Polycom Trio 8000 Series
Generic SIP Anyphone guide

Note: This guide is intended for HPBX 3.0

Additional info: This phone does follow Polycom standards and therefore uses the default login info: username: admin and password: 456.

1. Locate the IP address of the phone.
   a. Ask the customer for it.
   b. Use a network IP scanner.
2. Open a web browser and log into the web interface for the phone.
   a. Example address: https://192.0.0.0
3. Navigate to Settings > SIP
   a. Local Settings
      i. Local SIP port: The local SIP port is 5 + the last 3 digits of the extension.
         1. If the extension were 111, then the local SIP port would be 5111
      ii. Digitmap timeout: 3|3|3|6|3|3|3|3
   b. Click Save. The phone will reboot, proceed to the next step when it has finished rebooting.

SIP

Local Settings

* Local SIP Port	0
Calls Per Line Key	12
Enable Roaming buddies for	None ▼
New SDP Type	○ Enable ⦿ Disable
Live Communication Server Support	○ Enable ⦿ Disable
* Non Standard Line Seize	⦿ Enable ○ Disable
Disable Forward For Shared Line	⦿ Enable ○ Disable
Digitmap	[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT|**x.T|+x.T
* Digitmap Timeout	3|3|3|3|3|3|3
Remove End-of-Dial Marker	⦿ Enable ○ Disable
* Digitmap Impossible Match	0
Line Based Digitmap Switching	○ Enable ⦿ Disable

Outbound Proxy
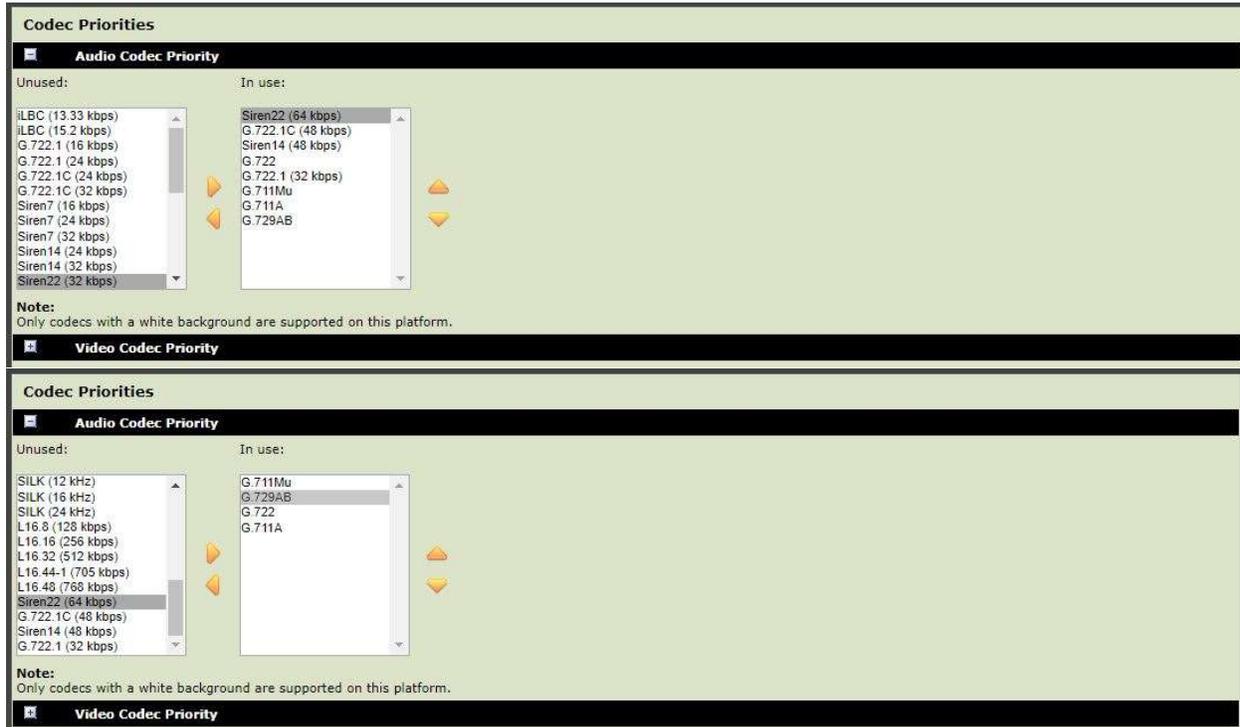
Server 1

Server 2

Server 3

Note:
* Fields require a phone reboot/restart.

4. Navigate to Settings > Codec Priorities
   a. For the "In use" box, remove all but the following and set in the following order:
      i. G.711Mu
      ii. G.729AB
      iii. G.722
      iv. G.711A
   b. Click Save.

Codec Priorities

Audio Codec Priority

Unused:
iLBC (13.33 kbps)
iLBC (15.2 kbps)
G.722.1 (16 kbps)
G.722.1 (24 kbps)
G.722.1C (24 kbps)
G.722.1C (32 kbps)
Siren7 (16 kbps)
Siren7 (24 kbps)
Siren7 (32 kbps)
Siren14 (24 kbps)
Siren14 (32 kbps)
Siren22 (32 kbps)

In use:
Siren22 (64 kbps)
G.722.1C (48 kbps)
Siren14 (48 kbps)
G.722
G.722.1 (32 kbps)
G.711Mu
G.711A
G.729AB

Note:
Only codecs with a white background are supported on this platform.

Video Codec Priority

Codec Priorities

Audio Codec Priority

Unused:
SILK (12 kHz)
SILK (16 kHz)
SILK (24 kHz)
L16.8 (128 kbps)
L16.16 (256 kbps)
L16.32 (512 kbps)
L16.44-1 (705 kbps)
L16.48 (768 kbps)
Siren22 (64 kbps)
G.722.1C (48 kbps)
Siren14 (48 kbps)
G.722.1 (32 kbps)

In use:
G.711Mu
G.729AB
G.722
G.711A

Note:
Only codecs with a white background are supported on this platform.

Video Codec Priority

5. Navigate to Settings > Lines
   a. Identification
      i. Display name: Is arbitrary and can be set to what ever the customer prefers
      ii. Address: enter the SIP User ID here
      iii. Label: enter the SIP User ID here
      iv. Line keys: can be set to a max of 3
         1. Note: this phone is capable of only 1 SIP registration and thus only one ITSP can be utilized.
   b. Authentication
      i. User ID: enter the SIP User ID here
      ii. Password: enter the SIP password here
   c. Outbound proxy
      i. Address: enter the sip server address
      ii. Port: 5060
      iii. Transport type: UDP Only
   d. Server 1
      i. Address: enter the sip server address
      ii. Port: 5060
      iii. Transport type: UDP Only
      iv. Expires(s): 300
   e. Message Center:
      i. Subscription address: enter the assigned extension
      ii. Callback Mode: Contact
      iii. Callback Contact: enter the assigned extension

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Line 1

▣ Identification

Display Name	
Address	
Label	
Type	⦿ Private ◯ Shared
Third Party Name	
Number of Line Keys	1
Calls Per Line	12
Enable SRTP	⦿ Yes ◯ No
Offer SRTP	◯ Yes ⦿ No
Require SRTP	◯ Yes ⦿ No
Server Auto Discovery	⦿ Enable ◯ Disable
TCP Fast Failover	◯ Enable ⦿ Disable

▣ Authentication

Use Login Credentials	◯ Enable ⦿ Disable
Domain	
User ID	
Password	

▣ Outbound Proxy

Address	
Port	0
Transport	DNSnaptr ▾

Line 1

▣ Identification
▣ Authentication
▣ Outbound Proxy
▣ Server 1

Special Interop	Standard ▾
Address	
Port	0
Transport	DNSnaptr ▾
Expires (s)	3600
Subscription Expires (s)	3600
Register	⦿ Yes ◯ No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30
Re-register on Failover	◯ Yes ⦿ No
Fail Back Mode	Duration ▾
Fail Back Timeout (s)	3600

▣ Server 2
▣ Server 3
▣ Call Diversion
▣ Message Center
▣ Ring Type
▣ Digitmap

Note:
* Fields require a phone reboot/restart.

[Cancel] [Reset to Default] [View Modifications] [Save]

Line 1

⊞	**Identification**
⊞	**Authentication**
⊞	**Outbound Proxy**
⊞	**Server 1**
⊞	**Server 2**
⊞	**Server 3**
⊞	**Call Diversion**
⊟	**Message Center**

Subscription Address [_____]

Callback Mode [Registration ▼]

Callback Contact [_____]

⊟	**Ring Type**

Ring Type [Low Trill ▼]

Private Line Ring Type [Silent Ring ▼]

⊞	**Digitmap**

Note:
* Fields require a phone reboot/restart.

[Cancel] [Reset to Default] [View Modifications] [Save]

Line 1

⊞	**Identification**
⊞	**Authentication**
⊞	**Outbound Proxy**
⊞	**Server 1**
⊞	**Server 2**
⊞	**Server 3**
⊟	**Call Diversion**

* Enforced by Server   ○ Yes   ● No

Signaling Method   [Subscribe As Feature Event ▼]

Always Forward   ○ Enable   ● Disable

Always Forward To Contact [_____]

If Busy, Forward   ○ Enable   ● Disable

If Busy, Forward To Contact [_____]

On No Answer, Forward   ○ Enable   ● Disable

On No Answer, Forward To Contact [_____]

On No Answer, Forward After Rings [9]

* On Do Not Disturb, Forward   ○ Enable   ● Disable

* On Do Not Disturb, Forward To Contact [_____]

* Disable Forward For Shared Lines   ● Yes   ○ No

* Forward Specific Caller   ● Enable   ○ Disable

⊞	**Message Center**
⊞	**Ring Type**
⊞	**Digitmap**

Note:
* Fields require a phone reboot/restart.

[Cancel] [Reset to Default] [View Modifications] [Save]

6. Setting the Time and Date:
    a. SNTP Server: pool.ntp.org
    b. Alternate SNTP Server: pool.ntp.org
    c. GMT Offset: select the local for the customer's location
    d. Alternate GMT Offset: select the local for the customer's location

Date & Time

Display Format	
Time Format	12 AM/PM ▼
Date Format	Monday, January 1 ▼

Time Synchronization	
Override DHCP's SNTP Server	○ Yes ◉ No
SNTP Server	▼
Alternate SNTP Server	▼
SNTP Resync Period (s)	86400
Time Zone ID	▼
Override DHCP's GMT Offset	○ Yes ◉ No
GMT Offset	(GMT 0:00) Western Europe Time ▼
Alternate GMT Offset	(GMT 0:00) Western Europe Time ▼

Daylight Savings

After the phone setup is complete and the phone is successfully registering, the top of the phone display will show the SIP User ID. Unfortunately, the field in the User Interface that causes this, is also part of the SIP registration process. While this field cannot be changed, it can at least be hidden so that this information is not displayed on the screen. It should be noted, that at this time, for HPBX 3.0 configurations for this phone, there is no way to have the assigned phone number, displayed on the phone.

To hide/prevent the SIP User/Registration ID from showing on or at the top of the screen, follow these steps from the phone screen itself, not the WebUI.

1. Tap on Settings
2. Tap on Advanced
    a. Enter the password.
    b. Should be "456" unless changed during setup.
3. Tap on Administration Settings
4. Tap on Home Screen Label
5. Tap on Type
    a. Select: None
    b. Tap on Done
6. Tap on Save

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If it is desired that this phone should be able to participate via auto answer in regards to the Intercom function from the hosted phone platform, the following lines will need to be added to the primary config file of the phone.
Follow these steps:

1. In the web interface, navigate to Utilities > Import & Export Configuration
2. Export Configuration section and from the drop-down list, select "All Configuration (except Device Settings) and click on the export button.
   a. Ensure that this is saved to a know location that can be accessed.
3. Open the saved config file in Notepad ++
   a. If you do not have this program, you can acquire it here: https://notepad-plus-plus.org/
   b. Add the following lines anywhere within the <ALL ~ />
   c. Make sure they line up with the other entries present.
      i. se.rt.custom1.name="Paging"
      ii. se.rt.custom1.ringer="ringer2"
      iii. se.rt.custom1.timeout="5"
      iv. se.rt.custom1.type="answer"
      v. voIpProt.SIP.alertInfo.1.class="custom1"
      vi. voIpProt.SIP.alertInfo.1.value="paging"
      vii. se.pat.ringer.ringer15.inst.1.value="2"
   d. Save the file, do not change the file name.
4. Back to the phones web interface, Utilities > Import & Export Configuration
5. Import configuration
   a. Click on Choose File, and look for the file: Export_all.cfg
   b. Click on Import
6. Test the feature by going to another phone and pressing *90 + ext of the Trio 8000 series phone.
7. If done correctly, a single tone should be heard and then a voice channel will be automatically opened.