

# SIP VoIP Settings User Guide

Version 1.0; October 10, 2007

VoIP

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## Change history

October 10, 2007	Version 1.0	Initial document release

# 1 Introduction

This user guide describes the [SIP Voice over IP \(VoIP\) Settings](#) [1] application and how to use it. The application version 1.0 is applicable for Nokia S60 VoIP Release 2.0 and 2.1 products. This document is intended for very experienced application developers.

This user guide provides:

- Introduction to the SIP VoIP Settings application
- List of features
- Application settings

For more detailed information on the settings configuration of Nokia S60 VoIP implementation, see [Nokia S60 VoIP Implementation Configuration Tutorial](#) [2].

**Note:** The modification of the SIP VoIP settings described in this document should be tried using a Nokia S60 VoIP device. All screen shots and menu examples in this document have been taken using the Nokia N80 Internet Edition. The menu contents or the location of applications and folders may vary according to the device.

## 2 SIP VoIP Settings application

SIP VoIP Settings is a standalone application with which the VoIP settings can be configured on the terminal User Interface (UI). It can be used to create new VoIP profiles and modify the existing ones. It can also be used to create NAT firewall (FW) settings and modify the VoIP parameters, including the codec and NAT firewall settings, which are normally not visible on the terminal UI.

With the SIP VoIP Settings application it is also possible to save all the VoIP profile settings to a text file.

**Note:** You can only edit your own VoIP settings. The provisioned VoIP settings cannot be edited nor displayed, only copied.

The services provided include the following:

### VoIP services

- Create VoIP profile: The user can manually create a VoIP profile by using a default or an existing VoIP profile.
- Delete VoIP profile: The user can manually delete a VoIP profile.
- Save settings to file: The user can save all VoIP profile settings to a text file.
- Edit VoIP profile settings: The user can view and modify VoIP profile-specific settings.
- Select SIP profile: The user can view the name of the SIP profile used in the VoIP profile and select another SIP profile to be used instead of the current one.
- Create audio codec: The user can manually create an audio codec.
- Delete audio codec: The user can manually delete an audio codec.
- Move audio codec: The user can change the order (priority) of the codecs.
- Edit audio codec settings: The user can view and modify audio codec settings.

### NAT firewall settings

- Create domain-specific settings: The user can manually create domain-specific settings.
- Edit domain-specific settings: The user can view and modify domain-specific settings.
- Delete domain-specific settings: The user can manually delete the domain-specific settings.
- Create IAP-specific settings: The user can manually create IAP-specific settings.
- Edit IAP-specific settings: The user can view and modify IAP-specific settings.
- Delete IAP-specific settings: The user can manually delete the IAP-specific settings.

### 3 VoIP services

VoIP profile includes VoIP service-specific settings, such as:

- VoIP service name
- VoIP service parameters
- Speech codec settings
- SIP profiles used

The VoIP profile name is the same as the VoIP service name shown on the terminal UI. If a VoIP service provider also configures the access networks used that have, for example, different billing or connectivity mode, the VoIP service can be divided into one or multiple VoIP profiles. If the service provider does not set up the access networks, only one VoIP and one SIP profile are needed. The VoIP profile settings are linked to the access point selected when the user is successfully registered to the service.

To access the VoIP services, select **SIP VoIP Settings > VoIP services > Options > Open** (see Figure 1 and Figure 2).

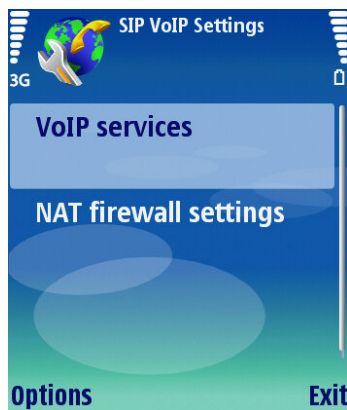


Figure 1: Select VoIP services

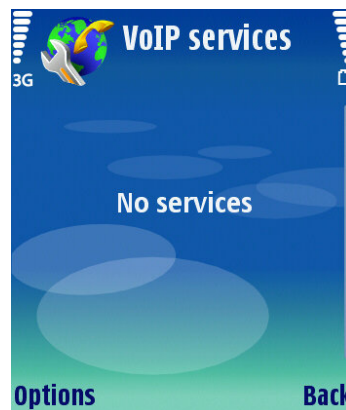


Figure 2: No VoIP services available

To create a new VoIP profile, select **VoIP services > Options > New > Use default profile/Use existing profile** (see Figure 3). If using an existing VoIP profile, select one to be copied (see Figure 4 – Figure 5).

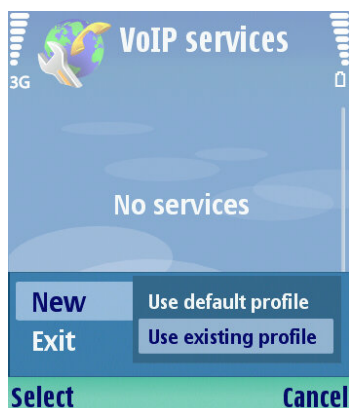


Figure 3: Create a new VoIP profile

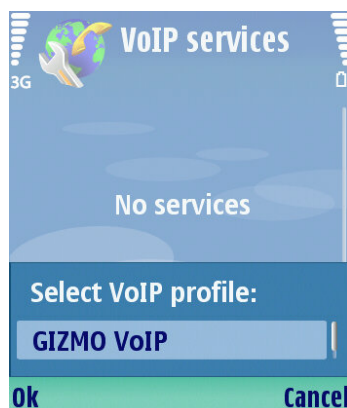


Figure 4: Select the VoIP profile

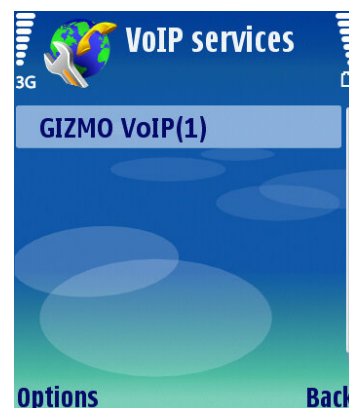


Figure 5: X VoIP profile created

To delete a VoIP profile or to save all VoIP profile settings to a text file, select **VoIP services > X VoIP profile > Options > Delete/Save to file** (see Figure 6). Select the memory and location for the file to be saved (see Figure 7 and Figure 8). The saved text file can be used, for example, in creating an XML file for OMA device management (DM).

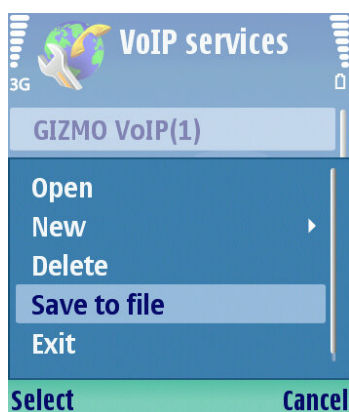


Figure 6: Save VoIP profile settings to a file

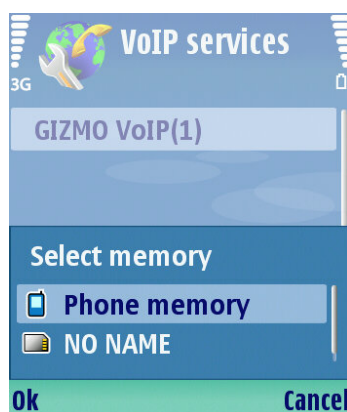


Figure 7: Select the memory

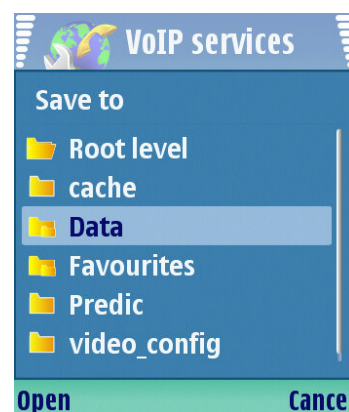


Figure 8: Select the location for the file to be saved

### 3.1 Profile settings

To modify the VoIP profile-specific settings, select **X VoIP profile > Profile settings > X parameter > Options > Change** (see Figure 9 – Figure 11).

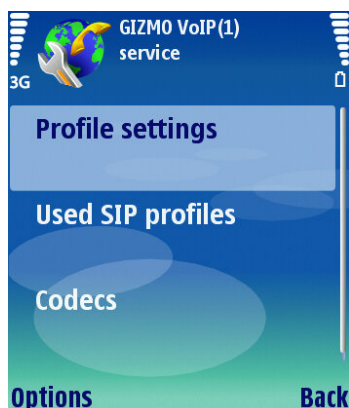


Figure 9: VoIP profile settings

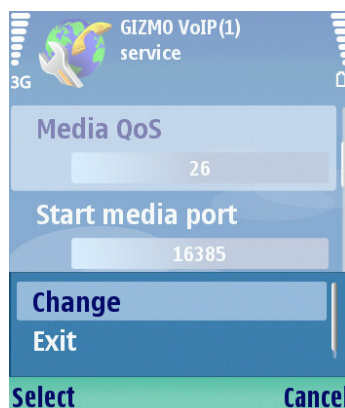


Figure 10: Select the profile parameter to be modified

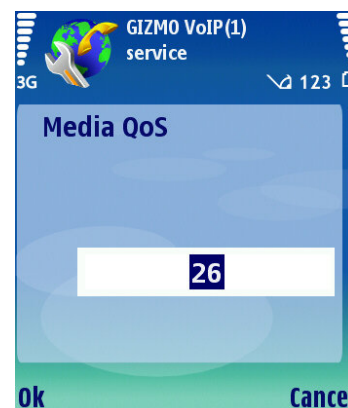


Figure 11: Modify the profile parameter value

- **Provider Name:**
  - Text, the maximum length is 32.
  - Provider of the VoIP profile settings as described in /R.1/ OMA-WAP-ProvCont-v1\_1-20021112-C.
  - This text is displayed on the terminal UI as the sender of the settings and cannot be edited.
- **Profile Name:**
  - Text, the maximum length is 32.
  - Based on the provisioning parameter NAME as described in /R.1/ OMA-WAP-ProvCont-v1\_1-20021112-C.
  - This text is displayed on the terminal UI as the service name.
- **Media QoS:**
  - A number in range of 0–63.
  - Quality of Service for VoIP media. DiffServ Code Point (Diffserv, DSCP bits) QoS values used in IP headers (Ipv4 TOS and Ipv6 TC). IETF RFC 2598, an Expedited Forwarding PHB. The IETF [3] and WMM [4] specifications conflict on the QoS values used for voice packets. From S60 3rd Edition, FP1 onwards, the U-APSD power save scheme of WMM is also enabled with the IETF default value (46), if the feature is supported by the terminal and the WLAN access point.
  - Default value: '46'
- **Start media port:**
  - An even number in range of 1024–65534.
  - The lower limit of the RTP port range.
  - Default value: '49152'
- **End media port:**
  - An even number in range of 1024–65534.
  - The upper limit for the allocated RTP ports. The value shall be at least 4 over the 'Start media port number' to guarantee two simultaneous calls.
  - Default value: '65534'

- **DTMF inband:**
  - DTMF tones are sent as compressed audio; they are part of the actual VoIP call audio stream. Note that the DTMF tones may be degraded if a high-compression rate codec (AMR-NB, G.729 or iLBC) is in use for a VoIP call.
  - It is not recommended to change this value because if enabled (see below) and if supported by the other peer in the VoIP call, the DTMF tones are sent as out-band.
  - 'On': Enabled
  - 'Off': Disabled
  - Default value: 'On'
- **DTMF outband:**
  - DTMF tones are sent as RTP payload as specified in IETF RFC 2833. If both in- and out-band DTMF signalling methods are enabled (setting value '1'), the DTMF out-band mode is used if the peer supports it.
  - Typically, both in- and out-band DTMF should be enabled; however, disabling the out-band signalling is required in some special cases.
  - 'On': DTMF digits out-band are generated, if requested by the remote side.
  - 'Off': DTMF digits out-band are not generated.
  - Default value: 'On'
- **Secure call preference:**
  - This parameter enables media security (secure RTP) if SIP TLS has been used for signalling. Supported from Nokia S60 VoIP Release 2.1 onwards.
  - 'Prefer secure': A secure call is preferred. If the other end does not support security, a fallback to non-secure call takes place.
  - 'Prefer non-secure': A non-secure call is preferred.
  - 'Use secure only': Security is mandatory for mobile originated (MO) call establishment.
  - Default value: 'Prefer non-secure'
- **Allow VoIP over WCDMA:**
  - If this setting is enabled, the Internet telephone application shows also the available WCDMA access points.
  - 'On': VoIP over WCDMA is allowed.
  - 'Off': VoIP over WCDMA is not allowed.
  - Default value: 'Off'
- **RTCP reporting:**
  - This setting enables the Real-Time Transport Control Protocol (RTCP) reports defined in RFC 3550.
  - 'On': RTCP reporting is enabled.
  - 'Off': RTCP reporting is disabled.
  - Default value: 'Off'
- **UA Header: term. type:**
  - 'On': The terminal type is appended to the UA header.
  - 'Off': The terminal type is not appended to the UA header.

- Default value: 'On'
- **UA Header: MAC address:**
  - 'OFF': The MAC address is not appended to the UA header.
  - 'ON': The MAC address is appended to the UA header.
  - Default value: 'OFF'
- **UA Header: free string:**
  - Text, the maximum length is 32.
  - User agent information string that is appended to the SIP UA header, for example, to separate two different configurations using different IAPs.
- **Count of VoIP digits:**
  - This parameter defines the meaningful count of caller ID characters for caller identification. Supported from Nokia S60 VoIP Release 2.1 onwards.
  - '0': All caller ID characters are meaningful.
  - '3–20': Number of meaningful caller ID characters.
  - Default value: '0'
- **Ignoring domain part:**
  - This parameter defines the rule for displaying the domain part of an address (URI) for incoming Internet calls in the user interface. Supported from Nokia S60 VoIP Release 2.1 onwards.
  - 'On': The domain part is not displayed if only E.164 numbers are used in the user part of the URI.
  - 'Numbers only': The domain part is never displayed.
  - 'Off': The domain part is displayed.
  - Default value: 'Off'

### 3.2 Used SIP profiles

IDs of the SIP profiles used by this VoIP profile. The VoIP profile can refer to one or many SIP profiles. To view the name of the SIP profile used in the VoIP profile or to select another SIP profile to be used instead of the current one, select **VoIP services > X VoIP profile > Used SIP profiles** (see Figure 12 and Figure 13). If there are several SIP profiles, select the profile to be used from the list.

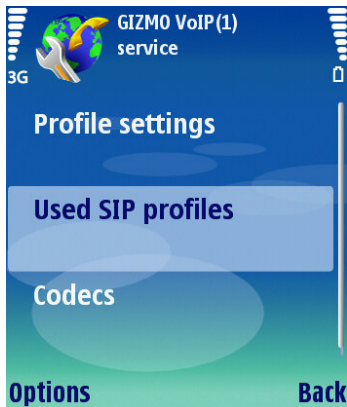


Figure 12: View the used SIP profiles

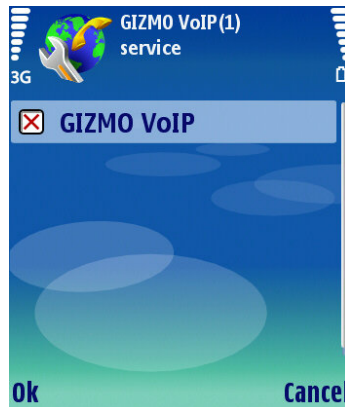


Figure 13: Select a SIP profile

### 3.3 Codecs

The VoIP profile includes settings for one or multiple speech codecs. For codec order, see Section 3.1 in [Nokia S60 VoIP Implementation Configuration Tutorial](#) [2].

To modify the speech codec settings, select **VoIP services > X VoIP profile > Codecs > X codec > X parameter > Options > Change** (see Figure 14 – Figure 19).

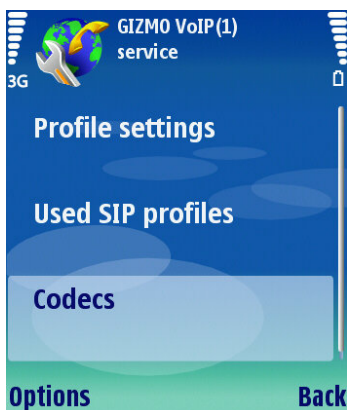


Figure 14: Speech codecs

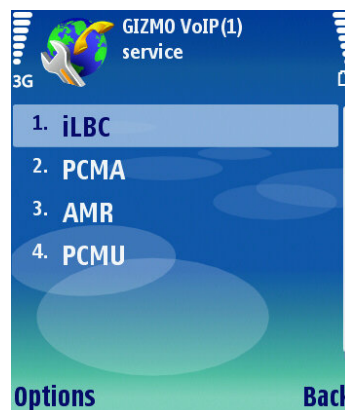


Figure 15: Codecs available

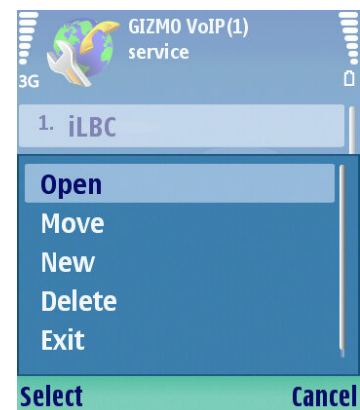


Figure 16: Open the X codec

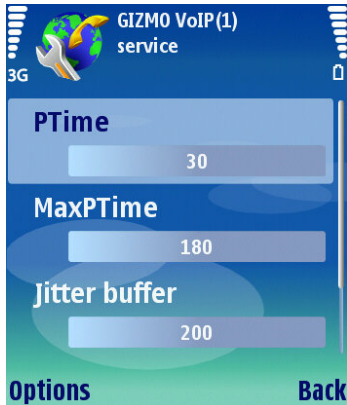


Figure 17: Codec parameters available

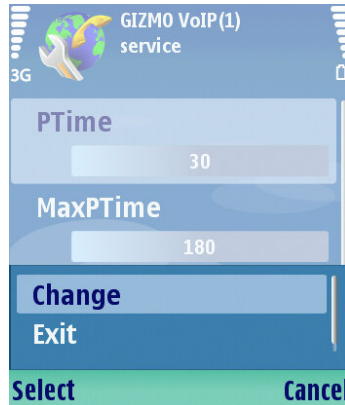


Figure 18: Select the codec parameter to be modified

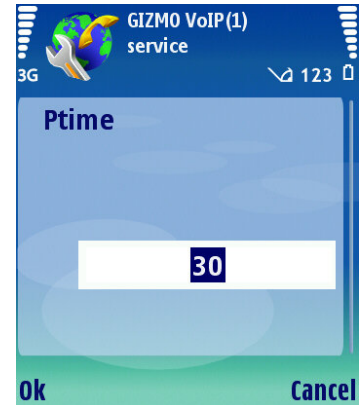


Figure 19: Modify the codec parameter value

To change the order of the speech codecs, select **Codecs > X codec > Options > Move** (see Figure 20). Select a new position for the codec (see Figure 21 and Figure 22).

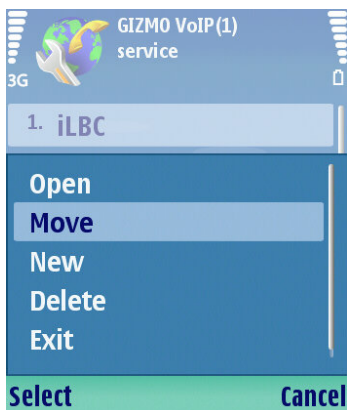


Figure 20: Move a speech codec

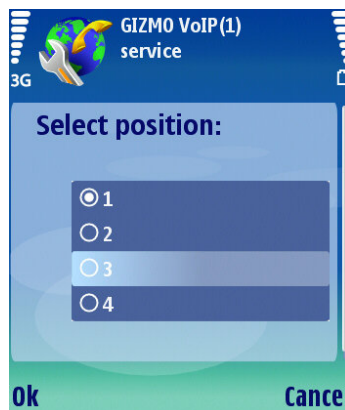


Figure 21: Select a new position for the codec

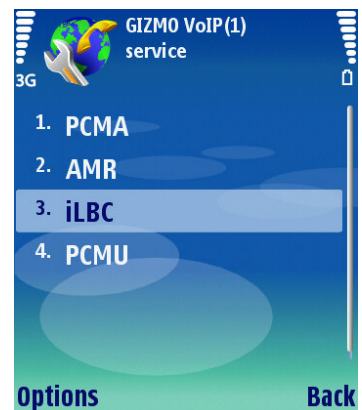


Figure 22: Codec position changed

To create a new speech codec or to delete one, select **Codecs > Options > New/Delete** (see Figure 23). Select the codec to be added (see Figure 24 and Figure 25).

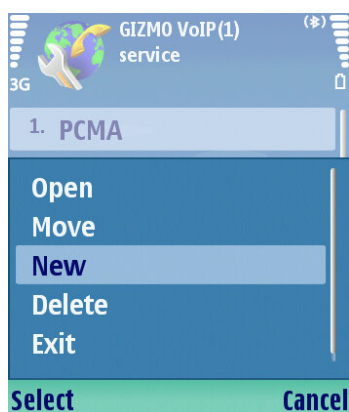


Figure 23: Add a new speech codec

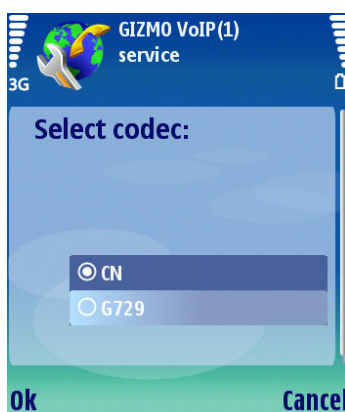


Figure 24: Select the codec to be added



Figure 25: A new codec added

### 3.3.1 AMR NB codec

- **PTime:**
  - The length of time in milliseconds represented by the media in a packet. The ptime may vary between the codec's default ptime and maxptime so that the ptime is increased by the multiples of its allowed values. If other allowed values are not mentioned, the default value and its multiples should be considered as the allowed value.
  - Default value: '20', which means a 20 ms speech block in one RTP packet.
- **MaxPTime:**
  - Time in milliseconds; a value in range of 20–200.
  - The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. If this parameter is not present, the sender may encapsulate any number of speech frames into one RTP packet. This attribute is probably only meaningful for audio data, but may be used with other media types if it makes sense. It is a media attribute, and is not dependent on the charset. Note that this attribute was introduced after RFC 2327, and non-updated implementations will ignore this attribute.
  - Default value: '200'
- **Jitter buffer:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **Voice activ. detection:**
  - Enabling VoIP Discontinuous Transmission (DTX), that is, RTP packets are not sent during silent periods; AMR generates Silence Description (SID) packets also during inactivity, but the packet frequency is reduced.
  - 'On': Enabled
  - 'Off': Disabled
  - Default value: 'Off'

- **Octet align:**
  - 'On': Enabled. Octet-aligned framing used according to RFC 3267.
  - 'Off': Disabled. Bandwidth efficient framing is employed.
  - Default value: 'Off'

### 3.3.2 PCMU (G.711 $\mu$ -law) codec

- **PTime:**
  - The length of time in milliseconds represented by the media in a packet. The ptime may vary between the codec's default ptime and maxptime so that the ptime is increased by the multiples of its allowed values. If other allowed values are not mentioned, the default value and its multiples should be considered as the allowed value.
  - Default value: '20'
- **MaxPTime:**
  - Time in milliseconds; a value in range of 20–200.
  - The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. If this parameter is not present, the sender may encapsulate any number of speech frames into one RTP packet. This attribute is probably only meaningful for audio data, but may be used with other media types if it makes sense. It is a media attribute, and is not dependent on the charset. Note that this attribute was introduced after RFC 2327, and non-updated implementations will ignore this attribute.
  - Default value: '200'
- **Jitter buffer:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **Voice activ. detection:**
  - Enabling VoIP DTX, that is, RTP packets are not sent during silent periods; Comfort Noise packets are also generated during inactivity if enabled as CN codec, but the packet frequency is reduced.
  - 'On': Enabled
  - 'Off': Disabled
  - Default value: 'Off'

### 3.3.3 PCMA (G.711 A-law) codec

- **PTime:**
  - The length of time in milliseconds represented by the media in a packet. The ptime may vary between the codec's default ptime and maxptime so that the ptime is increased by the multiples of its allowed values. If other allowed values are not mentioned, the default value and its multiples should be considered as the allowed value.
  - Default value: '20'
- **MaxPTime:**
  - Time in milliseconds; a value in range of 20–200.

- The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. If this parameter is not present, the sender may encapsulate any number of speech frames into one RTP packet. This attribute is probably only meaningful for audio data, but may be used with other media types if it makes sense. It is a media attribute, and is not dependent on the charset. Note that this attribute was introduced after RFC 2327, and non-updated implementations will ignore this attribute.
- Default value: '200'
- **Jitter buffer:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **Voice activ. detection:**
  - Enabling VoIP DTX, that is, RTP packets are not sent during silent periods; Comfort Noise packets are also generated during inactivity if enabled as CN codec, but the packet frequency is reduced.
  - 'On': Enabled
  - 'Off': Disabled
  - Default value: 'Off'

#### 3.3.4 iLBC codec

- **PTime:**
  - The length of time in milliseconds represented by the media in a packet. The ptime may vary between the codec's default ptime and maxptime so that the ptime is increased by the multiples of its allowed values. If other allowed values are not mentioned, the default value and its multiples should be considered as the allowed value. The allowed values for this codec are 20 and 30 or their multiples.
  - Default value: '30'
- **MaxPTime:**
  - Time in milliseconds; a value in range of 20–200.
  - The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. If this parameter is not present, the sender may encapsulate any number of speech frames into one RTP packet. This attribute is probably only meaningful for audio data, but may be used with other media types if it makes sense. It is a media attribute, and is not dependent on the charset. Note that this attribute was introduced after RFC 2327, and non-updated implementations will ignore this attribute.
  - Default value: '180'
- **Jitter buffer:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **Voice activ. detection:**
  - 'On': Enabled

- 'Off': Disabled
- Default value: 'Off'

### 3.3.5 G.729 codec

- **PTime:**
  - The length of time in milliseconds represented by the media in a packet. The ptime may vary between the codec's default ptime and maxptime so that the ptime is increased by the multiples of its allowed values. If other allowed values are not mentioned, the default value and its multiples should be considered as the allowed value. The allowed values for this codec are 10 or its multiples.
  - Default value: '20'
- **MaxPTime:**
  - Time in milliseconds; a value in range of 10–200.
  - The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. If this parameter is not present, the sender may encapsulate any number of speech frames into one RTP packet. This attribute is probably only meaningful for audio data, but may be used with other media types if it makes sense. It is a media attribute, and is not dependent on the charset. Note that this attribute was introduced after RFC 2327, and non-updated implementations will ignore this attribute.
  - Default value: '200'
- **Jitter buffer:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **Voice activ. detection:**
  - 'On': Enabled
  - 'Off': Disabled
  - Default value: 'Off'
- **AnnexB:**
  - A number. Enable enhancement according to IETF RFC 3555 annex-b.
  - 'On': Yes
  - 'Off': No
  - Default value: 'Off'

### 3.3.6 Comfort Noise (CN) codec

This codec is typically included if PCMU or PCMA is enabled.

## 4 NAT firewall settings

Nokia S60 VoIP implementation has STUN protocol support for NAT traversal and NAT binding refresh features. The NAT firewall traversal features enable the VoIP function behind NATs of certain type. With the SIP VoIP Settings application the user can also create the STUN settings.

NAT firewall settings by default refer to SIP domain-specific settings. Refresh timers are overridden with IAP-specific NAT firewall settings, if IAP-specific values are defined.

To access the NAT firewall domain or IAP-specific settings, select **SIP VoIP Settings > NAT firewall settings > Options > Open** (see Figure 26 and Figure 27).

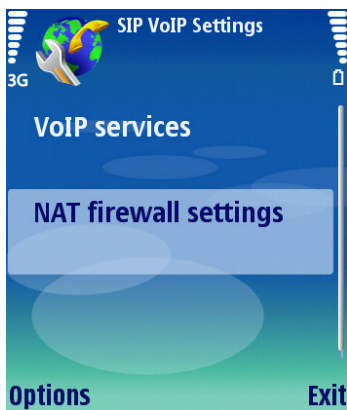


Figure 26: NAT firewall settings

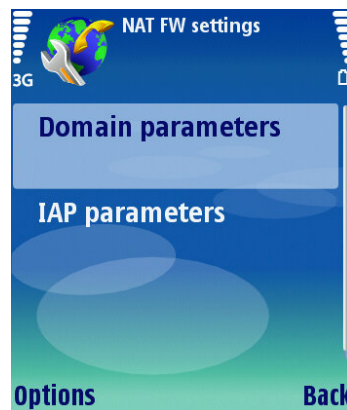


Figure 27: Domain and IAP parameters

### 4.1 Domain parameters

To create domain parameters, select **Domain parameters > Create parameters > Select Domain** (see Figure 28 – Figure 30).

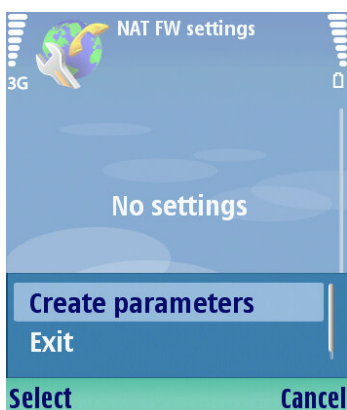


Figure 28: Create domain parameters

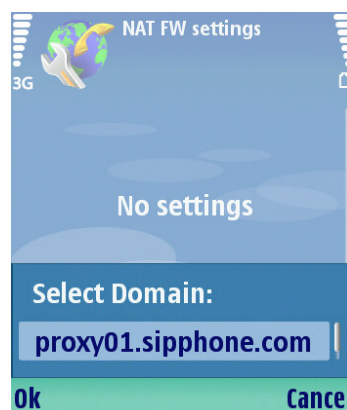


Figure 29: Select the domain parameters



Figure 30: X domain parameters created

To modify the domain-specific settings, select **X domain > X parameter > Options > Change** (see Figure 31 – Figure 33).

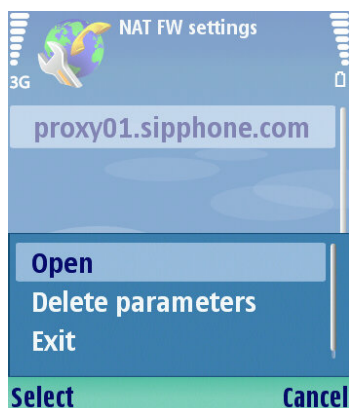


Figure 31: Open the X domain



Figure 32: Select the domain parameter to be modified



Figure 33: Modify the domain parameter value

- **STUN server name:**
  - This parameter defines the STUN server address in the domain-specific NAT-FW settings. Optional. By default, the DNS SRV query tries to find the STUN server.
  - Apply value 0.0.0.0 to disable the STUN server, for example, if a SBC is taking care of the NAT traversal.
  - Default value: not defined.
- **STUN server port:**
  - A number in range of 1024–65535.
  - This parameter defines the STUN server port in the domain-specific NAT-FW settings. Optional.
  - Default value: '3478'
- **TCP NAT bind. refresh:**
  - A number in range of 0–9999.
  - This parameter defines the NAT refresh interval for TCP in the domain-specific NAT-FW settings. The unit of the refresh interval is seconds. If an IAP-specific value for this interval is defined, it overrides this value. Optional.
  - Default value: '1200'
- **UDP NAT bind. refresh:**
  - A number in range of 0–9999.
  - This parameter defines the NAT refresh interval for UDP in the domain-specific NAT-FW settings. The unit of the refresh interval is seconds. If an IAP-specific value for this interval is defined, it overrides this value. Optional.
  - Default value: '28'
- **CRLF refresh:**
  - This parameter defines the usage of CRLF-based NAT binding refresh. This attribute enables the CRLF refresh to the outbound proxy (or to the registrar if no proxy is defined) over any transport. Optional, but enabling is strongly recommended if it is known that there is either a NAT or firewall on the route, or if the SIP proxy requires refresh to keep the persistent TCP/TLS connection alive.
  - 'On': Enabled

- 'Off': Disabled
- Default value: 'Off'

## 4.2 IAP parameters

As default, only those IAPs are available that have parameters defined.

To create IAP parameters, select **IAP parameters > Create parameters > Select IAP connection** (see Figure 34 – Figure 36).



Figure 34: Select the IAP parameters

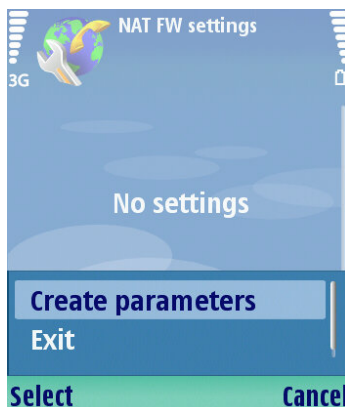


Figure 35: Create parameters for an access point



Figure 36: Select the access point to be modified

To modify the IAP-specific settings, select **X access point > X parameter > Options > Change** (see Figure 37 – Figure 39).



Figure 37: Open the access point to modify its parameters



Figure 38: Select the IAP parameter to be modified



Figure 39: Modify the IAP parameter value

- **TCP NAT bind. refresh:**
  - A number in range of 0–9999.
  - This parameter defines the NAT refresh interval for TCP in the IAP-specific NAT-FW settings. The unit of the refresh interval is seconds. The value overrides the domain-specific NAT Refresh TCP value, if it is defined. Optional.
  - Default value: '1200'

- **UDP NAT bind. refresh:**
  - A number in range of 0–9999.
  - This parameter defines the NAT refresh interval for UDP in the IAP-specific NAT-FW settings. The unit of the refresh interval is seconds. The value overrides the domain-specific NAT Refresh UDP value, if it is defined. Optional.
  - Default value: '28'
- **STUN retransmission:**
  - A number in range of 0–9999.
  - This parameter defines the STUN request retransmit timer (time in milliseconds) in the IAP-specific NAT-FW settings. Optional.
  - Default value: '250'

## 5 Terms and abbreviations

Term or abbreviation	Meaning
3GPP	The 3rd Generation Partnership Project
a-law	Name of G.711 PCMU algorithm (European)
AMR	Adaptive Multi Rate
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Comfort Noise
CP	Client Provisioning
DM	Device Management
DND	Do Not Disturb
DSCP	DiffServ Code Point
DTMF	Dual-Tone Multi Frequency
DTX	Discontinuous Transmission
FW	Firewall
HTTP	Hyper Text Transport Protocol
IAP	Internet Access Point
ID	Identity
IEEE	Institute of Electrical and Electronics Engineers ( <a href="http://www.ieee.org">www.ieee.org</a> )
IETF	The Internet Engineering Task Force ( <a href="http://www.ietf.org">www.ietf.org</a> )
iLBC	Internet Low Bitrate Codec
IMS	IP Multimedia System
IP	Internet Protocol
Maxptime	The maximum amount of media which can be encapsulated in a payload packet
NAT	Network Address Translation
NB	Narrow Band
OMA	Open Mobile Alliance ( <a href="http://www.openmobilealliance.org">www.openmobilealliance.org</a> )
OTA	Over-the-Air
PCMA	Pulse Code Modulation a-law
PCMU	Pulse Code Modulation $\mu$ -law
Ptime	Packetization interval
PHB	Per-Hop forwarding Behaviour
QoS	Quality of Service
RFC	Request For Comments

Term or abbreviation	Meaning
RTP	Real-Time Transport Protocol
SBC	Session Border Controller
SID	Silence Description
SIP	Session Initiation Protocol
SW	Software
STUN	Simple Traversal of UDP through NAT; a protocol that allows applications to detect that network address translation (NAT) is being used.
TEL	Telephony
TC	Traffic Class
TLS	Transport Layer Security
TOS	Type of Service
U-APSD	Unsolicited Automatic Power Save Delivery
URI	Uniform Resource Identifier
VAD	Voice Activation Detection
VoIP	Voice over IP
WLAN	Wireless LAN, Wireless Local Area Network
WMM	Wireless Multimedia
μ-law	Name of G.711 PCMU algorithm (North American)

## 6 References

- [1] [SIP VoIP Settings](http://www.forum.nokia.com) application available at <http://www.forum.nokia.com>
- [2] [Nokia S60 VoIP Implementation Configuration Tutorial](http://www.forum.nokia.com) available at <http://www.forum.nokia.com>
- [3] IETF RFC 2598, An Expedited Forwarding PHB, [www.ietf.org](http://www.ietf.org)
- [4] IEEE Standard 802.11e, Wireless Multimedia (WMM) Specification, [www.ieee.org](http://www.ieee.org)

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