



NetVanta Unified Communications Technical Note

Installing and Configuring the Grandstream BudgeTone 101/102

Introduction

The BudgeTone Series offers a family of affordable, next generation SIP phones that feature excellent audio quality and rich telephony features. The BudgeTone Series supports popular voice codecs and is designed to be fully interoperable with third party SIP providers. The BudgeTone phones have upgraded features including 3-way conferencing, full-duplex hands-free speakerphone (BT200), voice mail with indicator and custom ring tones. The BudgeTone SIP Phone Series is an affordable, easy to use VoIP solution for the home or SoHo office.

The BudgeTone 101/102 models require manual configuration.

Known Integration Issues

- BudgeTone 100 series phones cannot be auto-detected or auto-configured.
- The G722 codec causes audio issues. This codec does not work with Polycom phones and will result in poor audio. You must remove this codec manually if you use a BT100 series phone.

Upgrading the Firmware

Refer to the feature comparison matrix in *UC Server Interoperable SIP Device Features Comparisons* technical note, available online at <http://kb.adtran.com>, to determine the most recently supported version of firmware for the Grandstream telephones. To determine what version of firmware the Grandstream phone is using, perform the following steps:

1. Power on the phone.
2. Find the IP address of the phone:
 - Press **MENU**.
 - Press the down arrow until you see **[2] IP Addr.**
 - Press **MENU** again and the phone will display the IP address.*03

NOTE: The IP address may be displayed with extra characters or in an unusual format. For example, if it shows **192.168008133** the actual IP address would be **192.168.8.133**.

3. Enter the IP address in your Web browser.
4. When you see the password prompt, enter **admin** as the password and select **Login**.
5. Select the **Status** tab. The firmware version is listed beside **Program version**.

Should the firmware version on the phone be earlier than the version noted in [UC Server Interoperable SIP Device Features Comparisons](#) technical note, then the firmware should be upgraded. The firmware can be upgraded manually as described in the section below.

Manually Updating Firmware

1. Power on the phone.
2. Find the IP address of the phone:
 - Press **MENU**.
 - Press the down arrow until you see **[2] IP Addr**.
 - Press **MENU** again and the phone will display the IP address.

NOTE: The IP address may be displayed with extra characters or in an unusual format. For example, if it shows **192.168008133** the actual IP address would be **192.168.8.133**.

3. Enter the IP address in your Web browser.
4. When you see the password prompt, enter **admin** as the password and select **Login**.
5. Select the **Advanced Settings** tab.
6. Under **Firmware Upgrade and Provisioning** enter the following:
 - Choose Upgrade via **TFTP**
 - **Firmware Server Path:** enter the IP address of your UC server
 - **Allow DHCP Option 66 to override server:** No
7. Select **Update** and reboot when prompted.
8. Wait 10 minutes for the firmware to update. While the upgrade is in progress, you may see the message waiting indicator light up periodically and some of the function keys on the phone may not be usable. Do not disrupt the firmware update during this time.

Configuring Grandstream BT101/102 Series Phones

Gathering Information

To configure a BT101 or BT102 phone, you will need the following information.

SIP User ID: This is equivalent to the identity address (typically an extension number) you wish to associate with the Grandstream phone.

Authenticate ID: This is the SIP authentication identifier associated with the above identity. This is required by any SIP end-point to register with the SIP PBX.

Authenticate Password: This is the SIP authentication password associated with the above identity. This is typically the same value as your voice mail access PIN.

Determining the Authenticate ID and Password as the User (that owns the identity)

To determine the Authenticate ID and Password:

1. Launch the UC client.
2. Log in as the user you wish to associate to the phone.
3. In the left bottom pane, take note of the identity name.



4. Select the icon on the right and select **SIP Authentication**.

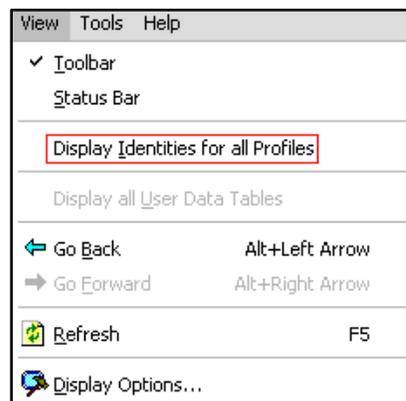


5. Record the **User/login name** and **Password** from the following dialog box. You will need them later when you configure the phone.



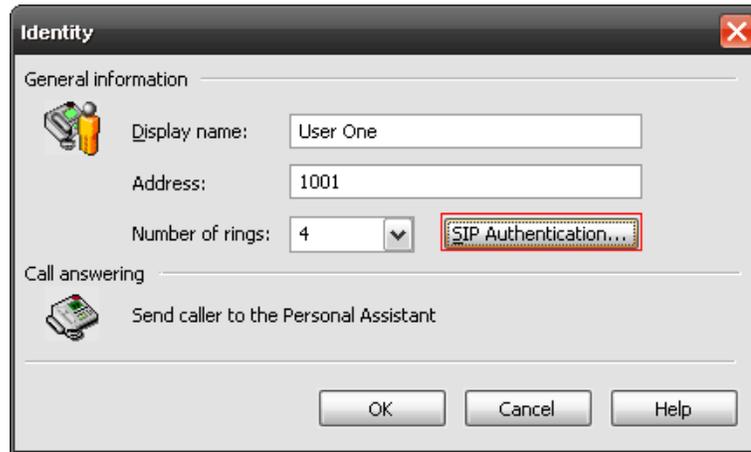
Determining the Authenticate ID and Password as the Administrator

1. Launch the UC client.
2. Log in as the admin user.
3. Select the **Identities** tab in the left pane.
4. Go to **View > Display Identities for all Profiles**.



5. Find the identity in the list that you wish to use and double-click on the entry.

6. Select **SIP Authentication**.



7. Record the **User/login name** and **Password** from the following dialog box. You will need them later when you configure the phone.



Configuring Phones

The BT101/102 series telephones are configured through the Web interface of the phone. From there, you can configure an identity and set options for voice mail.

To configure the phone:

1. Find the IP address of the phone:
 - Press **MENU**.
 - Press the down arrow until you see **[2] IP Addr**.
 - Press **MENU** again and the phone will display the IP address.

NOTE: The IP address may be displayed with extra characters or in an unusual format. For example, if it shows **192.168008133** the actual IP address would be **192.168.8.133**.

2. Enter this IP address in your Web browser.
3. When you see the password prompt, enter **admin** as the password and select **Login**.
4. Navigate to the **advanced settings** tab and enter the following:

SIP Server:	<IP Address of the UC server>
SIP User ID:	<SIP User ID> (see Gathering Information on page 3)
Authenticate ID:	<Authenticate ID> (see Gathering Information on page 3)
Authenticate Password:	<Authenticate Password> (see Gathering Information on page 3)
Name:	<Identity Name> (see Gathering Information on page 3)
Preferred Vocoder:	Change G722 under choice 7 to PCMU .
Voice Mail UserID:	*4386245
Send DTMF:	Check off <i>via RTP (RFC2833)</i>
Enable Call Features:	This allows you to execute features such as call forwarding using star codes. This should be disabled unless you need to use call forwarding. If this feature is enabled, it will cause a conflict with the UC server star codes and you must change the dial plan. See Changing Dial Plan Entries in the UC Server on page 8 for instructions on changing the dial plan.

5. Select **Update** and then **Reboot** when prompted.

The following figure depicts an example of how to complete the required fields.

Grandstream Device Configuration

STATUSBASIC SETTINGSADVANCED SETTINGS

Admin Password: (purposely not displayed for security protection)

SIP Server: (e.g., sip.mycompany.com, or IP address)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

Advanced Options:

Preferred Vocoder: (in listed order)

choice 1:	<input pcmu\""="" type="text" value="current setting is \"/>
choice 2:	<input pcma\""="" type="text" value="current setting is \"/>
choice 3:	<input g723\""="" type="text" value="current setting is \"/>
choice 4:	<input g729\""="" type="text" value="current setting is \"/>
choice 5:	<input g726-32\""="" type="text" value="current setting is \"/>
choice 6:	<input ilbc\""="" type="text" value="current setting is \"/>
choice 7:	<input type="text" value="PCMU"/>
choice 8:	<input pcmu\""="" type="text" value="current setting is \"/>

Voice Mail UserID: (User ID/extension for 3rd party voice mail system)

Enable Call Features: No Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)

Send DTMF: in-audio via RTP (RFC2833) via SIP INFO

Changing Dial Plan Entries in the UC Server

The Grandstream phones use star codes for some call features such as call forwarding and caller ID options. This may interfere with some of the dial plan entries in the UC server. The dial plan entry *82, the shortcode used to logout of a UC server ACD queue, is one that conflicts with the Grandstream call features and must be changed in the UC server to accommodate both functions. The Grandstream phone feature access codes are disabled by default; however, if they are enabled, then you must follow the instructions below.

To change the dial plan entries:

1. Navigate to **Communication Systems > UC Server > Routing**.
2. Find the entry with [*]82.* as the digits and double click to open the properties.
3. In the original digits field, change 82 to another number such as 84. For consistency, you may wish to change the star codes for the other call queuing star codes.

Dial Plan Entry

Routing rule

Original digits: [*]84.*

Description: Advanced Call Queue - Logout

Priority: 30

Destination

Gateway:

Host: pbxservices

Call next member after 0 seconds

Digit manipulation

Digits to skip: 3

agentlogout- Dialect number

Prefix to add Suffix to add

Options

Transport: udp

Source pattern: .*

OK Cancel Help