

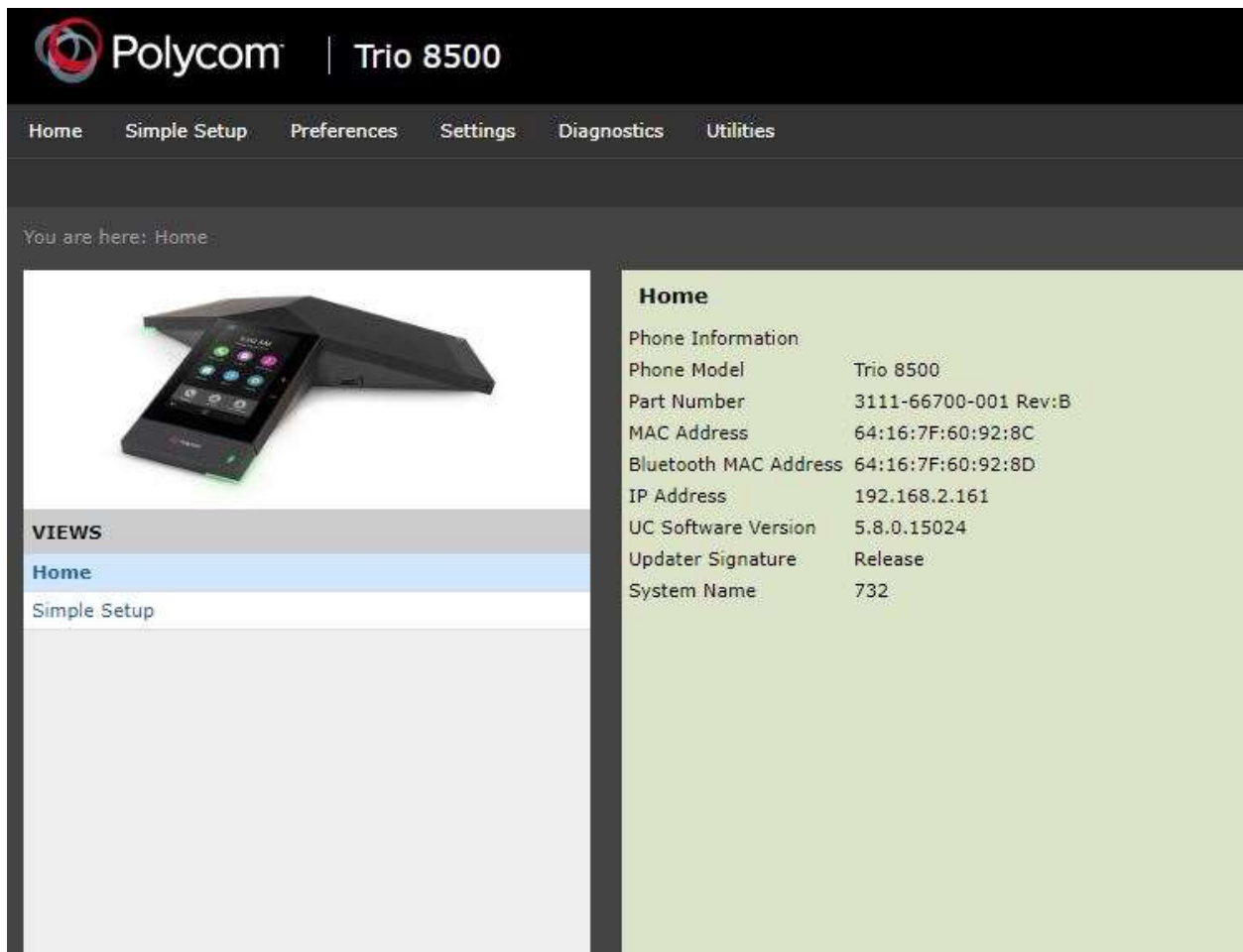
# Polycom Trio 8000 Series

## Generic SIP Anyphone guide

Note: This guide is intended for HPBX 2.0

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>



The screenshot shows the Polycom Trio 8500 web interface. At the top, there is a navigation bar with the Polycom logo and the text 'Trio 8500'. Below the navigation bar, there are several menu items: Home, Simple Setup, Preferences, Settings, Diagnostics, and Utilities. The main content area is divided into two columns. The left column features a large image of the phone and a 'VIEWS' section with 'Home' and 'Simple Setup' options. The right column displays 'Phone Information' with the following details:

Phone Information	
Phone Model	Trio 8500
Part Number	3111-66700-001 Rev:B
MAC Address	64:16:7F:60:92:8C
Bluetooth MAC Address	64:16:7F:60:92:8D
IP Address	192.168.2.161
UC Software Version	5.8.0.15024
Updater Signature	Release
System Name	732

3. Navigate to Settings > SIP
  - a. Local Settings

# Polycom Trio 8000 Series

## Generic SIP Anyphone guide

- i. Local SIP port: The local SIP port is 6 + the last 3 digits of the extension.
    1. If the extension were 111, then the local SIP port would be 6111
  - 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.

The screenshot shows the SIP configuration page with the following settings:

- 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]xxxxxxxx|[2-9]xxxxxxxx|[2-9]xxxT|\*+x.T
  - Digitmap Timeout: 3|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.

The first screenshot shows the Codec Priorities configuration page with the following settings:

- 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.

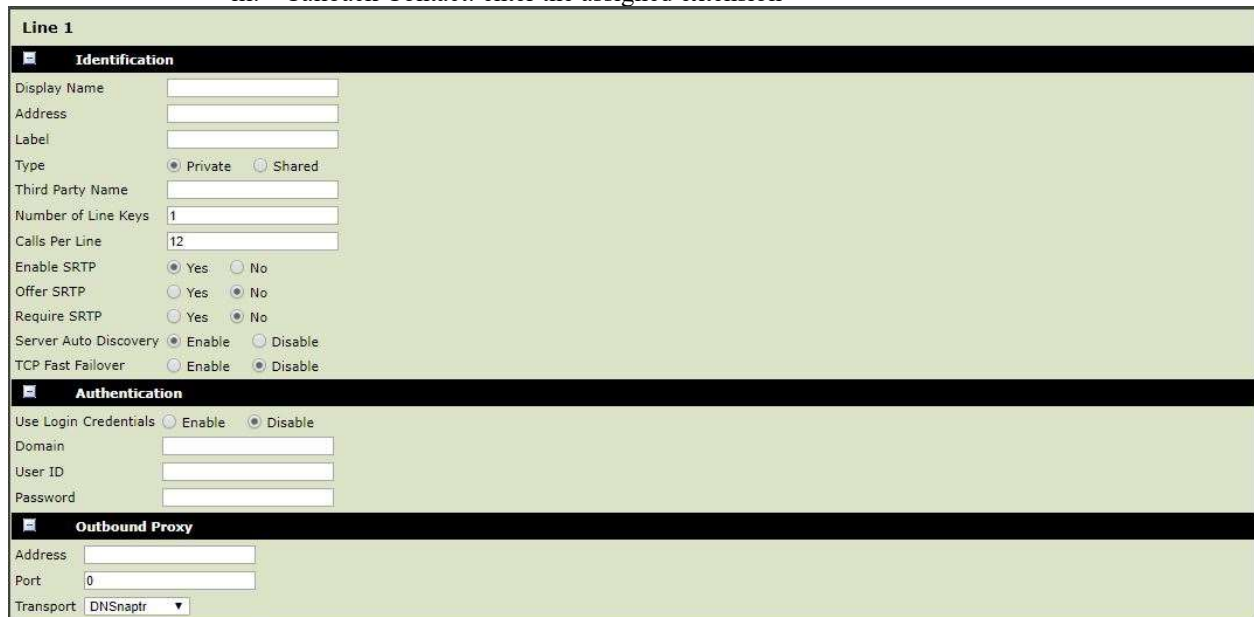
The second screenshot shows the Codec Priorities configuration page with the following settings:

- 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.

# Polycom Trio 8000 Series

## Generic SIP Anyphone guide

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 assigned extension
    - iii. Label: enter the assigned extension
    - 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: 6060
    - iii. Transport type: UDP Only
  - d. Server 1
    - i. Address: enter the sip server address
    - ii. Port: 6060
    - iii. Transport type: UDP Only
    - iv. Expires(s): 300
  - e. Message Center:
    - i. Subscription address: enter the assigned extension
    - ii. Callback Mode: Registration
    - iii. Callback Contact: enter the assigned extension



**Line 1**

**Identification**

Display Name

Address

Label

Type  Private  Shared

Third Party Name

Number of Line Keys

Calls Per Line

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

Transport

# Polycom Trio 8000 Series

## Generic SIP Anyphone guide

**Line 1**

- Identification
- Authentication
- Outbound Proxy
- Server 1

Special Interop: Standard

Address:

Port:

Transport: DNSnaptr

Expires (s):

Subscription Expires (s):

Register:  Yes  No

Retry Timeout (ms):

Retry Maximum Count:

Line Seize Timeout (s):

Re-register on Failover:  Yes  No

Fail Back Mode: Duration

Fail Back Timeout (s):

- 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

\* 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:

\* 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

# Polycom Trio 8000 Series

## Generic SIP Anyphone guide

**Line 1**

- Identification
- Authentication
- Outbound Proxy
- Server 1
- Server 2
- Server 3
- Call Diversion
- Message Center

Subscription Address:

Callback Mode:

Callback Contact:

- Ring Type

Ring Type:

Private Line Ring Type:

- Digitmap

**Note:**  
\* Fields require a phone reboot/restart.

### 6. Setting the Time and Date:

- SNTP Server: pool.ntp.org
- Alternate SNTP Server: pool.ntp.org
- GMT Offset: select the local for the customer's location
- Alternate GMT Offset: select the local for the customer's location

**Date & Time**

- Display Format

Time Format:

Date Format:

- Time Synchronization

Override DHCP's SNTP Server:  Yes  No

SNTP Server:

Alternate SNTP Server:

SNTP Resync Period (s):

Time Zone ID:

Override DHCP's GMT Offset:  Yes  No

GMT Offset:

Alternate GMT Offset:

- Daylight Savings