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SIP Trunking Configuration Guide for the Interactive Intelligence, Inc. Customer Interaction Center (CIC) PBX Version 3.0 SU13

Table of Contents

Table of Figures

Table of Tables

1 Audience

This document is intended for the SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities.

2 Introduction

This Configuration Guide describes configuration steps for Cox SIP trunking to an Interactive Intelligence (ININ) – Customer Interaction Center (CIC) IP PBX. Cox SIP trunking is a scalable and efficient IP trunking telecommunication solution for your business that provides all the traditional services such as Direct Inward Dialing, Hunting, Calling Line Name and Number, Local/Long Distance and Business Continuity options, including:

- Burstable Trunk Capacity – Dynamically increases call capacity during peak busy periods so your customers never receive a busy signal.
- Call Forward Always – On the trunk group pilot number for all calls in case of an outage (i.e., flood, fire, loss of power, etc.).
- Call Forward Not Reachable – On the trunk group pilot number that operates on a per-call contingency basis to forward the call to any PSTN number (i.e., call center or alternate office location) during temporary call completion impairments.
- Route Exhaustion – Automatic reroute of trunk group calls to any PSTN phone number (i.e., a call center) if calls can't be completed to the PBX.
- Support for geo-redundant PBX deployments and automatic reroute of SIP trunks to the backup customer data center.

All calls are routed over Cox's national fiber network with guaranteed Quality of Service (QoS); <u>calls never traverse</u> the Internet.

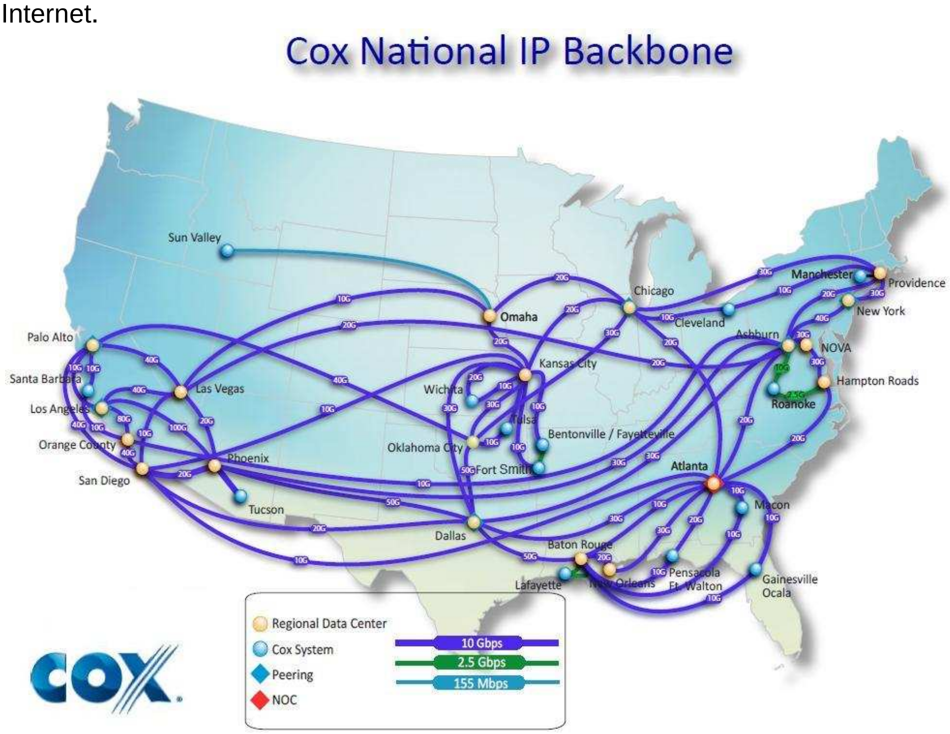

Figure 1 - Cox Fiber Network

2.1 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. ("tekVizion"). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- Verification Testing – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration ("solution Verification")
- Product Assessment – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our FastForward>> practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion's headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in the Telecom Corridor® in Richardson, Texas.
(*For more information on tekVizion and its practice areas, please visit tekVizion Labs's web site at www.tekVizionlabs.com.)*

3 SIP Trunking Network Components

The following SIP trunk reference network is representative of an ININ CIC PBX deployment:

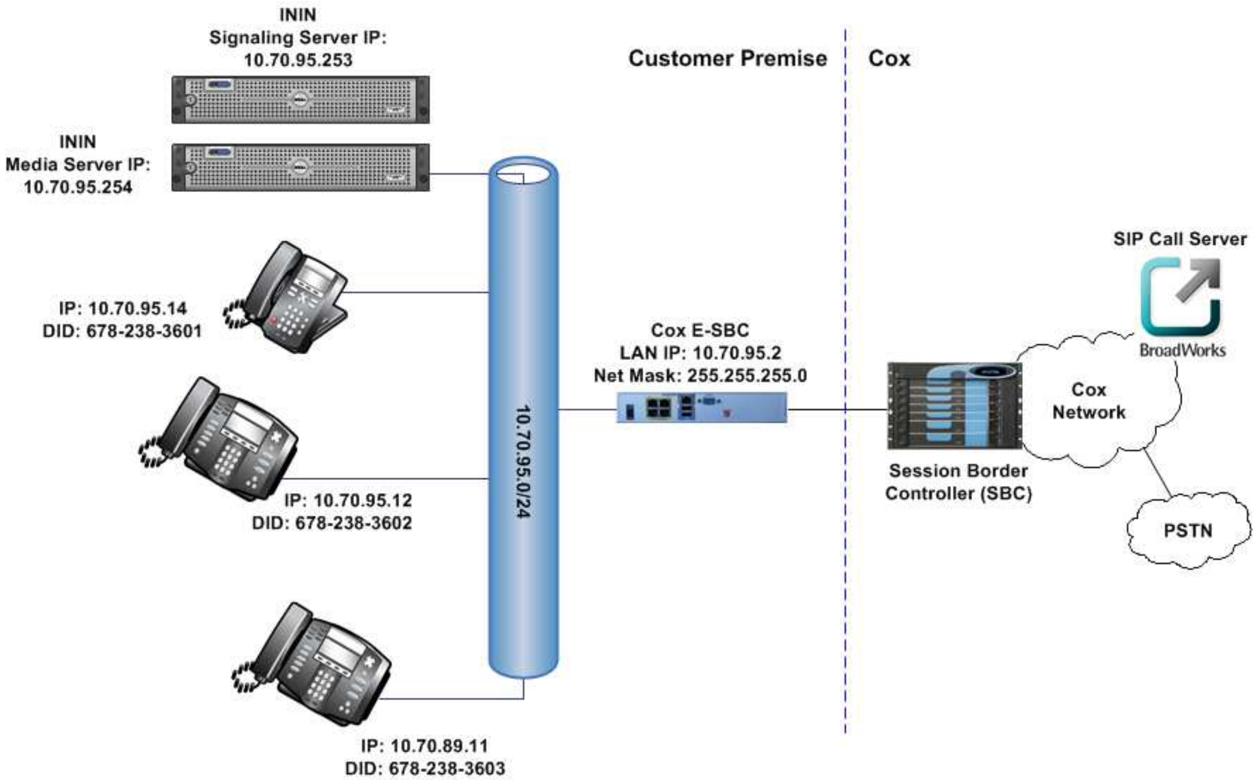

Figure 2 - SIP Trunk Lab Reference Network

Note: The ININ CIC does not offer DHCP server for dynamic IP address assignment for the SIP phones; however, the Cox Enterprise Session Border Controller (E-SBC) requires a static LAN IP address that must be manually assigned by the LAN network administrator. The DHCP server is provisioned on the Ethernet switch. The DHCP's IP address pool is constrained so that the E-SBC can be assigned an IP address outside of the pool.

The lab network consists of the following components:

- ININ PBX for voice features, SIP proxy and SIP trunk termination.
- Various SIP phones on the local LAN.
- The Cox E-SBC is the Edgewater Networks (www.edgewaternetworks.com) EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, B2BUA and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC.
- Acme Packet Net-Net 9200 Session Border Controllers (SBC).

3.1 Hardware Components

- SuperMicro Server with Intel Pentium Dual Core G6950
- EdgeMarc 4550 E-SBC
- PolyCom Phones (IP330, IP550, and IP650)

3.2 Software Requirements

- ININ Release 3.0 SU13
- EdgeMarc 4550 9.12.5 ReleaseFeatures

3.3 SIP Registration Method

Cox Network requires SIP REGISTER support to allow the IP-PBX to originate calls from the IP-PBX and to send calls to the IP PBX from the PSTN. ININ supports SIP Register with authentication. The Cox sales support team will specify the SIP Trunk Authorization Credentials.

3.4 Features Supported

- Basic calls using G.711ulaw
- Calling Party Number Presentation
- Anonymous call
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Pickup
- Call Waiting
- DND
- Call Park
- Hunt groups (Simultaneous and Sequential Ring)
- Three-Way Calling
- PBX Auto Attendant to Off-net Numbers
- PBX Account Codes
- PBX Authorization Codes
- Dial-Up Modem
- E911 Call
- RFC2833 transcoding
- PBX-Defined Caller ID (spoofing)

3.5 Features Not Supported

- Call Forward Busy
- Call Park
- G711 Pass Thru

4 Caveats and Limitations

- For any external transfer or call forward scenario without call putback enabled, when the first party hangs up the phone, the Customer Interaction Center (CIC) (“ININ PBX”) sends an extra INVITE after the parties have been disconnected. This does not cause any problems that a user would notice. There are no configuration changes that would prevent this unnecessary INVITE when call putback is disabled. This issue is being tracked with by SCR IC-91244 and slated to be resolved in CIC 4.0.

- One way audio occurs when the following conditions are true:
    - “SIP Prack/Update for EarlyMedia Support” is enabled on CIC.
    - The DTMF payload defined in CIC does not match the DTMF payload offered by the carrier.
    - Call putback is disabled.
    - Digest authentication is in use.
    - When CIC sends an UPDATE message to the carrier, using the same digest authentication credentials that succeeded in the INVITE message, the carrier responds with 401 UNAUTHORIZED to challenge the credentials. On external-external transfers, there is no-way audio.

  This issue is being tracked with SCR IC-91299. This patch is currently scheduled for R3 SU15. This issue can however be resolved by configuring the DTMF payload offered by the carrier to the same in CIC as shown in **Figure 6.** DTMF Payload 96 is delivered from the EdgeMarc SBC in the lab tests.

5 PBX Configuration Steps

5.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 1 – IP Addresses

Component	Cox Lab Value	Customer Value
EdgeMarc E-SBC		
• LAN IP Address	10.70.95.2	
• LAN Subnet Mask	255.255.255.0	
ININ IP PBX		
• System IP Address The Internet Connection will typically be on the same subnet as the LAN IP Address of the E-SBC. If this is not the case, then Layer 3 routing must be in place.	10.70.95.253	
• Media Server IP Address	10.70.95.254	
• Default Gateway The Default Gateway must be the LAN Network default Gateway. This will allow the administrator to log in via his\her workstation if the workstation is on a different network	10.70.95.1	
• DNS This is the DNS server for the Enterprise network. Cox Communications does not supply DNS services.	10.64.1.3	

5.2 CIC Detailed Configuration Steps

Equipment used for configuration setup:

- SuperMicro Server with Intel Pentium Dual Core G6950.
- ININ software version release V3.0 SU13.

5.2.1 Sip Carrier Options

Remember that the E-SBC LAN IP address may/will be different from this example. Please see **Figure 2** and **Table 1** for the IP address scheme.

1. Navigate to **ININSERVER1 > Lines**
2. Select **File > New**
3. In the **Add New Lines** dialog box**,** select **SIP**
4. Set **SIP Carrier Name**: SIPLine.
   This value can be any name that the Enterprise customer wants to use. For this example SIPLine was used.
5. Select **OK**
6. Under the **SIP Line Configuration** tab, select **Line**
7. Set **Domain Name:** MyDomain is used as a value to populate this field, as it may not be left blank. Cox SIP Trunking does not use domain names and any value may be input.
8. **Set Address:** 6782383600
9. **Set Name:** CoxSip

   NOTE: Many of the settings are default values. These values must be verified, as many of them are critical.

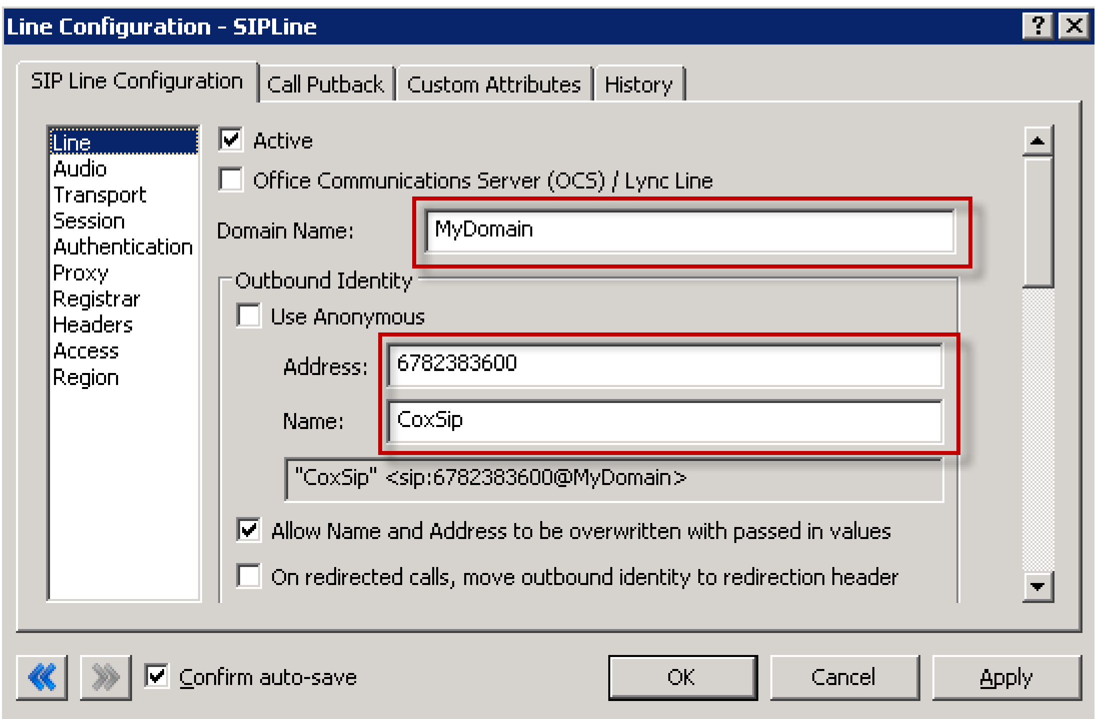

Figure 3 - SIP Carrier Options: Line

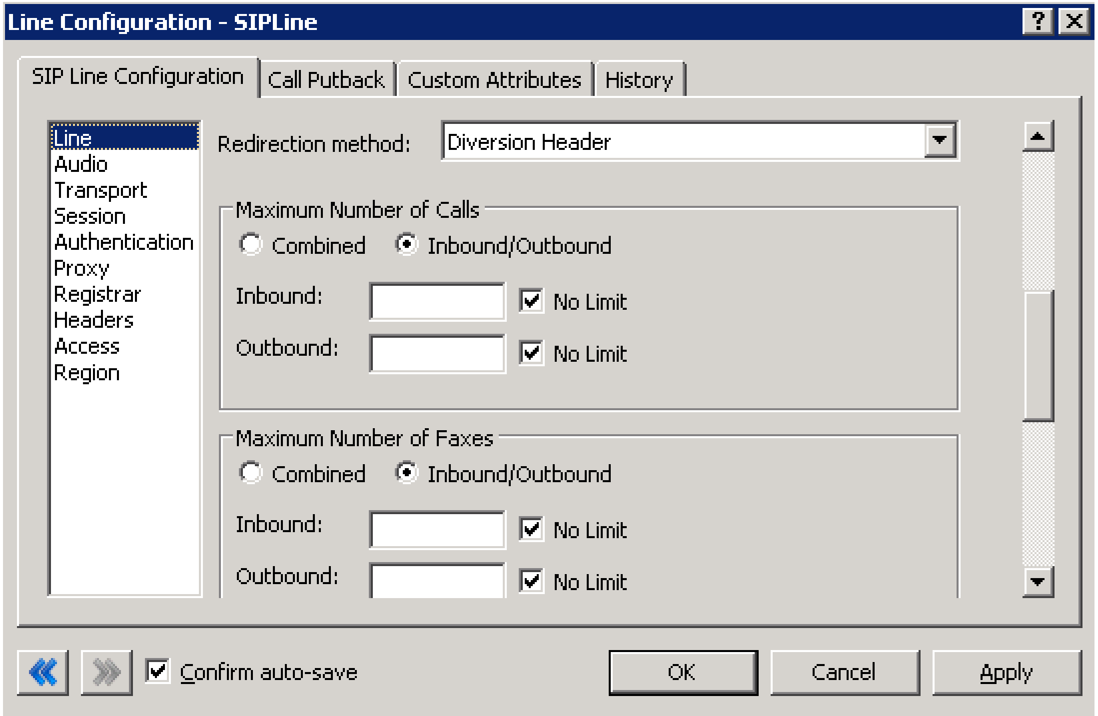

Figure 4 - SIP Carrier Options: Line

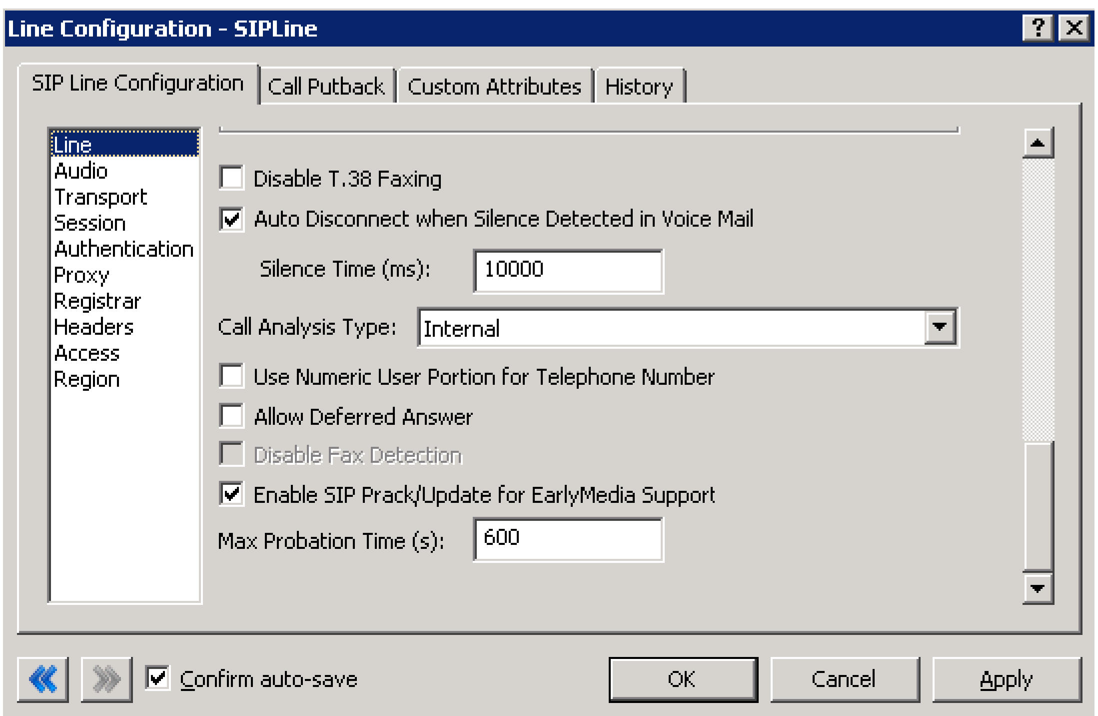

Figure 5 - SIP Carrier Options: Line

10. Select **Audio** and verify the settings.

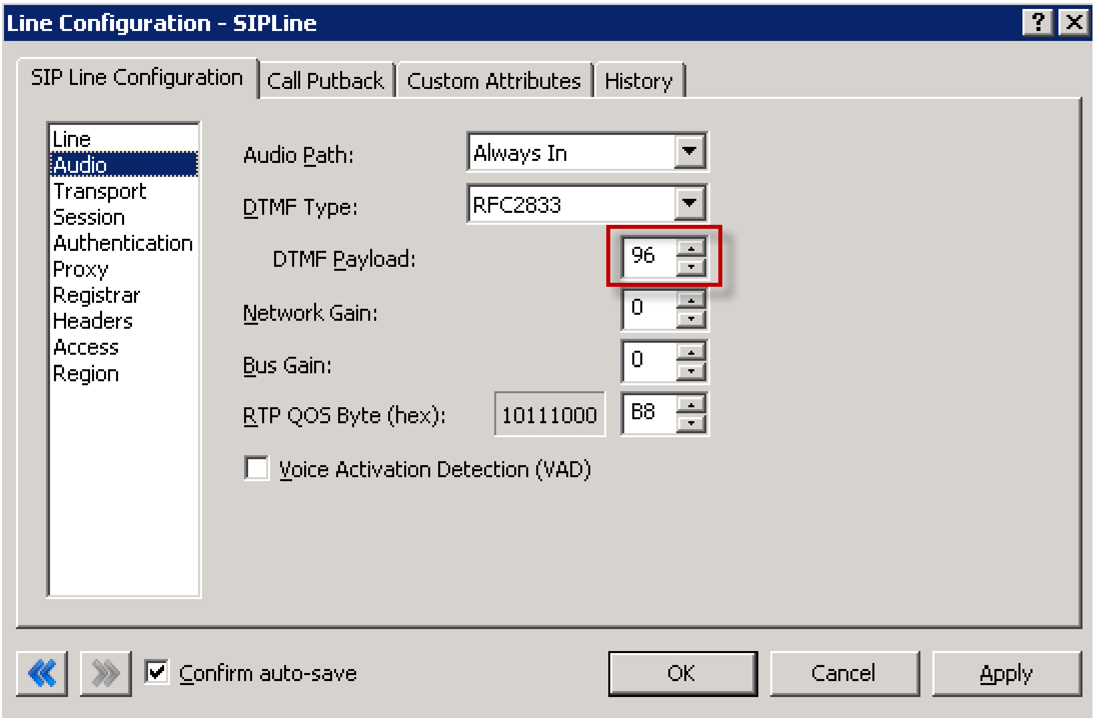

Figure 6 - SIP Carrier Options: Audio

11. Select **Transport** and verify the settings

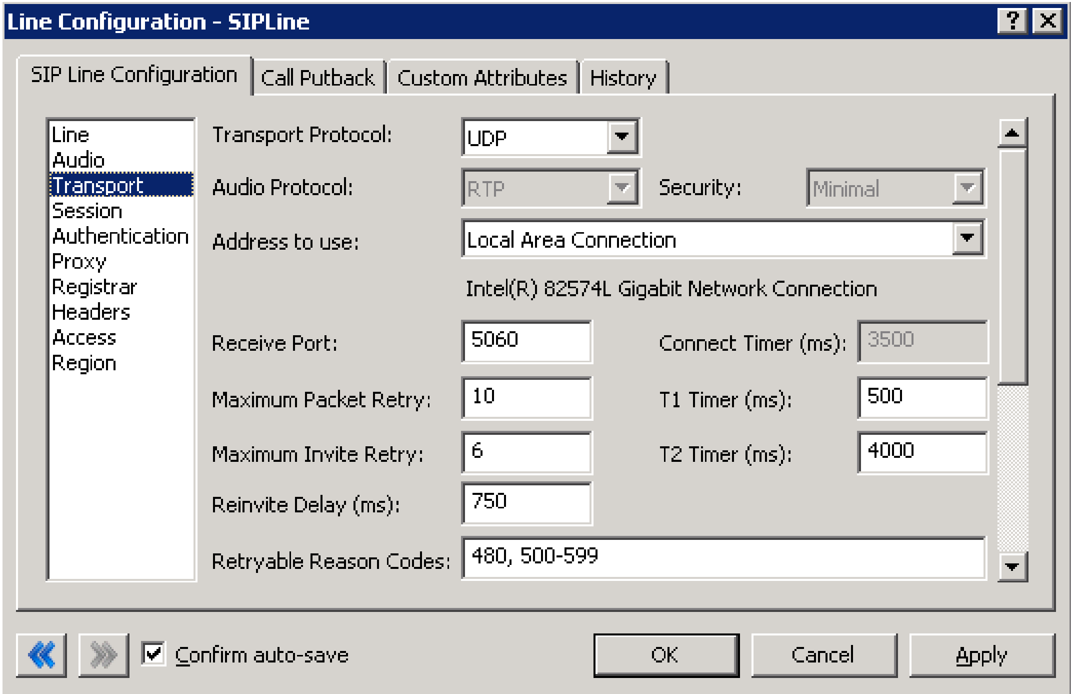

Figure 7 - SIP Carrier Options: Transport

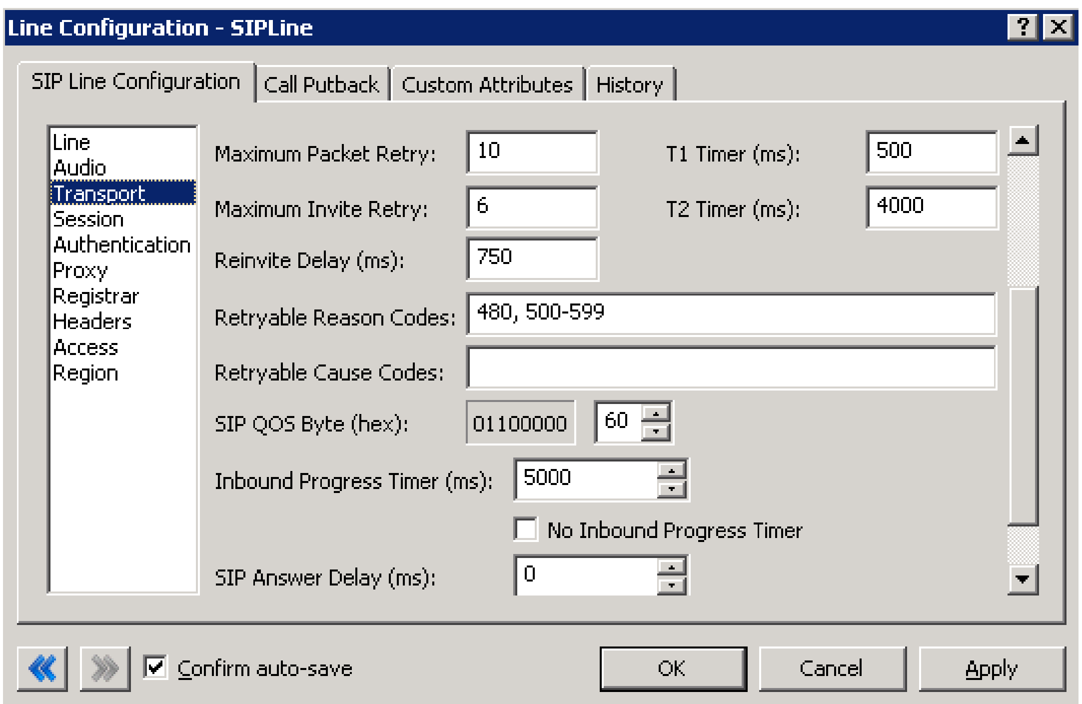

Figure 8 - SIP Carrier Options: Transport

12. Select **Session** and verify the settings

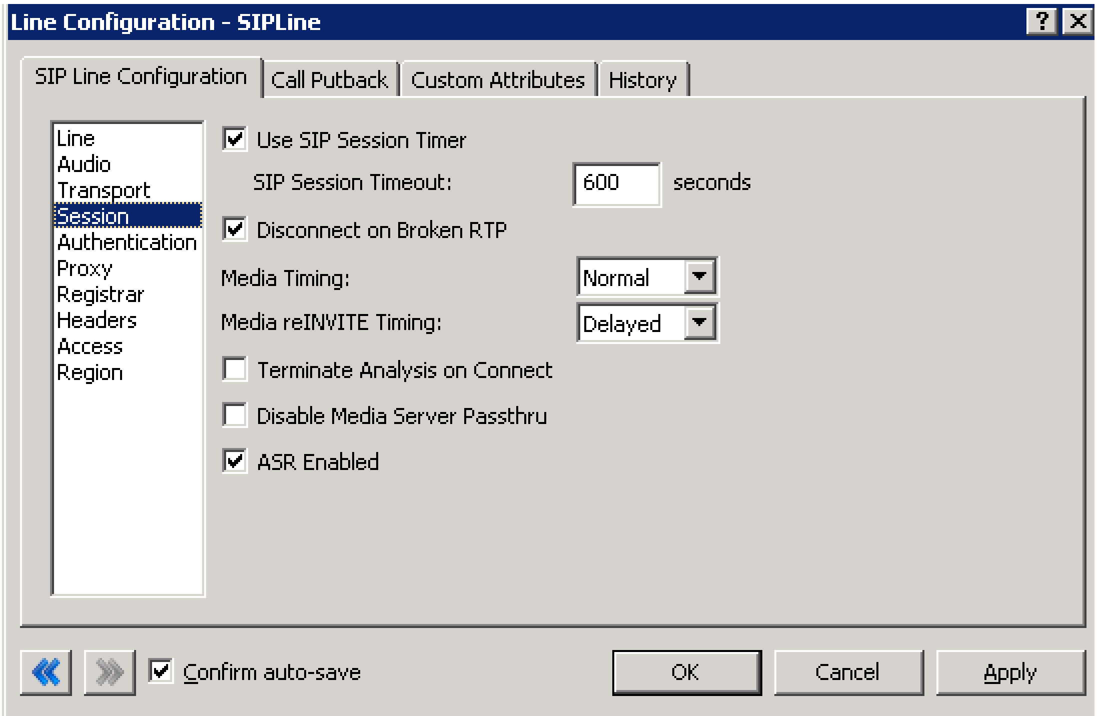

Figure 9 - SIP Carrier Options: Session

13. Select **Authentication**
14. Set **User Name:** 6782383600
15. Set **Password:** <provided by Cox>

The actual SIP Registration Username and Password will be provided by your Cox Account Representative and must be kept confidential. The Trunk Group Pilot Number (username) is used here for illustration purposes only!

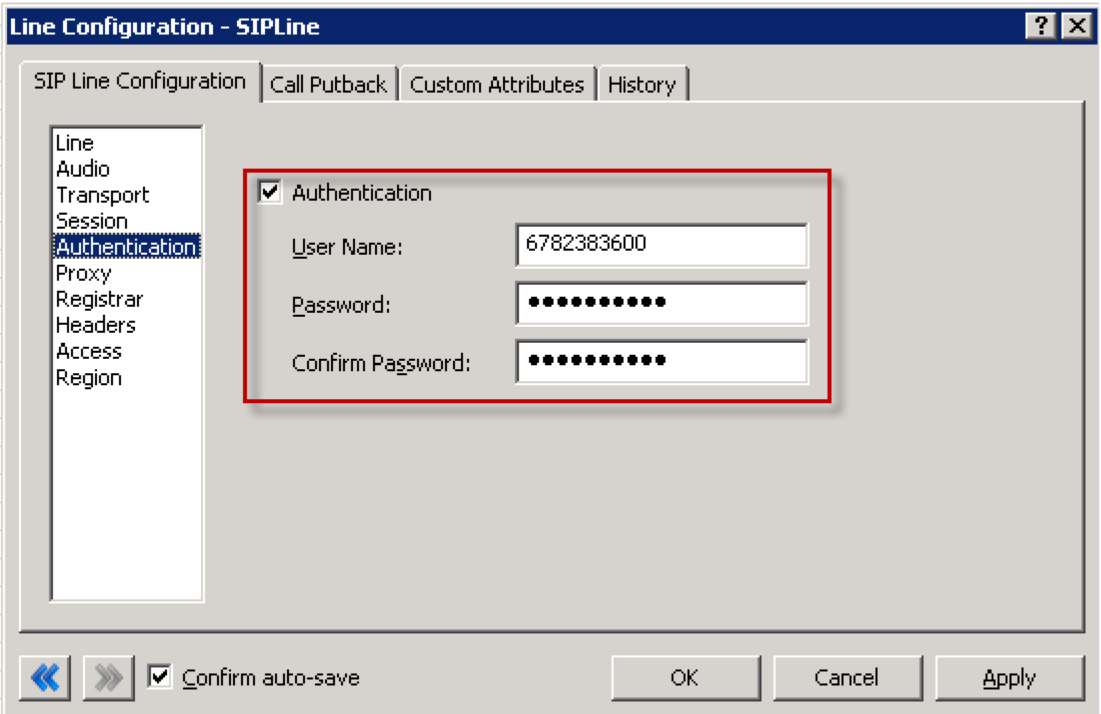

Figure 10 - SIP Carrier Options: Authentication

16. Select **Proxy**
17. Select **Add**
18. Set **IP Address:** 10.70.95.2

    This is the **static LAN IP address of the Cox E-SBC**. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.95.2
19. Select **OK**

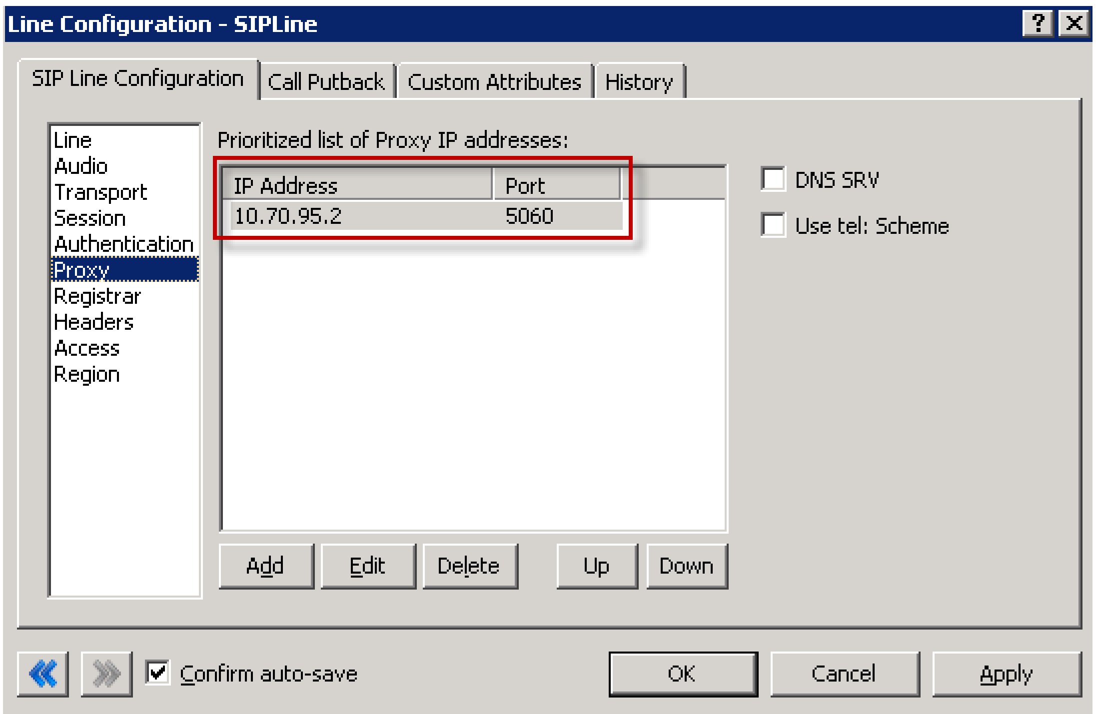

Figure 11 - SIP Carrier Options: Proxy

20. Select **Registrar**
21. Under **External Phone Numbers,** select **Add**
22. Set the **External Phone Number:** 6782383600
23. Select **OK**
24. Under **Prioritized list of Registrar IP Addresses,** set the **IP Address:** 10.70.95.2

    This is the **static LAN IP address of the Cox E-SBC**. Please use the actual E-SBC LAN IP for your network.
25. Select **OK** and **OK**

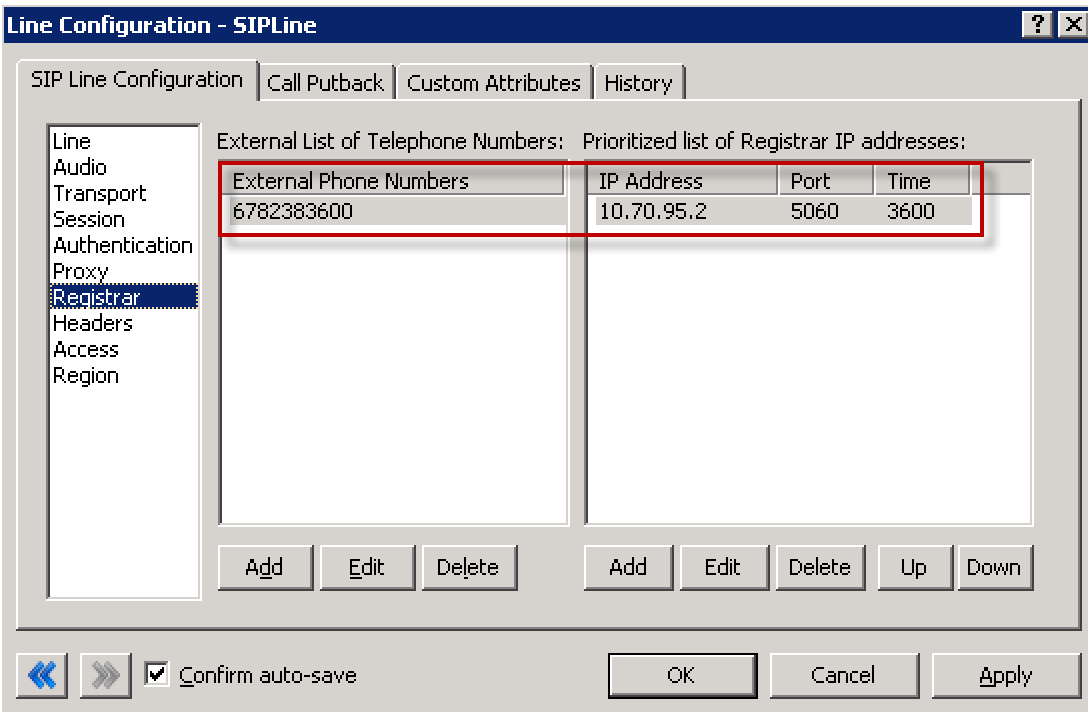

Figure 12 - SIP Carrier Options: Register

26. Verify the settings in the **Headers, Access, and Region** sections below.

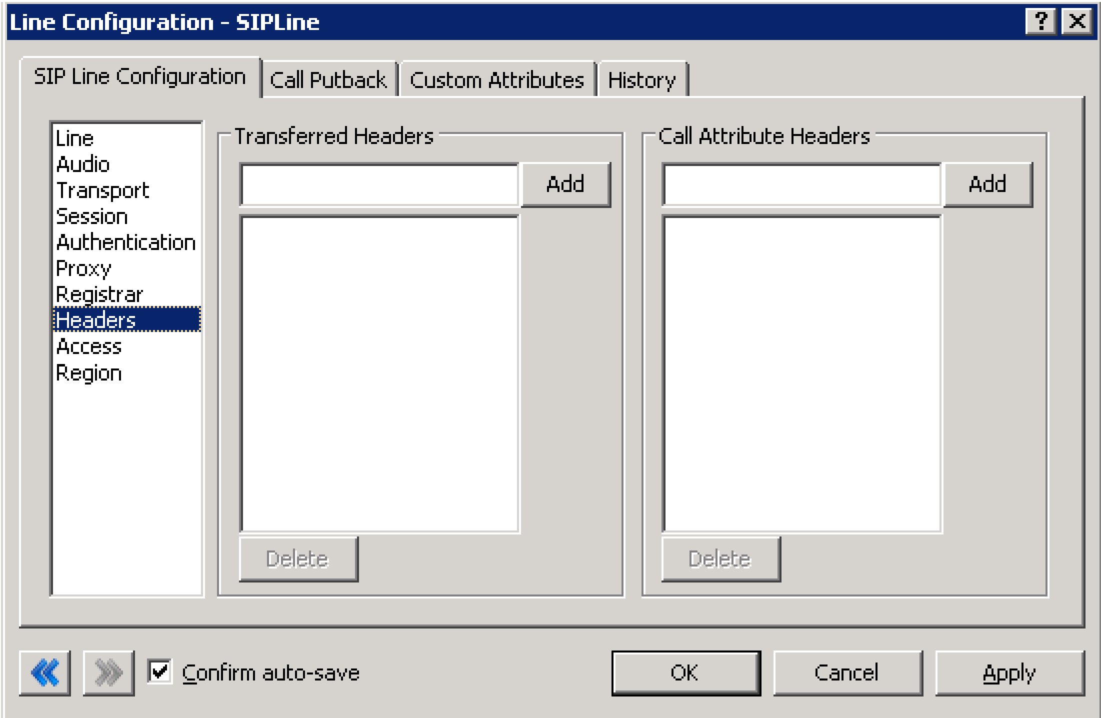

Figure 13 - SIP Carrier Options: Headers

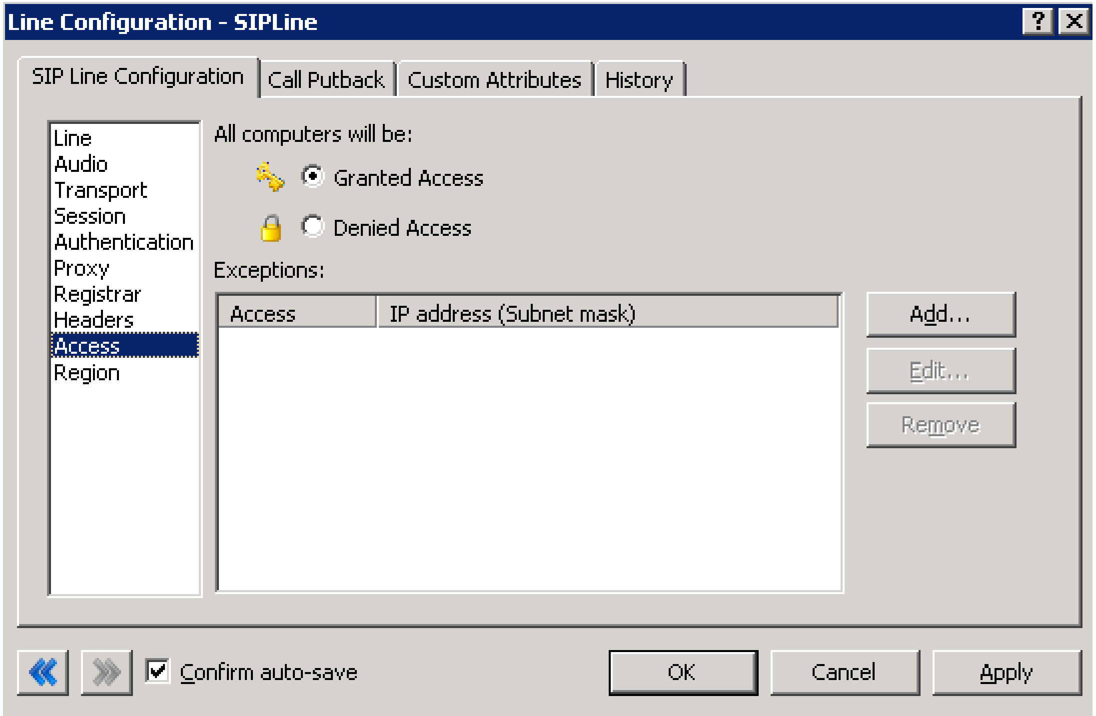

Figure 14 - SIP Carrier Options: Access

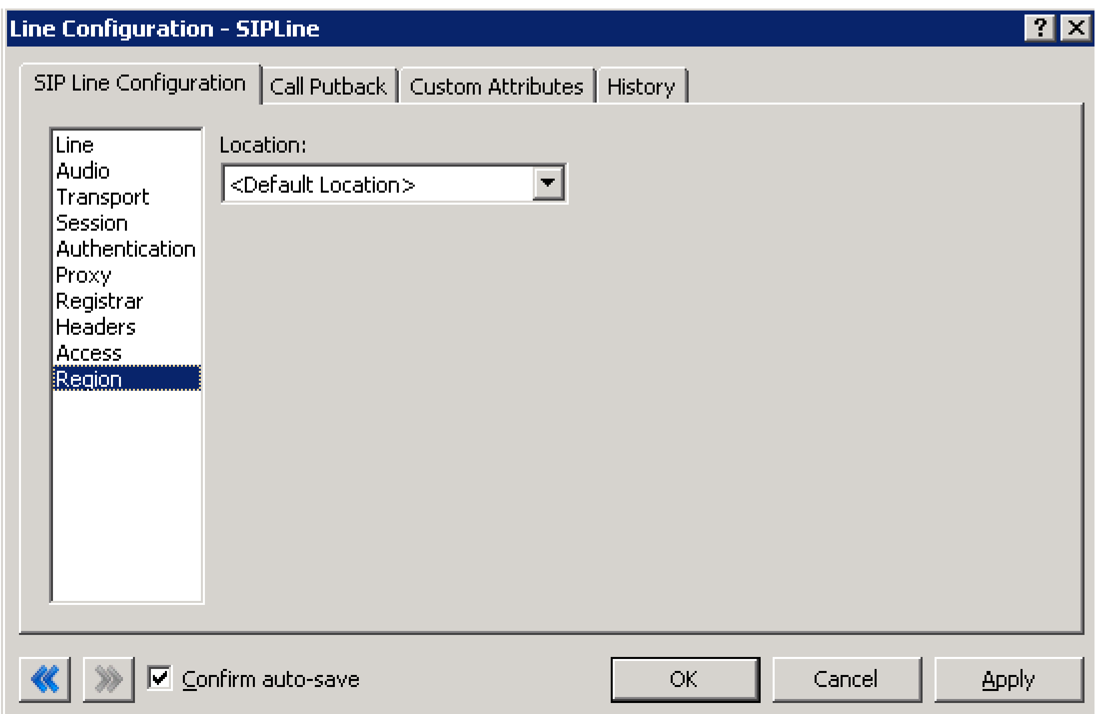

Figure 15 - SIP Carrier Options: Region

Date	Version	Author	Description
12/13/2012	1.01	John Mielko	Removed T.38 as supported