

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:

80

(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:

HT7XX

(optional)

☐ use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

Preferred DNS server:

0

.

0

.

0

.

0

☐ statically configured as:

IP Address:

192

.

168

.

0

.

160

Subnet Mask:

255

.

255

.

0

.

0

Default Router:

0

.

0

.

0

.

0

DNS Server 1:

0

.

0

.

0

.

0

DNS Server 2:

0

.

0

.

0

.

0

Time Zone:

Using self-defined Time Zone

Self-Defined Time Zone:

MTZ+5MDT+4,M3.2.0,M11.1.0

(For example: MTZ+6MDT+5,M4.1.0,M11.1.0)

Allow DHCP server to set Time Zone:

☒ No

☐ Yes

Language:

English

Reset Type:

Full Reset

Reset

Update

Cancel

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
Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Primary SIP Server: <input type="text" value="voip.directnet.ca"/>		(e.g., sip.mycompany.com, or IP address)	
Failover SIP Server: <input type="text"/>		(Optional, used when primary server no response)	
Prefer Primary SIP Server: <input checked="" type="radio"/> No <input type="radio"/> Yes (yes - will register to Primary Server if Failover registration expires)			
Outbound Proxy: <input type="text"/>		(e.g., proxy.myprovider.com, or IP address, if any)	
SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)			
NAT Traversal: <input type="radio"/> No <input checked="" type="radio"/> Keep-Alive <input type="radio"/> STUN <input type="radio"/> UPnP			
SIP User ID: <input type="text" value="15198040000"/>		(the user part of an SIP address)	
Authenticate ID: <input type="text"/>		(can be identical to or different from SIP User ID)	
Authenticate Password: <input type="password" value="....."/>		(purposely not displayed for security protection)	
Name: <input type="text" value="Display Name"/>		(optional, e.g., John Doe)	
DNS Mode: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV			
Tel URI: <input type="text" value="Disabled"/>			
SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Outgoing Call without Registration: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Register Expiration: <input type="text" value="60"/>		(in minutes. default 1 hour, max 45 days)	
SIP Registration Failure Retry Wait Time: <input type="text" value="20"/>		(in seconds. Between 1-3600, default is 20)	
Local SIP port: <input type="text" value="5060"/>		(default is 5060 for UDP and TCP; 5061 for TLS)	
Local RTP port: <input type="text" value="5004"/>		(even number between 1024-65535, default 5004)	
Use Random SIP Port: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Use Random RTP Port: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Refer-To Use Target Contact: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Transfer on Conference Hangup: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Disable Bellcore Style 3-Way Conference: <input checked="" type="radio"/> No <input type="radio"/> Yes (Using star code *23 for 3-way conference)			
Remove OBP from Route Header: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Support SIP Instance ID: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Validate Incoming SIP Message: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Check SIP User ID for incoming INVITE: <input checked="" type="radio"/> No <input type="radio"/> Yes (no direct IP calling if Yes)			
Allow Incoming SIP Messages from SIP Proxy Only: <input type="radio"/> No <input checked="" type="radio"/> Yes (no direct IP calling if Yes)			
SIP T1 Timeout: <input type="text" value="0.5 sec"/>			
SIP T2 Interval: <input type="text" value="4 sec"/>			
DTMF Payload Type: <input type="text" value="101"/>			

SIP T1 Timeout: 0.5 sec ▼

SIP T2 Interval: 4 sec ▼

DTMF Payload Type: 101

Preferred DTMF method: (in listed order)
 Priority 1: RFC2833 ▼
 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: (in listed order)
 choice 1: PCMU ▼
 choice 2: G729 ▼
 choice 3: PCMU ▼
 choice 4: PCMU ▼
 choice 5: PCMU ▼
 choice 6: PCMU ▼

G722 Pass: ☒ 6.214k pass-discontinuity ☐ 5.214k pass-discontinuity