# T440/T640 Simplecall Business

# Simplecall Business

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Feature	
Outgoing Calls	Yes
Incoming Calls	Yes
CLIP incoming	Yes
CLIP outgoing	Yes
Call Forwarding	Yes
Call Transfer	Yes
Call Waiting	Yes
DTMF	Yes
Anonymous Call	Yes
A-number forwarding	-



- Simplecall Business
  - Gigaset T440/T640 PRO settings.
  - Prerequisites
  - SIP Gateway
  - SIP GatewayGateway Group
  - Inbound Routes
  - Outbound Routes
  - o sip.conf File



# Important

If outbound CLI is required then from Version 1.0.1 only

# Gigaset T440/T640 PRO settings.

When configuring a SIP trunk into the T640/440 you must complete the following settings:

- 1. Prerequisites
- 2. SIP Gateway
- 3. Gateway Group
- 4. Inbound Routes
- 5. Outbound Routes
- 6. Amend sip.conf file

# Prerequisites

Simplecall SIP trunks are an IP authenticated connection.

This will require that the installation site has a permanent public IP address, and that the router with this address has port forwarded 5060 to the T640 /440.

# SIP Gateway

The SIP Gateway are the settings to connect to your ISP, for Simplecall

The SIP Gateway setting can be found in Administration/Routes/SIP gateways

#### Title

Information only, enter Simplecall

#### Name

Created by the T640/440

# Registar

The IP address of Simplecall = 91.146.112.10

# Proxy

Not needed

# User

IP authenticated account therefore N/A

# **Password**

IP authenticated account therefore N/A

# Allow outbound calls

Enable, assuming you wish to dial out.

# Register

IP authenticated account therefore off

# Language

en - English(en-US)

# Dial command

SIP/{prefix}{number:1}@{gateway}

Remove {prefix} if no prefix is configured in Outbound routes

Remove :1 if no trunk access digit is configured in Outbound routes

# Source of Destination number

INVITE request line

#### Title Simplecall Group Name gw\_2\_simplecall Gateway group configured for this account 91.146.112.10 Registar Port Proxy [1] 5060 User [2] NAT Password yes ON Allow outbound calls Redirect RTP stream Register Do not reroute RTP stream en - English (en-US) Language **Check availability** Dial comand [3] SIP/{number:1}@{gateway} Off Source of destination number INVITE request line Simultaneous calls Group [4] Simplecall set to limit the number of calls, 0 = unlimited 5060 Port [5] DTFM mode NAT rfc2833-RTP meta data Redirect RTP stream Do not reroute RTP stream (default) From user Check availability If you wish to send anonymous CLI, set this option to anonymous. 0 for unlimited (default) Simultaneous calls This is a global setting and cannot be turned off/on on a per user basis DTFM mode rfc2833 - RTP meta data From Domain From user The dedicated Public IP address the account has been allocated to. From Domain Your Public IP T38 support ON T38 support On no - Deactivated (default) Insecure Update remote party ID (CLIP) Use P-Asserted-Identity header no - deactivated Trust remote party ID Update remote part ID (CLIP) Ø G.711a ☐ G.711u Codecs [7] ■ G.722 G.729 Use P-Asserted-Identity header ☐ GSM H.261 Simplecall also authenticate on your CLI via the P-Asserted-ID Allowed IP subnet 91.146.112.10 Trust remote party ID no - deactivated Advanced parameters Codecs Value Select your requirements session-timers = originate session-expires = 1800 Allowed IP subnet

Simplecall's IP address 91.146.112.10/0

Always limit this setting to your ISP's IP address

**Advanced Parameters** 

Simple call requires a session timer for calls longer than 30 minutes.

session-timers = originate

session-expires = 1800

# **Gateway Group**

The Gateway group amends the inbound/outbound CLI and number received.

For inbound calls the Gateway group receives the call from the SIP gateway

For outbound calls the Gateway group receives the call from the Outbound routes

The Gateway group setting can be found in Administration/Routes /Gateway groups

# Title

Information only, enter Simplecall

#### Permit inbound calls

Enable, assuming you wish to receive calls

# **Outbound caller ID**

Whatever you receive you wish to send, therefore

S/ ^(.\*) / \$1 /

# **Asserted Identity**

Whatever you receive you wish to send, therefore

S/ ^(.\*) / \$1 /

#### **Inbound DIDs**

Whatever you receive you wish to send, therefore

S/ ^(.\*) / \$1 /

# Inbound caller ID

Whatever you receive you wish to send, therefore

S/ ^(.\*) / \$1 /

If you are using a trunk access code insert it here for redial access on the handsets.

For example if you dial 9 for an outside line:

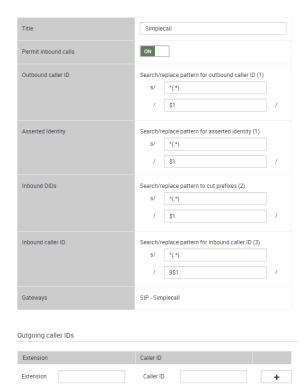
S/ ^(.\*) / 9\$1 /

# Gateways

The Simplecall SIP gateway allocated to this Gateway group

# **Outgoing caller IDs**

Set the specific CLI for each extension here



# **Inbound Routes**

The inbound routes maps the the DDI number you receive to a destination within the T640/440

The Inbound route setting can be found in Administration/Routes /Inbound routes

#### **Gateway Group**

Which Gateway group you wish to configure, select Simplecall



# Advanced options

Enable to see all options

# Rule

Information only, name for this inbound route

# Active

Displays if rule is active.

Route can be turned off/on

#### Date

Date range for rule to be active

# Weekdays

Day of week for rule to be active

# Time

Time for rule to be active

# **Profile**

Which profile is the rule active

# Pattern

The DDI number you have been sent from the ISP.

For Simplecall this will be the full number beginning with 0

# Target

The Extension number of the user/ Hunt Group/ Queue/ IVR you wish to route the call to.

# **Outbound Routes**

The outbound routes match the digits you dial to an external trunk.

The Outbound routes setting can be found in Administration/Routes /Outbound routes

# **Advanced Options**

Enable to see all options

# Rule

Information only, name for this outbound route

# Active

Enable/Disable rule

# Weekdays

Select which day the rule is active

#### Time

Select what time the rule is active

# Pattern



The number dialed

For example all numbers beginning with 0:

^0.\*

For trunk access of 9:

^90.\*

# Group

Permission group allocated to this rule

# **Gateway Group**

The Gateway group for the call to be presented to up to a maximum of three.

#### **Add Prefix**

Digits to add to the number dialed (local STD), this will be inserted in the SIP gateway dial command if it has {prefix}

# sip.conf File

For correct operation with Simplecall the sip.conf file needs additional confuguration

The sip.conf file can be found:

opt/galilei/etc/asterisk/

Under the [general] insert:

externip=<the public IP address>

localnet=<x.x.x.x/x> the local subnet for example localnet=192.168.1.0 /24

This amends the SDP message contact information to be the public address for all traffic apart from traffic to the localnet setting

Once you have amended the file either reboot the T640/440 or use the Asterisk CLI and the reload command



This is a global setting and will effect all SIP gateways

;callerid-asterisk callerid-galilei context-default ;allowguest-no allowguest-no autocreatepeer-no ;allowoverlap-no ;allowtransfer-no ; default "asterisk" ; Default context for incoming calls ; Allow or reject guest calls (default is yes) ; Disable overlap dialing support. (Default is yes)
; Disable all transfers (unless enabled in peers or users)
; Default is enabled
; Realm for digest authentication
; defaults to "asterisk". Thy ous et a system name in
; asterisk.comf, it defaults to that system name
; Realam MSIT be globally undque according to RFC 3261
; Set this to your host name or domain name ;realm=mydomain.tld realm=galilei.local bindport=5060 realmegalilei.local bindport=5000 ; UDP Port to bind to (SIP standard port is 5060) bindport=5000 ; bindport is the local UDP port that Asterisk will listen on ;udpbindaddr=0.0.0 ; IP address to bind UDP listen socket to (0.0.0.0 binds to all) ; by the standard port number, 192.168.1.1:5002 (default is port 5060) #exec /opt/galilei/etc/asterisk/Dindaddr.conf.pbp sip