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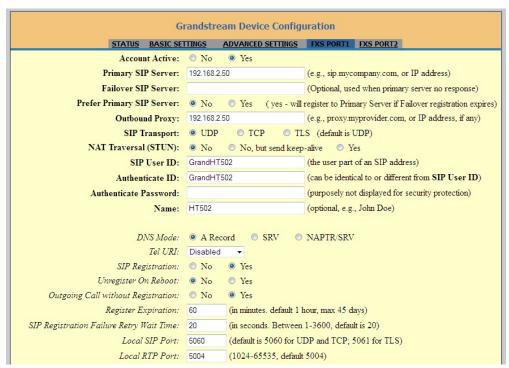
To guarantee a proper operation, please update the device first to the latest sw-version (V1.0.8.4, 05.02.2013).

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.

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 configuraration 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**.

3. The basic and advanced settings can be kept by default.

4. 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

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   [i]

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

MAC address: ---

Firmware version: ---

Extended Configuration   [i]

NAT: default ▾

Codecs: alaw,ulaw

Restrict to IP:

Door Intercom   [i]

○ Enable   ◉ Disable

Camera URL: [Test Connection]

DTMF:

[Save] [Apply] [Cancel]

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

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

We strongly advice 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 200OK SDP:	⦿ No	○ Yes
Preferred Vocoder:	choice 1:	PCMA ▾
(in listed order)	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 ▾	