

ALICE Receptionist Guide for ALICE Cloud Communications Services

To connect the ALICE Receptionist system to use the ALICE Cloud Communications Service for communications, the following steps must be completed:

- 1. Ensure your **network and firewall** are configured to support SIP traffic (Video and Audio packets) between the ALICE Receptionist System and the ALICE Cloud Communications Servers:
 - Add the necessary rules to your corporate firewall to allow both UDP and TCP traffic inbound and outbound to the following addresses on the following ports:
 URL ADDRESS: na.sip01.alicereceptionist.com PORT: 5060 (North America)
 URL ADDRESS: na.sip02.alicereceptionist.com PORT: 5060 (North America)
 URL ADDRESS: na.sip03.alicereceptionist.com PORT: 5060 (International)
- 2. Ensure the ALICE Receptionist system can reach the following ALICE License Servers:
 - a. ALICE License Service:
 - URL ADDRESS: https://webservices.alicereceptionist.com Port: 443 b. Counterpath License Service:
 - URL ADDRESS: https://secure.counterpath.com
- 3. **OPTIONAL: Test the SIP Extension provisioned for the ALICE system:** Download and install counterpath's **free** x-lite softphone from counterpaths (www.counterpath.com) website or from this address:

http://counterpath.s3.amazonaws.com/downloads/X-Lite 4.9.5.1 81564.exe

Once installed configure the X-lite softphone to use the SIP Extension provided for the ALICE Directory. Verify you are able to do the following in the X-Lite software:

- a. Register the SIP extension on the ALICE Cloud Communication Service
- b. Place a call from the softphone to a desk phone or another softphone.

Attempt the type of call you intend to use ALICE for (audio or video) to one of your endpoints and verify that the SIP traffic is

able to successfully traverse your network. Once you are able to verify calls are successful with the X-Lite client, you're network is correctly configured for the ALICE Receptionist system.

SIP Account				
				×
Account Voicemail	lopology	Presence	Transport	Advanced
Account name: SIP_T	est			
Protocol: SIP				
Allow this account fo	or ———			
Call				
IM / Presence				
User Details				
* User ID:	100			
* Domain:	na.sip01ali	ereceptioni	st.com	
Password:	•••••			
Display name:	100			
Authorization name:	100			
Domain Proxy				
Register with don	nain and rece	eive calls		
Send outbound via:				
Domain				
Proxy Address	:			
Dial plan: #1\a\a.T;ma	itch=1;presti	rip=2;		
			_	_
			ок	Cancel

Port: 443



If your test show that the audio or video communications is not working on the x-lite client or the endpoint, you will need to review your firewall configuration to determine what is preventing the SIP traffic from successfully traversing your network.