

Note: A single Anyphone line has a maximum concurrent call capacity of 4 calls. Hardware capacity should also be taken into consideration. This means that SNOM PA1 can handle up to have 4 concurrent calls.

Configuring the **SNOM PA1** Paging Server

1. Obtain SIP Credentials from Host Pilot
2. Connect and Login to the SNOM PA1 from the web interface
3. Configure provisioning information
4. Create Paging Groups (Optional)

2. Connect and Login to the **SNOM PA1**

The **SNOM PA1** can be connected to a POE switch (Power Over Ethernet) with an Ethernet cable or a power adapter. This ability allows the SNOM to be more versatile in where it can be placed. A power cable for the device can also be purchased from SNOM separately if yours did not ship with one. When you have power to the device the power light will turn red.

To begin setting up your PA1 Paging Server, ensure you have:

- Connected the Paging Server to your network
- Provided power to the Paging Server:
 - The **SNOM PA1** does not ship with a power cable
 - The Paging Server must be connected to a POE Switch or have a POE injector
 - A

Power cord can be purchased from Amazon below

- <http://www.amazon.com/Snom-Power-Supply-Phone-Phihong/dp/B007RTKWYG>

- Provided an IP address to the Paging Server via DHCP

Once your Paging Server has an IP address, you may use a web browser to login to the device.

- NOTE: An easy way to find out the IP address of the Paging Server is to connect a speaker to it in either the Line Out port or into the Red and Black speaker wires.
- Once an audio source is connected you can then press the IP/Reset once and the device will announce the IP Address over the connected speaker

You will then see the main page below:

- There is no password or username needed on initial login.

Operation

Home
Directory

Setup

Preferences
Speed Dial
Function Keys
Identity 1
Identity 2
Identity 3
Identity 4
Action URL Settings
Advanced
Certificates
Software Update

Status

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

Manual



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Security Advice

We strongly recommend that you secure the web interface in order to protect your phone against remote attacks. Therefore the HTTP User and Password as well as the Administrator Password should be changed from the default value.

Security:

Administrator Password: ?

Administrator Password (Confirmation): ?

HTTP Server:

User: ?

Password: ?

Additionally you should protect the web interface with hidden security tags against remote attackers trying to change phone settings with faked HTTP POST requests.

Use hidden tags: on off ?

Press to save the settings as shown above.

Press to ignore the warning until reboot.

Press to ignore the warning forever.

If you would like to setup Administrator login credentials for the device you can do so here.

3. Configure provisioning information

You will now need to enter provisioning information from your Host Pilot for the Paging Server. The information needs to be entered for Identity 1 under Setup from the left hand menu.

SNOM's terminology does differ from ours, please refer to the image and tables below.

Configuration Identity 1

VERSION 8

Operation

Home
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[Login](#) [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: ?

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](#)

Credential Name	SNOM Credential Name
EXT number (current line extension number)	Account
SIP Authorization ID	Authentication Username
Password	Password
SIP Domain	Registrar AND Outbound Proxy

Once you have entered these settings select Apply and then Save. In addition to the SIP credentials provided by Intermedia the following settings must also be applied.

- Setup > Identity 1 > SIP tab > Support broken Registrar > **On** ○ NOTE: This does not refer to anything being broke on Intermedia's network

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

SIP Identity Settings:

Music on hold server: ?
 Send hold as inactive: on off ?
 Alert Info URL: ?
 User picture URL: ?
 Dial-Plan String: ?
 Count all groups in Dial-Plan: on off ?
 ENUM Support: on off ?
 Countrycode: ?
 Areacode: ?
 Proxy Require: ?
 Additional supported headers: ?
 Q-Value: ?
 Proposed Expiry: ?
 Auto Answer: on off ?
 Long SIP-Contact (RFC3840): on off ?
 Support broken Registrar: on off ?
 Shared Line: on off ?
 Publish Presence on bootup: on off ?
 DTMF via SIP INFO: ?
 Send display name on INVITE: on off ?
 Extension Monitoring Call Pickup List URI: ?
 Contact List: on off ?
 Contact List URI: ?
 Force sendrecv on INVITE with no SDP: on off ?
 Server Type Support: ?
 Remove all bindings on unregister: on off ?
 Subscription Expiry (s): ?
 Failed Subscription Retry Time (s): ?
 Enable hook flash: on off ?
 Identity can receive calls: on off ?
 Allow incoming extension monitoring: on off ?
 Extension monitoring group ID: ?
 Device Feature Key Synchronisation: on off ?

Apply

The SNOM Paging Server should now be registered with us and should be able to receive calls. When calling this device you will not hear ringing and might hear a beep. This is normal as this device is setup by default to automatically answer calls and transmit audio to the Intercom Speakers that are connected.

In addition to the SIP credentials provided by Intermedia, the following settings are recommended.

- Setup > Identity 1 > NAT > Keepalive interval (seconds) > **300**

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NAT Identity Settings:

Offer ICE: on off ?

STUN server (IP-addr:port): ?

STUN interval (seconds): ?

Keepalive interval (seconds): ?

- Setup > Identity 1 > RTP > Codec > g729,pcmu,telephone-event
 - Copy and Paste the above string into the Codec field

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RTP Identity Settings:

Codec: ?

Packet Size: ?

Full SDP Answer: on off ?

Symmetrical RTP: on off ?

RTP Encryption: on off ?

G.726 Byte Order: RFC3551 AAL2 ?

SRTP Auth-tag: AES-32 AES-80 ?

RTP/SAVP: ?

Media Transport Offer: ?

Media Transport Offer Setup: ?

Multicast relay address: ?

- Setup > Identity 1 > SIP > DTMF via SIP INFO > **ON**

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SIP Identity Settings:

Music on hold server: ?

Send hold as inactive: on off ?

Alert Info URL: ?

User picture URL: ?

Dial-Plan String: ?

Count all groups in Dial-Plan: on off ?

ENUM Support: on off ?

Countrycode: ?

Areacode: ?

Proxy Require: ?

Additional supported headers: ?

Q-Value: 1.0 ?

Proposed Expiry: 3600 ?

Auto Answer: on off ?

Long SIP-Contact (RFC3840): on off ?

Support broken Registrar: on off ?

Shared Line: on off ?

Publish Presence on bootup: on off ?

DTMF via SIP INFO: ?

Send display name on INVITE: on off ?

Extension Monitoring Call Pickup List URI: ?

Contact List: on off ?

Contact List URI: ?

Force sendrecv on INVITE with no SDP: on off ?

Server Type Support: Default ?

Remove all bindings on unregister: on off ?

Subscription Expiry (s): 3600 ?

Failed Subscription Retry Time (s): 600 ?

Enable hook flash: on off ?

Identity can receive calls: on off ?

Allow incoming extension monitoring: on off ?

Extension monitoring group ID: ?

Device Feature Key Synchronisation: on off ?

- 1. Setup > Advanced > SIP/RTP > Network identity (port)
 - 6xxx, xxx is the Ext #
 - I.E. your Ext is 895, your Network identity port becomes $6xxx + 895 = 6895$

- 2. Setup > Advanced > SIP/RTP > Dynamic RTP port start > 30000
- 3. Setup > Advanced > SIP/RTP > Dynamic RTP port stop > 50000

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

SIP:

Network identity (port): 1. ?

SIP T1 (ms): ?

Timer Support (RFC4028): on off ?

SIP Session Timer (s): ?

SIP Dirty Host TTL (s): ?

SIP Max Forwards: ?

ENUM Suffix: ?

Retry interval after failed registration (s): ?

Use user:phone: on off ?

Publish Presence: on off ?

Refer-To Brackets: on off ?

Require PRACK: on off ?

Send PRACK: on off ?

Offer GRUU: on off ?

Offer MPO: on off ?

Use Outbound: on off ?

Use SIP Compact Headers: on off ?

Listen on SIP TCP port: on off ?

Register HTTP contact: on off ?

Disable blind transfer (REFER): on off ?

Disable deflection (code 302): on off ?

Show History-Info: on off ?

Show Diversion: on off ?

Use NAPTR on SIP URIs: on off ?

Encode display name: on off ?

Voice Quality Report Collector: ?

RTCP-XR Report Format: ?

Check SDP Version: on off ?

Release Transferred Party On: ?

Retrieve Transferred Party On: ?

Allow SIP Settings: on off ?

Minibrowser:

XML NOTIFY Support: on off ?

RTP/RTCP:

Dynamic RTP port start: 2. ?

Dynamic RTP port stop: 3. ?

RTCP Support: on off ?

RTP Keepalive: on off ?

The above settings will require a reboot of your SNOM Paging Server. Once complete your Paging Server will be reachable from its phone number and 3 digit extension assigned by your Host PBX Service with Intermedia.

Now you are good to go! Please feel free to review the following articles for Firewall Rules and Port Requirements for the service to work correctly

- <https://kb.intermedia.net/Article/3119>
- <https://kb.intermedia.net/Article/3042>