

Siemens Gigaset 580

Configuration Notes

Version 2, 2012-09-19

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1 BCS Configuration

Enable third-party SIP phone for the BCS user.

3rd party SIP phone

Enable 3rd party SIP phone

Username

Password *

Retype password *

Appear as Analog phone Desk phone

Ensure that call routing for user includes routing to Desktop phone/Analog phone.

2 Siemens Gigaset Configuration

1. Open Siemens Gigaset web administration : <http://www.gigaset-config.com>
2. Go to Home > Settings > Telephony > Connections
3. Click Edit for IP1.

Fill in the account details as below using the authentication details for third-party SIP phone:

Note:

- STUN server and, Outbound server address: public service address (Edge node)
- STUN server port: 5060
- Domain: user's domain

Home	Settings	Status	Log Off
Network			
Telephony			
Connections			
Audio			
Number Assignment			
Call Divert			
Dialling Plans			
Network Mailboxes			
Advanced VoIP settings			
Messaging			
Info Services			
Directories			
Management			
		1. IP Connection	
		Assign a connection name or actual phone number for identification.	
		Connection Name or Number:	<input type="text" value="IP1"/>
		VoIP Configuration / Profile Download	
		Start Configuration Assistant	
		Provider:	Other Provider
		Profile Version:	
		Personal Provider Data	
		Authentication name:	<input type="text" value="device.mikael"/>
		Authentication password:	<input type="password" value="*****"/>
		Username:	<input type="text" value="device.mikael"/>
		Display name:	<input type="text" value="Mikael"/>
		Hide Advanced Settings	
		General data of your service provider	
		Domain:	<input type="text" value="company.com"/>
		Proxy server address:	<input type="text"/>
		Proxy server port:	<input type="text" value="5060"/>
		Registration server:	<input type="text"/>
		Registration server port:	<input type="text" value="5060"/>
		Registration refresh time:	<input type="text" value="180"/> sec
		Network data for your service provider	
		STUN enabled:	<input checked="" type="radio"/> Yes <input type="radio"/> No
		STUN server address:	<input type="text" value="bcs.serviceprovider.com"/>
		STUN server port:	<input type="text" value="5060"/>
		STUN refresh time:	<input type="text" value="240"/> sec
		NAT refresh time:	<input type="text" value="20"/> sec
		Outbound proxy mode:	<input checked="" type="radio"/> Always <input type="radio"/> Automatic <input type="radio"/> Never
		Outbound server address:	<input type="text" value="bcs.serviceprovider.com"/>
		Outbound proxy port:	<input type="text" value="5060"/>
		Set Cancel Delete Connection	

4. Click Audio and then 'Own Codec preference'. Remove all selected codecs except G.711 a-law (may differ depending on deployment)

Settings for individual VoIP connections

IP1

Volume for this VoIP Account: Low Normal High

Selected codecs

G.711 a law

Available codecs

G.711 μ law
G.728
G.729
G.722

< Add
Remove >
Up
Down

IP2

Volume for this VoIP Account: Low Normal High

Selected codecs

G.711 a law

Available codecs

G.711 μ law
G.728
G.729
G.722

< Add
Remove >
Up
Down

Gigaset.net

Volume for this VoIP Account: Low Normal High

Selected codecs

G.722
G.711 μ law
G.711 a law
G.726
G.729

Available codecs

< Add
Remove >
Up
Down

Settings for Codecs

able Annex B for codec G.729: Yes No

- Go to "Number Assignment" and select only to ring only on IPx for incoming calls. Otherwise all handsets connected to the same base will ring for incoming calls

Home
Settings
Status
Log Off

Network

Telephony

- Connections
- Audio
- Number Assignment
- Call Divert
- Dialling Plans
- Network Mailboxes
- Advanced VoIP settings

Messaging

Info Services

Directories

Management

?

Select the connection for outgoing calls and also one or more connections for incoming calls for each handset.

Handsets

INT 1 Name

Connection	for outgoing calls	for incoming calls
IP1	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>
IP2	<input type="radio"/>	<input type="checkbox"/>
Gigaset.net	<input type="radio"/>	<input type="checkbox"/>
Fixed Line	<input type="radio"/>	<input type="checkbox"/>
Select line for each outgoing call	<input type="radio"/>	

INT 2 Name

Connection	for outgoing calls	for incoming calls
IP1	<input type="radio"/>	<input type="checkbox"/>
IP2	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>
Gigaset.net	<input type="radio"/>	<input type="checkbox"/>
Fixed Line	<input type="radio"/>	<input type="checkbox"/>
Select line for each outgoing call	<input type="radio"/>	

Alternative Connection

Alternate connection in case the VoIP connection is offline.

Automatic Fallback to Fixed Line: Yes No

Call Manager

Select the connection and the associated handset for your PC Call Manager.

Connection	Enable Call Manager	Handset
IP1	<input type="text" value="No"/> ▼	<input type="text" value="Mikael"/> ▼
IP2	<input type="text" value="No"/> ▼	<input type="text" value="Jonas"/> ▼

Set
Cancel

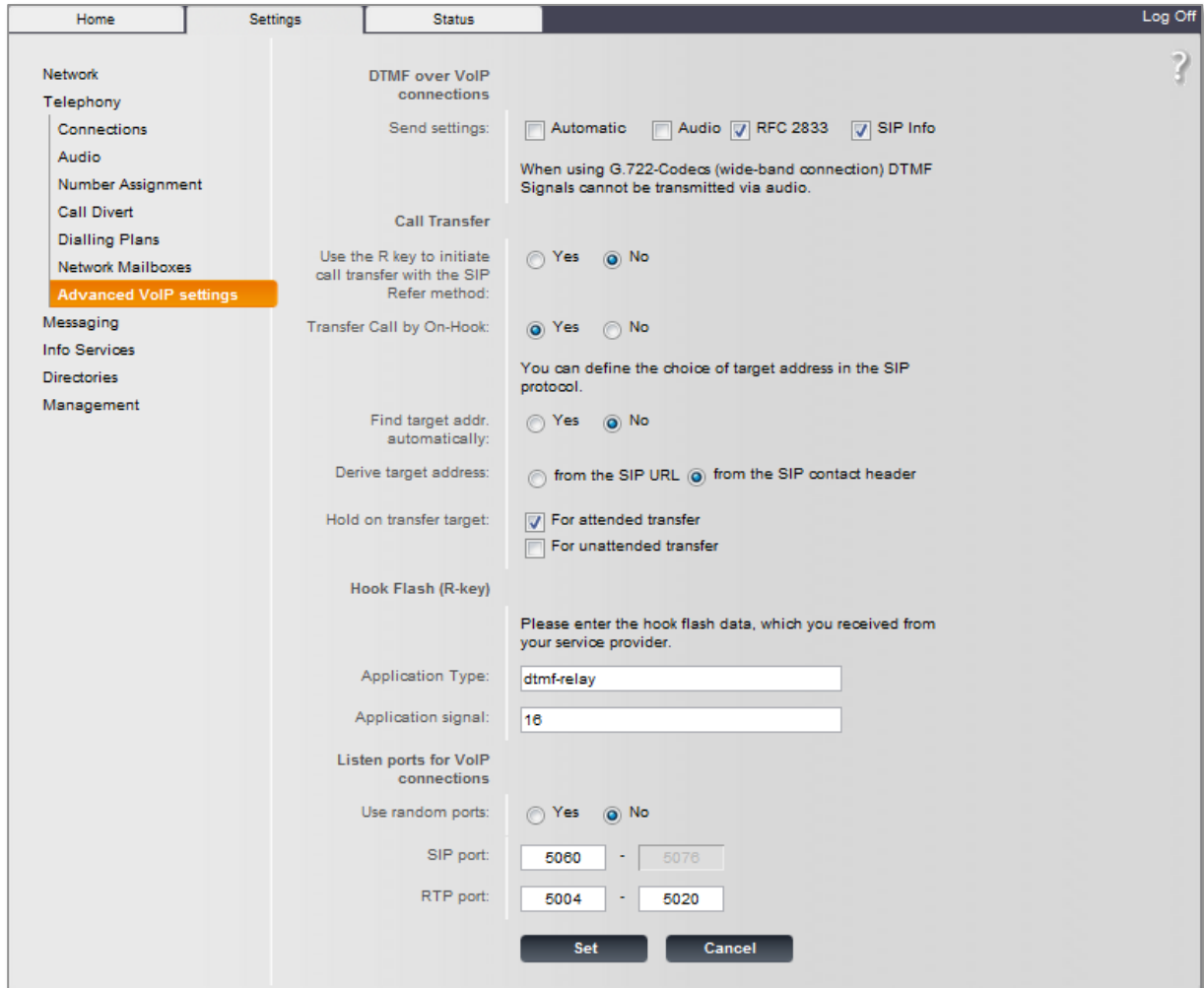
6. Go to "Dialling Plans" and change connection for 112 and 110 to IP1

<ul style="list-style-type: none"> Connections Audio Number Assignment Call Divert <li style="background-color: #f4a460;">Dialling Plans Network Mailboxes Advanced VoIP settings Messaging Info Services Directories 	<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">Phone Number</th> <th style="text-align: left;">Area Code</th> <th style="text-align: left;">Connection</th> <th style="text-align: left;">Comment</th> <th style="text-align: left;">Active</th> </tr> </thead> <tbody> <tr> <td>112</td> <td><input type="checkbox"/></td> <td>IP1</td> <td><input type="text"/></td> <td>Delete <input checked="" type="checkbox"/></td> </tr> <tr> <td>110</td> <td><input type="checkbox"/></td> <td>IP1</td> <td><input type="text"/></td> <td>Delete <input checked="" type="checkbox"/></td> </tr> <tr> <td>113</td> <td><input type="checkbox"/></td> <td>Block</td> <td><input type="text"/></td> <td>Delete <input checked="" type="checkbox"/></td> </tr> <tr> <td><input type="text"/></td> <td><input checked="" type="checkbox"/></td> <td>IP1</td> <td><input type="text"/></td> <td>Add</td> </tr> </tbody> </table>	Phone Number	Area Code	Connection	Comment	Active	112	<input type="checkbox"/>	IP1	<input type="text"/>	Delete <input checked="" type="checkbox"/>	110	<input type="checkbox"/>	IP1	<input type="text"/>	Delete <input checked="" type="checkbox"/>	113	<input type="checkbox"/>	Block	<input type="text"/>	Delete <input checked="" type="checkbox"/>	<input type="text"/>	<input checked="" type="checkbox"/>	IP1	<input type="text"/>	Add
Phone Number	Area Code	Connection	Comment	Active																						
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<input type="text"/>	<input checked="" type="checkbox"/>	IP1	<input type="text"/>	Add																						

7. Go to "Network Mailboxes" and write (short) number to voicemail (eg 555)

<ul style="list-style-type: none"> Telephony Connections Audio Number Assignment Call Divert Dialling Plans <li style="background-color: #f4a460;">Network Mailboxes 	<table border="1" style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">Connection</th> <th style="text-align: left;">Call number</th> <th style="text-align: left;">Active</th> </tr> </thead> <tbody> <tr> <td>Fixed Line</td> <td><input type="text"/></td> <td></td> </tr> <tr> <td>IP1</td> <td>555</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>IP2</td> <td>555</td> <td><input checked="" type="checkbox"/></td> </tr> </tbody> </table>	Connection	Call number	Active	Fixed Line	<input type="text"/>		IP1	555	<input checked="" type="checkbox"/>	IP2	555	<input checked="" type="checkbox"/>
Connection	Call number	Active											
Fixed Line	<input type="text"/>												
IP1	555	<input checked="" type="checkbox"/>											
IP2	555	<input checked="" type="checkbox"/>											

8. Go to "Advanced VoIP settings" and change the following:



Home Settings Status Log Off

Network
Telephony
Connections
Audio
Number Assignment
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Advanced VoIP settings
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DTMF over VoIP connections

Send settings: Automatic Audio RFC 2833 SIP Info

When using G.722-Codecs (wide-band connection) DTMF Signals cannot be transmitted via audio.

Call Transfer

Use the R key to initiate call transfer with the SIP Refer method:

Transfer Call by On-Hook: Yes No

You can define the choice of target address in the SIP protocol.

Find target addr. automatically: Yes No

Derive target address: from the SIP URL from the SIP contact header

Hold on transfer target: For attended transfer For unattended transfer

Hook Flash (R-key)

Please enter the hook flash data, which you received from your service provider.

Application Type:

Application signal:

Listen ports for VoIP connections

Use random ports: Yes No

SIP port: -

RTP port: -

Set Cancel