SIP Trunking Configuration Guide
for IntelePeer with
3CX Phone System
Version 14.0.44241.523

04/21/2016

Table of Contents

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 IntelePeer SIP Trunking to a 3CX IP-PBX.

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
- 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 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 Plano, Texas.

For more information on tekVizion and its practice areas, please visit tekVizion's website at www.tekvizion.com

3 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a 3CX configuration.

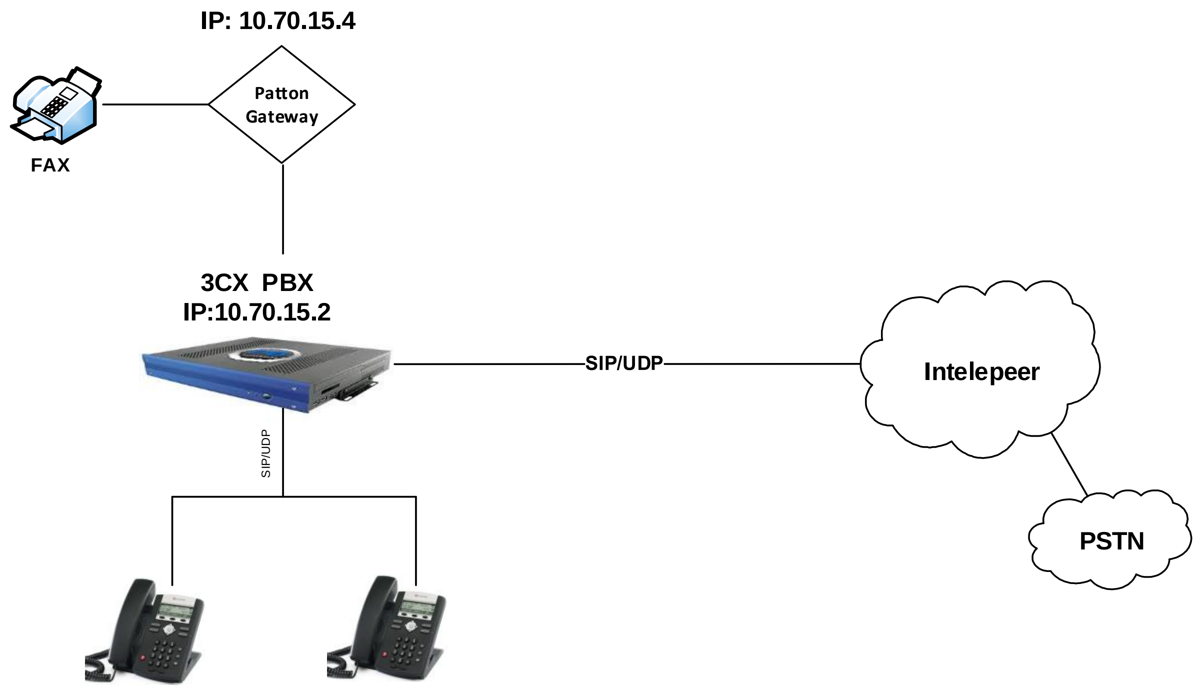

Figure 1: SIP Trunk Lab Reference Network

3.1 Hardware Components

- 3CX server
- Patton Gateway

3.2 Software Requirements

- 3CX Version 14.0.44241.523

4 Features

4.1 Features Supported

- Basic calls using G.711ulaw
- International Call
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Waiting
- Call Park
- Do Not Disturb
- Three-Way Calling
- Casual dialing
- Fax Send/Receive

4.2 Features Not Supported by PBX

- Inband DTMF

4.3 Caveats and Limitations

- INVITE with replaces for transfer scenario: B Party initiates the transfer when C Party is ringing. PBX sends INVITE with replaces. On the PSTN side, we see two calls hitting the phone where first call with B party's DID and second call with A party's DID. IntelePeer has to replace the first call with the second. Confirmation needed if IntelePeer will support INVITE with replaces for call transfer scenario.

5 Configuration

5.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure 3CX PBX for SIP Trunking as well as all features that were tested

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	SIP Trunks	Section 5.2.1
Step 2	Edit SIP Trunks	Section 5.2.2
Step 3	Extension Setup	Section 5.2.3
Step 4	Inbound Rules	Section 5.2.4
Step 5	Outbound Rule	Section 5.2.5
Step 6	International Dialing	Section 5.2.6

5.2 3CX Detailed Configuration Steps

5.2.1 SIP Trunks

1. From the **3CX Server Manager** main menu, click **SIP Trunks**
2. Click **Add Provider**

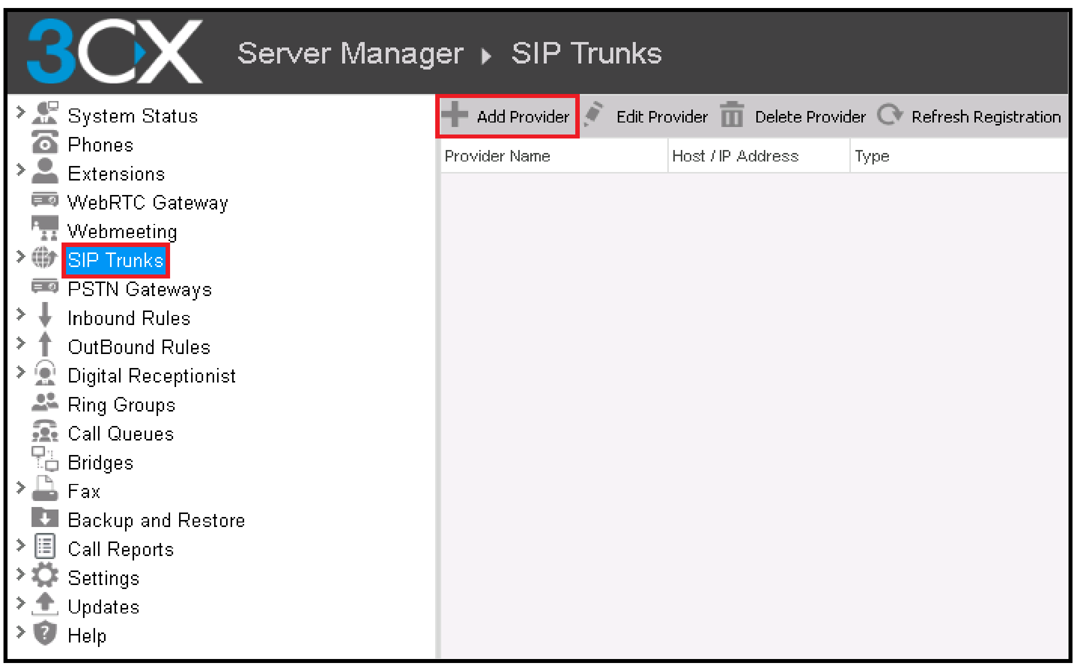

Figure 2: SIP Trunks

3. Set **Name of Provider**: Intelepeer is used in this example
4. Set **Country:** Generic is used in this example
5. Set **Provider**: Generic SIP Trunk is used in this example
6. Click **Next**

Figure 3: SIP Trunks – Cont.

7. Set **SIP Server Hostname or IP:** Enter the IP address of IntelePeer
8. Click **Next**

Figure 4: SIP Trunks – Cont.

9. Set **External Number:** Enter the Pilot number or any one of your DIDs to be used as a pilot number
10. Set **Maximum Simultaneous Calls:** 5 is used in this example
11. Click **Next**

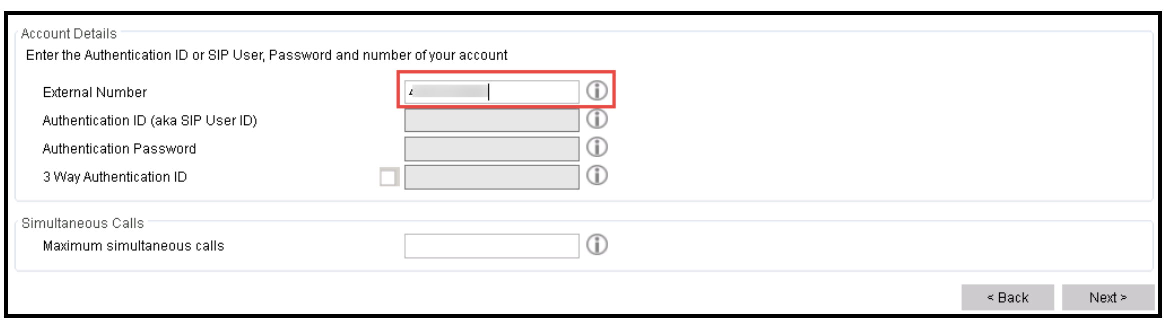

Figure 5: SIP Trunks – Cont.

12. **Office Hours** fields are left as default
13. Click **Next**

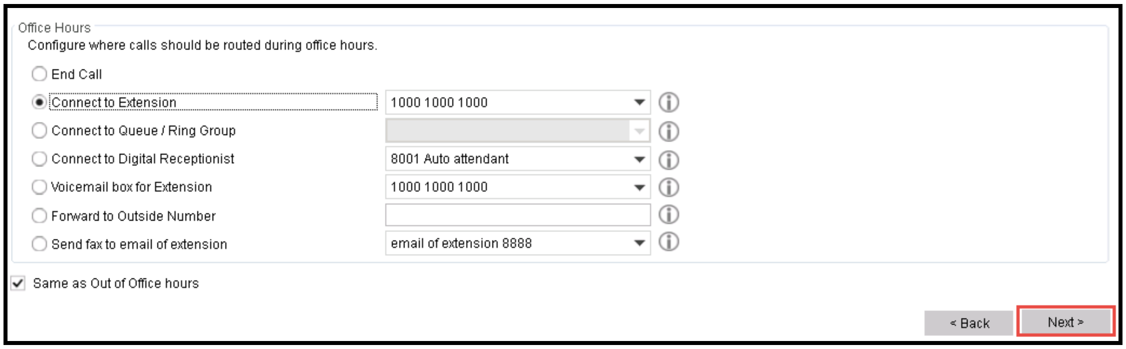

Figure 6: SIP Trunks – Cont.

5.2.2 Edit SIP Trunks

1. From the **3CX Server Manager** main menu, click **SIP Trunks**
2. Highlight **Intelepeer**
3. Click **Edit Provider**

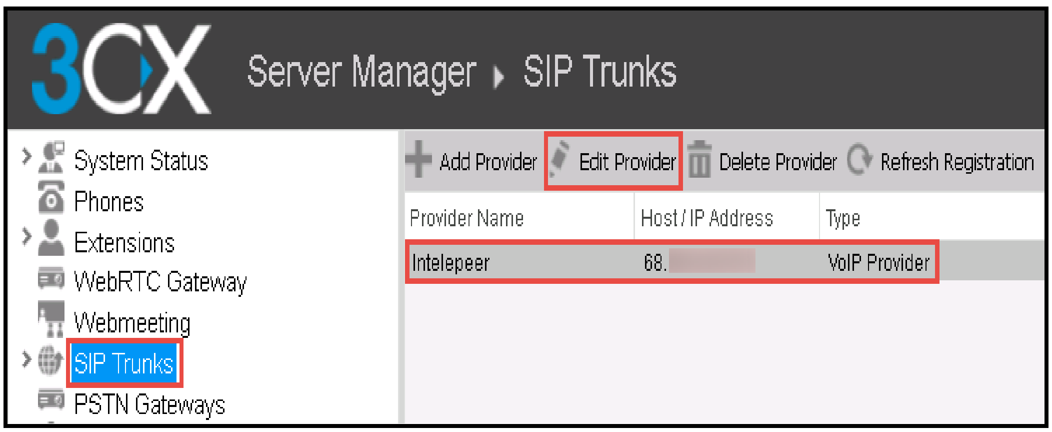

Figure 7: Edit SIP Trunks

4. Select the **General** tab
5. Set **SIP Server Hostname or IP:** Enter IntelePeer's IP address
6. Set **SIP Server Port:** Enter IntelePeer's listening port number
7. Set **Outbound Proxy Hostname or IP:** Enter IntelePeer's IP address
8. Set **Outbound Proxy Port (default is 5060):** 5060 is used in this example

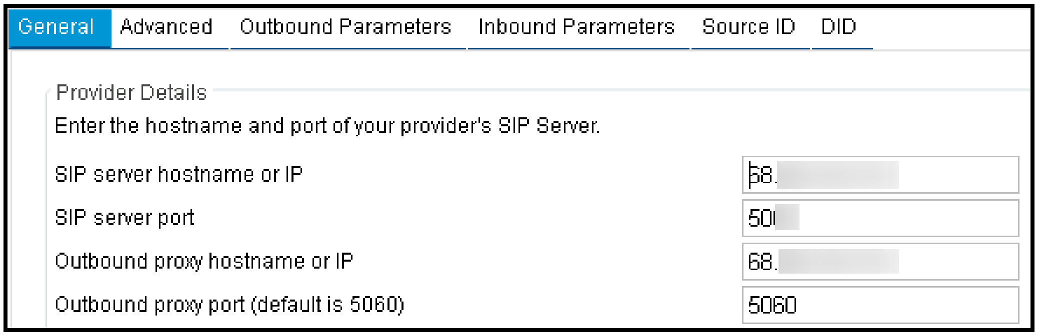

Figure 8: Edit SIP Trunks – Cont.

9. Select the **Advanced** tab
10. **PBX Delivers Audio:** Checked
11. Set **Assigned Codecs**: Assign supported codecs

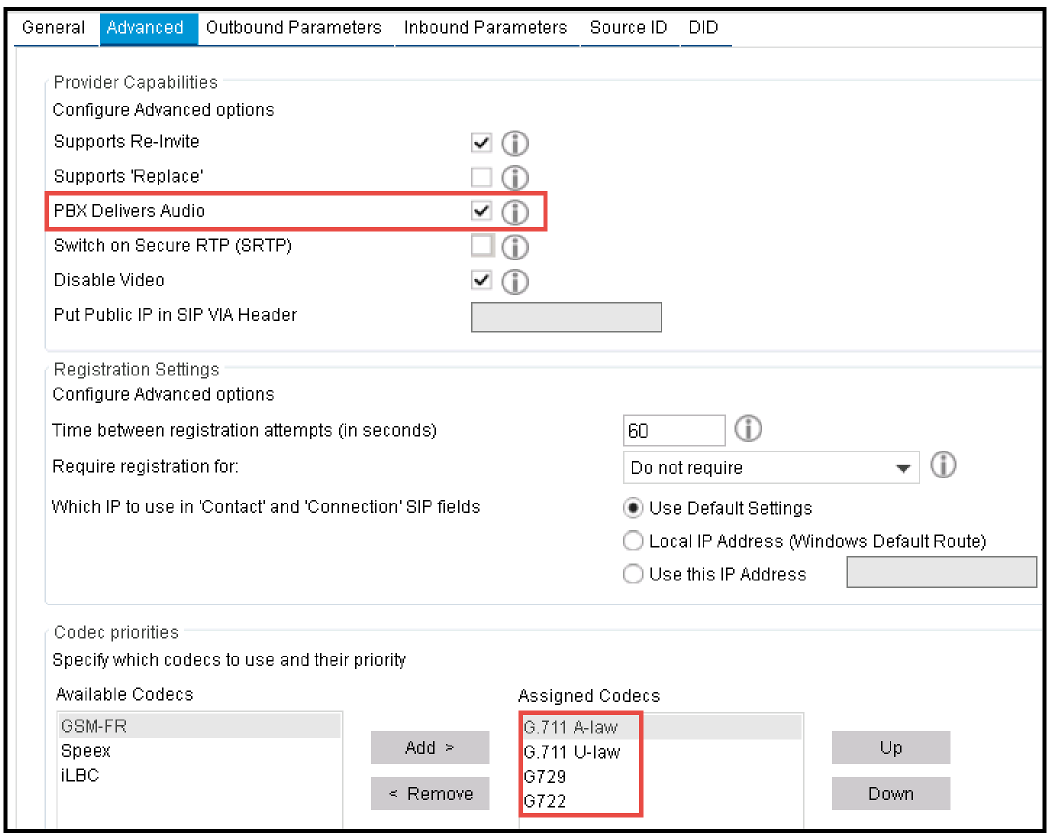

Figure 9: Edit SIP Trunks – Cont.

12. Select the **DID** tab
13. Add DID numbers into the blank DID fields
14. Click **Add**
15. Repeat the process for all required DIDs
16. Click **Apply** at bottom right of screen

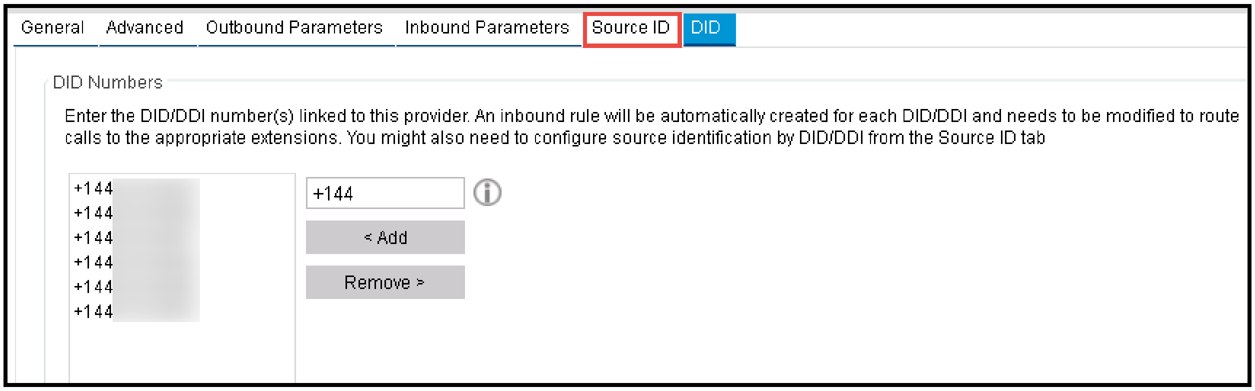

Figure 10: Edit SIP Trunks – Cont.

17. Select the **Source ID** tab
18. **Source Identification by DID**: Checked
19. Click **Add DID**
20. In the pop up window, select all desired DIDs
21. Click **OK**
22. Click **Apply** at bottom right of screen

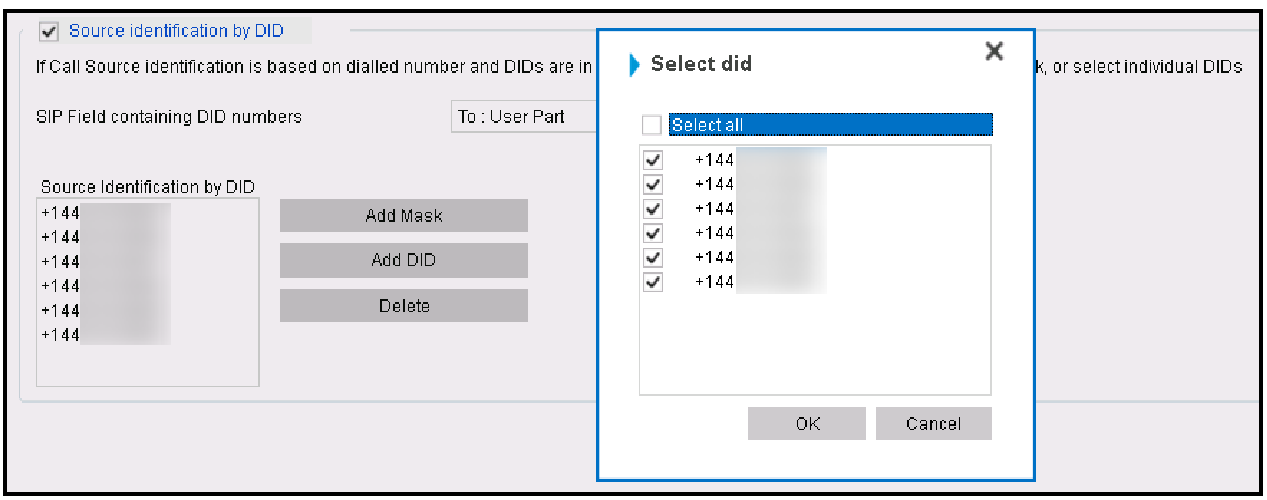

Figure 21: Edit SIP Trunks – Cont.

5.2.3 Extension Setup

1. From the 3CX Server Manager main menu, click **Extensions**
2. Click **Add Extension**

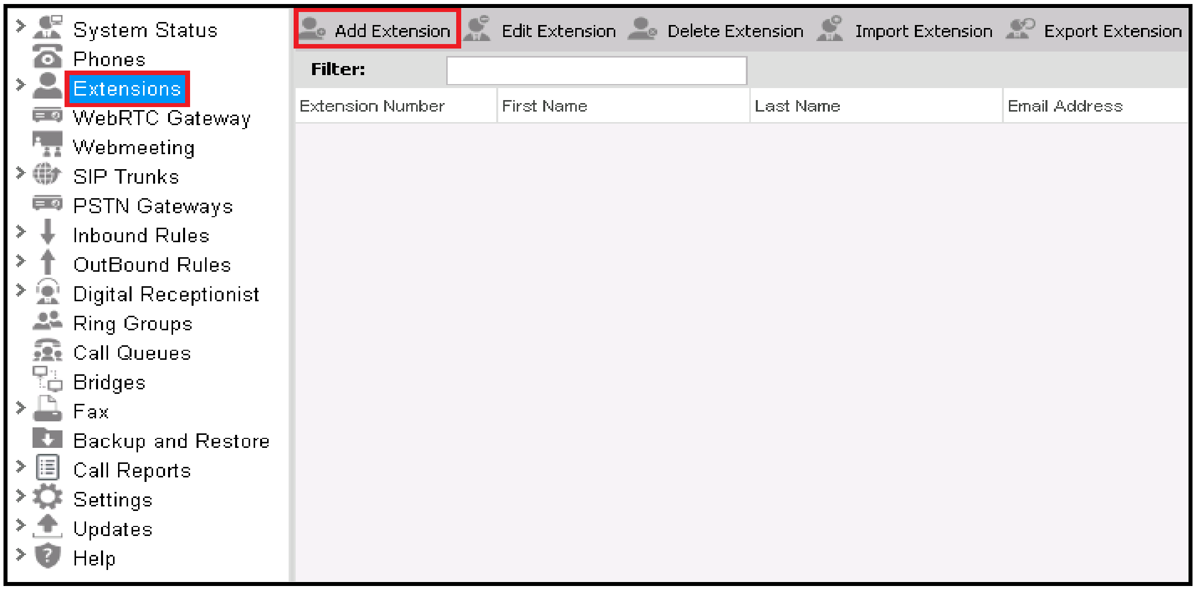

Figure 12: Extension Setup

3. Set **Extension Number:** 1001 is used as an example
4. Set **First Name:** Test is used as an example
5. Set **Last Name:** Phone1 is used as an example
6. Set **ID**: 1001 is used as an example
7. Set **Password:** 456 is used as an example

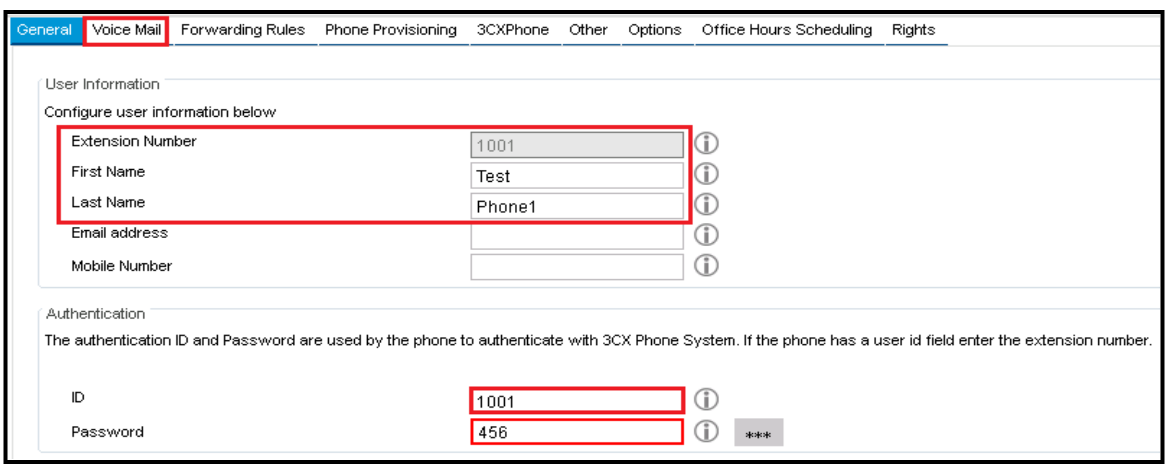

Figure 13: Extension Setup – Cont.

8. Select the **Voice Mail** tab
9. **Enable Voice mail:** Checked
10. Set **PIN Number:** 8724 is used as an example

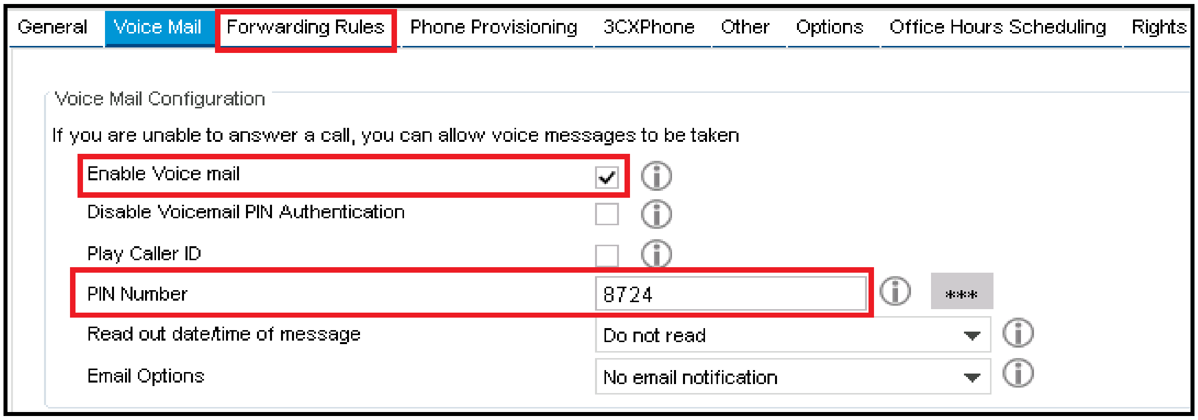

Figure 3: Extension Setup – Cont.

11. Select the **Forwarding Rules** tab
12. Select the **Available** tab
13. **I want to be able to accept more than 1 call at the same time – uses Phone Status:** Checked
14. Select the **Phone Provisioning** tab

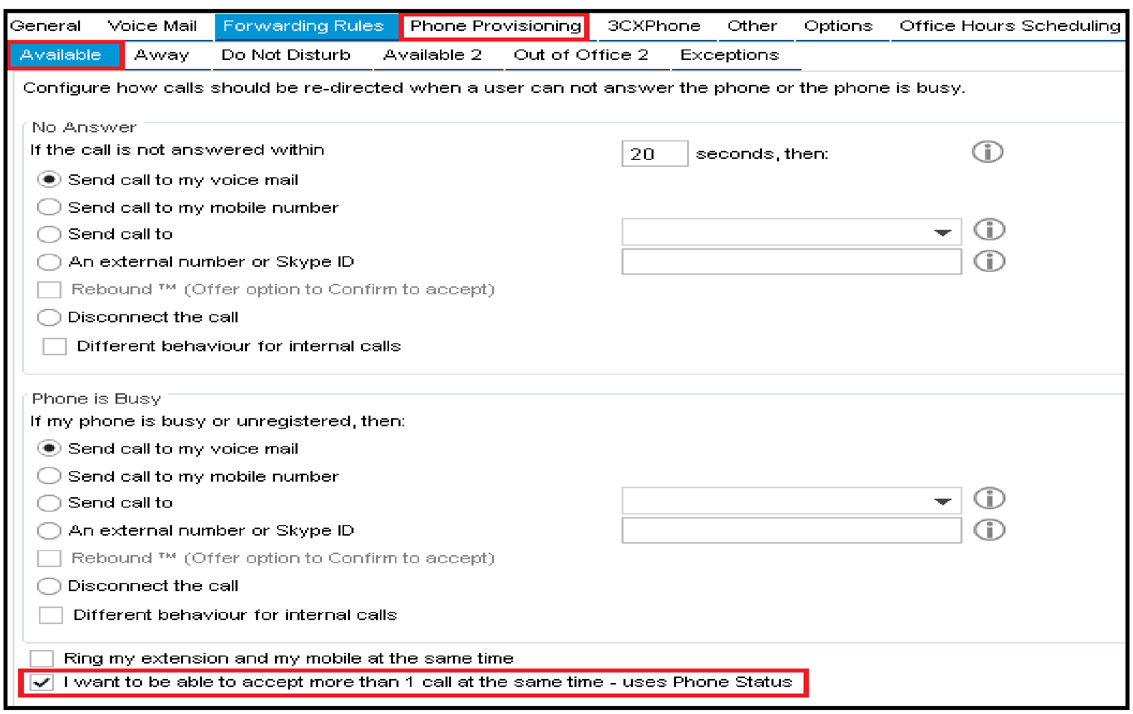

Figure 15: Extension Setup – Cont.

15. Set **MAC Address:** Enter the MAC address of the phone
16. Set **Model**: Polycom SPIP335(Firmware 4) is used as an example
17. Verify **Select Interface** Field is set to the PBX IP **10.70.15.2**, the 3CX IP address
18. Verify Preferred Codec is set to **PCMU**

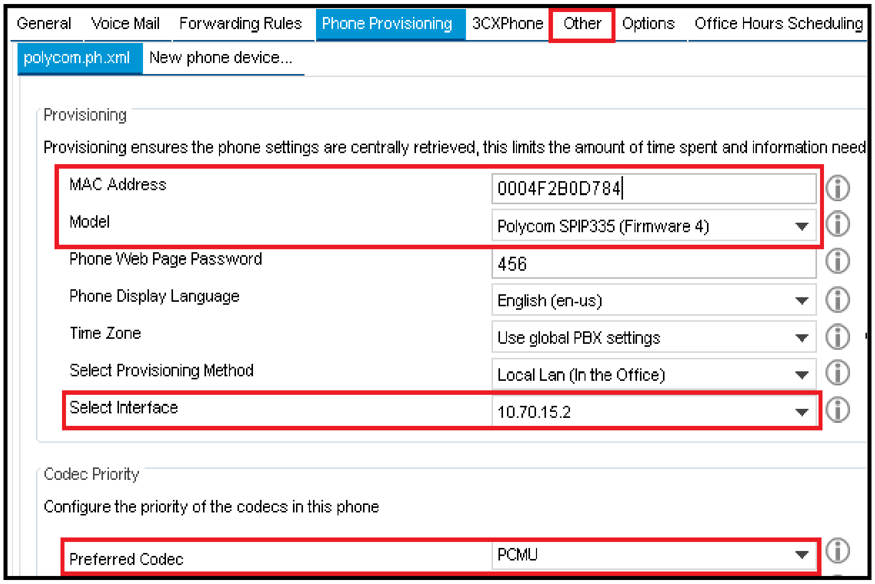

Figure 16: Extension Setup – Cont.

19. Select the **Other** tab
20. Verify **Current Status** is **Available**
21. Verify **Queues Status** is set to **Logged In**
22. Set **Outbound Caller ID**: +144XXXXXXXXX is used as an example
23. Click **Apply** and **OK** at bottom right of screen
24. Repeat the above process for each phone

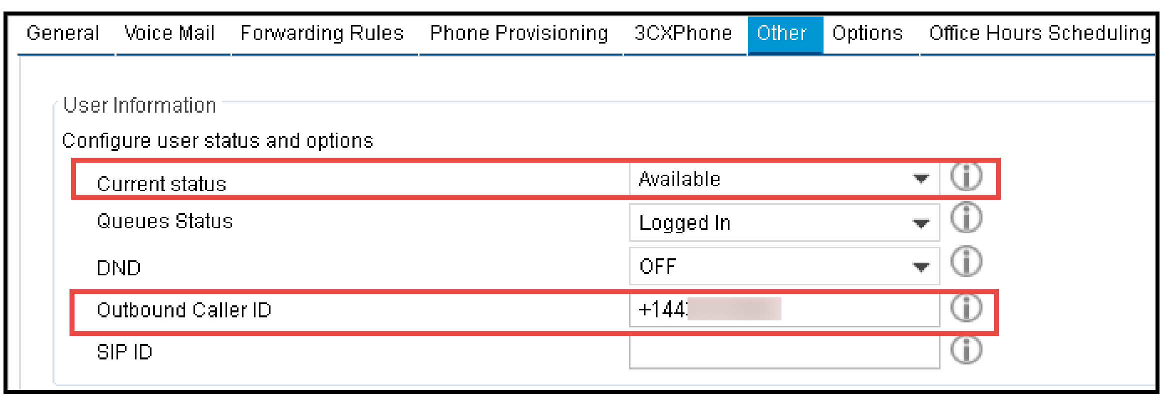

Figure 17: Extension Setup – Cont.

5.2.4 Inbound Rules

1. From the 3CX Server Manager main menu, navigate to **Inbound Rules**
2. Click **Add DID**

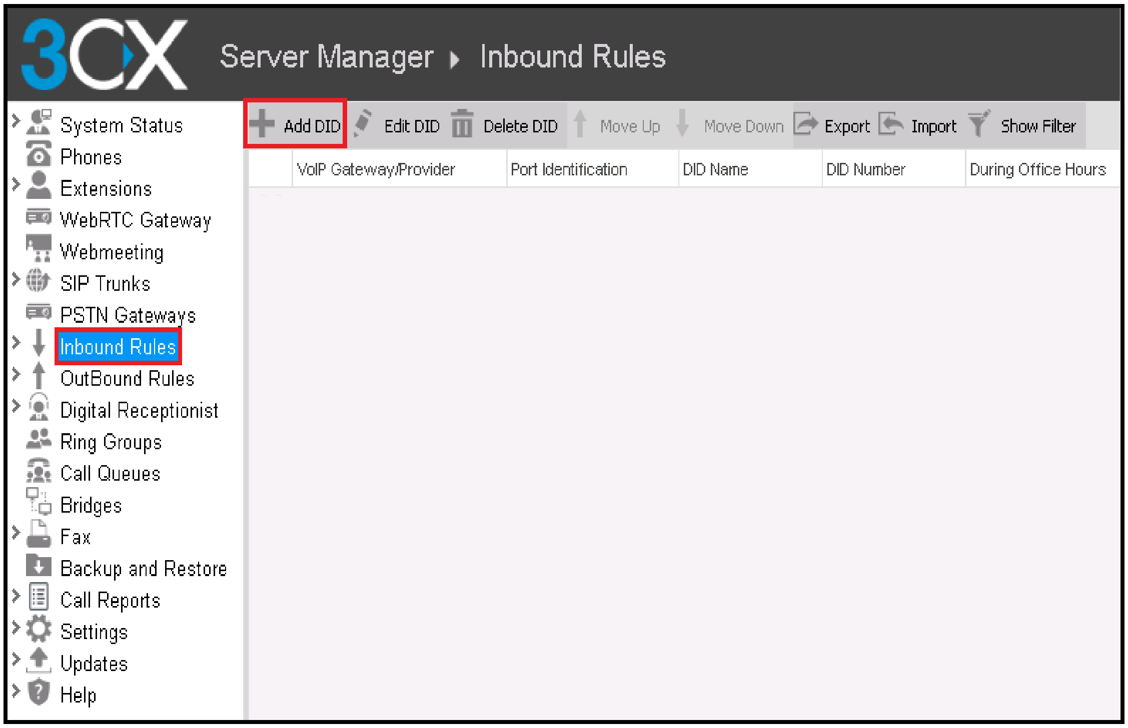

Figure 18: Inbound Rules

3. Set **Inbound Rule Name: Test Phone1** is used as an example
4. Set **DID/DDI number/mask: +144XXXXXXXX** is used as an example
5. **Connect to Extension:** Checked
6. **1001 TestPhone1** is selected as an example name from the drop down arrow
7. Click **Apply** and **OK** at bottom right of screen
8. Repeat process for each additional extension

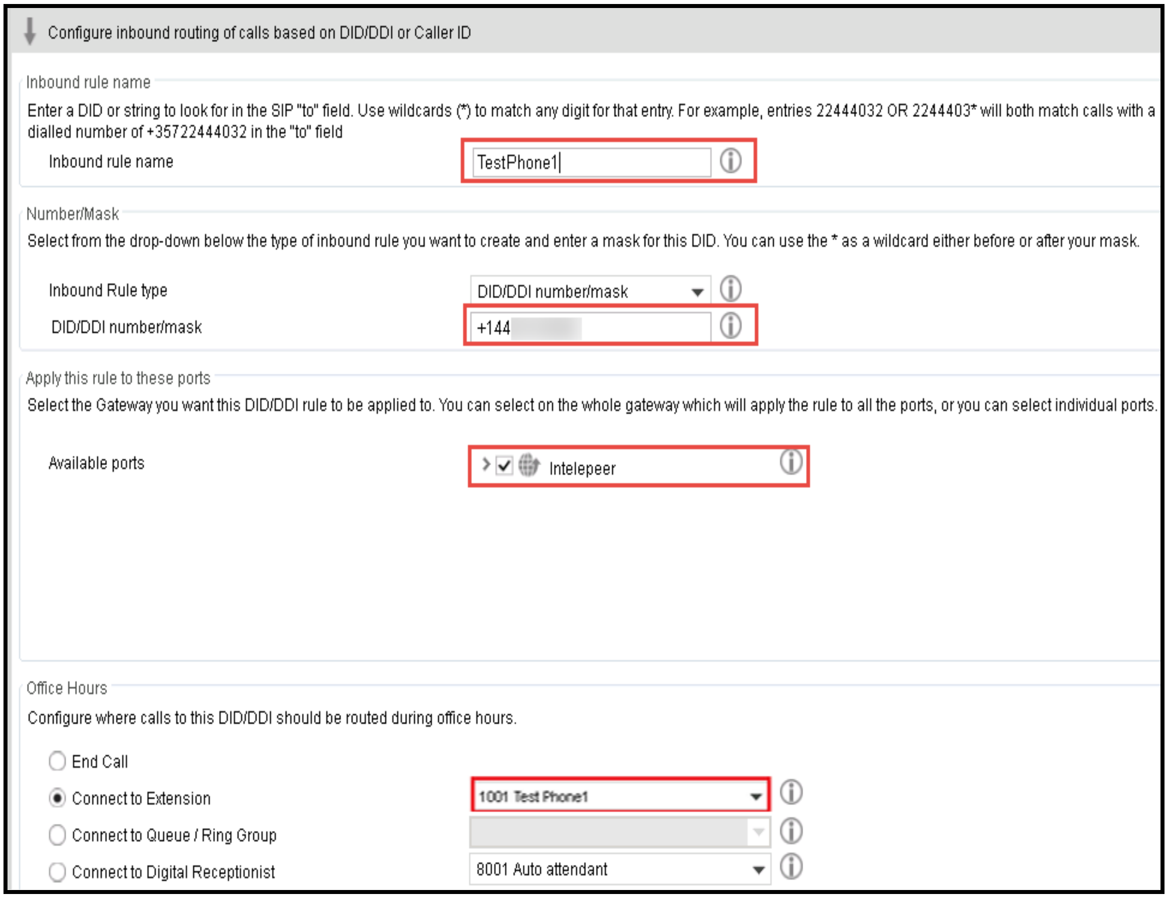

Figure 19: Inbound Rules – Cont.

5.2.5 Outbound Rule

1. From the 3CX Server Manager main menu, click **OutBound Rules**
2. Click **Add Outbound Rule**

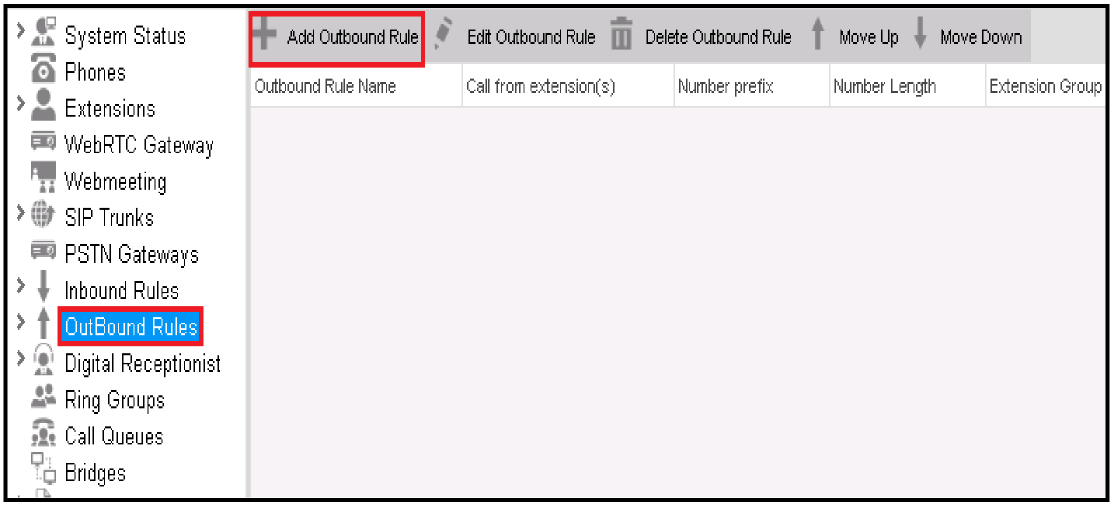

Figure 20: Outbound Rule

3. Set **Rule Name**: Rule for Intelepeer is used in this example
4. Set **Route 1:** Select **Intelepeer** using the drop down arrow in this example
5. Set **Strip Digits:** Let the default value be 0
6. Set **Prepend:** Add '+' to be added with the dialed number to represent E.164 format
7. Click **Apply** and **OK**

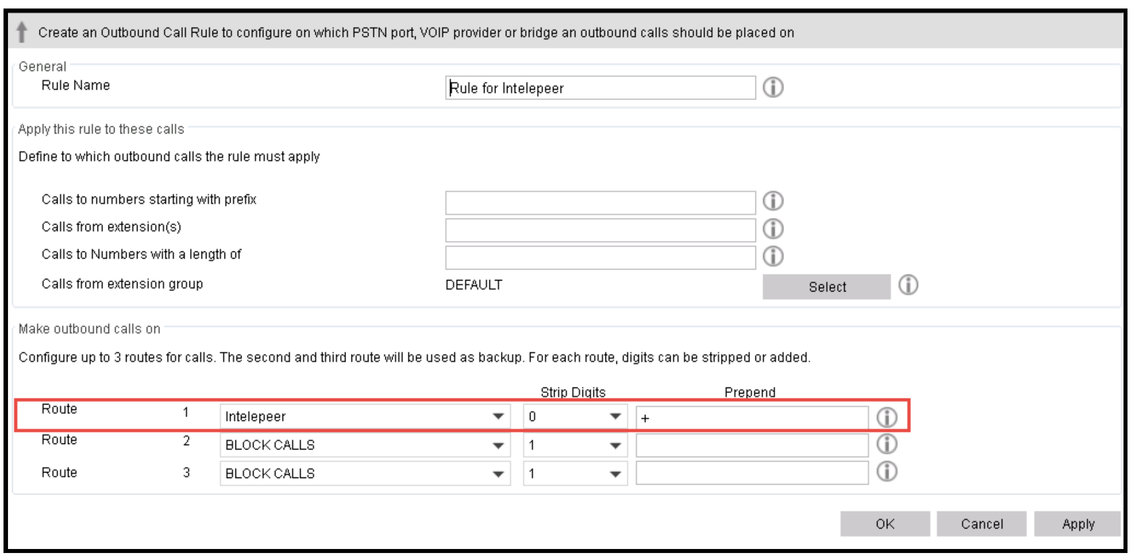

Figure 21: SIP Trunks – Cont.

8. Check **Default**
9. Click **OK**
10. Click **Apply** and **OK**

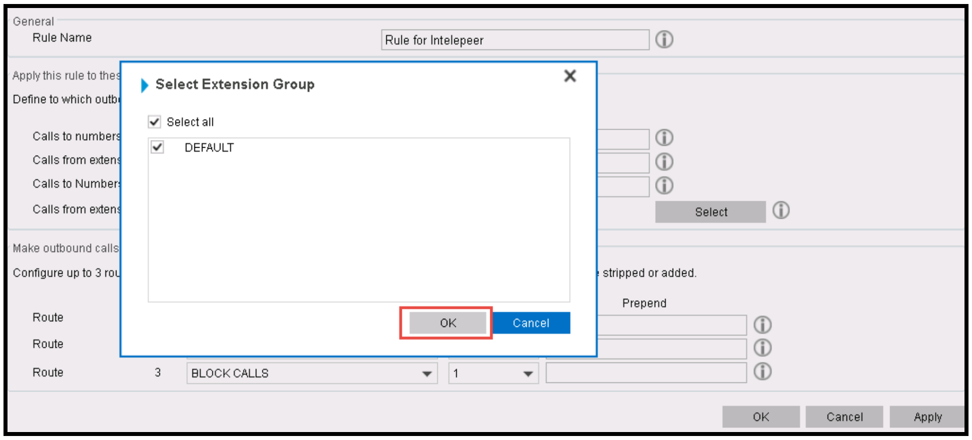

Figure 22: SIP Trunks – Cont.

5.2.6 International Dialing

1. From the 3CX Server Manager main menu tree, navigate to **Settings** > **Security**
2. Click **Allowed Country Codes** tab
3. Set **Region or Country**: Select based on site requirements. North America and Europe are selected here as an example.
4. Click **Apply** and **OK** at bottom right of screen

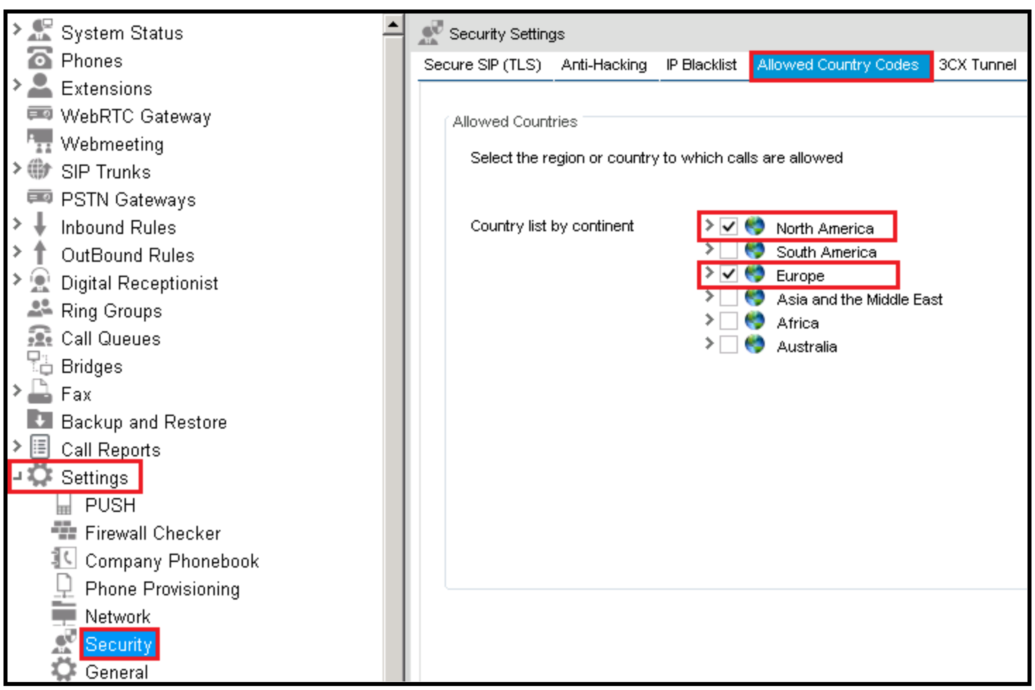

Figure 23: International Dialing

6 Summary of Tests and Results

N/S = Not Supported   N/T= Not Tested   N/A= Not Applicable

Test Case #	Test Case Description	Results	Notes
1.1	Registration	N/A	IntelePeer SIP Trunk is non-registering
2.1	Calling Party Disconnects Before Answer	Pass	
2.2	Calling Party Disconnects After Answer	Pass	
2.3	Called Party Disconnects After Answer	Pass	
2.4	Calling Party Receives RNA - Call Times Out	Pass	
2.5	Calling Party Receives Busy	Pass	
2.6	Calling Party Places Call on Hold	Pass	
2.7	Three Way Calling	Pass	
2.8	Calling Party Presentation Restricted	Pass	
3.1	Calling Party Disconnect Before Answer	Pass	
3.2	Calling Party Disconnects after Answer	Pass	
3.3	Called Party Disconnects after Answer	Pass	
3.4	Calling Party Receives Busy	Pass	
3.5	Called Party Unprovisioned Subscriber	Pass	
3.6	Calling Party Presentation Restricted	Pass	
3.7	Calling Number Presentation	Pass	
4.1	Inbound Calling Party Sends Fax Inband	Pass	
4.2	Outbound Calling Party Sends Fax Inband	Pass	
4.3	Inbound Calling Party Sends Fax using t.38	Pass	
4.4	Outbound Calling Party Sends Fax via t38	Pass	
5.1	International Outbound Dialing	Pass	
5.2	800/866/877/888 Outbound Dialing	Pass	
6.1	Outbound Call Forward Always	Pass	
6.2	Outbound Call Forward Always to an Out of Service Subscriber	Pass	
6.3	Outbound Call Forward Busy	Pass	
6.4	Outbound Call Forward Not Available (Ring No Answer)	Pass	
6.5	Outbound Blind Call Transfer	Pass	
6.6	Outbound Consultive Call Transfer	Pass	
6.7	Outbound Semi Attended Call Transfer	Fail	IntelePeer does not support INVITE with Replaces
6.8	Outbound Consultive Call Transfer to Local Extension	N/T	Test case is required only when PBX does not anchor media
6.9	Outbound Three Way Calling	Pass	
6.10	Outbound Call Hold	Pass	
6.11	Call Waiting	Pass	

7.1	Terminate Early Media Outbound Call Before Answer	Pass	
7.2	Early Media Forward Call	Pass	
8.1	Outbound, Wait for Session Audit	Pass	
8.2	Outbound, DUT Places Call on Hold, Wait for Session Audit	Pass	
8.3	Inbound, PBX Holds, Wait for Session Audit	Pass	
8.4	Inbound, Wait for Session Audit	Pass	
9.1	Outbound DTMF (RTPevent)	Pass	
9.2	Inbound DTMF(RTPevent)	Pass	
9.3	Outbound Inband (G711)	N/S	Inband DTMF Not Supported in 3CX software version 14.0
9.4	Inbound Inband (G711)	N/S	Inband DTMF Not Supported in 3CX software version 14.0
10.1	Codec Support	Pass	