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



Table of Contents

1	Audience	3
2	Introduction	3
2.1	tekVizion Labs	3
3	SIP Trunking Network Components	4
3.1	Hardware Components	5
3.2	Software Requirements	5
4	Features	5
4.1	Features Supported	5
4.2	Features Not Supported	5
4.3	Caveats and Limitations	5
5	Configuration	6
5.1	Configuration Checklist	6
5.2	IP Address Worksheet	6
5.3	Initial System Setup	7
5.4	SIP Trunk Setup	10
5.5	Create Trunk Line Groups	15
5.6	Create Trunk DIDs	19
6	Summary of Tests and Results	21

Table of Figures

Figure 1:	SIP Trunk Lab Reference Network	4
Figure 2:	IP Configuration	7
Figure 3:	Card Assignment	8
Figure 4:	L/M/G IPU Configuration	9
Figure 5:	SIP Trunk Setup	10
Figure 6:	Channel Group Setting	11
Figure 7:	Create Index Value	12
Figure 8:	Service Definition	13
Figure 9:	Service Assignment	14
Figure 10:	URI Settings	14
Figure 11:	Incoming Line Group Assignment – Cont.	15
Figure 12:	Incoming Line Group Assignment	16
Figure 13:	Outgoing Line Groups	17
Figure 14:	Outgoing Line Groups – Cont.	18
Figure 15:	Create Trunk DIDs	19
Figure 16:	Direct Inward Dialing	20

Table of Tables

Table 1 –	PBX Configuration Steps	6
Table 2 –	IP Addresses	6



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 IntelPeer 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 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 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.

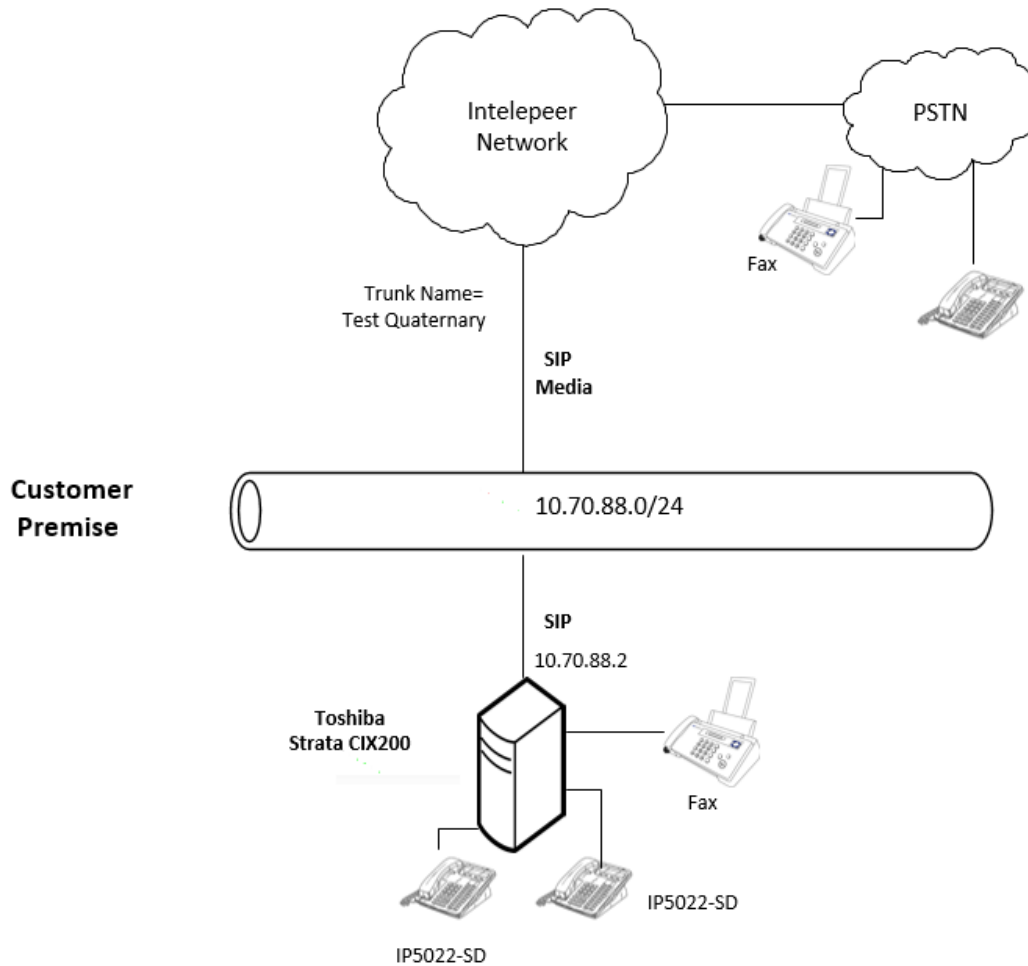


Figure 1: SIP Trunk Lab Reference Network

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.

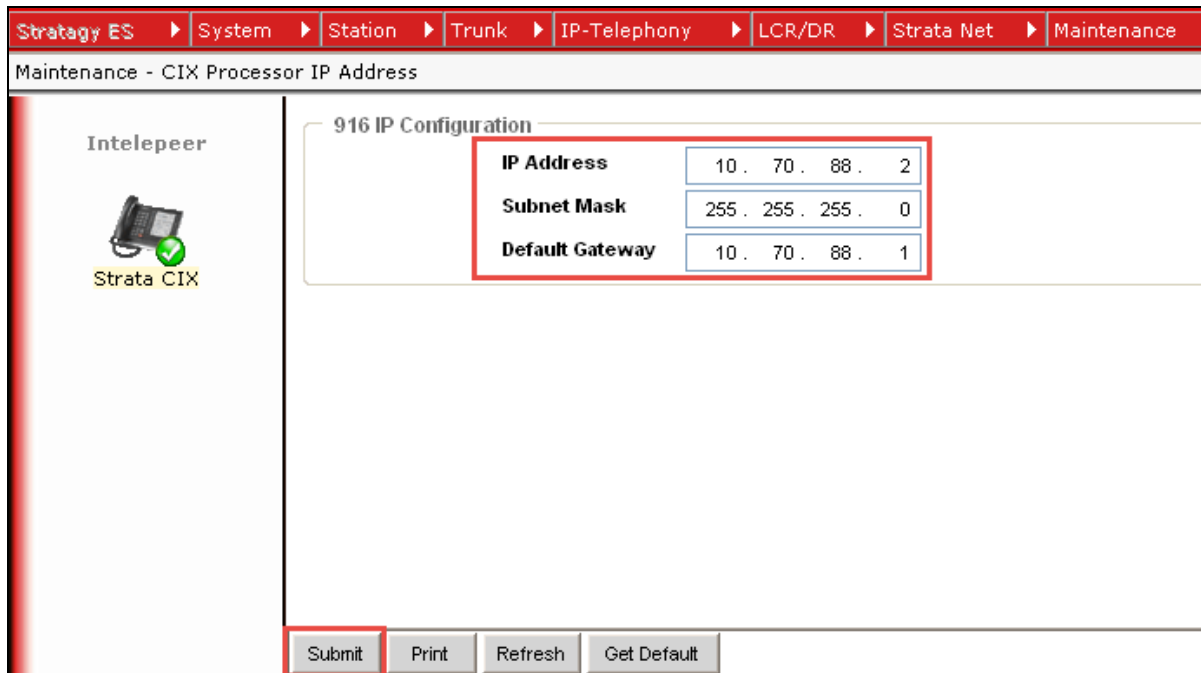
Table 2 – IP Addresses

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**



The screenshot shows a web interface for configuring IP settings. At the top, a red navigation bar contains the following menu items: Strategy ES, System, Station, Trunk, IP-Telephony, LCR/DR, Strata Net, and Maintenance. Below this, the page title is "Maintenance - CIX Processor IP Address". On the left side, there is a sidebar with the "Intelepeer" logo and a "Strata CIX" icon. The main content area is titled "916 IP Configuration" and contains a table with the following data:

IP Address	10 . 70 . 88 . 2
Subnet Mask	255 . 255 . 255 . 0
Default Gateway	10 . 70 . 88 . 1

At the bottom of the form, there are four buttons: "Submit", "Print", "Refresh", and "Get Default". The "Submit" button is highlighted with a red border.

Figure 2: IP Configuration

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

Strategy ES > System > Station > Trunk > IP-Telephony > LCR/DR > Strata Net > Maintenance > Alarm/Traffic

System - Card Assignment

100 CIX/CTX CABINET SLOT PCB ASSIGNMENTS

Cabinet **Slot** Assign Remove

PCB Type: ▼

Cabinet	Slot	PCB Type
01	01	MIPU16 - 16 IP Station
01	02	BDKU or BWDKU - 8 DKTs without Spkr UCA
01	03	RSTU BSTU or IVP8 - 8 standard telephone ports
01	04	Empty Slot or RRCU
02	01	Empty Slot or RRCU
02	02	Empty Slot or RRCU
02	03	Empty Slot or RRCU
02	04	Empty Slot or RRCU

Print Refresh

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**

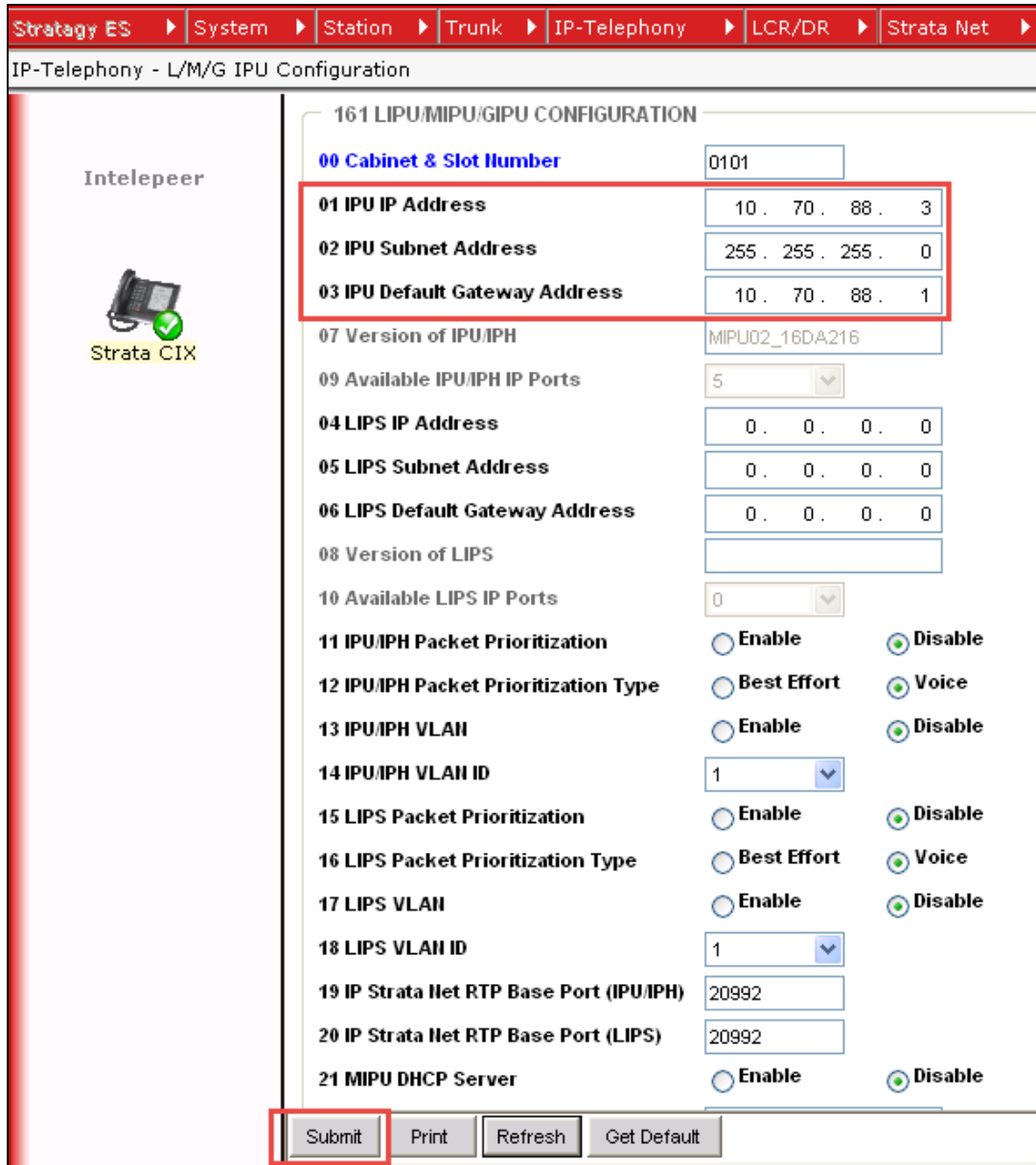


Figure 4: L/M/G IPU Configuration

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**

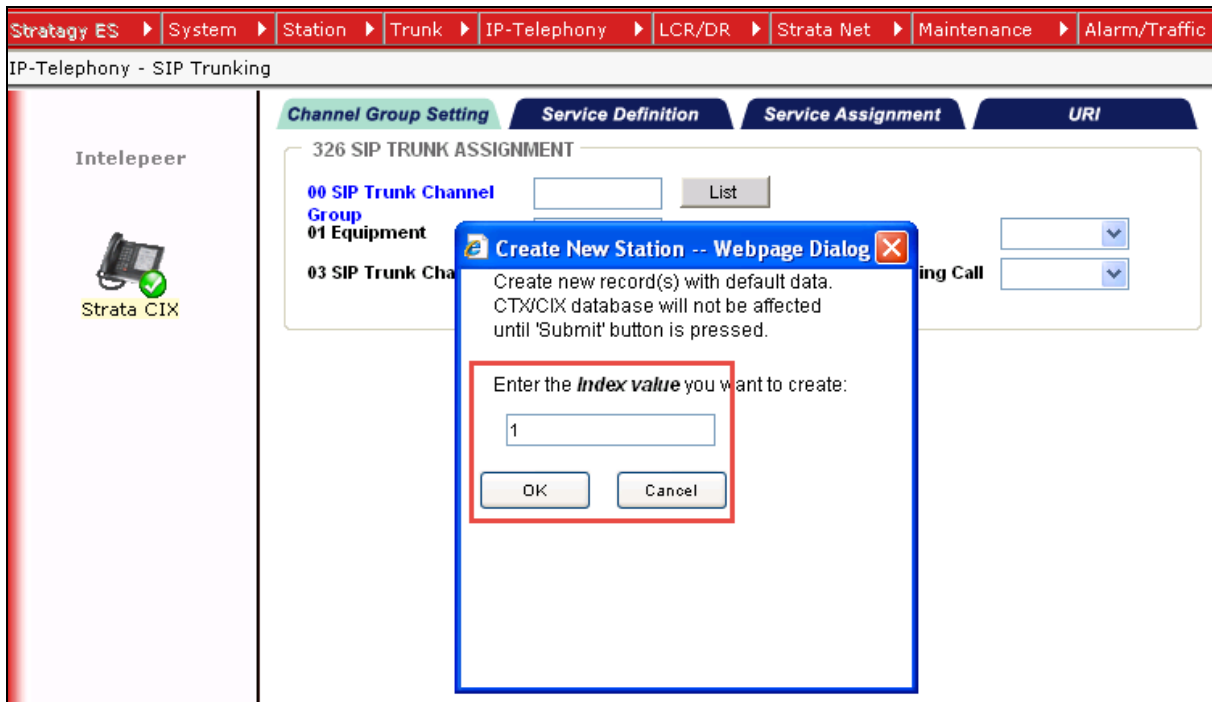
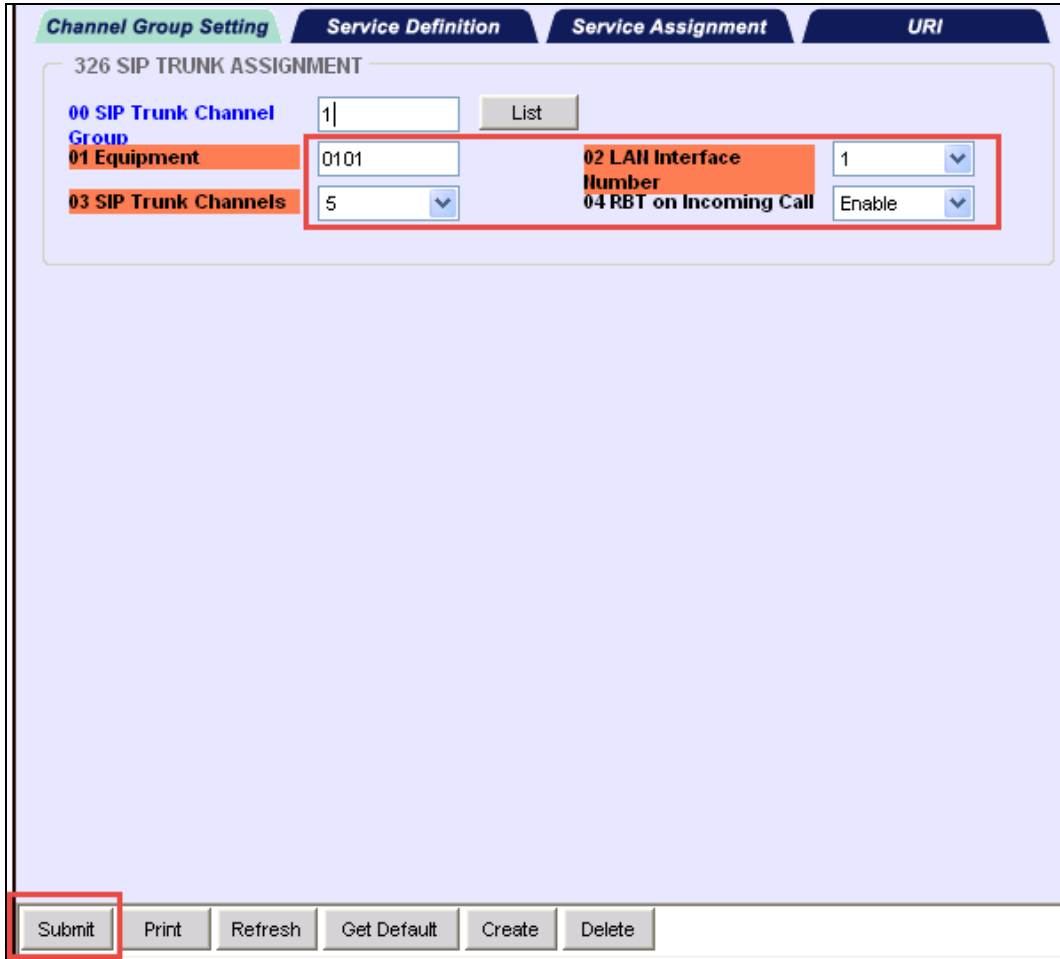


Figure 5: SIP Trunk Setup

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**



Channel Group Setting Service Definition Service Assignment URI

326 SIP TRUNK ASSIGNMENT

00 SIP Trunk Channel Group

01 Equipment

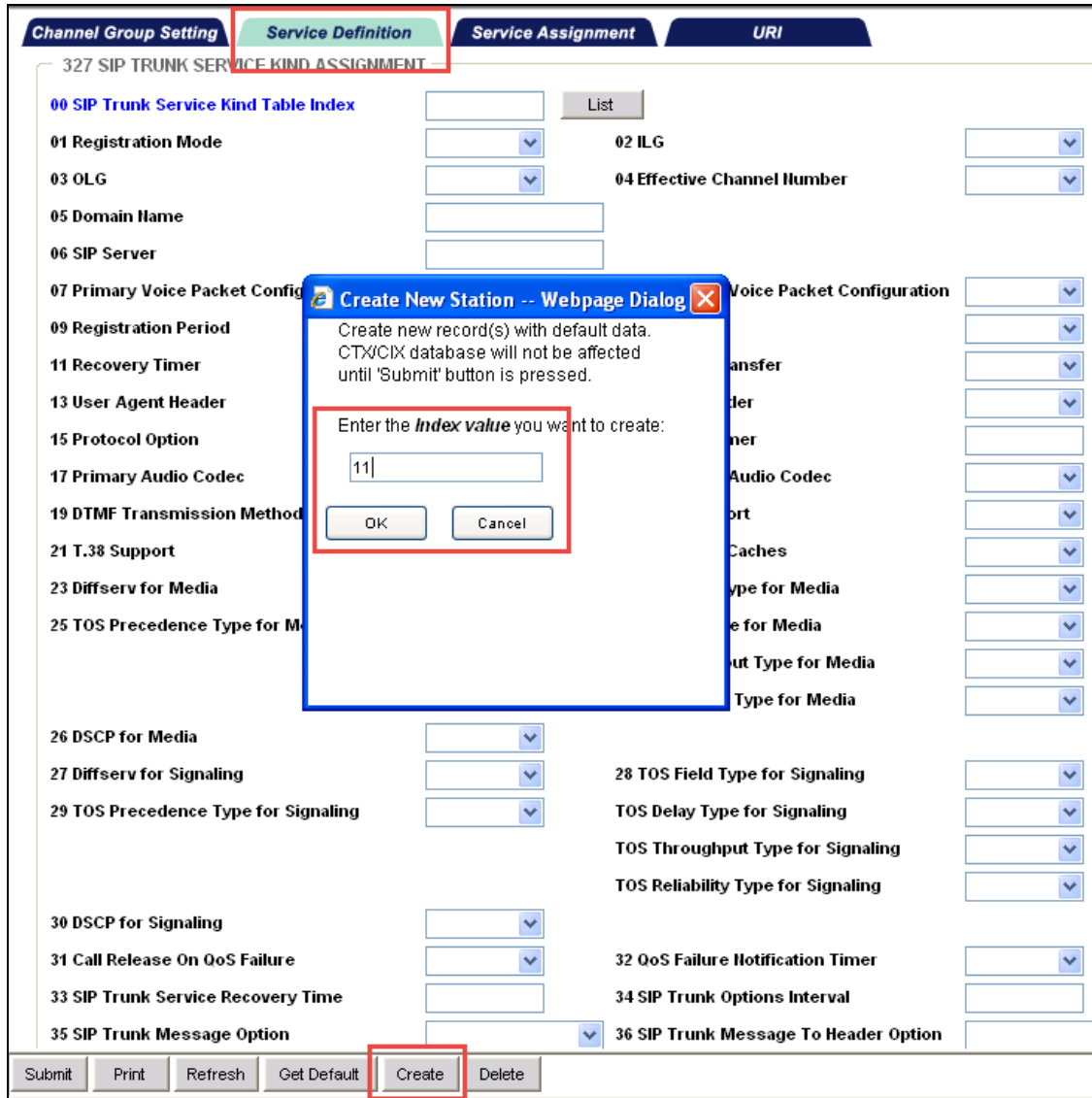
02 LAN Interface Number

03 SIP Trunk Channels

04 RBT on Incoming Call

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**



The screenshot shows the 'Service Definition' tab in a configuration interface. A dialog box titled 'Create New Station -- Webpage Dialog' is open, prompting the user to enter an 'Index value'. The value '11' is entered in the input field. The dialog also contains 'OK' and 'Cancel' buttons. In the background, the 'Create' button at the bottom of the configuration page is highlighted with a red box.

327 SIP TRUNK SERVICE KIND ASSIGNMENT

00 SIP Trunk Service Kind Table Index List

01 Registration Mode 02 ILG

03 OLG 04 Effective Channel Number

05 Domain Name

06 SIP Server

07 Primary Voice Packet Configuration Voice Packet Configuration

09 Registration Period

11 Recovery Timer

13 User Agent Header

15 Protocol Option

17 Primary Audio Codec

19 DTMF Transmission Method

21 T.38 Support

23 Diffserv for Media

25 TOS Precedence Type for Media

26 DSCP for Media

27 Diffserv for Signaling

29 TOS Precedence Type for Signaling

30 DSCP for Signaling

31 Call Release On QoS Failure

33 SIP Trunk Service Recovery Time

35 SIP Trunk Message Option

02 ILG

04 Effective Channel Number

Transfer

Server

Port

Audio Codec

Port

Caches

Type for Media

Type for Media

Type for Media

Type for Media

28 TOS Field Type for Signaling

TOS Delay Type for Signaling

TOS Throughput Type for Signaling

TOS Reliability Type for Signaling

32 QoS Failure Notification Timer

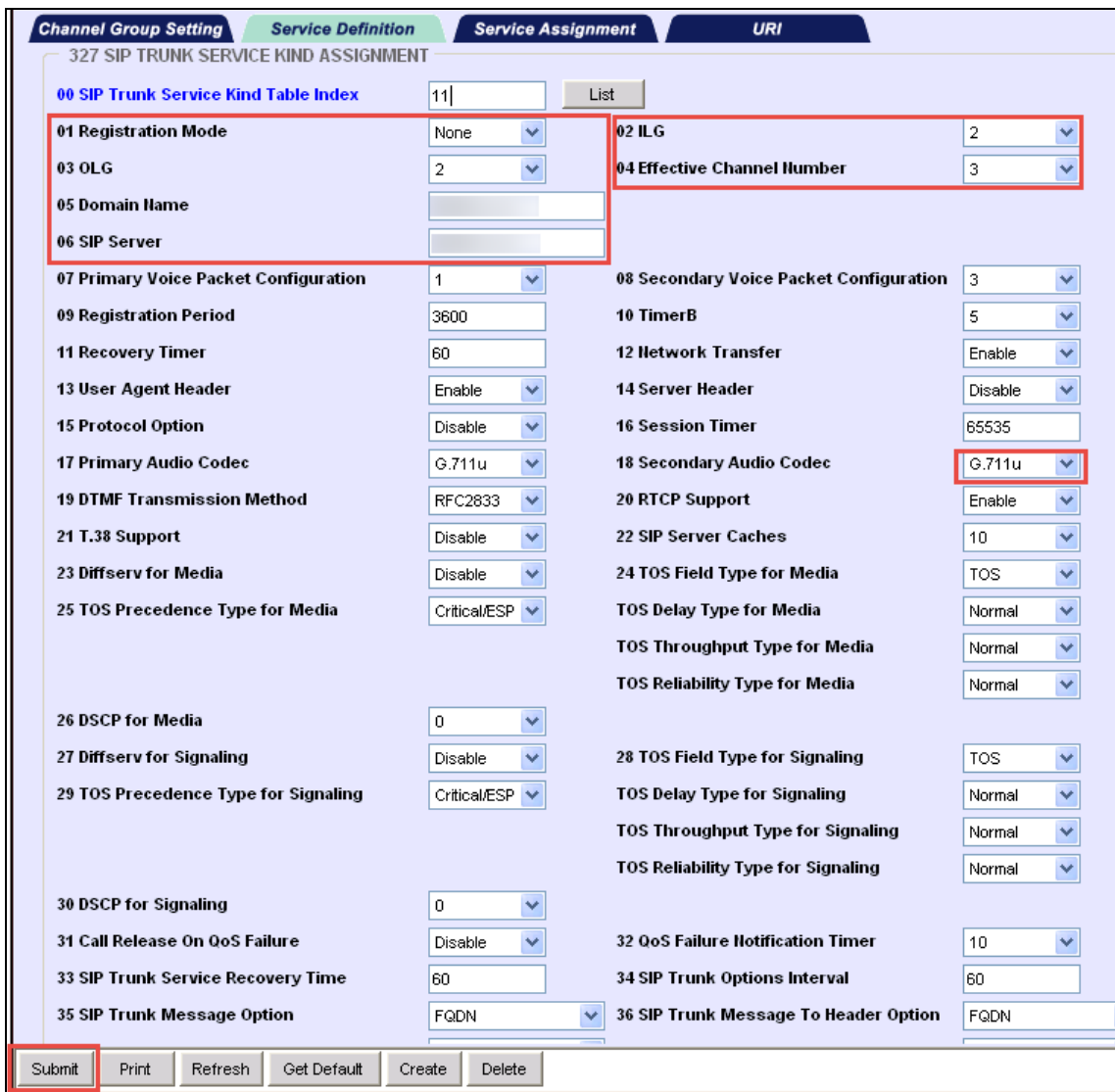
34 SIP Trunk Options Interval

36 SIP Trunk Message To Header Option

Submit Print Refresh Get Default **Create** Delete

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**



Channel Group Setting Service Definition Service Assignment URI

327 SIP TRUNK SERVICE KIND ASSIGNMENT

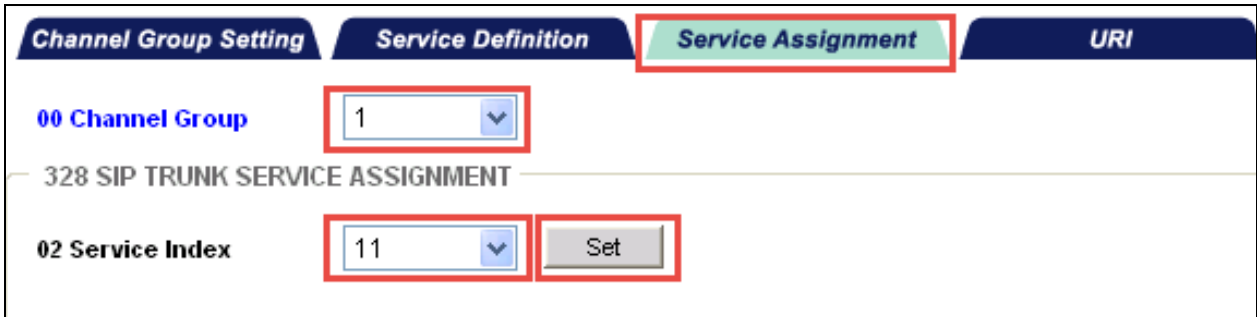
00 SIP Trunk Service Kind Table Index: 11 | List

01 Registration Mode	None	02 ILG	2
03 OLG	2	04 Effective Channel Number	3
05 Domain Name			
06 SIP Server			
07 Primary Voice Packet Configuration	1	08 Secondary Voice Packet Configuration	3
09 Registration Period	3600	10 TimerB	5
11 Recovery Timer	60	12 Network Transfer	Enable
13 User Agent Header	Enable	14 Server Header	Disable
15 Protocol Option	Disable	16 Session Timer	65535
17 Primary Audio Codec	G.711u	18 Secondary Audio Codec	G.711u
19 DTMF Transmission Method	RFC2833	20 RTCP Support	Enable
21 T.38 Support	Disable	22 SIP Server Caches	10
23 Diffserv for Media	Disable	24 TOS Field Type for Media	TOS
25 TOS Precedence Type for Media	Critical/ESP	TOS Delay Type for Media	Normal
		TOS Throughput Type for Media	Normal
		TOS Reliability Type for Media	Normal
26 DSCP for Media	0	28 TOS Field Type for Signaling	TOS
27 Diffserv for Signaling	Disable	TOS Delay Type for Signaling	Normal
29 TOS Precedence Type for Signaling	Critical/ESP	TOS Throughput Type for Signaling	Normal
		TOS Reliability Type for Signaling	Normal
30 DSCP for Signaling	0	32 QoS Failure Notification Timer	10
31 Call Release On QoS Failure	Disable	34 SIP Trunk Options Interval	60
33 SIP Trunk Service Recovery Time	60	36 SIP Trunk Message To Header Option	FQDN
35 SIP Trunk Message Option	FQDN		

Submit Print Refresh Get Default Create Delete

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



Channel Group Setting Service Definition **Service Assignment** URI

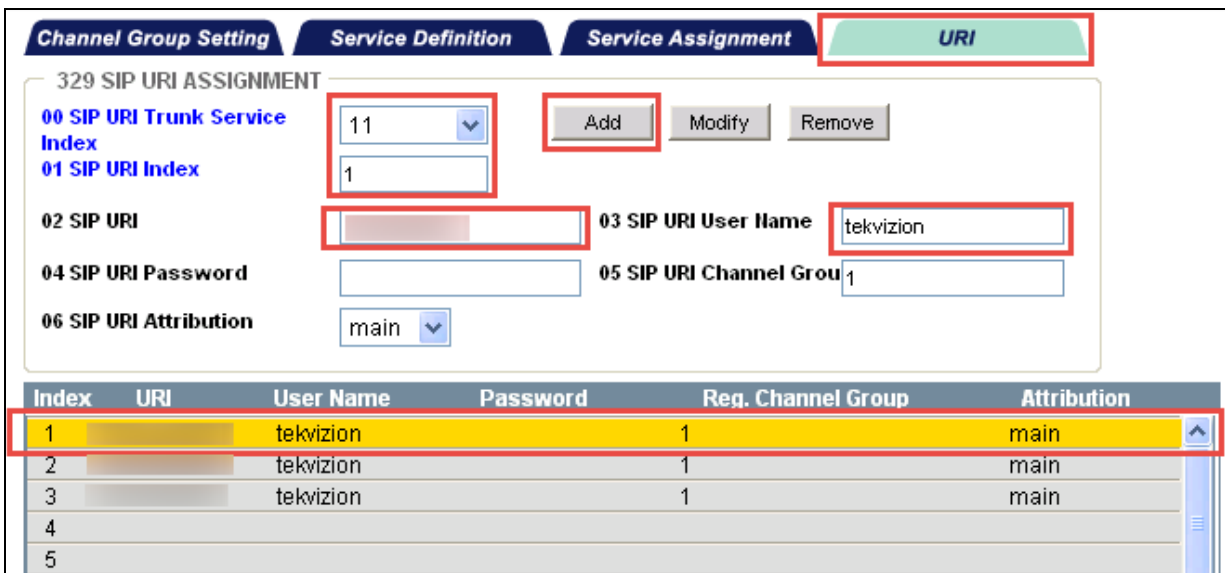
00 Channel Group 1

328 SIP TRUNK SERVICE ASSIGNMENT

02 Service Index 11 Set

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



Channel Group Setting Service Definition Service Assignment **URI**

329 SIP URI ASSIGNMENT

00 SIP URI Trunk Service Index 11 Add Modify Remove

01 SIP URI Index 1

02 SIP URI [Redacted] **03 SIP URI User Name** tekvizion

04 SIP URI Password [Redacted] **05 SIP URI Channel Group** 1

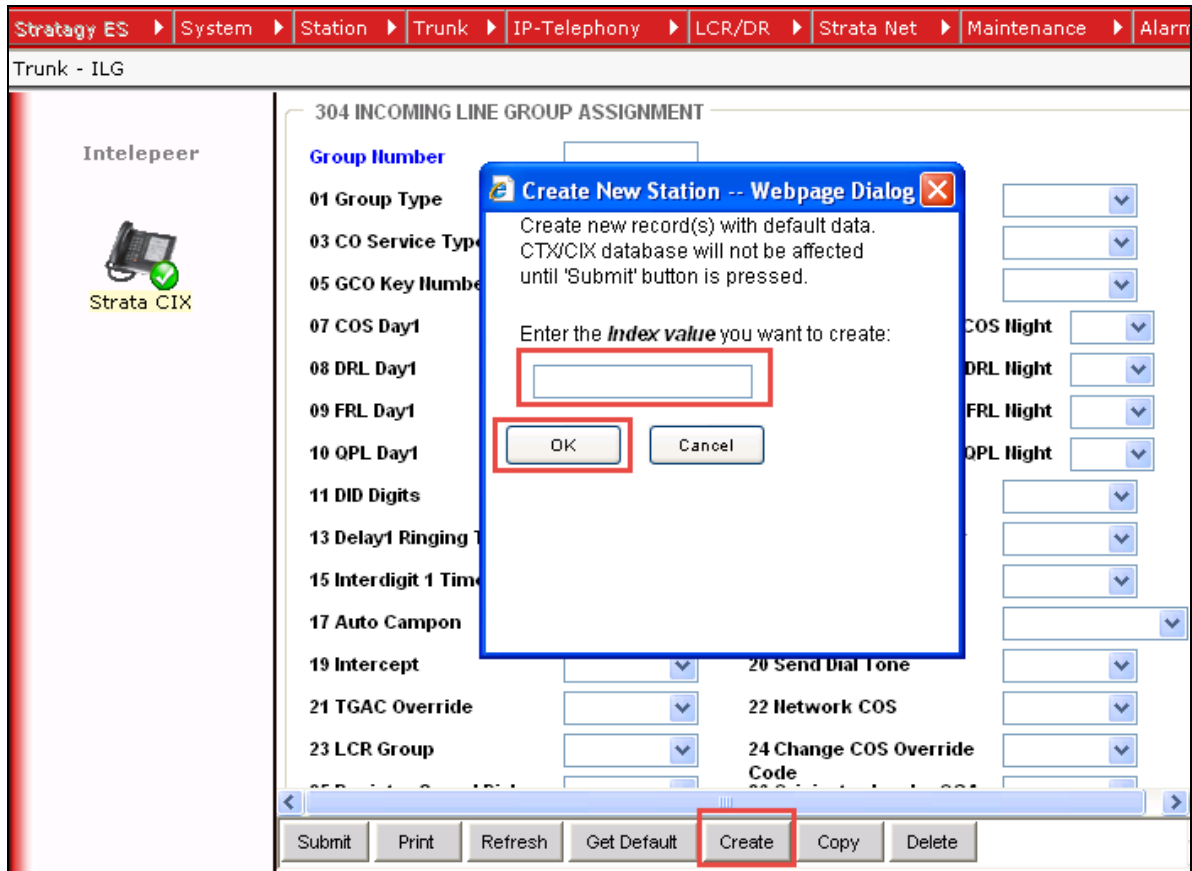
06 SIP URI Attribution main

Index	URI	User Name	Password	Reg. Channel Group	Attribution
1	[Redacted]	tekvizion	[Redacted]	1	main
2	[Redacted]	tekvizion	[Redacted]	1	main
3	[Redacted]	tekvizion	[Redacted]	1	main
4	[Redacted]	[Redacted]	[Redacted]	[Redacted]	[Redacted]
5	[Redacted]	[Redacted]	[Redacted]	[Redacted]	[Redacted]

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



The screenshot displays the '304 INCOMING LINE GROUP ASSIGNMENT' form in the IntelPeer web application. A modal dialog titled 'Create New Station -- Webpage Dialog' is open, asking the user to 'Enter the *Index value* you want to create:'. The dialog has an 'OK' button and a 'Cancel' button. In the background, the form includes fields for 'Group Number', '01 Group Type', '03 CO Service Type', '05 GC0 Key Number', '07 COS Day1', '08 DRL Day1', '09 FRL Day1', '10 OPL Day1', '11 DID Digits', '13 Delay1 Ringing Time', '15 Interdigit 1 Time', '17 Auto Campon', '19 Intercept', '20 Send Dial Tone', '21 TGAC Override', '22 Network COS', '23 LCR Group', and '24 Change COS Override Code'. At the bottom of the form, there are buttons for 'Submit', 'Print', 'Refresh', 'Get Default', 'Create', 'Copy', and 'Delete'. The 'Create' button is highlighted with a red box.

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**

304 INCOMING LINE GROUP ASSIGNMENT

Group Number	<input type="text" value="2"/>		
01 Group Type	<input type="text" value="SIP"/>	02 Line Type	<input type="text" value="CO"/>
03 CO Service Type	<input type="text" value="DID"/>	04 Private Service Type	<input type="text" value="Standard"/>
05 GC0 Key Number	<input type="text" value="0"/>	06 Pool Key Number	<input type="text" value="0"/>
07 COS Day1	<input type="text" value="1"/>	COS Day2	<input type="text" value="1"/>
08 DRL Day1	<input type="text" value="1"/>	DRL Day2	<input type="text" value="1"/>
09 FRL Day1	<input type="text" value="1"/>	FRL Day2	<input type="text" value="1"/>
10 OPL Day1	<input type="text" value="1"/>	OPL Day2	<input type="text" value="1"/>
11 DID Digits	<input type="text" value="4"/>	12 Speech/3.1KHz	<input type="text" value="Audio"/>
13 Delay1 Ringing Timer	<input type="text" value="12"/>	14 Delay2 Ringing Timer	<input type="text" value="24"/>
15 Interdigit 1 Timer	<input type="text" value="15"/>	16 Interdigit 2 Timer	<input type="text" value="5"/>
17 Auto Campon	<input type="text" value="Enable"/>	18 Calling Number ID	<input type="text" value="User Provided"/>
19 Intercept	<input type="text" value="Disable"/>	20 Send Dial Tone	<input type="text" value="Disable"/>
21 TGAC Override	<input type="text" value="Disable"/>	22 Network COS	<input type="text" value="1"/>
23 LCR Group	<input type="text" value="1"/>	24 Change COS Override Code	<input type="text" value="Disable"/>
25 Register Speed Dial Codes	<input type="text" value="Disable"/>	26 Originator Invoke OCA	<input type="text" value="Disable"/>
27 Senderized Tone Mode	<input type="text" value="Dial Tone"/>	28 Emergency Call Group	<input type="text" value="1"/>
29 Tenant Number	<input type="text" value="1"/>	30 Call-By-Call Cause	<input type="text" value="UserBusy"/>

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

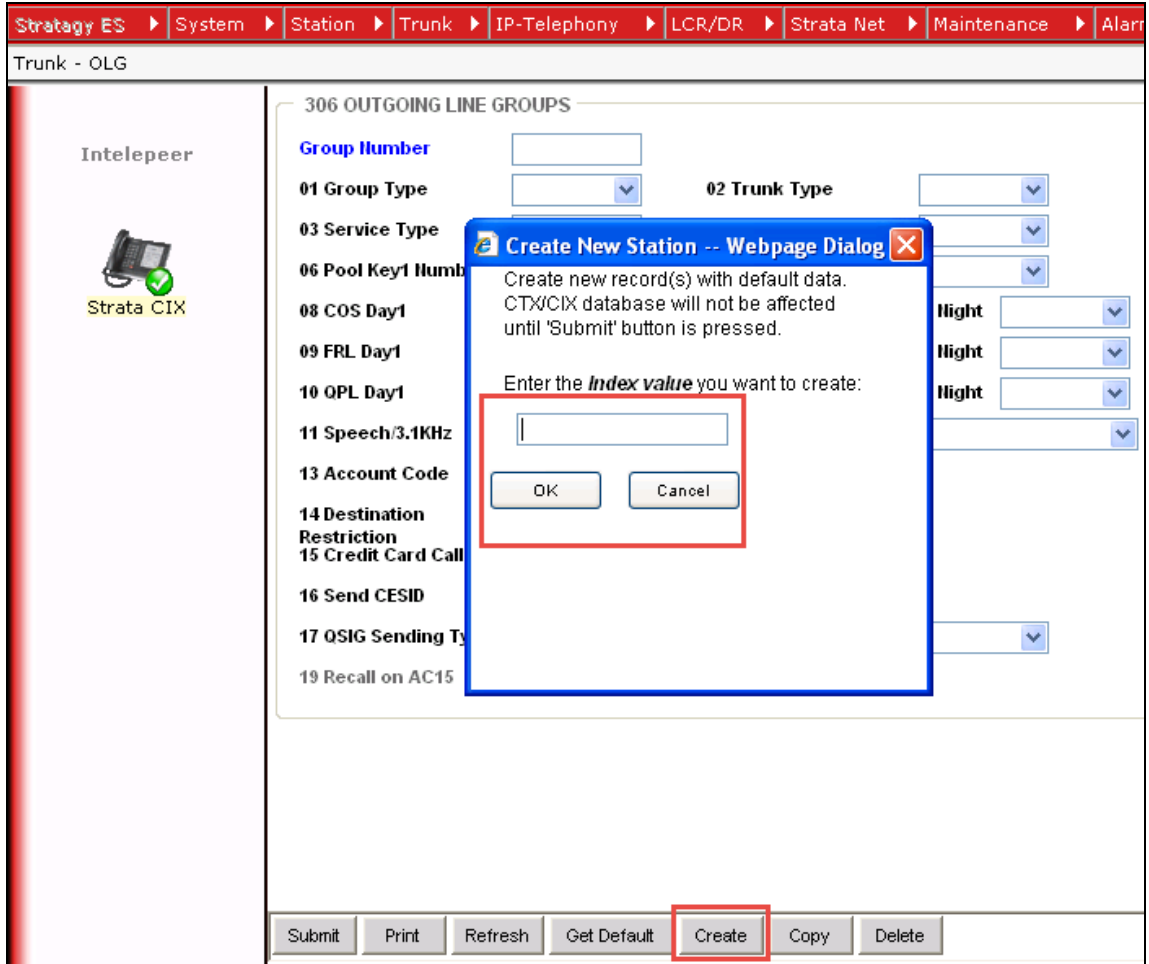
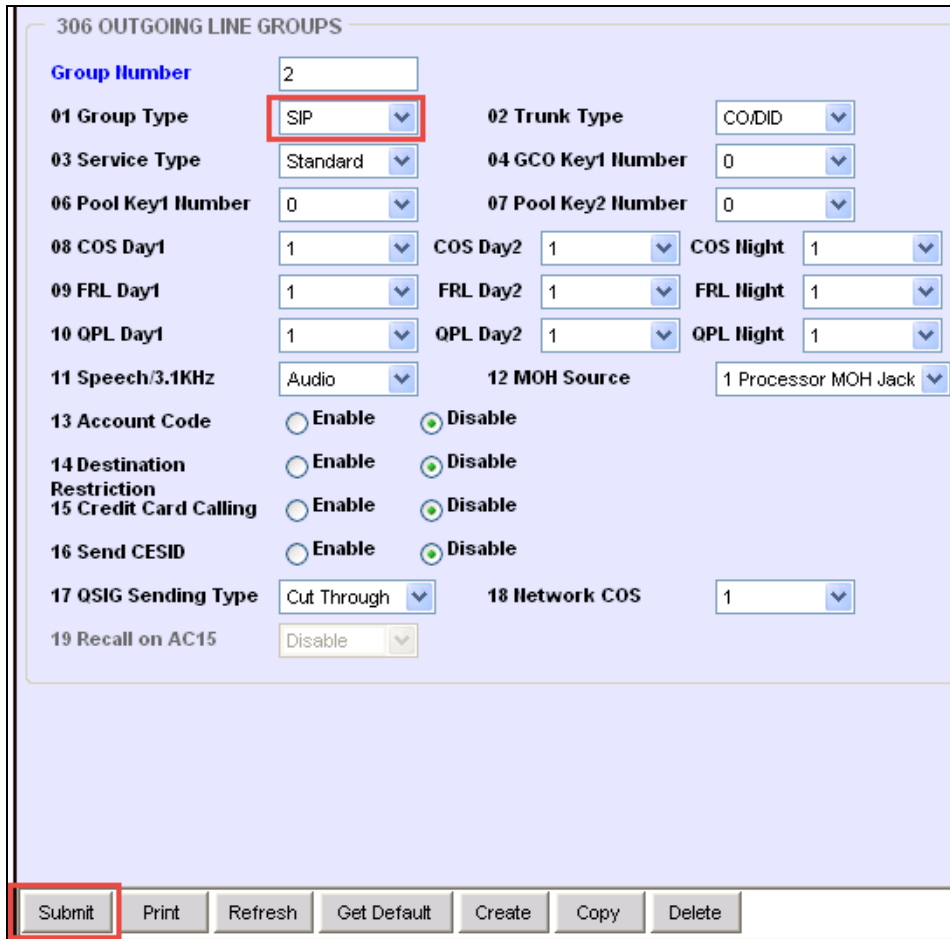


Figure 13: Outgoing Line Groups

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**



306 OUTGOING LINE GROUPS

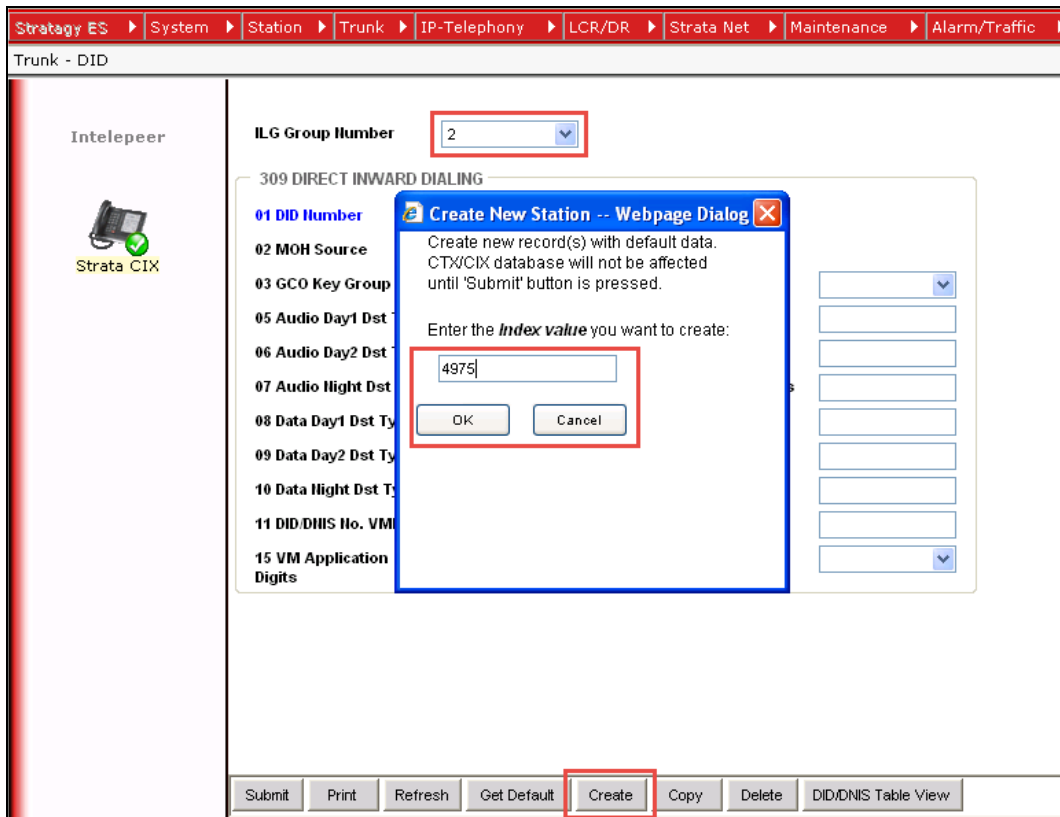
Group Number	2	02 Trunk Type	CO/DID
01 Group Type	SIP	04 GC0 Key1 Number	0
03 Service Type	Standard	07 Pool Key2 Number	0
06 Pool Key1 Number	0	08 COS Day1	1
08 COS Day1	1	COS Day2	1
09 FRL Day1	1	COS Night	1
10 QPL Day1	1	FRL Day2	1
11 Speech/3.1KHz	Audio	FRL Night	1
13 Account Code	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	QPL Day2	1
14 Destination	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	QPL Night	1
15 Credit Card Calling	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	12 MOH Source	1 Processor MOH Jack
16 Send CESID	<input type="radio"/> Enable <input checked="" type="radio"/> Disable	17 QSIG Sending Type	Cut Through
18 Network COS	1	19 Recall on AC15	Disable

Submit Print Refresh Get Default Create Copy Delete

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**



The screenshot displays the 'Trunk - DID' configuration page in the IntelePeer web interface. The breadcrumb navigation at the top reads: Strategy ES > System > Station > Trunk > IP-Telephony > LCR/DR > Strata Net > Maintenance > Alarm/Traffic. The main content area is titled 'Trunk - DID' and features a left sidebar with the IntelePeer logo and 'Strata CIX' status. The configuration form includes the following fields:

- ILG Group Number:** A dropdown menu with the value '2' selected.
- 309 DIRECT INWARD DIALING:** A section header.
- 01 DID Number:** A field with a blue dialog box overlaid on it.
- 02 MOH Source:** A dropdown menu.
- 03 GCO Key Group:** A dropdown menu.
- 05 Audio Day1 Dst:** A dropdown menu.
- 06 Audio Day2 Dst:** A dropdown menu.
- 07 Audio Night Dst:** A dropdown menu.
- 08 Data Day1 Dst Ty:** A dropdown menu.
- 09 Data Day2 Dst Ty:** A dropdown menu.
- 10 Data Night Dst Ty:** A dropdown menu.
- 11 DID/DNIS No. VM:** A dropdown menu.
- 15 VM Application Digits:** A dropdown menu.

The 'Create New Station -- Webpage Dialog' is a blue window with the following text: 'Create new record(s) with default data. CTX/CIX database will not be affected until 'Submit' button is pressed.' Below this text is a prompt: 'Enter the *Index value* you want to create:' followed by a text input field containing '4975'. At the bottom of the dialog are 'OK' and 'Cancel' buttons. The 'Create' button in the main form's footer is highlighted with a red box.

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

ILG Group Number	2	
309 DIRECT INWARD DIALING		
01 DID Number	4975	List
02 MOH Source	1 Processor MOH Jack	
03 GCO Key Group	0	04 Pooled Key Group 0
05 Audio Day1 Dst Type	Dialing Digits	Audio Day1 Dst Digits 216
06 Audio Day2 Dst Type	Dialing Digits	Audio Day2 Dst Digits 216
07 Audio Night Dst Type	Dialing Digits	Audio Night Dst Digits 216
08 Data Day1 Dst Type	No Data	Data Day1 Dst Digits
09 Data Day2 Dst Type	No Data	Data Day2 Dst Digits
10 Data Night Dst Type	No Data	Data Night Dst Digits
11 DID/DNIS No. VMID		12 DID/DNIS Name
15 VM Application Digits		16 Tenant Number 1
<input type="button" value="Submit"/> <input type="button" value="Print"/> <input type="button" value="Refresh"/> <input type="button" value="Get Default"/> <input type="button" value="Create"/> <input type="button" value="Copy"/> <input type="button" value="Delete"/> <input type="button" value="DID/DNIS Table View"/>		

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	<p>The PBX was configured with either g711 or g729 preferred and responded with primary codec in 183 and 200 on incoming calls</p> <p>In this case, incoming calls would offer g711 and g729 (or reverse order) and the PBX would respond with the preferred (primary) codec.</p>