

SNOM 300/320 Configuration Manual



Latest update



Unknown macro: 'page-info'

Procedure for upgrade firmware



The official firmware supported by PrivateWave is 8.4.35 .

a. Open the Web User Interface of the snom and navigate to the Software Update page.

b. Copy and paste this URL:

<http://provisioning.snom.com/download/fw/snom300-8.4.35-SIP-f.bin>

into the Firmware field and press load

c. The phone reboots and may ask you to perform the update, click 'Yes'.



Do not disconnect the power at anytime!

After that, the phone is upgraded to version 8.4.35.

Basic Configuration of the SNOM 300/320:

Open the Web User Interface of the SNOM

Step 1	Step 2
<p>Navigate to the Setup/Identity1 page, login tab:</p> <ul style="list-style-type: none">• Set Account, Authentication Username and password with the correct data that you have.• Set Registrar with the IP Address (or DNS) of Server Sip.• Set Outbound Proxy in this form : 'sips:ip_of_the_srv(or dns):5061'	<p>Goto in the Setup/Identity1 page, SIP tab:</p> <ul style="list-style-type: none">• Set Support Broken registrar to ON.• DTMF via SIP Info set to ON.

Login	SIP	NAT	RTP
<p>Login Information:</p> <p>Identity active: <input checked="" type="radio"/> on <input type="radio"/> off ?</p> <p>Displayname: <input type="text" value="XXXXXX"/> ?</p> <p>Account: <input type="text" value="###"/> ?</p> <p>Password: <input type="text" value="*****"/> ?</p> <p>Registrar: <input type="text" value="pbx.example.it"/> ?</p> <p>Outbound Proxy: <input type="text" value="sips:pbx.example.it:5061"/> ?</p> <p>Failover Identity: <input type="text" value="None"/> ?</p> <p>Authentication Username: <input type="text" value="###"/> ?</p> <p>Mailbox: <input type="text"/> ?</p> <p>Ringtone: <input type="text" value="Custom Melody"/> ?</p> <p>Custom Melody URL: <input type="text"/> ?</p> <p>Display text for idle screen: <input type="text" value="XXX (###)"/> ?</p> <p>Ring After Delay (sec): <input type="text"/> ?</p> <p>Record Missed Calls: <input checked="" type="radio"/> on <input type="radio"/> off ?</p> <p>Record Dialed Calls: <input checked="" type="radio"/> on <input type="radio"/> off ?</p> <p>Record Received Calls: <input checked="" type="radio"/> on <input type="radio"/> off ?</p> <p><input type="button" value="Save"/> <input type="button" value="Re-Register"/> <input type="button" value="Play Ringer"/></p> <p><input type="button" value="Remove Identity"/> <input type="button" value="Remove All Identities"/></p>			
<p>SIP Identity Settings:</p> <p>Music on hold server:</p> <p>Send hold as inactive:</p> <p>Alert Info URL:</p> <p>User picture URL:</p> <p>Dial-Plan String:</p> <p>ENUM Support:</p> <p>Countrycode:</p> <p>Areacode:</p> <p>Proxy Require:</p> <p>Additional supported headers:</p> <p>Q-Value:</p> <p>Proposed Expiry:</p> <p>Auto Answer:</p> <p>Long SIP-Contact (RFC3840): <input type="text" value="Support broken Registrar"/></p> <p>Shared Line:</p> <p>Publish Presence on bootstrap: <input type="text" value="DTMF via SIP INFO"/></p> <p>Send display name on INVITE:</p> <p>Extension Monitoring Call Pickup List:</p> <p>Extension Monitoring Call Pickup List:</p> <p>Contact List:</p> <p>Contact List URI:</p> <p>Server Type Support:</p> <p>Remove all bindings on unregister:</p> <p>Subscription Expiry (s):</p> <p>Failed Subscription Retry Time (s):</p> <p>Enable hook flash:</p> <p><input type="button" value="Save"/></p>			

Step 3	Step 4
<p>Goto in the Setup/Identity1 page, RTP tab:</p> <ul style="list-style-type: none"> • Set RTP Encryption to ON. • Set SRTP Auth-tag to AES-32. • Set RTP/SAVP to Mandatory. • Set Packet Size to 20ms. • Set Media Transport Offer to UDP. 	<p>Goto in the Certificates:</p> <ul style="list-style-type: none"> • Enable the server identity check in TLS connection by pushing the "TLS" button in "Unknown Certificates".

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

RTP Identity Settings:

Codec 1: ?
 Codec 2: ?
 Codec 3: ?
 Codec 4: ?
 Codec 5: ?
 Codec 6: ?
 Codec 7: ?

Packet Size: ?
 Full SDP Answer: on off ?
 Symmetrical RTP: on off ?
 RTP Encryption: on off ?
 Dynamic G.726 payload: on off ?
 G.726 Byte Order: RFC3551 AAL2 ?
 SRTP Auth-tag: AES-32 AES-80 ?
 RTP/SAVP: ?
 Media Transport Offer: ?
 Media Transport Offer Setup: ?

Advanced Configuration of the SNOM 300/320:

With the Advanced configuration we enable on the SNOM phone the "3-way call" and the "call transfer" functions.

Step 1	Step 2
Navigate to the Setup/Function Keys page: <ul style="list-style-type: none"> Choose a dial pad button to edit (in the example it's P5) Click on the drop-down menu of the "Action" (i.e. the second column) 	<ul style="list-style-type: none"> Choose "DTMF" Save by pushing the "Save" button at t

Call Lists

Prev. Outgoing ID

Next Outgoing ID

Clear Pickup Info

Menu

Redial

<input type="checkbox"/>	P1	Active	Line	
<input type="checkbox"/>	P2	Active	Line	
<input type="checkbox"/>	P3	Active	Line	
<input type="checkbox"/>	P4	Active	Key Event	Directory
<input type="checkbox"/>	P5	Active	DTMF	**
<input type="checkbox"/>	P6	Active	Key Event	Mute

Save

Prev. Outgoing

Clear Pickup Info

<input type="checkbox"/>	P1	Active	
<input type="checkbox"/>	P2	Active	
<input type="checkbox"/>	P3	Active	
<input type="checkbox"/>	P4	Active	
<input type="checkbox"/>	P5	Active	
<input type="checkbox"/>	P6	Active	

Step 3

Insert the "**1*" string in the third column: These DTMF activate the transfer mode.

<input type="checkbox"/>	P5	Active	DTMF	*1*
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Step 4

Act the same way for another dial button (i.e. the "3-way call")

<input type="checkbox"/>	P4	Active	
<input type="checkbox"/>	P5	Active	
<input type="checkbox"/>	P6	Active	

Save