

PortaDial Door Intercom

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Tested with: Gigaset T300/T500 PRO: Software version 4.6.5.0
PortaDial type 1 with 1 pushbutton
PortaDial Interface 3 SIP



1. Create SIP account in Gigaset T300/T500

Go to: Configuration - Phones - Add additional Telephones

Users	Details	
Groups	Telephone Type: Standard Sip	Functionality: Phone
Phones	Telephone name: Portadial	Password: <SIP Password> Random password
Modules	Server IP-Address: 192.168.2.10	Last known Device-IP: 192.168.2.153
Voicemail	Extended Configuration	
Conference	Options	
Addressbook	nat: --choose--	
Phone numbers	Codecs: alaw,ulaw	
Lines	restr. to IP:	

Use the settings like in the example above. Use the Password generator to create a secure Password

2. Open the Fasttel Wizard Elite door intercom web-interface

Go to *Advanced Settings 1*

SIP Server:	IP address of the Gigaset T300/T500
Outbound Server:	IP address of the Gigaset T300/T500
SIP User ID:	Telephone name defined in Gigaset T300/T500
Authenticate ID:	Telephone name defined in Gigaset T300/T500
Authenticate Password:	Password defined in Gigaset T300/T500
Send DTMF:	Only activate the option via RTP(RFC2833)

Grandstream Device Configuration

STATUS **BASIC SETTINGS** **ADVANCED SETTINGS 1** **ADVANCED SETTINGS 2**

Admin Password: (purposely not displayed for security protection)
SIP Server: (e.g., sip.mycompany.com, or IP address)
Outbound Proxy: (e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID: (the user part of an SIP address)
Authenticate ID: (can be identical to or different from **SIP User ID**)
Authenticate Password: (purposely not displayed for security protection)
Name: (optional, e.g., John Doe)
Home NPA:

Advanced Options:

Preferred Vocoder: (in listed order)
choice 1:
choice 2:
choice 3:
choice 4:
choice 5:
choice 6:
choice 7:
G723 rate: ☒ 6.3kbps encoding rate ☐ 5.3kbps encoding rate
iLBC frame size: ☒ 20ms ☐ 30ms
iLBC payload type: (between 96 and 127, default is 97)
Silence Suppression: ☒ No ☐ Yes
Voice Frames per TX: (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)
Fax Mode: ☒ T.38 (Auto Detect) ☐ Pass-Through
Layer 3 QoS: (Diff-Serv or Precedence value)
Layer 2 QoS: 802.1Q/VLAN Tag 802.1p priority value (0-7)
Allow incoming SIP messages from SIP proxy only: ☒ No ☐ Yes
Use DNS SRV: ☒ No ☐ Yes
User ID is phone number: ☒ No ☐ Yes
SIP Registration: ☒ Yes ☐ No
Unregister On Reboot: ☐ Yes ☒ No
Register Expiration: (in seconds. default 1 hour, max 45 days)
Early Dial: ☒ No ☐ Yes (use "Yes" only if proxy supports 484 response)

Allow outgoing call without Registration: ☒ No ☐ Yes

Dial Plan Prefix: (this prefix string is added to each dialed number)

No Key Entry Timeout: (in seconds, default is 4 seconds)

Use # as Dial Key: ☐ No ☒ Yes (if set to Yes, "#" will function as the Dial key)

local SIP port: (default 5060)

local RTP port: (1024-65535, default 5004)

Use random port: ☒ No ☐ Yes

SIP Registration Failure Retry Wait Time: (in seconds. Between 1-3600, default is 20)

NAT Traversal: ☐ No ☒ Yes, STUN server is: (URI or IP:port)

keep-alive interval: (in seconds, default 20 seconds)

Use NAT IP (used in SIP/SDP message if specified)

Use STUN keep-alive to detect networks connectivity: ☒ No ☐ Yes, total STUN response misses (minimum=3) before restart

Proxy-Require:

SUBSCRIBE for MWI: ☒ No, do not send SUBSCRIBE for Message Waiting Indication ☐ Yes, send periodical SUBSCRIBE for Message Waiting Indication

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Enable Call Features: ☐ No ☒ Yes
 (if yes, call features using star codes will be supported locally)

Use Bell-style 3-way Conference: ☒ No ☐ Yes (if Yes, *23 will be disabled)

Disable Call-Waiting: ☒ No ☐ Yes

Disable Call-Waiting Caller-ID: ☐ No ☒ Yes

Send DTMF: ☐ in-audio ☒ via RTP (RFC2833) ☐ via SIP INFO

DTMF Payload Type:

Send Flash Event: ☒ No ☐ Yes (Flash will be sent as a DTMF event if set to Yes)

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Go to Advanced Settings 2

NTP Server:	IP address of the Gigaset T300/T500
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Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS 1 ADVANCED SETTINGS 2

Onhook Threshold:

FXS Impedance:

Caller ID Scheme:

Onhook Voltage:

Polarity Reversal: ☐ No ☒ Yes (reverse polarity upon call establishment and termination)

NTP Server: (URI or IP address)

Send Anonymous: ☒ No ☐ Yes (caller ID will be blocked if set to Yes)

Anonymous Method: ☒ Use From Header ☐ Use Privacy Header

Time to ring:

Special Feature:

CBCOM Encode SIP RT(C)P T38

CBCOM Encoder 1.1 Key: (not shown for security reason)

Syslog Server:

Syslog Level:

Session Expiration: (in seconds, default 180 seconds)

Min-SE: (in seconds, default and minimum 90 seconds)

Caller Request Timer: ☐ Yes ☒ No (Request for timer when making outbound calls)

Callee Request Timer: ☐ Yes ☒ No (When caller supports timer but did not request one)

Force Timer: ☐ Yes ☒ No (Use timer even when remote party does not support)

UAC Specify Refresher: ☐ UAC ☐ UAS ☒ Omit (Recommended)

UAS Specify Refresher: ☒ UAC ☐ UAS (When UAC did not specify refresher tag)

Force INVITE: ☐ Yes ☒ No (Always refresh with INVITE instead of UPDATE)

Firmware Upgrade and Provisioning: Upgrade Via ☐ TFTP ☒ HTTP

Firmware Server Path:

Configure Server Path:

Firmware File Prefix:

Firmware File Postfix:

Config File Prefix:

Config File Postfix:

Retry-after(minutes): if server unavailable

Automatic Upgrade:

☒ No ☐ Yes, check upgrade every minutes (default 7 days)

- ☒ Always Check for New Firmware
☐ Check New Firmware only when F/W pre/suffix changes
☐ Always Skip the Firmware Check

Firmware Key: (in Hexadecimal Representation)

Authenticate Conf File: ☒ No ☐ Yes (cfg file would be authenticated before acceptance if set to Yes)

Lock keypad update: ☒ No ☐ Yes (configuration update via keypad is disabled if set to Yes)

Allow conf SIP Account
in Basic Settings: ☒ No ☐ Yes

Override MTU Size:

Volume Amplification: TX RX

Powerline Ring Tone: Frequency (Hz) ON (x10ms) OFF (x10ms)
(Allowed: 15-100) (Allowed: 5-800) (Allowed: 5-800)

Call Progress Tones:

	Frequency 1 (Hz)	Frequency 2 (Hz)	ON (x10ms) (C1;C2;C3)	OFF (x10ms) (C1;C2;C3)
Dial Tone	440	440	0	0
Recall Dial Tone	440	440	10	10
Message Waiting	440	440	10	10
Confirmation	440	440	10	10
Audible Ringing	440	440	100	400
Busy Tone	440	440	50	50
Reorder Tone	440	440	50	50
Receiver Offhook Tone	1400	2600	10	10

Disable Line Echo Canceller (LEC): ☒ No ☐ Yes (If set Yes, echo canceller is not used)

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Basic Settings

Grandstream Device Configuration

STATUS **BASIC SETTINGS** ADVANCED SETTINGS 1 ADVANCED SETTINGS 2

End User Password: (purposely not displayed for security protection)

Web Port: (default for HTTP is 80)

IP Address: ☒ dynamically assigned via DHCP (default) or PPPoE:
(will attempt PPPoE if DHCP fails and following is non-blank)

DHCP hostname:

DHCP domain:

DHCP vendor class ID:

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

Preferred DNS server:

☐ statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone: ▼

Daylight Savings Time: ☒ No ☐ Yes

Optional Rule:

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Status

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS 1 ADVANCED SETTINGS 2

MAC Address: 00:0B:82:2A:CD:79

IP Address: 192.168.2.153

Product Model: HT287 Rev4.1

Software Version: Program-- 1.1.0.37 Bootloader-- 1.1.0.1 HTML-- 1.1.0.37 VOC-- 1.0.0.13

System Up Time: 0 day(s) 2 hour(s) 45 minute(s)

Registered: Yes

PPPoE Link Up: disabled

NAT: detected NAT type is open Internet

NAT Mapped IP: 0.0.0.0

NAT Mapped Port: 0

Total Inbound Calls: 0

Total Outbound Calls: 0

Total Missed Calls: 0

Total Call Time (in minutes): 0

Total SIP Message Sent: 177

Total SIP Message Received: 183

Total RTP Packet Sent: 0

Total RTP Packet Received: 0

Total RTP Packet Loss: 0

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3. Gigaset User account.

Without a Gigaset user account, the door intercom can be used to dial an internal number. Incoming and Outgoing calls are not possible.

Incoming calls to the door intercom are only possible when the door intercom is assigned to a user.

In the Gigaset T300/T500 Go to: Configuration - Users and create a new user, only to be used for the door intercom (user license).

Users	User DataRedirectPhonesPhone NumbersGroup MembershipRights									
Groups	Primary Phone: SIP/PortadialCall Waiting Indication: ☒ activate									
Phones										
Modules	<table><thead><tr><th>Telephone</th><th>Ringing Number</th><th>Active</th></tr></thead><tbody><tr><td>SIP/Portadial</td><td></td><td>☒</td></tr></tbody></table>				Telephone	Ringing Number	Active	SIP/Portadial		☒
Telephone	Ringing Number	Active								
SIP/Portadial		☒								
Voicemail										