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.

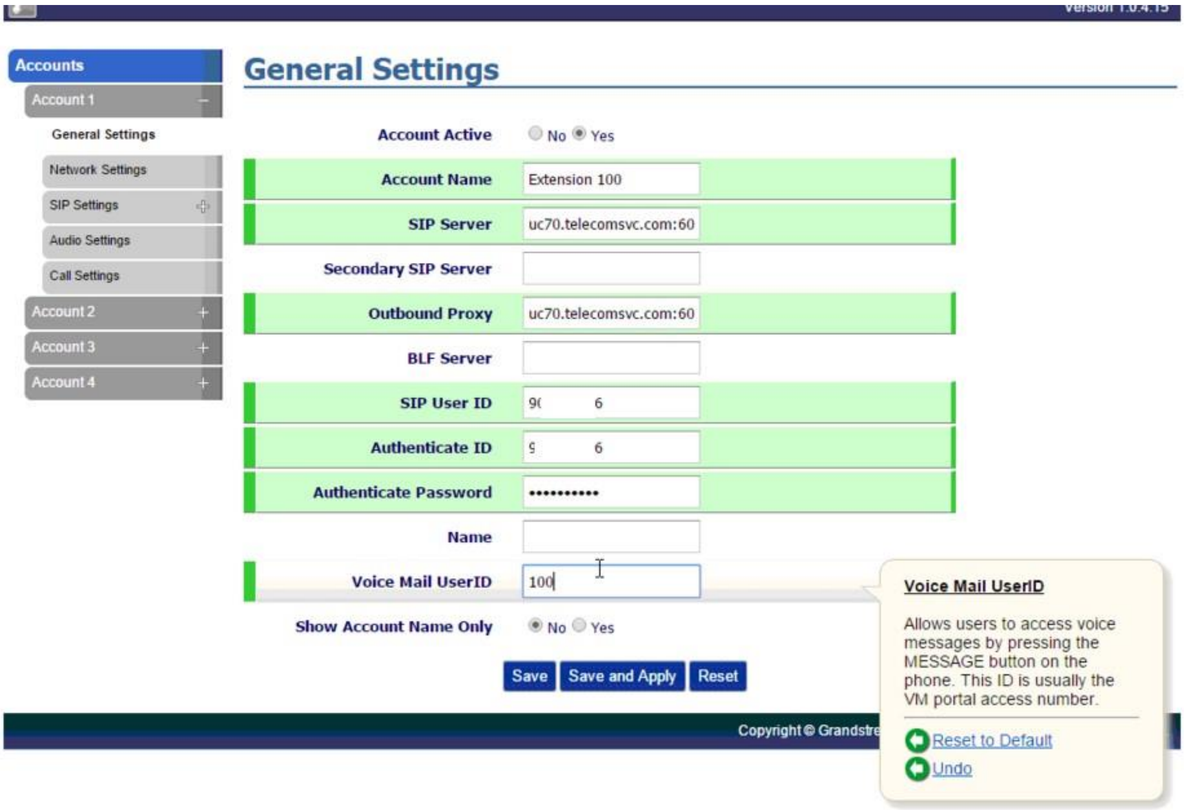
Once logged in, go to Accounts > Account 1. Enter the following settings:

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:

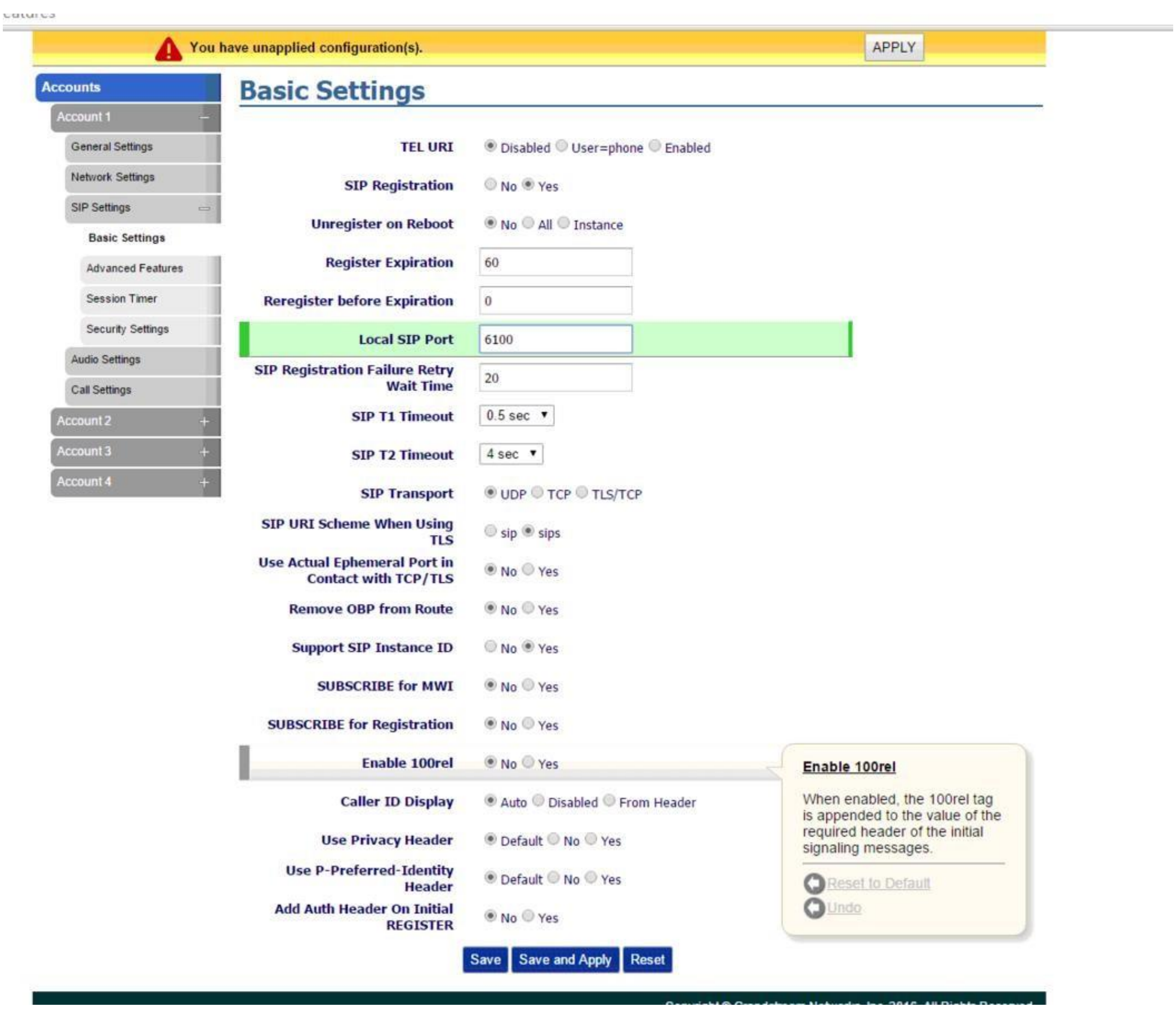


SIP Settings > Basic Settings

- **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:

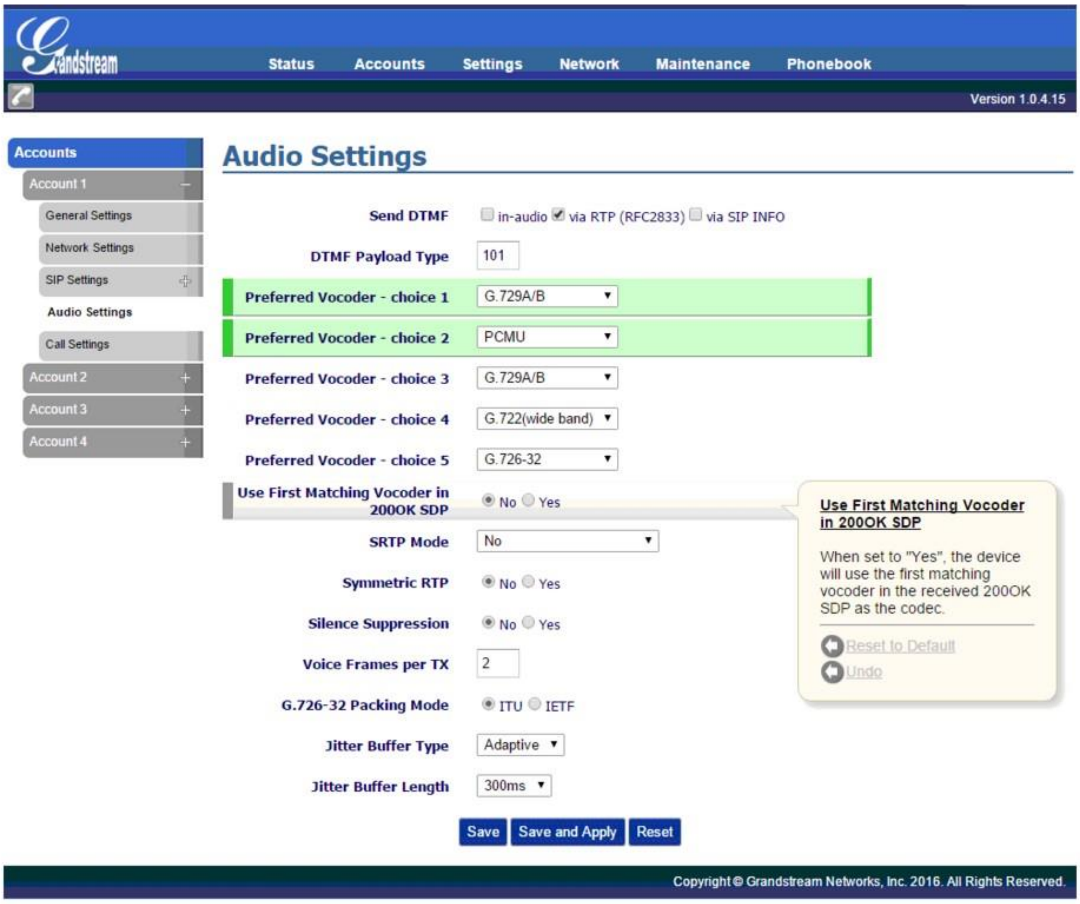


Audio Settings

- **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:


Grandstream
Status
Accounts
Settings
Network
Maintenance
Phonebook
Version 1.0.4.15
Accounts
Account 1
General Settings
Network Settings
SIP Settings
Audio Settings
Call Settings
Account 2
Account 3
Account 4
Audio Settings
Send DTMF
in-audio
via RTP (RFC2833)
via SIP INFO
DTMF Payload Type
101
Preferred Vocoder - choice 1
G.729A/B
Preferred Vocoder - choice 2
PCMU
Preferred Vocoder - choice 3
G.729A/B
Preferred Vocoder - choice 4
G.722(wide band)
Preferred Vocoder - choice 5
G.726-32
Use First Matching Vocoder in 200OK SDP
No
Yes
SRTP Mode
No
Symmetric RTP
No
Yes
Silence Suppression
No
Yes
Voice Frames per TX
2
G.726-32 Packing Mode
ITU
IETF
Jitter Buffer Type
Adaptive
Jitter Buffer Length
300ms
Save
Save and Apply
Reset
Use First Matching Vocoder in 200OK SDP
When set to "Yes", the device will use the first matching vocoder in the received 200OK SDP as the codec.
Reset to Default
Undo
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