

Optimum Business Trunking and the AltiGen Max1000 IP PBX version 6.7 Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Altigen 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:

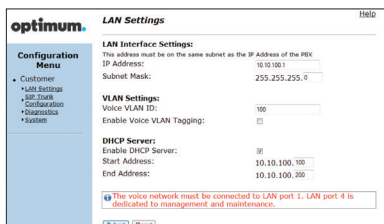
Click on the LAN Settings Link

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.

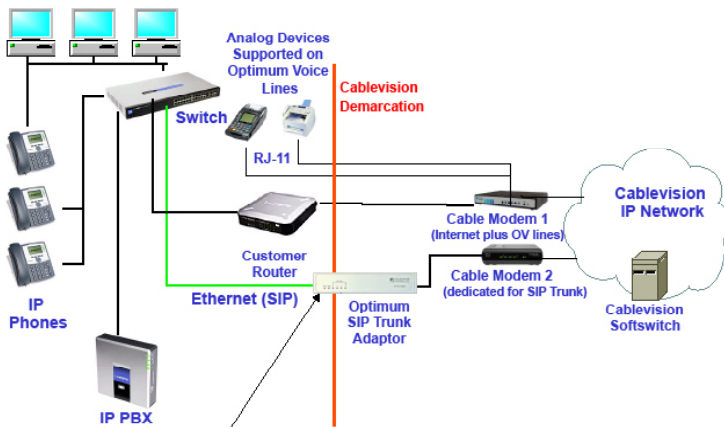
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

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



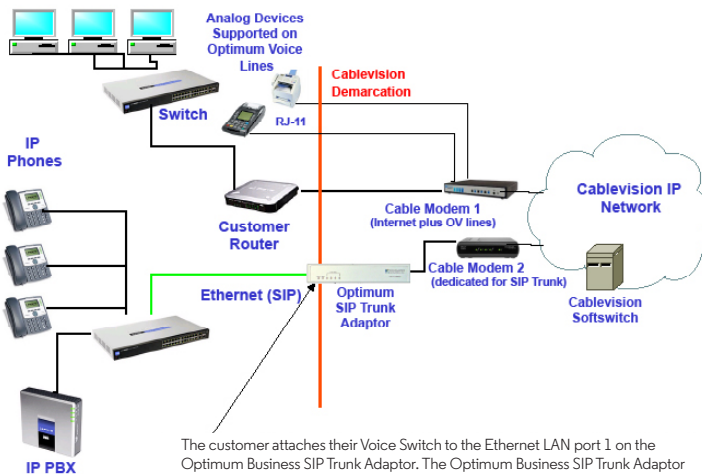
3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
4. Click 'Submit'.

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'.

Step 4:

Diagnostics Link

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

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

optimum.

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.

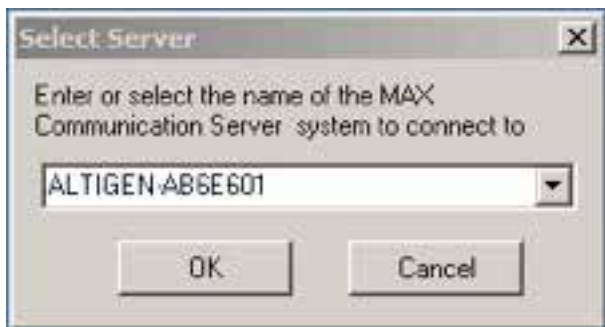
5 PBX Configuration

This configuration guide provides the steps for PBX registration mode. Static (non-registration) mode of PBX operation is not supported on the AltiGen PBX.

- AltiGen configuration GUI – Max Administrator version 6.7.0.205.
- AltiGen software version – 6.7.

The steps below describe the minimum configuration required to enable the AltiGen 6.7 PBX to use Optimum Business SIP trunking for inbound and outbound calling. Please refer to the AltiGen product documentation for more information on other advanced PBX features. The configuration described here assumes that the AltiGen is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on AltiGen Version 6.7.

1. Start the Max Administrator software application by double clicking the icon from your desktop.
2. Select the appropriate Max Communication Server system to connect to.



3. Login to the Max Administrator. The default password is "22222". The initial screen defaults to the User Configuration screen.



4. Select “System.”

- a. Select “General.”
- b. Under “Country” select U.S.A. & Canada.
- c. Under “System Home Area Code” enter the area code of the DID’s you will be using.
- d. Under “System Main Number” enter the number of your main DID (without area code).
- e. Under “System ID” select the number “1.”

Click “ok” or “Apply” to save settings.

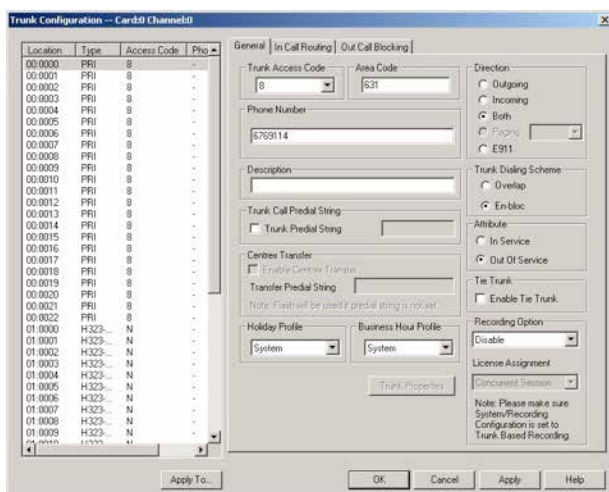
The screenshot shows the "System Configuration" dialog box with the "General" tab selected. The dialog has a title bar with a close button (X). Below the title bar is a tabbed interface with the following tabs: Account Code, Call Reports, Country Relevant, Audio Peripheral, Activity, Feature Profiles, General, Number Plan, Business Hours, Holiday, System Speed, and Call Restriction. The "General" tab is active, displaying various configuration fields:

- System ID:** A dropdown menu with "1" selected.
- Country:** A dropdown menu with "U.S.A. & Canada" selected.
- Manager Extension:** A dropdown menu.
- System Home Area Code:** A text field containing "678".
- System Main Number:** A text field containing "2384025".
- PRt Calling Number:** A checkbox labeled "Send initiator's ID in ONA and FWD call" which is unchecked.
- Operator:** A section with a label "Select an extension or group as operator" and a dropdown menu showing "Extension 258".
- Group Members:** A text area.
- Distinctive Ring:** Three checkboxes: "Enable Distinctive Ring", "Enable Operator Call Priority Ringing", and "Enable Workgroup Call Priority Ringing", all of which are unchecked.
- Conference Bridge Option:** A checkbox labeled "End Conference If No Extensions Participating" which is unchecked.
- System Call Park:** A section with a "Timeout Ring Back in" spinner set to "2" and a "Play Greeting Phrase" dropdown menu showing "phrase0401".
- Call Supervision:** A checkbox labeled "Allow Supervisor to Monitor, Barge-in, Coach, and Record agent's non-workgroup call" which is unchecked.

At the bottom of the dialog are four buttons: "OK", "Cancel", "Apply", and "Help".

5. Select “Trunk.”

- Scroll to the bottom and select “SIP-Trunk.”
- Under “Trunk Access Code” select the number “9.”
- Under “Area Code” enter the area code of the DID’s you will be using.
- Under “System Main Number” enter the number of your main DID (without area code).
- Under “Direction” select the “Both” option.



- Select “Trunk Properties” then “SIP Trunk Configuration.”
- Select a Trunk Group to configure and click “Edit.”

SIP Trunk - Id=0, Logical Channel Id=72

SIP Server IP Address: 10.10.125.1

User Name: 6782384025

Password: [masked]

Domain: 10.10.125.1

SIP Register Period: 180 Sec.

SIP Trunk Profile: Default

SIP Source Port: 5060

SIP Destination Port: 5060

☐ Automatic NAT Traversal

☒ Enable Channel

OK Cancel

h. Enter the Sip Trunk data for each field

SIP Server IP Address: Enter the IP address that was assigned to the Optimum Business Sip Trunk Adaptor. This is the IP address that was entered in step 2 of the Optimum Business Sip Trunk Set-Up Guide.

User Name: Enter the Pilot DID number. The Pilot DID should also be entered as the User ID in the Optimum Business Sip Trunk Adaptor. This is step 3 of the Optimum Business Sip Trunk Set-Up Guide.

Password: Enter the password. This password must match the password entered in the Optimum Business Sip Trunk Adaptor. This is step 3 of the Optimum Business Sip Trunk Set-Up Guide.

Domain: Enter the IP address that was assigned to the Optimum Business Sip Trunk Adaptor. This is the IP address that was entered in step 2 of the Optimum Business Sip Trunk Set-Up Guide.

SIP Register Period: Enter the registration interval to the upstream provider. Recommended: 180 Seconds.

SIP Trunk Profile: When set to the “Default” option, the Pilot DID will be used for the outbound Caller ID. This can be over ridden if the outbound Caller ID is manually configured in the phone extensions.

SIP Source Port: Set to 5060 (Commonly used for SIP traffic)

SIP Destination Port: Set to 5060 (Commonly used for SIP traffic)

Automatic NAT Traversal: Make sure this box is unchecked.

Enable Channel: Make sure this box is checked.

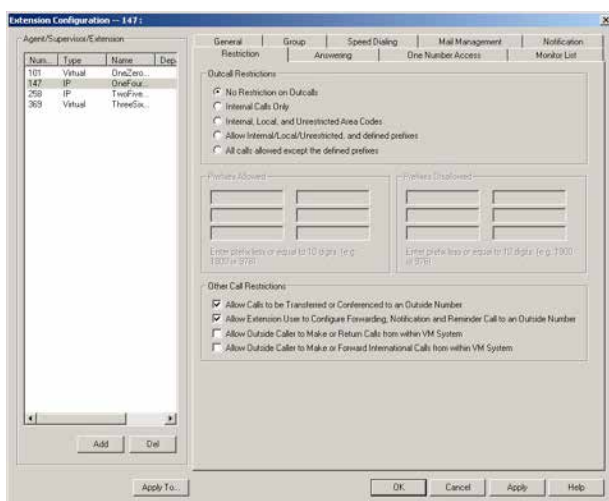
NOTE: The AltiGen only supports Registration mode. Static mode is not supported.

6. Select “Extension.”

- Select the “**General**” tab and choose the extension you would like to edit.
- Assign that extension a DID by adding the number to the “Description” and “**DID Number**” section. In this example we used **(6316769114)**.
- Under “**IP Extension**” select “Enable IP Extension” and “Dynamic IP Address.”

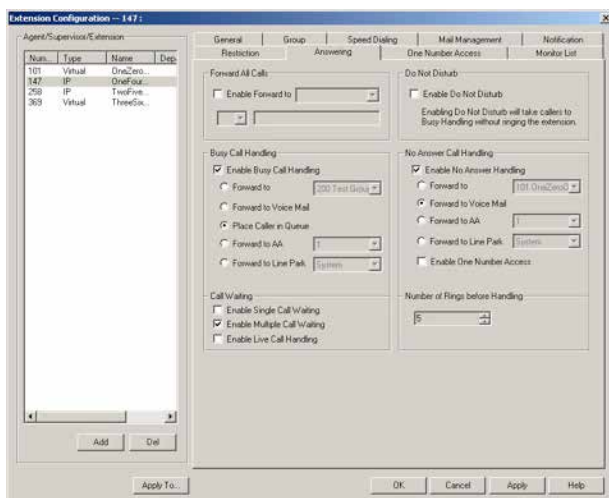
The screenshot shows the 'Extension Configuration' window for a 'Virtual' extension. The 'General' tab is selected. On the left, a table lists existing extensions: 101 (Virtual, OneZero), 142 (Virtual, OneFour), 250 (Virtual, TwoFive), and 309 (Virtual, ThreeSix). The main configuration area includes fields for Personal Information (First Name, Last Name, Password, Department, Description, DID Number, Transmitted CID, E911 CID), Account Code (Enable Forced Account Code, Override Allowed, Account Code W/ Address, For Long Distance Call Only, Block Account Code Display), Call Recording Options (License Assignment, Concurrent Session, Record, Recording tone), IP Extension (Enable IP Extension, Connect Voice Stream to Server, Dynamic IP Address, Static IP Address, Login IP Address, Home Media Server ID, Enable 3rd Party Sip Device, Enable fallback to Mobile Extension, Mobile Extension Channel), and Phone Display (Number Line, Caller Number, Name Line (IP Phone), Caller Name). Buttons for 'Add', 'Del', 'Apply To...', 'OK', 'Cancel', 'Apply', and 'Help' are at the bottom.

- d. Select the **“Restriction”** tab.
- e. Under **“Outcall Restrictions”** select “No Restriction on Outcall.”
- f. Under **“Other Call Restrictions”** select “Allow Calls to be Transferred” or “Conferenced to an Outside Number” and “Allow Extension User to Configure Forwarding, Notification and Reminder Call to an Outside Number.”



- g. Select the **“Answering”** tab.
- h. Under **“Call Waiting”** select “Enable Multiple Call Wating.”
- i. Under **“Busy Call Handling”** select “Enable Busy Call Handling” and “Place Caller in Queue.”
- j. Under **“No Answer Call Handling”** select “Enable No Answer Handling” and “Forward to Voice Mail.”

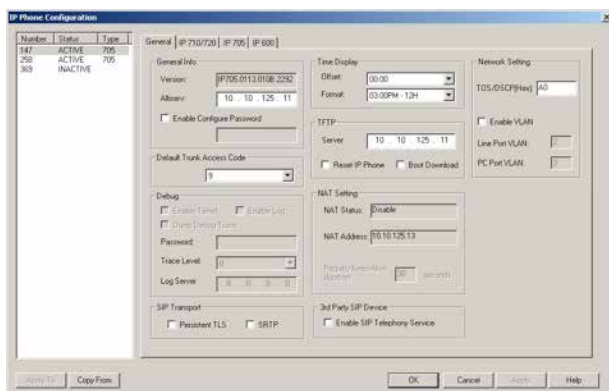
Click “Ok” or “Apply” to save settings.



7. Select "IP Phone."

- Select the extension you would like to configure.
- Under **"General Info"** put in the PBX IP address in the box next to "Altiserv." In this example we used (10.10.125.11).
- Under **"TFTP"** put in the PBX IP address in the box next to "Server." In this example we used (10.10.125.11).
- Under **"Default Trunk Access Code"** select the number "9."

Click "Ok" or "Apply" to save settings.



- For Call Forwarding go to “Extension” select the “Answering” tab and select the extension you would like to configure. Under “Forward All Calls” select “Enable Forward to” and select “Outside Number.” Now enter the number you would like to forward to in the box below.
- For Call Park, while on a call hit the “Flash” button and enter #41. This will place the call into system park. To pick up the call, from any phone in system dial #41.

Important

Inband DTMF:

The Cablevision network only supports inband DTMF tones. The AltiGen PBX only supports sending out-of-band DTMF tones. In order for the AltiGen PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be enabled to convert out-of-band DTMF tones sent by the AltiGen PBX to inband DTMF tones. To enable this conversion, log into the Optimum SIP Trunk Adaptor and click on the Convert Inband DTMF checkbox, and click the Submit button to update this setting. This is step 3 in the Optimum Business SIP Trunk Set-Up Guide.

DTMF Tone Duration:

The DTMF tone duration generated by the phones needs to be increased from the default value of 180ms-200ms to 600ms. The AltiGen PBX does not have access to change the DTMF settings on the PBX, you must change it on each phone.