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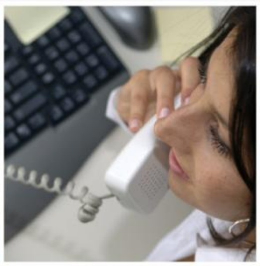
XO SIP Service

Customer Configuration Guide for
Interactive Intelligence Customer Interaction Center (CIC)
with XO SIP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1 Introduction

This document contains the test results of the XO SIP Trunking used to certify Interactive Intelligence – Customer Interaction Center IP PBX, which is primarily marketed to call center applications. This test verified the basic interoperability, features of the appliance, and the integration of the architecture framework and requires the Interaction SIP Proxy to be configured with the IP PBX.

2 Executive Summary

This report provides the test results found to date for the Interactive Intelligence – CIC (Customer Interaction Center) IP PBX SIP Trunking evaluation. The following is a summary of the issues and limitations found while performing the test.

Issues/Limitations:

- Call transfer (blind/consult) to PSTN need SCR patch to work properly.

  Solution: Issue was fixed after applying a patch to CIC system.

  "Issue was resolved in SCR IC-84759. SCR IC-84759 is available as part of the ES named SU12-ES_IC-85496 and this is available for the current release which is 3.0SU12. This SCR will also be included in the next Service Update which is 3.0SU13 and is scheduled to be released in about 2 weeks (end of Aug 2011)."

- DTMF requires RFC2833 to be enabled to work correctly. Do not check In-band DTMF box.

- If you want to use fax with package 1, please contact XO for configuration change.

- Call transfer (blind and consult) – Do not check Call Putback box if you plan to use IP Phones.

- Only Call Forward Always tested. Call forward busy/No answer was not tested.

- For outbound, if multiple CODECs are configured, only the first CODEC will be used to complete the call.

- Outbound caller ID block doesn't work.

Notes:

- By default '0' (operator) is for internal use only. Changes requires to send '0' out to XO.

- Interactive Intelligence doesn't require to dial '9' (or any digit) to grab outside line.

- Interactive Intelligence supports SIP 'Diversion' header. Original caller ID can be forwarded to PSTN in call forward scenarios.

- Interactive Intelligence preferred fax method is T.38 (relay). Fax over G.711 (pass through) was not tested.

- Analog phone and modem was not tested.

- Basic hunt group features were tested and passed.

Registration Method

Static registration is utilized between the Interactive Intelligence - CIC and the XO call agent.

XO SIP Service Packages Supported

Pkg	Codec	DTMF	Fax
1	G.711	RFC 2833 (In-band RTP DTMF not supported)	T38; G.711 pass-through not tested.
2	G.729a	RFC2833	T38; G.711 pass-through not tested.

3 Software and Hardware Equipment Requirements for Testing

1. Interactive Intelligence CIC (Customer Interaction Center)

   Version: 3.0.12

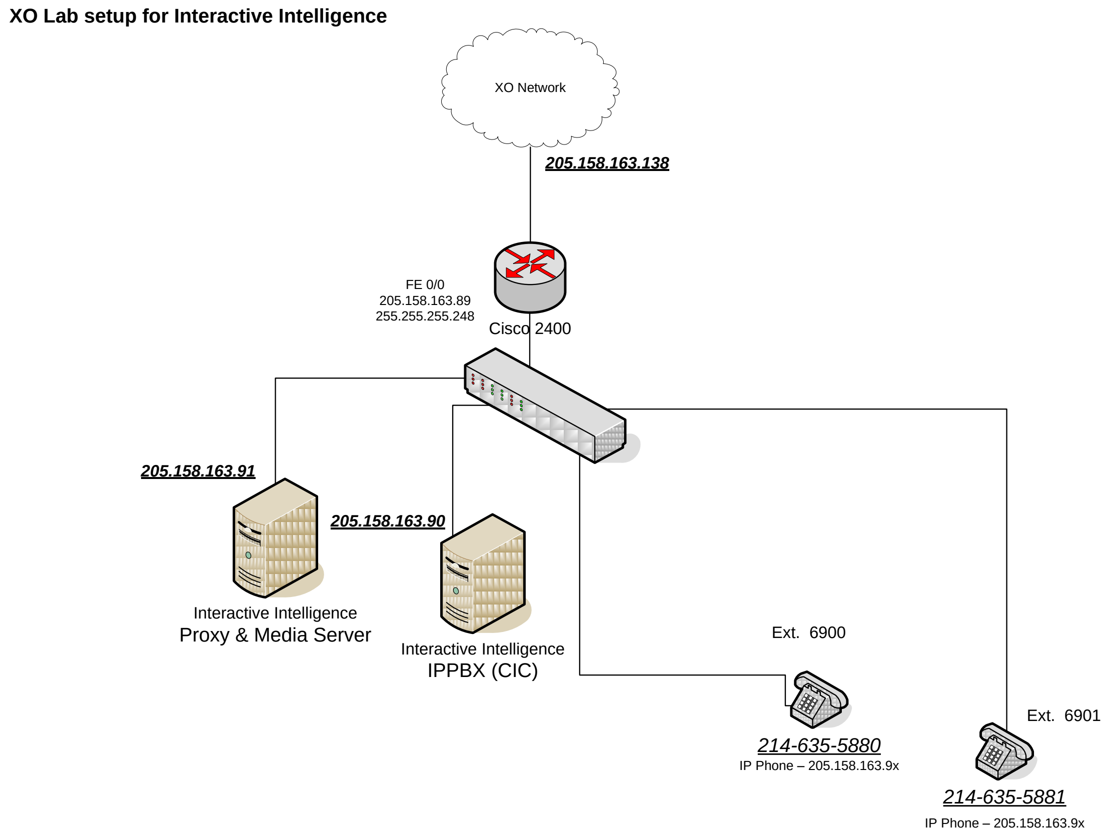
   Media Server version: 3.0SU12
   Proxy Server version: 4.6.1302.11055

2. SIP Phones

   Polycom Soundpoint IP 331 SIP

4 Test Configurations

4.1 The following diagram shows the configuration used during lab testing.



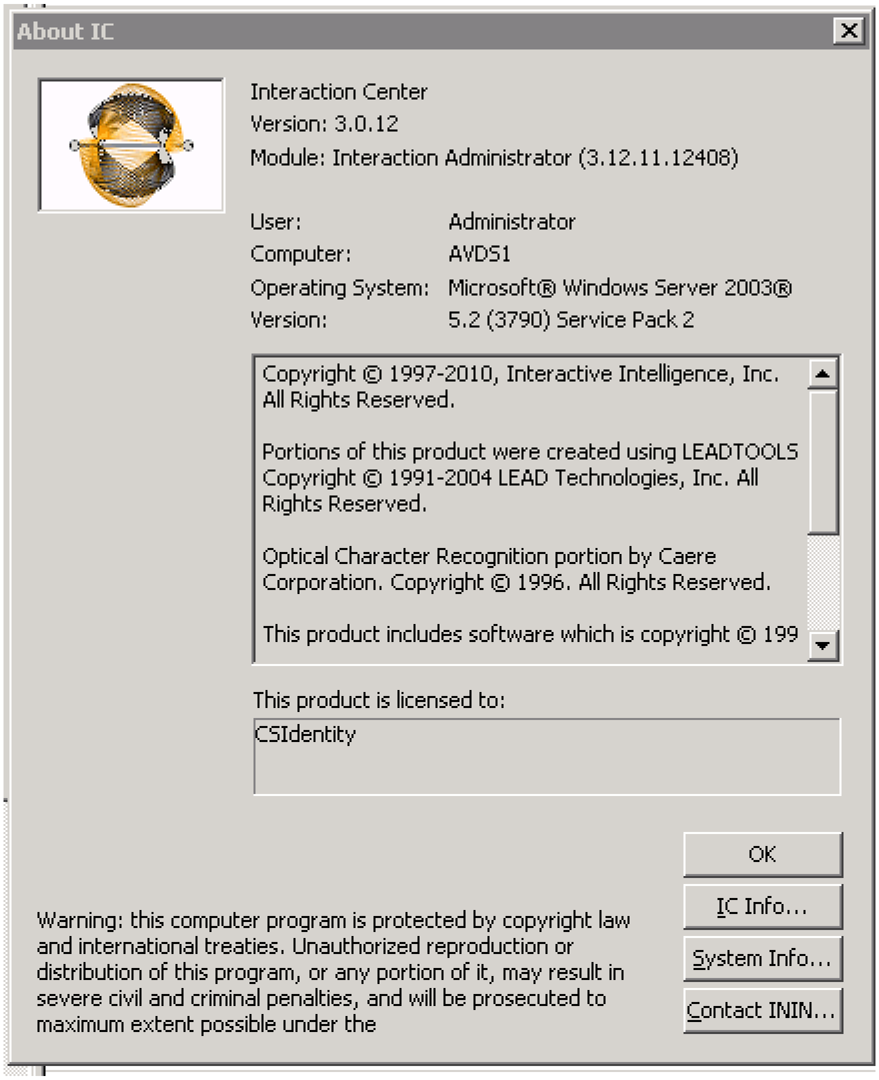
Notes:

- Above lab setup only shows main lab network elements.
- For this lab setup public IPs were used for Proxy, CIC and IP Phones so that vendor can access system remotely to help in lab testing. In real environment that may not be the case hence device with NAT functionality or NAT setup on XO router may require.
- For lab testing only one Interactive Intelligence CIC was used. Interactive Intelligence informed us that, CIC always gets setup in pair for redundancy. Interactive Intelligence proxy in front of those 2 CIC is must in that case as XO only supports one IP (as source and destination) at each SIP Trunk site.

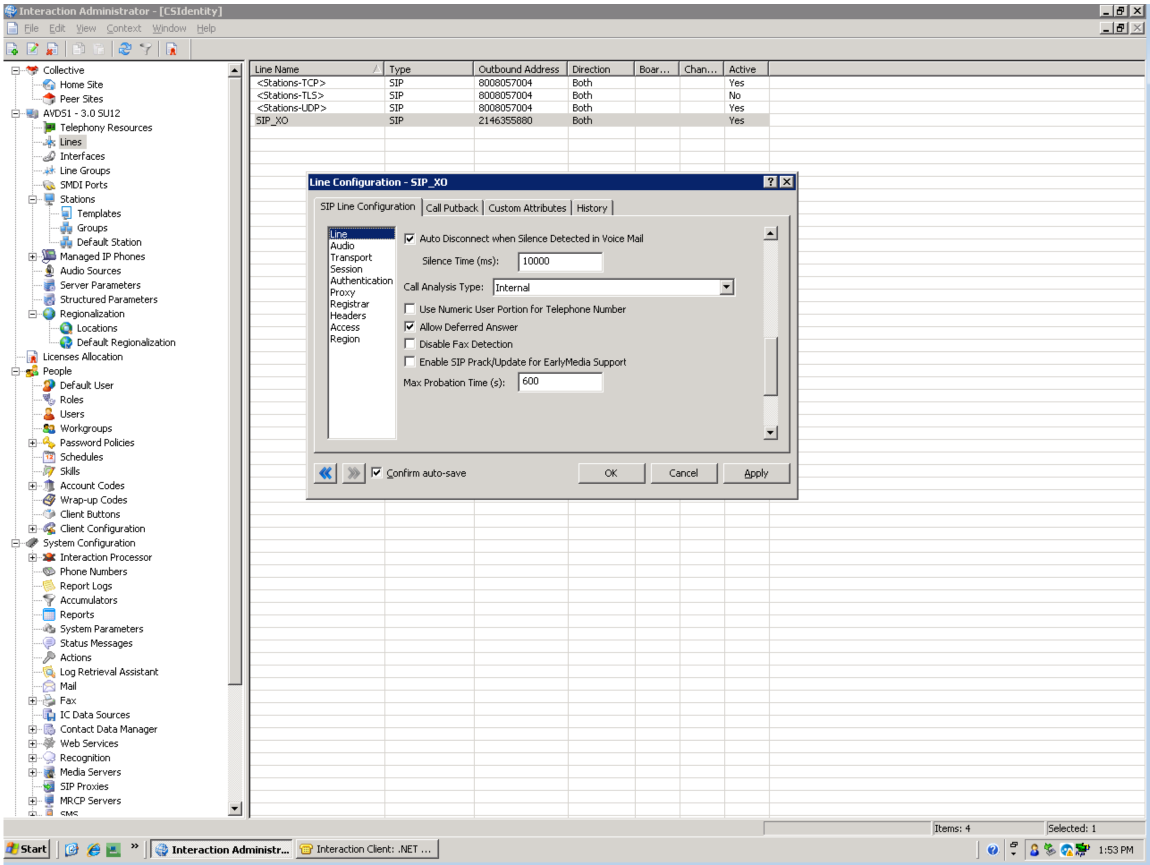
5 Test Bed Configuration Files

5.1 Interactive Intelligence – CIC (customer Interaction Center) basic Configuration.

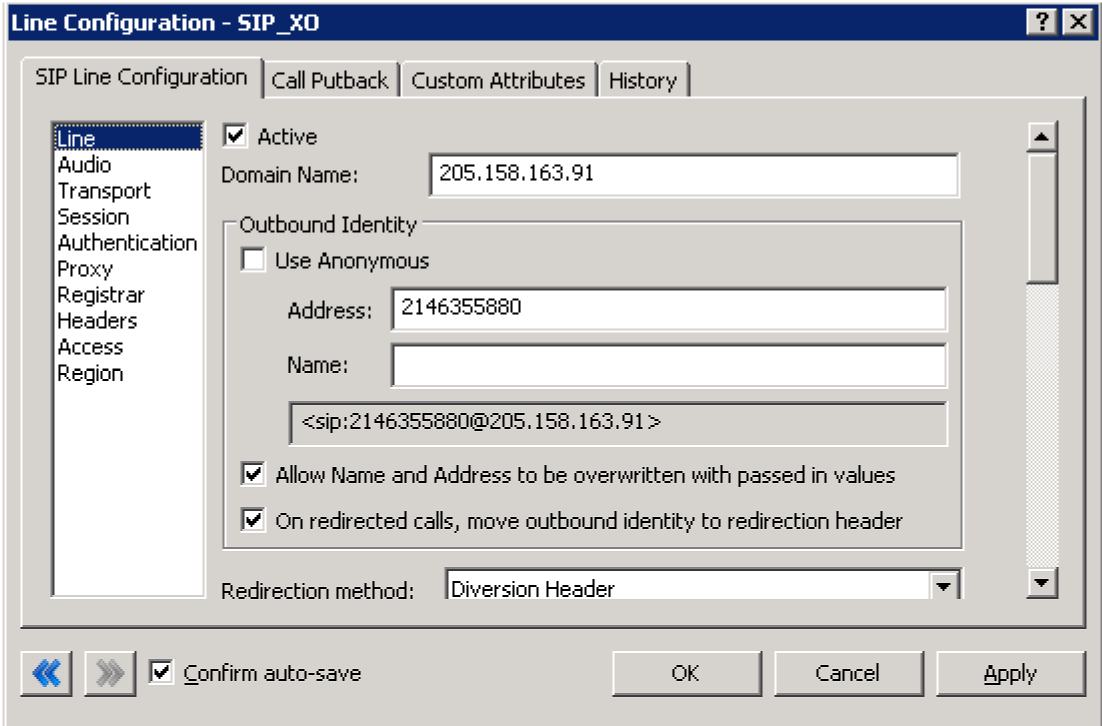
Interactive Intelligence – CIC version



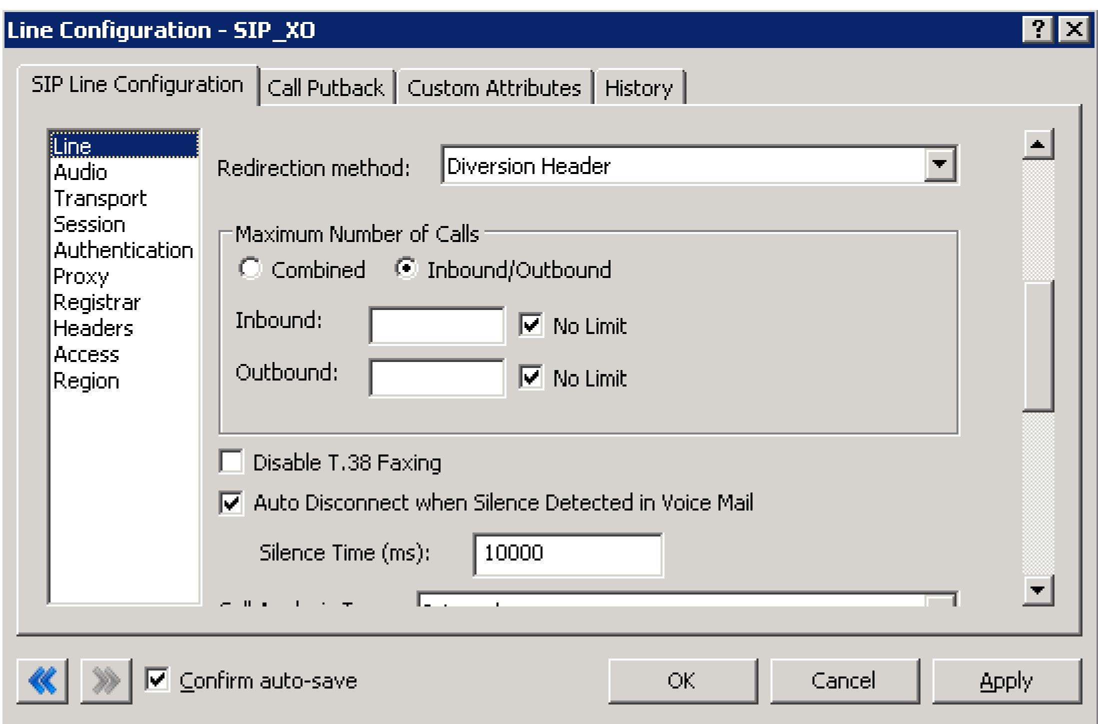
Line (Trunk) configuration:



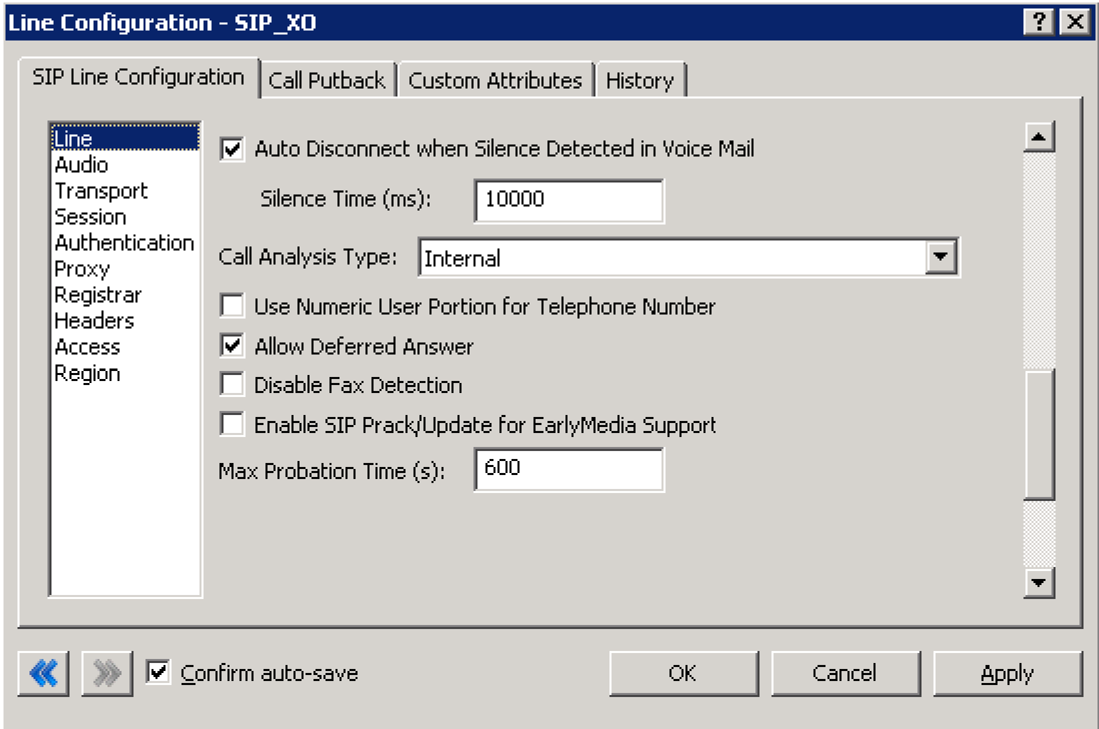
SIP Line configuration: Line



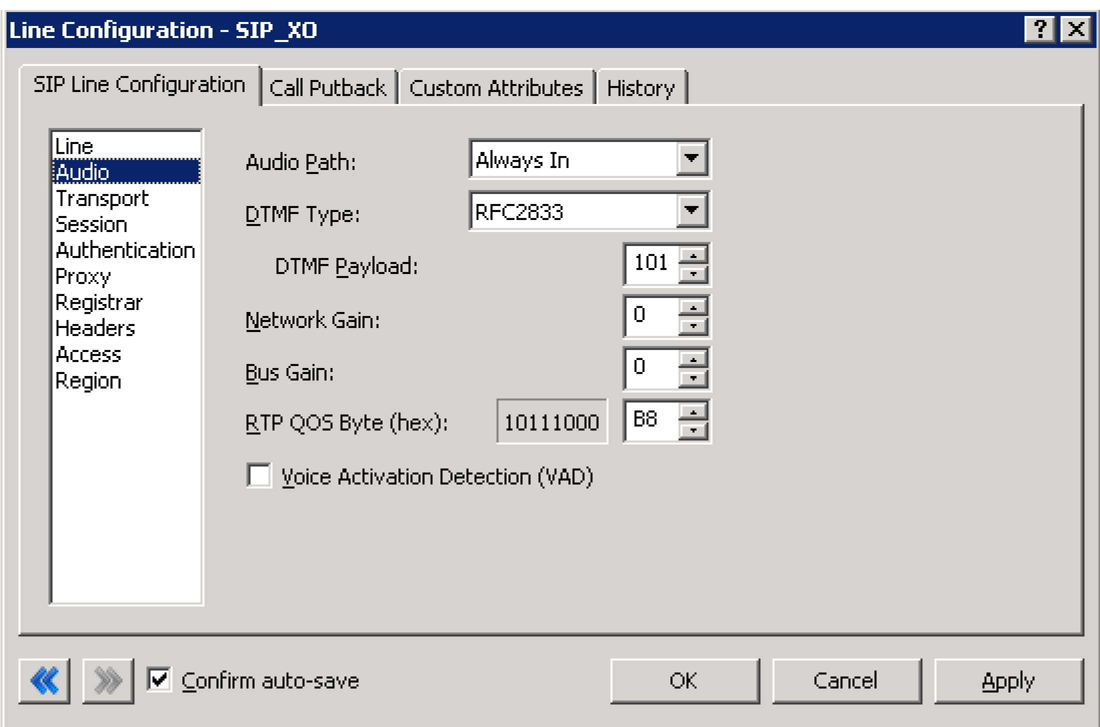
SIP Line configuration: Line (cont.)



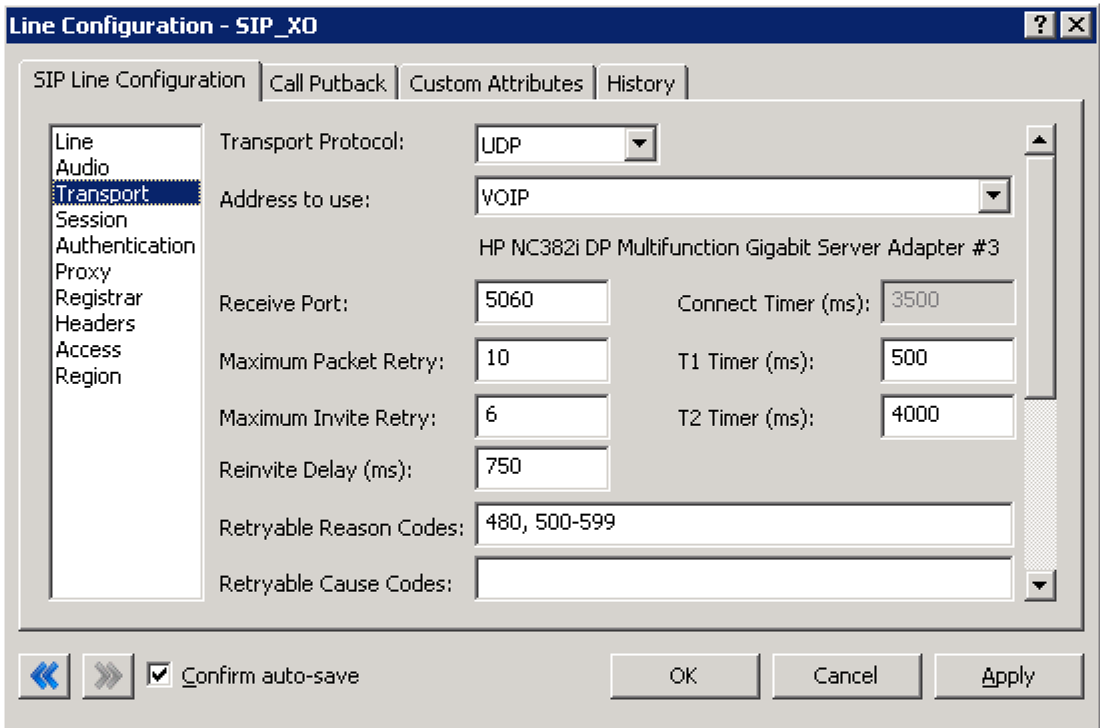
SIP Line configuration: Line (cont.)



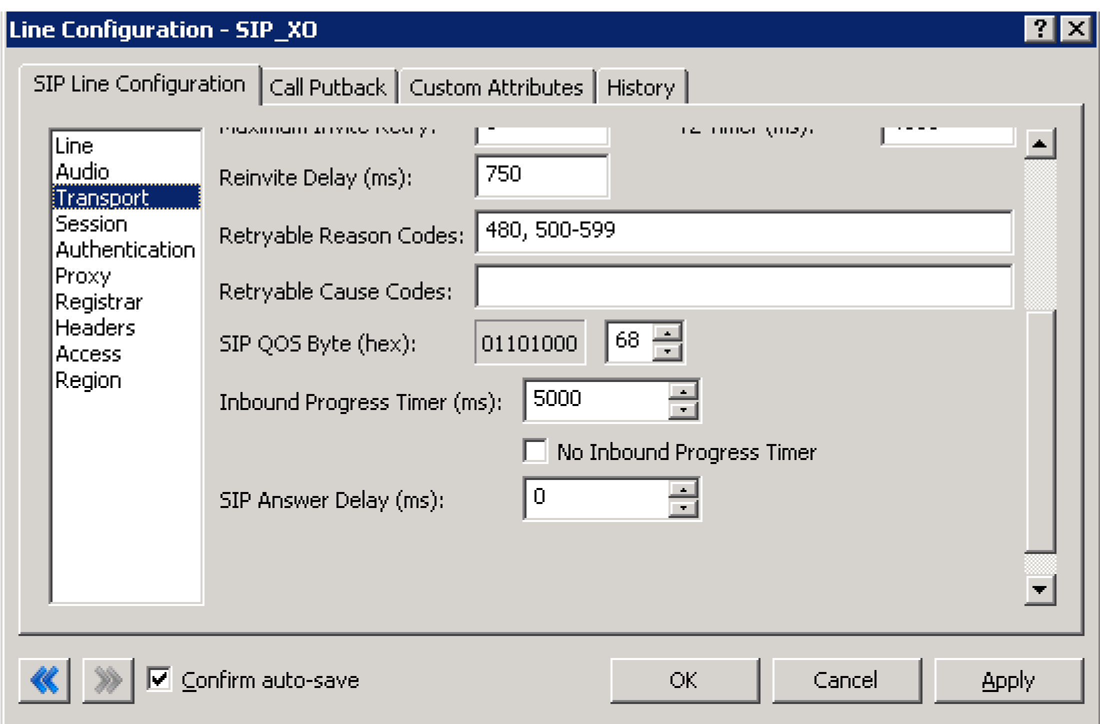
SIP Line configuration: Audio



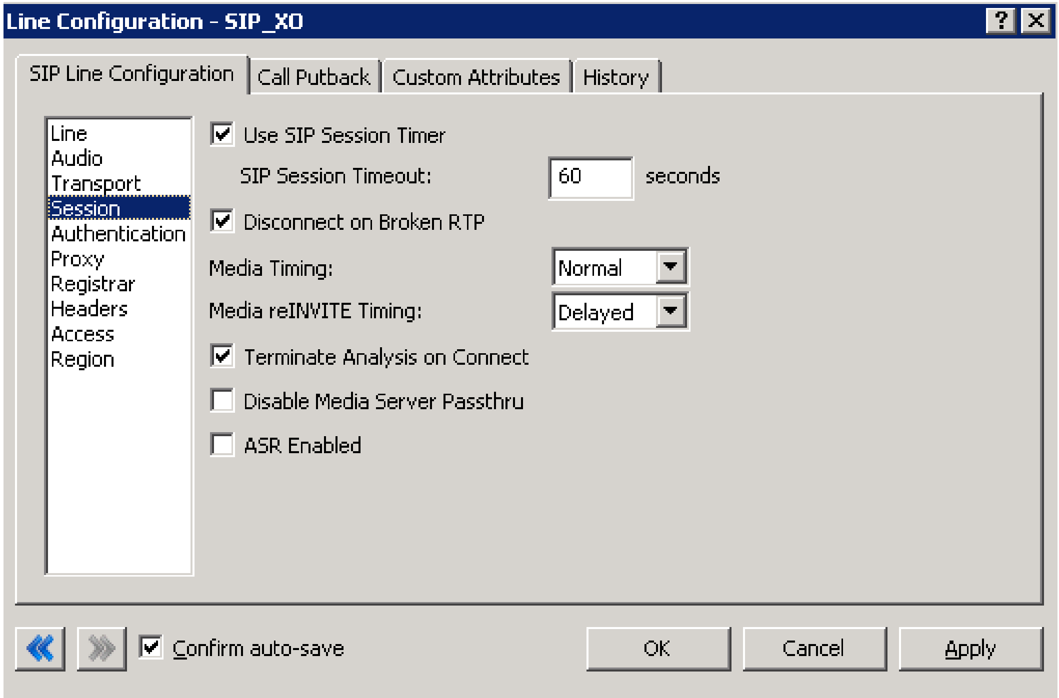
SIP Line configuration: Transport



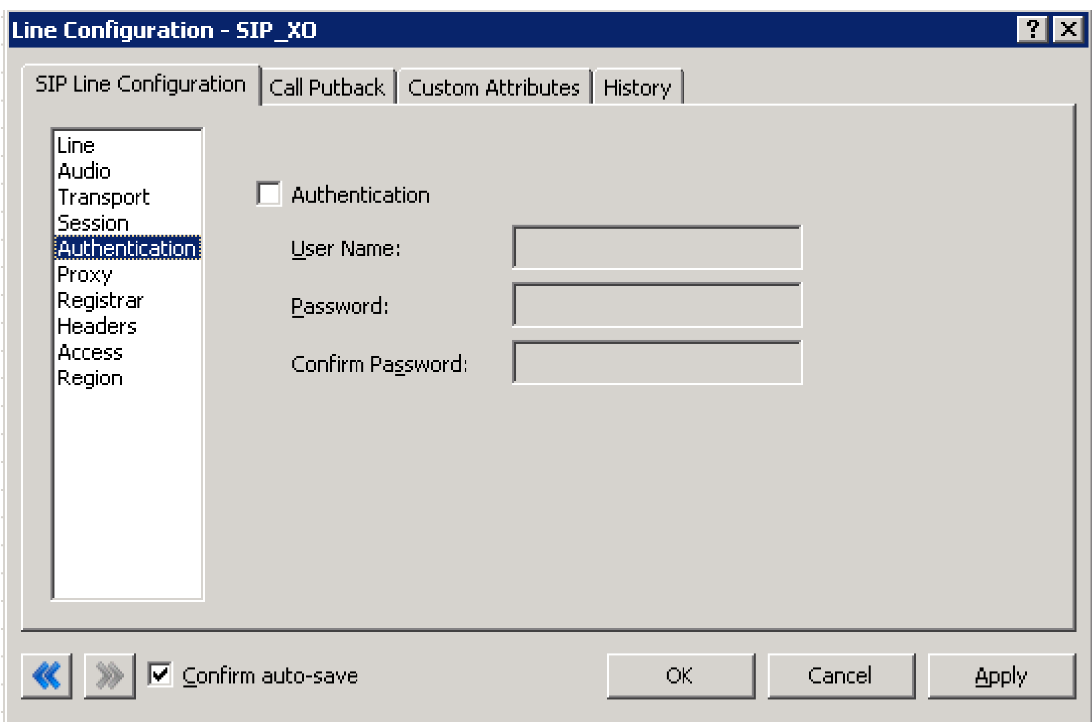
SIP Line configuration: Transport (cont.)



SIP Line configuration: Session

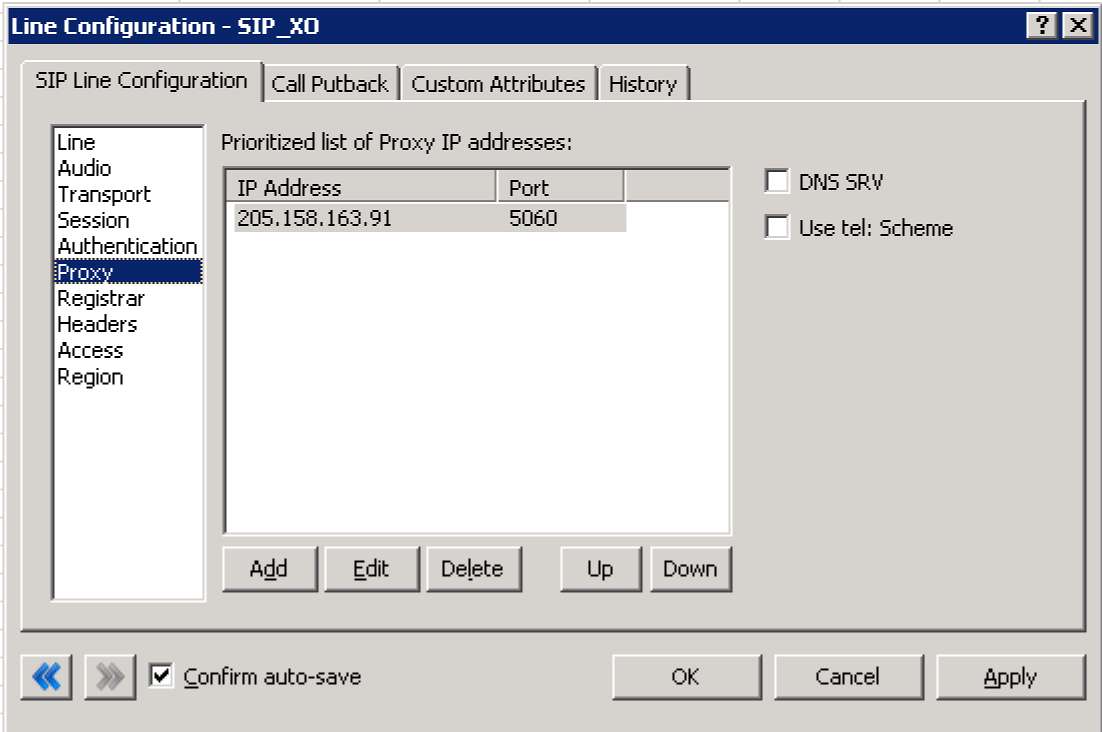


SIP Line configuration: Authentication

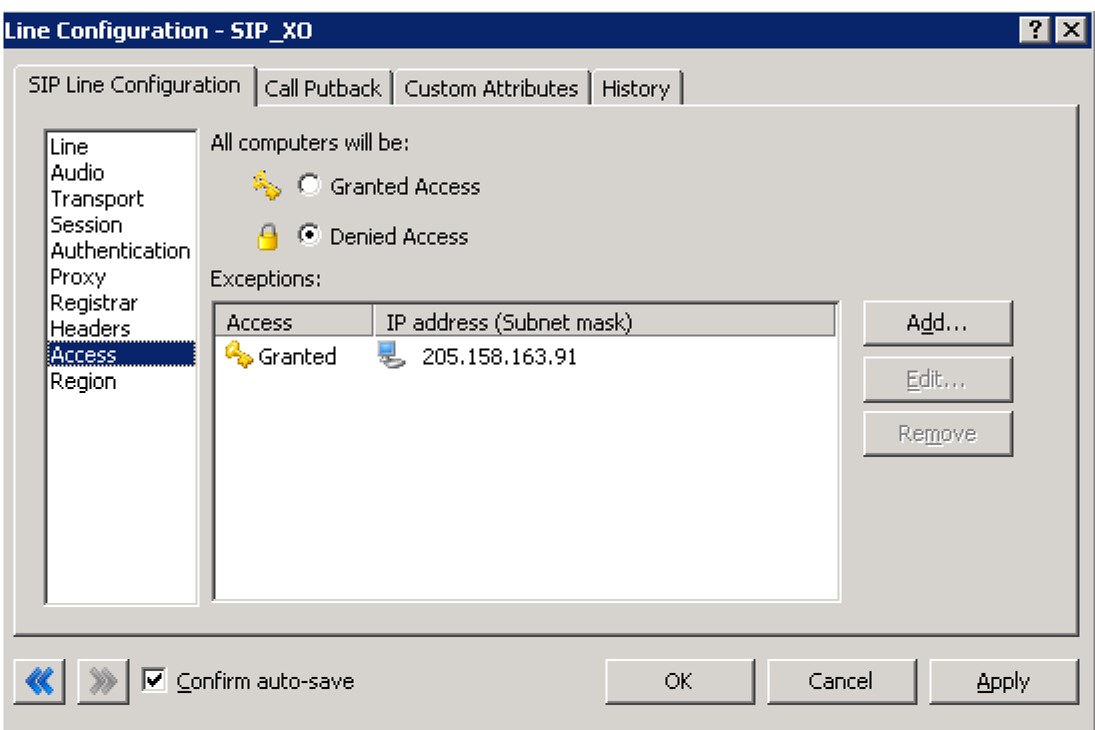


No authentication (username and password) requires with XO.

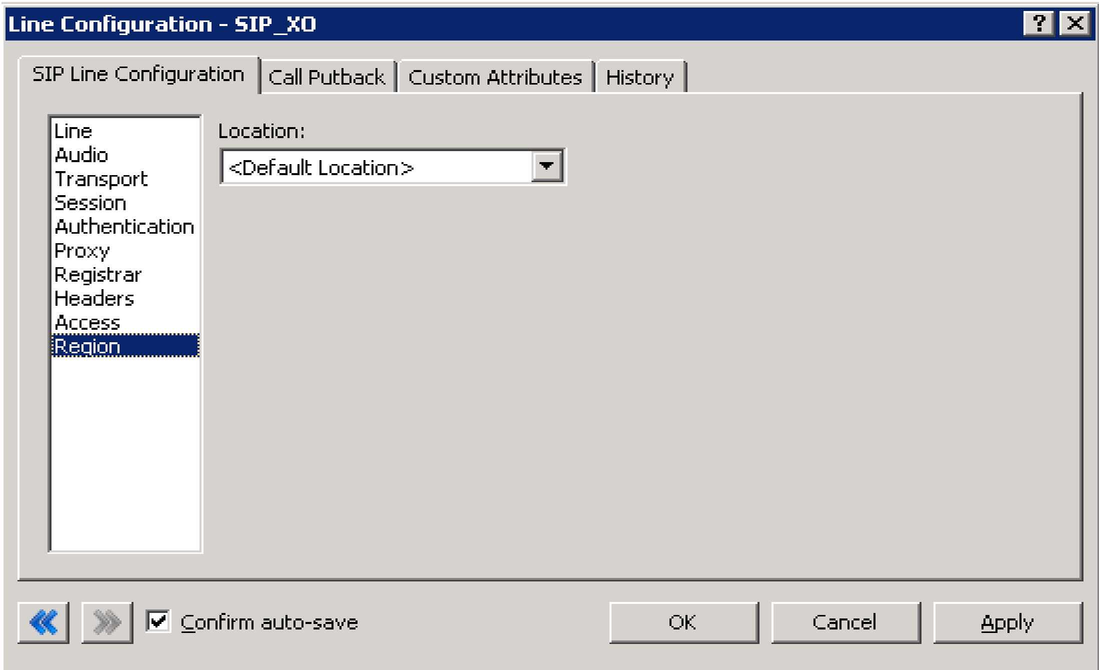
SIP Line configuration: Proxy



SIP Line configuration: Access

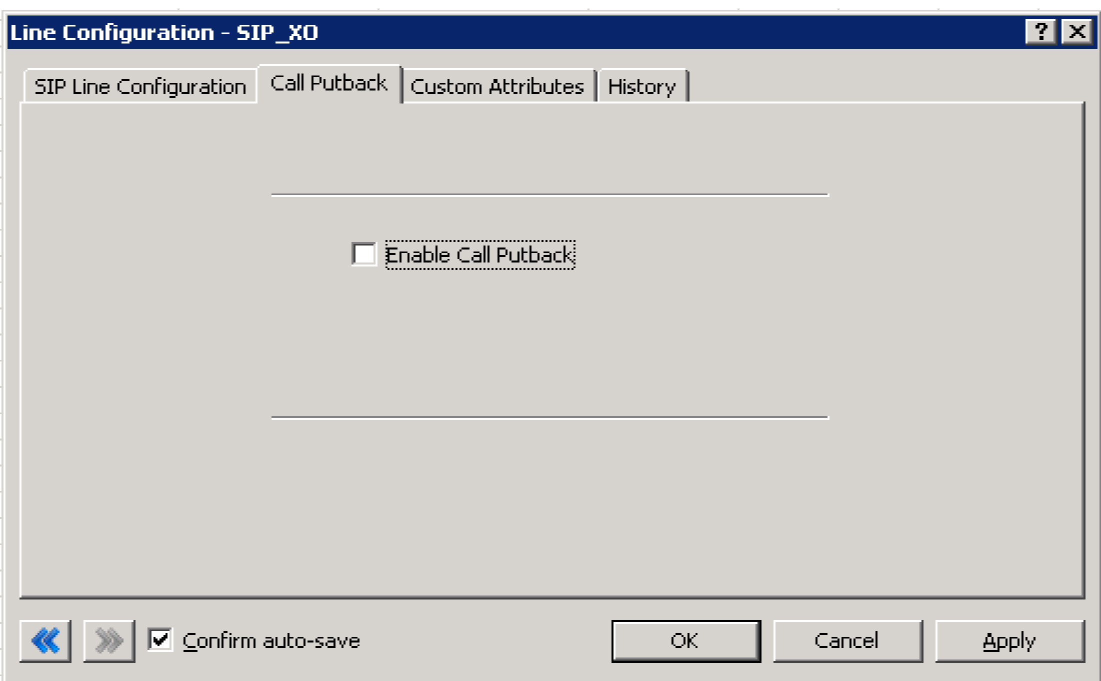


SIP Line configuration: Region

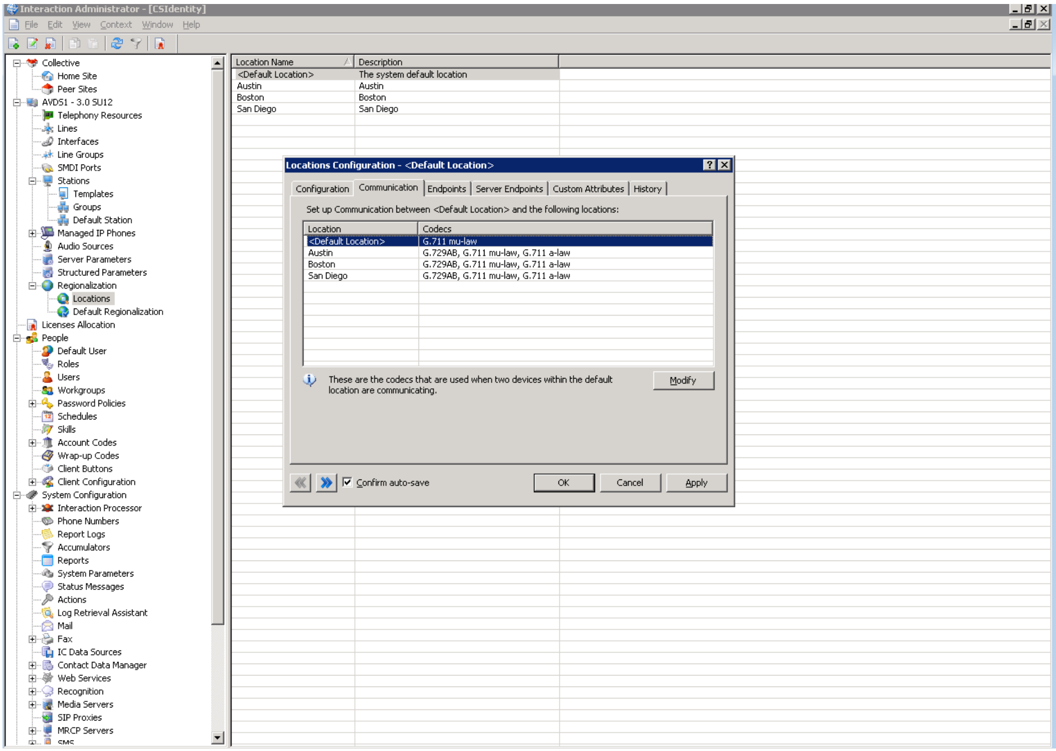


Line configuration – Call Putback:

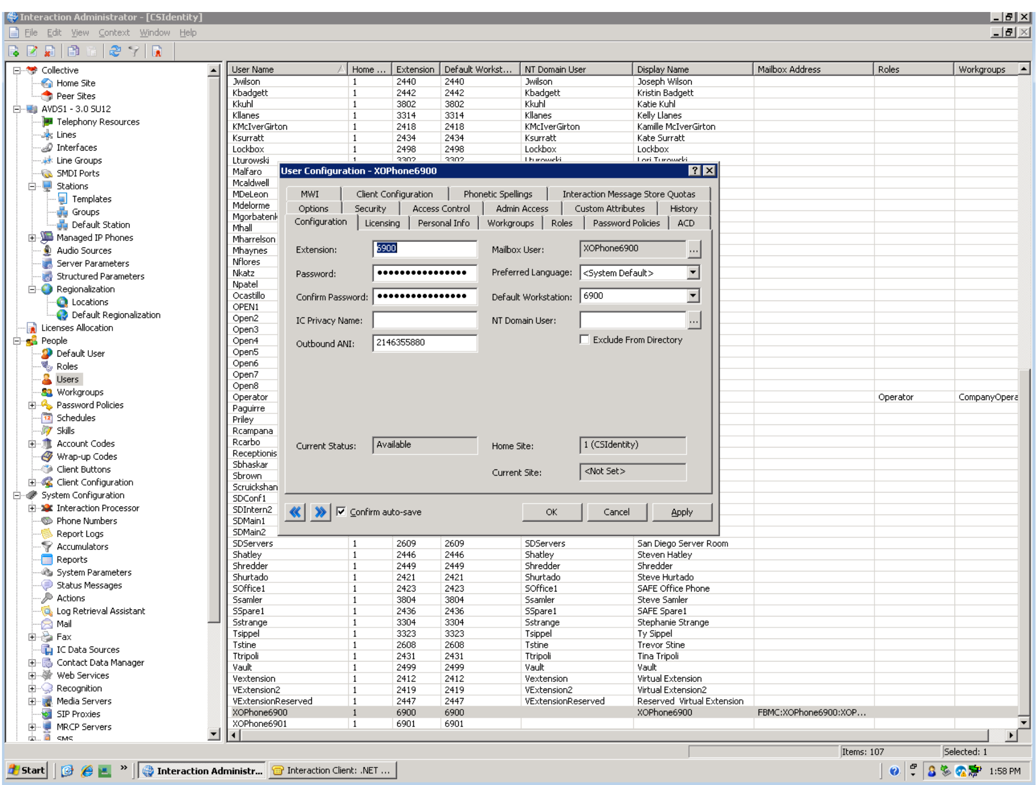
Check "Enable Call Putback" flag if transfer (blind/consult) with SIP REFER requires.



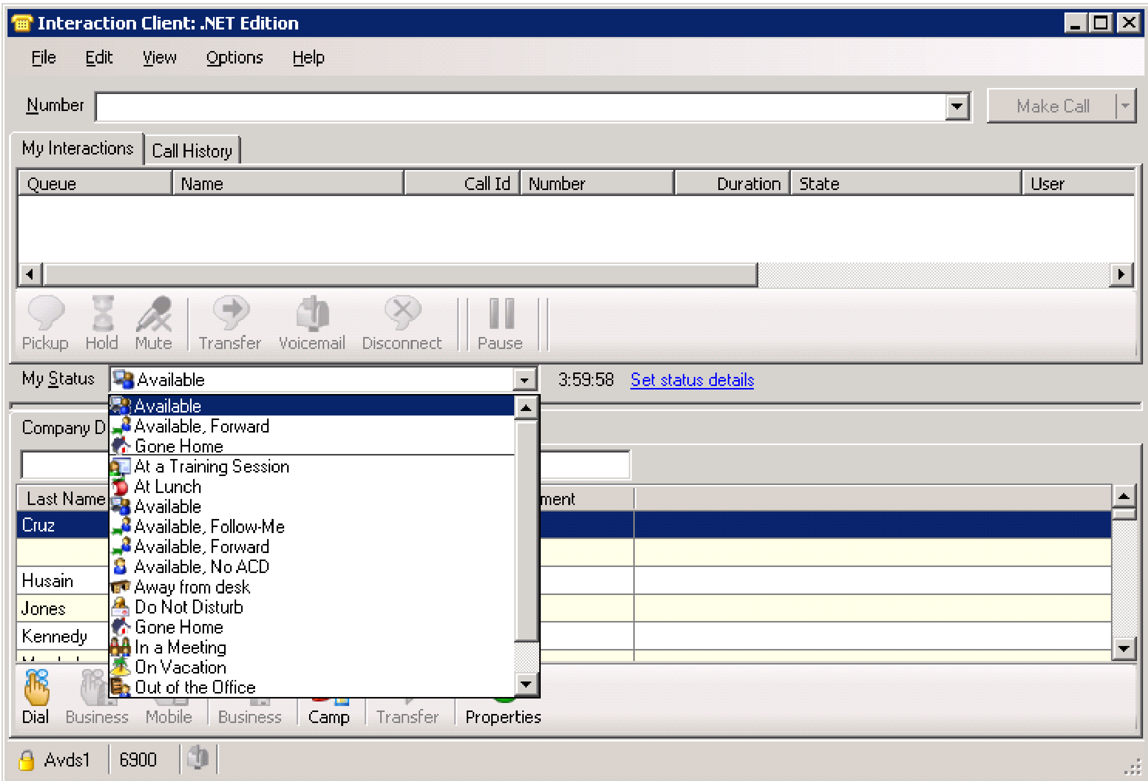
Regionalization >> Location >> <default location>



User >> XOPhone6900



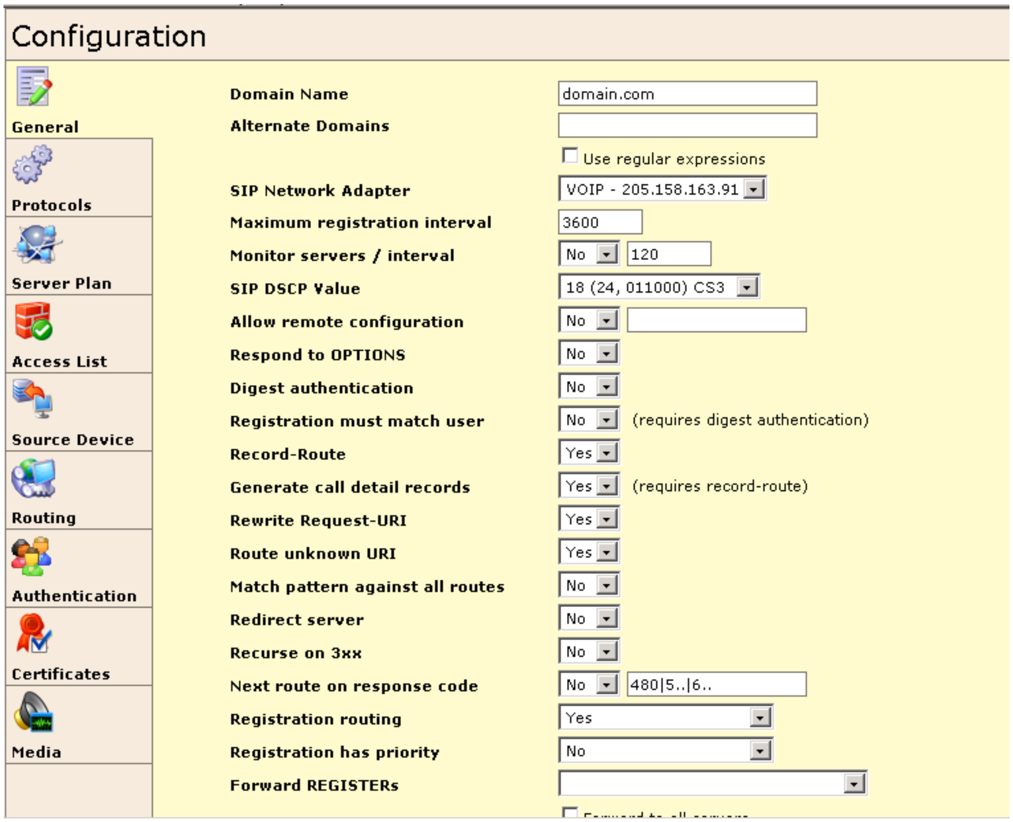
Interaction Client:



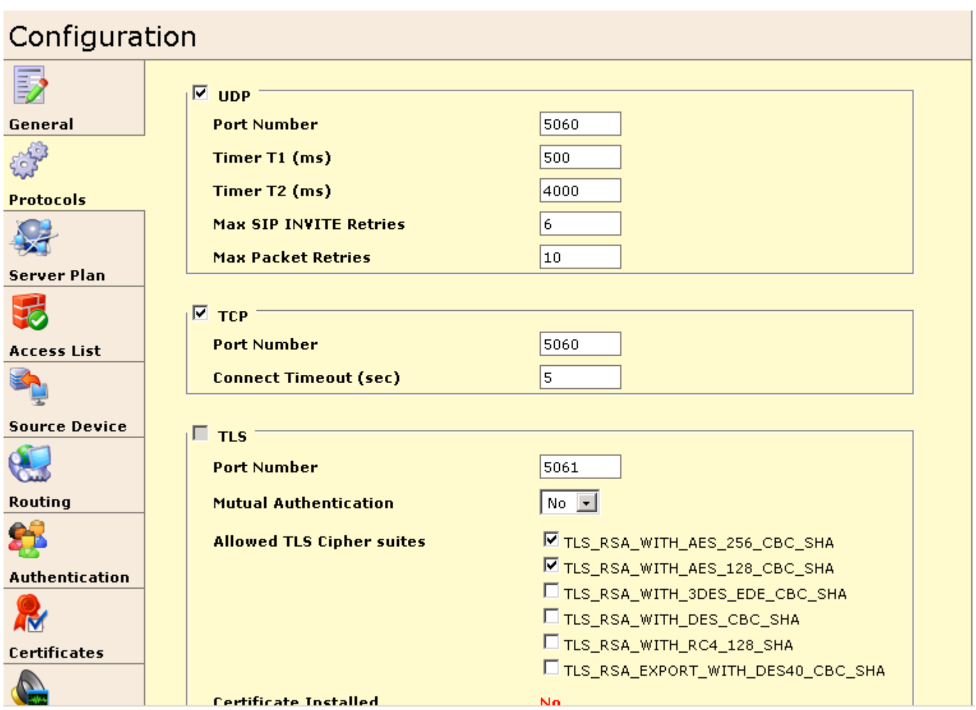
Interactive Intelligence SIP Proxy configuration

Version:

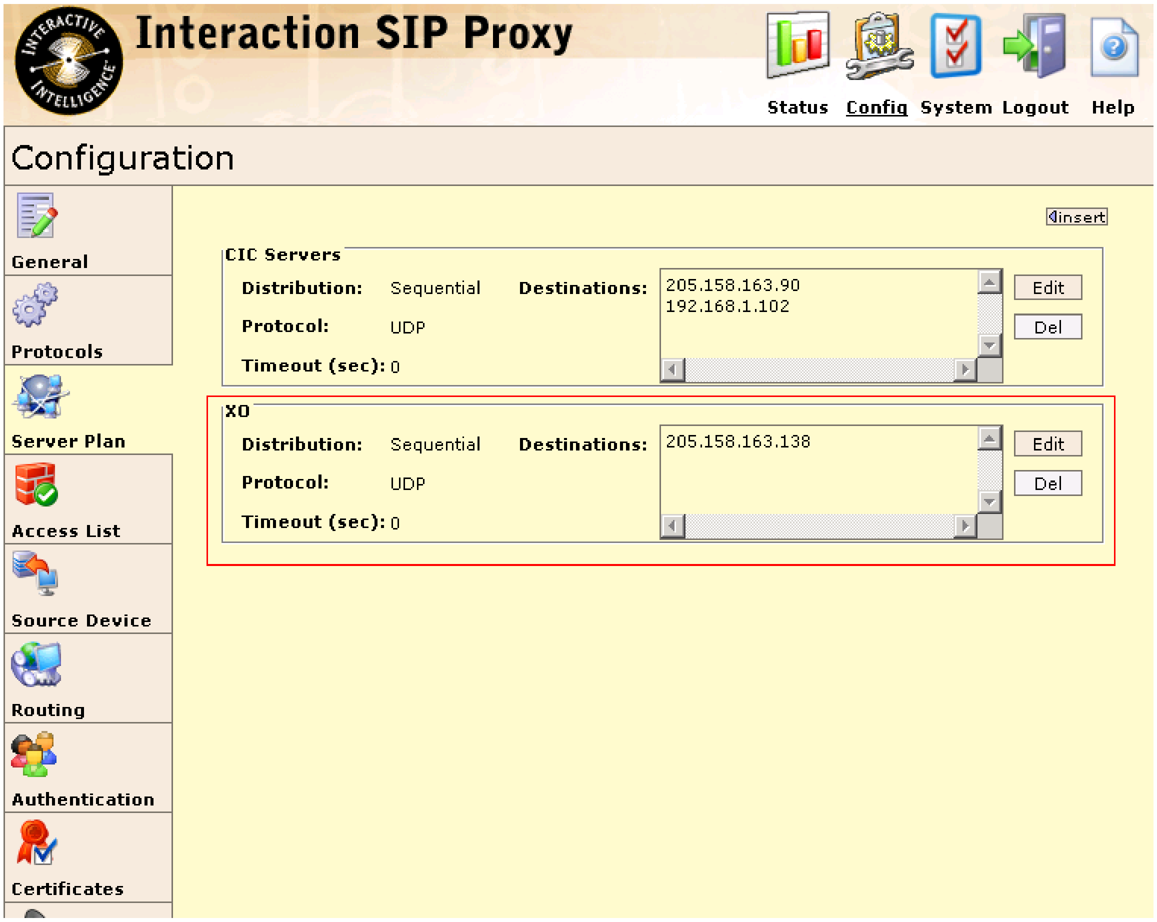
Configuration:



Protocols:

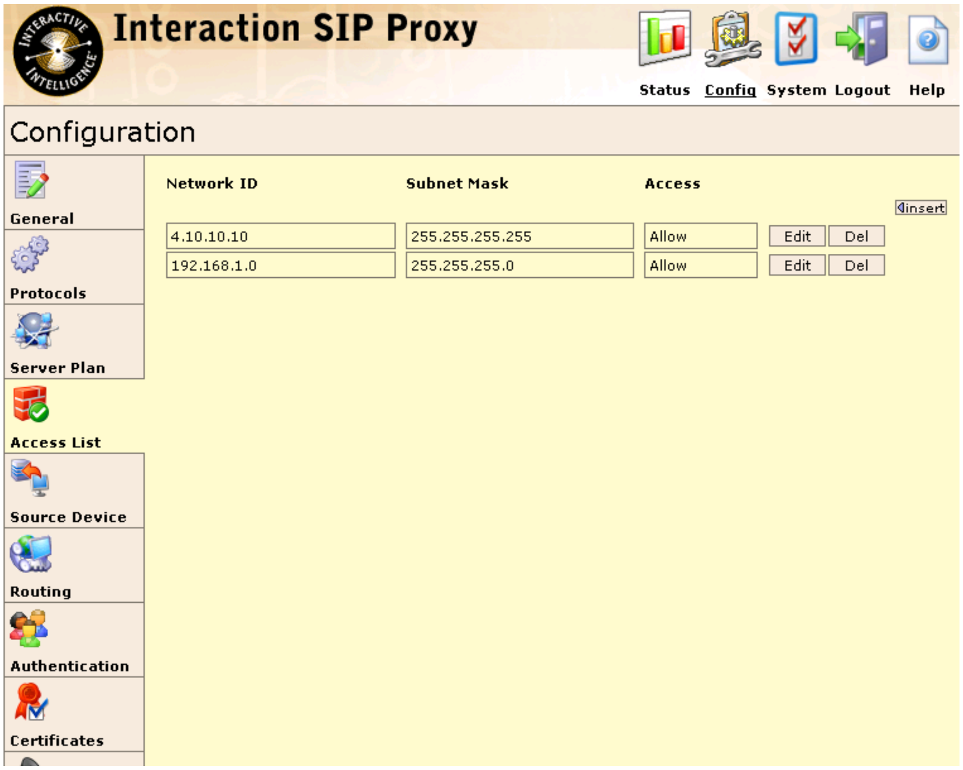


Server Plan:

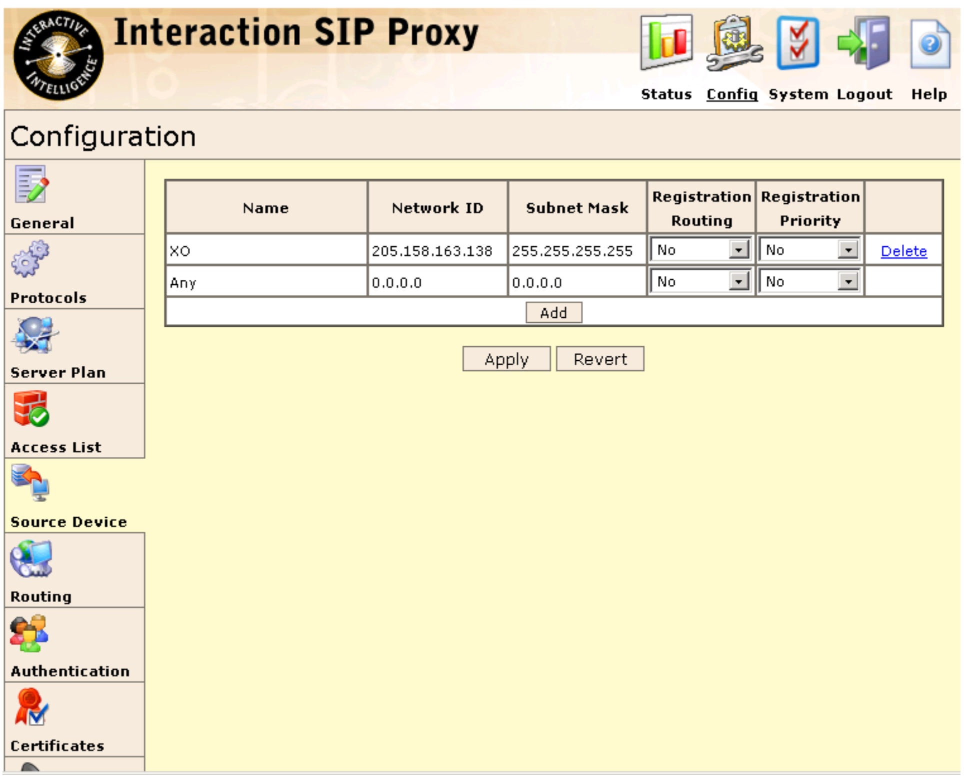


XO NBS IP needs to be added as shown in red box.

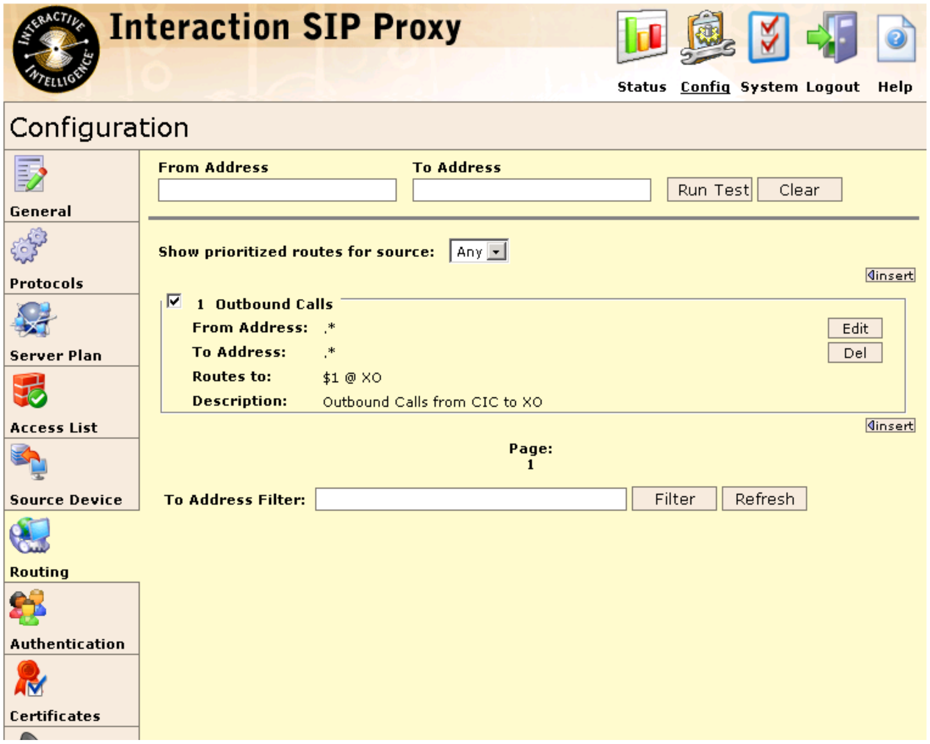
Access List:



Source device:



Routing:


Interaction SIP Proxy
Status
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Logout
Help
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Access List
Source Device
Routing
Authentication
Certificates
From Address
To Address
Run Test
Clear
Show prioritized routes for source:
Any
insert
1 Outbound Calls
From Address: .*
To Address: .*
Routes to: $1 @ XO
Description: Outbound Calls from CIC to XO
Edit
Del
insert
Page:
1
To Address Filter:
Filter
Refresh