FREQUENTLY ASKED QUESTION

snom phones used together with Asterisk PBX software

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snom

The SIP Experts

1.0 Asterisk in general

Asterisk is a complete PBX in software. It runs on Linux and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in three protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response and Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway).

More information about supported features can be found at http://www.asterisk.org. Asterisk was originally written by Mark Spencer of Digium dba Linux Support Services Inc. Code has been contributed from Open Source coders around the world.

Hardware for Asterisk can be purchased from Digium Inc. (http://www.digium.com).

2.0 Asterisk in terms of SIP

As SIP protocol is much simpler and doesn't rule out any of the features required in the PBX world, nowadays many people prefer it. Basic SIP building blocks include, for example, SIP proxy and registrar. Asterisk can't really be described as either of these. It is closer to a media gateway with SIP proxy/registrar- type features that make it possible to build a feature-rich PBX system (or network).

3.0 Supported phones

All snom phone models can be used with Asterisk.

4.0 Steps required in order to set up a snom phone to work with Asterisk

4.1 Possible topology

Obviously, there are a lot of different PBX network topologies. The following example should provide an idea of what needs to be taken into account when using snom phones.

Example

Asterisk PBX is at the address 10.1.1.1.

The phones are 10.1.1.2 and 10.1.1.3.

```
ASTERISK      SNOM1      SNOM2
    |           |          |
----------------------------------------
 1 0.1.1.1    10.1.1.2   10.1.1.3
```

4.2 Asterisk configuration

The relevant files for SIP phones in Asterisk are sip.conf, extensions.conf and voicemail.conf.

The contents depend heavily on what kind of dial plan and usernames are wanted. While some people use the phone extension as a username, others prefer to keep this separate.

<u>sip.conf</u>

sip.conf describes some general SIP parameters and all the SIP devices in the Asterisk PBX system.

sip.conf

```
[general]
port = 5060            ; Port to bind to
bindaddr = 0.0.0.0     ; Address to bind to A
context = default      ;Default for incoming calls
disallow=gsm
allow=alaw             ; Allow a-law
;allow=g729
disallow=ulaw

[910]
type=friend            (inbound and outbound calls accepted)
secret=password_for_this_phone
host=dynamic
callerid=JOHN <910>
defaultip=10.1.1.2     (will be used if not yet registered)
dtmfmode=inband        (use this with inband mode DTMF)
;dtmfmode=rfc2833      (use this with outband mode DTMF)
mailbox=910            (Asterisk VM-system's mailbox #)

[maria] (here a name is used instead of #)
type=friend
secret=password_for_this_phone
host=dynamic
callerid=MARIA <916>
defaultip=10.1.1.3
dtmfmode=inband
mailbox=916

etc.
```

extensions.conf

Extensions.conf describes the dialplan for the Asterisk PBX system. It can be used in many ways. Please check the Asterisk sample files that come with the software.

To reach your snom phone, you can, for example, have the following three lines for extension 910:

extensions.conf

```
exten => 910,1,Dial,sip/${EXTEN}|30
exten => 910,2,voicemail2,u910
exten => 910,102,voicemail2,b910
```

${EXTEN} is an internal variable. Here it means that

you are going to dial sip/910.

Details for 910 (actually the username) are in sip.conf.

Although it is also possible to use a username other than the extension number, using the extension number makes management and documentation a little easier.

Voicemail.conf

Voicemail.conf describes how all the mailboxes should behave.

910 => 1234, John Fischer, support@snom.de

910 is the mailbox number.

1234 is the password.

Using the mail address makes it possible to receive voicemails as a mail attachment (wav).

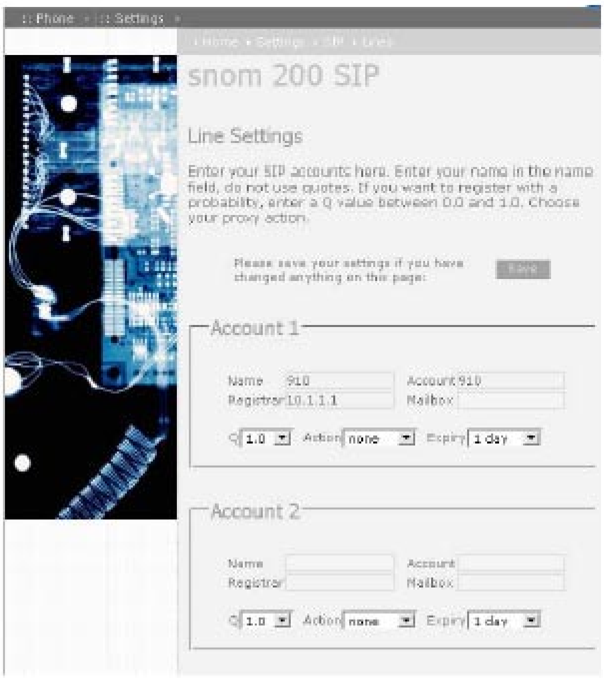
MWI also works with Asterisk.

If someone has left you a voicemail, you will receive indication of this (MWI). In snom 200, this will be displayed in two ways: a yellow LED will blink and there will be an MWI on the display. This is cleared only when you check your voicemails (and delete them).

5.0 snom configuration

In order to use snom phones with Asterisk, you will need to configure some SIP parameters.

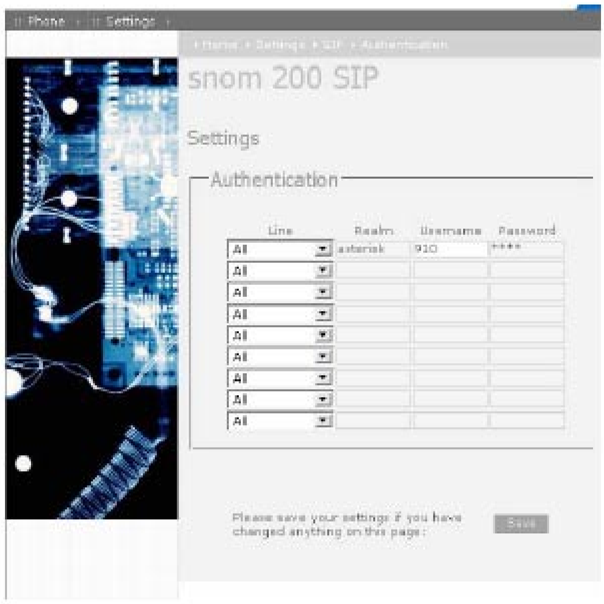
5.1 SIP Lines



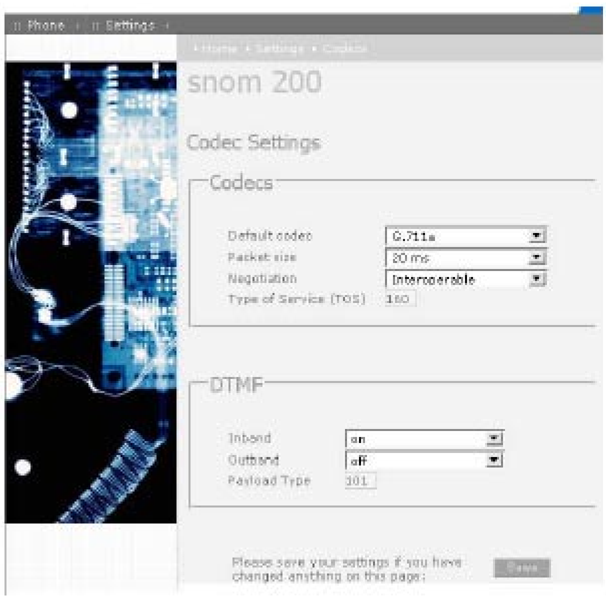
Fill in the name, account and registrar.

Use Asterisk's IP address in the registrar field. If you are using Asterisk, do not configure the mailbox here.

5.2 SIP Authentication



5.3 SIP codecs



snom and Asterisk both support several codecs but unlike snom, a separate license is required for Asterisk when using g.729 codec (Contact Digium inc.)

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