

Note: A single Anyphone line has a maximum concurrent call capacity of 4 calls. Hardware capacity should also be taken into consideration. This means that Grandstream GXP2140 can handle up to 4 concurrent calls.

This document will show how to configure Grandstream GXP2140 phone for use with Intermedia Hosted PBX service (AnyPhone BYOP).

The default credentials for this phone are:

Username: admin

Password: admin

If the phone was previously used with another service provider, you may need to factory reset it before logging in.

Once logged in, go to Accounts > Account 1. Enter the following settings:

General Settings

- Account Active:** yes
- **Account Name:** can be anything
- **SIP Server:** SIP Domain from the phone's SIP Configuration (usbc.telecomsvc.com:6060 or UC70.telecomsvc.com:6060)
- **Outbound Proxy:** Outbound Proxy from the phone's SIP Configuration
- **SIP User ID:** SIP User Name from the phone's SIP Configuration
- **Authenticate ID:** SIP Authorization ID from the phone's SIP Configuration
- **Voice Mail ID:** the phone's extension number

Press **Save** button when done.

General Settings configuration example:

version 1.0.4.15

Accounts

Account 1 -

General Settings

Network Settings

SIP Settings +

Audio Settings

Call Settings

Account 2 +

Account 3 +

Account 4 +

General Settings

Account Active No Yes

Account Name Extension 100

SIP Server uc70.telecomsvc.com:60

Secondary SIP Server

Outbound Proxy uc70.telecomsvc.com:60

BLF Server

SIP User ID 96 6

Authenticate ID 5 6

Authenticate Password

Name

Voice Mail UserID 100

Show Account Name Only No Yes

Voice Mail UserID

Allows users to access voice messages by pressing the MESSAGE button on the phone. This ID is usually the VM portal access number.

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SIP Settings > Basic Settings

- SIP Registration:** Yes
- Local SIP Port:** 6xxx, where xxx are the last 3 digits of a phone's extension.
- SIP Transport:** UDP

Press **Save** button when done.

Basic SIP Settings configuration example:

⚠ You have unapplied configuration(s). APPLY

Accounts

Account 1

- General Settings
- Network Settings
- SIP Settings
- Basic Settings**
- Advanced Features
- Session Timer
- Security Settings
- Audio Settings
- Call Settings

Account 2 +

Account 3 +

Account 4 +

Basic Settings

TEL URI Disabled User=phone Enabled

SIP Registration No Yes

Unregister on Reboot No All Instance

Register Expiration

Reregister before Expiration

Local SIP Port

SIP Registration Failure Retry Wait Time

SIP T1 Timeout

SIP T2 Timeout

SIP Transport UDP TCP TLS/TCP

SIP URI Scheme When Using TLS sip sips

Use Actual Ephemeral Port in Contact with TCP/TLS No Yes

Remove OBP from Route No Yes

Support SIP Instance ID No Yes

SUBSCRIBE for MWI No Yes

SUBSCRIBE for Registration No Yes

Enable 100rel No Yes

Caller ID Display Auto Disabled From Header

Use Privacy Header Default No Yes

Use P-Preferred-Identity Header Default No Yes

Add Auth Header On Initial REGISTER No Yes

Enable 100rel

When enabled, the 100rel tag is appended to the value of the required header of the initial signaling messages.

[Reset to Default](#)

[Undo](#)

Audio Settings

- **Send DTMF:** via RTP (RFC2833)
- **Preferred Vocoder – choice 1:** G.729A/B if the line's primary codec is G.729; PCMU if the line's primary codec is G.711
- **Preferred Vocoder – choice 2:** PCMU if the line's secondary codec is G.711; G.729A/B if the line's secondary codec is G.729

Press **Save and Apply** once configured.

Configuration example:

Accounts

- Account 1 -
- Account 2 +
- Account 3 +
- Account 4 +

- General Settings
- Network Settings
- SIP Settings
- Audio Settings
- Call Settings

Audio Settings

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

DTMF Payload Type

Preferred Vocoder - choice 1

Preferred Vocoder - choice 2

Preferred Vocoder - choice 3

Preferred Vocoder - choice 4

Preferred Vocoder - choice 5

Use First Matching Vocoder in 2000K SDP No Yes

S RTP Mode

Symmetric RTP No Yes

Silence Suppression No Yes

Voice Frames per TX

G.726-32 Packing Mode ITU IETF

Jitter Buffer Type

Jitter Buffer Length

Use First Matching Vocoder in 2000K SDP

When set to "Yes", the device will use the first matching vocoder in the received 2000K SDP as the codec.