

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 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**.

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
Account Active: <input type="radio"/> No <input checked="" type="radio"/> 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:	<input checked="" type="radio"/> No <input type="radio"/> 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:	<input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)
NAT Traversal (STUN):	<input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> 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: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV	
Tel URI:	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 type="radio"/> No <input checked="" type="radio"/> 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	
Users	Edit
Groups	Details
Phones	Telephone Type: Standard Sip Telephone name: GrandHT502 Applied server address: 192.168.2.50
Modules	Functionality: Phone Password: 0mpbUN4j81hSdM73 Last known Device-IP: 192.168.2.30
Voicemail	Autoprovisioning
Conference	Last provisioning: Jan 1, 1970 12:00:00 AM
Addressbook	MAC address: ---
Phone Numbers	Firmware version: ---
Lines	Extended Configuration
Routing	NAT: default
Server	Codecs: alaw,ulaw
Statistic	Restrict to IP: ---
Interconnection	Door Intercom
Security	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Extended Settings	Camera URL: <input type="text"/> <input type="button" value="Test Connection"/> DTMF: <input type="text"/>
<input type="button" value="Save"/> <input type="button" value="Apply"/> <input type="button" value="Cancel"/>	

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:	<input checked="" type="radio"/> No	<input type="radio"/> 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:	<input checked="" type="radio"/> 6.3kbps encoding rate	<input type="radio"/> 5.3kbps encoding rate	
iLBC Frame Size:	<input checked="" type="radio"/> 20ms	<input type="radio"/> 30ms	
iLBC Payload Type:	97	(between 96 and 127, default is 97)	
VAD:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	
Symmetric RTP:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	
Fax Mode:	<input checked="" type="radio"/> T.38	<input type="radio"/> Pass-Through	
Re-INVITE After Fax Tone Detected:	<input checked="" type="radio"/> Enabled	<input type="radio"/> Disabled	
Jitter Buffer Type:	<input type="radio"/> Fixed	<input checked="" type="radio"/> Adaptive	
Jitter Buffer Length:	<input type="radio"/> Low	<input checked="" type="radio"/> Medium	<input type="radio"/> High
SRTP Mode:	<input checked="" type="radio"/> Disabled	<input type="radio"/> Enabled but not forced	<input type="radio"/> Enabled and forced
SLIC Setting:	GERMANY ▼		
Caller ID Scheme:	Bellcore/Telcordia ▼		