

# T300/T500 - Grandstream HT-702/704 ATA - How To



This article will describe the configuration of the HandyTone 702/704 behind the Gigaset T300/500 PBX system.

To guarantee a proper operation, please update the device first to the latest sw-version (V1.0.4.8, 05.02.2013).

The automatic (one-touch) setup is supported with the HT702 and HT704(PBX version 5.3.x.x or higher). The HT701 must be configured manually. The manual steps show the HT701.

## **ATTENTION**

**Only setup for this device is supported by Gigaset.**

**No further support from Gigaset in operational state for this device !!!**

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**This section describes the manual setup of the HT701, HT702 and HT704 behind the Gigaset T300/500 PBX system.**

### **For the automatic setup please scroll down.**

In order to use a Grandstream VoIP ATA to connect a fax machine or analogue telephone to the T300/T500 PBX you have to take the following steps:

1. Connect the ATA Adapter to your network and connect the powercable. To get the IP address of the HT70X connect an analog phone to the line connector of the box and dial \*\*\*. The voice menu asks you to choose the wanted 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:	GrandHT701 (the user part of an SIP address)
Authenticate ID:	GrandHT701 (can be identical to or different from SIP User ID)
Authenticate Password:	(purposely not displayed for security protection)
Name:	HT701 (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 GrandHT701, we use a random password.

Gigaset PRO Administration - Mozilla Firefox

192.168.2.50/config/display.do?url=undefined

Configuration	
Edit	
Users	Details
Groups	
Phones	Telephone Type: Standard Sip Telephone name: GrandHT701 Applied server address: 192.168.2.50 Functionality: Phone Password: rbZ3ca5MnjXFa40k Last known Device-IP: 192.168.2.24
Modules	
Voicemail	
Conference	
Addressbook	
Phone Numbers	
Lines	Autoprovisioning Last provisioning: Jan 1, 1970 12:00:00 AM MAC address: --- Firmware version: ---
Routing	Extended Configuration NAT: default Codecs: alaw,ulaw Restrict to IP:
Server	
Statistic	
Interconnection	Door Intercom <input type="radio"/> Enable <input checked="" type="radio"/> Disable Camera URL: Test Connection DTMF:
Security	
Extended Settings	

Save Apply Cancel

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

We strongly advice only to use the G.711 alaw/ulaw codec for use with connected fax machines.

Use First Matching Vocoder in 200OK 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

This section describes the automatic setup of the HT702 and HT704 behind the Gigaset T300/500 PBX system.

In order to use the one-touch provisioning, please open the admin-section of the PBX and go to *Phones*. Start here the phonesearch by pressing the 'Search'-Button in the right-lower corner

Configuration

List of configured phones

Settings Configured Phones Extended Settings

Search Rows: 10 Page 1/3

State	Device Type	Device Name	IP	Assigned Persons		
✓	DE310 IP PRO	1081.DE310IP	192.168.16.133	Lahm, Phillip	✎	✕
✓	DE410 IP PRO	1079.DE410IP	192.168.16.131	Schweinsteiger, Bastian	✎	✕
✓	DE700 IP PRO	1151.DE700IP	192.168.16.195	Oezil, Mesut	✎	✕
✓	DE900 IP PRO	1080.DE900IP	192.168.16.132	Neuer, Manuel	✎	✕
✓	N510 IP PRO	1117.N510IP	192.168.16.194	Klose, Miroslav	✎	✕
✓	N510 IP PRO	1119.N510IP	192.168.16.194	Mueller, Thomas	✎	✕
✓	N510 IP PRO	1118.N510IP	192.168.16.194	---	✎	✕
✓	N510 IP PRO	1122.N510IP	192.168.16.194	---	✎	✕
✓	N510 IP PRO	1120.N510IP	192.168.16.194	---	✎	✕
✓	N510 IP PRO	1121.N510IP	192.168.16.194	---	✎	✕

Add Additional Telephones: + Manually Search Close

You can now specify the IP-range you want to search for the device or just start the search. When the device is found it will be automatically provisioned with the SIP-account.

After this you can start assigning the accounts to the users.

## Additional Information:

## External Link

Configuration

Users

Groups

Phones

Modules

Voicemail

Conference

Addressbook

Phone Numbers

Lines

Routing

Server

Statistic

Interconnection

Security

Extended Settings

Automatic Scan for Telephones

Search now

Applied server address: 192.168.16.191

0%

IP range to search: 192.168.16.0 - 255

Configured Devices

Quantity

Type

Add Additional Telephones: Manually

Specify IP range to search

Close