

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.OSU12. This SCR will also be included in the next Service Update which is 3.OSU13 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

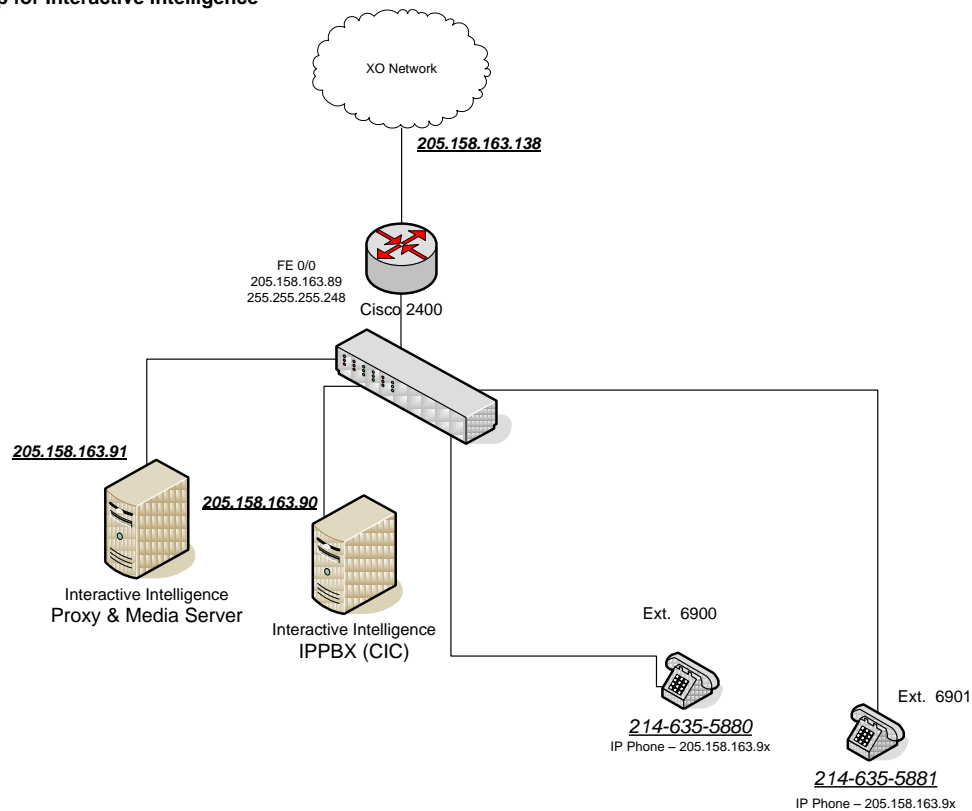
2. SIP Phones

Polycom Soundpoint IP 331 SIP

4 Test Configurations

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

XO Lab setup for Interactive Intelligence



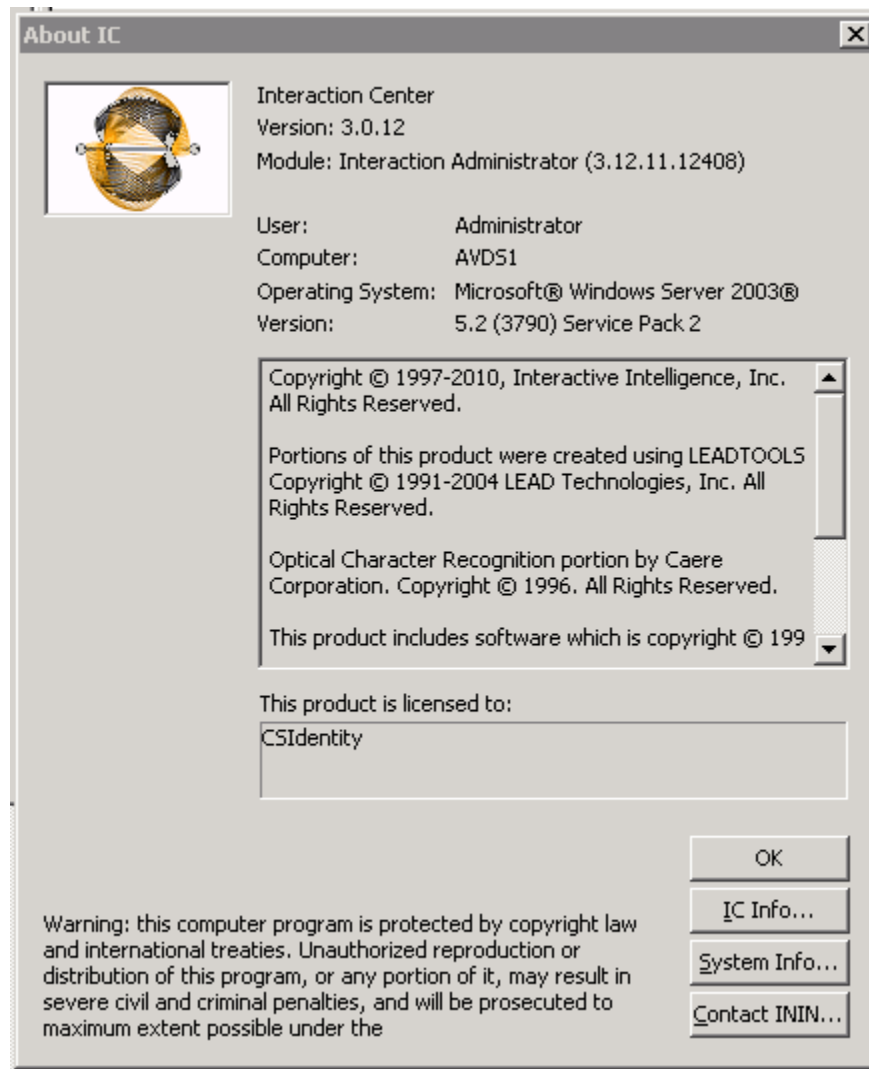
Notes:

- 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:

The screenshot displays the Interaction Administrator interface. On the left is a tree view of system components. The main area shows a table of SIP lines. A 'Line Configuration - SIP_XO' dialog box is open, showing configuration options for the selected line.

Line Name	Type	Outbound Address	Direction	Boar...	Chan...	Active
<Stations-TCP>	SIP	8008057004	Both			Yes
<Stations-TLS>	SIP	8008057004	Both			No
<Stations-UDP>	SIP	8008057004	Both			Yes
SIP_XO	SIP	2146355880	Both			Yes

Line Configuration - SIP_XO

- Auto Disconnect when Silence Detected in Voice Mail
- Silence Time (ms): 10000
- Call Analysis Type: Internal
- Use Numeric User Portion for Telephone Number
- Allow Deferred Answer
- Disable Fax Detection
- Enable SIP PreCall/Update for EarlyMedia Support
- Max Probation Time (s): 600

Buttons: Confirm auto-save, OK, Cancel, Apply

SIP Line configuration: Line

Line Configuration - SIP_XO

SIP Line Configuration | Call Putback | Custom Attributes | History

Line

Active

Domain Name: 205.158.163.91

Outbound Identity

Use Anonymous

Address: 2146355880

Name:

< sip:2146355880@205.158.163.91 >

Allow Name and Address to be overwritten with passed in values

On redirected calls, move outbound identity to redirection header

Redirection method: Diversion Header

Confirm auto-save

OK Cancel Apply

SIP Line configuration: Line (cont.)

Line Configuration - SIP_XO

SIP Line Configuration | Call Putback | Custom Attributes | History

Line

Redirection method: Diversion Header

Maximum Number of Calls

Combined Inbound/Outbound

Inbound: No Limit

Outbound: No Limit

Disable T.38 Faxing

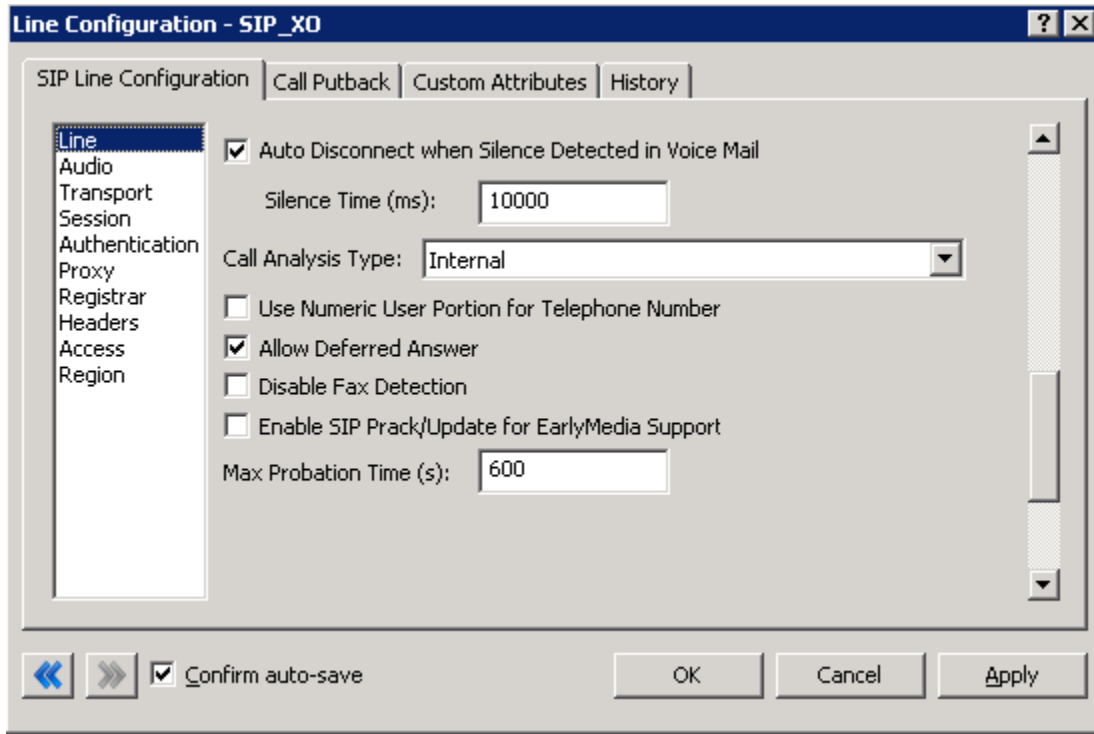
Auto Disconnect when Silence Detected in Voice Mail

Silence Time (ms): 10000

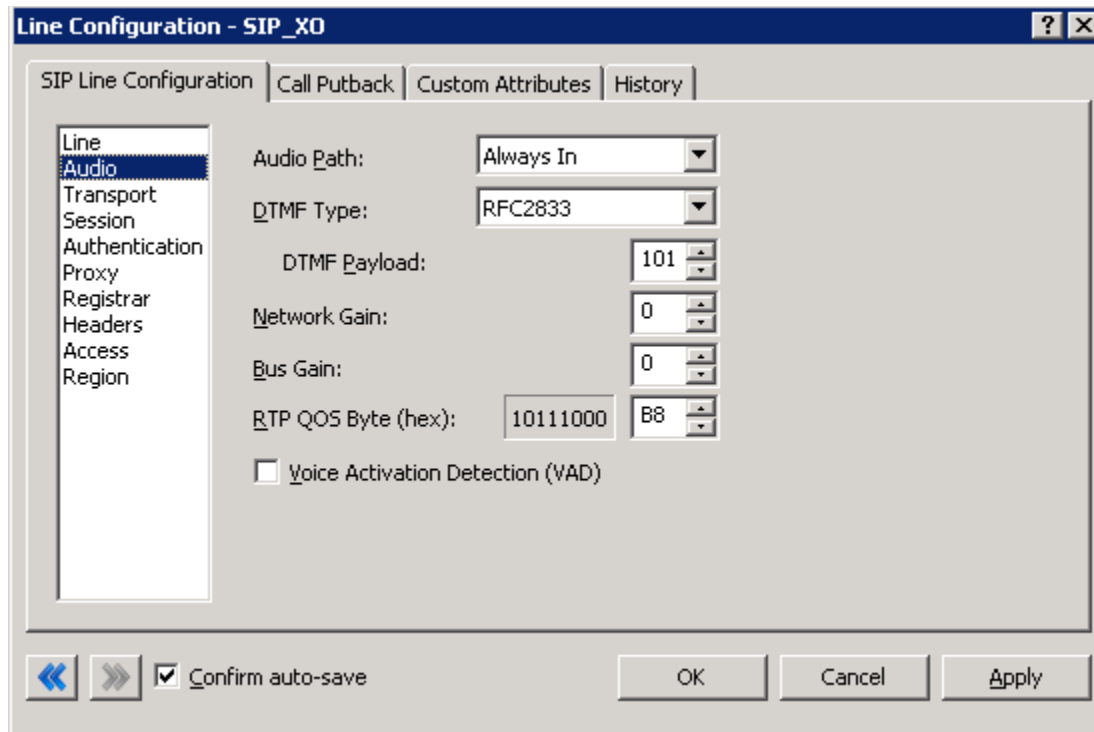
Confirm auto-save

OK Cancel Apply

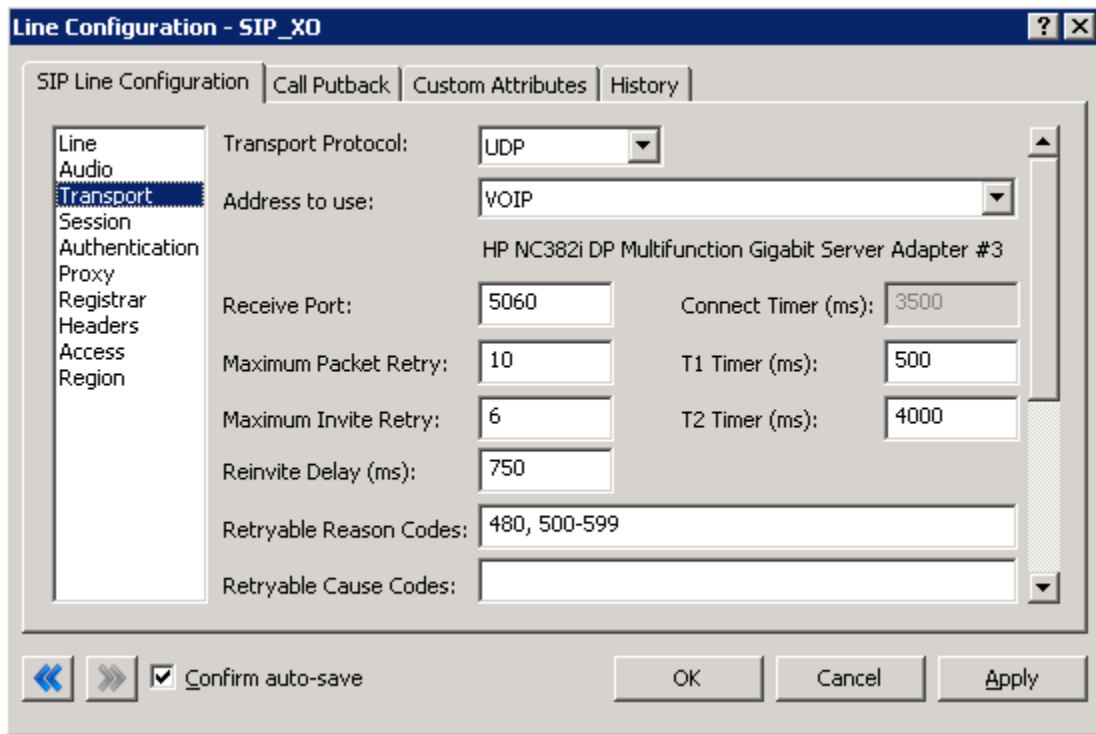
SIP Line configuration: Line (cont.)



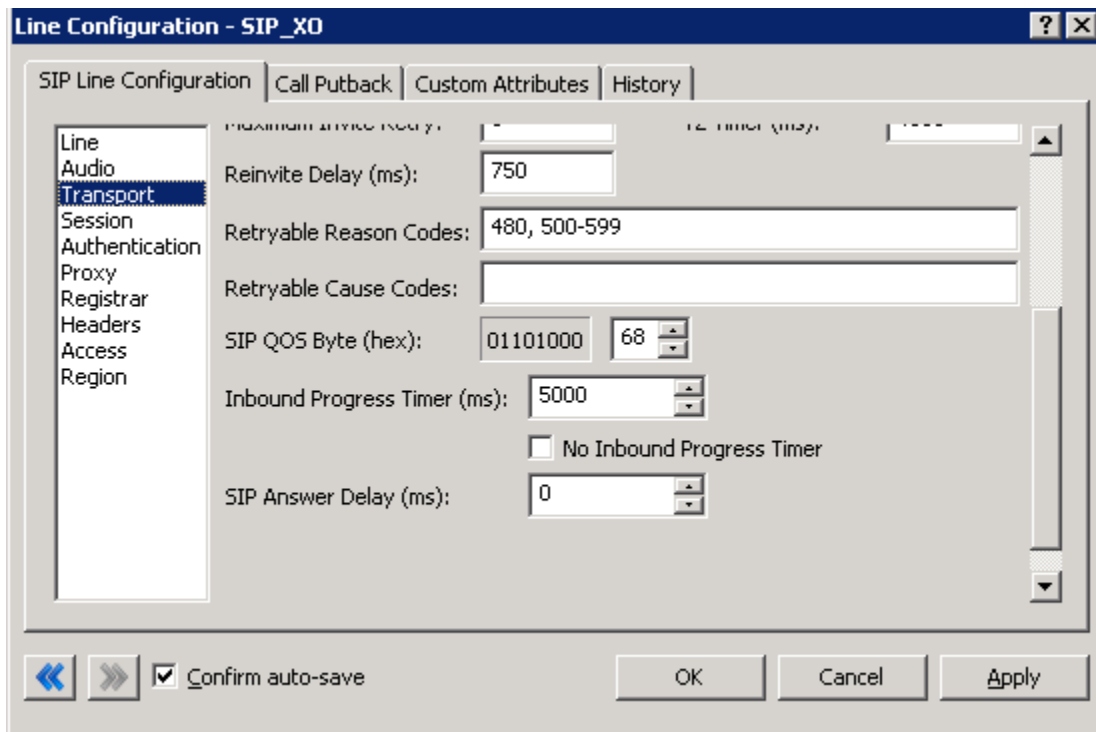
SIP Line configuration: Audio



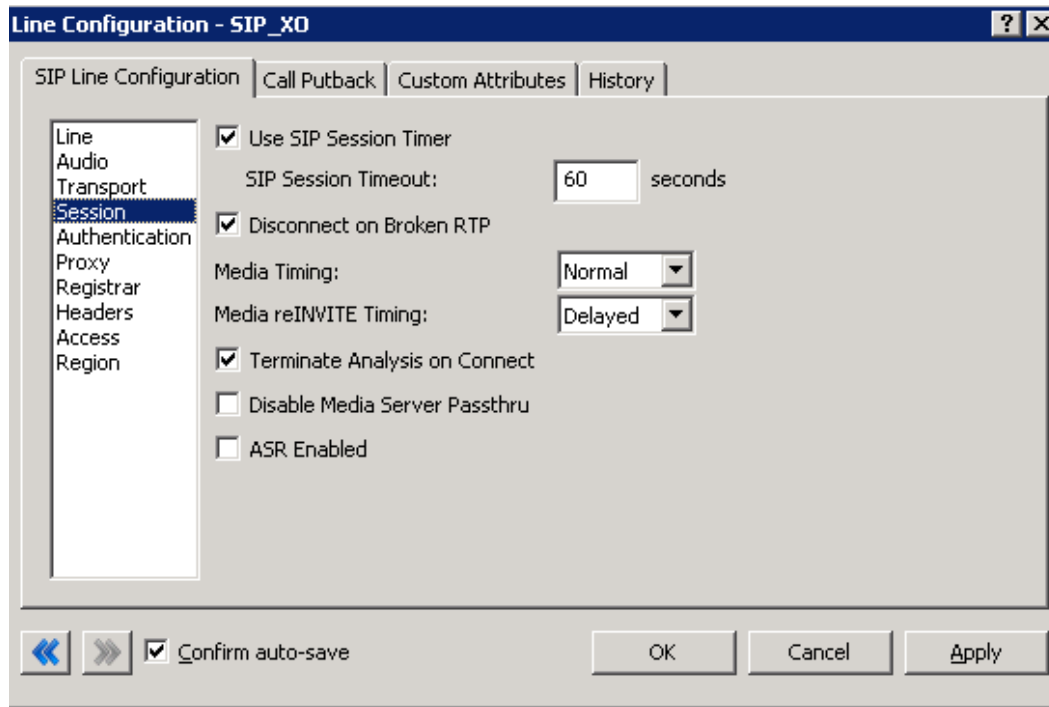
SIP Line configuration: Transport



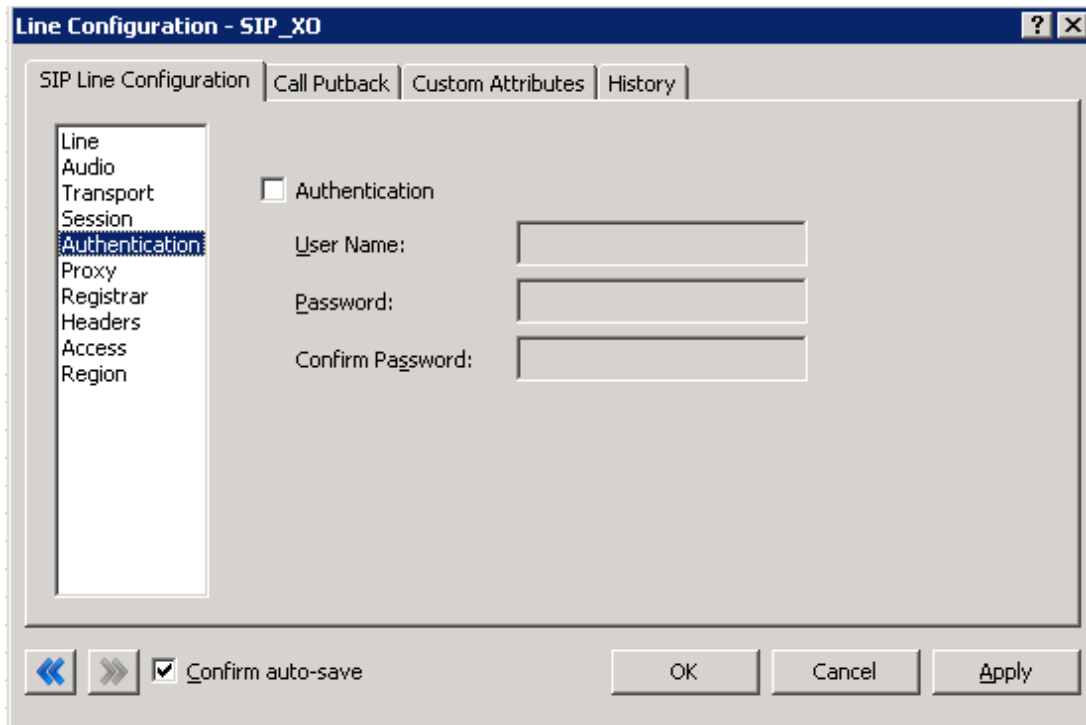
SIP Line configuration: Transport (cont.)



SIP Line configuration: Session

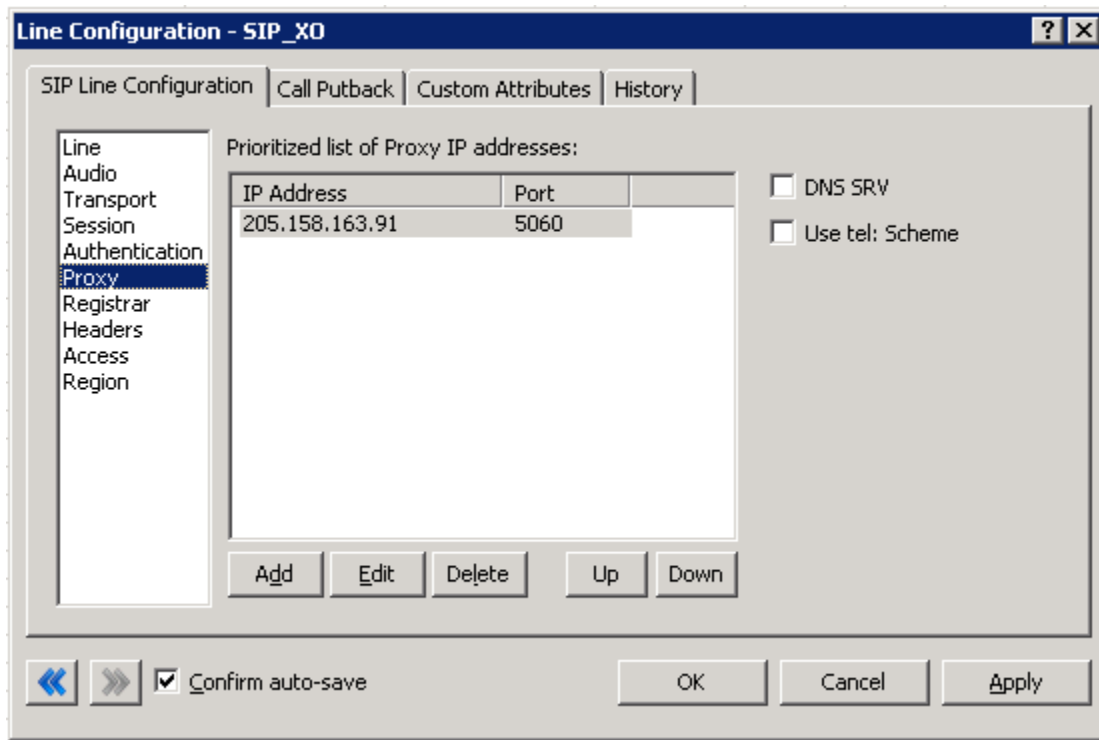


SIP Line configuration: Authentication

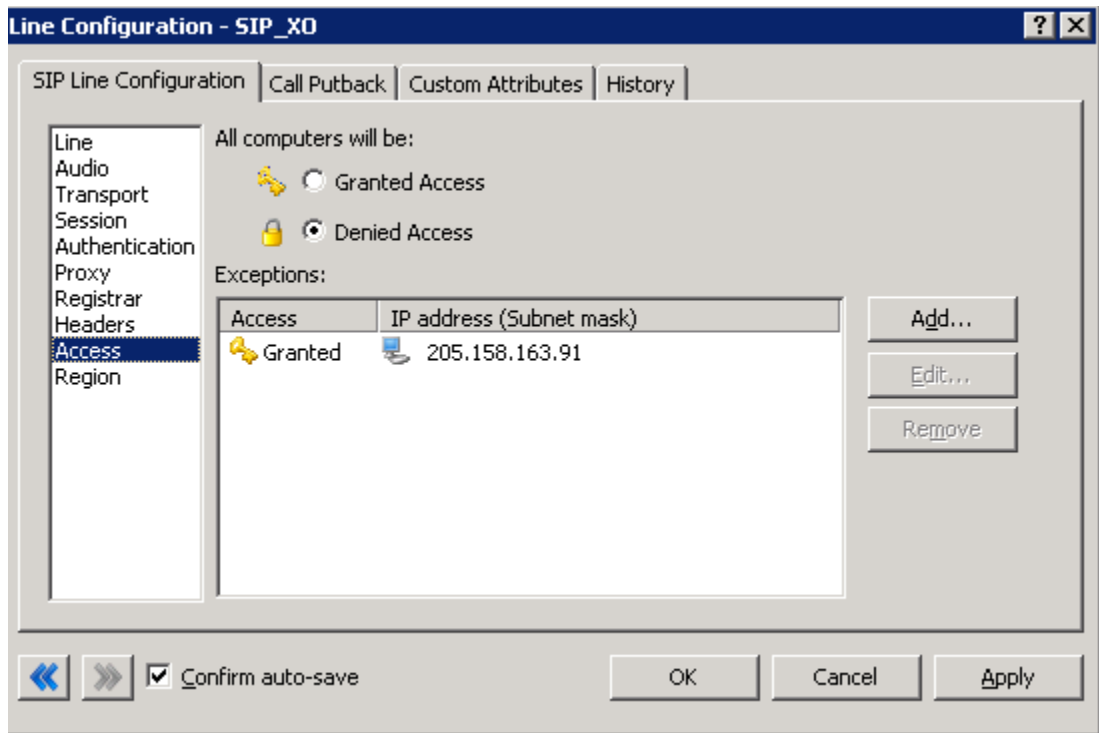


No authentication (username and password) requires with XO.

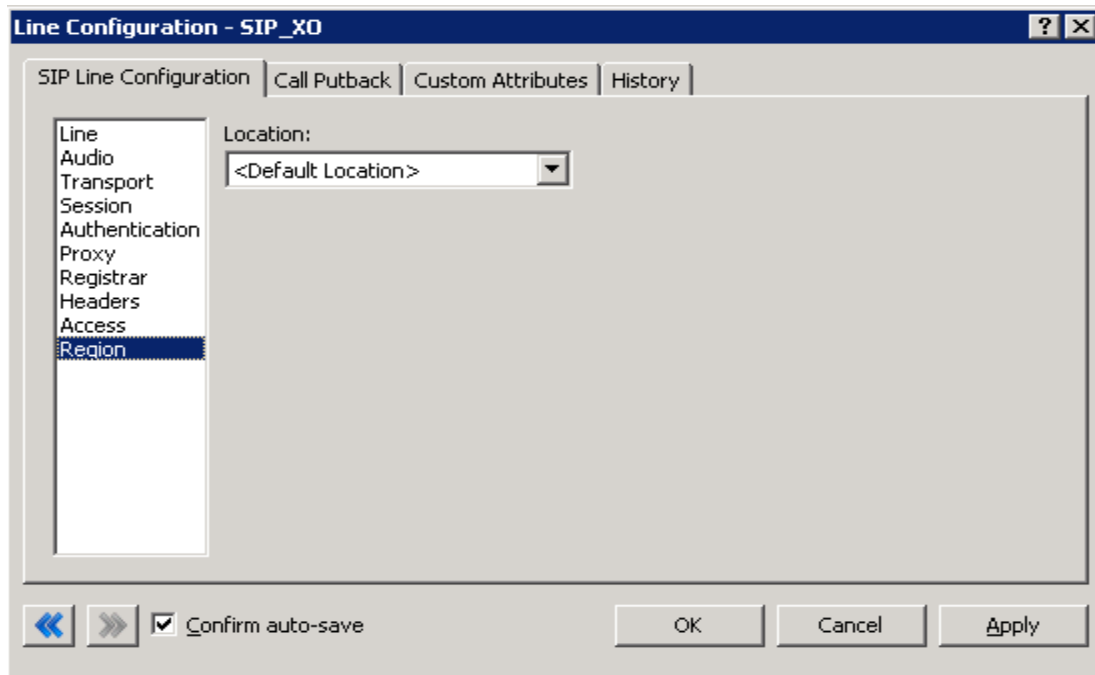
SIP Line configuration: Proxy



SIP Line configuration: Access

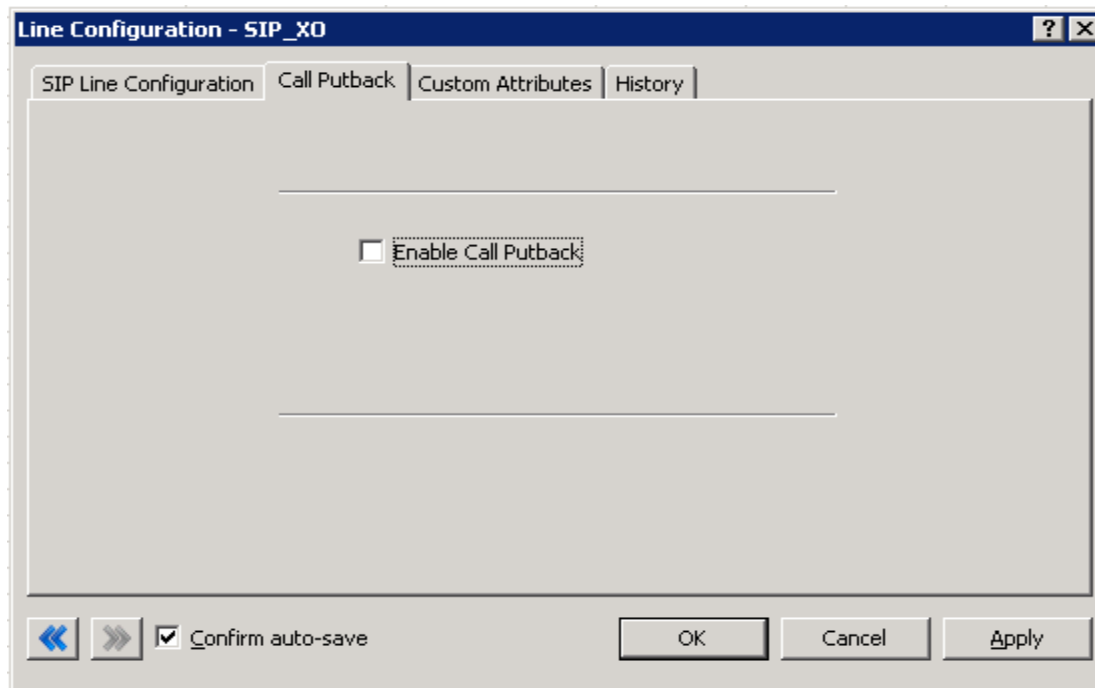


SIP Line configuration: Region

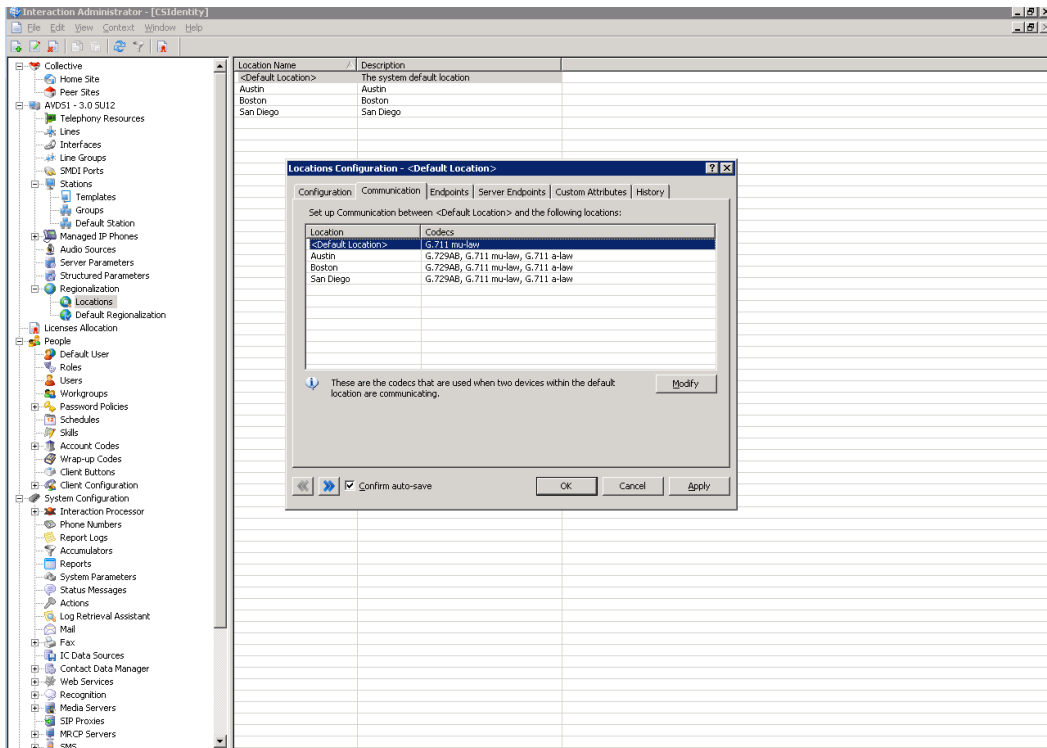


Line configuration – Call Putback:

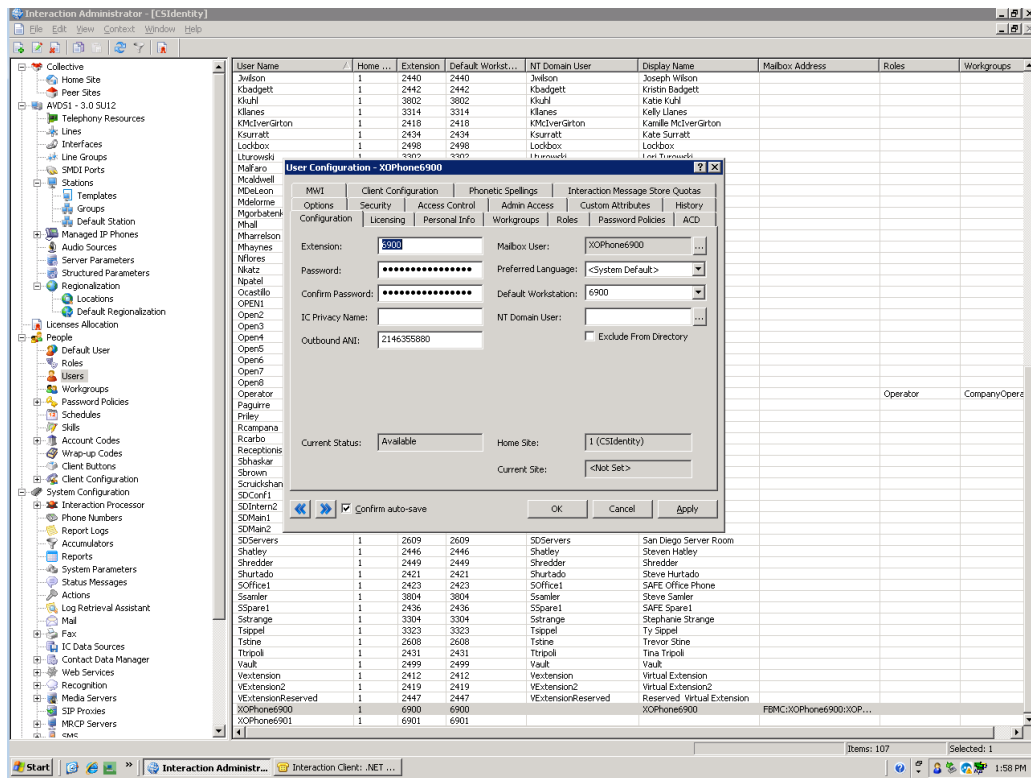
Check “Enable Call Putback” flag if transfer (blind/consult) with SIP REFER requires.



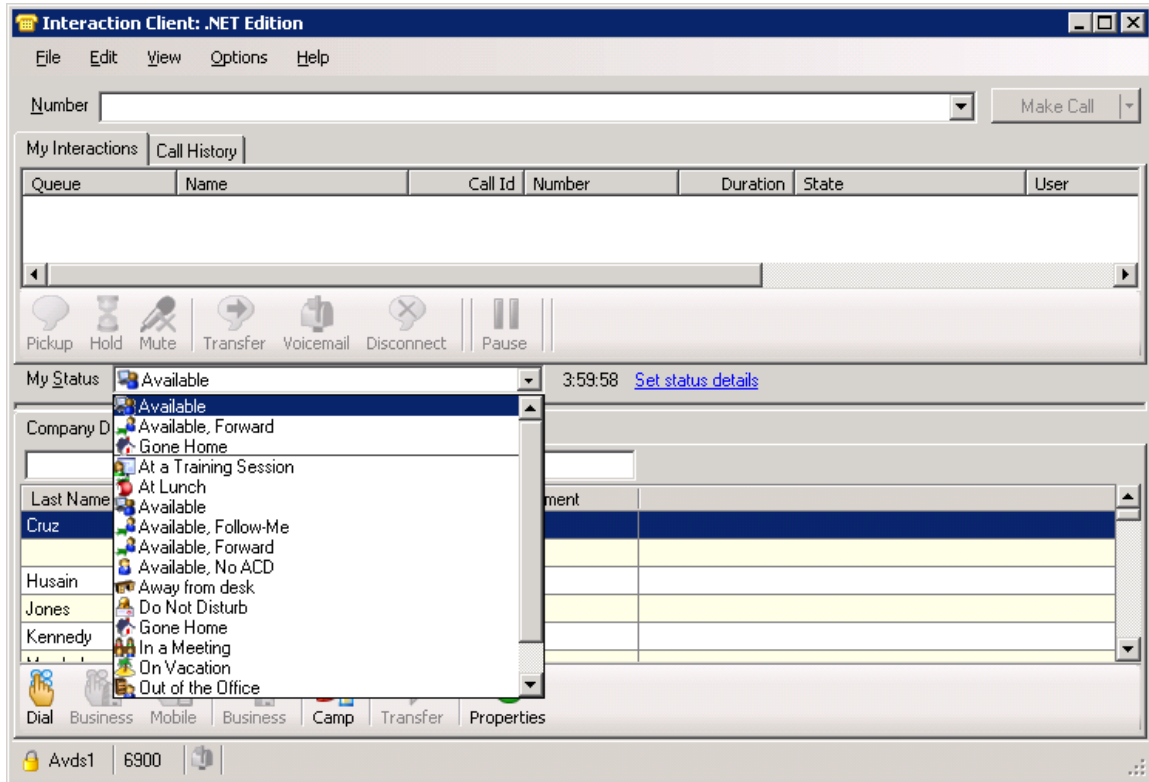
Regionalization >> Location >> <default location>



User >> XOPhone6900



Interaction Client:



Interactive Intelligence SIP Proxy configuration

Version:

Interaction SIP Proxy

[Status](#) [Config](#) [System](#) [Logout](#) [Help](#)

Status

Product Version 4.6.1302.11055 built on Jun 27 2011

License Production - permanent

Running Since 2011-07-21 15:56:57 (11d 22h 32m 52s)

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[Interactive Intelligence, Inc](#)

- About
- Sessions
- Registrations
- Server Status
- Statistics

Configuration:

Configuration	
	General
	Protocols
	Server Plan
	Access List
	Source Device
	Routing
	Authentication
	Certificates
	Media
	<p>Domain Name <input type="text" value="domain.com"/></p> <p>Alternate Domains <input type="text"/></p> <p><input type="checkbox"/> Use regular expressions</p> <p>SIP Network Adapter <input type="text" value="VOIP - 205.158.163.91"/></p> <p>Maximum registration interval <input type="text" value="3600"/></p> <p>Monitor servers / interval <input type="text" value="No"/> <input type="text" value="120"/></p> <p>SIP DSCP Value <input type="text" value="18 (24, 011000) CS3"/></p> <p>Allow remote configuration <input type="text" value="No"/> <input type="text"/></p> <p>Respond to OPTIONS <input type="text" value="No"/></p> <p>Digest authentication <input type="text" value="No"/></p> <p>Registration must match user <input type="text" value="No"/> (requires digest authentication)</p> <p>Record-Route <input type="text" value="Yes"/></p> <p>Generate call detail records <input type="text" value="Yes"/> (requires record-route)</p> <p>Rewrite Request-URI <input type="text" value="Yes"/></p> <p>Route unknown URI <input type="text" value="Yes"/></p> <p>Match pattern against all routes <input type="text" value="No"/></p> <p>Redirect server <input type="text" value="No"/></p> <p>Recurse on 3xx <input type="text" value="No"/></p> <p>Next route on response code <input type="text" value="No"/> <input type="text" value="480 5.. 6.."/></p> <p>Registration routing <input type="text" value="Yes"/></p> <p>Registration has priority <input type="text" value="No"/></p> <p>Forward REGISTERS <input type="text"/></p> <p><input type="checkbox"/> Forward to all servers</p>

Protocols:

Configuration	
	General
	Protocols
	Server Plan
	Access List
	Source Device
	Routing
	Authentication
	Certificates
	<p><input checked="" type="checkbox"/> UDP</p> <p>Port Number <input type="text" value="5060"/></p> <p>Timer T1 (ms) <input type="text" value="500"/></p> <p>Timer T2 (ms) <input type="text" value="4000"/></p> <p>Max SIP INVITE Retries <input type="text" value="6"/></p> <p>Max Packet Retries <input type="text" value="10"/></p> <p><input checked="" type="checkbox"/> TCP</p> <p>Port Number <input type="text" value="5060"/></p> <p>Connect Timeout (sec) <input type="text" value="5"/></p> <p><input type="checkbox"/> TLS</p> <p>Port Number <input type="text" value="5061"/></p> <p>Mutual Authentication <input type="text" value="No"/></p> <p>Allowed TLS Cipher suites</p> <ul style="list-style-type: none"> <input checked="" type="checkbox"/> TLS_RSA_WITH_AES_256_CBC_SHA <input checked="" type="checkbox"/> TLS_RSA_WITH_AES_128_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_3DES_EDE_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_DES_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_RC4_128_SHA <input type="checkbox"/> TLS_RSA_EXPORT_WITH_DES40_CBC_SHA <p>Certificate Installed <input type="text" value="No"/></p>

Server Plan:

The screenshot shows the 'Interaction SIP Proxy' configuration interface. On the left is a navigation menu with options: General, Protocols, Server Plan, Access List, Source Device, Routing, Authentication, and Certificates. The 'Server Plan' option is selected. The main area displays two server configurations. The top one is for 'CIC Servers' with destinations 205.158.163.90 and 192.168.1.102. The bottom one is for 'XO' with destination 205.158.163.138. This 'XO' configuration is enclosed in a red rectangular box. Each configuration includes fields for Distribution (Sequential), Protocol (UDP), and Timeout (sec) (0). Buttons for 'Edit' and 'Del' are present for each entry. An 'Insert' button is located at the top right of the configuration area.

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

Access List:

The screenshot shows the 'Interaction SIP Proxy' configuration interface. The 'Access List' tab is selected in the left sidebar. The main area displays a table with columns for 'Network ID', 'Subnet Mask', and 'Access'. There are two rows of data, each with 'Edit' and 'Del' buttons. An 'Insert' button is located at the top right of the table area.

Network ID	Subnet Mask	Access
4.10.10.10	255.255.255.255	Allow
192.168.1.0	255.255.255.0	Allow

Source device:

The screenshot shows the 'Interaction SIP Proxy' configuration interface. The 'Source Device' tab is selected in the left sidebar. The main area displays a table with columns for 'Name', 'Network ID', 'Subnet Mask', 'Registration Routing', and 'Registration Priority'. There are two rows of data, each with a 'Delete' link. An 'Add' button is located below the table, and 'Apply' and 'Revert' buttons are at the bottom.

Name	Network ID	Subnet Mask	Registration Routing	Registration Priority
XO	205.158.163.138	255.255.255.255	No	No
Any	0.0.0.0	0.0.0.0	No	No

Routing:

The screenshot displays the 'Interaction SIP Proxy' configuration interface. The main title is 'Interaction SIP Proxy' with a logo on the left and navigation links (Status, Config, System Logout, Help) on the right. The 'Configuration' section is active, showing a sidebar with menu items: General, Protocols, Server Plan, Access List, Source Device, Routing, Authentication, and Certificates. The main content area includes a 'From Address' and 'To Address' input field with 'Run Test' and 'Clear' buttons. Below this is a 'Show prioritized routes for source:' dropdown set to 'Any'. A table lists one route: '1 Outbound Calls' with 'From Address: .*', 'To Address: .*', 'Routes to: \$1 @ XO', and 'Description: Outbound Calls from CIC to XO'. The table has 'Edit' and 'Del' buttons. At the bottom, there is a 'Page: 1' indicator and a 'To Address Filter:' input field with 'Filter' and 'Refresh' buttons.