# **Fasttel Door Intecom**

## Fasttel Door intercom - Interoperability

Tested with: Gigaset T300/T500 PRO: Software version 4.6.5.0 Fasttel Wizard Elite

## Create SIP account in Gigaset T300/T500

Go to: Configuration - Phones - Add additional Telephones

### blocked URL

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

## Open the Fasttel Wizard Elite door intercom web-interface

Go to Advanced Settings

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)		
NTP Server	IP address of the Gigaset T300/T500		

Grandstream Device Configuration				
51	ATUS BASIC SETTINGS	ADVANCED SETTINGS		
Admin Password:		(purposely not displayed for security protection)		
SIP Server:	192.168.2.10	(e.g., sip.mycompany.com, or IP address)		
Outhound Proxy:	192.168.2.10	(e.g., proxy.myprovider.com, or IP address, if any)		
SIP User ID:	fasttel	(the user part of an SIP address)		
Authon ticata ID:	fasttel	(can be identical to or different from <b>SIP User</b>		
Autoenticale ID:	<b>D</b> )			
Authenticate Password:		(purposely not displayed for security protection)		
Name:	Deurintercom	(optional, e.g., John Doe)		
Home NPA:				
Advanced Options:	desire to become extended	DOLLAR		
(in listed order)	choice 1: current setting is choice 2: current setting is "	G723" V		
	choice 3: current setting is "	G723" 💌		
	choice 4: current setting is "	G729" 💙		
	choice S: current setting is "	G726-32" ¥		
	choice 7: current setting is			
G723 rate:	<ul> <li>6 3kbps encoding rate</li> </ul>	S 3kbps encoding rate		
iLBC frame size:	<ul> <li>20ms</li> <li>30ms</li> </ul>			
iI.RC novload type:	97 Chetween 96 and	127 default is 97)		
Silence Suppression:		,,		
Wrice Frames ver IX:	2 (up to 10/20/32/	64 for G711/G726/G723/other codecs respectively)		
Fax Mode:	• T.38 (Auto Detect)	Pass-Through		
Layer 3 QoS:	48 (Diff-Serv or Precedence value)			
Layer 2 QoS:	802.1Q/VLAN Tag 0	802.1p priority value 0 (0-7)		
Allow incoming SIP				
messages	💿 No  🔘 Yes			
from SIP proxy only:	0-0-			
Use LIVS SRV:	⊙ No O Yes			
User ID is phone number:	⊙ No O Yes			
SIP Registration:	Ves Vo			
Unregister On Reboot:	Ves 🕑 No			
Register Expiration:	3600 (in seconds. defa	ult 1 hour, max 45 days)		
Early Dial:	💌 No 💛 Yes (use "Y	es" only if proxy supports 484 response)		
Allow outgoing call without Registration:	⊙ No OYes			
Dial Plan Prefix:	(this prefix string	g is added to each dialed number)		
No Key Entry Timeout:	1 (in seconds, defa	ult is 4 seconds)		
Use #as Dial Key:	🔘 No 🛛 💿 Yes (if set f	to Yes, "#" will function as the Dial key)		
local SIP port:	5060 (default 5060)			
local RTP port:	5004 (1024-65535, de	fault 5004)		
Use random port:	💿 No 🙁 Yes			
NAT Traversal:	💿 No			

	Vec STIIN convertice (IIPL or IP: port)				
keen aline internal:	20 (in seconds, default 20 seconds)				
The MAT ID	(wed in SIP(DP mercure if creatified)				
Drave Pravira:	(Bed it off rob T the stage it specified)				
STIRS CRIPTIRE for MUT.					
50D5C22D2 J07 22W2.	No, do not send SUBSCRIBE for Message Waiting Indication				
	View View View View View View View View				
Offhook Auto-Dial:	User ID/extension to dial automatically when				
Enable Call Features:	No Sec (if Yes, Call Forwarding & Call-Waiting Disable are guported				
	locally)				
Use Bell-style	No Ves (if Ves *23 unil he disabled)				
3-way Conference:					
LASAOIE Call-waining:	· ● No ● Yes				
Send DIMF:	🗋 in-audio 🗹 via RTP (RFC2833) 🗋 via SIP INFO				
DIMF Payload Type:	101				
Send Flash Event:	• No • Yes (Flash will be sent as a DTMF event if set to Yes)				
Onhook Threshold:	800 ms				
FXS Impedance:	600 Ohm (North America)				
Caller ID Scheme:	ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)				
Onhook Voltage:	36V 💌				
Polarity Reversal:	• No • Yes (reverse polarity upon call establishment and termination)				
NTP Server:	192.168.2.10 (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:	60 seconds 💌				
Special Feature:	Standard 💙				
Syslog Server:					
Syslog Level:	NONE				
Session Expiration:	180 (in seconds. default 180 seconds)				
Min-SE:	90 (in seconds. default and minimum 90 seconds)				
Caller Request Timer:	○ Yes ③ No (Request for timer when making outbound calls)				
Callee Request Timer:	○ Yes				
Force Timer:	○ Yes				
UAC Specify Refresher:	○ UAC ○ UAS ⊙ Omit (Recommended)				
UAS Specify Refresher:	• UAC UAS (When UAC did not specify refresher tag)				
Force DWITE:	• Yes • No (Always refresh with INVITE instead of UPDATE)				
Firmware Upgrade and	IInerade Via 💿 TETP 🚫 HTTP				
Provisioning:	Firmware Server Path: 192.168.13.200				
	Config Server Path: fm.grandstream.com/gs				
	Firmware File Prefix: Firmware File Postfix:				
Config fue Frenx: Config fue Postfix:					
	Automatic Upgrade:				
	• No • Yes, check for upgrade every 10080 minutes (default 7 days)				

	Check Nev	v Firmware onl in the Firmwar	ly when F/W pr e Check	e/suffix change	25
Firmware Key:	(in Hexadecimal Representation)				
Authenticate Conf File:	No Ves (cfg file would be authenticated before acceptance if set t Yes)				
Lock keypad update:	No O Yes (configuration update via keypad is disabled if set to Yes)				
Allow conf SIP Account in Basic Settings:	💿 No 🛛 🔿	Yes			
Override MTU Size:	0				
Volume Amplification:	TX OdB default 😒	RX -4dB	~		
Call Progress Tones:		Frequency l (Hz)	Frequency 2 (Hz)	ON (x10ms) (C1;C2;C3)	OFF (x10ms) (C1;C2;C3)
	Dial Tone	425	425	0	0
	Recall Dial Tone	425	425	10	10
	Message Waiting	425	425	10	10
	Confirmation	425	425	10	10
	Audible Ringing	425	425	100	400
	Busy Tone	425	425	50	50
	Reorder Tone	425	425	25	25

#### **Basic Settings**

blocked URL

#### Status

blocked URL

### 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)

blocked URL

## Programming the Fasttel using DTMF.

Call the extension number of the Doorintercom.

The Fasttel will answer the call and will play DTMF tones.

Then you enter the programming access code: \*1996\*

But, the Gigaset T300/T500 already uses the code \*1 for Call recording. This means when you press \*1996\*, directly after the \*1, the PBX will start recording the call and you will not be able to enter the programming mode. You need to change the \*1 code in the Gigaset T300/T500 to an other code.

How to change the \*1 code.

- Login the PBX using SSH.
  Go to the directory /home/asterisk
  Open the file features.conf and change the automon => \*1 to \*4
  Restart the PBX services to activate your chang

#### [featuremap]

blindxfer => NONEXISTINGEXTENSION1 ; Blind transfer, default is #

·disconnect -> *0	Disconnect
	. Diaconnect

#### automon => \*4 ; One Touch Record

;atxfer => \*2 ; Attended transfer

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    - o Gigaset User account.
      o Programming the Fasttel using DTMF.

