Configuring Interactive Intelligence

ININ IP PBX

For **tw telecom**

SIP Trunking service

USER GUIDE

Version 1.0

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DISCLAIMER: tw telecom is not a reseller, provider, or maintenance support company for the telephone PBX system referenced in this guide. The following example was tested in a controlled environment and represents a basic configuration that works with tw telecom's eSBC and SIP Trunking service. Any change in the telephone system configuration is solely up to the owner of the equipment. Any support needed for the telephone system should come directly from the original equipment manufacture or authorized reseller.

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1 Introduction

The adoption of SIP Trunking has steadily grown because of cost saving initiatives and the explosion of IP Unified Communications. The growth of SIP Trunking however, created interoperability and edge demarcation (NAT/PAT) challenges between the service provider network and enterprise local area network. The enterprise Session Border Controller (eSBC) installed at the customer premises is solving these challenges by providing a demarcation point which protects both the enterprise local network as well as the service provider trunk. The eSBC plays the role of a Back-to-Back User Agent (B2BUA) and provides topology hiding, SIP ALGs, NAT/PAT translation, SIP normalization for IP-PBX interoperability, Remote Monitoring & Management capabilities and much more.

This document describes a basic configuration of the telephone systems listed below for interconnecting with the **tw telecom** SIP Trunking product. The provided configuration is an example only, and must be modified with customer specific network and application details.

- ININ version 4.0 to connect to **tw telecom** provided eSBC

1.1 Document Scope

The verification only applies to the product versions used in the test. The PBX is classified as interoperable in a configuration described in this document and for the features and functions communicated in the product literature.

This document describes the configuration to perform SIP Trunk interoperability between the **tw telecom** provided FVX eSBC and the listed telephone PBX system. The document contains short notes on SIP Trunk and the necessary PBX configuration, followed by any notes or limitations encountered during the testing process. The configuration examples used in this document are based on ININ version 4.0.

1.2 Assumptions

This document assumes user parameters and basic configurations are already in place in the IP PBX. Such configurations include:

- License Requirements
- System Configuration
- Server Configuration
- Dial Plan
- Phone Numbers (DID/EXT)
- Region
- Network Interfaces
- Additional Trunk Interfaces
- Codecs
- Trusted IP Ranges

1.3 tw telecom SIP Trunking Service Standards

The following table summarizes the supported protocol and signaling standards required for interoperability with the tw telecom SIP Trunking service. Guidance on configuring to these standards is contained in the following sections.

Codec	G.711ulaw		Numbering Plan	10-digit NANP
SIP port	5060			011 International
Transport Type	UDP		Secure Transport	No
RTP ports	20,000-39,999		Secure Signaling	No
Packet Size	20ms		Fax	G.711 Pass-through
DTMF	RFC2833 or inband media		Silence Suppression	No
Privacy	Remote-Party-id		SIP Early Offer	Yes
Redirected Calls	Diversion Header		SIP Early Media	Yes

2 Test Configuration

In the ININ SIP trunk architecture, shown in Figure 1 below, the ININ IPPBX acts as a SIP proxy through which all incoming and outgoing SIP messages flow through the FVX eSBC to the tw telecom network. There is no direct SIP signaling path between the eSBC network and the endpoints. ININ is configured to anchor the calls on the IP PBX. The phones are provisioned on a private network.

2.1 Network Topology

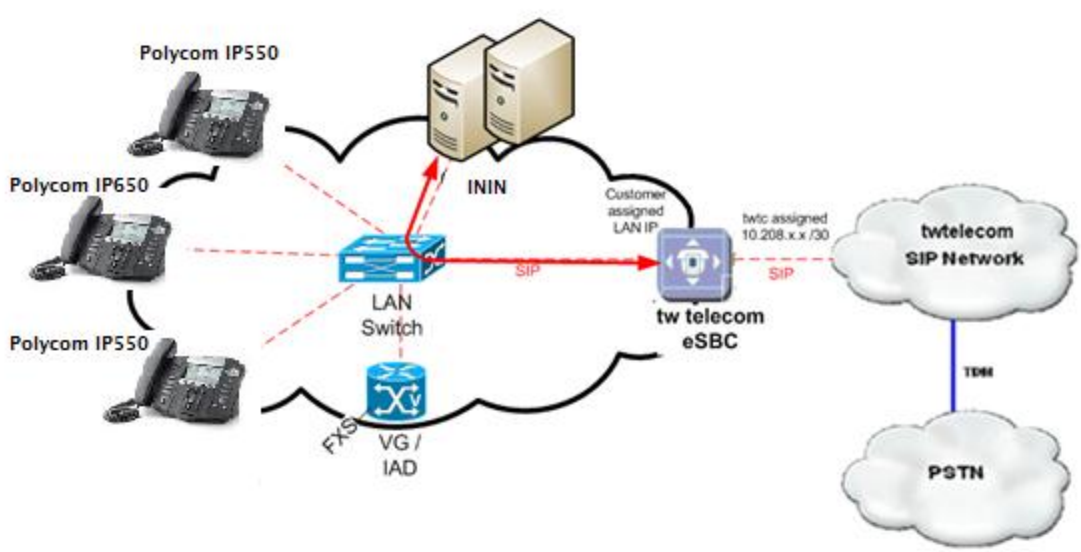


Figure 1: ININ

2.2 Customer Premise Devices

CPE Devices	Version
ININ IP PBX	4.0
FVX eSBC	5.0.1

2.3 Customer Phone Models/Versions

3rd Party Product/Components	Version
Polycom IP550/650 IP Phones	SIP 3.2.5.0643

2.4 System Components

Hardware Components

- ININ IC Server
- Polycom IP550/650 IP Phones
- The eSBC depicted in figure 1 is a tw telecom managed device. This device will be installed on the customer premise, and is the service point of demarcation.

Software Requirements

- Interaction Administrator 4.0.17.389

Consult your Interactive Inteligence representative for the correct hardware and license requirements.

2.5 Features

Testing was performed in accordance to **tw telecom**'s test plan with the following features verified to be supported:

Features Supported

- Basic G.711ulaw calls
- Inbound and Outbound local, long distance and international calls
- Calling Party Number Presentation and Restriction
- Calling Name
- Intra- and Inter-site Call Transfer
- Intra- and Inter-site Conference
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 Pass-through
- Outbound calls to IP and TDM networks

2.6 Application Notes / Known Limitations

Application Notes

The following are the application notes that apply for this configuration:

- **Anchor media.** The SIP Line requires setting the Audio Path as Always In (section 3.1.2)

Known Limitations

These are the ININ limitations:

ININ generates a Race Condition during an Inter-Site Call Transfer when one of the external parties hangs up. External User A calls ININ user. ININ User transfers to External User B. Either External User A or B hangs up the eSBC sends a BYE message to ININ and ININ responds to the eSBC with a 200 OK followed by an INVITE SDP message. At this point the ININ is out of RFC not following the Basic Call Setup flow. When the eSBC receives the INVITE SDP, it gets confused and tries to renegotiate the media with the BW sending the same INVITE SDP message. Since there is no call at the other end, BW keeps the other party waiting while sends multiple 200 OK SDP to the eSBC awaiting for more instructions until times out and finally releases the other party about 20 seconds later.

3 Sip Trunk Configuration in the ININ IP PBX

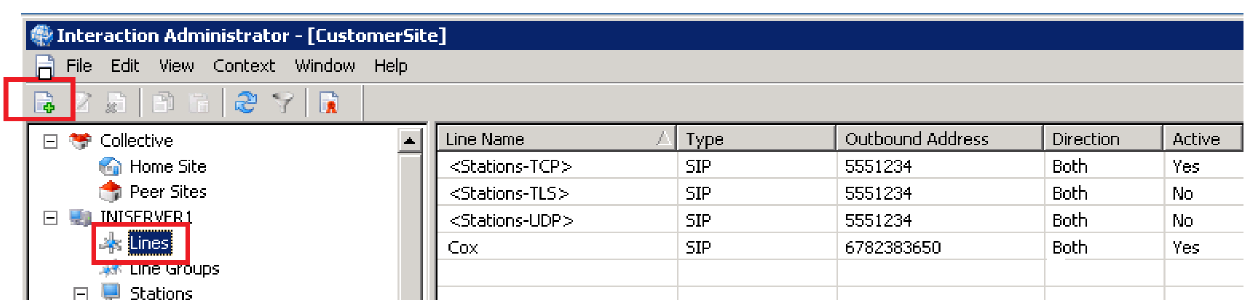
This section shows the configuration for a SIP Trunk on ININ with the eSBC. In order to start any configuration you must Login to the ININ server and open up the Interaction Administrator (IA) tool. **Note: any parameter not mentioned in this configuration guide will be treated with its default value.**

The following shows the sequence of steps to create a SIP Trunk using the IA tool:

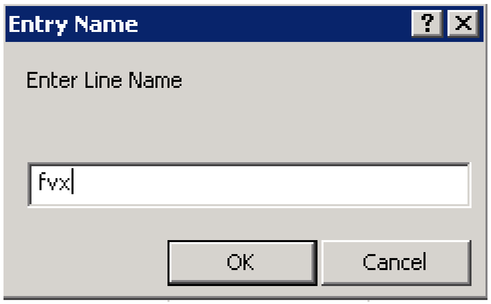
1. **Lines**
2. **Line Groups**

3.1 Lines

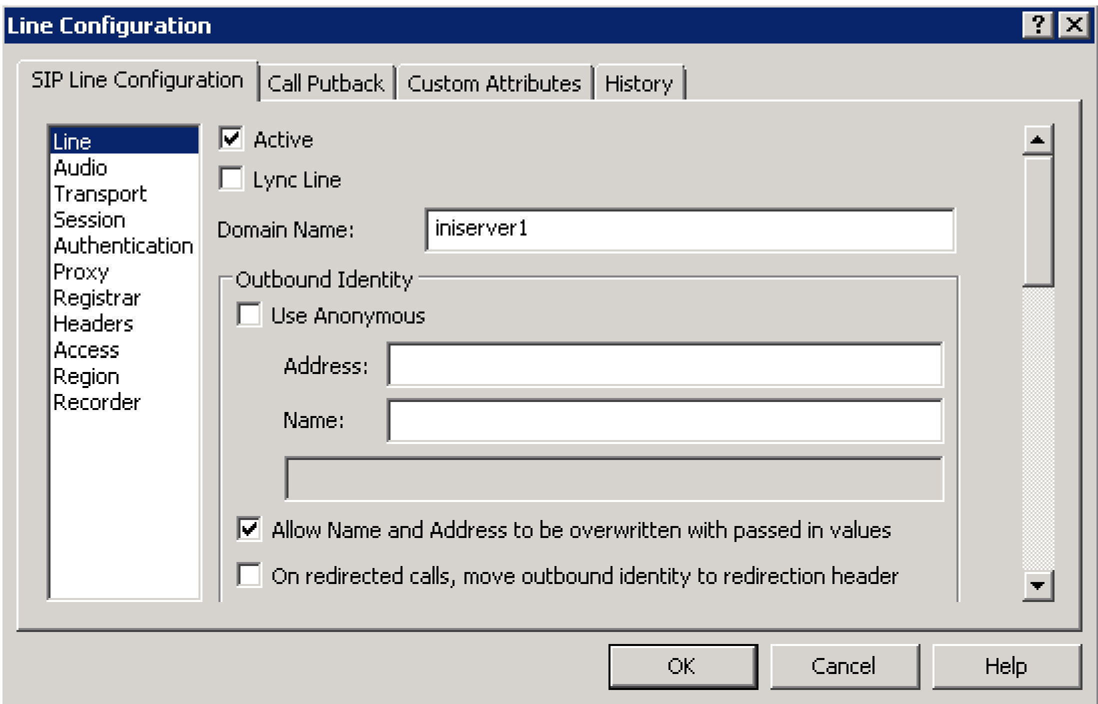
In the IA tool expand your ININ Server (INISERVER1 in this example) and click Lines. This will display all the Lines configured in the system on the right window and will give you access to create /edit your SIP Trunk. To add a new Line click on the little icon with the green plus sign on the upper left corner as shown in the image below.



Enter the significant name you want to call your SIP Line (e.g. eSBC, FVX, etc)



Once you click OK this will bring a new window with the parameters we need to configure.

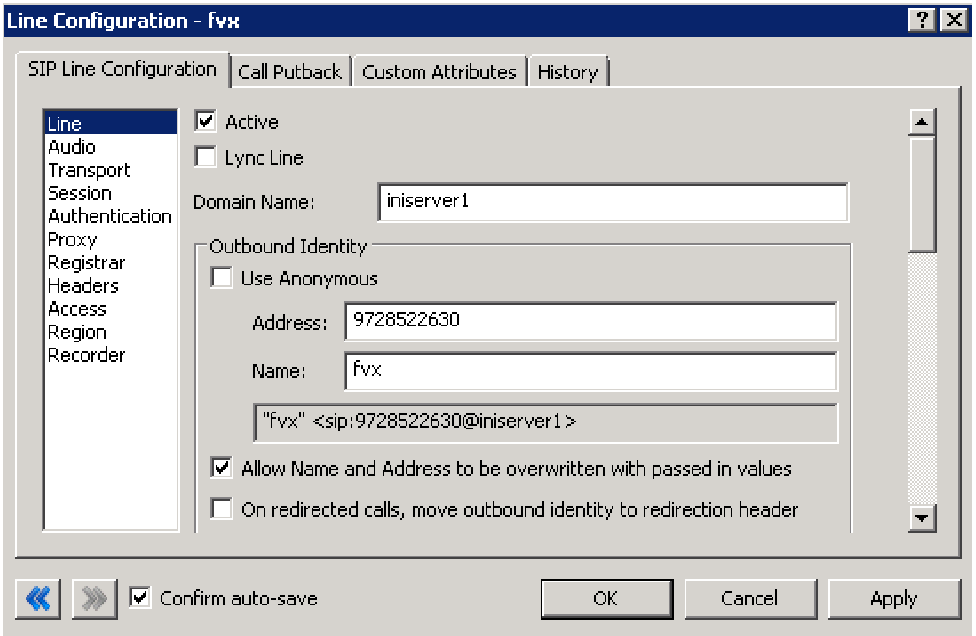


We are going to work only on the SIP Line Configuration tab. These are the significant parameters:

1. **Line**
2. **Audio**
3. **Transport**
4. **Session**
5. **Proxy**

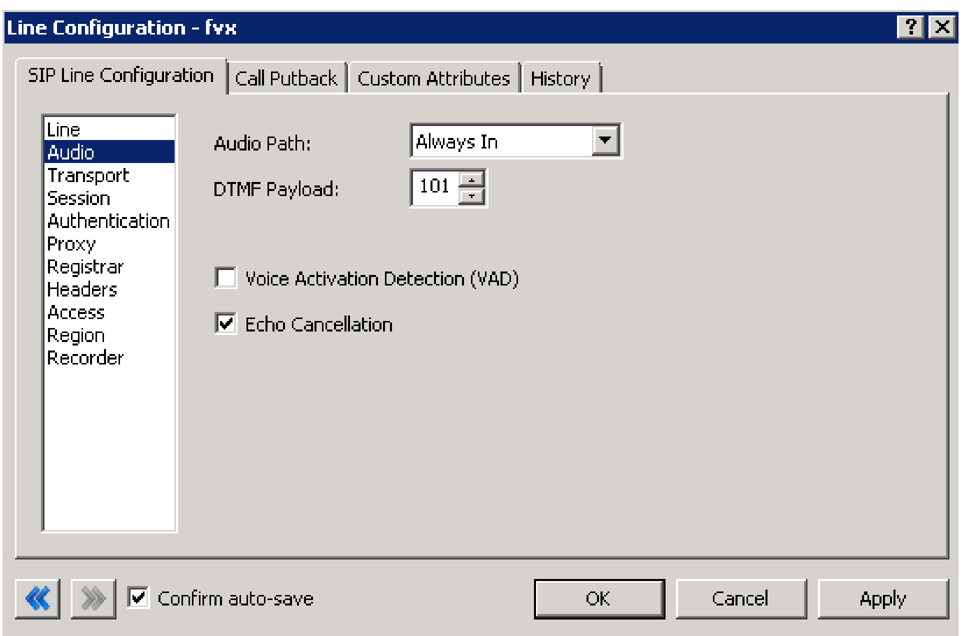
3.1.1 Line

Address: Enter your Pilot Number. For this example is 9728522630
Name: Significant name to call your Outbound Identity. For this example is fvx.



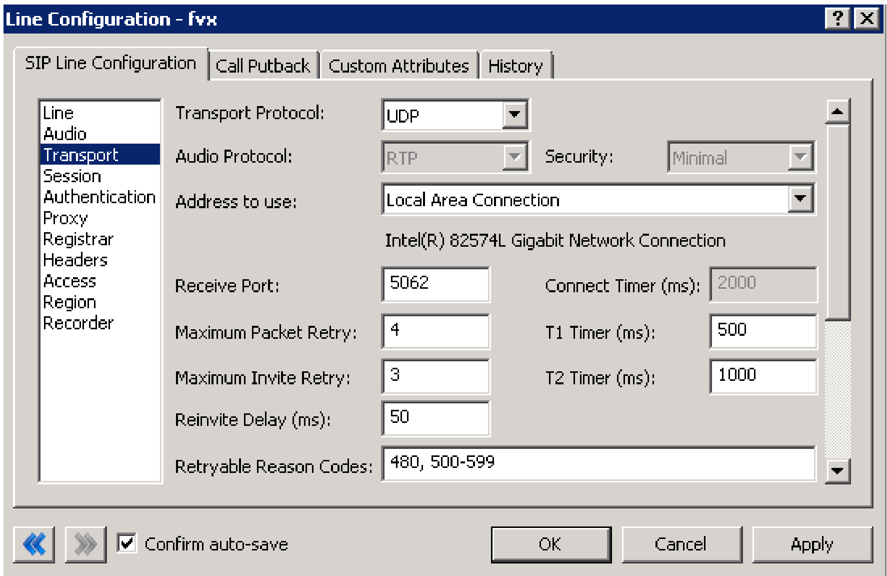
3.1.2 Audio

Audio Path: Always In



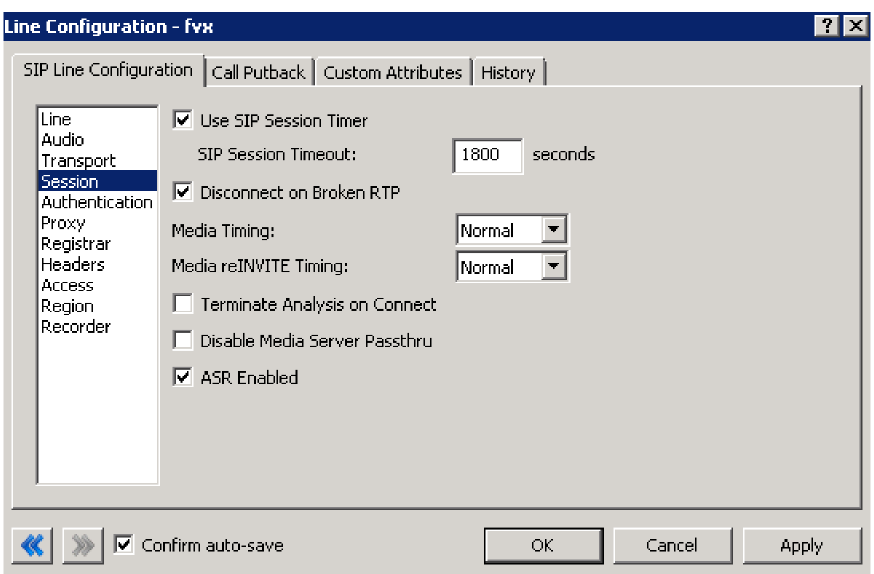
3.1.3 Transport

Transport Protocol: UDP
Address to Use: Always select the network interface that is enabled. In this example is Local Area Connection.
Receive Port: Check what is the next UDP port available. ININ allows one port per SIP trunk. For this example is 5062



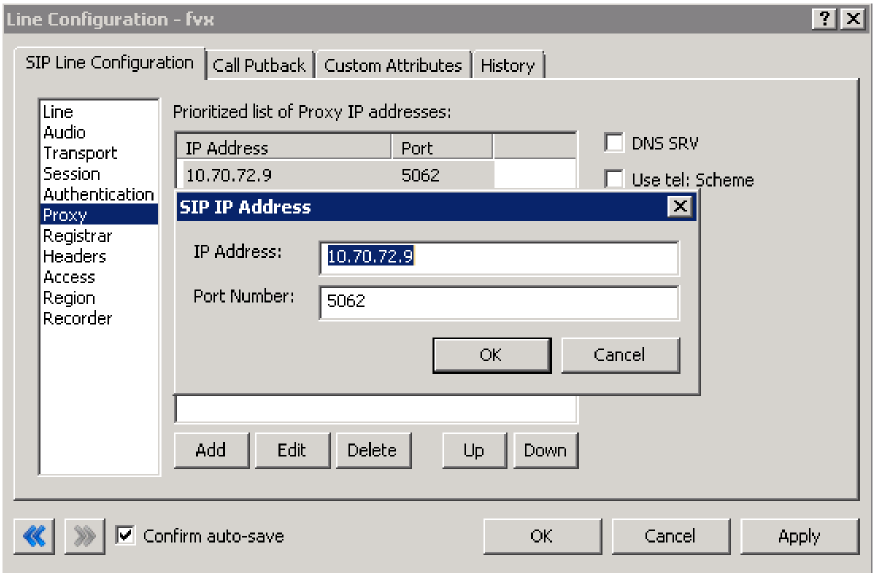
3.1.4 Session

Use SIP Session Timer: Checked
SIP Session Timeout: 1800 seconds

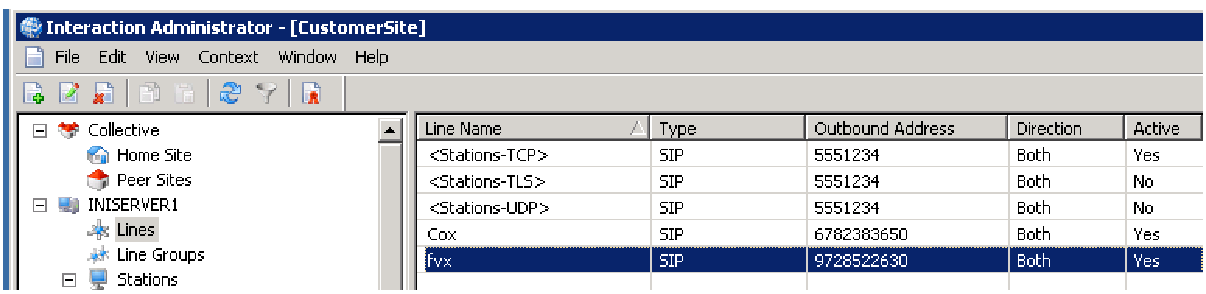


3.1.5 Proxy

You need to enter the LAN IP address of your eSBC and its UDP port. In this example IP is 10.70.72.9 and port 5062

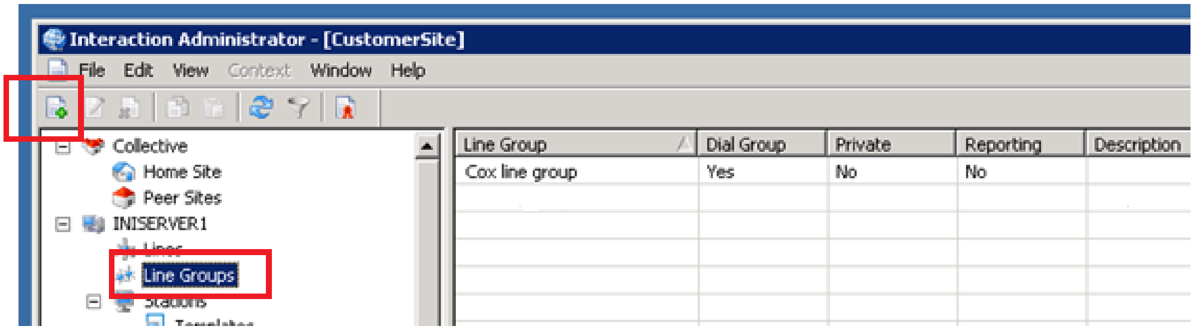


Click Apply then OK. You should see your new SIP Line listed.



3.2 Line Groups

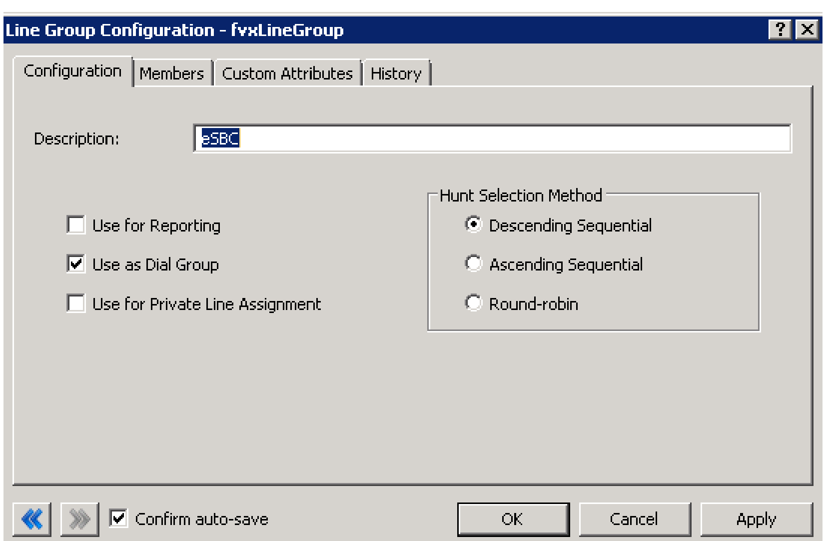
On the IA tool, switch to Line Groups as shown in the image below. To add a new Line Groups click on the little icon with a green plus sign located in the upper left corner.



Enter a significant name you want to call your Line Groups. In this example is fvxLineGroup.

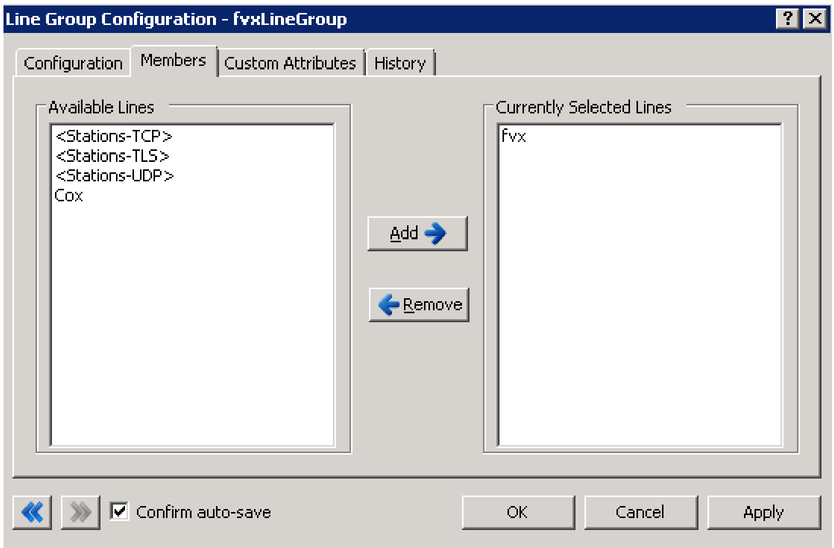
3.2.1 Configuration

Description: Enter a description that best suit the purposes of this group
Use as Dial Group: checked



3.2.2 Members

From the “Available Lines” Add your SIP Line to the right side “Currently Selected Lines” For this example is fvx.



Click Apply then OK and you should be able to see your Line Groups now listed.