

Optimum Business Trunking and the Grandstream UCM6102 IP- PBX Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Grandstream UCM6102 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



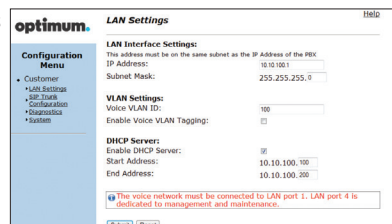
Step 2:

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.

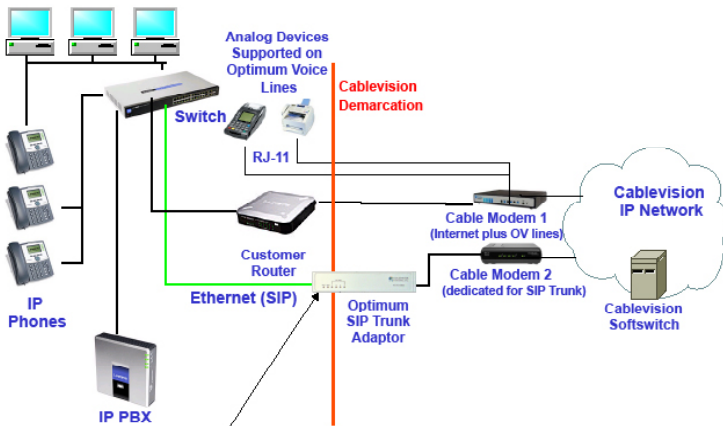
Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



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.

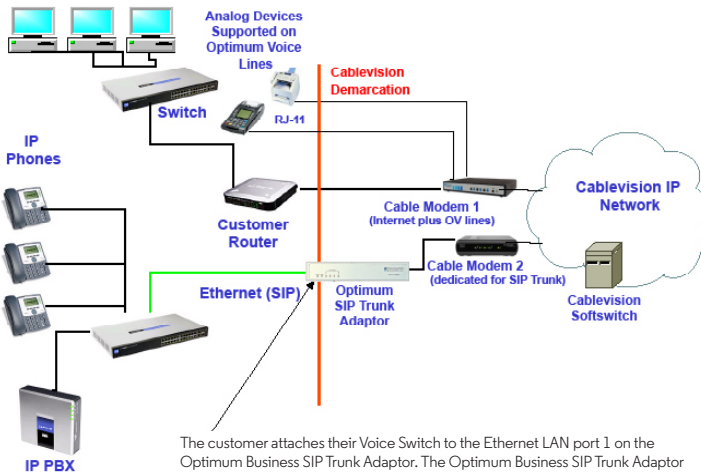
4. Click 'Submit'.

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'.

The screenshot shows the 'SIP Trunk Configuration' page. On the left is a 'Configuration Menu' with links: Customer, LAN Settings, SIP Trunk Configuration (highlighted), 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'. At the bottom, there is a 'DID's' list with a scrollable view of numbers: 0164030809, 0164030760, 0164030769, 0164030765, and 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 the previous screenshot, with 'Diagnostics' (highlighted) and 'System' as options. The main content area has a heading '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:' (0164030809) 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 buttons for 'Enabled' (selected) and 'Disabled'. 3. 'Ping Test:' with a description 'IP Address to Ping:' and a 'Ping' button. Below that is a '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



Configuration Menu

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

System
[Help](#)

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



Configuration Menu

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

Set Password
[Help](#)

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 knowledge base solution provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.

The steps below describe the basic configuration required to enable the Grandstream UCM6102 IP-PBX to use Optimum Business SIP trunking for inbound and outbound calling. Please refer to the Grandstream UCM6102 documentation for other advanced PBX features.

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Grandstream UCM6102 version 1.0.4.7.

PBX Information

Manufacturer:	Grandstream Networks
Model:	UCM6102
Software Version:	1.0.4.7
Does the PBX send SIP Registration messages (Yes/No)?	Yes

6 Network Access

The default IP of the LAN port of the PBX was 192.168.2.1 /24 and was unchanged. As for the WAN port, it was given a static IP address of 10.10.154.11 /24 with the Optimum Business SIP Trunk Adaptor being 10.10.154.1 /24. SIP traffic flow was via the WAN port.

The default username/password of the device was admin/admin. The IP address of the PC managing the device should fall in network 192.168.2.0/24. Once configured with an appropriate PC IP, the device can be configured via a Web browser through its LAN IP address.



The image shows a login interface with a dark blue background. It features two input fields: a yellow one for the username 'admin' and an orange one for the password, which is masked with six dots. Below these fields is a blue 'Login' button. To the right of the button is a language selection menu showing 'English' with a downward arrow.

To configure network settings navigate to **Settings ▶ Network Settings ▶ Basic Settings**. Under **Basic Settings** next to **Method** select **Route**. Under **WAN**, first enter **Static** next to **IP Method**. Enter the address of the Optimum Business SIP Trunk Adaptor next to **Gateway IP** and the address of the PBX next to **IP Address**. Under **LAN**, enter the address of the PBX's LAN port next to the **IP Address** field. For the phones, the DHCP range must be specified in the **Allow IP Address From** and **Allow IP Address To** fields.

Network Settings

- Basic Settings
- 802.1X
- Port Forwarding

Firewall

Change Password

LDAP Server

HTTP Server

Email Settings

Time Settings

NTP Server

Basic Settings

Method: Route

Preferred DNS Server:

WAN

IP Method: Static

Gateway IP: 10.10.154.1

Subnet Mask: 255.255.255.0

IP Address: 10.10.154.11

DNS Server 1: 8.8.8.8

DNS Server 2: 4.2.2.2

Layer 2 QoS 802.1Q/VLAN Tag: 0

Layer 2 QoS 802.1p Priority Value: 0

LAN

IP Address: 192.168.2.1

Subnet Mask: 255.255.255.0

DHCP Server Enable: ☒

DNS Server 1: 8.8.8.8

DNS Server 2: 208.67.222.222

Allow IP Address From: 192.168.2.100

Allow IP Address To: 192.168.2.254

Default IP Lease Time: 43200

Cancel

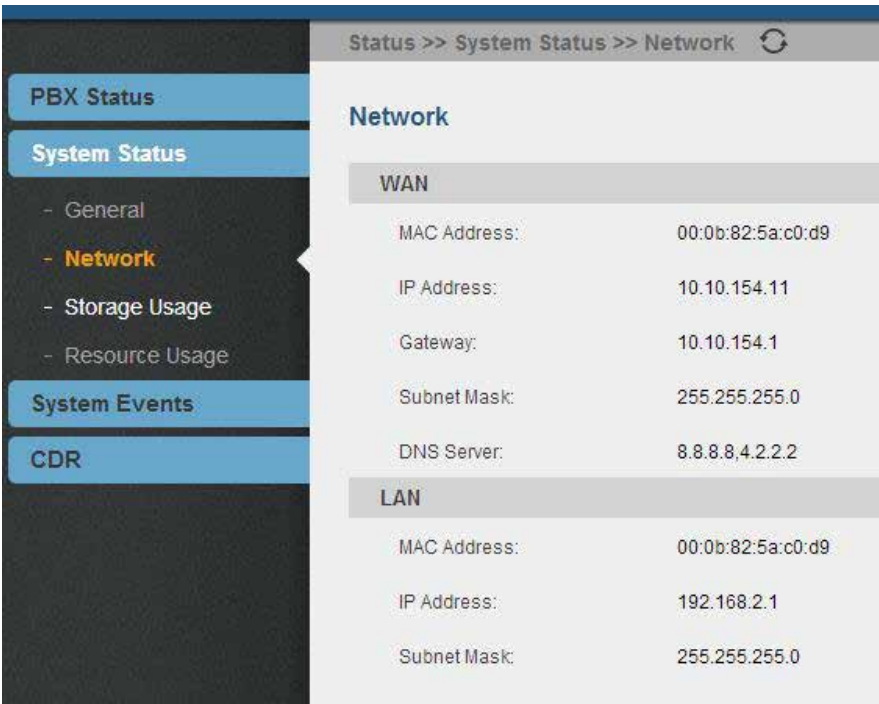
Save

When done click **Save**. Thereafter there will be a prompt to restart the device for the network settings to take effect.



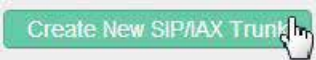
Click **OK** and upon restart the new settings will take effect.

Once the system has restarted, the new network configuration can be viewed by navigating to **Status ▶ System Status ▶ Network** as shown.



7 SIP Trunk

To register the Optimum Business SIP Trunk Adaptor to the PBX, navigate to **PBX ▶ Basic/Call Routes ▶ VoIP Trunks** and click on **Create New SIP/IAX Trunk**.



Select **Register SIP Trunk** next to **Type**. Enter a name for the Optimum Business SIP Trunk Adaptor next to **Provider Name** (EM-4552 was use in this example). The **Host Name** field should follow the address of the Optimum Business SIP Trunk Adaptor. **Keep Trunk CID** was left unchecked and is optional. If checked the Pilot DID will be the Caller ID for the calls. The **Username**, **Password** and **AuthID** will be the given credentials between the PBX and the Optimum Business SIP Trunk Adaptor. The **Outbound Proxy** should also be the address of the Optimum Business SIP Trunk Adaptor. **Auto Record** is optional.

Create New SIP/IAX Trunk

More details will be shown when editing trunk.

Type:

Register SIP Trunk

① Provider Name:

EM-4552

① Host Name:

10.10.154.1

① Keep Trunk CID:

☐

① Username:

4085551234

Password:

① AuthID:

4085551234

① Outbound Proxy:

10.10.154.1

① Auto Record:

No

Cancel

Save

When done click **Save** then **Apply Changes** from above.



For Static Mode select **Peer SIP Trunk** instead next to **Type**. Enter a name for the Optimum Business SIP Trunk Adaptor next to **Provider Name** and enter the address of the Optimum Business SIP Trunk Adaptor next to **Host Name** as shown.

More details will be shown when editing trunk.

Type: Peer SIP Trunk

Provider Name: EM-4552

Host Name: 10.10.154.1

Keep Trunk CID: ☐

Auto Record: No

Cancel Save

When done click **Save** then **Apply Changes** from above.

8 DID/Extensions

To provision phones and assign extensions, navigate to **PBX ▶ Basic/Call Routes ▶ Extensions** and click **Create New User**. Here the basic settings for a phone can be configured. Under **General**, the **Extension** field specifies the extension for the phone. **CallerID Number** and **CallerID Name** will be the DID number that will be mapped to this extension. **Permission** is the permission level the user will have for outgoing calls. **Internal** was used for all extensions. Under **Technology** check **SIP**. Under **SIP Settings** enter the user's DID next to **AuthID** and set **DTMF Mode** to **Inband** as shown. All other settings can be configured according to preference.

Note: DTMF tone duration cannot be configured on the PBX.

When done click **Save** then **Apply Changes** from above.

The devices can also use Auto Provisioning where the PBX will discover a device upon boot up, automatically assign it an extension, and return a URL of the config file for the device to download.

To do this, from **Basic/Call Routes** navigate to **Zero Config** and click **Auto Provision Settings**. Once here check **Enable Zero Config**. Check **Automatically Assign Extension** and specify the beginning extension next to **Start extension**. A default password can also be set next to **Default Password**.

Auto Provision Settings

Auto provision automatically provides an extension to the device. There are three methods of auto provision: SIP SUBSCRIBE, DHCP Option 55 and mDNS.

For example, when the device boots up, it will send SIP SUBSCRIBE multicast in the LAN. The PBX will find it, create an account and return a URL of the config file for the device to download.

Enable Zero Config:

☒

① Automatically Assign Extension:

☒

① Start Extension:

① Enable Pick Extension:

☐

① Extension Segment:

-

① Pick Extension Period (hour):

① Generate Random Password:

☐

① Default Password:

Cancel

Save

When done click **Save** then **Apply Changes** from above.

By clicking on Auto Discover, the PBX can automatically discover the device using a specific Scan Method of **Ping**, **ARP**, or **SIP-Message**. Once that is selected the **Scan IP** field can be filled to scan the entire network segment by entering **255**.

Auto Discover

The PBX can automatically discover the new devices by ARP or PING. It can scan the entire network segment or a single IP address.

① Scan Method:

① Scan IP:

- - -

Cancel

Save

When done click **Save** and a message will be presented prompting to check the result.

Prompt information

Scan Done! Do you want to check the result now?

Cancel

OK

Click **OK** and the result will be displayed.

No.	MAC Address	IP Address	Extension	Version	Model	Connection Status	Create Config	Actions
1	B82E2D40F	192.168.2.100	extension 1-100		Grandstream GSP 1400 v2.0.0.0	Connected	Yes	  

A user may also manually enter device information prior to performing the Auto Discover step above. This can be done by clicking on **Create New Device**. Here an extension may be entered along with a MAC and IP Address of the device.

Create New Device

Enable Hot-Desking:

☐

① MAC Address:

① IP Address:

① Version:

① Model:

① Account Select:

Account 1: None

Cancel

Save

When done click **Save** then **Apply Changes** from above.

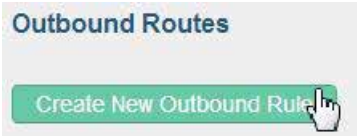
By going back to **Extensions** the status of this discovered phone can be viewed. It now shows green which implies that it is registered and free.

IP Status	Extension	Label Name	Technology	Phone Port	Actions
	100	EXT1000000	SIP	192.168.2.100:1080	  

When finally done click **Save** then **Apply Changes** from above.

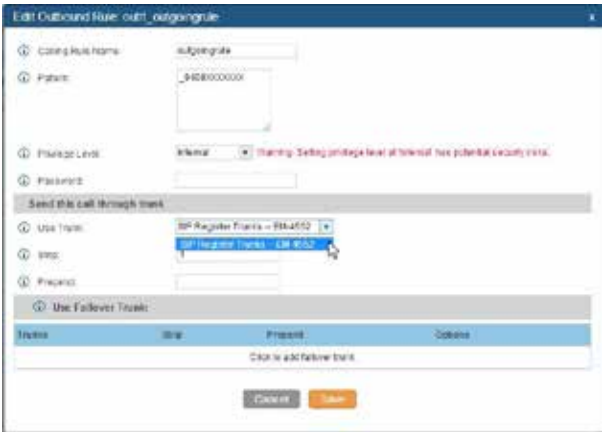
9 Dial Plan

To configure Dial Plan, Outbound and Inbound routes need to be configured. First navigate to **PBX ▶ Basic Call Routes ▶ Outbound Routes** and click **Create New Outbound Rule**.



Initially, a rule name needs to be given. Next to **Pattern** is where the dial pattern needs to be entered. All patterns are prefixed by “_”: Entering “[12345]” allows any single digit within these brackets. “N” allows any digit between 2 and 9. The “.” character matches one or more digits. The “!” character matches zero or more digits. “X” allows any digit between 0 and 9 and finally “Z” allows any digit between 1 and 9.

In this example “9408XXXXXX” was entered to allow all numbers beginning with area code 408. The digit “9” was entered in the beginning and will be stripped by entering “1” next to **Strip. Internal** was selected next to **Privilege Level** and next to **Use Trunk** the trunk for the Optimum Business SIP Trunk Adaptor was selected, in this case “EM-4552”.



When done click **Save** then **Apply Changes** from above. Other outbound rules may be configured in a similar manner.

As for inbound rules, click on **Inbound Routes** and then **Create New Inbound Rule**.



First select the trunk for the Optimum Business SIP Trunk Adaptor next to **Trunks**. With the same character values as outbound rules, enter a pattern next to **DID Pattern**. In this example, “4085555556” was entered to permit a call to extension 101. **Internal** was again selected next to **Privilege Level**. **Extension** was selected next to **Default Destination** following the extension this rule is to be forwarded to, in this case 101. Other extensions may be configured similarly.

Create New Inbound Rule

Trunks:

SIP Register Trunks -- EM-4552

① DID Pattern:

4085555556 /

① Privilege Level:

Internal

① Default Destination:

Extension

101

Time Condition:

Time	Destination	Options
Click to add Time Condition		

Cancel

Save

When done click **Save** then **Apply Changes** from above.

10 Auto Attendant

To configure Auto Attendant navigate to **PBX ▶ Call Features ▶ IVR** and click on **Create New IVR**. Enter a name next to **Name** and a valid extension number next to **Extension**. **Welcome Prompt** may be left as is or manually created by clicking on **Prompt**. A customized recording may be uploaded here. Below under **Key Pressing Events** is where an extension can be matched to a specific digit. In this example digit 1 was mapped to extension 100. The rest of the fields may be configured according to preference. When done click **Save** then **Apply Changes** from above.

Edit IVR : Sally

1

Name

Sally

1

Extension

7000

1

Dial Other Extensions

☐

1

Dial Trunk

☐

1

Permission

Internal

1

Welcome Prompt

welcome

Prompt

1

Digit Timeout

5

1

Response Timeout

10

1

Response Timeout Prompt

ivr-create-timeout

1

Invalid Prompt

invalid

1

Response Timeout Repeat Loops

3

1

Invalid Repeat Loops

3

1

Language

Default

Key Pressing Events

Press 0

Select an Option

Press 1

Extension

Extension - 100

Press 2

Select an Option

Press 3

Select an Option

Press 4

Select an Option

Press 5

Select an Option

Press 6

Select an Option

Press 7

Select an Option

Press 8

Select an Option

Press 9

Select an Option

Press *

Select an Option

Cancel

Save

11 Backup/Restore

To back up or restore the configuration file, navigate to **Maintenance ▶ Backup ▶ Local Backup**. Here the option of **Create New Backup** and **Upload Backup File** will be displayed. If **Create New Backup** is selected, a window will prompt for a file name. When entered simply click **Backup**.



If **Upload Backup File** is selected, a window will prompt for a file to upload. When entered simply click **Upload**.

