

SIP Trunking using the Optimum Business SIP Trunk Adaptor and the Vertical SBX IP320

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Vertical SBX IP320 IP PBX for proper operation Optimum Business Sip Trunking.

2 SIP Trunk Adaptor Set-up Instructions

These instructions describe the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

Step 1:

Log on to the Optimum Business SIP Trunk Adaptor

1. Connect a PC to port 4 of the Optimum Business SIP Trunk Adaptor, the silver device labeled Edgewater Networks, 4550 series.



2. Open a Web browser and go to IP Address <http://10.10.200.1>. A login box will appear.
3. Enter login and password and click 'OK'.
Login: pbxinstall
Password: slptrunk



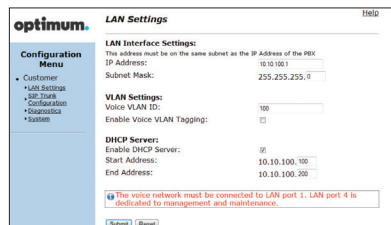
Step 2:

Click on the LAN Settings Link

1. Assign an IP Address to the LAN interface of the SIP Trunk Adaptor.
The IP address must be on the same subnet as the IP PBX. This changes the address on port 1 of the Optimum Business SIP Trunk Adaptor.
Note: This will become your local SIP proxy IP address. No other IP addresses will be provided by Cablevision.

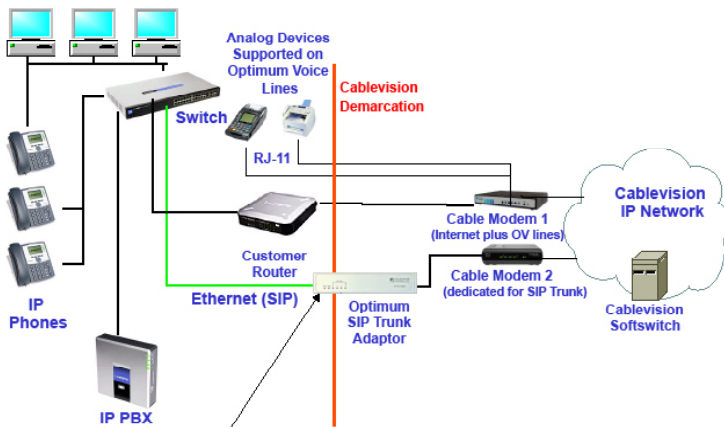
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



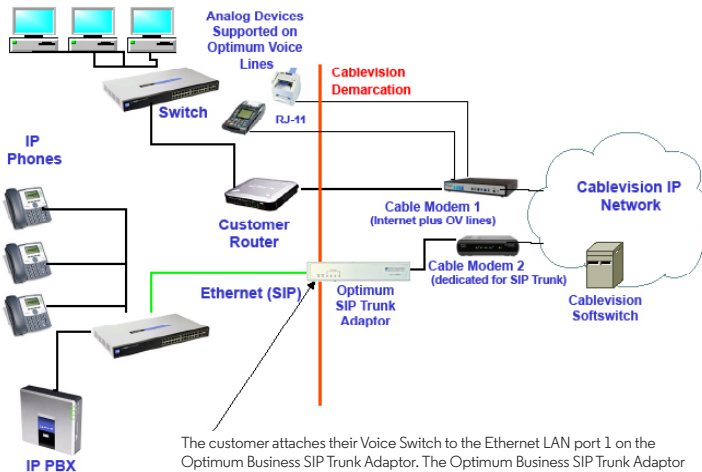
- Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
- Click 'Submit'.

Diagram 1 SIP Trunk Adaptor for IP-PBXs
Example: Single LAN Configuration



Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

Diagram 2 SIP Trunk Adaptor for IP-PBXs
Example: Separate Voice and Data Networks Configuration



The customer attaches their Voice Switch to the Ethernet LAN port 1 on the Optimum Business SIP Trunk Adaptor. The Optimum Business SIP Trunk Adaptor can be enabled as a DHCP server to provide routing for the separate voice network.

Step 3:

Click on the SIP Trunk Configuration Link

1. Select your IP PBX make and model from the drop-down menu.
2. Specify how the IP PBX will register to the Optimum Business SIP Trunk Adaptor.
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.
4. Click 'Submit'.

The screenshot shows the 'SIP Trunk Configuration' page. On the left is a 'Configuration Menu' with links: Customer, LAN Settings, SIP Trunk Configuration (selected), Diagnostics, and System. The main content area has a 'Select your PBX:' dropdown menu with 'Asterisk' selected. Below this are two radio button options: 'Passive connection using the local, private IP address of the PBX interface' (selected) and 'Active connection using registration'. The 'Active connection' section includes fields for 'User Id:' (set to 'secret') and 'Password:' (masked with asterisks). There is a checkbox for 'Convert Inband DTMF:' which is currently unchecked. Below these are 'Submit' and 'Reset' buttons. A 'Status:' section shows 'Trunk Status:' as 'Not Registered' and a list of 'DID's' (0164030809, 0164030760, 0164030769, 0164030765, 0164030841).

Step 4:

Diagnostics Link

You can make a test call directly from your phone or use the test call application under the Diagnostics link.

The screenshot shows the 'Network Test Tools' page. On the left is the same 'Configuration Menu' as in Step 3. The main content area has a title 'Network Test Tools' and a description: 'A network administrator may use the test tools on this page to verify connectivity of the System and trace the path of data throughout the network.' Below this are three test sections: 1. 'Outbound Call Test:' with a description 'This test will place a call to the provided telephone number and play a series of tones for 30 seconds.' and fields for 'Pilot Number:' and 'Telephone Number:'. 2. 'Inbound Call Test:' with a description 'When this test is enabled calls received for the pilot number are diverted to the internal Test UK for 15 minutes, after this elapsed time the test is automatically disabled.' and radio button options for 'Enabled' (selected) and 'Disabled'. 3. 'Ping Test:' with a description 'IP Address to Ping:' and a 'Ping' button. Below that is 'Traceroute Test:' with a description 'IP Address to Trace:' and a 'Traceroute' button. Each test section also has a 'Reset' button.

Step 4 continued

Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
Call	Initiates a call outbound to a telephone number entered or inbound to the pilot number displayed.
Inbound Call Test (radio button)	Indicates whether inbound test call will be enabled or disabled. If inbound test calls are enabled, calls made to the pilot number will be redirected to the test UA for fifteen minutes. When the pilot number is dialed, you will hear a test message play.
Submit	Enables or disables the inbound call test.
IP Address to Ping	Verifies basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address.
Ping Button	Sends a ping to the IP address specified in the field "IP Address to Ping".
IP Address to Trace	Tracks the progress of a packet through the network. The packet can be tracked through the WAN or LAN interfaces of the adaptor.
Interface (radio button)	Indicates whether a packet will be tracked through the LAN or the WAN.
Traceroute Button	Initiates a traceroute to the specified IP address on either the LAN or the WAN.
Reset	Clears all fields and selections and allows you to enter new information. Reset applies to outbound call test, ping and traceroute.

3 Additional Set-up Information Systems

optimum.

System[Help](#)

Configuration Menu

- Customer
 - LAN Settings
 - SIP Trunk Configuration
 - Diagnostics
 - System

Software Version:
Version 11.6.14.1 -- Fri Jan 4 17:49:28 PST 2013

Hostname:
5164939899

Model:
EdgeMarc 4552

Vendor:
Cablevision

LAN Interface MAC Address:
A8:70:A5:00:D8:18

Registration Status:
The ALG feature is registered. View [license key](#).

System Date:
02/29/2016 15:03:40 UTC

Change Password:

- [pbxinstall](#)

Field	Description
Pbxinstall Link	Select to change the default password for the pbxinstall login ID. Only the password can be changed. The login ID cannot be changed.

Password

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Set Password[Help](#)

Configuration Menu

- Customer
 - LAN Settings
 - SIP Trunk Configuration
 - Diagnostics
 - System

Change the GUI password by filling in the fields below. The password must be between 6 and 8 characters in length.

Username:

Current Password:

New Password:

Confirm Password:

Field	Description
Username	Specifies the username for which the password can be changed.
Current Password	Specifies the current password.
New Password	Specifies the new password.
Confirm Password	Confirms the new password.
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

4 International Calling

Optimum Voice Business Trunking offers an optional International Calling Service for direct-dialed calls made from the Customer's business or from any phone via the Optimum Voice International Calling remote access number to destinations outside of the United States, Puerto Rico, Canada and the U.S. Virgin Islands at per minute rates. The Customer must login to the Optimum Business Account Center and activate the service on the Optimum Business Trunk Pilot telephone number to activate the service and manage the monthly International spending limit for the account.

Activating International calling on the Pilot TN will enable International calling for all Direct Inward Dial (DIDs) telephone numbers as well. Blocking International calling for one or more DIDs is managed by the customer directly from the PBX phone system configuration. To minimum the exposure to fraudulent calling, It is recommended to limit International calling capability to those DID's that require it and set up an account spending limit that reflects what is necessary to run the business.

It is the Customer (and/or the Customer Agent's) responsibility to properly secure the customer's PBX to prevent the PBX from being compromised and fraudulent calls from being made by unauthorized (internal or external) users. If fraudulent calls are detected, Cablevision reserves the right to disable International Calling until the PBX is properly secured by the customer.

Goal

The purpose of this document is to describe the steps needed to configure the Vertical SBX IP320 IP PBX for proper operation with Optimum Business SIP Trunking.

Prerequisites

Please follow the instructions in the Optimum Business SIP Trunk Set-up Guide. The Set-up Guide was left by the Optimum Business technician at installation. If you do not have the Set-up Guide, go to optimumbusiness.com/sip to download a copy.

This configuration guide provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.

IP PBX Information

Manufacturer:	Vertical Communications
Model:	Vertical SBX IP320
Software Version:	MPB-VD78P-4.0Ad APR/13 PC ADM-GSVAD D.0Aa 2012.08.30
Does the PBX send SIP Registration messages (Yes/No)?	Yes

Vertical SBX IP320 Configuration

The steps below describe the minimum configuration required to enable the PBX to use a SIP trunk for inbound and outbound calling. Please refer to the Vertical SBX IP320 product documentation for more information on other advanced PBX features.

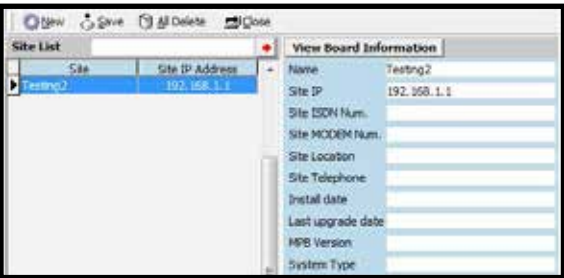
Network Settings

You need to install PCADMIN on your PC. This will be the tool you will use to configure the KSU device. Programming through the phones is possible, but PCADMIN is a much easier option. You will first need to create a site for the KSU and assign an IP address. For the certification test, the IP address was left at its default of 192.168.1.1 and the Optimum Business SIP Trunk Adaptor was configured with 192.168.1.11. The VoIP card was configured for 192.168.1.2. This is the IP address that will essentially be used for SIP signaling with the Optimum Business SIP Trunk Adaptor. As for the KSU address of 192.168.1.1, this will be used to access the device and manage configuration.

Navigate to **“Tools”** from the top and select **“Site Information”**.



Then click **“New”** and you will be prompted to enter information for your KSU. For simplicity we just gave it a random name and assigned it an IP address of 192.168.1.1.



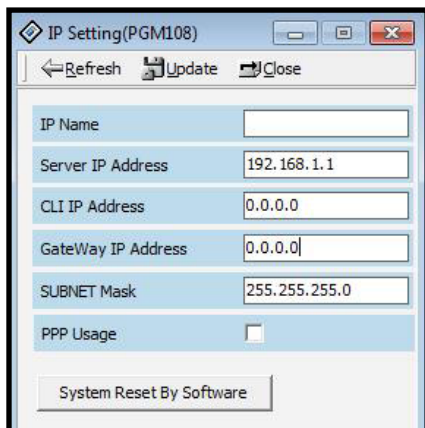
When finished, save changes. Then right click the site and select **“Connect (LAN)”** to connect to the KSU through PCADMIN. When connected to the KSU you will be prompted for a password, by default it is 0000.



When you see the Connect icon turn green it means you have successfully connected.



Navigate to **“Pre-Programmed ▶ IP Settings”** and enter the IP address of the KSU next to **“Server IP Address”** along with the subnet. You can leave other fields as 0.0.0.0 for now.

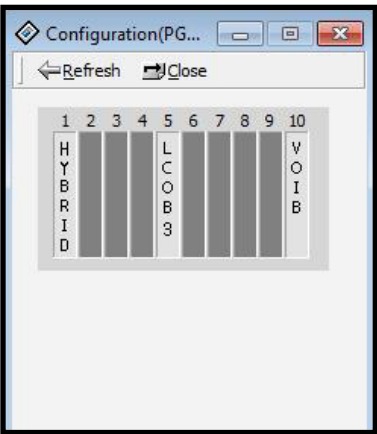


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Navigate to **“Pre-Programmed ▶ Configuration(PGM 101-103)”**.
Right click slot 10, go to **“Select Board ▶ COL ▶ VOIB”**.

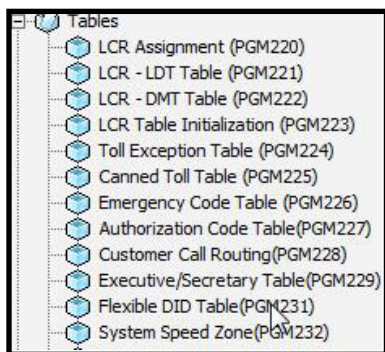


You should see **“VOIB”** written under 10 above after doing this. You must do this to enable the VOIB even if you have it installed on the KSU.



Phone DID and Extension assignment

Now navigate to **“Tables ▶ Flexible DID Table”**.



The two station extensions we used were 100 and 101. First, right click Index 100 and select **“Update Tool”**. For **“Day Type”**, **“Night Type”**, **“Weekend Type”**, and **“Lunch Mode Type”** enter **“Station”**. For **“Day Dest.”**, **“Night Dest.”**, **“Weekend Dest.”**, and **“Lunch Mode Dest.”** enter 100. Enter 0 and choose 1 as displayed below on top right. When done click **Update** then **Refresh**. Repeat this for station 101.

Update Tool

Name:

Update Delete All Inset All Delete Close

Edit with Range

Day Type: Station

Night Type: Station

Weekend Type: Station

Remote Type: Not Assigned

Launch Mode Type: Station

Day Dest.: 100

Night Dest.: 100

Weekend Dest.: 100

Remote Dest.:

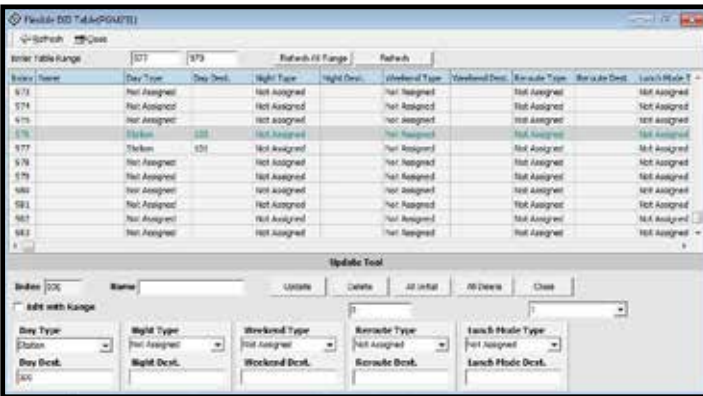
Launch Mode Dest.:

Launch Mode Dest.: 100

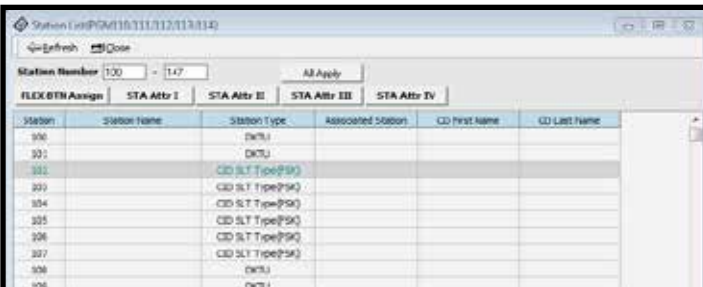
Site	Location	DOB	Follow-up	DOB	Follow-up	DOB	Follow-up	DOB	Follow-up	DOB	Follow-up
011	MAHAR	01	3/20/00	011	3/20/00	011	Not Assigned	MAHAR	01	0	3

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You will then under **“Flexible DID Table(PGUM231)”** need to assign these extensions to DIDs. The DIDs used were 6314488976 for station 100 and 6314488977 for station 101. To assign to them to their corresponding stations, you need to go to the last 3 digits of each DID, in this case 976 and 977 and assign them to 100 and 101 respectively. Click on 976 and change the **“Day Type”** to **“Station”**, and the **“Day Dest.”** to 100. Also enter 0 and 1 at the top as shown below. When done click **Update** then **Refresh**. Repeat for remaining stations.



Navigate to **“Station Base Program ▶ Station List(PGM 110/111/112/113/114)”** and enter 100-107 next to **“Station Number”**. Right click Station 102 and select **“Update Tool”**. Under **“Station Type”** select **“CID SLT Type(FSK)”** then click **Update**. We did this for Stations 102-107 for visual purposes. When done click **Refresh**.



Now navigate to **“Flex Button Assignment(PGM115/125)”** and enter 100 next to **“Current Station”**. For flex buttons 1-7 we entered the following:

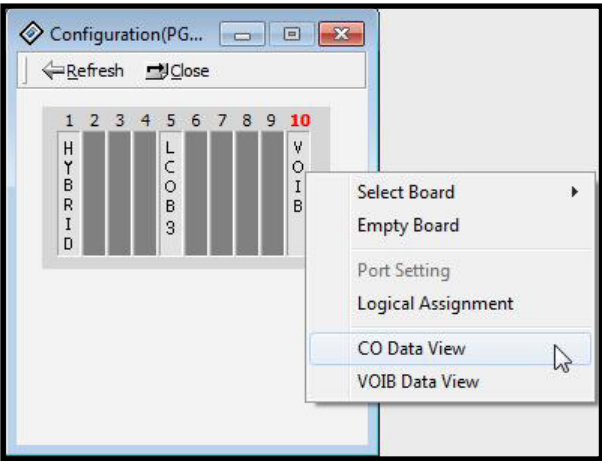
Current Station 100		Copy To DSS (PGM125)
Flex Button	Type	Value
1	{CO xx} Button	4
2	{CO xx} Button	5
3	{CO xx} Button	6
4	{CO xx} Button	7
5	Not Assigned	
6	Not Assigned	
7	{LOOP}	

For station 101 we entered the following:

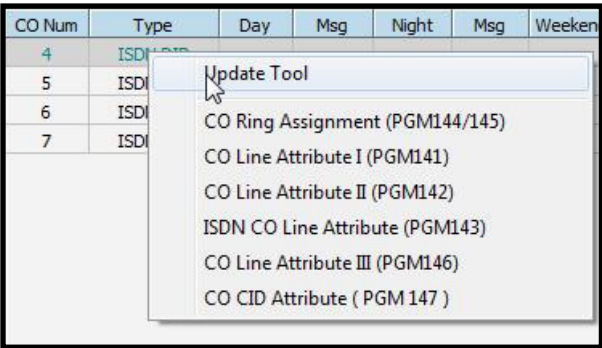
Flex Button Assignment(PGM115/125)		
Refresh Close		
Current Station 101		Copy To DSS (PGM125)
Flex Button	Type	Value
1	{CO xx} Button	1
2	{CO xx} Button	2
3	{CO xx} Button	3
4	{CO xx} Button	4
5	{CO xx} Button	5
6	{CO xx} Button	6
7	{LOOP}	

SIP Registration

Navigate to **“Pre-Programmed ▶ Configuration(PGM101-103)”**. Right click slot 10 and select **“CO Data View”**.

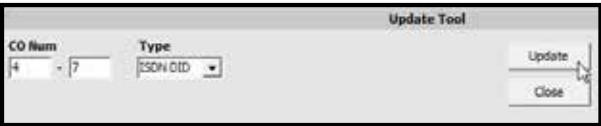


Next to **“CO Num”** enter range 4-7, then click **“Refresh”**. Right click column 4 and select **“Update Tool”**.



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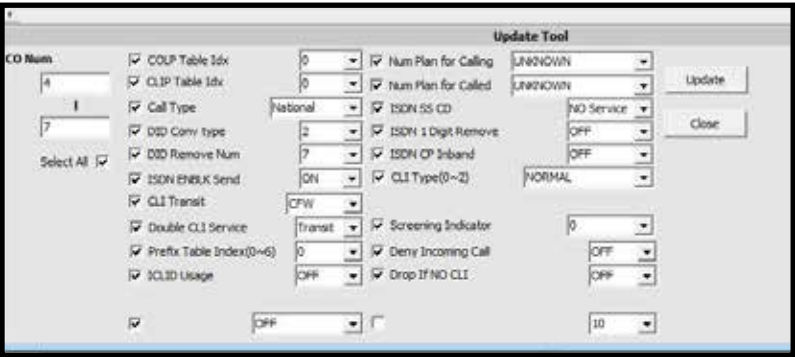
Where it says “CO Num” again enter 4-7. Under “Type” select “ISDN DID”. Click **Update** then **Refresh**.



It should look like this when finished:

CO Nums													
CO Ring Assigns													
Normal													
CO Num	Type	Day	Mng	Night	Mng	Weekend	Mng	In Denial	Mng	Signal Type	Lunch	Mng	
4	ISDN DID												
5	ISDN DID												
6	ISDN DID												
7	ISDN DID												

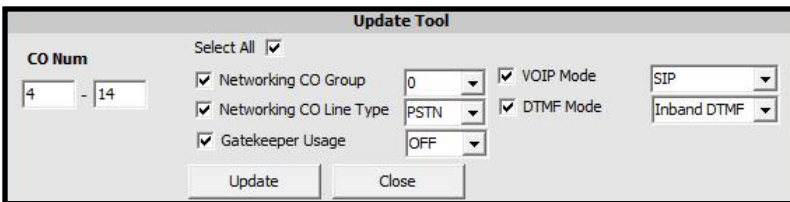
Then select “CO ISDN Attr” which is same as PGM143. Enter 4-7 next to “CO Num” and simply replicate the following values:



When done click **Update** then **Refresh**.

Now navigate to **“Network ▶ Networking CO Line Attribute(PGM322)”** and enter 4-14 next to **“CO Num”**. Right click **“CO Num 4”** and select **“Update Tool”**. Enter 4-14 next to **“CO Num”** and select **“SIP”** next to **“VOIP Mode”**.

Leave **“DTMF Mode”** to **“Inband DTMF”** as shown. The Cablevision network does not support out of band DTMF tones. You must select **“Inband DTMF”**. When done click **Update** then **Refresh**.



Update Tool

CO Num: 4 - 14

Select All ☒

☒ Networking CO Group: 0

☒ Networking CO Line Type: PSTN

☒ Gatekeeper Usage: OFF

☒ VOIP Mode: SIP

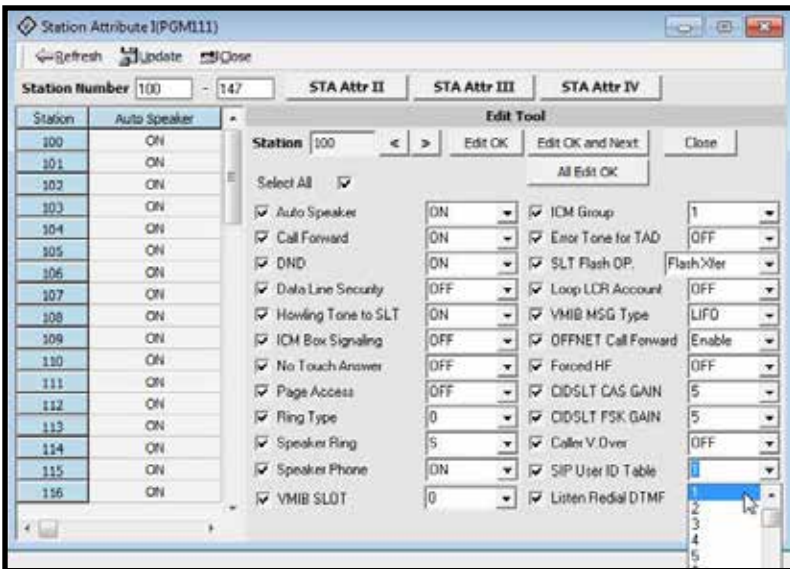
☒ DTMF Mode: Inband DTMF

Update Close

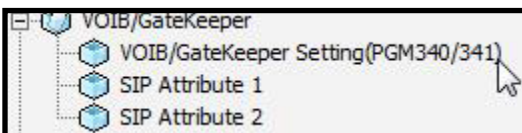
4	0	PSTN	OFF	SIP	Inband DTMF
5	0	PSTN	OFF	SIP	Inband DTMF
6	0	PSTN	OFF	SIP	Inband DTMF
7	0	PSTN	OFF	SIP	Inband DTMF
8	0	PSTN	OFF	SIP	Inband DTMF
9	0	PSTN	OFF	SIP	Inband DTMF
10	0	PSTN	OFF	SIP	Inband DTMF
11	0	PSTN	OFF	SIP	Inband DTMF
12	0	PSTN	OFF	SIP	Inband DTMF
13	0	PSTN	OFF	SIP	Inband DTMF
14	0	PSTN	OFF	SIP	Inband DTMF

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Now navigate to **“Station Base Program ▶ Station List (PGM110/111/112/113/114)”** and next to **“Station Number”** enter 100-147 then click **Refresh**. Click on Station 100. Make sure to check the **“SIP User ID Table”** on bottom left corner and to select **“1”**. You may replicate other values then click **“All Edit OK”**. Thereafter click **Update** then **Refresh**.



Now navigate to **“VOIB/GateKeeper”** and click **“VOIB/GateKeeper Setting(PGM340/341)”**.



Next to **“IP Address”** will be the address of the VOIB which will be used for SIP signaling. Next to **“Gateway Address”** will be the address of the Optimum Business SIP Trunk Adaptor. The Codec used was G.711Ulaw and next to **“VOIB”** mode make sure **“SIP”** is selected.

Due to the Cablevision DTMF network requirements, the DTMF tone duration generated by the phones and/or PBX may need to be increased from the default value of 180ms-200ms to 400ms-600ms. If you are interested in modifying DTMF durations, you can change the Jitter Buffer field under DTMF Mode below.

VOIB/GateKeeper Setting (PGM340/341)

Refresh Update Close

SIP Attr 1 SIP Attr 2

IP Address: 192.168.1.2

GATEWAY Address: 192.168.1.11

SUBNET Mask: 255.255.255.0

DNS Address: 0.0.0.0

Default Codec: G.711_Ulaw

Default Gain: 31 1 - 62

No Delay (TOS): ☐

Throughput (TOS): NORMAL

Reliability (TOS): NORMAL

Trace Password:

Firewall IP Address: 0.0.0.0

VOIB Mode: SIP

DSP Use Silence Detection: ☐

DSP Use Echo Canceler: ☒

DTMF Mode: Inband DTMF

Jitter Buffer: 150 50 - 300(msec)

Voice Monitor: ☐

GK Usage: ☐

GK Call Mode: Direct

GK Open H245: ☐

GK H245 Tunneling: ☐

GK Pregranted Arg: ☐

GK Out of Band Flash: ☐

GK Time to live(sec): 30 0 - 250

GK Address: 0.0.0.0

GK Find Address: 0.0.0.0

GK Find Port: 1718 0 - 9999

GK RAS Signal Port: 1719 0 - 9999

GK Signal Port: 1720 0 - 9999

VOIB GK ID(~23chs):

VOIB H323 ID(~23chs):

VOIB E164 Addr.(~23dgt):

VOIB Terminal Alias:

1.

2.

3.

4.

Fax Mode: ☐

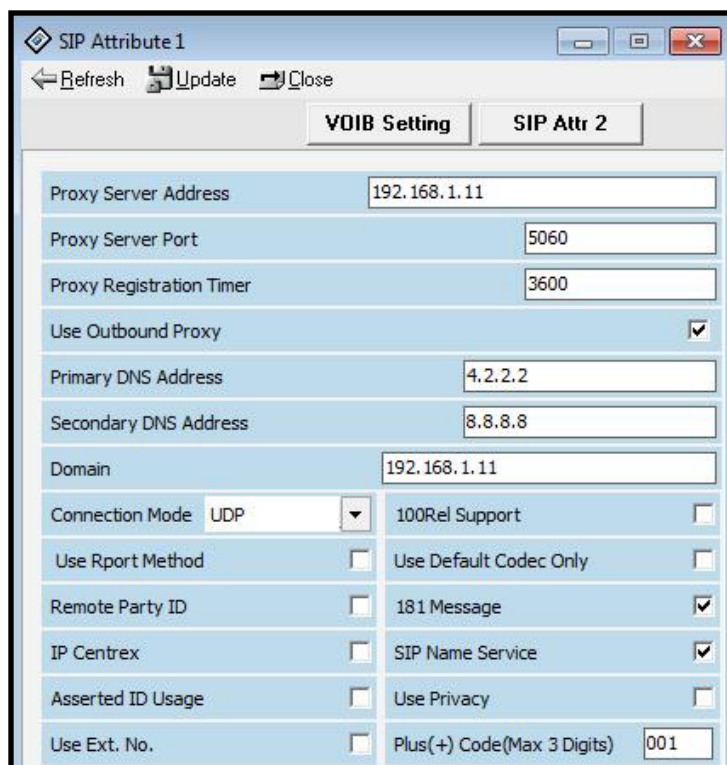
H.323 Mode: FAST

Early H.245: ☐

H245Tunneling: ☐

TOS Precedence: 0 0 - 7

Now navigate to **“VOIB/GateKeeper”** and click **“SIP Attribute 1”**. Next to **“Proxy Server Address”** and **“Domain”** enter the address of the Optimum Business SIP Trunk Adaptor. This is the address you entered in step 2 of the Optimum Business SIP Trunk Set-Up Guide. The connection type is UDP and the Server port is 5060. When done click **Update** then **Refresh**.



SIP Attribute 1	
<div> Refresh Update Close </div>	
<div> <div>VOIB Setting</div> <div>SIP Attr 2</div> </div>	
Proxy Server Address	192.168.1.11
Proxy Server Port	5060
Proxy Registration Timer	3600
Use Outbound Proxy	<input checked="" type="checkbox"/>
Primary DNS Address	4.2.2.2
Secondary DNS Address	8.8.8.8
Domain	192.168.1.11
Connection Mode	UDP
100Rel Support	<input type="checkbox"/>
Use Rport Method	<input type="checkbox"/>
Use Default Codec Only	<input type="checkbox"/>
Remote Party ID	<input type="checkbox"/>
181 Message	<input checked="" type="checkbox"/>
IP Centrex	<input type="checkbox"/>
SIP Name Service	<input checked="" type="checkbox"/>
Asserted ID Usage	<input type="checkbox"/>
Use Privacy	<input type="checkbox"/>
Use Ext. No.	<input type="checkbox"/>
Plus(+) Code(Max 3 Digits)	001

Now navigate to **“VOIB/GateKeeper”** and click **“SIP Attribute 2”**. Enter the PD next to **“Contact Number”**. Next to **“User ID Registration”** select **“Register”**. Check **“User ID Usage”** and enter 100 next to **“Asc Station”**. Just below under **“User ID”** enter the PD username followed by the address of the Optimum Business SIP Trunk Adaptor as the domain. Enter the authentication username and password in the fields that follow. The username and password must match the User ID and Password entered in the Optimum Business SIP Trunk Adaptor. This is step 3 of the Optimum Business Sip Trunk Set-Up Guide.

Index	User ID	Authentication User Name	Authentication User Password	Contact Number	User ID Registration	User ID
1	631468576@252.258.1.12:5060	631468576	631468576	631468576	Register	On
2					Provision	OFF
3					Provision	OFF
4					Provision	OFF
5					Provision	OFF
6					Provision	OFF
7					Provision	OFF
8					Provision	OFF
9					Provision	OFF
10					Provision	OFF
11					Provision	OFF
12					Provision	OFF

Update Tool

Index: 1 Contact Number: 631468576 User ID Registration: Register User ID Usage: Asc Sta Asc Sta: 100 Update Close

User ID: 631468576@252.258.1.12:5060 Authentication User Name: 631468576

Authentication User Password: Authentication User Password Repeat:

When done click **Update** then **Refresh**.

If you want to use static mode, change the **“User ID Registration”** from **“Register”** to **“Provision”**.

User ID Registration: Provision

Provision

Register

Navigate to **“ISDN System Base Program ▶ COLP Table”**. Right click Index 0 and select **“Update Tool”**. Under **“COLP Digits”** enter the authorization username you used for registration.

The screenshot shows a window titled "COLP Table (PGM201)" with a table of indices and COLP digits. The table has 15 rows, indexed 0 to 14. Row 0 is highlighted, and its COLP Digits are "6314488976". Below the table is an "Update Tool" section with a checkbox for "Update/Delete and next". Below this checkbox are two input fields: "Index" with the value "0" and "COLP Digits" with the value "6314488976". At the bottom of the "Update Tool" section are three buttons: "Update", "Delete", and "Close".

Index	COLP Digits
0	6314488976
1	
2	
3	
4	
5	
6	
7	
8	
9	
10	
11	
12	
13	
14	

Update Tool

☐ Update/Delete and next

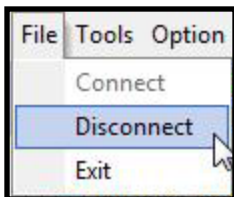
Index: 0 COLP Digits: 6314488976

Update Delete Close

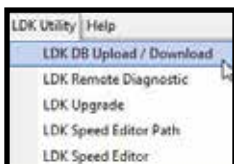
When done click **Update** then **Refresh**.

Backup/Restore

To backup or restore you will first need to disconnect from the KSU. Navigate from the top to **“File”** then click **“Disconnect”**.



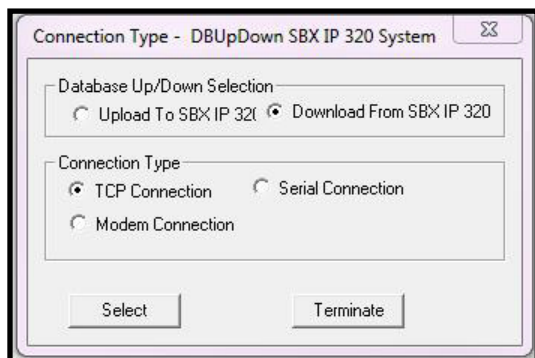
Then when the KSU is disconnected navigate from the top to **“LDK Utility”** and click on **“LDK DB Upload/Download”**.



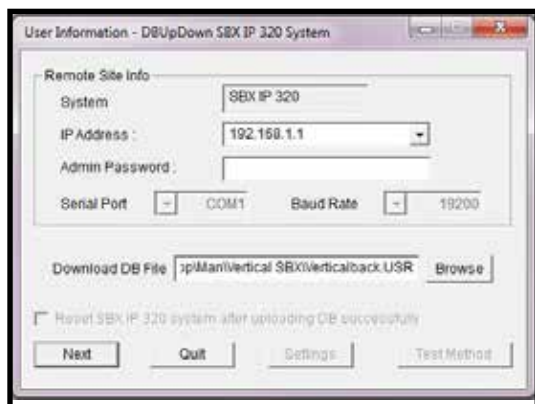
A system selection window will then appear for SBX IP 320. When you see it click **OK**.



When the second window appears, specify whether it is a download or an upload then choose the connection type, which in our case was TCP. If you want to download, select **“Download From SBX IP 320”** then select **“TCP Connection”**.



When the second window appears, type the address of the KSU and leave password blank.



Specify the directory then click **Next** and your download will begin. If you want to upload, simply select **“Upload to SBX IP 320”** and proceed in a similar manner. Be sure that .USR is the extension of your file.