VoIP Configuration Overview

Agents using VoIP phones must be located by the Virtual Contact Center system, so that calls may be sent to them. Two VoIP scenarios are supported:

Agents register with 8x8

In this common scenario, the agent's SIP phone registers with the 8x8 SIP registrar. This registration is done automatically, and allows the Virtual Contact Center system to locate the Agent's phone.

Note: Agents will be able to place outbound calls from the application only - not directly from the phone itself.

Agents register with a local SIP server

Some work environments have a complete SIP-based telephony system in place. In this case, the agent phone registers with the local SIP server. Calls to the agent are sent from Virtual Contact Center to this SIP server, which is able to forward the call to the agent.

Note: The only SIP server supported by Virtual Contact Center is Asterisk (www.asterisk.org).

See Technical Requirements for a list of supported SIP phones.

Configuring your SIP phone

The main idea behind SIP phone configuration is to make sure that there is a unique identifier that will locate your phone. This is done in two steps:

1. Configure your phone with a unique username, and some network information.
2. Enter a corresponding URI into the Agent Profile in the Virtual Contact Center application.

For detailed information on configuring Avaya, Cisco, Linksys, and Eyebeam VOIP phones, please refer to the respective documentation.

For information on registering with a local SIP server, please refer to the document - ‘Using a Local Registrar’.