

# PHONE SETTINGS DATATAL GATEWAY

M = mandatory

O = Optional

D = Dependanc

## SNOM 870 (snom870-SIP 8.7.5.35)

### Identity setting

	SETTING	VALUE	DESCRIPTION
<b>M</b>	Account:	<extenstion>	Users extension
<b>M</b>	Password:	<password>	User password, define under login in Admital Web
<b>M</b>	Registrar:	Datatal GW FQDN/IP	
<b>M</b>	Authentication Username:	<username>	User username, usually e-mail. Define under login in Admital Web
<b>O</b>	Mailbox:	<externsion>@< Datatal GW FQDN/IP>	This is to recive Message Indiction Lamp
<b>O</b>	Display text for idle screen:	<text if user is idle>	
<b>D</b>	STUN server (IP-addr:port):	<STUN server IP and port>	Phone should signal correct external IP in registartion and call setup. Datatal GW can be configure as STUN Server
<b>D</b>	STUN interval (seconds):	Refresh timer for stun request	

# Configuration Identity 1

SNOM

## Operation

- Home
- Directory

## Setup

- Preferences
- Speed Dial
- Function Keys
- Identity 1**

- Identity 2
- Identity 3
- Identity 4
- Identity 5
- Identity 6
- Identity 7
- Identity 8
- Identity 9
- Identity 10
- Identity 11
- Identity 12
- Action URL Settings
- Advanced
- Certificates
- Software Update

## Status

- System Information
- Log
- SIP Trace
- DNS Cache
- Subscriptions
- PCAP Trace
- Memory
- Settings

## Manual

[Login](#) [Features](#) [SIP](#) [NAT](#) [RTP](#)

**Login Information:**

Identity active:  on  off ?

Displayname:  ?

Account:  ?

Password:  ?

Registrar:  ?

Outbound Proxy:  ?

Failover Identity:  ?

Authentication Username:  ?

Mailbox:  ?

Ringtone:  ?

Custom Melody URL:  ?

Display text for idle screen:  ?

XML Idle Screen URL:  ?

Ring After Delay (sec):  ?

Record Missed Calls:  on  off ?

Record Dialed Calls:  on  off ?

Record Received Calls:  on  off ?

Identity is hidden:  on  off ?

[Apply](#) [Re-Register](#) [Play Ringer](#)

[Remove Identity](#) [Remove All Identities](#)

# Configuration Identity 1

SNOM

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- Identity 6

[Login](#) [Features](#) [SIP](#) [NAT](#) [RTP](#)

**NAT Identity Settings:**

Offer ICE:  on  off ?

STUN server (IP-addr:port):  ?

STUN interval (seconds):  ?

Keepalive interval (seconds):  ?

[Apply](#)

## Advanced

	SETTING	VALUE	DESCRIPTION
<input type="radio"/>	Update Policy:	Update automatically	If update should be automatically
<input type="radio"/>	Setting URL:	URL to Datatal provsioning service	https://<Datatal GW FQDN/IP>/admin/p_prov/fetch.php?mac={mac}
<input type="radio"/>	Settings refresh timer:	Refresh timer	

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### Manual

[Network](#) [Behavior](#) [Audio](#) [SIP/RTP](#) [QoS/Security](#) **Update**

**Update:**

Update Policy:  ?

Setting URL:  ?

Settings refresh timer:  ?

PnP Config:  on  off ?

By clicking on the **Load** button below the phone will **RESET** its settings, load the new settings from the specified file and reboot. **So all current settings will be lost!**

Upload Setting File manually:  Ingen fil har valts

Load TR-069 Parameter Map Manually:  Ingen fil har valts

Load Dialplan XML Manually:  Ingen fil har valts

## Function Keys

SETTING	VALUE	DESCRIPTION
<input type="radio"/> Context	Line that user is registered	
<input type="radio"/> Type	Extension	We are interested in extensions
<input type="radio"/> Number	<extension>	The extension that we want to supervise, a colleague or a hunt group
<input type="radio"/> Short Text	<name>	Name of group or colleague
<input type="radio"/> MWI Button	<number to voicemail>	Number to voicemail in Flexi (internal logon extension)

## Function Keys

snom

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### Manual

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Some settings are not yet stored permanently. [Save](#) [View Changes](#) [?](#)

### Key Settings:

On this page you can specify the settings for programmable keys on your snom phone. Use **Context** to specify the identity context for that key e.g. this identity will be used to subscribe for a particular extension. **Type** will select the actual functionality of a particular key. In the last argument field **Number**, the actual telephone number, sip url, dtmf sequence, action url or key type can be stored. Please refer to your phone manual for more details.

Context	Type	Number	Short Text	
214@192.168.50.80	Extension	sip:101@192.168.50.80;user	Sales	P1
214@192.168.50.80	Extension	sip:102@192.168.50.80;user	Support	P2
Active	None			P3
Active	None			P4
Active	Line			P5
Active	Line			P6
Active	Line			P7
Active	Line			P8
Active	Line			P9
Active	Line			P10
Active	Line			P11
Active	Line			P12
Active	Line			P13
Active	Line			P14
Active	Line			P15

None

Accepted Calls Missed Calls

Redial

None

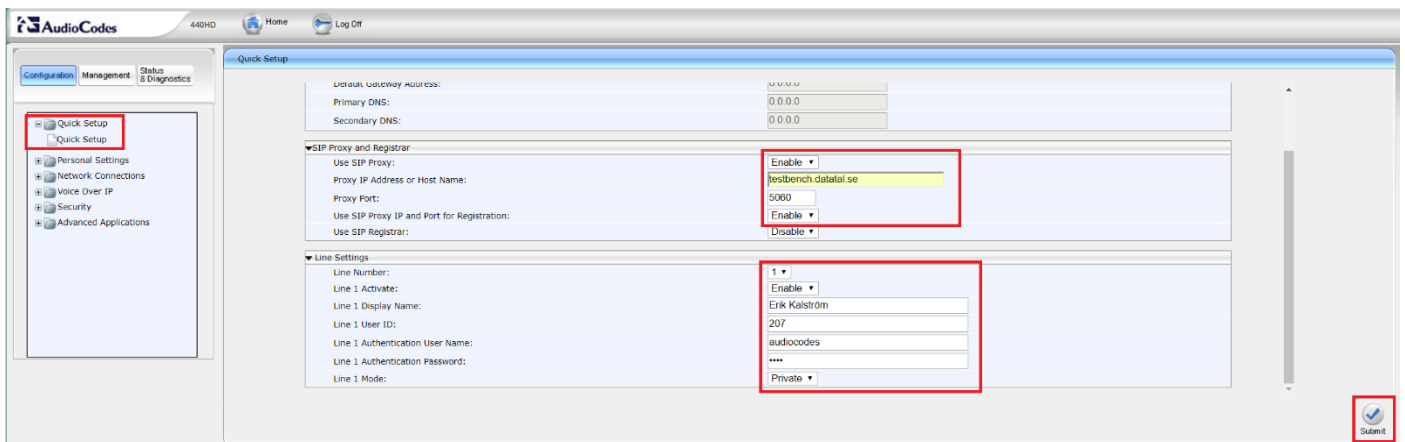
Type	Number	
Speed Dial	1001	Retrieve
Key Event	DND	DND
Key Event	Directory	Directory
Key Event	Menu	Menu
Key Event	Transfer	Transfer
Key Event	Hold	Hold

Apply

# AudioCodes 440HD (2.2.12.126)

## Quick setup

SETTING	VALUE	DESCRIPTION	
M	Use SIP Proxy	Enable	
M	Proxy IP address or host name	<Datatal GW FQDN or IP>	
M	Use SIP proxy IP and Port for Registration	Enable	
M	Line Number	1	
M	Line 1 Active	Enable	
M	Line 1 Display Name	<Users first and surname>	
M	Line 1 User Id	<User extension>	
M	Line 1 Authentication User Name	<Users username>	Can be edit under User->Login in Admital Web
M	Line 1 Authentication Password	<Users password>	
M	Line 1 Mode	Private	



## Services

### Server Type

SETTING	VALUE	DESCRIPTION
<input type="radio"/> Type	Asterisk	This will activate SIP subscribe on Presence events for BLF line

#### ▼Application Server

Type: Asterisk ▼

### MWI settings

SETTING	VALUE	DESCRIPTION
<input type="radio"/> Voice Mail Number	<number to voicemail>	Number to voicemail (internallogonextension)
<input type="radio"/> Activate	Enable	
<input type="radio"/> Subscribe To MWI	Enable	Phone will SIP subscribe on MWI
<input type="radio"/> MWI Server IP Address or Host name	<Datatal GW FQDN or IP>	
<input type="radio"/> MWI Server port	5060	
<input type="radio"/> MWI Subscribe Expiry time	3600	

The screenshot shows the AudioCodes configuration interface. The left sidebar contains a tree view with 'Services' highlighted. The main panel displays the 'Services' configuration page. A red box highlights the 'Message Waiting Indication (MWI)' section, which includes the following settings:

- Voice Mail Number: 1001
- Activate: Enable
- Subscribe To MWI: Enable
- MWI Server IP Address or Host Name: 192.168.50.80
- MWI Server Port: 5060
- MWI Subscribe Expiry Time: 3600 Seconds

Below the MWI section, the 'BLF Support' section is visible with the following settings:

- Activate: Enable
- Call Pick Up: Enable
- Access code: \*\*
- BLF Subscription Period: 3600 Seconds

# Polycom vvx410 (5.7.0.11768)

## Simple Setup

	SETTING	VALUE	DESCRIPTION
M	Address	<Datatal GW FQDN or IP>	
M	Port	5060	
M	Display Name	Enable	
M	Address	<User extension>	
M	Authentication User Id	<Users username>	
M	Authentication Password	<Users password>	
M	Label	<Label on phone for this line>	
M	Line 1 Authentication User Name	<Users username>	Can be edit under User->Login in Admital Web
M	Base Profile	Generic	

**Polycom | VVX 410**

Home **Simple Setup** Preferences Settings Diagnostics Utilities

You are here: Simple Setup

**Simple Setup**

**Language**  
 Phone Language: English (Internal) [v]  
 Web Configuration Utility Language: Add

**Time Synchronization**  
 Alternate SNTP Server: time.windows.com [v]  
 Time Zone: (GMT 0:00) Western Europe Time [v]

**SIP Server**  
 Address: 192.168.50.80  
 Port: 5060

**SIP Outbound Proxy**  
 Address:  
 Port: 0

**SIP Line Identification**  
 Display Name: Johnny Pettersson  
 Address: 205  
 Authentication User ID: Polycom  
 Authentication Password: \*\*\*\*  
 Label: Line 1

**Base Profile**  
 \* Base Profile: Generic [v]

**Note:**  
 \* Fields require a phone reboot/restart.

Cancel Reset to Default View Modifications Save


## Line 1 advance settings

### Diversion state and MWI settings

	SETTING	VALUE	DESCRIPTION
<input type="radio"/>	Enforced by Server	Yes	SIP subscription in diversion state
<input type="radio"/>	Signaling Method	Subscribe As Feature Event	
<input type="radio"/>	Subscription Address	<Datatal GW FQDN or IP>	
<input type="radio"/>	Callback mode	Contact	
<input type="radio"/>	Callback Contact	<number to voicemail>at<Datatal GW FQDN or IP>	

Home Simple Setup Preferences Settings Diagnostics Utilities

You are here: Settings > Lines > Line 1



**Line 1**

**Identification**

Display Name: Johnny Pettersson  
Address: 205  
Label: Line 1  
Type:  Private  Shared  
Third Party Name:   
Number of Line Keys: 1  
Calls Per Line: 24  
Enable SRTP:  Yes  No  
Offer SRTP:  Yes  No  
Require SRTP:  Yes  No  
Server Auto Discovery:  Enable  Disable

**Outbound Proxy**

**Server 1**

**Server 2**

**Call Diversion**

\* Enforced by Server:  Yes  No  
Signaling Method: Subscribe As Feature Event  
Skype for Business Forward: Disable Call Forwarding  
Skype for Business Forward Contact:   
**Message Center**

Subscription Address: 192.168.50.80  
Callback Mode: Contact  
Callback Contact: 1001@testbench.datatal.se

**Ring Type**

**Note:**  
\* Fields require a phone reboot/restart.

Cancel Reset to Default View Modifications Save



# Ascom IPBS2 DECT

Version tested: 10.3.5

Configuration	Info	Admin	NTP	Kerberos	Certificates	License	EULA
<b>General</b>	<b>Version</b> IPBS2[10.3.5], Bootcode[10.3.5], Hardware[IPBS2-A5A/1A]						
LAN	<b>Serial Number</b> [REDACTED]						
IP4	<b>MAC Address (LAN)</b> [REDACTED]						
IP6	<b>DRAM</b> 48 MB						
LDAP	<b>FLASH</b> 16 MB						
DECT	<b>Coder</b> 4 Channels of G.711,G.729,G.723,G.722.2						
VoIP	<b>SNTP Server</b> 216.239.35.8						
Unite	<b>Time</b> 06.03.2020 11:12						
Services	<b>Uptime</b> 0d 0h 23m 31s						

Set NTP server

## IP-DECT Base Station

Configuration	Info	Admin	NTP	Kerberos	Certificates	License	EULA
<b>General</b>	<b>Time Server</b> <input type="text" value="time.google.com"/>						
LAN	<b>Alt. Time Server</b> <input type="text"/>						
IP4	<b>Interval [min]</b> <input type="text" value="60"/>						
IP6	<b>Timezone</b> <input type="text" value="Europe - Central European Time (UTC+1)"/>						
LDAP	<b>String</b> <input type="text" value="CET-1CEST-2,M3.5.0/2,M10.5.0/3"/>						
DECT	<b>Current Server</b> time.google.com -> 216.239.35.8						
VoIP	<b>Last Sync</b> 06.03.2020 10:48						
Unite	<input type="button" value="OK"/> <input type="button" value="Cancel"/>						
Services							
<b>Administration</b>							
Users							

	SETTING	VALUE	DESCRIPTION
O	Coder	G722.2/G711A	G722.2 is set for HD audio
O	Coder	G711A	If only G711
M	Disable ICE	Check	

## IP-DECT Base Station

Configuration
**System**
Suppl. Serv.
Master
Crypto Master
Mobility Master
Radio
Radio config

- General
- LAN
- IP4
- IP6
- LDAP
- DECT**
- VOIP
- Unite
- Services
- Administration
- Users
- Device Overview
- DECT Sync
- Traffic
- Gateway
- Backup
- Update
- Diagnostics
- Reset

System Name:

Password:

Confirm Password:

Subscriptions:

Authentication Code:

Tones:

Default Language:

Frequency:

Enabled Carriers: 9 8 7 6 5 4 3 2 1 0

Local R-Key Handling:

No Transfer on Hangup:

No On-Hold Display:

Display Original Called:

Early Encryption:

RFP Location:

Unite Data Channel:

Disable ICE:

Coder:  Frame (ms)  Exclusive  SC

Secure RTP Key Exchange:

	SETTING	VALUE	DESCRIPTION
M	Protocol	SIP/TCP	Transport UDP and TLS is supported but TCP preferred
M	Proxy	<FQDN-Gateway-server>	Servename or IP to Gateway
O	Max. Internal Number Length	Number length in system	
O	STUN Server	IP and port to Stun server	If DECT base is placed external site and cannot communicate with Gateway on internal address this is required
M	Register With number	Checked	

## IP-DECT Base Station

Configuration
System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Radio
Radio config
PARI
SARI
Air Sync

- General
- LAN
- IP4
- IP6
- LDAP
- DECT
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Mode Active

Multi-Master
 

Master ID

Enable PARI Function

Region Code

IP-PBX
 

Protocol SIP/TCP

Proxy

Alt. Proxy

Alt. Proxy

Alt. Proxy

Domain

Max. Internal Number Length

International CPN Prefix

Registration with system password

Enbloc Dialing

Enable Enbloc Send-Key

Send Inband DTMF

Allow DTMF Through RTP

Short Disconnect Tone

Treat rejected calls as Busy

Configured With Local GK

SIP Interoperability Settings
 

Registration Time-To-Live  [sec]

STUN server

Hold Signalling inactive

Hold Before Transfer

Accept Inbound Calls Not Routed Via Home Proxy

Register With Number

AOR as Line Identity

KPML support

SETTING	VALUE	DESCRIPTION
M	Send Early Progress Response	Check SIP/TSIP

## IP-DECT Base Station

Configuration

**SIP**

General

LAN

IP4

IP6

LDAP

DECT

**VoIP**

Unite

Services

Administration

Users

Device Overview

DECT Sync

Traffic

Gateway

Backup

Update

Diagnostics

Reset

Add Instance ID To The User Registration With The IP-PBX	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
IP-PBX Supports Redirection Of Registration When Registered To Alternative Proxy	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Use Local Contact Port As Source Port For TCP/TLS Connections	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Prefer P-Asserted-Identity As Calling Party Identity	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Use SBC for NAT traversal	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
No Server Certificate Subject Check For TLS Connections	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Accept Hold Signaling Using Remote Media Address 0.0.0.0	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Remove SRTP Lifetime in SDP	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Allow Multiple Codecs in Answer SDP	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Send Early Progress Response	<input checked="" type="checkbox"/> SIP	<input checked="" type="checkbox"/> TSIP	<input type="checkbox"/> SIPS
Ignore Retry-After in Registration Responses	<input type="checkbox"/> SIP	<input type="checkbox"/> TSIP	<input type="checkbox"/> SIPS

Note: All settings require reset

## DECT USER

	SETTING	VALUE	DESCRIPTION
M	Long name	Arbitrary name	
M	Display name	Name displayd for user	
M	Name	Name	
M	Number	Extension	Users extension in Flexi
M	Auth.Name	Username	Username on user in Flexi
M	Password	User password	Password for user, can use Terminal Password instead of users own password
M	IPEI	IPEI for dect handset	
O	Idle Display	Display text when idle	
M	Auth. Code	Pincode	Used when registerd a handset to base

Edit User - Google Chrome

Inte säker | 192.168.50.108/GW-DECT/mod\_cmd\_login.xml?cmd=show&user-guid=186...

User type

User

User Administrator

Long Name: 230

Display Name: Ascom Dect1

Name: 230

Number: 230

Auth. Name: asc (SIP only)

Password: .....

Confirm Password: .....

IPEI / IPDI: [Redacted]

Idle Display: Ascom Dect1

Auth. Code: 0000

Feature Status

Call Waiting On

OK Apply Delete Unsubs. Cancel