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SIP Trunking Configuration Guide
for
Sphericall
7.0.3 .106 patched 7.0.109

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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 the NEC Sphericall 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 Name, Calling 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); calls never traverse the Internet.

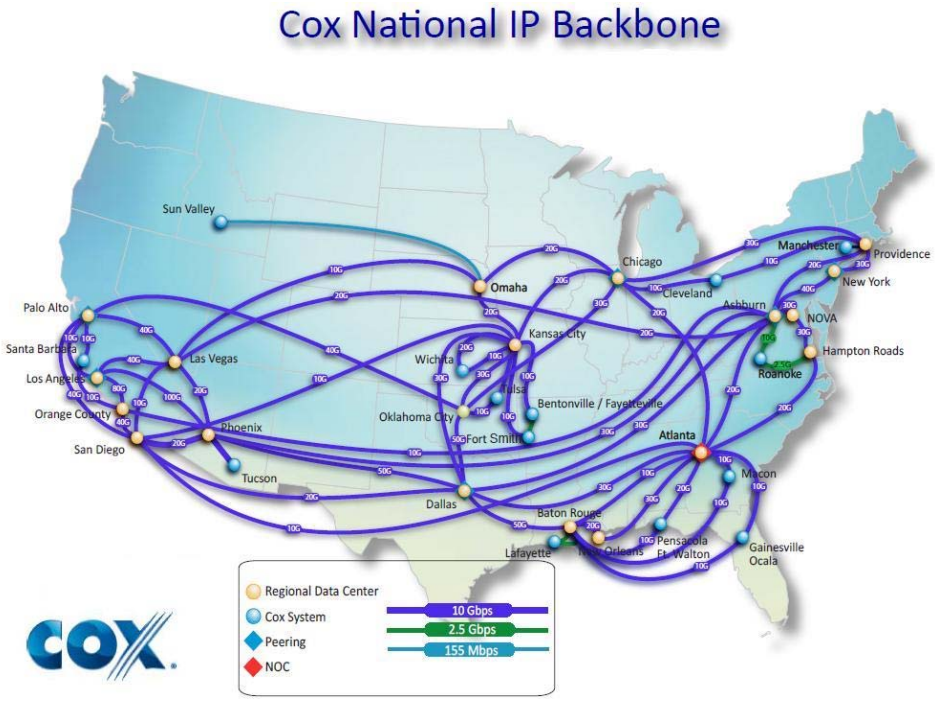

Figure 1 Cox Fiber Network

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 network for the SIP trunk reference configuration is illustrated below and is representative of a NEC Sphericall configuration

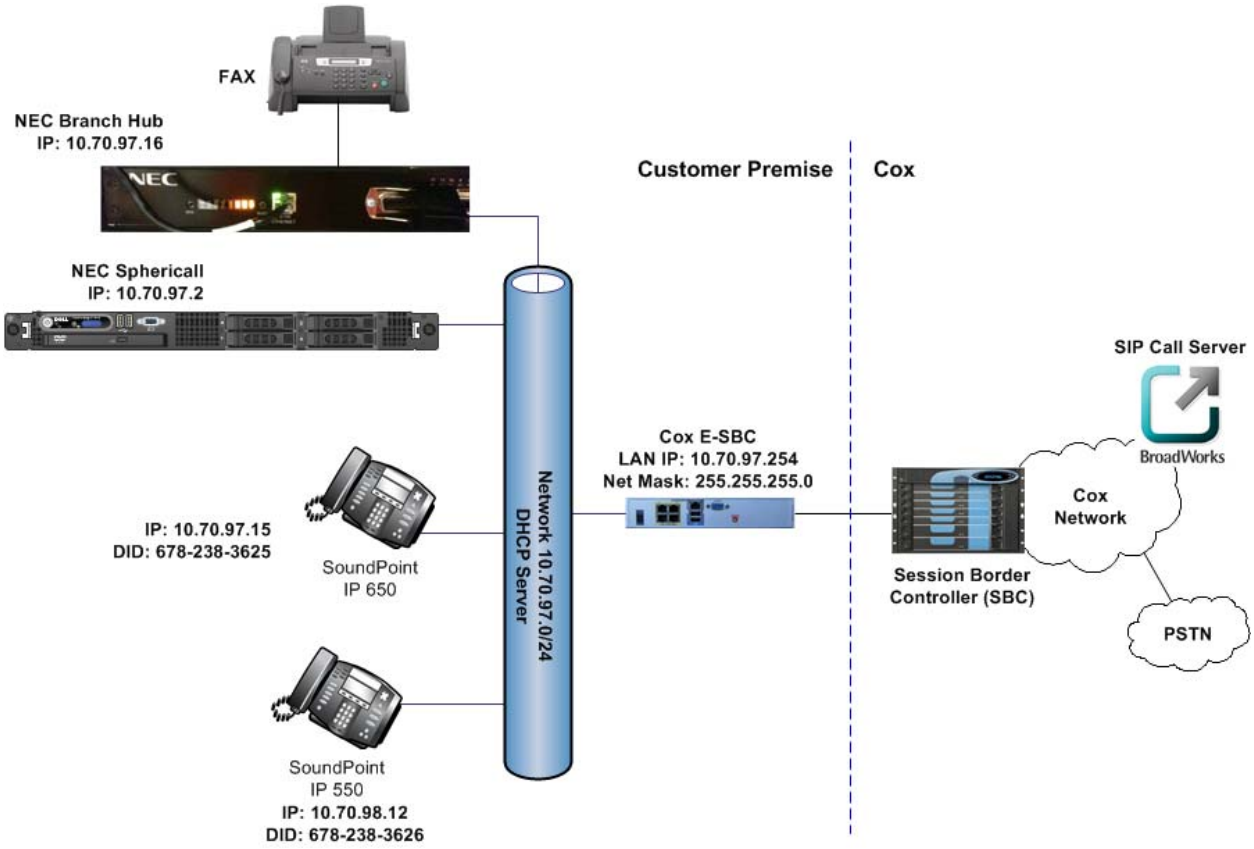

Figure 2 SIP Trunk Lab Reference Network

Note: The NEC Sphericall 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:

- NEC Sphericall PBX for voice features, SIP proxy and SIP trunk termination.
- NEC Branch Hub serves as an analog gateway
- 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

- NEC Sphericall
- NEC Branch Hub
- Analog fax machine
- EdgeMarc 4550 E-SBC

3.2 Software Requirements

- NEC Sphericall Release 7.0.3.106
- NEC Sphericall MGC patch 7.0.3.109
- NEC Sphericall Administrator 7.0.3.106
- EdgeMarc 4550 9.12.5 Release

4 Features

4.1 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 PBX from the PSTN. NEC Sphericall supports SIP Register with authentication. Cox implementation team provides the Pilot number and the authentication key, which should be provisioned in the NEC Sphericall. How to configure these in the NEC Sphericall are shown in Section 6.3.4.

4.2 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
- G711 Fax only
- Dial-Up Modem
- E911 Call
- RFC2833 transcoding

4.3 Features Not Supported

- T38 Fax
- Anonymous Call
- PBX Defined Caller ID (CLID spoofing)

5 Caveats and Limitations

- NEC Sphericall hairpins both call legs during call transfer, meaning the SIP sessions are not released after transfer. The sessions are released when the calls are released.
- PBX Auto Attendant(AA) to Off-net Numbers. When an Incoming call is addressed for the AA Sphericall immediately sends out 100Trying. But waits for two full seconds before sending a 18x message to the Network. This is per design from NEC. NEC explains this behavior as follows: Sphericall may have multiple nodes in the network. When the call is terminating to AA or a Sequential Ring group Sphericall queries the network for possible destinations before sending 18x to the Network. The Cox network is viewing the delay as a failed node in the network and cancels the call.
- Sequential Ring same failure as AA.
- When NEC Sphericall places a call on hold for any reason, it sends an INVITE to the network. The INVITE tells the network I am placing you on hold and sends a=sendonly with c=<ip address of device that placed call on hold>. The network responds with a 200OK SDP and sends a=recvonly. Less than on second later Sphericall sends a Second INVITE to the network. This INVITE changes the c=<to the ip address where the MOH is delivering the music from> with a=sendonly. This time the network sends a 200OK SDP with a=inactive. The end result is that the original PSTN party hears silence until the call is answered. This behavior does not cause calls to be dropped or talk path to be affected. This is not viewed as a defect by the Sphericall team.

  The result of this is that the originating PSTN caller hears silence in the following call flows:
  - Being Placed on Hold
  - Being transferred, Blind, or Consultative.
  - Being Placed in a Conference

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	10.70.97.254	
• LAN Subnet Mask	255.255.255.0	
NEC Sphericall **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.	10.70.97.2	
• NEC Branch Hub This is the IP address for the Branch Hub. This must be in the same network as the NEC IP-PBX	10.70.97.16	
• 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	10.70.97.1	

6 NEC Sphericall Detailed Configuration Steps

Equipment used for configuration setup:

- NEC Sphericall.
- NEC Sphericall software version release 7.0.3.106
- NEC Sphericall softare patch mgc 7.0.3.109.
- NEC Branch Hub

6.1.1 Sphericall Configuration

Sphericall is a software based IP-PBX solution. The Sphericall software is administered via Sphericall Administrator. This software is installed on the Sphericall server and can be accessed via Remote Desktop, or via a console connected directly to the Sphericall server. Once logged on to the server, execute the following steps to start "Sphericall Administrator".

- Double click "**Sphericall Administrator**"

Figure 3 Sphericall Icon

- The User Account Control window may appear, if it does appear click **Yes**

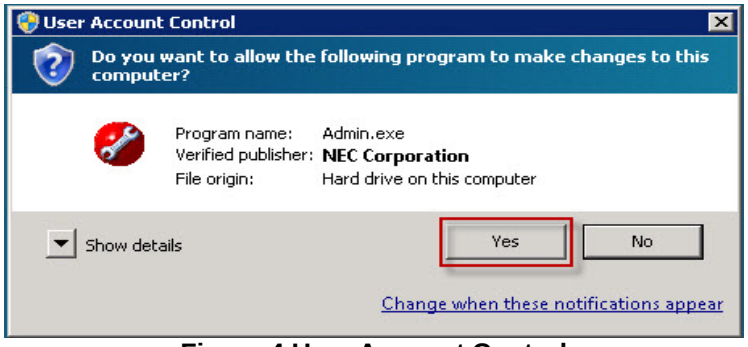

Figure 4 User Account Control

- The Sphericall Administrator window is now seen. The following sections will have detailed steps on how to provision extensions, sip trunk, dial plan, etc.

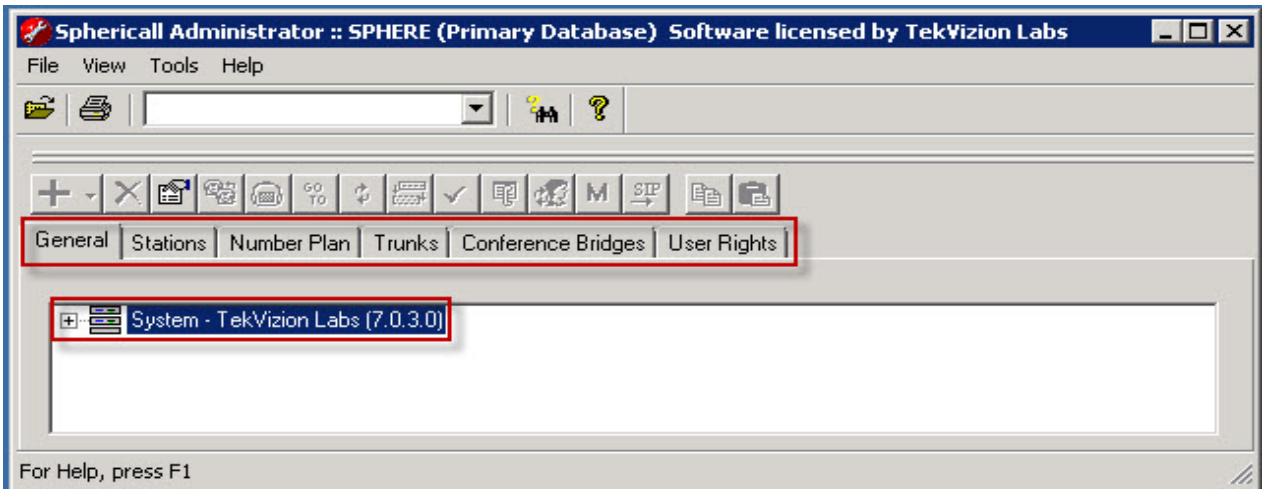

Figure 5 Sphericall Administrator

6.2 Create SIP Trunks

Note: 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. In the SA Click on **General**
2. In the Configuration Pane right click on **System**

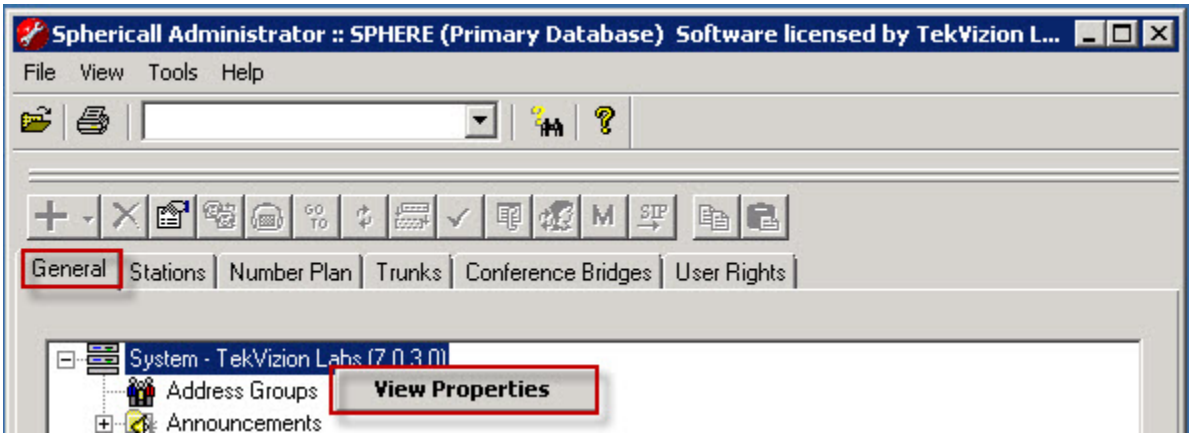

Figure 6 View Properties

3. The **User Agent Profile** dialog window appears. There are User Agent Parameters that can be modified based on the User Agent defined in the sip messages. If there is no User-Agent defined in the incoming SIP message then Generic SIP Trunk is the User Agent Profile that is used to dictate signaling.
4. The Screen shots below will show which options were modified from their default config. The User Agent Parameters that have been modified are:
    - Set **OPTIONS Request** = Supported
    - Set **Send Forwarding Information** = Using Diversion Header
    - Set **Send Transferring Information** = Using Diversion Header
    - Set **Xpidf + xml support for Presence** = Unsupported
5. Click **OK** once the parameters are modified

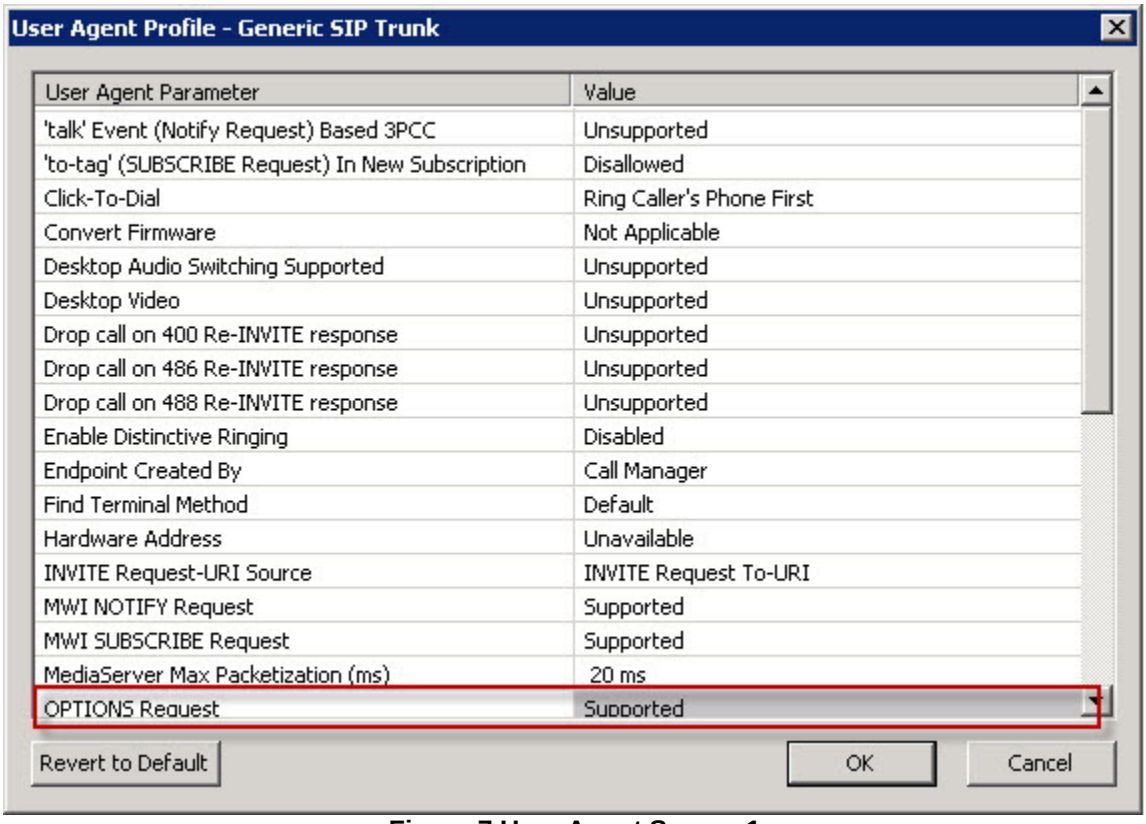

Figure 7 User Agent Screen 1

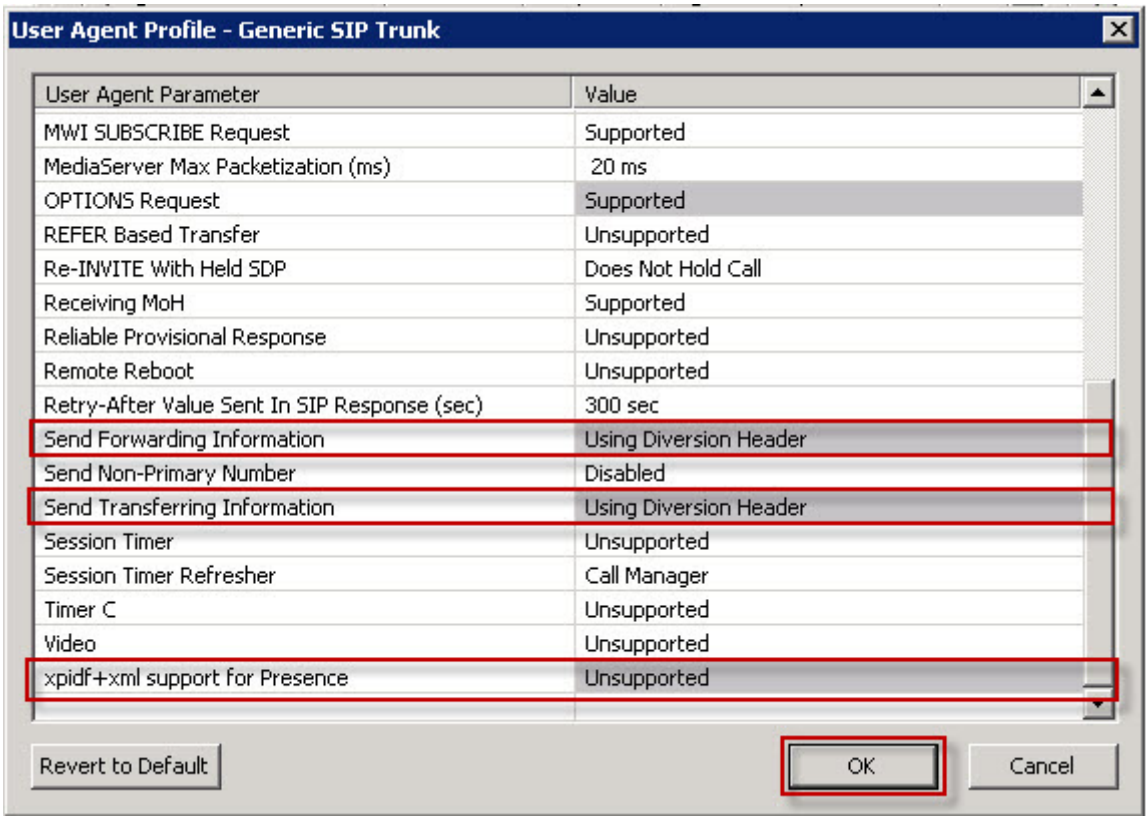

Figure 8 User Agent Screen 2

1. In the SA Click on **Trunks**
2. In the Configuration Pane right click **Add** > **SoftTrunk**
3. The **Configure Service Provider** dialog window appears.
4. Set the following values:
   - **Description**: Cox Sphericall. This is used for identification purposes only.
   - **Account**: 6782383625 This is for example purposes only.
   - **Service Provider Domain**: This is the static LAN IP address of the Cox E-SBX. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.97.254
   - **Outbound Proxy**: This is the static LAN IP address of the Cox E-SBX. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.97.254
   - **Port**: 5060
   - **Registration Type**: Outbound This is set as default
   - **Primay** MGC Sphere This is set as default
   - **Secondary** MGC: This is set as default

- **Contact Domain**: This is the static LAN IP address of the Cox E-SBX. Please use the actual E-SBC LAN IP for your network. The IP Address used in this configuration is 10.70.97.254
- **Preferred Transport**: UDP
- **User Agent:** Generic SIP Trunk

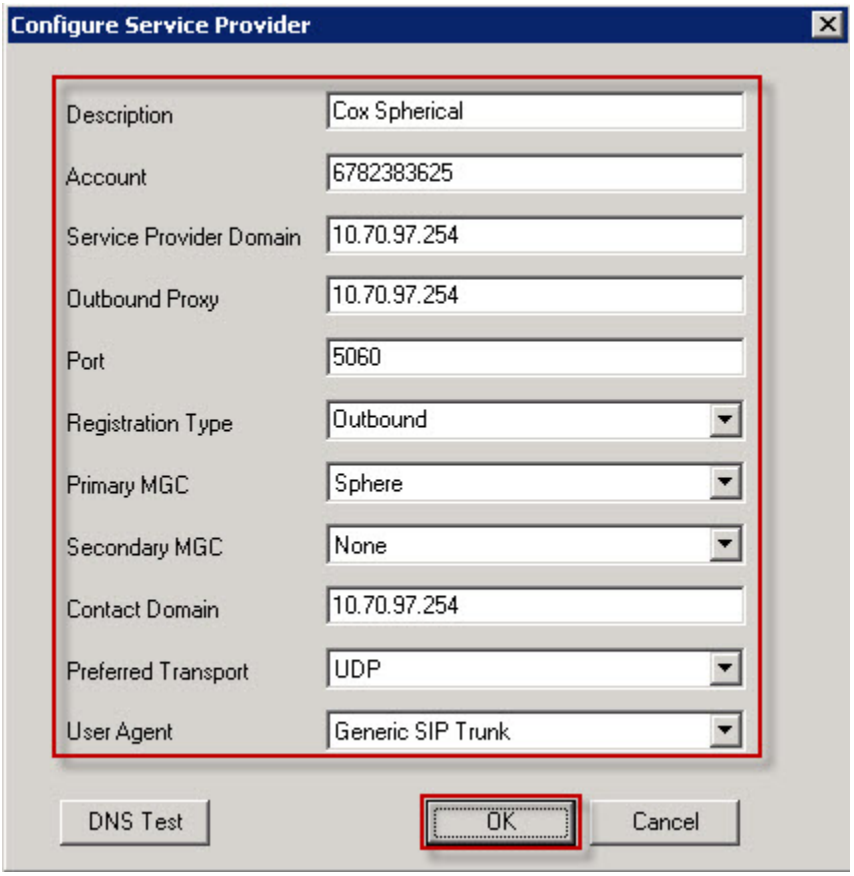

Figure 9 Configure Sip Trunk

5. In the SA Click on **General**
6. Navigate to **Telephony Areas > Default Area**
7. Right Click on **Default Area**

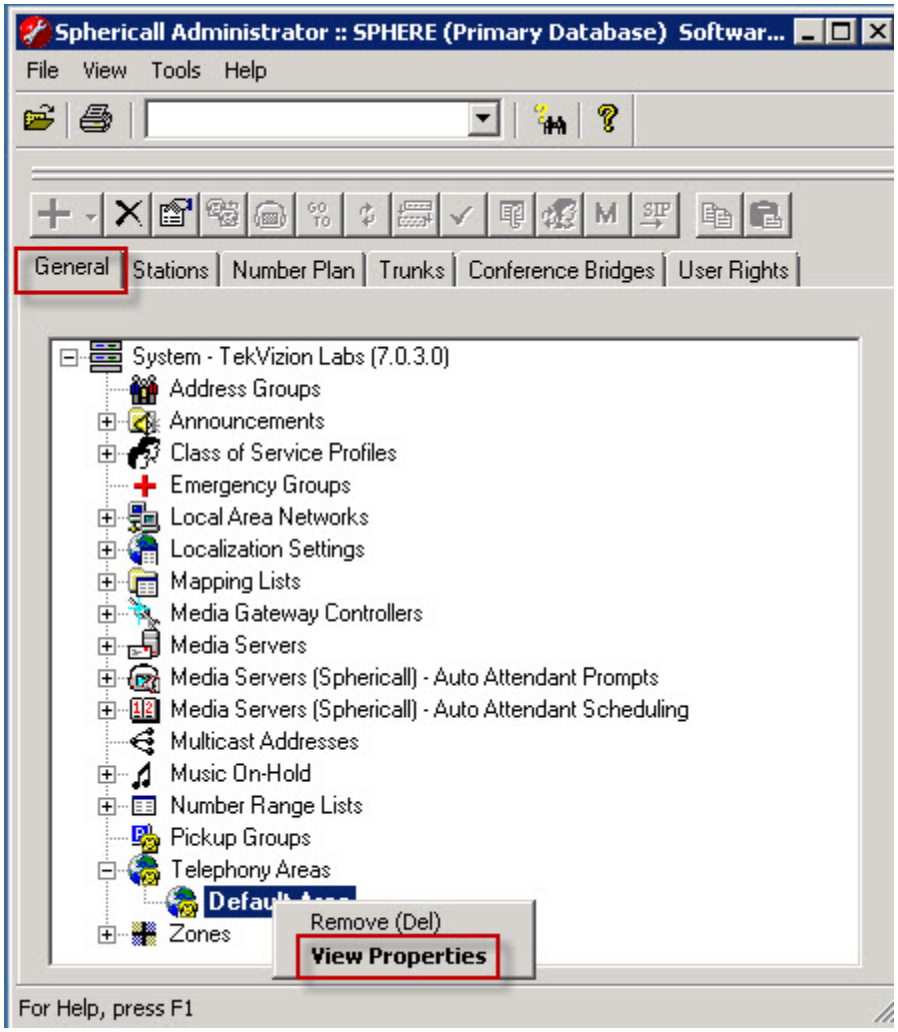

Figure 10 SA Telephony Areas Properties

8. The **Properties for Telephony Area:Default Area** dialog window appears
9. Click on **General**
10. Set **Template**: Template – US 10 digit local dialing
11. Set **Area Code**: 678. This is the Area Code used for this example.
12. Set **PBX Number**: 2383625 This is the number used for this example
13. Confirm **Default Outside Service**: 9
14. Set **Local Area Code**: 678 This is the Area Code used for this example.
15. Click on **OK**

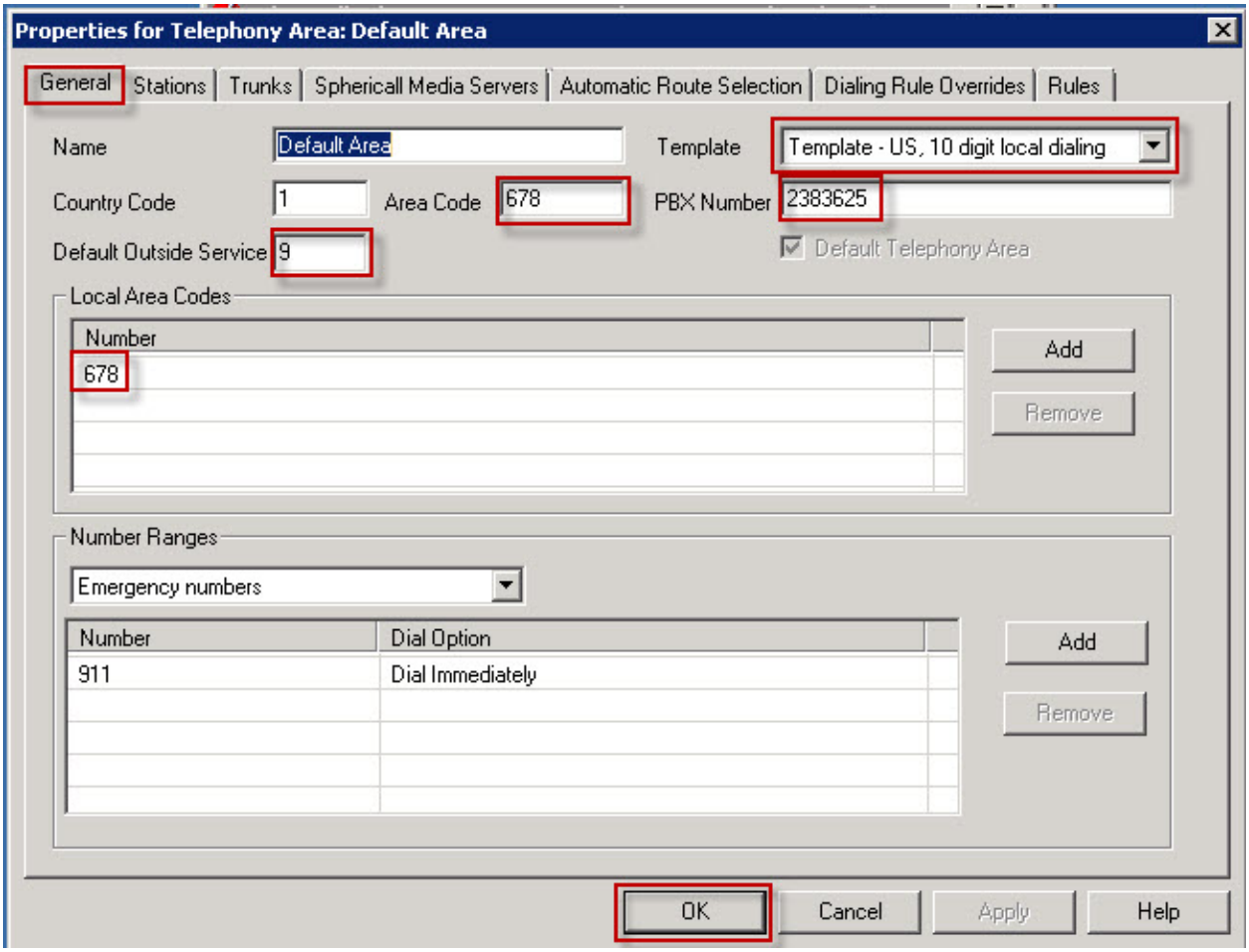

Figure 11 Telephony Default Area

16. In the SA Click on **Trunks**
17. In the Configuration Pane right click on **Port 1 Hub 5 port 1** These values may be different on the system that is being configured.
18. Select **View Properties**
19. The **Properties for Trunk hub 5 port 1** dialog window appears.
20. Click on **General**
21. Set **Total Capacity**: 10 The capacity numbers are depending on the service that were purchased from Cox Communication.
22. Set **Inbound Capacity**: 10
23. Set **Outbound Capicity**: 10
24. Set **Outbound Caller ID**: All Numbers
25. Click **OK**

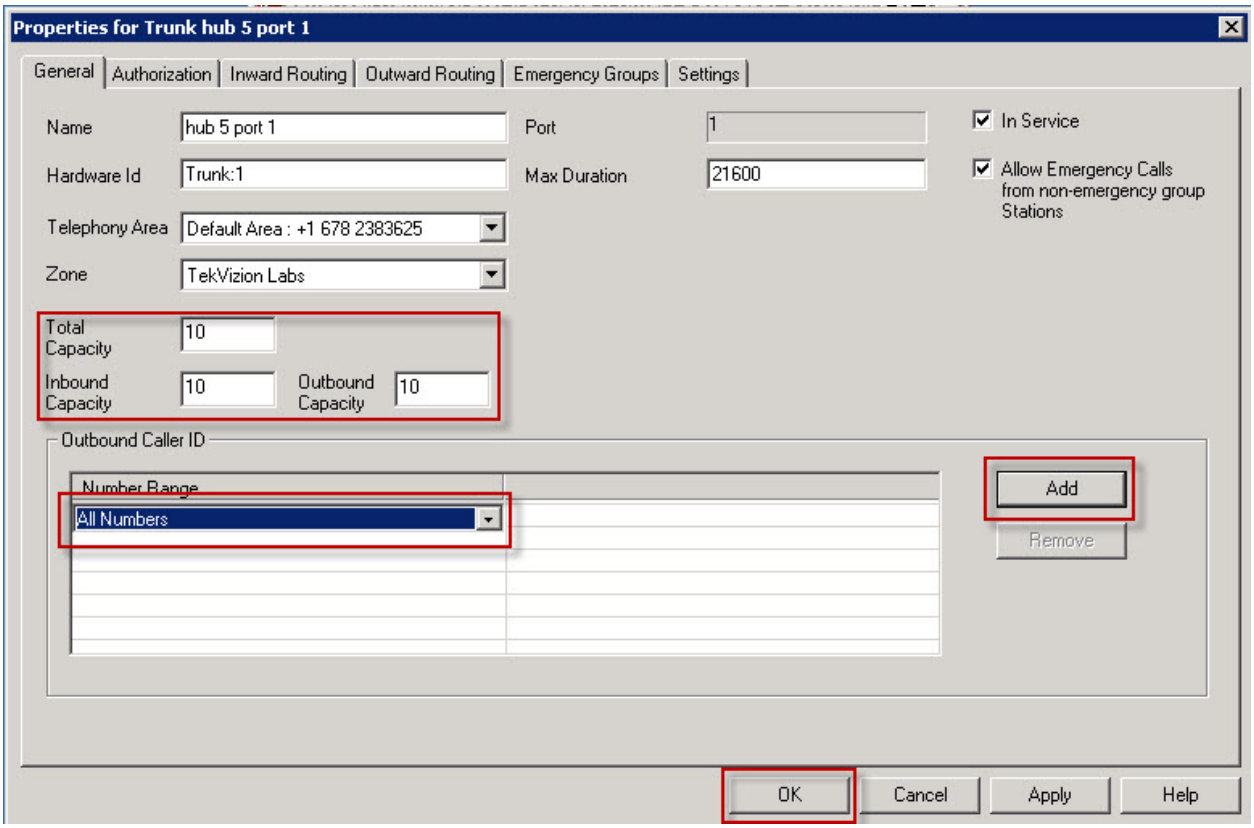

Figure 12 Properties for SIP Trunk

26. Click on **Authorization**
27. Check **Use Authorization**:
28. Set **Account**: 6782383625
29. Set **Password**: **********
30. Verify **Password**:**********
31. Set **Realm**: Broadworks Sphericall picks this up from the 401 UnAuthorized information that is sent from COX.
32. Set **Authorization Type**: To Respond
33. Click **OK**

    The actual SIP Registration Password and Username 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!

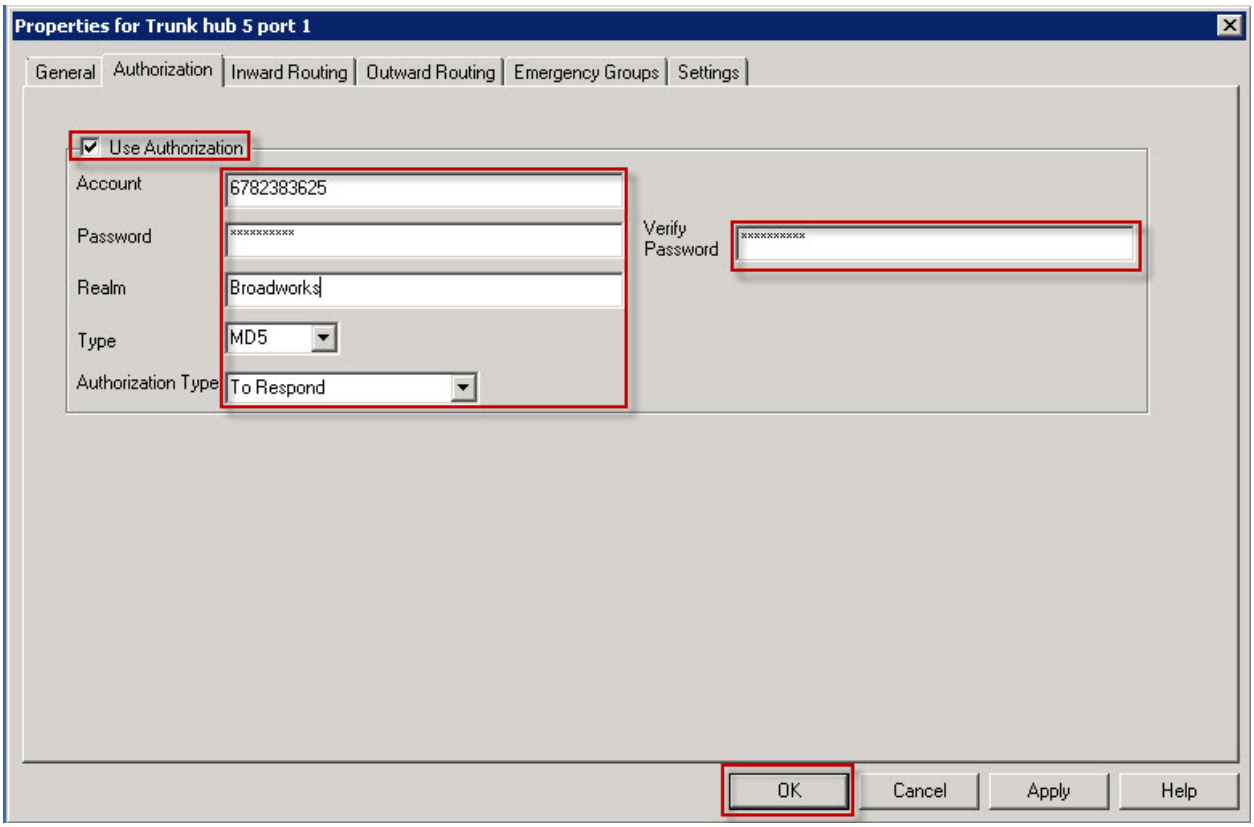

Figure 13 Authorization Information For Trunk

34. Click on **Settings**
35. Click on **Add**
36. In the Scroll down menu under Name select **Maximum Call Duration Timer**: 70.
37. In the Scroll down menu under Name select: **SIP> Auto Switch To TCP**=Disabled.
38. Select **OK**

NOTE*** In step 37 Disable NEC Sphericall from switching from UDP to TCP once the packet gets to big. This comes into play with forwarding test cases. The result of step 37 not being executed is basic calls will work with no problem but call forward calls will not.

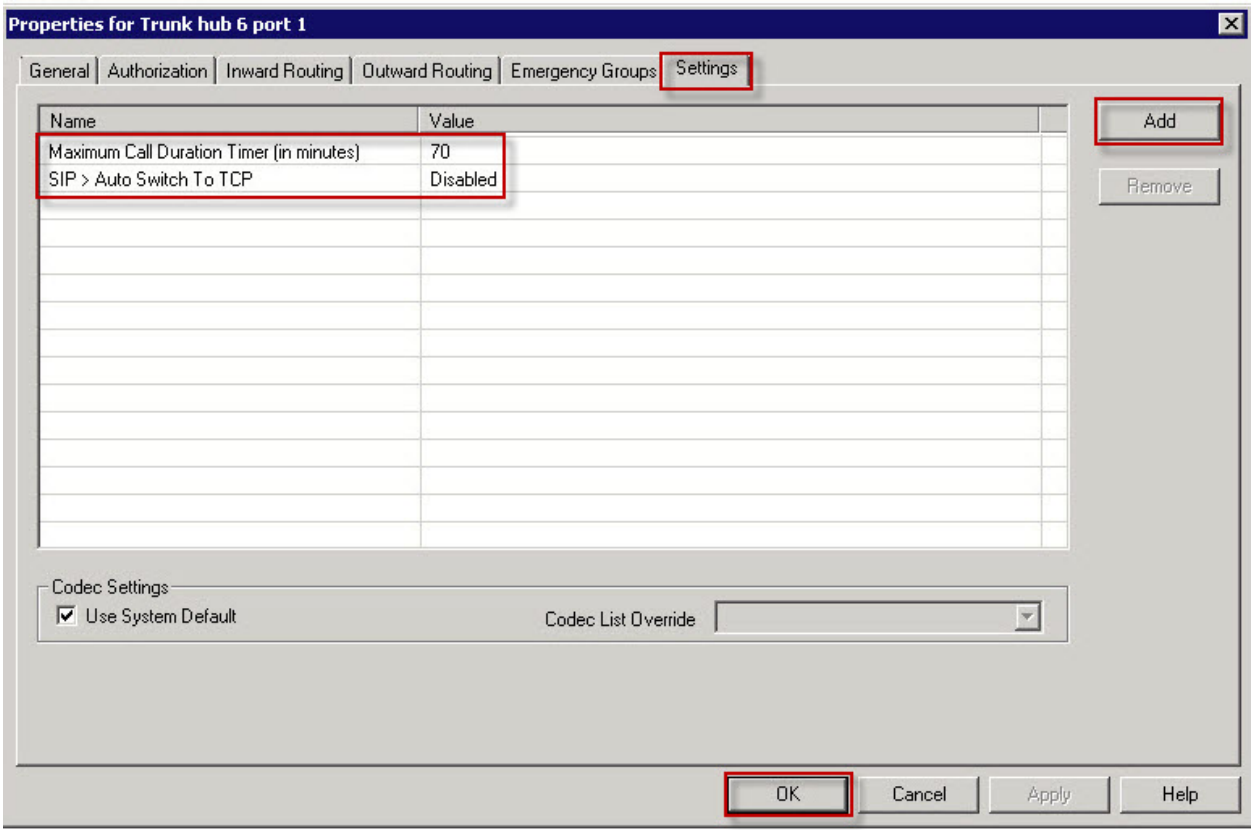

Figure 14 SIP Trunk Settings