Interoperable SIP Trunk Providers

While SIP interoperability in general has made significant progress, it is still required to test interoperability with each and every ITSP to make sure all the required features for successful SIP trunking work. the following is the current list of ITSPs that we were able to test against. This list highly depends on the availability of suitable test accounts. Should your preferred provider be missing and you would like to get it added, please say so on the sipx-dev mailing list. If you can provide a test account together with your request, then that would increase the chances of getting help with resolving interoperability issues.

Required parameters for different ITSP

Provider (ITSP)	Domain ITSP	Server Address	Global addr	Registrar	Reg On Init	Default Asserted Identity	SIP keepalive	RTP keepalive
bandwidth.com	ot.bandwidth.com	n/a (blank)	true	n/a	false	true	CR-LF	NONE
att.com	Specified by AT&T	n/a (blank)	true	n/a	false	true	CR-LF	NONE
bt.com	sip.ser-001.nat.bt.com	81.144.230.5	false	default	true	true	CR-LF	replay last sent packet
cbeyond.net	sipconnect-fca.atl0.cbeyond.net	sip-proxy-fca.atl0.cbeyond.net	true	default	true	true	CR-LF	NONE
voxitas.com	wdc01a.netlogic.net	wdc01a.netlogic.net	false	default	true	true	CR-LF	NONE
sipcall.ch	voipgateway.org	n/a (blank)	true	default	true	true	CR-LF	replay last sent packet
callwithus.com	sip.callwithus.com	n/a (blank)	true	default	true	true	CR-LF	empty packet
voipuser.org	sip.voipuser.org	n/a (blank)	false	default	true	true	CR-LF	empty packet
les.net	did.voip.les.net	n/a (blank)	false	default	true	true	CR-LF	empty packet
eutelia.it	voip.eutelia.it	n/a (blank)	true	default	true	true	CR-LF	empty packet
vitelity.net	vitelity.net	outbound.vitelity.net	true	inbound8.vitelity.net (supplied by ITSP)	true	true	CR-LF	empty packet
vitaltalk.com	vitaltalk.com	chicago-1a.vtnoc.net	true	default	true	true	CR-LF	replay last sent packet
voip.ms	voip.ms	n/a (blank)	true	default	true	true	CR-LF	NONE
callcentric.com	callcentric.com	204.11.192.31 (pick same registrar and proxy address) false	204.11.192.31	true	true	CR-LF	NONE`	
simplesignal.com	sip.myvtel.com	n/a (blank)	true	default	true	uncheck (leave PAI blank)	CR-LF	NONE
bluesip.net	bluesip.net	leave blank	true	default	true	true	CR-LF	NONE
unlimitel.ca	unlimitel.ca	sip.unlimitel.ca	true	default	true	uncheck (leave PAI blank)	CR-LF	replay last sent packet
flowroute.com	sip.flowroute.com	n/a (leave blank)	true	default	true	true	CR-LF	replay last sent packet

An entry in the table does not guarantee that the ITSP is well behaved or of good quality.

An entry in the table indicates that somebody (for example, member of the sipXecs user community) has tried out the ITSP and has reported that it worked and has shared their settings.

Templates for many of the above providers are provided for easy configuration of the system.

SipXbridge has been tested successfully for interoperability with Nortel CS1000 (release6) systems. In addition to the settings in the table above, you will also need to configure your Caller-ID setting for the account as described below.

The SIP trunking capabilities of sipXecs (or sipXbridge) should extend far beyond the list of ITSP included in the table above. There are simply too many ITSPs so that we cannot test with them all. You can help extend the list. We are in the process of defining test procedures for ITSP interoperability testing and certification.