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SIP Trunking Configuration Guide
for
Toshiba CIX200
Version AR5.20 MT076.00

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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 IntelePeer SIP Trunking to a Toshiba Strata CIX 200 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 Labs 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 Toshiba Strata CIX PBX configuration.

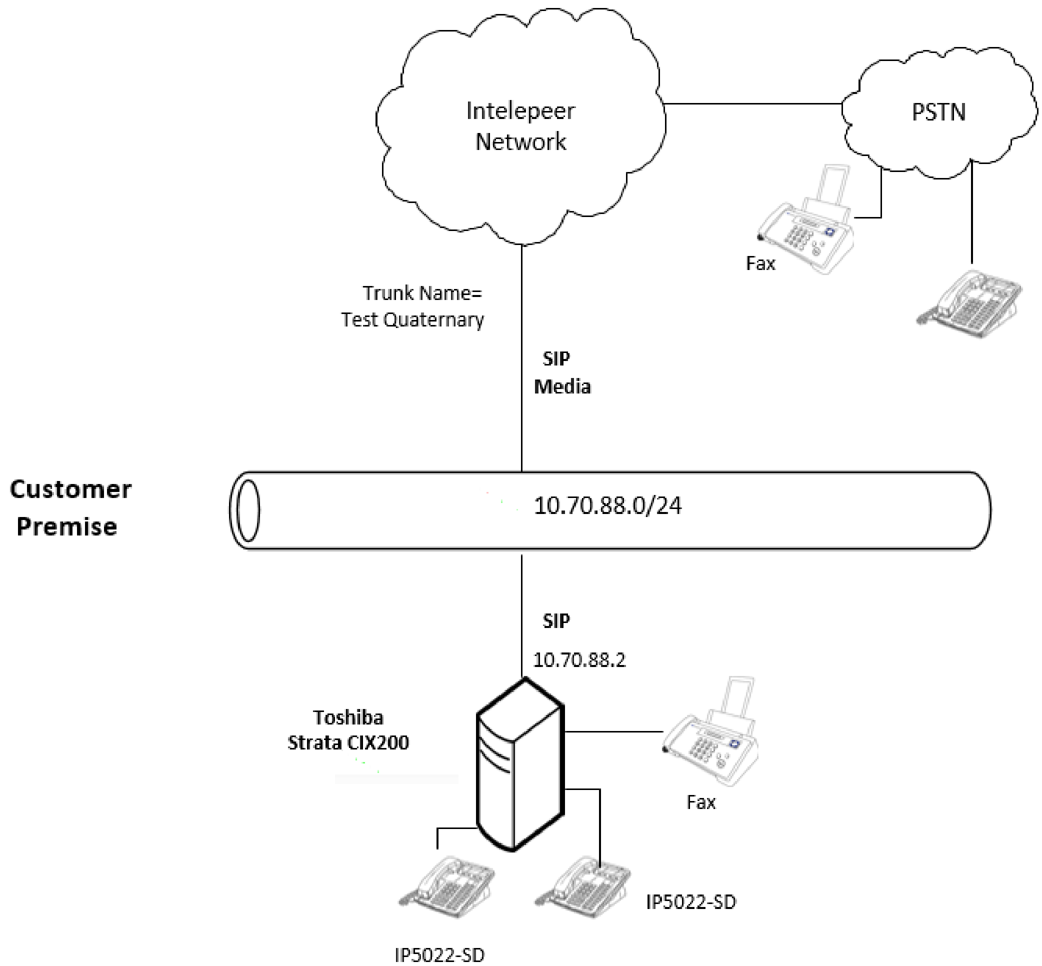

Figure 1: SIP Trunk Lab Reference Network

The lab network consists of the following components:

- Toshiba CIX PBX for voice features, SIP proxy and SIP trunk termination
- Toshiba IP phones connected to the local LAN
- Fax connected to the PBX and PSTN

3.1 Hardware Components

- Toshiba Strata CIX 200
- Toshiba IP phones

3.2 Software Requirements

- Toshiba Strata CIX Firmware AR5.20 MT076.00

4 Features

4.1 Features Supported

- Basic calls using G.711ulaw
- Calling and Called Party Number Presentation
- Call Transfer
- Call Forwarding
- Call Hold and Resume
- Call Waiting
- Conference call
- Three-Way Calling
- Fax calls using G711ulaw
- Inband DTMF
- RFC2833
- Session Audit using Re-INVITE messages

4.2 Features Not Supported

- Privacy (Calling Party restriction)
- Fax Calls using T38

4.3 Caveats and Limitations

- The PBX SIP trunk configuration for primary and secondary codec fields have to be set the same codec for outgoing calls to successfully complete. Refer to the SIP trunk configuration.
- Intermittent speech path with inbound calls to the Toshiba CIX PBX was observed throughout the testing. When the issue occurs, there is no speech path present at either endpoint.
  The issue was seldom seen during this test
- The Toshiba CIX 200 PBX only supports g711ulaw codec for fax tests
- Refer to Section 6 Summary of Tests and Results

5 Configuration

5.1 Configuration Checklist

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

Table 1 – PBX Configuration Steps

Step	Description	Reference
Step 1	IP Address Worksheet	Section 5.2
Step 2	Initial System Setup	Section 5.3
Step 3	SIP Trunk Setup	Section 5.4
Step 4	Create Trunk Line Groups	Section 5.5
Step 5	Create Trunk DIDs	Section 5.6

5.2 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 2 – IP Addresses

Component	Lab Value	Customer Value
IntelePeer SIP Trunk		
• Signaling IP Address	68.68.123.XXX	
• Media IP Address	68.68.123.XXX	
• Protocol/Port	UDP/5060	
Toshiba Strata CIX PBX		
• System LAN 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.	10.70.88.2	
• LAN 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	10.70.88.1	

5.3 Initial System Setup

The initial setup involves setting up the IP configuration

1. Navigate to **Maintenance > CIX Processor IP Address**
2. Set **IP Address:** Enter the IP address of the LAN interface
3. Set **Subnet Mask**: Enter the Subnet mask for the LAN interface
4. Set **Default Gateway:** Enter the default gateway address for the LAN interface
5. Click **Submit**

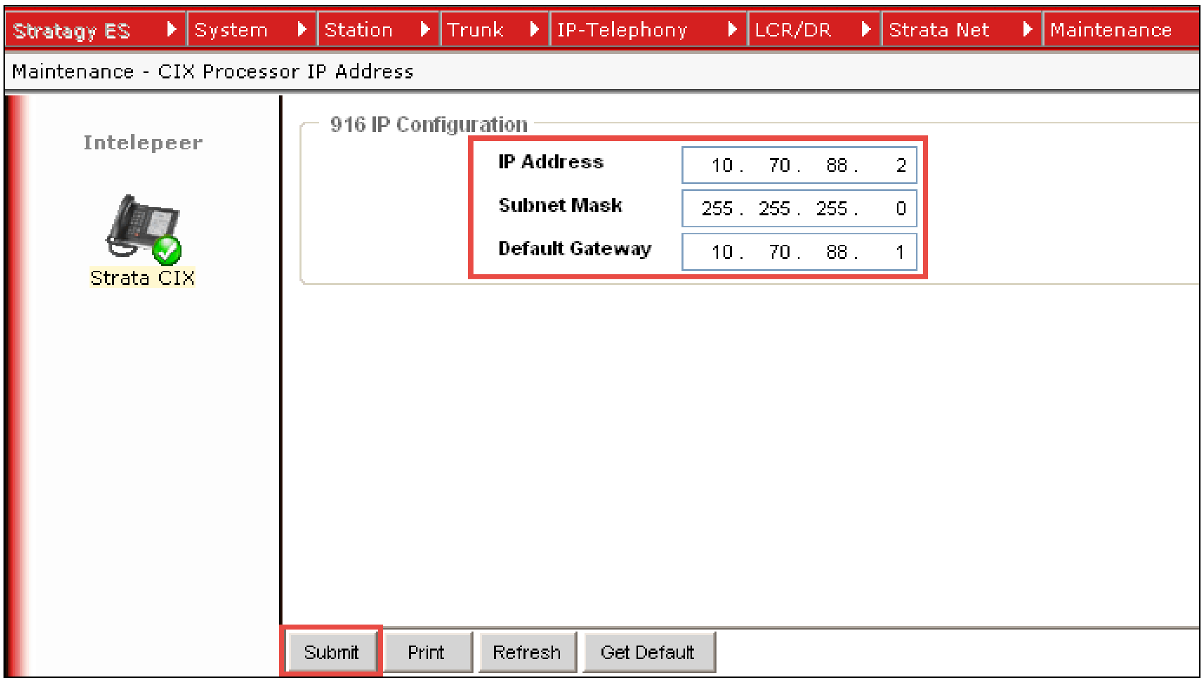

Figure 2: IP Configuration

6. Navigate to **System > Card Assignment**
7. In this example, both the **Cabinet** and **Slot** have a value of “01”

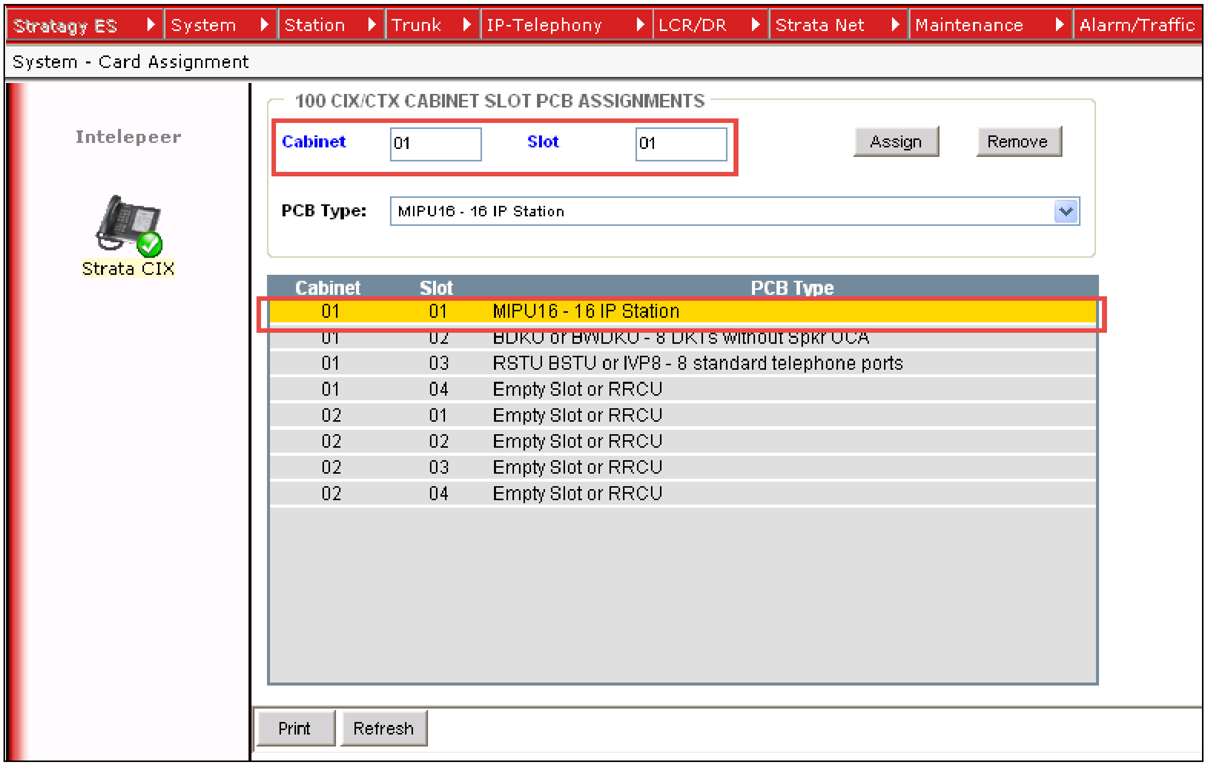

Figure 3: Card Assignment

8. Navigate to **IP-Telephony > L/M/G IPU Configuration**
9. **00 Cabinet and Slot Number:** Enter “0101”
10. **01 IPU IP Address**: Enter another IP address that is in same subnet as the LAN IP address
11. **02 IPU Subnet Address:** Enter same subnet mask as the LAN IP address
12. **03 IPU Default Gateway Address:** Enter the same gateway IP address as the LAN IP Address
13. Click **Submit**

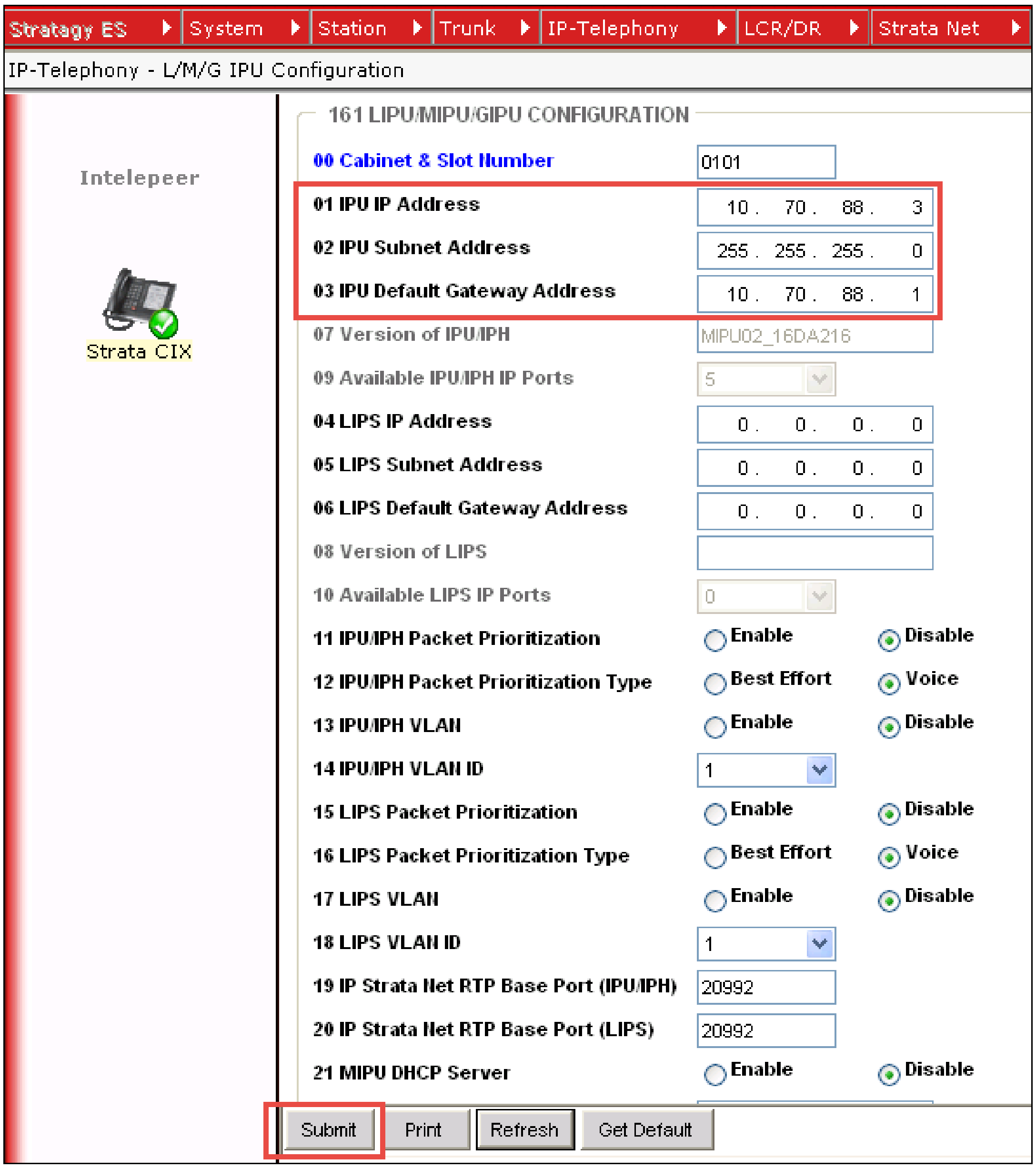

Figure 4: L/M/G IPU Configuration

5.4 SIP Trunk Setup

1. Navigate to **IP-Telephony > SIP Trunking**
2. Click **Create**
3. **Index Value:** 1 is used in this example
4. Click **OK**

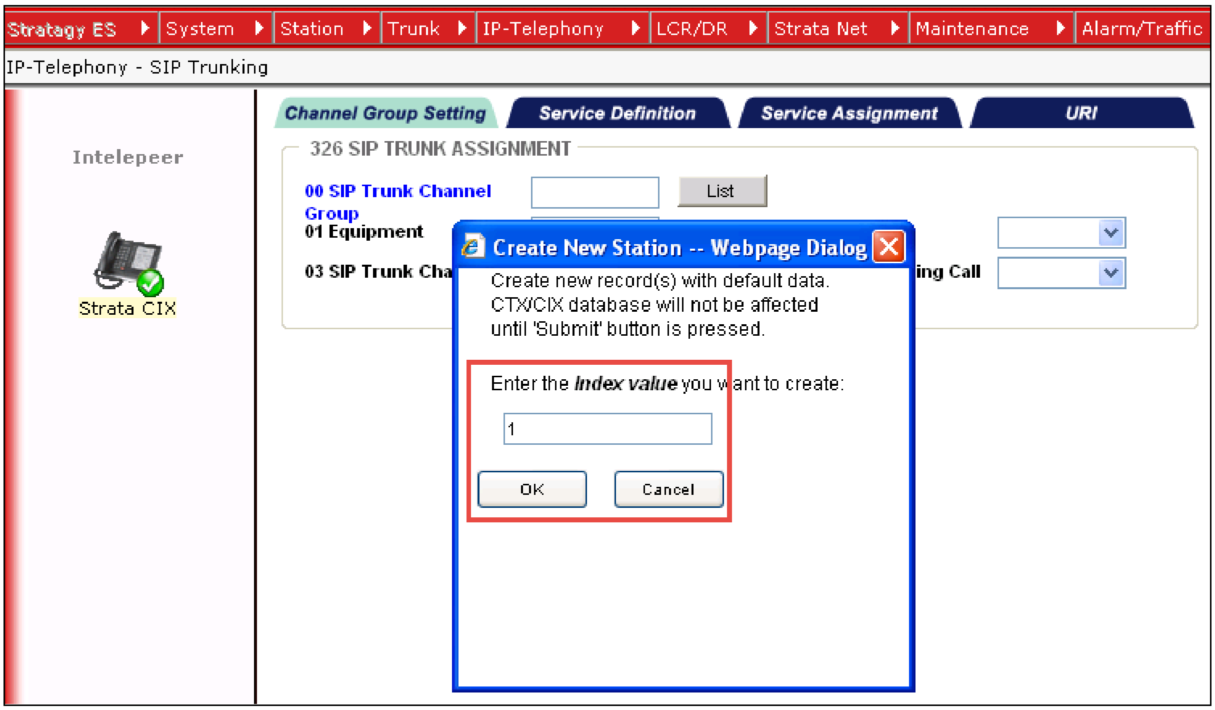

Figure 5: SIP Trunk Setup

5. **01 Equipment**: 0101 is used in this example
6. **02 LAN Interface:** 1 is used in this example
7. **03 SIP Trunk Channels**: 5 is used in this example
8. **04 RBT on Incoming Call**. Enable is selected in this example
9. Click **Submit**

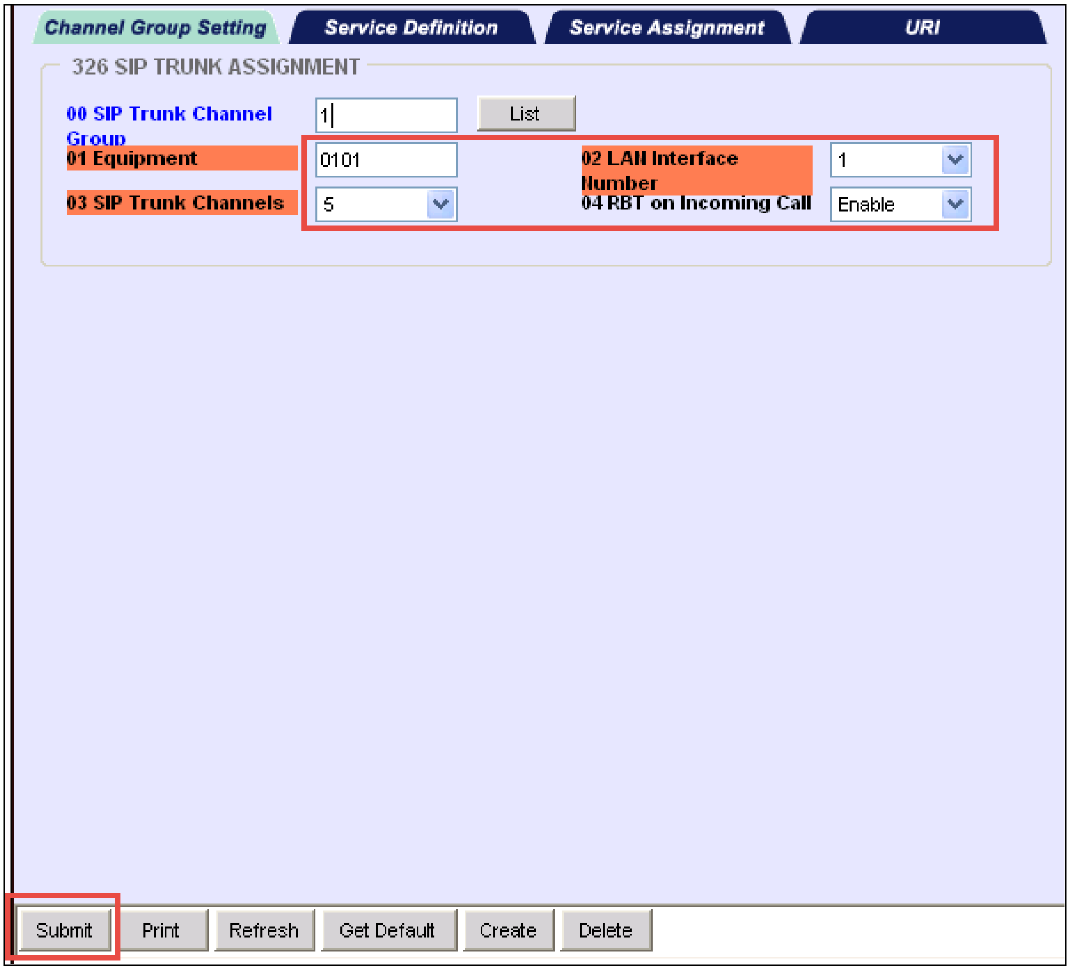

Figure 6: Channel Group Setting

10. Select the **Service Definition** tab
11. Click **Create**
12. **Index Value**: 11 was used in this example
13. Click **OK**

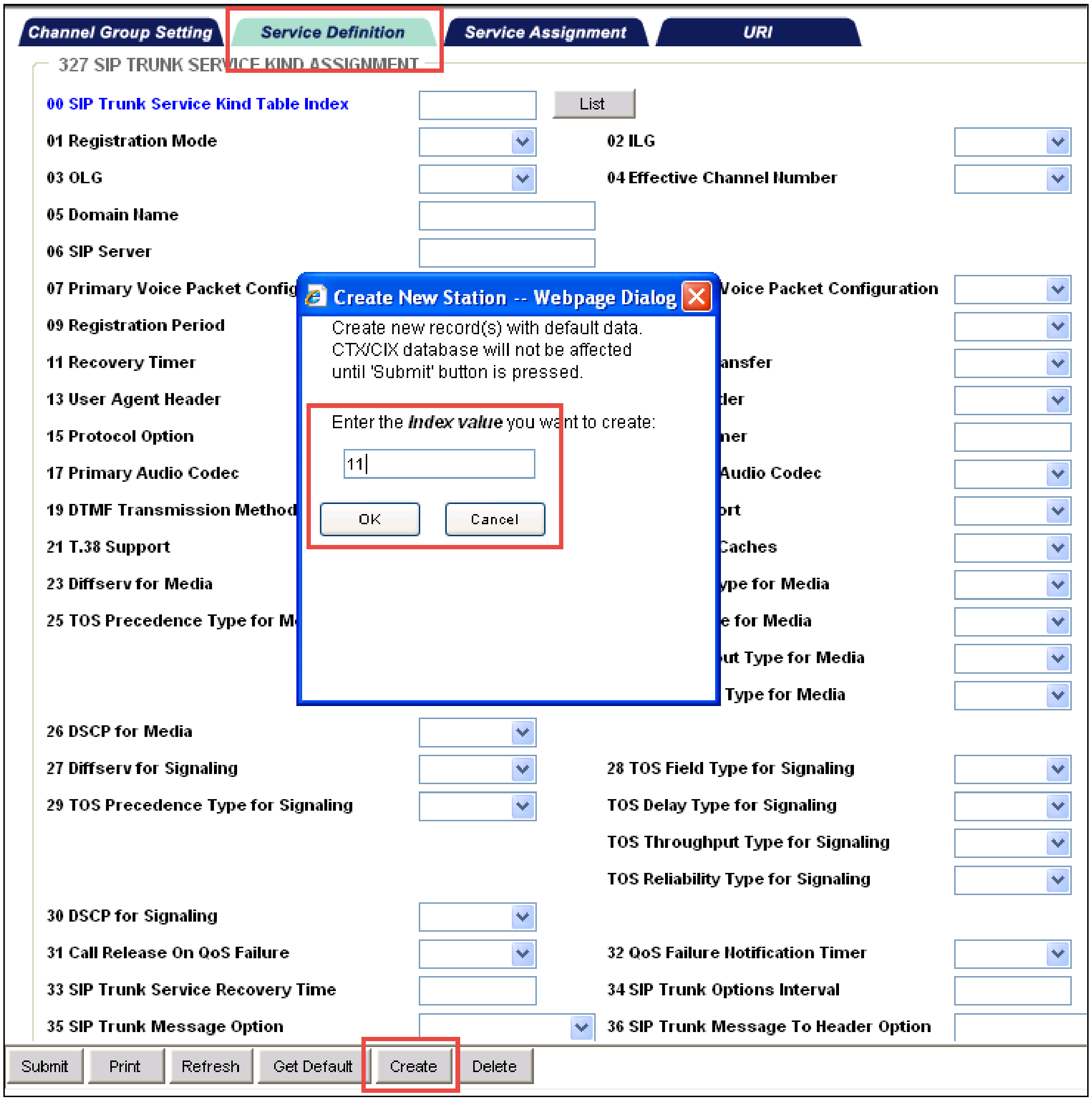

Figure 7: Create Index Value

14. **01 Registration Mode**: Select *None*
15. **02 ILG**: Select 2
16. **03 OLG**: Select = 2
17. **04 Effective Channel Number**: Select 3
18. **05 Domain Name**: Enter Service Provider trunk IP address
19. **06 SIP Server**: Enter Service Provider trunk IP address
20. **18 Secondary Audio Codec:** Select G.711u
    Note: *This codec must match the setting in **17 Primary Audio Codec** for outgoing calls*
21. The default settings were used for the remaining fields
22. Click **Submit**

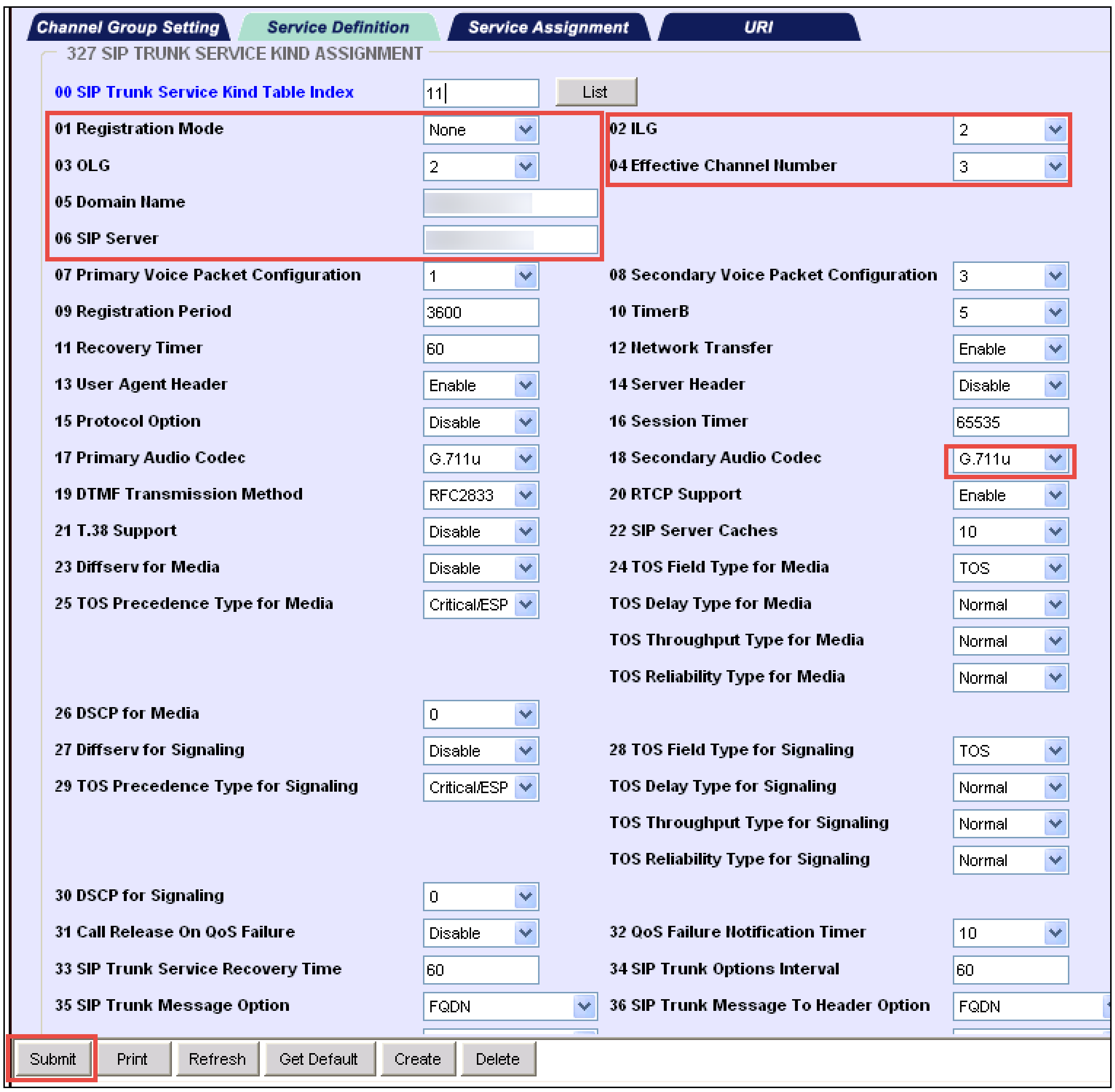

Figure 8: Service Definition

23. Select the **Service Assignment** tab
24. **00 Channel Group:** Select 1
25. **02 Service Index:** Select 11
26. Click **Set** to update the service index table

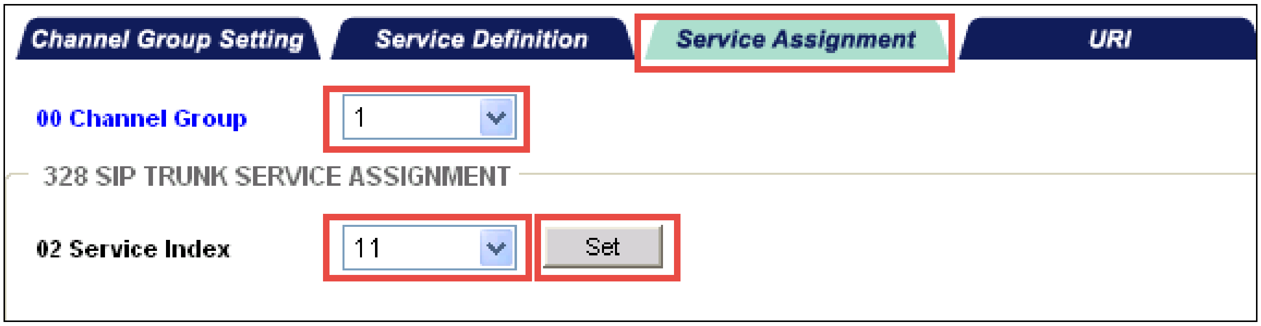

Figure 9: Service Assignment

27. Select the **URI** tab
28. **00 SIP URI Trunk Service:** Select 11
29. **01 SIP URI Index**: Select 1
30. **02 SIP URI**: Enter the DID provided by Service Provider
31. **03 SIP URI User Name**: Enter the User Name
32. Click **Add** to add entry to index
33. Repeat steps for additional DIDs

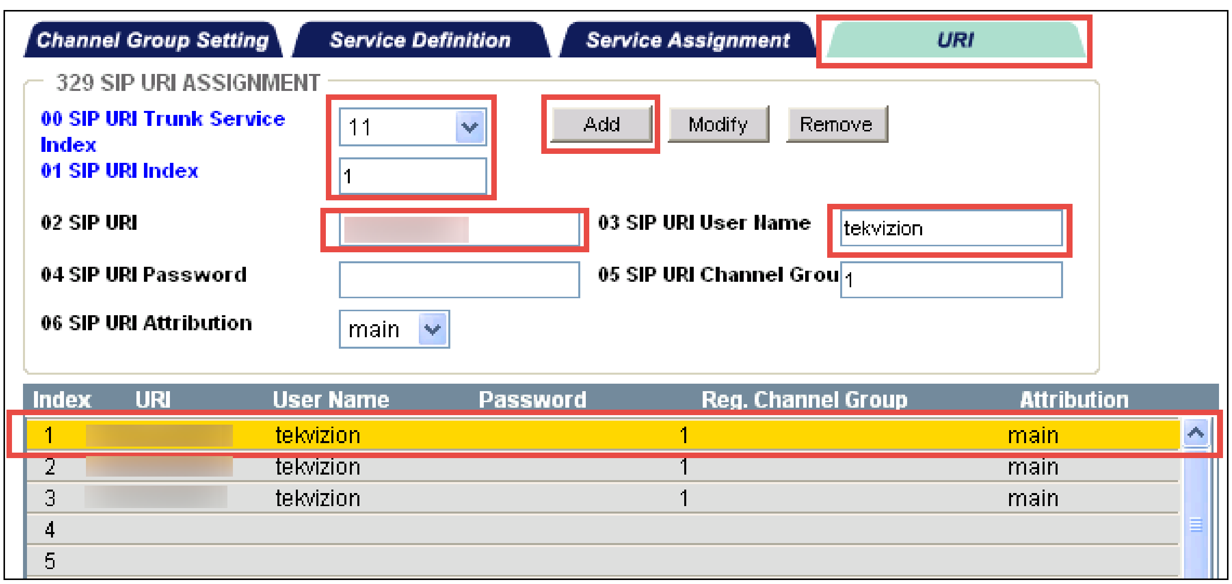

Figure 10: URI Settings

5.5 Create Trunk Line Groups

34. Navigate to **Trunk > ILG**
35. Click **Create**
36. Enter the **Index Value** for the Incoming Group Number being created

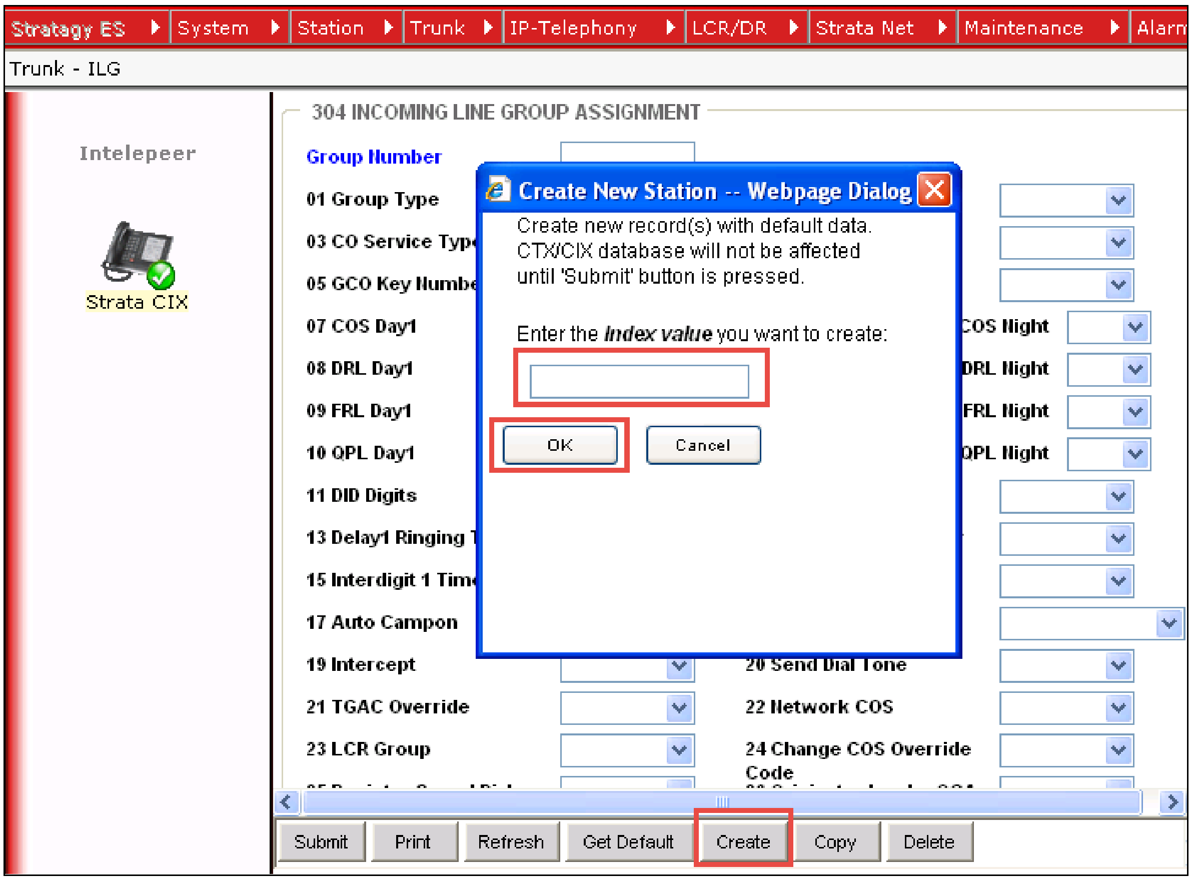

Figure 11: Incoming Line Group Assignment – Cont.

Note: **Group Number** *2* is shown in the ILG example

37. **01 Group Type**: Select *SIP*
38. **02 Line Type:** Select *CO*
39. **03 CO Service Type**: Select *DID*
40. **11 DID Digits**: Select 4
41. The remaining default values were used
42. Click **Submit**

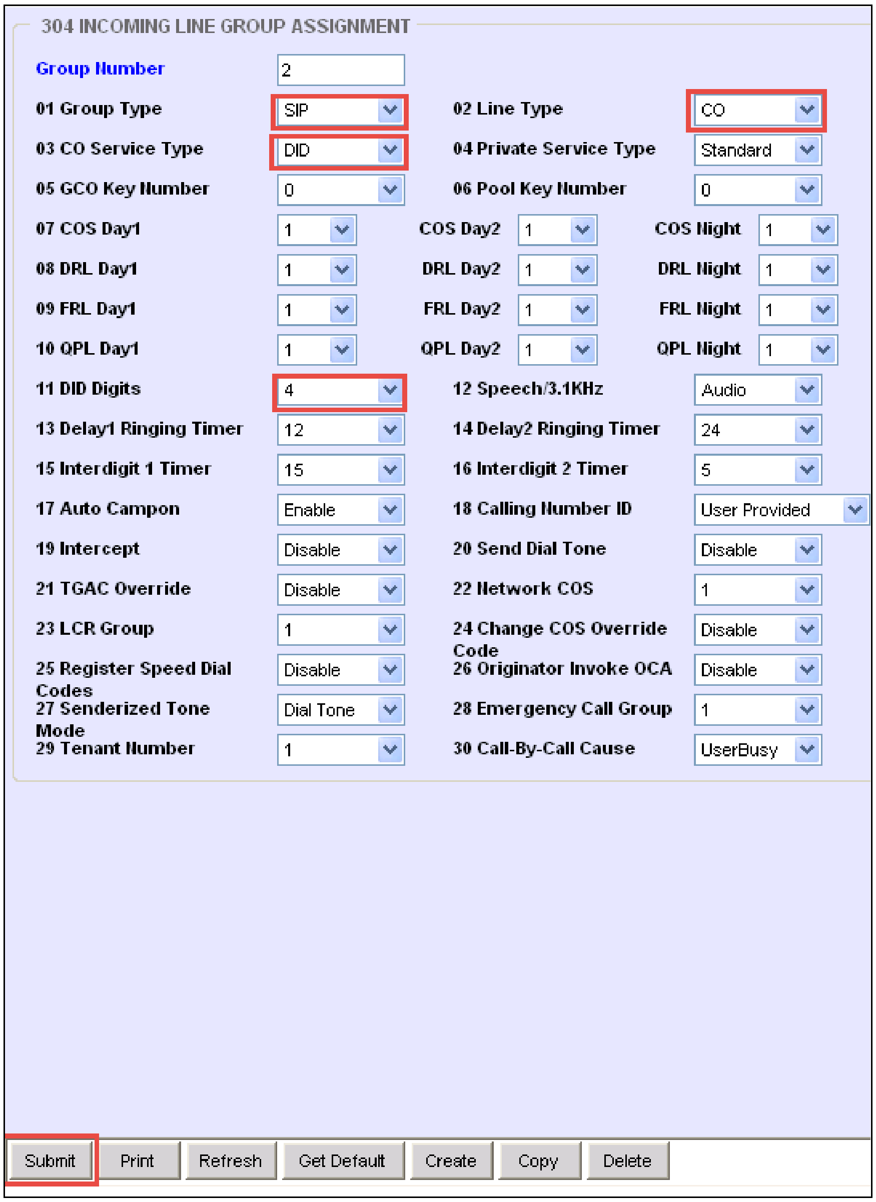

Figure 12: Incoming Line Group Assignment

43. Navigate to **Trunk > OLG**
44. Click **Create**
45. Enter the **Index Value** for the Outgoing Group Number being created

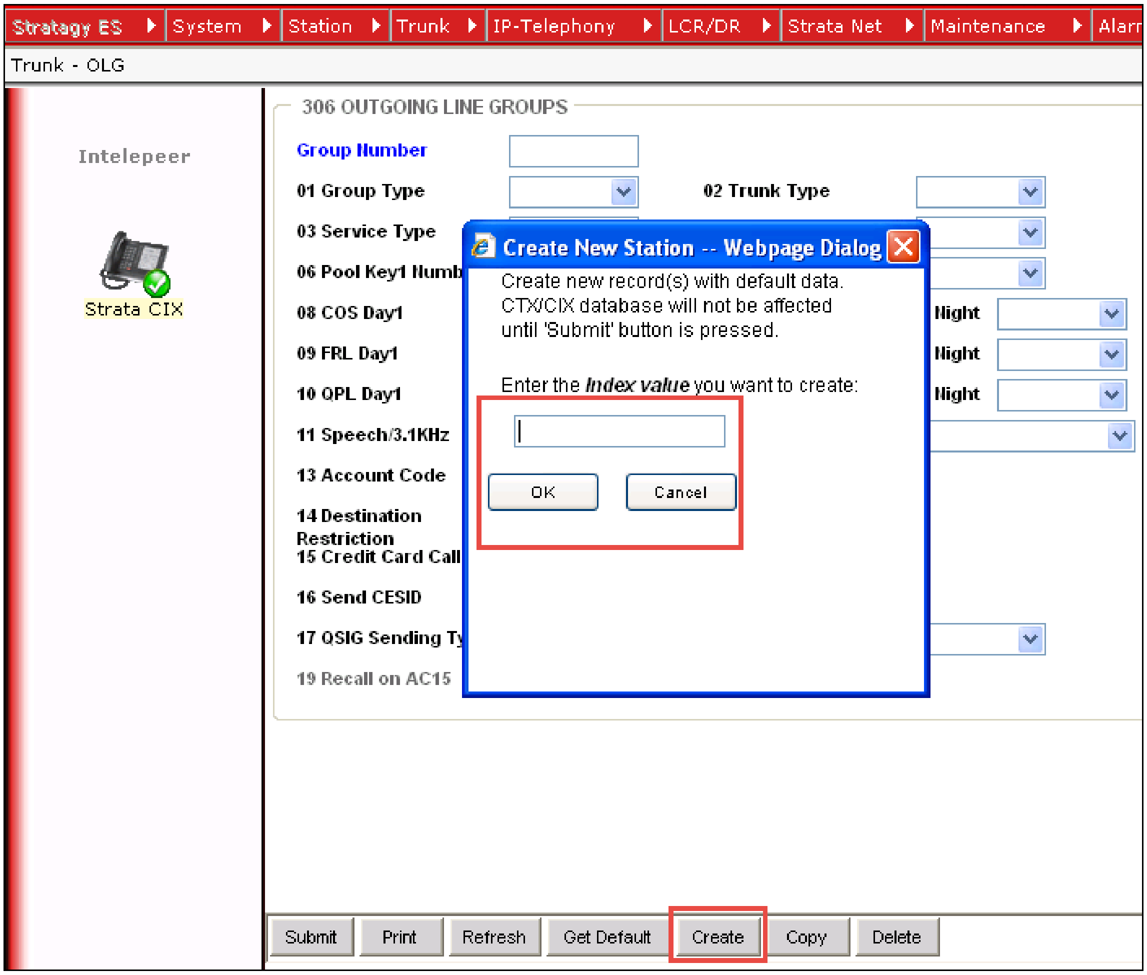

Figure 13: Outgoing Line Groups

Note: The **Group Number** *2* is shown in the OLG example

46. **01 Group Type**: Select *SIP*
47. The remaining default values were used
48. Click **Submit**

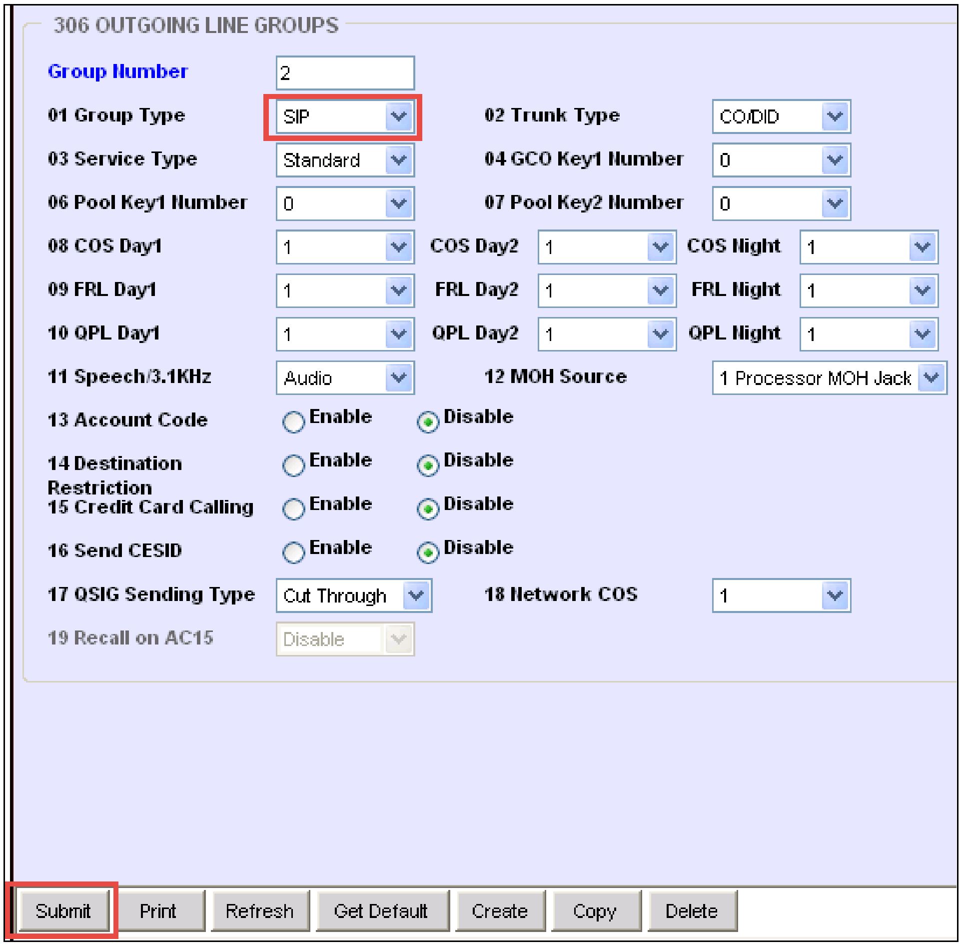

Figure 14: Outgoing Line Groups – Cont.

5.6 Create Trunk DIDs

1. Navigate to **Trunk > DID**
2. Enter the **ILG Group Number.** In the example, 2 is shown.
3. Click **Create**
4. Enter the **Index Value** or last 4-digits of the DID (provided by Service Provider). In the example, 4975 is shown.
5. Click **OK**

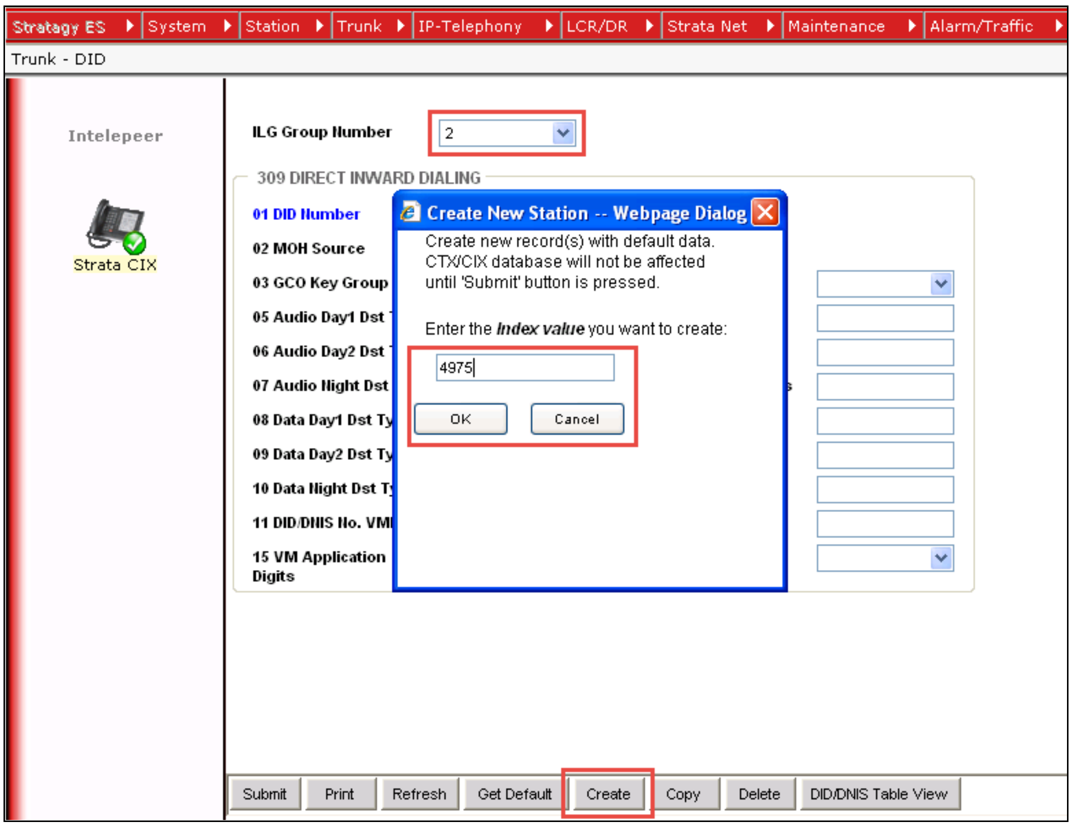

Figure 15: Create Trunk DIDs

6. **05 Audio Day1 Dst Type**: Select *Dialing Digits*
7. **06 Audio Day2 Dst Type**: Select *Dialing Digits*
8. **07 Audio Night Dst Type**: Select *Dialing Digits*
9. **Audio Day1 Dst Digits**: Enter a 3-digit extension in the field.
   (In this example, 216 is an IP-phone extension).
10. **Audio Day2 Dst Digits**: Enter a 3-digit extension in the field.
    Audio Night Dst Digits: Enter a 3-digit extension in the field.
    (In these examples, 216 is an IP-phone extension)
11. Select the default values for the other fields
12. Click **Submit**
13. Repeat steps for additional DIDs

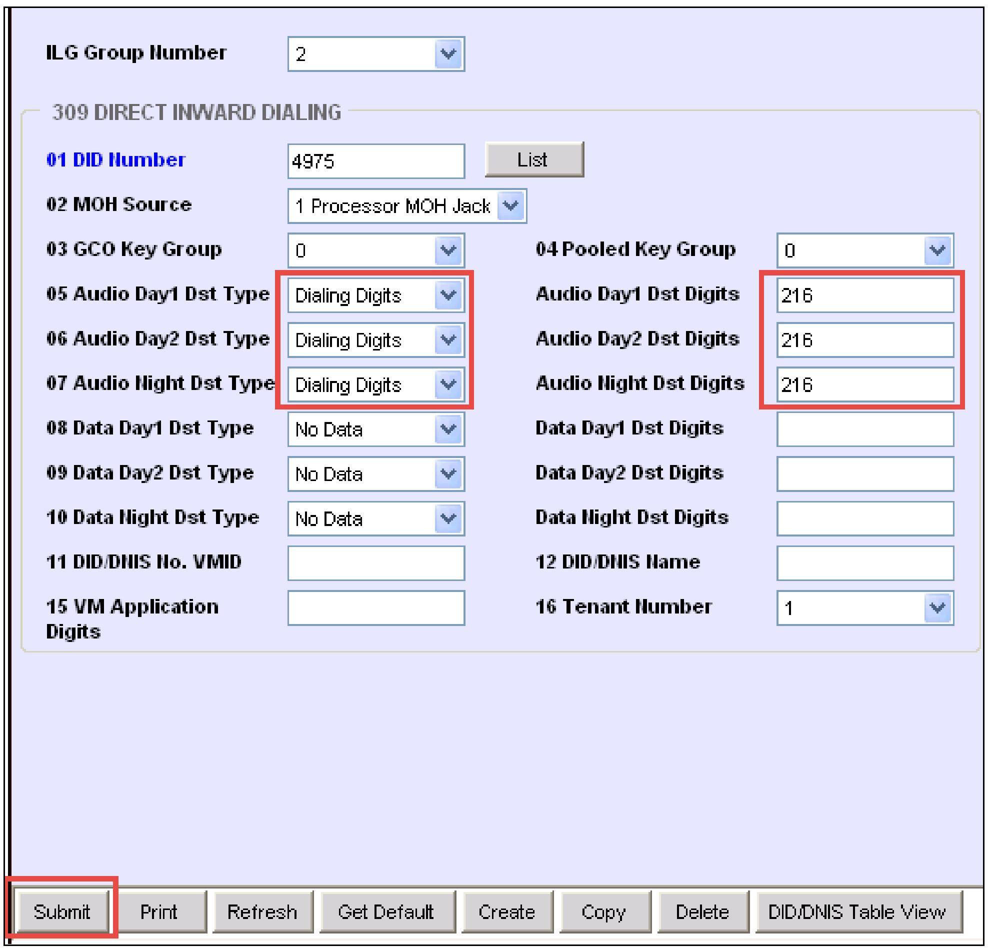

Figure 16: Direct Inward Dialing

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
1.4	Registration	N/A	Registration mode is not configured with SIP trunk
Inbound Calling Test Cases			
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	Note: No Bye is seen once PSTN Conference Controller leaves conference
2.8	Calling Party Presentation Restricted	Pass	
Outbound Calling Test Cases			
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	N/S	Not supported by CIX PBX
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	N/S	CIX does not support T38

4.4	Outbound Calling Party Sends Fax via t38	N/S	CIX does not support T38
5.1	International Outbound Dialing	Pass	Tested with INBAND and RFC2833 configuration on PBX side. Tested with an International IVR bank system. Verification was limited to audio prompts.
5.2	800/866/877/888 Outbound Dialing	Pass	Tested with INBAND and RFC2833 configuration on PBX side
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	PBX IP phone initiates transfer and hangs up before 2nd PSTN phone starts ringing. Transfer is completed to 2nd PSTN phone.
6.6	Outbound Consultative Call Transfer	Pass	
6.7	Outbound Semi Attended Call Transfer	Pass	
6.8	Outbound Consultative Call Transfer to Local Extension	Pass	
6.9	Outbound Three Way Calling	Pass	
6.10	Outbound Call Hold	Pass	PSTN endpoint connects, but no hold-Invite seen from PBX
6.11	Call Waiting	Pass	The PBX station shows audible and visual CWT notification when PBX station makes first call using SDN and second call rings into PDN. Otherwise there is only visual CWT notification.
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	No MOH heard during hold
8.3	Inbound, PBX Holds, Wait for Session Audit	Pass	Note: The procedure was changed to an inbound call to match the test title
8.4	Inbound, Wait for Session Audit	Pass	

9.1	Outbound DTMF (RTPevent)	Pass	Primary and Secondary codecs have to be set the same. For this test, the codecs were set to g711u.
9.2	Inbound DTMF(RTPevent)	Pass	
9.3	Outbound Inband (G711)	Pass	
9.4	Inbound Inband (G711)	Pass	
10.1	Codec Support	Pass	The PBX was configured with either g711 or g729 preferred and responded with primary codec in 183 and 200 on incoming calls In this case, incoming calls would offer g711 and g729 (or reverse order) and the PBX would respond with the preferred (primary) codec.