

# T300/T500 - Grandstream HT-502/503 VoIP-ATA - How To



This article will describe the configuration of the HandyTone 502/503 behind the Gigaset T300/500 PBX system.

To guarantee a proper operation, please update the device first to the latest sw-version (V1.0.8.4, 05.02.2013).

The automatic (one-touch) setup is not supported with these devices. If you want to use this mechanism, please use the Patton or Grandstream HT702/704 devices.

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To use a Grandstream VoIP ATA to connect a fax machine to the T300/T500 PBX you have to take the following steps:

1. Connect the ATA Adapter to your network using the WAN port of the box and connect the powercable. To start the configuration connect an analog phone to the phone1 connector of the box and dial **\*\*\***. The voice menu asks you to choose the wanted menu. Dial **12** and in the next step **9**. This enables WAN side http access. Hang up the phone. Restart the box by switching the power. Then take the phone and dial **\*\*\***. At the voice menu dial **02**. The voice system will tell you the current IP address of the box.
2. Open your webbrowser and type in the address, e. g. 192.168.2.24. The default password of the box is **admin**.

- The basic and advanced settings can be kept by default.
- The FXS Port Menu has to be configured as shown

**Grandstream Device Configuration**

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT1 FXS PORT2

Account Active:  No  Yes

Primary SIP Server: 192.168.2.50 (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: 192.168.2.50 (e.g., proxy.myprovider.com, or IP address, if any)

SIP Transport:  UDP  TCP  TLS (default is UDP)

NAT Traversal (STUN):  No  No, but send keep-alive  Yes

SIP User ID: GrandHT502 (the user part of an SIP address)

Authenticate ID: GrandHT502 (can be identical to or different from SIP User ID)

Authenticate Password: (purposely not displayed for security protection)

Name: HT502 (optional, e.g., John Doe)

DNS Mode:  A Record  SRV  NAPTR/SRV

Tel URI: Disabled

SIP Registration:  No  Yes

Unregister On Reboot:  No  Yes

Outgoing Call without Registration:  No  Yes

Register Expiration: 60 (in minutes. default 1 hour, max 45 days)

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

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

Local RTP Port: 5004 (1024-65535, default 5004)

In our example the T500 PBX uses 192.168.2.50 as IP address.

The SIP name configured in the PBX is GrandHT502, we use a random password.

Configuration

Edit

Users Groups Phones Modules Voicemail Conference Addressbook Phone Numbers Lines Routing Server Statistic Interconnection Security Extended Settings

**Details**

Telephone Type: Standard Sip Functionality: Phone

Telephone name: GrandHT502 Password: 0mpbUN4j81hSdM73 Random password

Applied server address: 192.168.2.50 Last known Device-IP: 192.168.2.30

Autoprovisioning

Last provisioning: Jan 1, 1970 12:00:00 AM

MAC address: ---

Firmware version: ---

Extended Configuration

NAT: default

Codecs: alaw,ulaw

Restrict to IP: ---

Door Intercom

Enable  Disable

Camera URL: Test Connection

DTMF: ---

Save Apply Cancel

We advice to check the codec settings at the bottom of the FXS port menu.

We strongly advise only to use the G711 alaw /ulaw codec for use with connected fax machines.

### Additional Information:

#### External Link

**ATTENTION: No further support for this device from Gigaset!!**

Use First Matching Vocoder in 2000K SDP:  No  Yes

Preferred Vocoder: (in listed order)

choice 1: PCMA ▼

choice 2: PCMU ▼

choice 3: PCMA ▼

choice 4: PCMA ▼

choice 5: PCMA ▼

choice 6: PCMA ▼

G723 Rate:  6.3kbps encoding rate  5.3kbps encoding rate

iLBC Frame Size:  20ms  30ms

iLBC Payload Type: 97 (between 96 and 127, default is 97)

VAD:  No  Yes

Symmetric RTP:  No  Yes

Fax Mode:  T.38  Pass-Through

Re-INVITE After Fax Tone Detected:  Enabled  Disabled

Jitter Buffer Type:  Fixed  Adaptive

Jitter Buffer Length:  Low  Medium  High

SRTP Mode:  Disabled  Enabled but not forced  Enabled and forced

SLIC Setting: GERMANY ▼

Caller ID Scheme: Bellcore/Telcordia ▼