



ALICE™ RECEPTIONIST

Communications Guide

ALICE Receptionist guide for Avaya phone systems

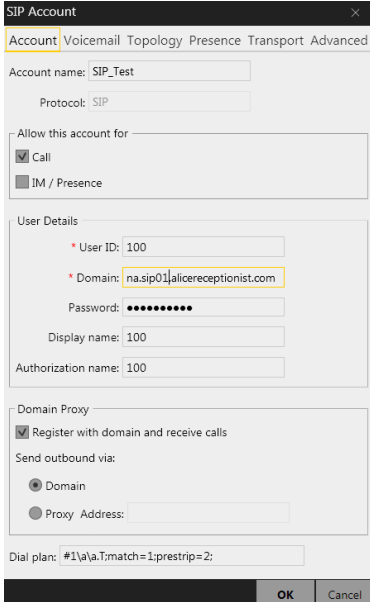
To connect the ALICE Receptionist system to use your Avaya phone system for communications, the following steps must be completed:

1. **Provision** a SIP softphone extension on the Avaya system that can be used by the ALICE Receptionist Directory software.
2. Ensure your **network and firewall** are configured to support SIP traffic (Video and Audio packets) between the ALICE Receptionist System and other endpoints on your network. Endpoints can include phones or softphones that are SIP enabled. Typical configurations include:
 - a. Use of port 5060 for VoIP SIP traffic across UDP or TCP protocols
3. 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`
PORT: 443

4. **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 provisioned SIP Extension that was created in step #1 of this document. 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.

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.



SIP Account

Account: Voicemail Topology Presence Transport Advanced

Account name: SIP_Test

Protocol: SIP

Allow this account for:

Call

IM / Presence

User Details

* User ID: 100

* Domain: na.sip01.alicereceptionist.com

Password: ●●●●●●

Display name: 100

Authorization name: 100

Domain Proxy

Register with domain and receive calls

Send outbound via:

Domain

Proxy Address:

Dial plan: #1|a.Tmatch=1;prestrip=2;

OK Cancel