN cannot be canceled as long as this trunk is active, and should be referenced whenever contacting support.

Lines

A SIP Trunk can have multiple public IP's associated with it. Each public IP address defined as eligible to send/receive traffic on this trunk will need to have port capacity associated to it. Calls will be sent to the first available IP that has not reached capacity, and will not roll over to subsequent IP's until all capacity has been used. Any IP defined as a member of the trunk is eligible to send an outbound call.

Ports

All SIP trunks are assigned a specific number of available ports. Every concurrent outgoing or incoming call will use one port. Most customers will have only one IP endpoint and all ports will be assigned to that endpoint, however it is possible to configure multiple endpoints and assign ports across them to facilitate load balancing and fail-over. Each product will include a specific number of ports, and additional ports can be provisioned at an additional charge.

DID's

All SIP trunks can have a virtually unlimited number of DID's associated with them. Calls to these DID's use ports but do not incur any additional charges. DID's are available anywhere within the Broadvoice footprint and can be ported in from other carriers. DID's each have a monthly recurring charge associated with them to maintain the number. Each DID provisioned will come with the following:

- LIDB registration
- CNAM
- E911 Service

Toll Free

All SIP trunks can have a virtually unlimited number of toll free numbers associated with them. Calls to toll free numbers will use ports on your trunk as well as be billed at the rate defined in your service plan. Toll Free numbers each have a monthly recurring charge associated with them to maintain the number.

Origination

All calls originated via a SIP trunk will have:

- Calling Number (when not marked private)
- Calling Name (CNAM) when available

Termination

SIP trunks include a volume of minutes per month to be used for termination within the lower 48 states and Canada. All calls within this footprint are not charged up to the included minute amount.

- All minutes beyond this are charged at the blended domestic rate specified in the service plan.
- Calls to Alaska, Hawaii, Caribbean islands, and international destinations are billed at the current price indicated on our international rates page:
[http:// www.Broadvoice.com/cloud_pbx_global.aspx](http://www.Broadvoice.com/cloud_pbx_global.aspx)
- Directory assistance calls are billed at \$0.99 per call and \$0.10 per minute after 2 minutes if call completion is requested.
- Operator service calls are billed at \$3.00 per call

All these rates are subject to change. Please refer to our terms of service page for details:
<http://www.Broadvoice.com/terms.html>

5. Technical Guidelines

Below are the guidelines for what call flows are and are not supported with Broadvoice SIP Trunking. Please review these completely.

Signaling requirements and behavior

In order to ensure complete interoperability and experience the best possible results with Broadvoice SIP trunking please ensure the following behavior in your switch signaling:

Invites are sent using 10D format as described below

1. Domestic Calls: 1 + area code + number
2. OR Domestic Calls: area code + number
3. OR Domestic Calls: 7D number IF within the same area code as destination
4. International Calls: 011 + country code + number
5. Canada: 1 + area code + number
6. Caribbean islands: 1 + area code + number
7. 211 – Calls to 211 United Way / Human services are supported based on the calling number sent at no charge
8. 311 – Non emergency city services are supported based on the calling number sent at no charge
9. 411 – Information / Directory assistance calls are supported with an additional charge per call
10. 511 – Traffic information calls are supported based on the calling number as no charge
11. 611 – This will go to phone power support
12. 711 – TTY service for the hearing impaired are supported at no charge
13. 811 – Dig alert calls are supported based on the calling number sent at no charge
14. 911 can be supported if inbound numbers are provisioned. If a 911 call is sent from a non-provisioned number it will be answered but a charge of \$75 will be applied.

All signaling traffic contains public IP's in all headers described below:

1. From
2. To
3. Contact
4. Topmost VIA
5. SDP

Privacy

1. P-Asserted-Identity or Remote-party-ID are used to convey all privacy. It is recommended that a PAI header be inserted on all Invites but is not required.
2. Calls sent with privacy MUST have a PAI header inserted.
3. Inbound calls with privacy enabled will NOT have a PAI header describing the original

caller

4. Wholesale customers will receive PAI headers on inbound calls that have privacy enabled.
5. Calls sent with an anonymous from header should include a valid RPID or PAI header describing the true calling party for services that do not respect anonymity. If this header is not included the Trunk ID will be used.

Caller ID:

Valid options to set for the from number in traffic are:

1.
 1. Any 10 digit number provisioned on your trunk.
 2. In conjunction with the "Truth in Caller ID Act of 2010" all calls sent with a caller ID other than one of the numbers on your trunk will have their caller ID changed to the trunk ID.
 3. Customers may set any CNAM value desired, however delivery is left to the discretion of the terminating telco, and as such is best effort.
 4. Wholesale customers may set any 10 digit North American number.
2. SIP Options keep-alives:
 1. To ensure reliable service Broadvoice will send SIP Options messages to all configured customer endpoints to determine their availability. Any 2xx or 4xx class response to these messages will confirm that your equipment is up and ready to take calls. Failure to respond to these keep-alives will result in that endpoint not being offered calls until a keep-alive is successfully responded to.
3. Voice Codec
 1. Broadvoice supports only G711 and G729
 2. Silence suppression is not supported
4. DTMF
 1. Broadvoice supports both DTMF via RFC 2833 and InBand. SIP INFO is not supported.
5. Session Timers
 1. Broadvoice will use a re-invite to detect orphaned calls every 15 minutes. If the re-invite is not responded we will assume the call leg is orphaned and tear it down.
 2. Broadvoice supports a maximum call length of 4 hours.
6. Bandwidth and connectivity
 1. The customer is responsible for managing their own bandwidth and connectivity.
 2. A typical G711uLaw call will consume 84kbps of bandwidth. Please provision ports appropriately
7. Quality monitoring
 1. Broadvoice provides RTP statistics in its CDR's between the customer endpoint and its border elements purely for customer diagnostic purposes.

Supported Call Flows

- Call established from Customer endpoint to PSTN
- Call established from PSTN to Customer endpoint
- Call established from PSTN to Customer endpoint, Customer endpoint plays ring back
- Call established from Customer endpoint to PSTN, Broadvoice plays ring back
- RE-INVITE changing media preferences after call is established
- UPDATE changing media preferences during ring back but before 200OK
- Call established from Customer endpoint to PSTN with privacy
- Call established from PSTN to Customer endpoint with privacy
- SIP options health checks

Unsupported Call Flows

- Call transfer bridged in the phone power network
- Call hold via RFC 2543

Billing Behavior

- Billing on all outgoing calls commences at the 200OK
- Billing on all Toll-free calls commences at the 200OK
- Broadvoice is aware that some international destinations MAY send a 200ok while they are still playing ring-back. This call will be considered answered and billable.
- Once the customer has acknowledged their trunk is in production, they will be held responsible for all calls originating from their configured IP address.

6. Suggested Interoperability Testing

It is recommended that customers perform the following sample call flows to ensure complete functionality of their Broadvoice SIP Trunk prior to going into production, since improper behavior in any of these flows may result in improper billing treatment or impaired service.

It is encouraged that customers perform packet captures from their IP PBX and submit them to their interop engineer at Broadvoice to validate the correct behavior, however if this is not an option, customers can work with their interop engineers to lookup the call flows in our cache and validate the correct behavior for any of the tests they wish to perform.

6.1 Outbound Calls

Test Case 1.1.1:

Normal Call, Ring Back From PSTN End, 18x with SDP, With Answer , Hang-up from PSTN.

Time	66.159.235.87	208.64.8.13	
123.301	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)		
123.304	100 Trying		SIP Status
	(5060) <----- (5060)		
123.325	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
123.353	RTP (g711U)		RTP Num packets:134 Duration:2.660s SSRC:0x319EB48F
	(13442) -----> (49734)		
123.435	RTP (g711U)		RTP Num packets:322 Duration:6.419s SSRC:0x5A02423F
	(13442) <----- (49734)		
126.017	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
126.020	ACK		SIP Request
	(5060) -----> (5060)		
126.034	RTP (g711U)		RTP Num packets:192 Duration:3.819s SSRC:0x2C5812E3
	(13442) -----> (49734)		
129.867	BYE		SIP Request
	(5060) <----- (5060)		
129.870	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 1.1.2:

Normal Call, Ring Back From PSTN End, 18x with SDP, With Answer , Hang-up from Customer Proxy.

Time	66.159.235.87	208.64.8.13	
123.301	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)		
123.304	100 Trying		SIP Status
	(5060) <----- (5060)		
123.325	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
123.353	RTP (g711U)		RTP Num packets:134 Duration:2.660s SSRC:0x319EB48F
	(13442) -----> (49734)		
123.435	RTP (g711U)		RTP Num packets:322 Duration:6.419s SSRC:0x5A02423F
	(13442) <----- (49734)		
126.017	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
126.020	ACK		SIP Request
	(5060) -----> (5060)		
126.034	RTP (g711U)		RTP Num packets:192 Duration:3.819s SSRC:0x2C5812E3
	(13442) -----> (49734)		
129.867	BYE		SIP Request
	(5060) -----> (5060)		
129.870	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 1.2.1:

Ring No Answer, SIP End Hangup During Ring back, SIP Resp Cancel & SIP Resp 487

Time	66.159.235.87	208.64.8.13	
3.654	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) ----->	(5060)	
3.657	100 Trying		SIP Status
	(5060) <-----	(5060)	
3.680	183 Session Progress SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
(3.701	RTP (g711U)		RTP Num packets:81 Duration:1.599s SSRC:0x3BF31B37
	(10514) ----->	(49742)	
3.777	RTP (g711U)		RTP Num packets:78 Duration:1.538s SSRC:0x303272F2
	(10514) <-----	(49742)	
5.321	CANCEL		SIP Request
	(5060) ----->	(5060)	
5.322	200 OK		SIP Status
	(5060) <-----	(5060)	
5.328	487 Request Terminated		SIP Status
	(5060) <-----	(5060)	
5.331	ACK		SIP Request
	(5060) ----->	(5060)	

Test Case 1.2.2:

Ring No Answer, Timeout

Time	66.159.235.87	208.64.8.13	
78.655	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) ----->	(5060)	
78.658	100 Trying		SIP Status
	(5060) <-----	(5060)	
78.758	180 Ringing		SIP Status
	(5060) <-----	(5060)	
138.818	504 Server Time-out		SIP Status
	(5060) <-----	(5060)	
138.821	ACK		SIP Request
	(5060) ----->	(5060)	

Test Case 1.2.3:

User Busy, SIP Resp 486

Time	66.159.235.87	208.64.8.13	
78.655	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) ----->	(5060)	
78.658	100 Trying		SIP Status
	(5060) <-----	(5060)	
78.758	486 Busy Here		SIP Status
	(5060) <-----	(5060)	
78.858	ACK		SIP Request
	(5060) ----->	(5060)	

Test Case 1.3.1:

1 Hour hold time & SIP Session Timers

Time	66.159.235.87	208.64.8.13	
0.000	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)		
0.003	100 Trying		SIP Status
	(5060) <----- (5060)		
0.105	180 Ringing		SIP Status
	(5060) <----- (5060)		
6.020	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
6.022	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
6.026	ACK		SIP Request
	(5060) -----> (5060)		
936.060	INVITE SDP (g711U)		SIP Request
	(5060) <----- (5060)		
936.063	100 Trying		SIP Status
	(5060) -----> (5060)		
936.063	200 OK SDP (g711U)		SIP Status
	(5060) -----> (5060)		
936.070	ACK		SIP Request
	(5060) <----- (5060)		
1866.105	INVITE SDP (g711U)		SIP Request
	(5060) <----- (5060)		
1866.109	100 Trying		SIP Status
	(5060) -----> (5060)		
1866.109	200 OK SDP (g711U)		SIP Status
	(5060) -----> (5060)		
1866.118	ACK		SIP Request
	(5060) <----- (5060)		
2796.175	INVITE SDP (g711U)		SIP Request
	(5060) <----- (5060)		
2796.179	100 Trying		SIP Status
	(5060) -----> (5060)		
2796.179	200 OK SDP (g711U)		SIP Status
	(5060) -----> (5060)		
2796.185	ACK		SIP Request
	(5060) <----- (5060)		

Test Case 1.4.1:

Single choice, g711 20ms

Time	66.159.235.87		
		208.64.8.13	
8.597	INVITESDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:3234931422@208.64.8.13
	(5060) ----->	(5060)	
8.600	100 Trying		SIP Status
	(5060) <-----	(5060)	
9.136	RTP (g711U)		RTP Num packets:2985 Duration:59.696s SSRC:0x8B93B6CF
	(14874) <-----	(49796)	
10.125	183 Session Progress SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
10.154	RTP (g711U)		RTP Num packets:329 Duration:6.559s SSRC:0x4561C605
	(14874) ----->	(49796)	
16.723	200 OK SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
16.726	ACK		SIP Request
	(5060) ----->	(5060)	
16.734	RTP (g711U)		RTP Num packets:2608 Duration:52.136s SSRC:0x2B5A9D00
	(14874) ----->	(49796)	
68.872	BYE		SIP Request
	(5060) <-----	(5060)	
68.875	200 OK		SIP Status
	(5060) ----->	(5060)	

```
v=0
o=root 1562814183 1562814183 IN IP4 66.159.235.87
s=Asterisk PBX 1.6.0.26-FONCORE-r78
c=IN IP4 66.159.235.87
b=CT:384
t=0 0
m=audio 11332 RTP/AVP 0 8
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

Test Case 1.4.2:

InBand DTMF, g711 20ms

Time	66.159.235.87	208.64.8.13	
154.676	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:13234931422@208.64.8.13
	(5060) -----> (5060)		
154.678	100 Trying		SIP Status
	(5060) <----- (5060)		
155.230	RTP (g711U)		RTP Num packets:1498 Duration:29.958s SSRC:0x9BA14863
	(19568) <----- (49840)		
156.151	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
156.179	RTP (g711U)		RTP Num packets:260 Duration:5.181s SSRC:0x3202F5F9
	(19568) -----> (49840)		
161.364	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
161.367	ACK		SIP Request
	(5060) -----> (5060)		
161.378	RTP (g711U)		RTP Num packets:1192 Duration:23.818s SSRC:0x2F7DB77E
	(19568) -----> (49840)		
	185.211 BYE		SIP Request
	(5060) <----- (5060)		
185.214	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 1.4.3:

DTMF RFC2833 named events, g711 20ms

Time	66.159.235.87	208.64.8.13	
9.582	INVITE SDP (g711U)		SIP From: sip:8886076937@66.159.235.87 To:sip:13234931422@208.64.8.13
	(5060) -----> (5060)		
9.584	100 Trying		SIP Status
	(5060) <----- (5060)		
10.066	RTP (g711U)		RTP Num packets:2637 Duration:52.736s SSRC:0x12C74C40
	(15990) <----- (49852)		
11.008	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
	11.032 RTP (g711U)		RTP Num packets:453 Duration:9.039s SSRC:0x130D55E3
	(15990) -----> (49852)		
20.082	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.085	ACK		SIP Request
	(5060) -----> (5060)		
20.093	RTP (g711U)		RTP Num packets:2138 Duration:42.736s SSRC:0x217E7D92
	(15990) -----> (49852)		
62.835	BYE		SIP Request
	(5060) <----- (5060)		
62.838	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 1.4.4:

Fax InBand, g711 20ms

Time	66.159.235.87	208.64.8.13	
154.676	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:13234931422@208.64.8.13
	(5060) ----->	(5060)	
154.678	100 Trying		SIP Status
	(5060) <-----	(5060)	
155.230	RTP (g711U)		RTP Num packets:1498 Duration:29.958s SSRC:0x9BA14863
	(19568) <-----	(49840)	
156.151	183 Session Progress SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
156.179	RTP (g711U)		RTP Num packets:260 Duration:5.181s SSRC:0x3202F5F9
	(19568) ----->	(49840)	
161.364	200 OK SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
161.367	ACK		SIP Request
	(5060) ----->	(5060)	
161.378	RTP (g711U)		RTP Num packets:1192 Duration:23.818s SSRC:0x2F7DB77E
	(19568) ----->	(49840)	
185.211	BYE		SIP Request
	(5060) <-----	(5060)	
185.214	200 OK		SIP Status
	(5060) ----->	(5060)	

Test Case 1.4.5:

Fax Re-Invite to T.38, g711 20ms

Time	66.159.235.87	208.64.8.13	
0.000	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) ----->	(5060)	
0.044	100 Trying		SIP Status
	(5060) <-----	(5060)	
0.844	180 Ringing		SIP Status
	(5060) <-----	(5060)	
2.285	200 OK SDP (g711u)		SIP Status
	(5060) <-----	(5060)	
2.293	ACK		SIP Request
	(5060) ----->	(5060)	
2.659	INVITE SDP (t38)		SIP From: sip:8182644371@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) ----->	(5060)	
2.697	100 Trying		SIP Status
	(5060) <-----	(5060)	
2.866	200 OK SDP (t38)		SIP Status
	(5060) <-----	(5060)	
2.874	ACK		SIP Request
	(5060) ----->	(5060)	
61.085	BYE		SIP Request
	(5060) ----->	(5060)	
61.136	200 OK		SIP Status
	(5060) <-----	(5060)	

Test Case 1.4.6:

Single choice, g729 20ms, DTMF RFC283

Time	66.159.235.87	208.64.8.13	
16.219	INVITE SDP (g729a)		SIP From: sip:8886076937@66.159.235.87 To:sip:13234931422@208.64.8.13
	(5060) -----> (5060)		
16.222	100 Trying		SIP Status
	(5060) <----- (5060)		
16.727	RTP (g729a)		RTP Num packets:2183 Duration:43.657s SSRC:0x6F6EB7AF
	(12014) <----- (49782)		
17.598	183 Session Progress SDP (g729a)		SIP Status
	(5060) <----- (5060)		
17.632	RTP (g729a)		RTP Num packets:1449 Duration:28.958s SSRC:0x6FC75316
	(12014) -----> (49782)		
46.599	200 OK SDP (g729a)		SIP Status
	(5060) <----- (5060)		
46.602	ACK		SIP Request
	(5060) -----> (5060)		
46.610	RTP (g729a)		RTP Num packets:689 Duration:13.759s SSRC:0x5ABA6A78
	(12014) -----> (49782)		
60.383	BYE		SIP Request
	(5060) -----> (5060)		
60.389	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 1.5.1:

Normal Call with no FROM number

Please ensure you set privacy using either a P-Asserted-Identity header or a Remote-Party-ID header to denote the actual calling party for billing and compliance purposes.

Time	66.159.235.87	208.64.8.13	
123.301	INVITE SDP (g711U)		SIP From: sip:Unknown@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)		
123.304	100 Trying		SIP Status
	(5060) <----- (5060)		
123.325	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
123.353	RTP (g711U)		RTP Num packets:134 Duration:2.660s SSRC:0x319EB48F
	(13442) -----> (49734)		
123.435	RTP (g711U)		RTP Num packets:322 Duration:6.419s SSRC:0x5A02423F
	(13442) <----- (49734)		
126.017	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
126.020	ACK		SIP Request
	(5060) -----> (5060)		
126.034	RTP (g711U)		RTP Num packets:192 Duration:3.819s SSRC:0x2C5812E3
	(13442) -----> (49734)		
129.867	BYE		SIP Request
	(5060) <----- (5060)		
129.870	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 1.5.2:

Normal Call to toll free

Time	66.159.235.87	208.64.8.13	
12.741	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:18886076937@208.64.8.13
	(5060) -----> (5060)		
12.744	100 Trying		SIP Status
	(5060) <----- (5060)		
16.627	RTP (g711U)		RTP Num packets:524 Duration:10.499s SSRC:0xA452F3EF
	(11890) <----- (49868)		
20.329	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.355	RTP (g711U)		RTP Num packets:11 Duration:0.201s SSRC:0x2249B35A
	(11890) -----> (49868)		
20.568	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.571	ACK		SIP Request
	(5060) -----> (5060)		
20.578	RTP (g711U)		RTP Num packets:328 Duration:6.543s SSRC:0x17CDBA38
	(11890) -----> (49868)		
27.136	BYE		SIP Request
	(5060) -----> (5060)		
27.141	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 1.5.3:

International Call (011 + CC + XXXXXXXXXXX)

Hong Kong weather -011-852-187-8200

Speaking Clock (UK) – 011-44-871-789-3642

Time	66.159.235.87	208.64.8.13	
19.353	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:0118521878200@208.64.8.13
	(5060) -----> (5060)		
19.356	100 Trying		SIP Status
	(5060) <----- (5060)		
19.839	RTP (g711U)		RTP Num packets:37 Duration:0.719s SSRC:0x86D5AF18
	(11810) <----- (49880)		
20.995	RTP (g711U)		RTP Num packets:476 Duration:9.519s SSRC:0xA31AF737
	(11810) <----- (49880)		
22.425	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
22.463	RTP (g711U)		RTP Num packets:172 Duration:3.422s SSRC:0x747D29AB
	(11810) -----> (49880)		
25.894	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
25.897	ACK		SIP Request
	(5060) -----> (5060)		
25.903	RTP (g711U)		RTP Num packets:230 Duration:4.587s SSRC:0x2F289189
	(11810) -----> (49880)		
30.523	BYE		SIP Request
	(5060) -----> (5060)		
30.528	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 1.5.4:

Service Call (311)

Time	66.159.235.87	208.64.8.13	
5.722	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:311@208.64.8.13
	(5060) -----> (5060)		
5.725	100 Trying		SIP Status
	(5060) <----- (5060)		
7.503	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
7.526	RTP (g711U)		RTP Num packets:246 Duration:4.899s SSRC:0x1AF0D829
	(19174) <----- (49876)		
7.527	RTP (g711U)		RTP Num packets:20 Duration:0.380s SSRC:0x5D2A707F
	(19174) -----> (49876)		
7.915	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
7.919	ACK		SIP Request
	(5060) -----> (5060)		
7.928	RTP (g711U)		RTP Num packets:225 Duration:4.477s SSRC:0x761D4B27
	(19174) -----> (49876)		
12.431	BYE		SIP Request
	(5060) -----> (5060)		
12.436	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 1.5.5:

Emergency (911)

When placing this call indicate to the PSAP that you are placing a test call and ask to confirm the service address. This will NOT result in a 911 dispatch to your location. DO NOT HANG UP UNTIL YOU SPEAK WITH THE OPERATOR.

Time	66.159.235.87	208.64.8.13	
5.722	INVITE SDP (g711U)		SIP From: sip:8182644371@66.159.235.87 To:sip:911@208.64.8.13
	(5060) -----> (5060)		
5.725	100 Trying		SIP Status
	(5060) <----- (5060)		
7.503	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
7.526	RTP (g711U)		RTP Num packets:246 Duration:4.899s SSRC:0x1AF0D829
	(19174) <----- (49876)		
7.527	RTP (g711U)		RTP Num packets:20 Duration:0.380s SSRC:0x5D2A707F
	(19174) -----> (49876)		
7.915	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
7.919	ACK		SIP Request
	(5060) -----> (5060)		
7.928	RTP (g711U)		RTP Num packets:225 Duration:4.477s SSRC:0x761D4B27
	(19174) -----> (49876)		
12.431	BYE		SIP Request
	(5060) -----> (5060)		
12.436	200 OK		SIP Status
	(5060) <----- (5060)		

6.2 Inbound Calls

Test Case 2.1.1:

Normal Call, Ring Back From PSTN End, 18x with SDP, With Answer , Hang-up from PSTN.

Time	66.159.235.87	208.64.8.13	
123.301	INVITE SDP (g711U)		SIP From: sip:8188547456@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
123.304	100 Trying		SIP Status
	(5060) <----- (5060)		
123.325	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
123.353	RTP (g711U)		RTP Num packets:134 Duration:2.660s SSRC:0x319EB48F
	(13442) -----> (49734)		
123.435	RTP (g711U)		RTP Num packets:322 Duration:6.419s SSRC:0x5A02423F
	(13442) <----- (49734)		
126.017	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
126.020	ACK		SIP Request
	(5060) -----> (5060)		
126.034	RTP (g711U)		RTP Num packets:192 Duration:3.819s SSRC:0x2C5812E3
	(13442) -----> (49734)		
129.867	BYE		SIP Request
	(5060) <----- (5060)		
129.870	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 2.1.2:

Normal Call, Ring Back From PSTN End, 18x with SDP, With Answer, Hang-up from Customer Proxy.

Time	66.159.235.87	208.64.8.13	
123.301	INVITE SDP (g711U)		SIP From: sip:8188547456@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
123.304	100 Trying		SIP Status
	(5060) <----- (5060)		
123.325	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
123.353	RTP (g711U)		RTP Num packets:134 Duration:2.660s SSRC:0x319EB48F
	(13442) -----> (49734)		
123.435	RTP (g711U)		RTP Num packets:322 Duration:6.419s SSRC:0x5A02423F
	(13442) <----- (49734)		
126.017	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
126.020	ACK		SIP Request
	(5060) -----> (5060)		
126.034	RTP (g711U)		RTP Num packets:192 Duration:3.819s SSRC:0x2C5812E3
	(13442) -----> (49734)		
129.867	BYE		SIP Request
	(5060) -----> (5060)		
129.870	200 OK		SIP Status
	(5060) <----- (5060)		

Test Case 2.2.1:

Ring No Answer, SIP End Hangup During Ring back, SIP Resp Cancel & SIP Resp 487

Time	66.159.235.87	208.64.8.13	
3.654	INVITE SDP (g711U)		SIP From: sip:8188547456@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
3.657	100 Trying		SIP Status
	(5060) <----- (5060)		
3.680	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
3.701	RTP (g711U)		RTP Num packets:81 Duration:1.599s SSRC:0x3BF31B37
	(10514) -----> (49742)		
3.777	RTP (g711U)		RTP Num packets:78 Duration:1.538s SSRC:0x303272F2
	(10514) <----- (49742)		
5.321	CANCEL		SIP Request
	(5060) -----> (5060)		
5.322	200 OK		SIP Status
	(5060) <----- (5060)		
5.328	487 Request Terminated		SIP Status
	(5060) <----- (5060)		
5.331	ACK		SIP Request
	(5060) -----> (5060)		

Test Case 2.2.2:

Ring No Answer, Timeout

Time	66.159.235.87	208.64.8.13	
78.655	INVITE SDP (g711U)		SIP From: sip:8188547456@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
78.658	100 Trying		SIP Status
	(5060) <----- (5060)		
78.758	180 Ringing		SIP Status
	(5060) <----- (5060)		
138.818	504 Server Time-out		SIP Status
	(5060) <----- (5060)		
138.821	ACK		SIP Request
	(5060) -----> (5060)		

Test Case 2.2.3:

User Busy, SIP Resp 486

Time	66.159.235.87	208.64.8.13	
78.655	INVITE SDP (g711U)		SIP From: 8188547456 sip:@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
78.658	100 Trying		SIP Status
	(5060) <----- (5060)		
78.758	486 Busy Here		SIP Status
	(5060) <----- (5060)		
78.858	ACK		SIP Request
	(5060) -----> (5060)		

Test Case 2.3.1:

1 Hour hold time & SIP Session Timers

Time	66.159.235.87	208.64.8.13	
0.000	INVITE SDP (g711U)		SIP From: sip:8188547456@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) ----->	(5060)	
0.003	100 Trying		SIP Status
	(5060) <-----	(5060)	
0.105	180 Ringing		SIP Status
	(5060) <-----	(5060)	
6.020	183 Session Progress SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
6.022	200 OK SDP (g711U)		SIP Status
	(5060) <-----	(5060)	
6.026	ACK		SIP Request
	(5060) ----->	(5060)	
936.060	INVITE SDP (g711U)		SIP Request
	(5060) <-----	(5060)	
936.063	100 Trying		SIP Status
	(5060) ----->	(5060)	
936.063	200 OK SDP (g711U)		SIP Status
	(5060) ----->	(5060)	
936.070	ACK		SIP Request
	(5060) <-----	(5060)	
1866.105	INVITE SDP (g711U)		SIP Request
	(5060) <-----	(5060)	
1866.109	100 Trying		SIP Status
	(5060) ----->	(5060)	
1866.109	200 OK SDP (g711U)		SIP Status
	(5060) ----->	(5060)	
1866.118	ACK		SIP Request
	(5060) <-----	(5060)	
2796.175	INVITE SDP (g711U)		SIP Request
	(5060) <-----	(5060)	
2796.179	100 Trying		SIP Status
	(5060) ----->	(5060)	
2796.179	200 OK SDP (g711U)		SIP Status
	(5060) ----->	(5060)	
2796.185	ACK		SIP Request
	(5060) <-----	(5060)	

Test Case 2.4.1:

Single choice, g711 20ms

Time	66.159.235.87	208.64.8.13	
8.597	INVITESDP (g711U)		SIP From: sip:3234931422@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
8.600	100 Trying		SIP Status
	(5060) <----- (5060)		
9.136	RTP (g711U)		RTP Num packets:2985 Duration:59.696s SSRC:0x8B93B6CF
	(14874) <----- (49796)		
10.125	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
10.154	RTP (g711U)		RTP Num packets:329 Duration:6.559s SSRC:0x4561C605
	(14874) -----> (49796)		
16.723	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
16.726	ACK		SIP Request
	(5060) -----> (5060)		
16.734	RTP (g711U)		RTP Num packets:2608 Duration:52.136s SSRC:0x2B5A9D00
	(14874) -----> (49796)		
68.872	BYE		SIP Request
	(5060) <----- (5060)		
68.875	200 OK		SIP Status
	(5060) -----> (5060)		

```
v=0
o=root 1562814183 1562814183 IN IP4 66.159.235.87
s=Asterisk PBX 1.6.0.26-FONCORE-r78
c=IN IP4 66.159.235.87
b=CT:384
t=0
m=audio 11332 RTP/AVP 0 8
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=silenceSupp:off - - - -
a=ptime:20
a=sendrcv
```

Test Case 2.4.2:

InBand DTMF, g711 20ms

Time	66.159.235.87	208.64.8.13	
154.676	INVITE SDP (g711U)		SIP From: sip:13234931422@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
154.678	100 Trying		SIP Status
	(5060) <----- (5060)		
155.230	RTP (g711U)		RTP Num packets:1498 Duration:29.958s SSRC:0x9BA14863
	(19568) <----- (49840)		
156.151	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
156.179	RTP (g711U)		RTP Num packets:260 Duration:5.181s SSRC:0x3202F5F9
	(19568) -----> (49840)		
161.364	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
161.367	ACK		SIP Request
	(5060) -----> (5060)		
161.378	RTP (g711U)		RTP Num packets:1192 Duration:23.818s SSRC:0x2F7DB77E
	(19568) -----> (49840)		
185.211	BYE		SIP Request
	(5060) <----- (5060)		
185.214	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 2.4.3:

DTMF RFC2833 named events, g711 20ms

Time	66.159.235.87	208.64.8.13	
9.582	INVITE SDP (g711U)		SIP From: sip:13234931422@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
9.584	100 Trying		SIP Status
	(5060) <----- (5060)		
10.066	RTP (g711U)		RTP Num packets:2637 Duration:52.736s SSRC:0x12C74C40
	(15990) <----- (49852)		
11.008	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
11.032	RTP (g711U)		RTP Num packets:453 Duration:9.039s SSRC:0x130D55E3
	(15990) -----> (49852)		
20.082	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.085	ACK		SIP Request
	(5060) -----> (5060)		
20.093	RTP (g711U)		RTP Num packets:2138 Duration:42.736s SSRC:0x217E7D92
	(15990) -----> (49852)		
62.835	BYE		SIP Request
	(5060) <----- (5060)		
62.838	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 2.4.4:

Fax InBand, g711 20ms

Time	66.159.235.87	208.64.8.13	
154.676	INVITE SDP (g711U)		SIP From: sip:13234931422@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
154.678	100 Trying		SIP Status
	(5060) <----- (5060)		
155.230	RTP (g711U)		RTP Num packets:1498 Duration:29.958s SSRC:0x9BA14863
	(19568) <----- (49840)		
156.151	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
156.179	RTP (g711U)		RTP Num packets:260 Duration:5.181s SSRC:0x3202F5F9
	(19568) -----> (49840)		
161.364	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
161.367	ACK		SIP Request
	(5060) -----> (5060)		
161.378	RTP (g711U)		RTP Num packets:1192 Duration:23.818s SSRC:0x2F7DB77E
	(19568) -----> (49840)		
185.211	BYE		SIP Request
	(5060) <----- (5060)		
185.214	200 OK		SIP Status
	(5060) -----> (5060)		

Test Case 2.4.5:

Fax Re-Invite to T.38, g711 20ms

Time	66.159.235.87	
	208.64.8.13	
0.000	INVITE SDP (g729 g711U)	SIP From: sip:8182644371@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)	
0.044	100 Trying	SIP Status
	(5060) <----- (5060)	
0.844	180 Ringing	SIP Status
	(5060) <----- (5060)	
2.285	200 OK SDP (g711)	SIP Status
	(5060) <----- (5060)	
2.293	ACK	SIP Request
	(5060) -----> (5060)	
2.659	INVITE SDP (t38)	SIP From: sip:8182644371@66.159.235.87 To:sip:8188547456@208.64.8.13
	(5060) -----> (5060)	
2.697	100 Trying	SIP Status
	(5060) <----- (5060)	
2.866	200 OK SDP (t38)	SIP Status
	(5060) <----- (5060)	
2.874	ACK	SIP Request
	(5060) -----> (5060)	
61.085	BYE	SIP Request
	(5060) -----> (5060)	
61.136	200 OK	SIP Status
	(5060) <----- (5060)	

Test Case 2.5.1:

Normal Call with no FROM number

Time	208.64.8.13	
	66.159.235.87	
3.667	INVITE SDP (g711U g729 telephone-event)	SIP From: sip:anonymous@208.64.8.13:5060 To:sip:8182644371@66.159.235.87
	(5060) -----> (5060)	
3.673	100 Trying	SIP Status
	(5060) <----- (5060)	
4.087	180 Ringing	SIP Status
	(5060) <----- (5060)	
4.113	180 Ringing	SIP Status
	(5060) <----- (5060)	
8.407	200 OK SDP (g711U)	SIP Status
	(5060) <----- (5060)	
8.415	ACK	SIP Request
	(5060) -----> (5060)	
8.426	RTP (g711U)	RTP Num packets:41 Duration:0.819s SSRC:0x21393734
	(50108) <----- (14634)	
8.486	RTP (g711U)	RTP Num packets:39 Duration:0.779s SSRC:0xA93D5285
	(50108) -----> (14634)	
9.271	BYE	SIP Request
	(5060) <----- (5060)	
9.276	200 OK	SIP Status
	(5060) -----> (5060)	

INVITE sip:8182644371@66.159.235.87:5060;transport=udp SIP/2.0
 Via: SIP/2.0/UDP 208.64.8.13:5060;branch=z9hG4bKfi36h0203ov17fs2o3c0.1
 Allow-Events: message-summary, refer, dialog, line-seize, presence, call-info
 Max-Forwards: 69
 Call-ID: 333837CD@192.168.22.50
 From: "Anonymous" <sip:anonymous@208.64.8.13:5060>;tag=192.168.22.50+1+1d942f+fe35a170
 To: <sip:8182644371@66.159.235.87>
 CSeq: 111174205 INVITE
 Expires: 180
 Organization: Phone Power
 Supported: 100rel, resource-priority
 Content-Length: 164
 Content-Type: application/sdp
 Contact: "Anonymous" <sip:SD8f5rd-vp9pmjrfsar7sflvprmu1ptp9plqvsvp6hvjh0pgveip1or8fbo50sc4v1@208.64.8.65:5060;transport=udp>
 v=0
 o=- 2701192565 2701192565 IN IP4 208.64.8.13
 s=-
 c=IN IP4 208.64.8.65
 t=0 0
 m=audio 50108 RTP/AVP 0 18 101
 a=rtpmap:101 telephone-event/8000
 a=ptime:20

Test Case 2.5.2:

Call to Toll Free number

Time	66.159.235.87		
		208.64.8.13	
12.741	INVITE SDP (g711U)		SIP From: sip:18886076937@66.159.235.87 To:sip:8182644371@208.64.8.13
	(5060) -----> (5060)		
12.744	100 Trying		SIP Status
	(5060) <----- (5060)		
16.627	RTP (g711U)		RTP Num packets:524 Duration:10.499s SSRC:0xA452F3EF
	(11890) <----- (49868)		
20.329	183 Session Progress SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.355	RTP (g711U)		RTP Num packets:11 Duration:0.201s SSRC:0x2249B35A
	(11890) -----> (49868)		
20.568	200 OK SDP (g711U)		SIP Status
	(5060) <----- (5060)		
20.571	ACK		SIP Request
	(5060) -----> (5060)		
20.578	RTP (g711U)		RTP Num packets:328 Duration:6.543s SSRC:0x17CDBA38
	(11890) -----> (49868)		
27.136	BYE		SIP Request
	(5060) -----> (5060)		
27.141	200 OK		SIP Status
	(5060) <----- (5060)		



BROADVOICE™

1-800-991-5015

20847 Sherman Way, Winnetka, CA 91306

www.Broadvoice.com