SIP Testing Checklist

This document contains a test checklist for XO Enterprise SIP Trunking. The test verifies the basic interoperability, features of the appliance, and the integration of the architecture framework into XO Enterprise SIP. This certification test plan was executed in XO's Plano Certification Laboratory (Plano Lab) which is equipped with a variety of test equipment and is capable of simulating XO's VoIP network environment.

This may serve as a guide to SIP testing but is not intended as a certification or endorsement of any given device. XO always recommends testing directly with the SIP provider in order to determine interoperability.

Item	Description	Comments	Test Result
1	**Basic Calls (G711)**	**ESIP1 is set to G.711**	
1.1	Inbound call from PSTN to Enterprise SIP Location for ported and nonported TNs	Verify the call can be established and voice path is clear.	
1.2	Outbound call from Enterprise SIP Location (11-digits) to PSTN for ported and non-ported TNs	Verify the call can be established and voice path is clear.	
1.3	1+10-digits to PSTN for ported and non-ported TNs	Verify the call can be established and voice path is clear.	
1.4	Inbound Toll Free	Make a call from PSTN to 1-800-XXXXXXX. Verify the call can be terminated and voice path is clear.	
1.5	Outbound Toll Free to TFN	Verify the call can be established and voice path is clear.	
1.6	Operator 0-	Dial 0 from NPA-NXX-XXXX. Verify the call can be terminated to XO operator.	

1.7	Operator 0+11-digits	Dial 0+11 digits to NPA-NXX-XXXX. Verify the call can be terminated to operator.
1.8	International 011+IDDD	Verify the call can be established and voice path is clear.
1.9	DA	Dial 411. Verify the call can be terminated to operator.
1.10	Static E 911 calls (simulated from different customer locations) delivered to the right PSAP	Verify the call can be terminated to PSAP. Should be pre-arranged with PSAP.
1.11	Inbound / outbound calls between ESIP users and XO's Retail VOIP users complete	Verify the call can be established and voice path is clear.
1.12	Inbound / outbound calls between ESIP users and XO's WVOIP users complete	Verify the call can be established and voice path is clear.
2	**Basic Calls (G729a)**	**ESIP is set to G.729a**
2.1	Inbound call from PSTN to SESL for ported and non-ported TNs	Verify the call can be established and voice path is clear.
2.2	Outbound call from SESL (11-digits) to PSTN for ported and non-ported XO TNs	Verify the call can be established and voice path is clear.
2.3	1+10-digits to PSTN for ported and non-ported XO TNs	Verify the call can be established and voice path is clear.
2.4	Inbound Toll Free	Make a call from PSTN to 1-800-XXXXXXX.Verify the call can be terminated and voice path is clear

2.5	Outbound Toll Free to XO TFN and other carrier's TFN	Verify the call can be established and voice path is clear.
2.6	Operator 0-	Dial 0 from NPA-NXX-XXXX. Verify the call can be terminated to operator
2.7	Operator 0+11-digits	Dial 0+11 digits to NPA-NXX-XXXX. Verify the call can be terminated to operator.
2.8	International 011+IDDD	Verify the call can be established and voice path is clear
2.9	DA	Dial 411. Verify the call can be terminated to operator
2.10	static E 911 calls (simulated from different customer locations) delivered to the right PSAP	Verify the call can be terminated to PSAP. Should be pre-arranged with PSAP.
2.11	inbound / outbound calls between ESIP users and XO's Retail VOIP users complete	Verify the call can be established and voice path is clear.
2.12	inbound / outbound calls between ESIP users and XO's WVOIP users complete	Verify the call can be established and voice path is clear
3	**Caller ID (CLID) and Calling Name (CNAM) Presentation)**	
3.1	Inbound CLID from PSTN	Verify the PSTN CLID is displayed on the termination phone.
3.2	Outbound CLID to PSTN	Verify the CLID is displayed on the termination phone.

3.3	Inbound Caller ID Blocking	Verify the PSTN CLID is blocked.
3.4	Outbound Caller ID Blocking	Verify the CLID is blocked.
3.5	Inbound CNAM	Verify the PSTN CNAME is displayed on the termination phone.
4	**Call Forward: (1st party = calling; 2nd party = 1st called party; 3rd party = the call-forwarded-to party) Verify CLID delivery**	
4.1	Call Forward Always by Enterprise SIP Location to Extension on inbound call from PSTN	Verify the call is forwarded.
4.2	Call Forward Always by Enterprise SIP Location to PSTN on inbound call from PSTN	Verify the call is forwarded.
4.3	Call Forward Busy by by Enterprise SIP Location to Extension on inbound call from PSTN	Verify the call is forwarded.
4.4	Call Forward Busy by Enterprise SIP Location to PSTN on inbound call from PSTN	Verify the call is forwarded.
4.5	Call Forward No Answer by Enterprise SIP Location to Extension on inbound call from PSTN	Verify the call is forwarded.
4.6	Call Forward No Answer by Enterprise SIP Location to PSTN on inbound call from PSTN	Verify the call is forwarded.

5	**Dual Tone Multi-Frequency (DTMF) Tests**	**Call Forward: (1st party = calling; 2nd party = 1st called party; 3rd party = the call-forwarded-to party) Verify CLID delivery**
5.1	Outbound RFC2833 to PSTN (access IVR and verify) for G711 and G729a	Verify both menu selections works.
5.2	Inbound RFC2833 to SESL for G711 and G729a	Verify menu selection works.
5.3	Outbound in-band RTP DTMF to PSTN for G711	Verify menu selection works.
5.4	Inbound in-band RTP DTMF for G711	Verify menu selection works.
6	**Voicemail**	
6.1	Leave voice mail from PSTN: Verify that DTMF (inband RTP and RFC2833 or G729a and G711) works	Don't answer the call and leave a message. Verify voice mail works.
6.2	Retrieve voice mail from PSTN: Verify that DTMF (inband RTP and RFC2833 for G729a and G711) works	Make a call from PSTN to voice mail and retrieve voice mail. Verify voice mail retrieval works.
7	**Network Redundancy (use G729a calls) Loadsharing and Priority method ; have traffic in both directions. Measure PDD**	
7.1	Check that standard call flow works in non-failure mode	Set priority mode and make a call from PSTN to Location 1 and verify call is established and RTP is good.

7.2	Check that failover method works when the 1st Trunk Group is totally congested.	Make a call(s) to ensurelocation 1 is congested. Then make a call from PSTN to location 1 and Verify call is rerouted to location 2
8	**Auto Attendant using G711 and G729a**	
8.1	Dial by extension from PSTN	Verify that the call is established to the AA and verify the dialing by extension.
8.2	Dial by name from PSTN	Verify that the call is established to the AA and verify the dialing by Name. Note: The CLID last name and first name must be used for dial by name to work properly
9	**Conferencing**	
9.1	3rd party is extension (all G711)	Verify 3 parties are conference
9.2	Third party is PSTN (all G711)	Verify call is rerouted to location 2.
9.3	Third party is PSTN using G729a	Verify 3 parties are conference
9.4	Conferencing of participants using mix of G729a , G711	Verify 3 parties are conference
10	**Call Hold**	

10.1	PSTN call can be put on hold and then returned to successfully with and without Music On Hold	Make a call from 9723965051 to 4693876501. Verify call can be held on either side and then returned with and without Music On Hold.
11	**Call Transfer: 1st party = calling party; 2nd party = 1st called party; 3rd party = the call transferred-to party**	
11.1	Verify the call is established from PSTN Location extension and then blind transfer to 2nd extension	Verify the call is established from PSTN Location extension and then blind transfer to2nd extension
11.2	Call Transfer of an inbound call from PSTN to extension – consult	Verify the call is established from PSTN Location extension consulted transfer to 2nd extension
11.3	Call Transfer of an inbound call from PSTN to PSTN – blind	Verify the call is established from PSTN to Location extension and then blind transfer to PSTN
11.4	Call Transfer of an inbound call from PSTN to PSTN – consult	Verify the call is established from PSTN to Location and then consulted transfer to PSTN
11.5	Call Transfer of an inbound call from extension to PSTN – blind	Verify the call is established from extension and then blinded transfer to PSTN
11.6	Call Transfer of an inbound call from extension to PSTN - consult	Verify the call is established from extension to PSTN and then consulted transfer to PSTN
12	**Session Refresh**	
12.1	Check that Session Timer negotiation works correctly	Make a call and Verify session timer negotiation works correctly.

12.2	Check the frequency of session refresh reInvites / Updates as per the negotiated values	Set Session Expires=120 seconds. Make a call, answer the call and keep it up. Verify re-INVITE after 60 seconds
12.3	Session Tear Down: Keep a call active; disconnect the WAN link and wait for N=SESSION-EXPIRE seconds, if the call is cleared	Do the same step as 12.2, and unplug WAN link. Verify call will be cleared after 60 seconds.
13	**Fax**	
13.1	Check that fax calls using only G711 work correctly for calls originating from each direction	Make a fax call. Verify fax-thru succeeds.
13.2	Check that fax calls start with G729a codec for voice call and negotiate correctly for T38 for fax calls originating from each direction for Group3 and Super group3 fax machines	Make a fax call. Verify T.38 fax call succeeds.
14	**CALEA**	
14.1	Test with Subsentio that intercepts can be placed and ccd and cdd collected	Must be pre-arranged

XO makes no representation of warranty, expressed or implied, as to the accuracy or completeness of the information contained in this document, and nothing contained herein is, or shall be relied upon as, a promise or representation.

Give us a call today at 877-617-2111 or visit xo.com/esip

XO Communications is a leading nationwide provider of advanced communications, managed network and IT infrastructure services for business, large enterprise and wholesale customers. These customers include more than half of the Fortune 500, in addition to leading cable, mobile wireless and domestic and international telecommunications companies. XO offers a superior customer experience through its innovative data and IP solutions, its employees' focus on customers and the proven performance of its advanced network.

To learn more about XO Communications, visit www.xo.com or blog.xo.com. XO Communications is also on Twitter, YouTube, Facebook and LinkedIn.

XO communications™ Solutions you want. Support you need.