gnTel

Configuration manual

Grandstream
Type: Analog Telephone Adapter

Contents

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 aplicable to the following type of devices: Analog Telephone Adapter.

2 Key terms and abbreviations

The following key terms and abbreviations are used:

PI	Provisioning Interface from gnTel, `https://provisioning.gntel.nl`
Customer code	Customer code of the customer at gnTel, as stated in the PI.
Trunk account	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:

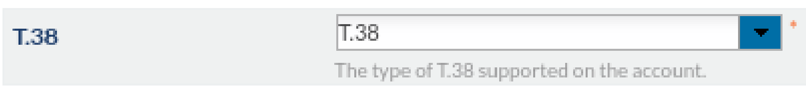
1. Connect the Analog Telephone Adapter (ATA) to an analog device, power supply and internet connection.
2. Pick up the handset and dial * * *.
3. Chose option 2 and note the IP address down.
4. Go to a web browser and enter http:// followed by the IP address.
   Please note: the IP address announced 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.

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

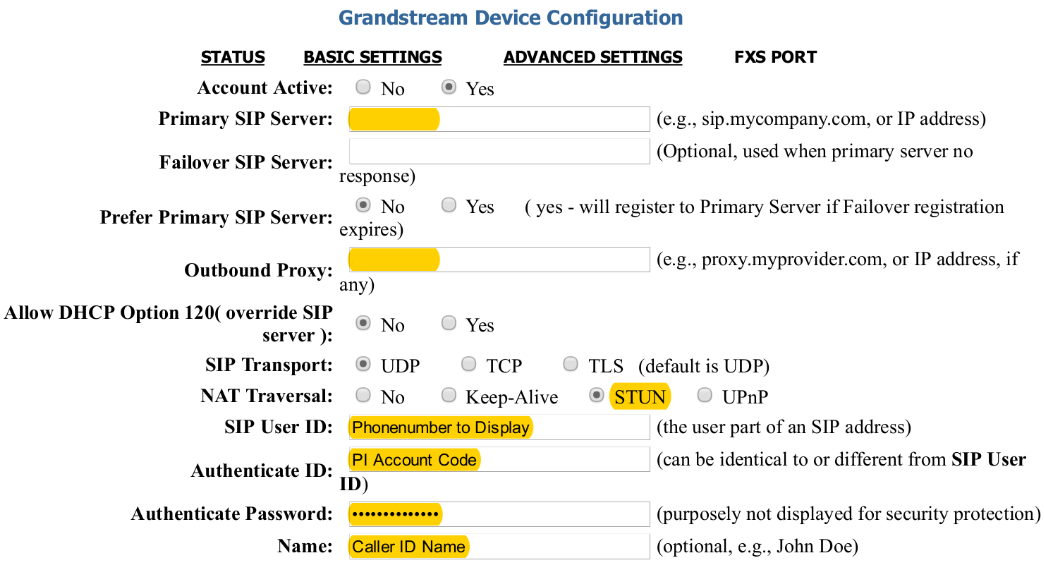


5 Web interface settings

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.



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:	30 (in seconds. Between 1-64800, default is 30)
SIP OPTIONS Keep Alive Max Lost:	3 (Number of max lost packets for SIP OPTIONS Keep Alive before re-registration. Between 3-10, default is 3)
Layer 3 QoS:	26 SIP DSCP (Diff-Serv value in decimal, default 24) 46 RTP DSCP (Diff-Serv value in decimal, default 46)
Local SIP port:	5060 (default is 5060 for UDP and TCP; 5061 for TLS)
Local RTP port:	5004 (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

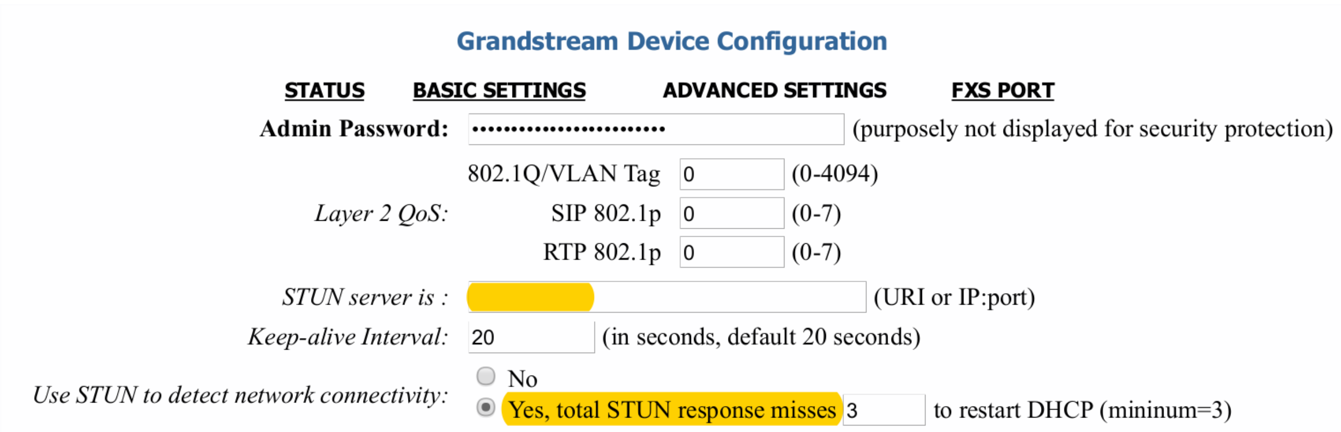
Fax Tone Detection Mode:	Caller Callee Caller or Callee
Re-INVITE After Fax Tone Detected:	Enabled Disabled
Jitter Buffer Type:	Fixed Adaptive
Jitter Buffer Length:	Low Medium High
SRTP Mode:	Disabled Enabled but not forced Enabled and forced
Crypto Life Time:	Disabled Enabled
SLIC Setting:	EUROPEAN CTR21
Caller ID Scheme:	ETSI-DTMF during ringing

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



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

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

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

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:	Yes
Disable Call-Waiting Caller ID:	Yes
Disable Call-Waiting Tone:	Yes