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

04/21/2016

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# 1 Audience

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

## 2 Introduction

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This Configuration Guide describes configuration steps for IntelPeer 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 help 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](http://www.tekvizion.com)*

### 3 SIP Trunking Network Components

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

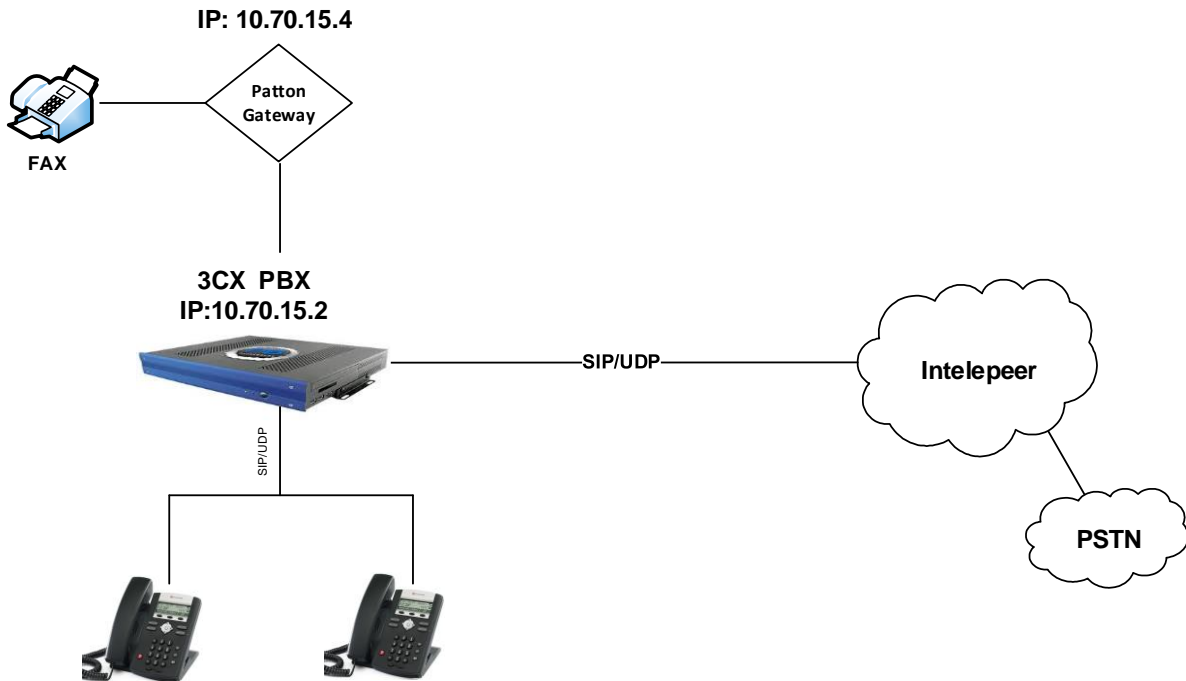


Figure 1: SIP Trunk Lab Reference Network

#### 3.1 Hardware Components

- 3CX server
- Patton Gateway

#### 3.2 Software Requirements

- 3CX Version 14.0.44241.523

## 4 Features

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### 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

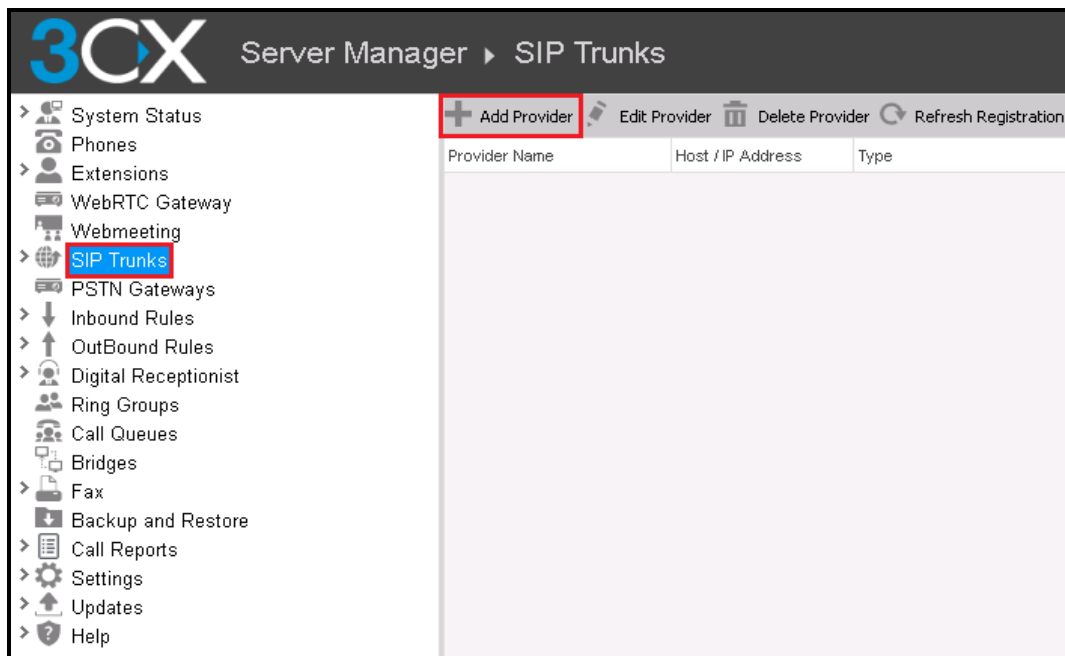
**Table 1 – PBX Configuration Steps**

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**

Add VOIP Provider Wizard

Name of Provider: Intelepeer ⓘ

Country: Generic ⓘ

Provider: Generic SIP Trunk ⓘ

URL: <http://www.3cx.com/partners/sip-trunks/>

3CX Supported VoIP Providers can be found here: <http://www.3cx.com/partners/sip-trunks/>

More 3rd party tested providers can be found here: <http://www.3cx.com/partners/voip-providers.html>

Cancel Next >

Figure 3: SIP Trunks – Cont.

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

VOIP Provider Details:

Enter the hostname and port for your VOIP Provider's SIP Server

SIP server hostname or IP: 66. ⓘ

SIP Server port: 5 ⓘ

Outbound proxy hostname or IP: ⓘ

Outbound proxy port (default is 5060): 5060 ⓘ

< Back Next >

Figure 4: SIP Trunks – Cont.

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

Account Details

Enter the Authentication ID or SIP User, Password and number of your account

External Number: 4 ⓘ

Authentication ID (aka SIP User ID): ⓘ

Authentication Password: ⓘ

3Way Authentication ID:  ⓘ

Simultaneous Calls

Maximum simultaneous calls: ⓘ

< Back Next >

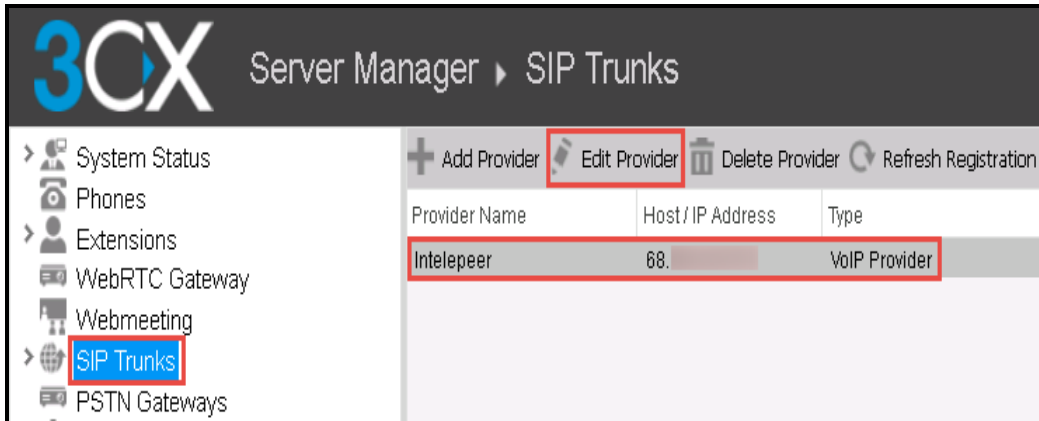
Figure 5: SIP Trunks – Cont.

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 IntelPeer's IP address
6. Set **SIP Server Port**: Enter IntelPeer's listening port number
7. Set **Outbound Proxy Hostname or IP**: Enter IntelPeer's IP address
8. Set **Outbound Proxy Port (default is 5060)**: 5060 is used in this example

The screenshot shows the 'General' tab of a configuration interface. Under the 'Provider Details' section, there are four input fields:

- SIP server hostname or IP: 68.
- SIP server port: 5060
- Outbound proxy hostname or IP: 68.
- Outbound proxy port (default is 5060): 5060

**Figure 8: Edit SIP Trunks – Cont.**

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

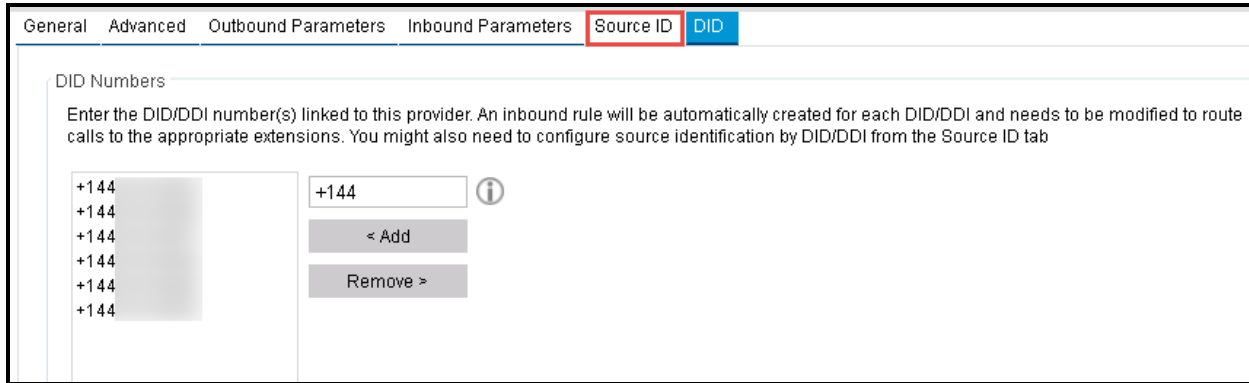
The screenshot shows the 'Advanced' tab of the configuration interface. It is divided into three main sections:

- Provider Capabilities:** Includes checkboxes for 'Supports Re-Invite' (checked), 'Supports Replace' (unchecked), 'PBX Delivers Audio' (checked), 'Switch on Secure RTP (SRTP)' (unchecked), and 'Disable Video' (checked). There is also a text input for 'Put Public IP in SIP VIA Header'.
- Registration Settings:** Includes a dropdown for 'Require registration for:' set to 'Do not require', and radio buttons for 'Which IP to use in Contact and Connection SIP fields' (selected: 'Use Default Settings').
- Codec priorities:** A table showing available codecs and assigned codecs.
 

Available Codecs	Assigned Codecs
GSM-FR	G.711 A-law
Speex	G.711 U-law
iLBC	G729
	G722

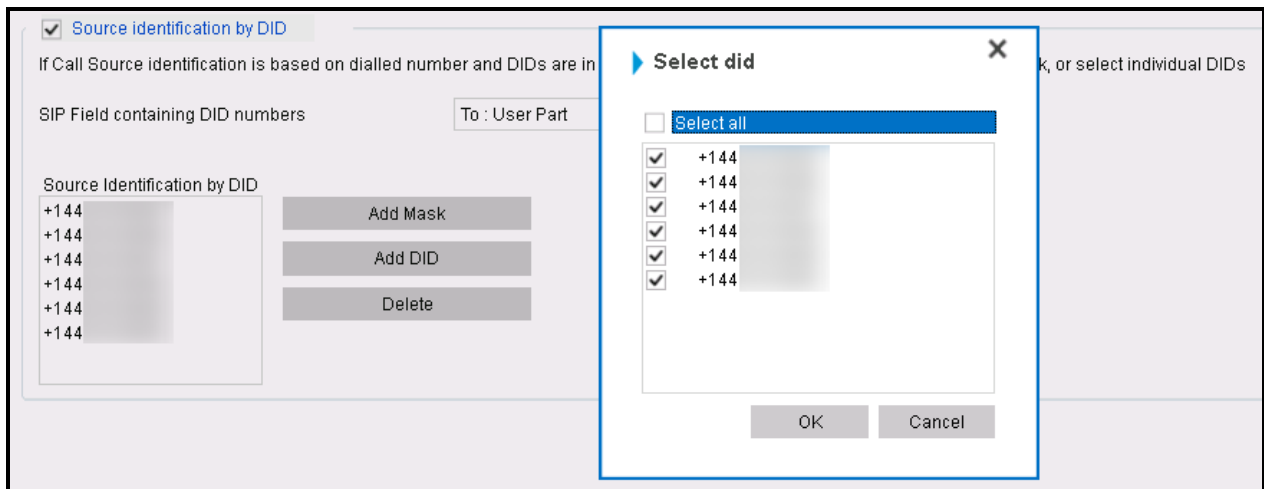
**Figure 9: Edit SIP Trunks – Cont.**

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**

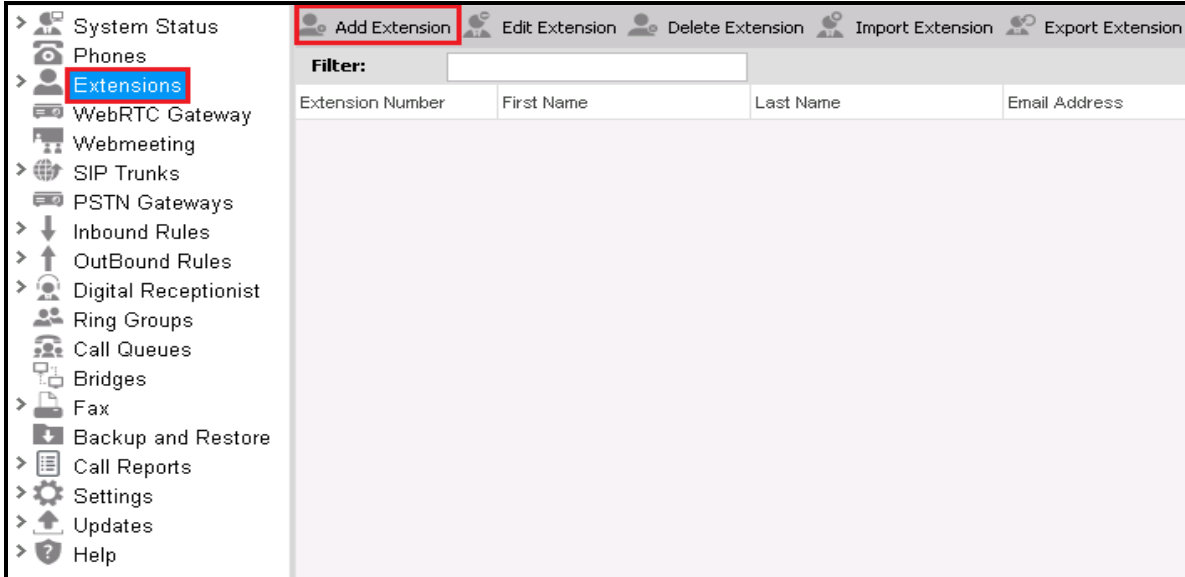


Figure 12: Extension Setup

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

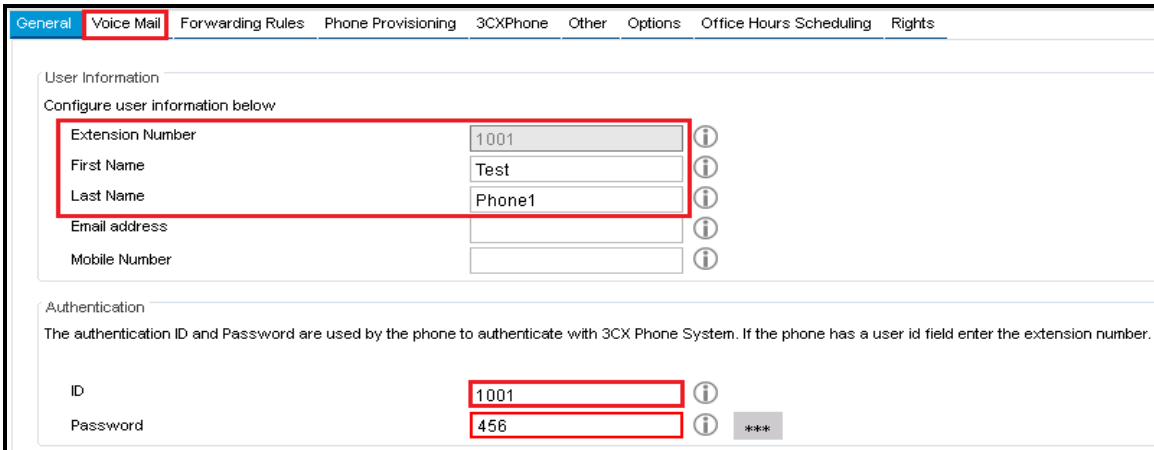


Figure 13: Extension Setup – Cont.

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

General **Voice Mail** Forwarding Rules Phone Provisioning 3CXPhone Other Options Office Hours Scheduling Rights

Voice Mail Configuration

If you are unable to answer a call, you can allow voice messages to be taken

Enable Voice mail  ⓘ

Disable Voicemail PIN Authentication  ⓘ

Play Caller ID  ⓘ

PIN Number  ⓘ \*\*\*

Read out date/time of message  ⓘ

Email Options  ⓘ

**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

General Voice Mail **Forwarding Rules** Phone Provisioning 3CXPhone Other Options Office Hours Scheduling

**Available** Away Do Not Disturb Available 2 Out of Office 2 Exceptions

Configure how calls should be re-directed when a user can not answer the phone or the phone is busy.

No Answer

If the call is not answered within  seconds, then: ⓘ

Send call to my voice mail

Send call to my mobile number

Send call to  ⓘ

An external number or Skype ID  ⓘ

Rebound™ (Offer option to Confirm to accept)

Disconnect the call

Different behaviour for internal calls

Phone is Busy

If my phone is busy or unregistered, then:

Send call to my voice mail

Send call to my mobile number

Send call to  ⓘ

An external number or Skype ID  ⓘ

Rebound™ (Offer option to Confirm to accept)

Disconnect the call

Different behaviour for internal calls

Ring my extension and my mobile at the same time

I want to be able to accept more than 1 call at the same time - uses Phone Status

**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**

General Voice Mail Forwarding Rules Phone Provisioning 3CXPhone **Other** Options Office Hours Scheduling

polycom.ph.xml New phone device...

Provisioning

Provisioning ensures the phone settings are centrally retrieved, this limits the amount of time spent and information needed

MAC Address	0004F2B0D784	(i)
Model	Polycom SPIP335 (Firmware 4)	(i)
Phone Web Page Password	456	(i)
Phone Display Language	English (en-us)	(i)
Time Zone	Use global PBX settings	(i)
Select Provisioning Method	Local Lan (In the Office)	(i)
Select Interface	10.70.15.2	(i)

Codec Priority

Configure the priority of the codecs in this phone

Preferred Codec	PCMU	(i)
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**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

General Voice Mail Forwarding Rules Phone Provisioning 3CXPhone **Other** Options Office Hours Scheduling

User Information

Configure user status and options

Current status	Available	(i)
Queues Status	Logged In	(i)
DND	OFF	(i)
Outbound Caller ID	+144:	(i)
SIP ID		(i)

**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**

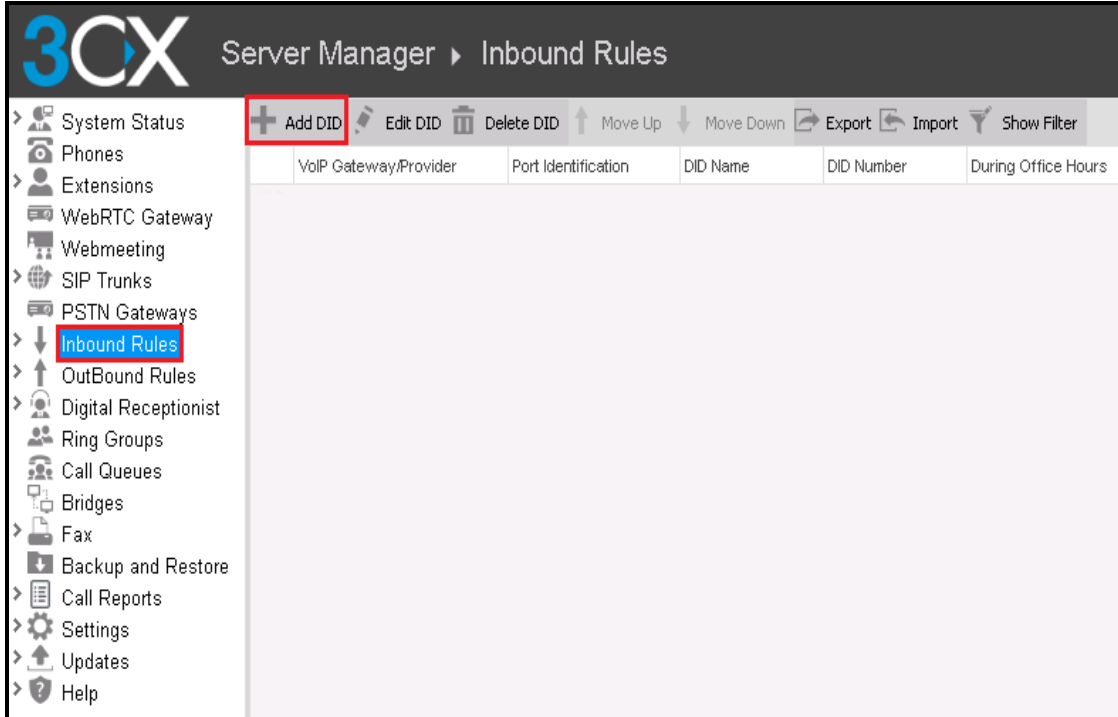


Figure 18: Inbound Rules

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

↓ Configure inbound routing of calls based on DID/DDI or Caller ID

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**Inbound rule name**  
 Enter a DID or string to look for in the SIP "to" field. Use wildcards (\*) to match any digit for that entry. For example, entries 22444032 OR 2244403\* will both match calls with a dialled number of +35722444032 in the "to" field

Inbound rule name  ⓘ

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**Number/Mask**  
 Select from the drop-down below the type of inbound rule you want to create and enter a mask for this DID. You can use the \* as a wildcard either before or after your mask.

Inbound Rule type DID/DDI number/mask ⓘ

DID/DDI number/mask  ⓘ

---

**Apply this rule to these ports**  
 Select the Gateway you want this DID/DDI rule to be applied to. You can select on the whole gateway which will apply the rule to all the ports, or you can select individual ports.

Available ports

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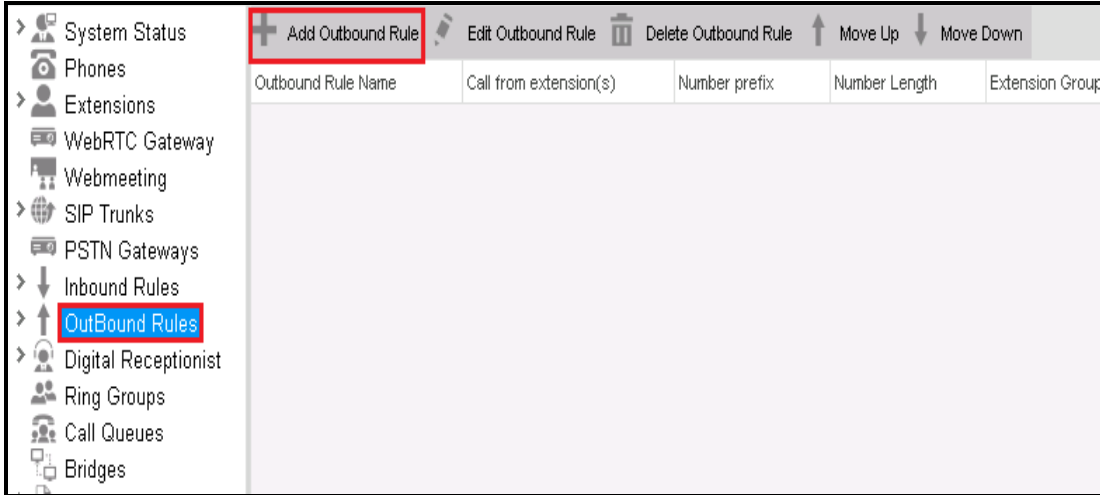
**Office Hours**  
 Configure where calls to this DID/DDI should be routed during office hours.

End Call  
 Connect to Extension  ⓘ  
 Connect to Queue / Ring Group  ⓘ  
 Connect to Digital Receptionist  ⓘ

**Figure 19: Inbound Rules – Cont.**

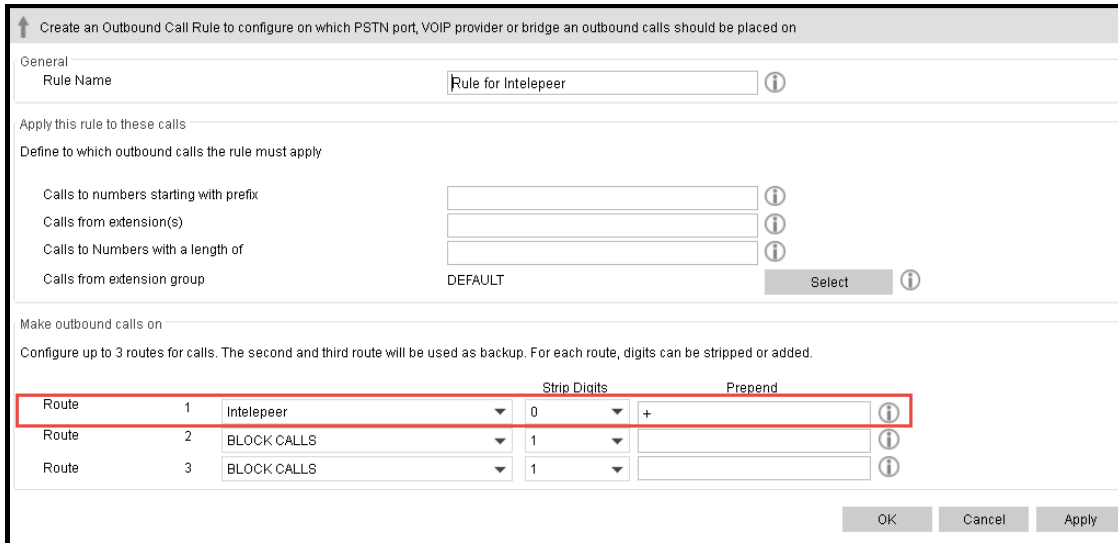
### 5.2.5 Outbound Rule

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



**Figure 20: Outbound Rule**

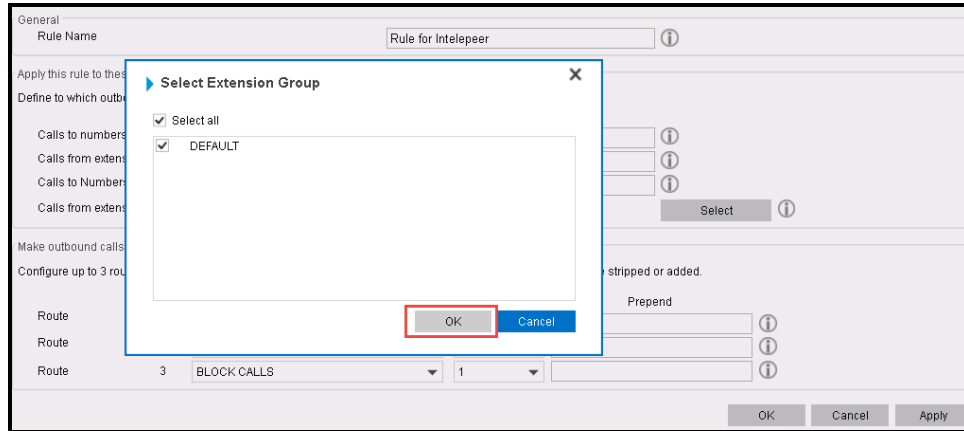
3. **Set Rule Name:** Rule for Intelepeer is used in this example
4. **Set Route 1:** Select **Intelepeer** using the drop down arrow in this example
5. **Set Strip Digits:** Let the default value be 0
6. **Set Prepend:** Add '+' to be added with the dialed number to represent E.164 format
7. Click **Apply**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



**Figure 21: SIP Trunks – Cont.**



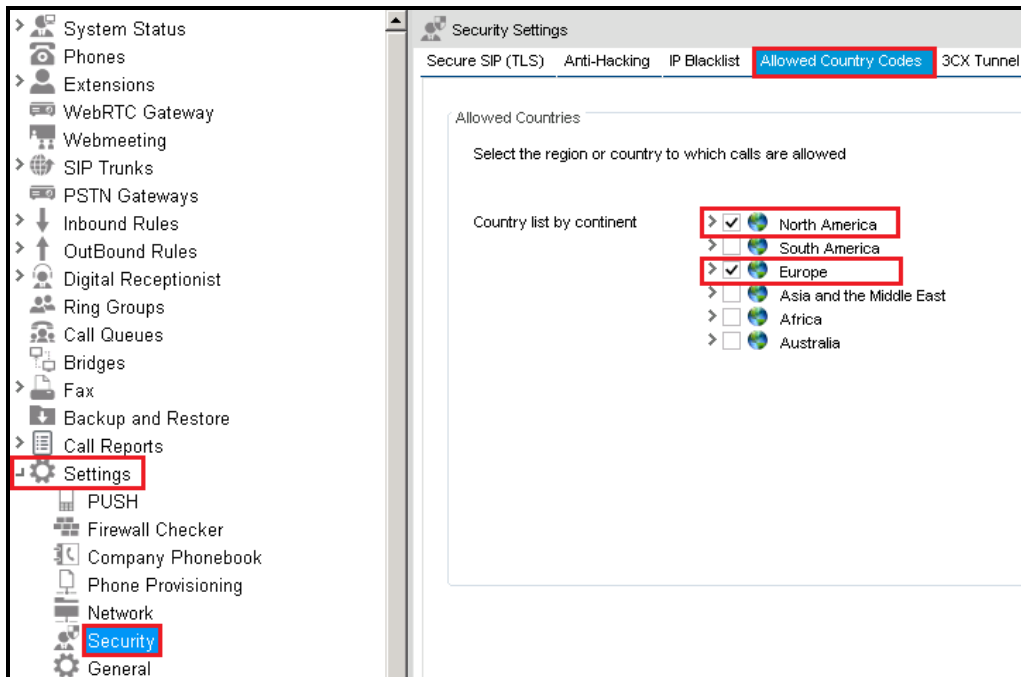
8. Check **Default**
9. Click **OK**
10. Click **Apply** and **OK**



**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	