

Grandstream HT-701 Configuration

Firmware 1.0.1.6+

To access the web interface: dial '***' then '02' from the phone attached to the ATA.
Next, enter the IP address provided in your web browser. Login password is **admin**

On the 'Basic Settings' page modify the 'Self-Defined Time Zone' box to set the appropriate time zone for your location. Time zone strings are provided below the screenshot.

Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS** **FXS PORT**

End User Password: (purposely not displayed for security protection)

Web Port: (default for HTTP is 80)

Telnet Server: No Yes

IP Address: dynamically assigned via DHCP

DHCP hostname: (optional)

DHCP domain: (optional)

DHCP vendor class ID: (optional)

use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

Preferred DNS server: . . .

statically configured as:

IP Address: . . .

Subnet Mask: . . .

Default Router: . . .

DNS Server 1: . . .

DNS Server 2: . . .

Time Zone: ▼

Self-Defined Time Zone: (For example: MTZ+6MDT+5,M4.1.0,M11.1.0)

Allow DHCP server to set Time Zone: No Yes

Language: ▼

Reset Type: ▼

All Rights Reserved Grandstream Network

- EST**
MTZ+5MDT+4,M3.2.0,M11.1.0
- PST**
MTZ+8MDT+7,M3.2.0,M11.1.0
- MST**
MTZ+7MDT+6,M3.2.0,M11.1.0
- CST**
MTZ+6MDT+5,M3.2.0,M11.1.0

Next, access the FXS Port page and update the highlighted fields:

Note: SIP User ID, Authenticate Password and Name will be unique for each user.

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT
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Account Active: No Yes

Primary SIP Server: (e.g., sip.mycompany.com, or IP address)

Failover SIP Server: (Optional, used when primary server no response)

Prefer Primary SIP Server: No Yes (yes - will register to Primary Server if Failover registration expires)

Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)

SIP Transport: UDP TCP TLS (default is UDP)

NAT Traversal: No Keep-Alive STUN UPnP

SIP User ID: (the user part of an SIP address)

Authenticate ID: (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: (optional, e.g., John Doe)

DNS Mode: A Record SRV NAPTR/SRV

Tel URI:

SIP Registration: No Yes

Unregister On Reboot: No Yes

Outgoing Call without Registration: No Yes

Register Expiration: (in minutes. default 1 hour, max 45 days)

SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)

Local SIP port: (default is 5060 for UDP and TCP; 5061 for TLS)

Local RTP port: (even number between 1024-65535, default 5004)

Use Random SIP Port: No Yes

Use Random RTP Port: No Yes

Refer-To Use Target Contact: No Yes

Transfer on Conference Hangup: No Yes

Disable Bellcore Style 3-Way Conference: No Yes (Using star code *23 for 3-way conference)

Remove OBP from Route Header: No Yes

Support SIP Instance ID: No Yes

Validate Incoming SIP Message: No Yes

Check SIP User ID for incoming INVITE: No Yes (no direct IP calling if Yes)

Allow Incoming SIP Messages from SIP Proxy Only: No Yes (no direct IP calling if Yes)

SIP T1 Timeout:

SIP T2 Interval:

DTMF Payload Type:

SIP T1 Timeout: 0.5 sec ▼

SIP T2 Interval: 4 sec ▼

DTMF Payload Type: 101

Preferred DTMF method: Priority 1: RFC2833 ▼
 (in listed order) Priority 2: In-audio ▼
 Priority 3: SIP INFO ▼

Disable DTMF Negotiation: No (negotiate with peer) Yes (use above DTMF order without negotiation)

Send Hook Flash Event: No Yes (Hook Flash will be sent as a DTMF event if set to Yes)

Enable Call Features: No Yes (if Yes, call features using star codes will be supported locally)

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

Ring Tone 1 ▼ used if incoming caller ID is

Distinctive Ring Tone: Ring Tone 1 ▼ used if incoming caller ID is

Ring Tone 1 ▼ used if incoming caller ID is

Disable Call-Waiting: No Yes

Disable Call-Waiting Caller ID: No Yes

Disable Call-Waiting Tone: No Yes

Disable Receiver Offhook Tone: No Yes (ROH tone will not be played after offhook for 60 seconds)

Disable Reminder Ring for On-Hold Call: No Yes

Disable Visual MWI: No Yes

Ring Timeout: 60 (10-300, default is 60 seconds)

Delayed Call Forward Wait Time: 20 (Allowed range 1-120, in seconds.)

No Key Entry Timeout: 2 (in seconds, default is 4 seconds)

Early Dial: No Yes (use "Yes" only if proxy supports 484 response)

Dial Plan Prefix: (this prefix string is added to each dialed number)

Use # as Dial Key: No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)

Dial Plan: { x+ | *x+ }

SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication
 Yes, send periodical SUBSCRIBE for Message Waiting Indication

Send Anonymous: No Yes (caller ID will be blocked if set to Yes)

Anonymous Call Rejection: No Yes

Lower down the page:

Send Re-INVITE After Fax: No Yes

Enable 100rel: No Yes

Use First Matching Vocoder in 200OK SDP: No Yes

Preferred Vocoder: choice 1: PCMU ▼
 (in listed order) choice 2: G729 ▼
 choice 3: PCMU ▼
 choice 4: PCMU ▼
 choice 5: PCMU ▼
 choice 6: PCMU ▼

G722 Rate: 6.25k... 5.25k...