

# Configuration manual

## Grandstream

Type: Analog Telephone Adapter





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This configuration manual is limited to the settings required to connect to the gnTel platform. For general manuals with detailed features we would like to refer to the manufacturers manual.

The screen shots used in this manual have been obtained using the Grandstream Analog Telephone Adapter.



## 1 Introduction

This manual describes the settings required to properly connect and work with the gnTel platform. The manual is applicable to the following type of devices: Analog Telephone Adapter.

## 2 Key terms and abbreviations

The following key terms and abbreviations are used:

<b>PI</b>	Provisioning Interface from gnTel, <a href="https://provisioning.gntel.nl">https://provisioning.gntel.nl</a>
<b>Customer code</b>	Customer code of the customer at gnTel, as stated in the PI.
<b>Trunk account</b>	Trunk account code from the SIP Account from the device, as stated in the PI. This code consists of the Customer code + 3 unique digits for each account.

## 3 Logging on to the web interface

A Grandstream device is approachable via the web interface if the PC and the Grandstream device are located in the same sub net.

Logging on to the web interface:

1. Connect the Analog Telephone Adapter (ATA) to an analog device, power supply and internet connection.
2. Pick up the handset and dial \* \* \*.
3. Chose option **2** and note the IP address down.
4. Go to a web browser and enter http:// followed by the IP address.  
**Please note:** the IP address announced contains 12 numbers, e.g. 192.168.001.010. The leading zeros need to be removed when navigating to the web interface, e.g. 192.168.1.10.
5. A pop-up appears requesting login information.
6. Log in details are either the default settings of settings made by the partner in the PI.

**Default:**

Password: admin

## 4 Trunk account details

The trunk account details can be found in the PI under 'Technical' → 'Trunk accounts'. Click on the option 'Advanced configuration' of the trunk account that will be used, the password can be found here.

▼ Trunk accounts (4)					Add account	
Account code	Allowed IP range	Created on	Options			
C 0		2016-12-30	> Edit	> Advanced configuration		
0 11		2016-06-21	> Edit	> Advanced configuration		
0 12		2017-06-14	> Edit	> Advanced configuration		

Within the advanced configuration the T.38 setting needs to be enabled and set to T.38, as shown in the figure below.





## 5 Web interface settings

The web interface allows the user to check and adjust the devices' settings.

### 5.1 Setting up a trunk account

When in the web interface, navigate to the 'FXS Port' tab. The following settings need to be adjusted.

**Grandstream Device Configuration**

<u>STATUS</u>	<u>BASIC SETTINGS</u>	<u>ADVANCED SETTINGS</u>	<u>FXS PORT</u>
Account Active:	<input type="radio"/> No <input checked="" type="radio"/> Yes		
Primary SIP Server:	<input type="text" value=""/>	(e.g., sip.mycompany.com, or IP address)	
Failover SIP Server:	<input type="text" value=""/>	(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" value=""/>	(e.g., proxy.myprovider.com, or IP address, if any)	
Allow DHCP Option 120( override SIP server):	<input checked="" type="radio"/> No <input type="radio"/> Yes		
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 type="radio"/> Keep-Alive <input checked="" type="radio"/> STUN <input type="radio"/> UPnP		
SIP User ID:	<input type="text" value="Phonenumber to Display"/>	(the user part of an SIP address)	
Authenticate ID:	<input type="text" value="PI Account Code"/>	(can be identical to or different from SIP User ID)	
Authenticate Password:	<input type="text" value="*****"/>	(purposely not displayed for security protection)	
Name:	<input type="text" value="Caller ID Name"/>	(optional, e.g., John Doe)	

Primary SIP Server: voip.gntel.nl  
Outbound Proxy: voip.gntel.nl  
SIP User ID: the trunk account code  
Authenticate ID: the same trunk account code  
Authenticate Password: The password belonging to the trunk account  
Name: Name of the account

Scrolling down in the FXS Port tab, the following settings also need to be set for a proper functioning on the gnTel platform.

Enable SIP OPTIONS Keep Alive: Yes  
Use Random SIP Port: Yes  
Use Random RTP Port: Yes  
Hold Target Before Refer: Yes

*Enable SIP OPTIONS Keep Alive:*  No  Yes  
*SIP OPTIONS Keep Alive Interval:*  (in seconds. Between 1-64800, default is 30)  
*SIP OPTIONS Keep Alive Max Lost:*  (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)  
*Layer 3 QoS:*  SIP DSCP (Diff-Serv value in decimal, default 24)  
 RTP DSCP (Diff-Serv value in decimal, default 46)  
*Local SIP port:*  (default is 5060 for UDP and TCP; 5061 for TLS)  
*Local RTP port:*  (even number between 1024-65535, default 5004)  
*Use Random SIP Port:*  No  Yes  
*Use Random RTP Port:*  No  Yes  
*Hold Target Before Refer:*  No  Yes

Fax Tone Detection Mode: Caller or Callee  
 USLIC Setting: EUROPEAN CTR21  
 Caller ID Scheme: ETS-DTMF during ringing

*Fax Tone Detection Mode:*  Caller  Callee  Caller or Callee  
*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  
*Crypto Life Time:*  Disabled  Enabled  
  
*SLIC Setting:*   
*Caller ID Scheme:*

## 5.2 Enabling or disabling STUN



STUN reserves a port for the audio traffic within the NAT table on the local router. This enables the start of audio traffic right from the start of the call. If the local router has "Symmetric NAT", STUN should be disabled!

To enable STUN, navigate to the 'Advanced settings' tab. Enter the following details, as shown in the screen shot.

- STUN Server: stun.gntel.nl
- Use STUN to detect network connectivity: Yes, total STUN response misses 3

### Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT
Admin Password:	..... (purposely not displayed for security protection)		
	802.1Q/VLAN Tag	<input type="text" value="0"/> (0-4094)	
Layer 2 QoS:	SIP 802.1p	<input type="text" value="0"/> (0-7)	
	RTP 802.1p	<input type="text" value="0"/> (0-7)	
STUN server is :	<input type="text"/> (URI or IP:port)		
Keep-alive Interval:	<input type="text" value="20"/> (in seconds, default 20 seconds)		
Use STUN to detect network connectivity:	<input type="radio"/> No		
	<input checked="" type="radio"/> Yes, total STUN response misses <input type="text" value="3"/> to restart DHCP (minimum=3)		

Navigate to the 'FXS Port' tab and search for the option 'NAT Traversal' and set this to 'STUN'.

NAT Traversal:  No  Keep-Alive  STUN  UPnP

### 5.3 Date and time settings

Time settings can be found in the web-interface in the 'Advanced settings' tab.

- NTP Server: ntp.gntel.nl

NTP Server:  (URI or IP address)

Next the Time zone has to be set, which can be done under the 'Basic Settings' tab.

Time Zone:

### 5.4 Disable logging missed calls

If a user doesn't desire to show missed calls, this feature can be disabled via the web interface. Navigate to the 'FXS Port' tab and scroll down to the following options.

Disable Call-Waiting:	<input type="radio"/> Yes	Disable Call-Waiting:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Disable Call-Waiting Caller ID:	<input type="radio"/> Yes	Disable Call-Waiting Caller ID:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Disable Call-Waiting Tone:	<input type="radio"/> Yes	Disable Call-Waiting Tone:	<input type="radio"/> No	<input checked="" type="radio"/> Yes