



INTUITIVE VOICE
TECHNOLOGY

Grandstream GXP2000
Phone Setup
User Guide

Version 1.0

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Overview

This guide provides an overview of setting up the Grandstream GXP2000 telephone.

General Support

If you have questions or need technical assistance with Evolution PBX call (888) 587-5750 or send an Email to support@intuitivevoice.com.

If you have questions about third party hardware or software please visit the following Websites:

- Grandstream IP Phone: <http://www.grandstream.com>
- Snom VoIP Wireless Telephone: <http://www.snom.com>
- Snap: <http://www.snapanumber.com>.
- iView: <http://www.i9technologies.com/>.
- Cisco VoIP Telephones: <http://www.cisco.com>.
- CounterPath Software Telephones - <http://www.counterpath.com>.
- Digium PCI Cards: <http://www.digium.com>.
- Linksys Analog Terminal Adapters and VoIP telephones: <http://www.linksys.com>.
- Polycom VoIP Telephones: <http://www.polycom.com>.

My Evolution PBX Doesn't Do That!

Some features and configuration options in this document may only be available with a recent software upgrade available from Intuitive Voice Technology. Email support@intuitivevoice.com to learn more about Evolution PBX upgrades (please explain the feature that you are interested in).

Errata

Please report errors or confusing descriptions by sending an Email to support@intuitivevoice.com.

Setting Up the Grandstream GXP2000 Telephone

The GXP2000 is an easy to use 4-line enterprise SIP telephone that supports integrated power-over-Ethernet. Expandable, secure and easy to manage, the GXP2000 offers superior audio quality, 4 individual SIP accounts, 7 programmable keys, visual message indicator, full duplex hands-free speakerphone, dual 10M/100Mbps Ethernet ports, intuitive user interfaces, large back-lit graphical LCD display with support for multiple languages, security and privacy protection, screen content customization using XML, automated phone book synchronization with directory server using XML.

Configuring the Grandstream GXP2000 in Evolution PBX

To Configure a Basic Extension for the Grandstream GXP2000 in Evolution PBX

1. Login to **Evolution PBX**, click the **Resources** tab then click **Phone Extensions**.
2. In the **New Device** field, select **Grandstream-GXP2000** and the Modify Phones window displays.

The screenshot displays the Evolution PBX web interface. At the top, there is a navigation bar with tabs for RESOURCES, CALL ROUTING, ADMINISTRATION, and REPORTING. Below this is a 'Phone Extensions List' table with columns for Extension, Type, Caller ID, MAC Address, Status, and Modify. The table contains several rows of data, including extensions 5001, 5003, 5007, 5008, 5015, 5710, 5711, and 5712. Below the table is a 'New Device' dropdown menu set to 'Select Type'. The 'Modify Phones' configuration screen is visible, divided into three sections: General, Phone Settings, and Settings. The General section includes fields for Extension (5000), Caller ID Name (Name), Caller ID Number (5551231234), Record Calls (No), and Parent Extension (Master). The Phone Settings section includes Phone Type (-GENERIC SIP-), MAC Address (not needed), Mailbox (Personal), Time Zone (Pacific), Paging (Disabled), and Presence (Disabled). The Settings section includes Remote Phone (No), IP Address (For Future Use), Heartbeat (2000), Outbound (Internal-Local-Toll), Codec (G711 (USA)), and Button Map (Default). A 'Save' button is located at the bottom right of the configuration screen.

Extension	Type	Caller ID	MAC Address	Status	Modify
5001	Polycom-50X	"Ron Home" <6022495750>	0004E2032F43	OK (82 ms) N/A	[Pencil] [X] [Globe]
5003	-GENERIC SIP-	"Snom Wireless" <6022495750>	not needed	OK (4 ms) N/A	[Pencil] [X] [Globe]
5007	-GENERIC SIP-	"Zach Softphone" <6022495750>	not needed	UNKNOWN N/A	[Pencil] [X] [Globe]
5008	-GENERIC SIP-	"Bill Softphone" <6022495750>	not needed	UNKNOWN N/A	[Pencil] [X] [Globe]
5015	-GENERIC SIP-	"Ron Softphone" <5551231234>	not needed	UNKNOWN N/A	[Pencil] [X] [Globe]
5710	Polycom-60X	"Chris Home" <6022495750>	0004E2053363	OK (101 ms) N/A	[Pencil] [X] [Globe]
5711	-GENERIC SIP-	"Chris Mobile" <6022495750>	not needed	UNKNOWN N/A	[Pencil] [X] [Globe]
5712	-GENERIC SIP-	"Chris Mobile2" <6022495712>	not needed	UNKNOWN N/A	[Pencil] [X] [Globe]

3. In the **Extension** field, select an available extension.
4. In the **Caller ID Name** field, enter the name you want to be passed as this extension's caller ID text.
5. In the **Caller ID Number** field, enter the number you want to be passed as this extension's caller ID number.
6. In the **Time Zone** field, select the appropriate time zone for the time zone where the phone will be located.
7. In the **Remote Phone** field, select **Yes** if the phone will not be on the same subnet as the Evolution PBX server (for example, if the phone belongs to a user that is located at a remote location). Select **No** if this is not the case.
Note: The other options on the Modify Phones frame are optional or advanced.
8. Click **Save**.
Note: The Evolution PBX Web configuration interface requires that your browser's popup blocker be turned off or disabled so that the Device Settings screen can be seen.
9. Write down the device settings for your new extension that are displayed on the screen. These settings will be used to configure your phone.

Configuring the Grandstream GXP2000 Telephone

To Configure the Grandstream GXP2000 Telephone

1. Connect the **Grandstream GXP2000** to your Ethernet network.
2. The Grandstream GXP2000 automatically acquires an IP Address from your network.
3. Write down the IP Address displayed on the phone, open a Web Browser then enter the IP Address (for example, http://192.168.2.150) into the Address bar.
4. Enter the default password (admin) then click **Login** or press **Enter** and the ADVANCED SETTINGS tab of the Grandstream Device Configuration window displays.

Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **ACCOUNT 1** **ACCOUNT 2** **ACCOUNT 3** **ACCOUNT 4** **EXT 1** **EXT 2**

Admin Password: (purposely not displayed for security protection)

G723 rate: 6.3kbps encoding rate 5.3kbps encoding rate

iLBC frame size: 20ms 30ms

iLBC payload type: (between 96 and 127, default is 97)

Silence Suppression: No Yes

Voice Frames per TX: (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)

Layer 3 QoS: (Diff-Serv or Precedence value)

Layer 2 QoS : 802.1Q/VLAN Tag 802.1p priority value (0-7)

No Key Entry Timeout: (in seconds, default is 4 seconds)

Use # as Dial Key: No Yes

local RTP port: (1024-65535, default 5004)

Use random port: No Yes

keep-alive interval: (in seconds, default 20 seconds)

Use NAT IP (if specified, this will be used in SIP/SDP message)

STUN server: (URI or IP:port)

Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP

Firmware Server Path:

Config Server Path:

Firmware File Prefix:

Firmware File Postfix:

Config File Prefix:

Config File Postfix:

Allow DHCP Option 66 to override server:
 No Yes

Automatic Upgrade:
 No Yes, check for upgrade every minutes (default 7 days)

5. In the **Firmware Upgrade and Provisioning** field, select **TFTP**.
6. In the **Config Server Path** field enter the IP Address of your Evolution PBX server.
7. In the **Allow DHCP Option 66 to override server** field, select **Yes**.

8. In the **NTP Server** field, enter the IP Address of your Evolution PBX server.
9. In the **Allow DHCP Option 42 to override NTP server** field, select **No**.

<i>Phonebook XML Download:</i>		Enable Phonebook XML Download: <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP
	Phonebook XML Server Path:	<input type="text"/>
	Phonebook Download Interval:	<input type="text" value="0"/> (0-720)
	Remove Manually-edited entries on Download:	<input checked="" type="radio"/> No <input type="radio"/> Yes
<i>Idle Screen XML Download:</i>		Enable Idle Screen XML Download: <input checked="" type="radio"/> No <input type="radio"/> YES, HTTP <input type="radio"/> YES, TFTP
	Idle Screen XML Server Path:	<input type="text"/>
<i>Offhook Auto Dial:</i>	<input type="text"/>	(User ID/extension to dial automatically when offhook)
<i>DTMF Payload Type:</i>	<input type="text" value="101"/>	
<i>Syslog Server:</i>	<input type="text"/>	
<i>Syslog Level:</i>	<input type="text" value="NONE"/>	
<i>NTP Server:</i>	<input type="text" value="192.168.2.225"/>	(URI or IP address)
	Allow DHCP Option 42 to override NTP server:	<input checked="" type="radio"/> No <input type="radio"/> Yes
<i>Distinctive Ring Tone:</i>	Custom ring tone 1, used if incoming caller ID is <input type="text"/>	
	Custom ring tone 2, used if incoming caller ID is <input type="text"/>	
	Custom ring tone 3, used if incoming caller ID is <input type="text"/>	
<i>System Ring Tone:</i>	<input type="text" value="f1=440,f2=480,c=200/400;"/>	
<i>Call Progress Tones:</i>	Dial Tone	<input type="text" value="f1=350,f2=440;"/>
	Message Waiting	<input type="text" value="f1=350,f2=440,c=10/10;"/>
	Ring Back Tone	<input type="text" value="f1=440,f2=480,c=200/400;"/>
	Call-Waiting Tone	<input type="text" value="f1=440,f2=440,c=25/525;"/>
	Busy Tone	<input type="text" value="f1=480,f2=620,c=50/50;"/>
	Reorder Tone	<input type="text" value="f1=480,f2=620,c=25/25;"/>
	Syntax: f1=val, f2=val [, c=on1/off1 [-on2/off2 [-on3/off3]]]; (Frequencies are in Hz and cadence on and off are in 10ms)	
<i>Disable Call-Waiting:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Disable Call-Waiting Tone:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Use Quick IP-call mode:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Lock Keypad Update:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (configuration update via keypad is disabled if set to Yes)	
<i>Display Language:</i>	<input type="text" value="English"/>	
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>		

10. Click the **ACCOUNT 1** tab and the account 1 configuration window displays.

Grandstream Device Configuration	
STATUS BASIC SETTINGS ADVANCED SETTINGS ACCOUNT 1 ACCOUNT 2 ACCOUNT 3 ACCOUNT 4 EXT 1 EXT 2	
Account Active:	<input type="radio"/> No <input checked="" type="radio"/> Yes
Account Name:	<input type="text"/> (e.g., MyCompany)
SIP Server:	<input type="text"/> 192.168.2.225 (e.g., sip.mycompany.com, or IP address)
Outbound Proxy:	<input type="text"/> 192.168.2.225 (e.g., proxy.myprovider.com, or IP address)
SIP User ID:	<input type="text"/> 5000 (the user part of an SIP address)
Authenticate ID:	<input type="text"/> 5000 (can be same or different from SIP UserID)
Authenticate Password:	<input type="text"/> (not displayed for security protection)
Name:	<input type="text"/> Name (optional, e.g., John Doe)
Use DNS SRV:	<input checked="" type="radio"/> No <input type="radio"/> Yes
User ID is phone number:	<input type="radio"/> No <input checked="" type="radio"/> Yes
SIP Registration:	<input type="radio"/> No <input checked="" type="radio"/> Yes
Unregister On Reboot:	<input type="radio"/> No <input checked="" type="radio"/> Yes
Register Expiration:	<input type="text"/> 2 (in minutes. default 1 hour, max 45 days)
local SIP port:	<input type="text"/> 5060 (default 5060)
SIP Registration Failure Retry Wait Time:	<input type="text"/> 20 (in seconds. Between 1-3600, default is 20)
SIP T1 Timeout:	<input type="text"/> 1 sec
SIP T2 Interval:	<input type="text"/> 4 sec
SIP Transport:	<input checked="" type="radio"/> UDP <input type="radio"/> TCP
Use RFC3581 Symmetric Routing:	<input checked="" type="radio"/> No <input type="radio"/> Yes
NAT Traversal (STUN):	<input type="radio"/> No <input type="radio"/> No, but send keep-alive <input checked="" type="radio"/> Yes
SUBSCRIBE for MWI:	<input checked="" type="radio"/> No <input type="radio"/> Yes
PUBLISH for Presence:	<input checked="" type="radio"/> No <input type="radio"/> Yes
Proxy-Require:	<input type="text"/>
Voice Mail UserID:	<input type="text"/> *97 (UserID for voice mail system)
Send DTMF:	<input type="checkbox"/> in-audio <input checked="" type="checkbox"/> via RTP (RFC2833) <input type="checkbox"/> via SIP INFO
Early Dial:	<input checked="" type="radio"/> No <input type="radio"/> Yes (use "Yes" only if proxy supports 484 response)
Dial Plan Prefix:	<input type="text"/> (this prefix string is added to each dialed number)
Delayed Call Forward Wait Time:	<input type="text"/> 20 (Allowed range 1-120, in seconds.)

11. In the **Account Active** field, select **Yes**.
12. In the **SIP Server** field, enter the IP Address (for example, 192.168.2.225) of your Evolution PBX server.
13. In the **Outbound Proxy** field enter the IP Address (for example, 192.168.2.225) of your Evolution PBX server.
14. In the **SIP User ID** field, enter the generic extension configured earlier in Evolution PBX.
15. In the **Authenticate ID** field, enter the generic extension configured earlier in Evolution PBX.
16. In the **User ID is phone number** field, select **Yes**.
17. In the **SIP Registration** field, select **Yes**.
18. In the **Unregister On Reboot** field, select **Yes**.

19. In the **Local SIP Port** field, enter **5060**.
20. In the **NAT Traversal (STUN)** field, select **Yes**.
21. In the **Voice Mail UserID** field, enter ***97**.
22. In the Send **DTMF** field, select **via RTP(RFC2833)**.
23. In the **Enable Call Features** field, select **Yes**.

<i>Enable Call Features:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes (if yes, call features using star codes will be supported locally)	
<i>Call Log:</i>	<input checked="" type="radio"/> Log All Calls <input type="radio"/> Log Incoming/Outgoing only (Missed calls NOT recorded) <input type="radio"/> Disable Call Log	
<i>Session Expiration:</i>	<input type="text" value="180"/>	(in seconds. default 180 seconds)
<i>Min-SE:</i>	<input type="text" value="90"/>	(in seconds. default and minimum 90 seconds)
<i>Caller Request Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Request for timer when making outbound calls)	
<i>Callee Request Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (When caller supports timer but did not request one)	
<i>Force Timer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Use timer even when remote party does not support)	
<i>UAC Specify Refresher:</i>	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)	
<i>UAS Specify Refresher:</i>	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)	
<i>Force INVITE:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (Always refresh with INVITE instead of UPDATE)	
<i>Enable 100rel:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Account Ring Tone:</i>	<input checked="" type="radio"/> system ring tone <input type="radio"/> custom ring tone 1 <input type="radio"/> custom ring tone 2 <input type="radio"/> custom ring tone 3	
<i>Send Anonymous:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)	
<i>Anonymous Method:</i>	<input checked="" type="radio"/> Use From Header <input type="radio"/> Use Privacy Header	
<i>Anonymous Call Rejection:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Auto Answer:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Allow Auto Answer by Call-Info:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes	
<i>Turn off speaker on remote disconnect:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Check SIP User ID for incoming INVITE:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Refer-To Use Target Contact:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Disable Multiple Media Attribute in SDP:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes	
<i>Preferred Vocoder: (in listed order)</i>	choice 1: <input type="text" value="PCMU"/>	choice 5: <input type="text" value="G.726-32"/>
	choice 2: <input type="text" value="PCMA"/>	choice 6: <input type="text" value="iLBC"/>
	choice 3: <input type="text" value="G.729A/B"/>	choice 7: <input type="text" value="G.722 (wide band)"/>
	choice 4: <input type="text" value="iLBC"/>	choice 8: <input type="text" value="GSM"/>
<i>SRTP Mode:</i>	<input checked="" type="radio"/> Disabled <input type="radio"/> Enabled but not forced <input type="radio"/> Enabled and forced	
<i>eventlist BLF URI:</i>	<input type="text"/>	
<i>Special Feature:</i>	<input type="text" value="Standard"/>	
<input type="button" value="Update"/> <input type="button" value="Cancel"/> <input type="button" value="Reboot"/>		

24. In the **Allow Auto Answer by Call-Info** field, select **Yes**.
25. In the Preferred Vendor field, select your vendor choices, in order: 1. PCMU 2. PCMA 3. G.729A/B 4. iLBC 5. G.726-32 6. iLBC 7. G.722(wide band) 8. GSM.
26. Click **Update** and the telephone will save, reboot, and automatically register with Evolution PBX.

Note: All other settings are optional.