

# Gigaset pro

## Third Party Interoperability Testing



### InterOperation & Configuration Notes For Gigaset pro IP DECT Systems Interworking With The Samsung OfficeServ 7200 PBX

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## Change History

Document revision	Date	Authored by	Sections affected	Reason for change
Rev 001	19 April 2013	JL	All	Initial release

# 1. Overview

## 1.1. Introduction

This document provides a summary of how the Samsung OfficeServ 7200 PBX can interoperate with Gigaset pro IP DECT Cordless systems and phones. This is a Gigaset pro "self-certification" document based on own testing with the Samsung PBX.

## 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.

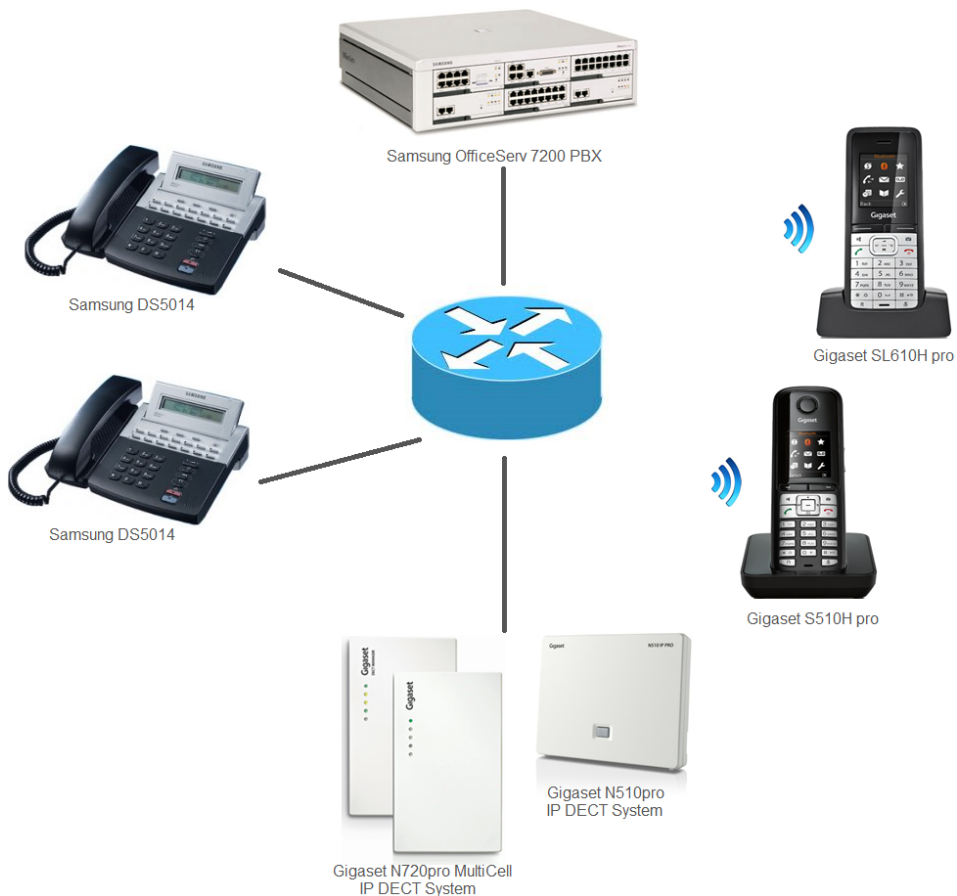
It is important to understand that SIP is not a standardised protocol but in fact is an IETF RFC (**R**equ**e**st **F**or **C**omment). An RFC is a document that describes the specifications for a recommended technology. If the specification is ratified it becomes a standards document. At the time of 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 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. Architecture Overview

The following is a diagram of the solution architecture showing the components used during the test.



### 2.2. Software versions

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

Device	Software version
Samsung OfficeServ 7200 Version	4.65
Gigaset N300IP & N510 pro	42.075
Gigaset N720DM pro	70.068

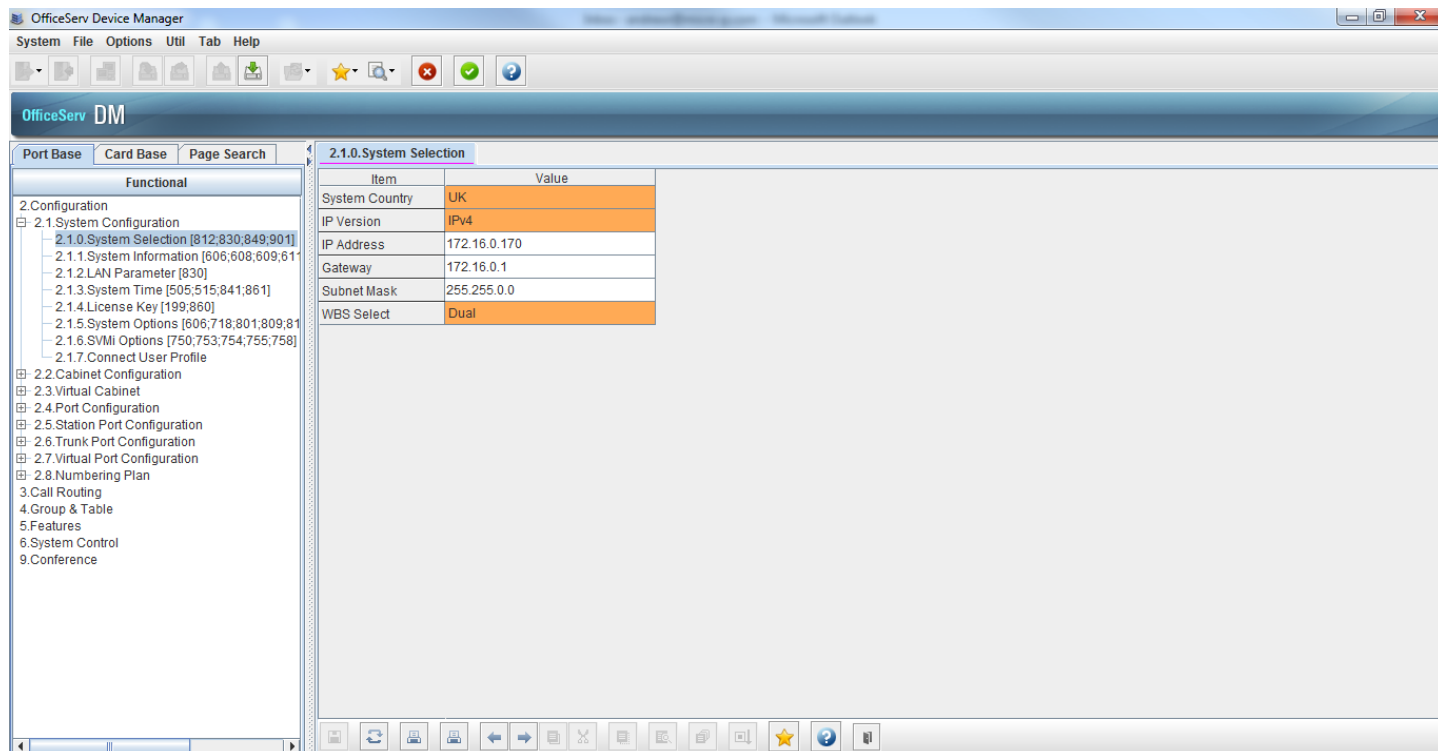
# 3. Configuration

## 3.1. Samsung

### Configure Global SIP Settings

The Samsung PBX used for the testing was an OfficeServ 7200 with MP20 processor along with an OAS card for the RTP gateway.

Ensure that the system is connected to the same network as the phones and contains a static IP address. The IP address can be sent in menu 2.1.0 or MMC830. This IP address is used for the signalling.



Assign an IP address to the OAS card (OfficeServ Applications Server). This is the card where the RTP traffic is dealt. Menu 2.2.2 or MMC831.

**2.2.2.MGI Card**

Item	Value
Card Type	MGI BASE
IP Address	172.16.0.171
Gateway	172.16.0.1
Subnet Mask	255.255.0.0
IP Type	Private with Public
MAC Address	00:21:4C:97:1F:60
Local RTP Port (start)	30000
Public IP Address 1	135.196.94.225
Public RTP Port 1	30000
Public IP Address 2	0.0.0.0
Public RTP Port 2	30000
Public IP Address 3	0.0.0.0
Public RTP Port 3	30000
QoS Monitor	Disable
Telnet ID	mgj
Telnet Password	mgj12345

Port No	Tenant No	Tel Number	Fixed User	Made Busy
1	1	3801		Idle
2	1	3802		Idle
3	1	3803		Idle
4	1	3804		Idle
5	1	3805		Idle

A 3<sup>rd</sup> Party CTI license will need to be installed in Menu 2.1.4 or MMC860:

**2.1.4.License Key**

Item	Value
Temporary License Type	Disable
Remaining License Time	0
License Key	GMWQJ0Z4-Y87F760E-ZGYBVUQG-ABL00E82-AYXOJ4DK-NLICHYEQ
License Status	OK
VMS	Allowed
Fax	Allowed
SIP Trunk	Max Count 1
SIP Phone	Max Count 2
3rd SIP Phone	Max Count 2
SIP Application	Max Count 2
WE VoIP	Max Count 0
Remote Dial	Max Count 0
Delphicom	Max Count 0
License Key	NONKG80H-16TNZHNB-1HUXVW2W-GSBG1A4Q-G90E7S3Q-U97DALMQ
License Status	OK
H.323	Allowed

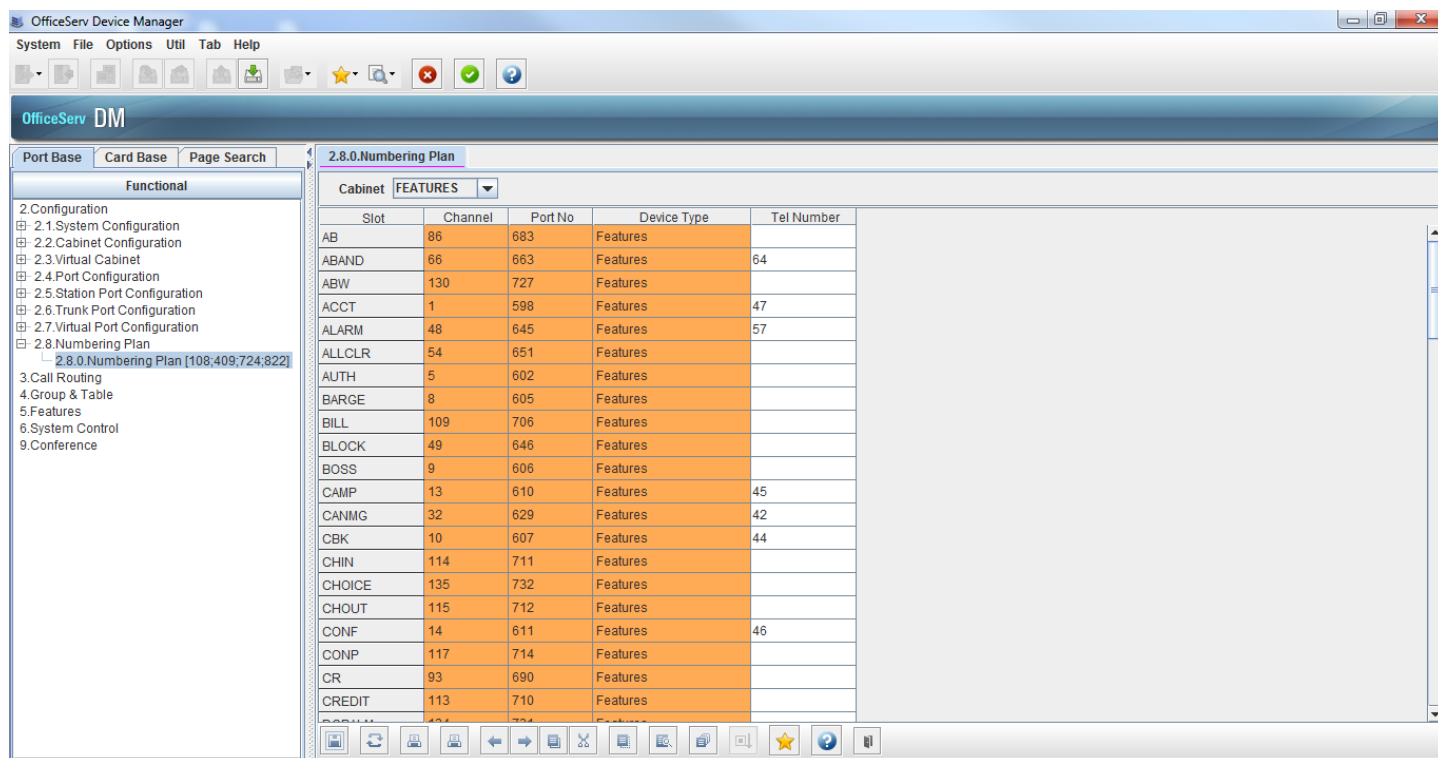
SIP username and passwords will be set as well as enabling call waiting. This is in menu 2.7.2 or MMC842:

Tel Number	User ID	Password	Tone Source	Call Waiting	Call Forward Unreachable	DTMF Type	Insert Trunk Port	Inse
3326	3326	*****	Use System Tone	Enable		RFC2833		Disabl
3327	3327	*****	Use System Tone	Enable		RFC2833		Disabl
3328			Use System Tone	Disable		RFC2833		Disabl
3329			Use System Tone	Disable		RFC2833		Disabl
3330			Use System Tone	Disable		RFC2833		Disabl
3331			Use System Tone	Disable		RFC2833		Disabl
3332			Use System Tone	Disable		RFC2833		Disabl

Enable the support for Re-Invite. 5.2.12 or MMC837. It's the second option from the bottom, the one above is the timer for when for re-invite is sent:

Item	Value
Register MSG Block Timer (Sec)	60
Register Retry Limit	2
SIP Trunk Configuration	
SIP Peering Codec PR1	G.729
SIP Peering Codec PR2	G.711a
SIP Peering Codec PR3	G.711u
SIP Peering Codec PR4	Disable
SIP Peering Use Alias	Disable
SIP Peering Max Channel	64
Outgoing Originator Codec Use	Disable
Incoming Call Fixed Codec	Disable
TLS Certificate Format	PEM
TLS Encrypt Private Key Use	Disable
TLS Encrypt Private Key Password	
SIP Diversion Header Accumulation	Enable
SIP Extension Option	
Response to Tag	Keep
SIP Connection Reuse	Disable
SIP Mutual TLS Enable	Disable
SIP Validate Any TLS Certificate	Disable
TCP Port	5060
TLS Port	5061
Session Expire Time (sec)	10
Session Timer	Re-Invite
UNREG. Guard Time	Disable

Feature codes can be set in menu 2.8.0 or MMC724:



The screenshot shows the OfficeServ Device Manager interface. The main window is titled '2.8.0.Numbering Plan'. On the left, there is a navigation tree with '2.8.0.Numbering Plan [108:409;724;822]' selected. The main area displays a table with the following columns: Slot, Channel, Port No, Device Type, and Tel Number. The table contains 20 rows of feature codes.

Slot	Channel	Port No	Device Type	Tel Number
AB	86	683	Features	
ABAND	66	663	Features	64
ABW	130	727	Features	
ACCT	1	598	Features	47
ALARM	48	645	Features	57
ALLCLR	54	651	Features	
AUTH	5	602	Features	
BARGE	8	605	Features	
BILL	109	706	Features	
BLOCK	49	646	Features	
BOSS	9	606	Features	
CAMP	13	610	Features	45
CANMG	32	629	Features	42
CBK	10	607	Features	44
CHIN	114	711	Features	
CHOICE	135	732	Features	
CHOUT	115	712	Features	
CONF	14	611	Features	46
CONP	117	714	Features	
CR	93	690	Features	
CREDIT	113	710	Features	

Default feature codes.

- Call Forward No Answer - 603
- Call Forward Busy - 602
- Call Forward Busy no Answer - 604
- Call Forward Immediate – 601
- Call Pickup - 66 followed by the pickup group. (Everyone is in pickup group 1 by default).
- Call Pickup direct extension – 65 followed by the extension number.
- DND - 605



## 3.2. 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/Samsung extensions 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 configuration page for a VoIP connection. The interface is divided into a left sidebar with navigation options and a main content area. The main content area is titled '1. IP Connection' and contains several sections of settings:

- 1. IP Connection:** Assign a connection name or actual phone number for identification. The 'Connection Name or Number' field is set to '3326'.
- VoIP Configuration / Profile Download:** Includes a 'Start Configuration Assistant' button.
- Provider:** Set to 'Other Provider'.
- Profile Version:** (Empty field)
- Personal Provider Data:**
  - Authentication name: 3326
  - Authentication password: (masked with dots)
  - Username: 3326
  - Display name: 3326
- General data for your service provider:**
  - Domain: 172.16.0.170
  - Proxy server address: 172.16.0.170
  - Proxy server port: 5060
  - Registration server: 172.16.0.170
  - Registration server port: 5060
  - Registration refresh time: 600 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 (dropdown menu)

At the bottom of the configuration area, there are three buttons: 'Set', 'Cancel', and 'Delete Connection'.

- Connection Name** = Name to be associated with this connection – default is IP1
- Authentication Name** = Extension number defined on Samsung under individual users
- Authentication Password** = Password as defined under 'SIP Password' under individual users
- User Name** = Extension number defined on Samsung under individual users
- Display Name** = Name that will be displayed as the CPND when calls are made from the DECT handset
- Domain** = SIP Domain / Realm that is configured on the Samsung Environment
- Proxy Server Address** = IP of Samsung switch providing SIP Proxy
- Proxy Server Port** = Default 5060
- Registration Server Address** = IP of Samsung switch providing SIP Proxy
- Registration Server Port** = Default 5060
- Registration Refresh Timer** = Must match the timer that has been configured in Samsung eg 600
- STUN** = Disabled
- Outbound Proxy** = Not required

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

The screenshot shows the 'Status' tab of the Samsung SIP settings. The 'Overview of connections' table lists six connections. Connections 1 and 2 are 'Registered', while connections 3 through 6 are 'Not configured'. Below the table, there is a 'Provider or PBX profile' section with an 'Automatic check for profile updates' option set to 'Yes' and an 'Update Profile' button. At the bottom, there are 'Set' and 'Cancel' buttons.

Name	Provider	Status	Active
1. 3326	Other Provider	Registered	<input checked="" type="checkbox"/>
2. 3327	Other Provider	Registered	<input checked="" type="checkbox"/>
3. IP3	Other Provider	Not configured	<input type="checkbox"/>
4. IP4	Other Provider	Not configured	<input type="checkbox"/>
5. IP5	Other Provider	Not configured	<input type="checkbox"/>
6. IP6	Other Provider	Not configured	<input type="checkbox"/>

Select the **Number Assignment** menu option:

Ensure that the "Samsung" connection is used for both outgoing and 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 3326

Connection	for outgoing calls	for incoming calls
3326	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>
3327	<input type="radio"/>	<input type="checkbox"/>

Select line for each outgoing call

INT 2 Name 3327

Connection	for outgoing calls	for incoming calls
3326	<input type="radio"/>	<input type="checkbox"/>
3327	<input checked="" type="radio"/>	<input checked="" type="checkbox"/>

Select line for each outgoing call

**Call Manager**

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

Connection	Enable Call Manager	Handset
3326	No	3326
3327	No	3327

Set Cancel

Select the **Network Mailboxes** menu option:

Enter the Samsung network voicemail access number.

Home Settings Status Log Off

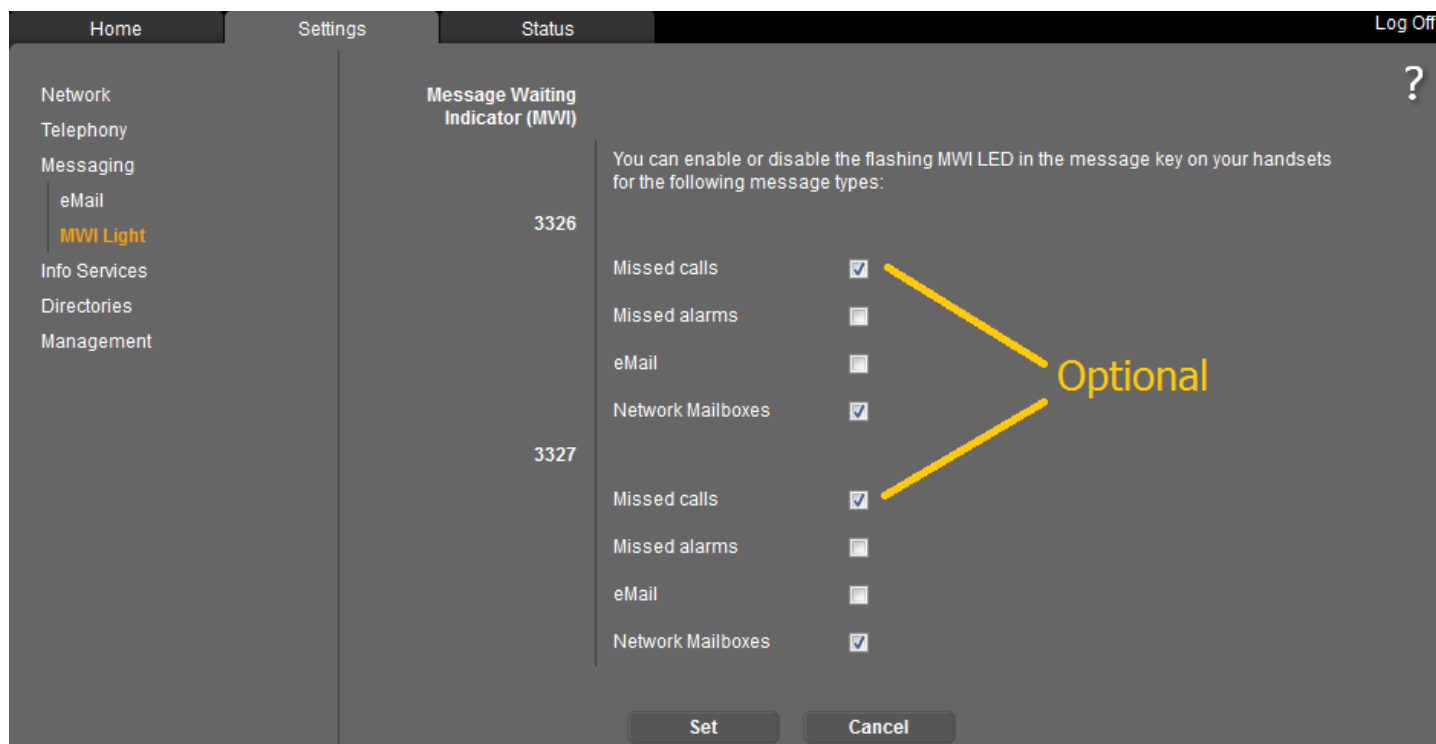
Network  
Telephony  
Connections  
Audio  
Number Assignment  
Call Divert  
Dialling Plans  
**Network Mailboxes**  
Advanced VoIP settings  
Messaging  
Info Services  
Directories  
Management

**Network Mailboxes**

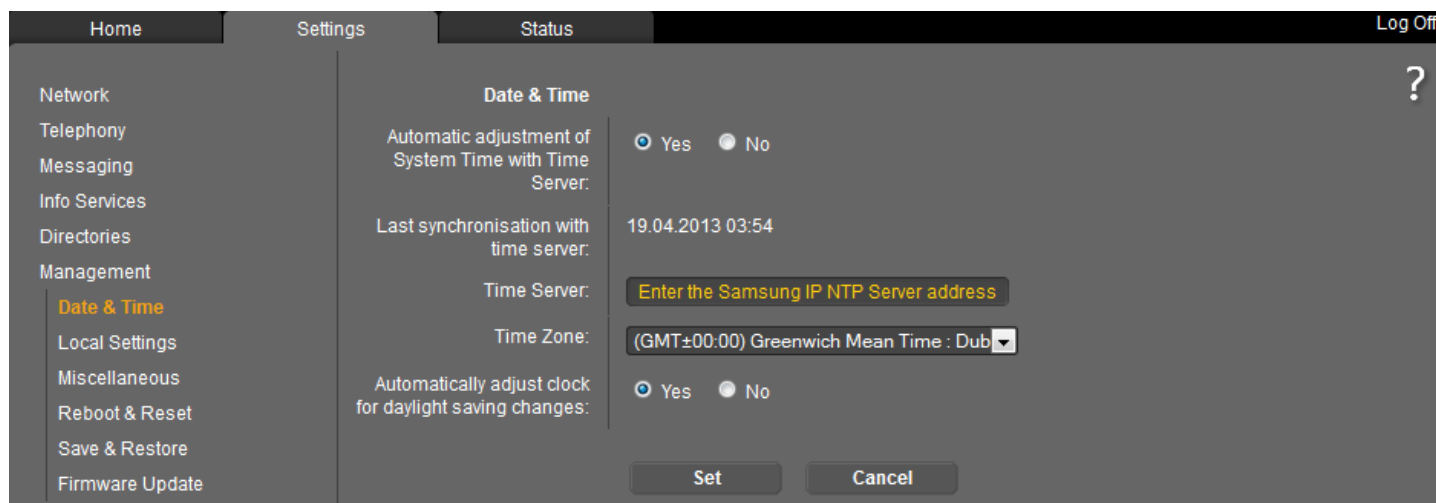
Connection	Call number	Active
3326	5039	<input checked="" type="checkbox"/>
3327	5039	<input checked="" type="checkbox"/>

Set Cancel

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



Select the **Date & Time** menu option:  
Ensure that the Samsung IP address is used for as the NTP server.



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

Home Settings Status Log Off

Network  
Telephony  
Messaging  
Info Services  
Directories  
Management  
Date & Time  
**Local Settings**  
Miscellaneous  
Reboot & Reset  
Save & Restore  
Firmware Update

Select Country ?

The international country code will be initialized when the country is selected.

Country: United Kingdom

Area Codes

International: Prefix 00 Code Number 44

Local: Prefix 0 Code Number

Use Area Code Numbers for Calls via VoIP:

Local  Yes  No

Additional International Code  Yes  No

Tone Selection

Tone Pattern: United Kingdom

Set Cancel

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

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

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

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

Set Cancel

**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.



### 3.3. 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

Issues deemed to require a resolution are displayed in BLUE

Feature	Success	Comments
Call – initiate	✓	
Call – accept	✓	
On hook dialling & handsfree speaker	✓	
Hold/unhold	✓	
Park/unpark	-	Untested
Transfer – consultative	✓	
Transfer – unattended	✓	
Call Forward	✓	
Do Not Disturb	✓	
Redial	✓	
Caller ID – outgoing	✓	
Caller ID – incoming	✓	
Redirect to VoiceMail	✓	
Voice mailbox access	✓	
VoiceMail new message indicator & counter	partial	N720 OK. N510 bug with Gigaset R&D
Call Waiting	✓	
3-Party Conference (as attendee)	✓	
3-Party Conference (initiate)	✘	Further testing required
Call Pick up (group)	✓	
Call Pick up (directed)	✓	
Broadcast – initiate	-	Untested
Broadcast – audible	-	Untested
DTMF signalling	✓	

[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.

Comments or questions in relation to this document should be addressed to the originator:

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