

Gigaset pro

Third Party Interoperability Testing



Desktop Phones
DE310 DE410 DE700 DE900



N510 pro
Business class DECT system



N720 pro
MultiCell DECT System



InterOperation & Configuration Notes For Gigaset pro IP Desktop Phones & DECT Systems Interworking With The Hello Telecom Hosted PBX Service

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Change History

Document revision	Date	Authored by	Sections affected	Reason for change
Rev 001	7 August 2013	JL	All	Initial release

1. Overview

1.1. Introduction

This document provides a summary of how the Hello Telecom Hosted PBX Service can interoperate with Gigaset pro IP DECT Cordless systems and phones. This is a Gigaset pro "self-certification" document based on own testing with Hello Telecom.

1.2. Session Initiation Protocol

Session Initiation Protocol (SIP) is a simple protocol that facilitates peer-to-peer communication sessions. Users (or, in general, any addressable entities) in a SIP framework are identified by Universal Resource Identifiers (URI). Each such Internet-style address (for example, sip: johndoe@proximitycomms.com) maps into one or more Contacts, each of which typically represents a device or service at which the corresponding user may be reached. The SIP framework is responsible for routing a request for a peer-to-peer session addressed to a given URL to one or more appropriate contacts for that URL. The framework may utilise information about the preferences, presence and location of the user identified by the URL, to determine the most appropriate contacts. The protocol also provides mechanisms to specify the type of session that is requested as well as means to change session parameters.

It is important to understand that SIP is not a standardised protocol but in fact is an IETF RFC (**R**equ**e**st **F**or **C**omment). An RFC is a document that describes the specifications for a recommended technology. If the specification is ratified it becomes a standards document. At the time of producing this document SIP still remains a RFC. Not all RFCs become standards; some are designated indefinitely with Informational or Experimental status. Therefore interoperability of two SIP devices is not guaranteed; this is why Gigaset pro has produced this document to explain the configuration and features available when using its products with third-party providers' services.

Full details of the SIP IETF RFC can be found here: <http://www.ietf.org/rfc/rfc3261.txt>

2. Testing Configuration

2.1. Software versions

The following software versions were used during the testing by Gigaset pro

Device	Software version
Hello Telecom	
Gigaset N300IP & N510 pro	42.075
Gigaset N720DM pro	70.068

3. Configuration

3.1. Gigaset

The screenshots are those of an N510pro however similar configuration parameters are shared across the Gigaset IP product portfolio.

Under the menu heading **Connections** edit the first VoIP account IP1 [note: up to six VoIP accounts/DECT Users can be configured on the N300IP and N510pro, whilst up to 100 Users on the N720 pro system]. Enter the VoIP account User credentials and global PBX settings:

The screenshot displays the 'Settings' page for '1. IP Connection'. The left sidebar contains a navigation menu with categories: Network, Telephony, Connections (highlighted), Audio, Number Assignment, Call Divert, Dialling Plans, Network Mailboxes, Advanced VoIP settings, Messaging, Info Services, Directories, and Management. The main content area is titled '1. IP Connection' and includes a help icon (?). It contains several sections of configuration fields:

- 1. IP Connection**: Assign a connection name or actual phone number for identification. Field: Hello 201.
- VoIP Configuration / Profile Download**: Start Configuration Assistant button.
- Provider**: Other Provider.
- Profile Version**: jl_chester 1371652560.
- Personal Provider Data**:
 - Authentication name: 201.test.net
 - Authentication password: [masked]
 - Username: 201.test.net
 - Display name: 201
- General data for your service provider**:
 - Domain: cbx08.htel.co.uk
 - Proxy server address: cbx08.htel.co.uk
 - Proxy server port: 5060
 - Registration server: cbx08.htel.co.uk
 - Registration server port: 5060
 - Registration refresh time: 180 sec
- Network data for your service provider**:
 - STUN enabled: Yes No
 - STUN server address: [empty field]
 - STUN server port: 3478
 - STUN refresh time: 240 sec
 - NAT refresh time: 20 sec
 - Outbound proxy mode: Always Automatic Never
 - Outbound server address: [empty field]
 - Outbound proxy port: 5060
 - Select Network Protocol: Automatic

At the bottom, there are three buttons: Set, Cancel, and Delete Connection.

Click **Set** and note the Status changes to **Registered**:

Name	Provider	Status	Active	
1. Hello 201	Other Provider	Registered	<input checked="" type="checkbox"/>	Edit
2. Hello 202	Other Provider	Registered	<input checked="" type="checkbox"/>	Edit
3. IP3	Other Provider	Not configured	<input type="checkbox"/>	Edit
4. IP4	Other Provider	Not configured	<input type="checkbox"/>	Edit
5. IP5	Other Provider	Not configured	<input type="checkbox"/>	Edit
6. IP6	Other Provider	Not configured	<input type="checkbox"/>	Edit

Provider or PBX profile

A profile contains all relevant settings for your provider or phone system (PBX).

Automatic check for profile updates: Yes No

Update Profile

Set Cancel

Select the **Number Assignment** menu option:

Ensure that the correct connection is used for both outgoing and incoming calls.

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

Handsets

INT 1 Name: 201

Connection	for outgoing calls	for incoming calls
Hello 201	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>
Hello 202	<input type="radio"/>	<input type="checkbox"/>
Select line for each outgoing call	<input type="radio"/>	

INT 2 Name: 202

Connection	for outgoing calls	for incoming calls
Hello 201	<input type="radio"/>	<input type="checkbox"/>
Hello 202	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>
Select line for each outgoing call	<input type="radio"/>	

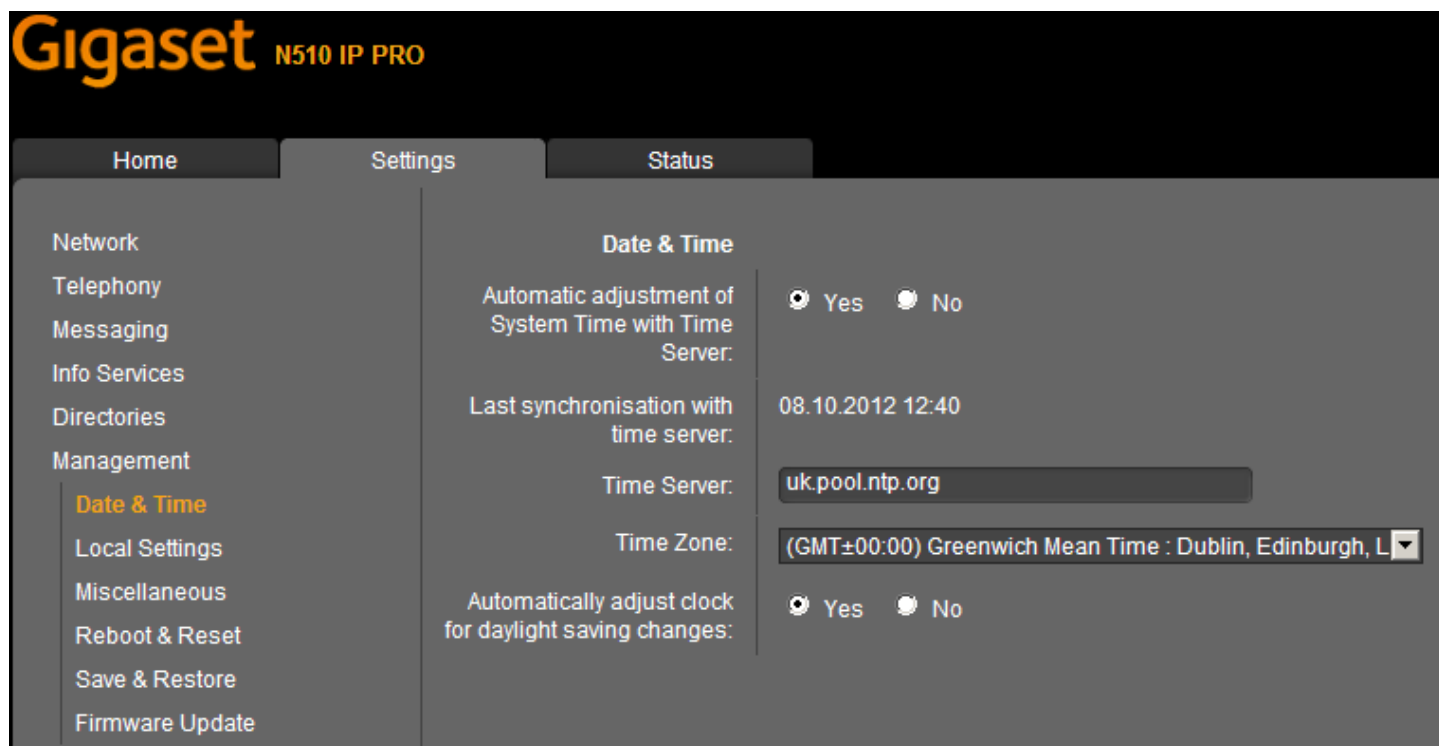
Select the **Network Mailboxes** menu option:
Enter the Hello Telecom network voicemail access number.

Connection	Call number	Active
Hello 201	1571	<input checked="" type="checkbox"/>
Hello 202	1571	<input checked="" type="checkbox"/>

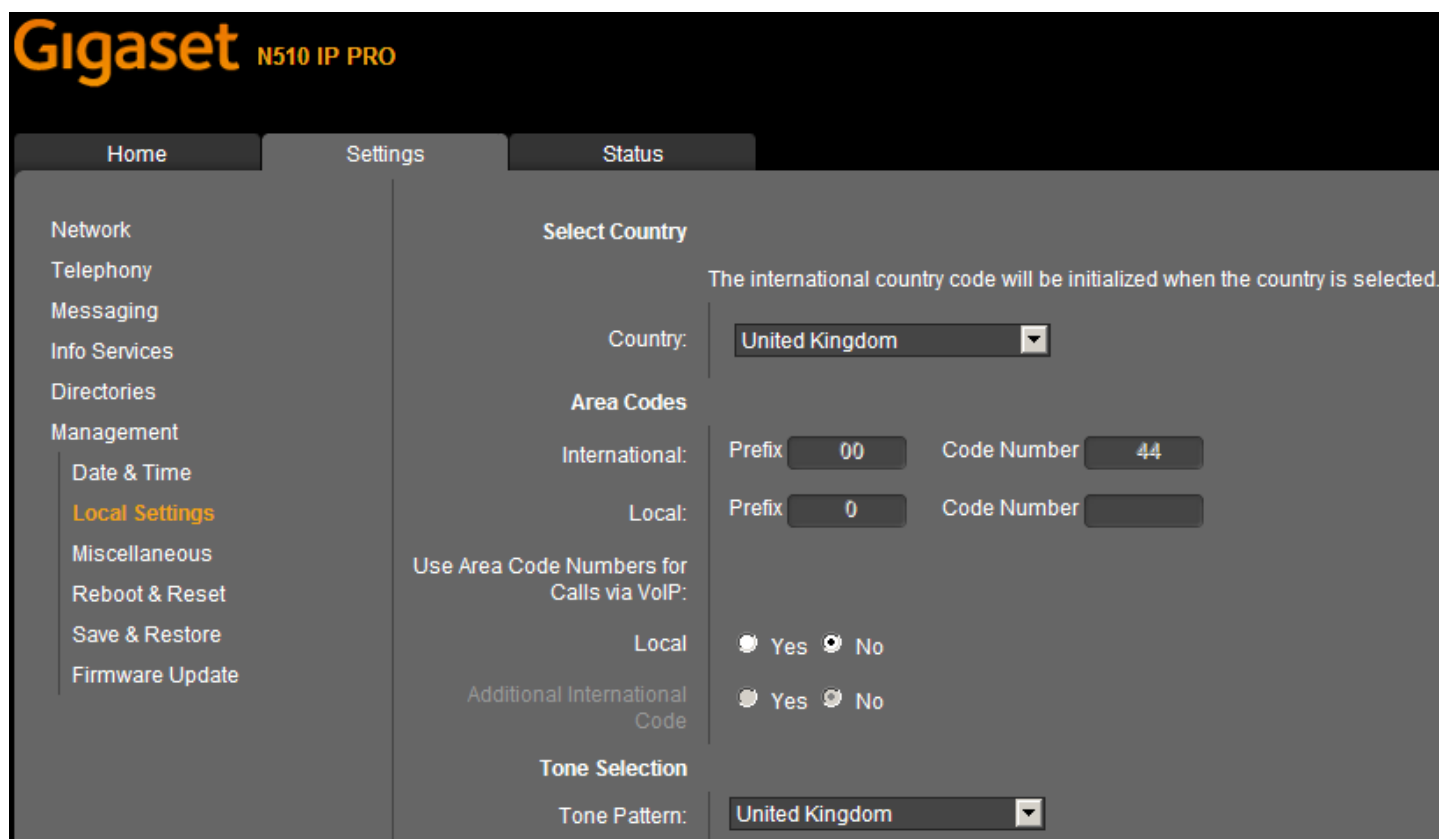
Select the **Messaging > MWI Light** menu option:
Ensure the Network Mailboxes is checked. Missed call notification is optional.

Connection	Missed calls	Missed alarms	eMail	Network Mailboxes
201	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>
202	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>

Select the **Date & Time** menu option:
Enter your preferred NTP server.



Select the **Local Settings** menu option:
Ensure that the UK Tone scheme is selected.



Select the **Advanced VoIP Settings** menu option:
Ensure that **Transfer Call By On-Hook** is selected

The screenshot shows the 'Settings' menu of a Gigaset device. The 'Advanced VoIP settings' option is highlighted in the left-hand navigation pane. The main content area displays various VoIP configuration options:

- DTMF over VoIP connections**
 - Automatic negotiation of DTMF transmission: Yes No
 - 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: Yes No
 - Transfer Call by On-Hook:** Yes No (This option is highlighted with a red box in the image)
 - 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)**
 - R-key settings are disabled because the R key is used for call transfer.
- Listen ports for VoIP connections**
 - Use random ports: Yes No
 - SIP port: 5060 - 5076
 - RTP port: 5004 - 5020
- Music on hold**
 - Yes No

Buttons for 'Set' and 'Cancel' are located at the bottom of the settings page.

INFO NOTE: All of the above settings can be Auto Provisioned into the Gigaset Device using plain XML via appropriate Redirection methods, thereby achieving a Zero-Touch experience with a new device for the End User.

3.2. Correct procedure for initiating Call Transfers from a Gigaset DECT handset:

During an established call, proceed as follows:

1. Press either the **R** key (Recall/Hookflash-telecoms terminology!) or the soft key **Ext.Call** (as indicated in the display during the call) to place the call on hold. Either will have the effect of signalling to the PBX to place the call on hold.
2. Enter the telephone number of the User you wish to call and wait for ringing.
3. At this stage you can either:
 - Blind Transfer - hang up to transfer the call unannounced
 - Consultative Transfer - wait for the other party to answer, then consult/announce the call and hang up. Or it could be that the other party doesn't wish to speak with the Caller in which case select the displayed option to **END ACTIVE CALL** and you will be connected to the Caller once again.



4. Test Results

See published results [here](#)

[Highlights only – full test plan results available upon request]

Further configuration details can be found in the product specific Admin Guides which are available for download in the Support area of the Gigaset pro website.

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