

Interop T640/T440 Ziggo zakelijk

Ziggo zakelijk

Valid for T440/T640 Firmware version 2.0.6 or higher

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	No*
A-number forwarding	Yes



- [Ziggo zakelijk](#)
 - [Gigaset T440/T640 settings](#).
 - Gateway group:
 - SIP Gateway:
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Important

* Outgoing Anonymous calls do not work at the moment, we are working on it, as the pilot number is shown as CLIP on the receiving device.

Gigaset T440/T640 settings.

Gateway group:

In the Gigaset PBX go to "Administration" -> "Routes" -> "Gateway groups" - Enter the desired name and click on "Create new group":

Gateway groups

Gateway group	Gateways	Caller IDs		
<input type="text"/>				<input type="button" value="Create new group"/>


SIP Gateway:

In the Gigaset PBX go to "Administration" -> "Routes" -> "SIP Gateways" - At "Gateway" enter the desired name and click on "Create new gateway":

Gateway	<input type="text"/>	<input type="button" value="Create new gateway"/>
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Fill out the provider details:

Edit SIP gateway : Ziggo_SBC

Title	<input type="text" value="Ziggo_SBC"/>
Name	<input type="text" value="gw_20_ziggosbc"/>
Registrar	<input type="text" value="sbc-sip.sipnl.net"/>
Proxy [1]	<input type="text"/>
User [2]	<input type="text" value="31 [REDACTED]"/>
Password	<input type="password" value="....."/> 
Allow outbound calls	<input checked="" type="checkbox"/> ON
Register	<input checked="" type="checkbox"/> ON
Language	<input type="text" value="nl - Dutch (nl-NL)"/>
Dial command [3]	<input type="text" value="PJSIP/{prefix}{number}@{gateway}"/>
Transport name	<input type="text" value="default-udp"/>
Source of destination number	<input type="text" value="INVITE request line"/>
Group [4]	<input type="text" value="Ziggo_SBC"/>
Port [5]	<input type="text" value="5060"/>
NAT	<input type="text" value="yes"/>



Important

With the "Dial command" in above screenshot there is no need to dial an extra 0 for an outside line

If it is wanted to select an outgoing line with an extra 0 the "Dial command" is as following: PJSIP/{prefix}{number:1}@[gateway]

Redirect RTP stream	Do not reroute RTP stream (default)
Check availability	ON
Simultaneous calls	0 0 for unlimited (default)
DTMF mode	rfc4733 - RTP meta data
From user	
From Domain	
T38 support	OFF
Update P-Asserted-Identity (CLIP)	no - Deactivated (default)
Update remote party ID (CLIP)	no - Deactivated (default)
Trust remote party ID	no - Deactivated (default)
Codecs [6]	<div><div><input type="checkbox"/> G.722</div><div><input type="checkbox"/> G.729</div><div><input checked="" type="checkbox"/> G.711a</div><div><input type="checkbox"/> G.711u</div></div> <div><div><input type="checkbox"/> GSM</div><div><input type="checkbox"/> H.261</div><div><input type="checkbox"/> H.263</div></div> <div><div><input type="checkbox"/> H.263+</div></div>

Advanced parameters

Value	
inband_progress=yes	
Value	<div><input type="text"/> +</div>



Important

Enter the advanced parameter "inband_progress=yes" in the Value field and press on the "+", the setting will be active after a reboot of the system or reload of Asterisk.