

SIP VoIP Release 3.x Settings Application User Guide

Version 2.0; 7 May 2009

VoIP

NOKIA

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Change history

1 October 2008	Version 1.0	Initial document release
7 May 2009	Version 2.0	Updates throughout the document to describe the use of SIP VoIP Release 3.x Settings 2.0

1 Introduction

This user guide describes the SIP VoIP Release 3.x Settings 2.0 application and how to use it. Version 2.0 of the application is designed to work on S60 3rd Edition devices that implement Nokia S60 VoIP Release 3.0 and 3.1. This document is intended for very experienced application developers.

The guide provides:

- An introduction to the SIP VoIP Release 3.x Settings application.
- A list of features.
- A description of application settings.

For more detailed information about the configuration of settings for Nokia S60 VoIP implementations, see the [Nokia S60 VoIP Release 3.0 Configuration Tutorial](#) [1].

Note: The modification of SIP VoIP settings described in this document should be done using a Nokia S60 VoIP-enabled device. All screen shots and menus are examples only. Menu contents and the location of applications and folders may vary according to the device.

2 SIP VoIP Settings application

SIP VoIP Settings is a UI plug-in for the S60 internet communication settings application that can be used to configure VoIP settings. The plug-in can be used to both create new VoIP services and modify existing ones. It can also be used to create NAT firewall (FW) settings and to modify the VoIP parameters, including the codec and NAT firewall settings, which are normally not visible on the device UI.

The SIP VoIP Release 3.x Settings application also makes it possible to export the VoIP profile settings to a file or send the settings to another device via Bluetooth connectivity.

To access the SIP VoIP Release 3.x Settings application, select **Menu > Connectivity > Net settings**, or, in some devices, **Menu > Control panel > Net settings** (see Figure 1).



Figure 1: The Net settings application on an S60 device.

The features provided by the application include:

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.
Export settings:	The user can export VoIP profile settings to a file or send them to another device via Bluetooth connectivity.
Import settings:	The user can import VoIP profile settings from a file or from another device via Bluetooth connectivity.
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.

Add audio codec:	The user can manually add 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 Internet Access Point (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

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

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

The VoIP provider name is the same as the VoIP service name shown on the device. The VoIP profile settings are linked to a network destination created automatically when the user has successfully activated the service from the Contacts application.

To access the VoIP service settings, select **Net Settings > Advanced VoIP settings > VoIP services > Options > Open** (see Figure 2 – Figure 4).

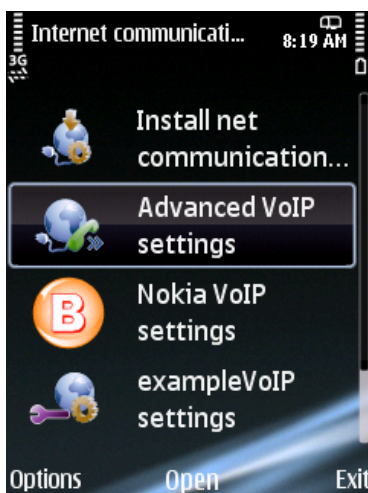


Figure 2: Select Sip VoIP settings

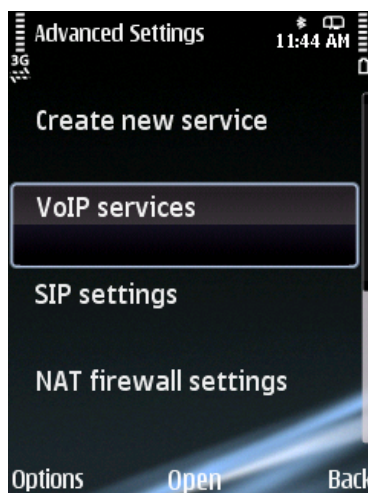


Figure 3: Select VoIP services

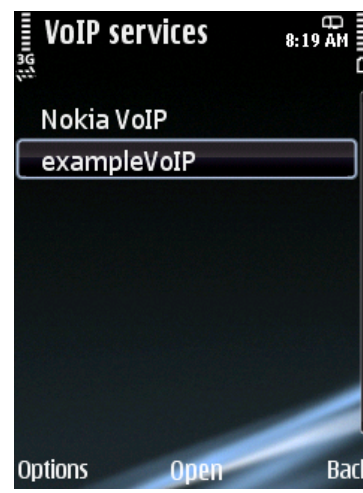


Figure 4: VoIP services available

3.1 Creating, editing, and removing VoIP settings

To create a new VoIP service, select **Create new service**, or **VoIP services > Options > New service > Use default profile/Use existing profile**. If using an existing VoIP profile, select one to be copied (see Figure 5 – Figure 7). When creating a VoIP service, a SIP profile must be selected for the VoIP profile. If no SIP profiles are available, the user must create one (see Figure 8 – Figure 10). Finally, when a SIP profile is selected or created, the user can add presence settings with default values to the service (see Figure 11 – Figure 13).



Figure 5: Create a new VoIP service

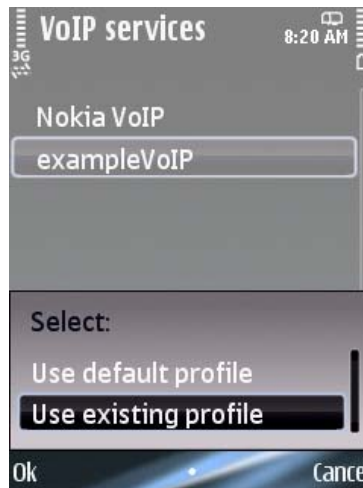


Figure 6: Select creation option

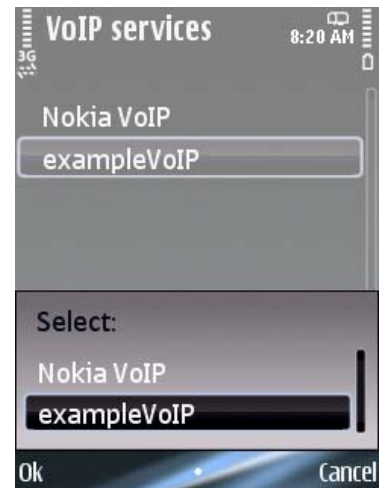


Figure 7: Select the VoIP profile



Figure 8: Create a new SIP profile (or select an existing one)



Figure 9: Type full SIP user name and password



Figure 10: Ok button becomes visible when full user name with domain is given (Password field is optional)

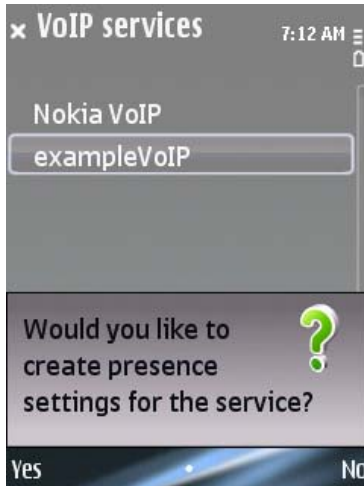


Figure 11: Create new presence settings

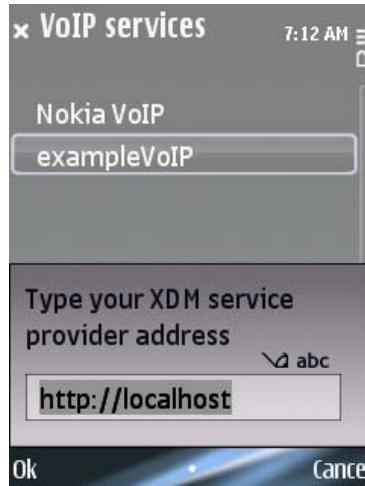


Figure 12: XDM URI is needed. Seed value http://localhost is automatically filled into URI field

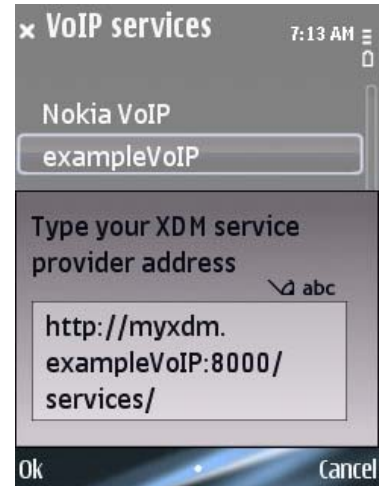


Figure 13: Type XDM URI

To delete a VoIP profile, select **VoIP services > X VoIP profile > Options > Remove** (see Figure 14 – Figure 16).

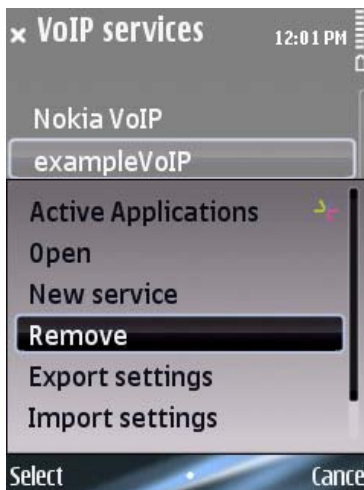


Figure 14: Select VoIP profile and Remove

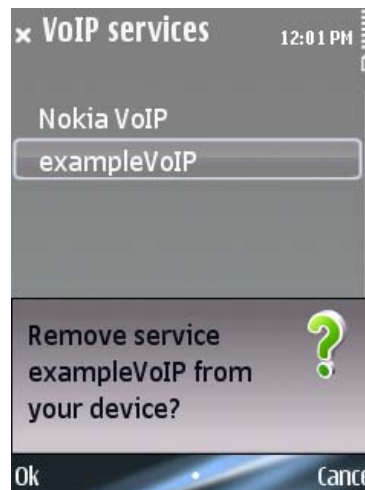


Figure 15: Confirm removal by selecting Ok

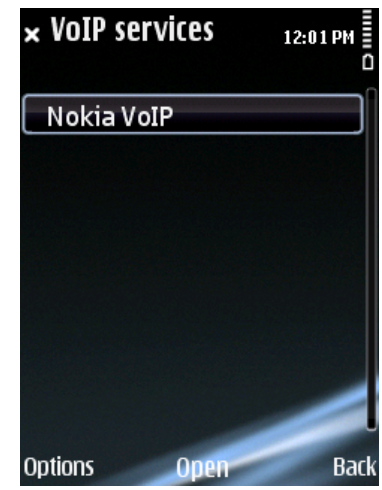


Figure 16: VoIP services available

3.2 Exporting and importing VoIP services

To export VoIP service to a file or send to another Nokia S60 VoIP Release 3.x device via Bluetooth connectivity, select **VoIP services > X VoIP profile > Options > Export settings** (see Figure 17 – Figure 18).

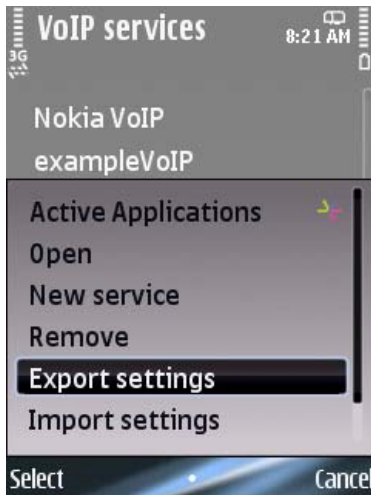


Figure 17: Export settings

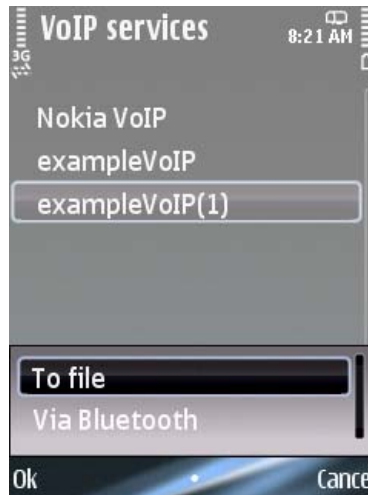


Figure 18: Select To file

If settings are to be exported to a file, select the memory and location in which the file will be saved (see Figure 19 – Figure 22). The saved file can be used for importing settings later.

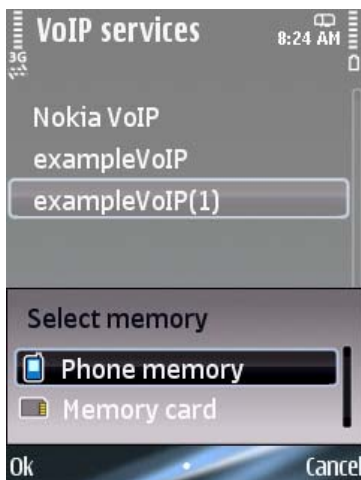


Figure 19: Select the memory

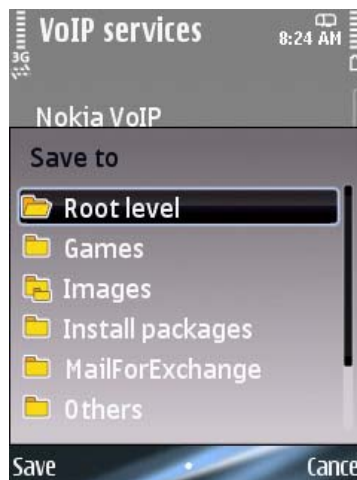


Figure 20: Select the location for the file to be saved

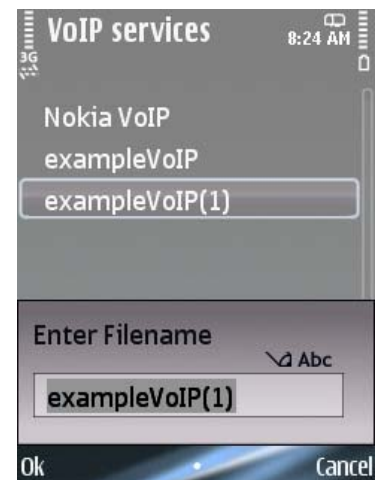


Figure 21: Enter name for the file to be saved



Figure 22: Export successful

If settings are exported to another device via Bluetooth connectivity, select a Bluetooth device from the list (see Figure 23 – Figure 25).

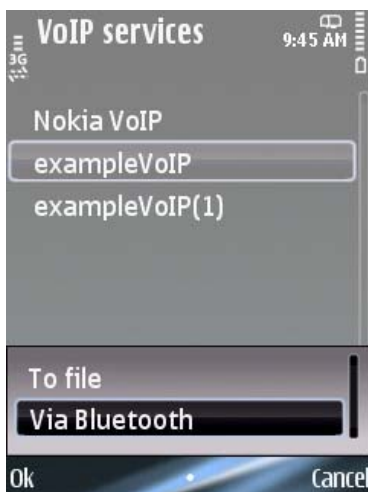


Figure 23: Select Via Bluetooth



Figure 24: Select device from list

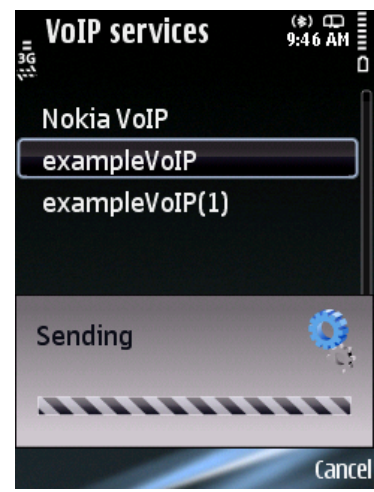


Figure 25: Settings are being sent

To import VoIP settings from a file or directly from another device, select **VoIP services > X VoIP profile > Options > Import** (see Figure 26).

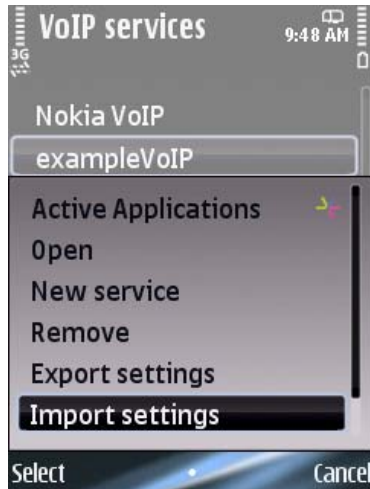


Figure 26: Select Import settings

If settings are to be imported from a file, select the memory and location of the file (see Figure 27 – Figure 29).

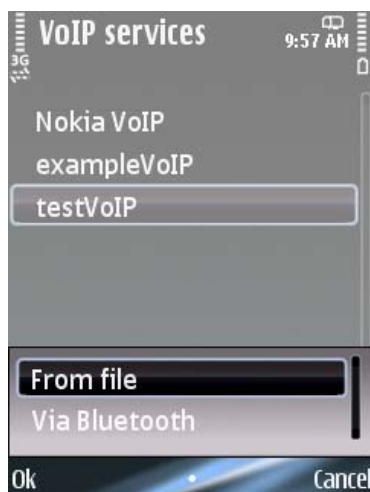


Figure 27: Select From file



Figure 28: Select the memory

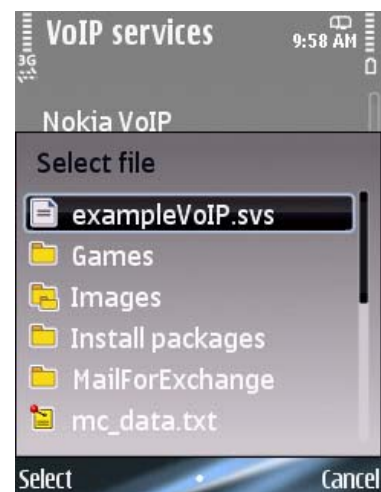


Figure 29: Select the file to be imported

If settings are to be imported from another device, select a Bluetooth device from the list (see Figure 30 – Figure 31).

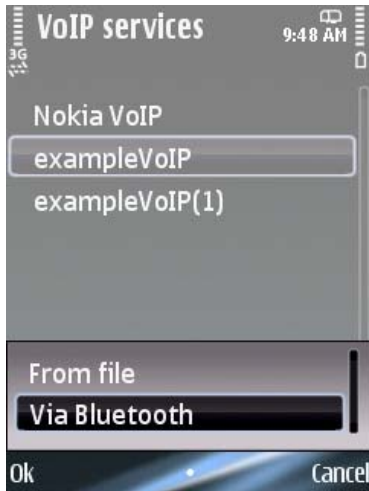


Figure 30: Select Via Bluetooth

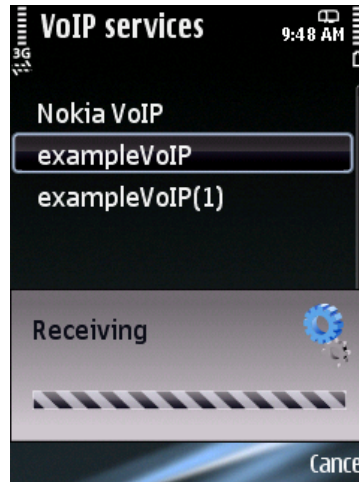


Figure 31: Settings are being received

After a successful import, the Contacts application is automatically launched.

3.3 Profile settings

In this view, the user can edit profile settings for provider-specific VoIP service.

To modify the VoIP profile-specific settings, select **X VoIP profile > Profile settings > X parameter > Options > Change** (see Figure 32 – Figure 34).



Figure 32: VoIP profile settings



Figure 33: Select the profile parameter to be modified



Figure 34: Modify the profile parameter value

- **Provider name:**
 - Text, the maximum length is 64.
 - 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.
 - If the VoIP profile was created using the SIP VoIP Release 3.x Settings application, then it can be edited.
 - This text is displayed on the device as the service name.
- **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.
- **Media QoS:**
 - A number in the 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 [2] and WMM [3] specifications conflict on the QoS values used for voice packets. 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 the range of 1024–65530.
 - The lower limit of the RTP port range.
 - Default value: '49152'.
- **End media port:**
 - An even number in the range of 1028–65534.
 - The upper limit for the allocated RTP ports. The value must be at least four over the 'Start media port number' to guarantee two simultaneous calls.
 - Default value: '65534'.
- **DTMF in-band:**
 - 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 out-band:**
 - 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 (SCP):**
 - Parameter that enables media security (secure RTP) if SIP TLS has been used for signalling.
 - 'Prefer secure': A secure call is preferred. If the other end does not support security, a fallback to a nonsecure call takes place.
 - 'Prefer non-secure': A nonsecure call is preferred.
 - 'Use secure only': Security is mandatory for mobile-originated (MO) call establishment.
 - Default value: 'Prefer nonsecure'.
- **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'.

- **SIP VoIP User-Agent header: Terminal type display (UATT):**
 - 'On': The terminal type is appended to the User-Agent header.
 - 'Off': The terminal type is not appended to the User-Agent header.
 - Default value: 'On'.
- **SIP VoIP User-Agent header: WLAN MAC address display (UAWMAC):**
 - 'On': The MAC address is appended to the User-Agent header.
 - 'Off': The MAC address is not appended to the User-Agent header.
 - Default value: 'Off'.
- **SIP VoIP User-Agent header: Free string (UAHS):**
 - Text, the maximum length is 32.
 - User agent information string that is appended to the SIP User-Agent header, for example, to separate two different configurations using different IAPs.
- **Count of VoIP digits:**
 - Parameter that defines the meaningful count of caller ID characters for caller identification.
 - '0': All caller ID characters are meaningful.
 - '3-20': Number of meaningful caller ID characters.
 - Default value: '0'.
- **Ignoring domain part:**
 - Parameter that defines the rule for displaying the domain part of an address (URI) in the Logs application.
 - 'On': The domain part is never displayed.
 - 'Off': The domain part is displayed.
 - 'Numbers only': The domain part is not displayed if only E.164 numbers are used in the user part of the URI.
 - Default value: 'Off'.
- **Voice mailbox settings ID:**
 - SIP settings name. If this parameter is not present, the SIP profile used by the VoIP profile is used.
- **Voice mailbox preferred IAP ID:**
 - Internet Access Point (IAP). If this parameter is not present, the IAP/SNAP used by the SIP settings is used.
- **Voice mailbox resubscribe:**
 - Parameter that defines the interval for sending a re-SUBSCRIBE message in seconds in the range of 0 – 86400.
 - Default value: 600.
- **Voice mailbox address (Vmbx MWI URI):**
 - A SIP or tel URI (IETF RFC 3261 or 3966) defining the Message Waiting Indication (MWI) address of the user's IP voice mailbox. For example, *sip:alice@voicemailbox.example.com*.

- **Voice mailbox listen address (Vmbx listening URI):**
 - A SIP or tel URI (IETF RFC 3261 or 3966) defining the listening address of the user's IP voice mailbox. For example, *sip:alice@listen.example.com*.
- **Presence settings ID:**
 - Name of the presence settings.
- **Auto-accept buddy requests:**
 - Parameter that defines the rule for automatic acceptance of presence subscription requests.
 - 'On': Buddy requests are accepted automatically.
 - 'Off': Buddy requests are not accepted automatically.
 - Default value: 'Off'.
- **Add 'user=phone' to all numbers:**
 - Parameter that defines the rule for adding the text string 'user=phone' to all SIP URIs containing only numbers as a user name.
 - 'On': User=phone is added to all numbers.
 - 'Off': User=phone is not added to all numbers.
 - Default value: 'Off'.
- **SIP connectivity test URI:**
 - Defines the address to be used in SIP connectivity test calls.
- **Provider bookmark URI:**
 - Parameter that defines the URI for the service provider bookmark, for example, *http://www.example.com*.
- **Minimum value for session interval:**
 - Parameter that defines the minimum value for a session interval in delta seconds in the range of 90 – 9999.
 - Default value: '90'.
- **SIP session interval:**
 - Parameter that defines the session interval for a SIP session in seconds; a value in the range of 0-3600.
 - Default value: '1800'.
- **Branding URI:**
 - Parameter that defines the URI for branding data.

3.4 Used SIP profiles

This is the SIP profile used by the VoIP profile being created or edited. The VoIP profile can only refer to one SIP profile and one SIP profile can only be used by one VoIP profile. 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 35 and Figure 36). If there are several SIP profiles, select the profile to be used from the list.



Figure 35: View the used SIP profiles



Figure 36: Select a SIP profile

3.5 Codecs

The VoIP profile includes settings for one or multiple speech codecs. For codec order, see Section 3.1 in the *Nokia S60 VoIP Release 3.0 Configuration Tutorial* [1].

To modify the speech codec settings, select **VoIP services > X VoIP profile > Codecs > X codec > X parameter > Options > Change** (see Figure 37 – Figure 42).



Figure 37: Speech codecs



Figure 38: Codecs available



Figure 39: Open the X codec



Figure 40: Available codec parameters



Figure 41: Select the codec parameter to be modified

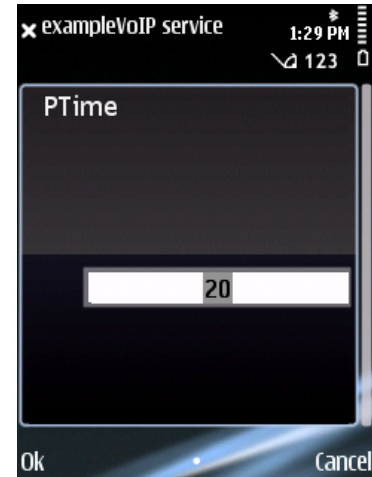


Figure 42: Modify the codec parameter value

To change the order of the speech codecs, select **Codecs > X codec > Options > Move** (see Figure 43). Select a new position for the codec (see Figure 44 and Figure 45).

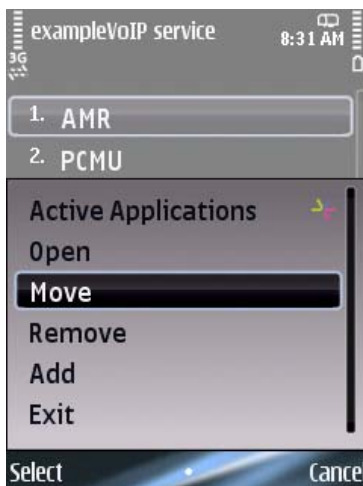


Figure 43: Move a speech codec

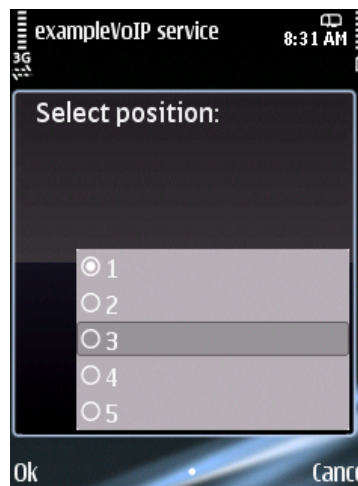


Figure 44: Select a new position for the codec

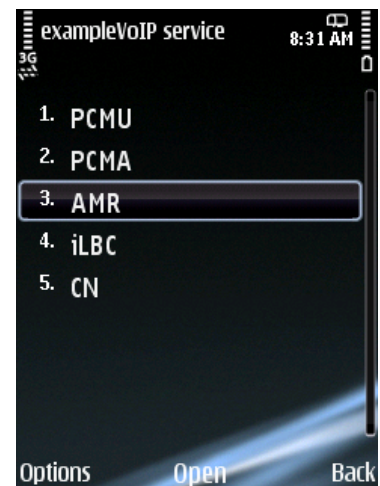


Figure 45: Codec position changed

To create a new speech codec or to delete one, select **Codecs > Options > Add/Remove** (see Figure 46). Select the codec to be added (see Figure 47 and Figure 48).



Figure 46: Add a new speech codec

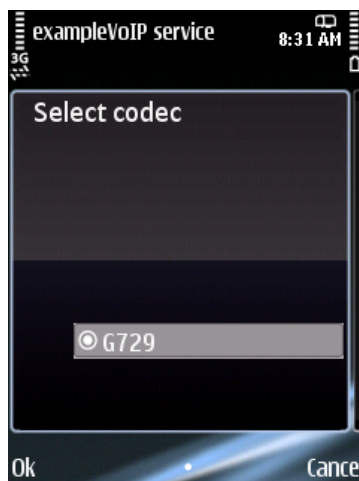


Figure 47: Select the codec to be added

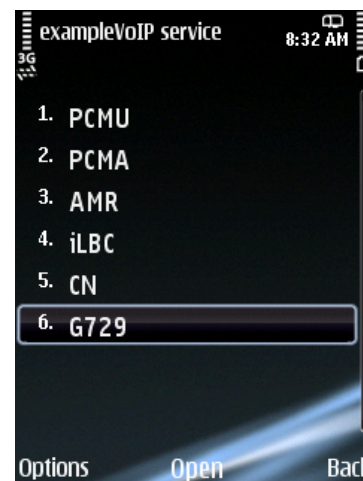


Figure 48: A new codec added

3.5.1 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 the range of 20–200.
 - The maximum amount of media that 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore this attribute.
 - Default value: '200'.
- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.

- **Voice activation detection (VAD):**
 - Enables 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 4867.
 - 'Off': Disabled. Bandwidth efficient framing is employed.
 - Default value: 'Off'.
- **Mode-set:**
 - Parameter that defines the requested AMR mode set. Restricts the active codec mode set to a subset of all modes. A possible value is a list of modes from the set 0–7 in ascending order.
 - 'Off': No restriction for used modes.
 - '0': AMR 4.75 kbit/s
 - '1': AMR 5.15 kbit/s
 - '2': AMR 5.90 kbit/s
 - '3': AMR 6.70 kbit/s (PDC-EFR)
 - '4': AMR 7.40 kbit/s
 - '5': AMR 7.95 kbit/s
 - '6': AMR 10.2 kbit/s
 - '7': AMR 12.2 kbit/s (GSM-EFR)
- **Mode-change-period:**
 - Parameter that defines a number of frame-blocks, N (1 or 2), that is, the frame-block period at which codec mode changes are allowed for the sender. The initial phase of the interval is arbitrary, but changes must be separated by a period of N frame-blocks, that is, a value of two allows the sender to change the mode every second frame-block. The value of N shall be either 1 or 2. If this parameter is not present, mode changes are allowed at any time during the session, that is, N=1.
 - '1': Mode changes are allowed at any time during the session.
 - '2': Mode changes are allowed every second frame-block.
- **Mode-change-neighbour:**
 - '0': Change is allowed between any two modes in the active codec mode set.
 - '1': Mode changes will only be made to the neighbouring modes in the active codec mode set. Neighbouring modes are the ones closest in bit rate to the current mode, either the next higher or the next lower rate.

- **Maxred:**
 - Parameter that defines the maximum length of time in milliseconds that elapses between the first transmission of a frame and any redundant transmission that the sender will use. This parameter allows the receiver to have a bounded delay when redundancy is used. The allowed values are between 0 and 100. If the parameter is omitted, no limitation on the use of redundancy is present. The value must be a multiple of the used frame time.
 - '0': No redundancy will be used.
 - '20–100': The maximum length of time in milliseconds that elapses between the first transmission of a frame and any redundant transmission that the sender will use.
 - Default value: Not used.

3.5.2 AMR-WB codec

The AMR-WB codec is supported from Nokia S60 VoIP Release 3.1 onwards.

- **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 the range of 20–200.
 - The maximum amount of media that can be encapsulated in each packet, expressed as time in milliseconds. The time will 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore this attribute.
 - Default value: '200'.
- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.
- **Voice activation detection (VAD):**
 - 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 4867.
 - 'Off': Disabled. Bandwidth efficient framing is employed.
 - Default value: 'Off'.

- **Mode-set:**
 - Parameter that defines the requested AMR mode-set. Restricts the active codec mode set to a subset of all modes. A possible value is a list of modes from the set 0–8 in ascending order.
 - 'Off': No restriction for used modes.
 - '0': AMR-WB 6.60 kbit/s
 - '1': AMR-WB 8.85 kbit/s
 - '2': AMR-WB 12.65 kbit/s
 - '3': AMR-WB 14.25 kbit/s
 - '4': AMR-WB 15.85 kbit/s
 - '5': AMR-WB 18.25 kbit/s
 - '6': AMR-WB 19.85 kbit/s
 - '7': AMR-WB 23.05 kbit/s
 - '8': AMR-WB 23.85 kbit/s
- **Mode-change-period:**
 - Parameter that defines a number of frame-blocks, N (1 or 2), that is, the frame-block period at which codec mode changes are allowed for the sender. The initial phase of the interval is arbitrary, but changes must be separated by a period of N frame-blocks, that is, a value of two allows the sender to change the mode every second frame-block. The value of N shall be either 1 or 2. If this parameter is not present, mode changes are allowed at any time during the session, that is, N=1.
 - '1': Mode changes are allowed at any time during the session.
 - '2': Mode changes are allowed every second frame-block.
- **Mode-change-neighbour:**
 - '0': Change between any two modes in the active codec mode set is allowed.
 - '1': Mode changes will only be made to the neighbouring modes in the active codec mode set. Neighbouring modes are the ones closest in bit rate to the current mode, either the next higher or the next lower rate.
- **Maxred:**
 - Parameter that defines the maximum length of time in milliseconds that elapses between the first transmission of a frame and any redundant transmission that the sender uses. It allows the receiver to have a bounded delay when redundancy is used. The allowed values are between 0 and 100. If the parameter is omitted, no limitation on the use of redundancy is present. The value must be a multiple of the used frame time.
 - '0': No redundancy will be used.
 - '20–100': The maximum length of time in milliseconds that elapses between the first transmission of a frame and any redundant transmission that the sender uses.
 - Default value: Not used.

3.5.3 PCMU (G.711 μ -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. The allowed values for this codec are 10 or its multiples.
 - Default value: '20'.
- **MaxPTime:**
 - Time in milliseconds; a value in the range of 20–200.
 - The maximum amount of media that 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore it.
 - Default value: '200'.
- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.
- **Voice activation detection (VAD):**
 - 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'.

3.5.4 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. The allowed values for this codec are 10 or its multiples.
 - Default value: '20'.
- **MaxPTime:**
 - Time in milliseconds; a value in the range of 20–200.
 - The maximum amount of media that 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore it.
 - Default value: '200'.

- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.
- **Voice activation detection (VAD):**
 - Enables 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'.

3.5.5 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 the range of 20–200.
 - The maximum amount of media that 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore it.
 - Default value: '180'.
- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.
- **Voice activation detection (VAD):**
 - 'On': Enabled.
 - 'Off': Disabled.
 - Default value: 'Off'.
- **Mode-set:**
 - The iLBC codec has support for two basic frame lengths (modes): 20 ms at 15.2 kbit/s and 30 ms at 13.33 kbit/s. When the codec operates at block lengths of 20 ms, it produces 304 bits per block. Similarly, for block lengths of 30 ms, it produces 400 bits per block. The mode value is configured in the same way as the mode-set in the AMR configuration.
 - Default value: '30'. If the mode-set=20 is configured, the terminal will offer the iLBC with the mode=20 (15.2 kbit/s), but will fall back to using the mode=30 (13.33 kbit/s) if the remote end does not include the mode=20 in its answer.

3.5.6 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 the range of 10–200.
 - The maximum amount of media that 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. Note that this attribute was introduced after RFC 2327, and nonupdated implementations will ignore it.
 - Default value: '200'.
- **Jitter buffer:**
 - A positive integer (milliseconds) in the range of 20–200.
 - Default value: '200'.
- **Voice activation detection (VAD):**
 - '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'.

3.5.7 Comfort Noise (CN) codec

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

4 NAT firewall settings

Nokia S60 VoIP implementation has Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs) (STUN) support for NAT traversal and NAT binding refresh features. 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 **Advanced VoIP Settings > NAT firewall settings > Options > Open** (see Figure 49 and Figure 50).



Figure 49: NAT firewall settings

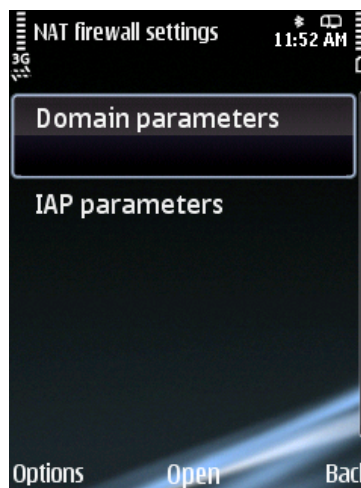


Figure 50: Domain and IAP parameters

4.1 Domain parameters

To create domain parameters, select **Domain parameters > Create parameters > Select domain** (see Figure 51 – Figure 53).

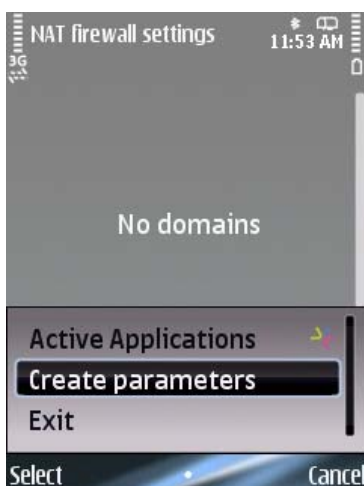


Figure 51: Create domain parameters

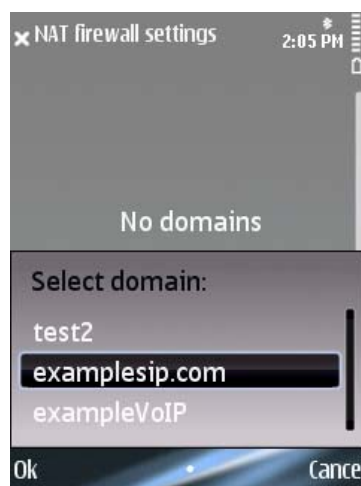


Figure 52: Select the SIP domain

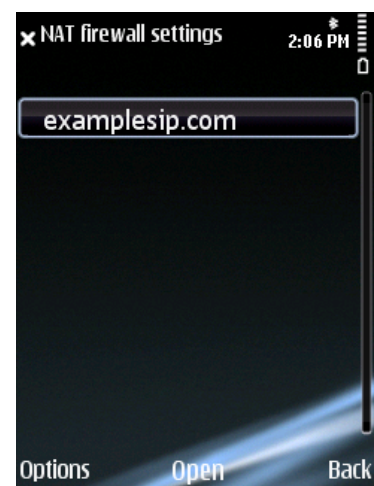


Figure 53: X domain parameters created

To modify the domain-specific settings, select X domain > X parameter > Options > Change (see Figure 54 –Figure 56) or X domain > STUN Settings > X parameter > Options > Change (see Figure 57 – Figure 59).

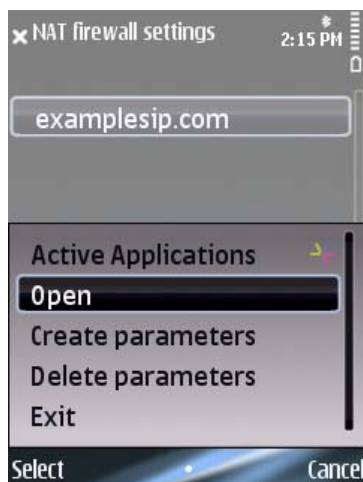


Figure 54: Open the X domain



Figure 55: Select the domain parameter to be modified



Figure 56: Modify the domain parameter value



Figure 57: Open the STUN settings



Figure 58: Select the parameter to be modified



Figure 59: Modify the parameter value

- **STUN settings:**
 - In this view, the user can change STUN settings.
- **TCP NAT bind. refresh:**
 - A number in the range of 0–9999.
 - Parameter that 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.
 - Default value: '1200'.

- **UDP NAT bind. refresh:**
 - A number in the range of 0–9999.
 - Parameter that 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.
 - Default value: '28'.
- **CRLF refresh:**
 - Parameter that defines the usage of CRLF-based NAT binding refresh. The attribute enables the CRLF refresh to the outbound proxy (or to the registrar if no proxy is defined) over any transport. Though optional, enabling is strongly recommended if 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'.
- **Used NAT protocol:**
 - Parameter that defines the NAT protocol used. Optional.
 - Default value: 'nokia.stun'.

4.1.1 STUN settings

- **STUN server address:**
 - Parameter that defines the STUN server address in the domain-specific NAT-FW settings. If no address is defined, the UE tries to resolve the STUN server using the DNS SRV query.
 - Apply value 0.0.0.0 to disable the STUN servers, for example, if an SBC is taking care of the NAT traversal.
- **STUN server port:**
 - Parameter that defines the STUN server port in the domain-specific NAT-FW settings. Optional.
 - Default value: '3478'.
- **STUN server user name:**
 - Needed for STUN server authentication with long-term credentials. Short-term credentials that do not need to be configured are generally used.
- **STUN server password:**
 - Needed for STUN server authentication with long-term credentials.
- **STUN shared secret:**
 - Parameter that indicates whether the STUN server supports TLS and shared secret or not.
 - 'Off': STUN shared secret mechanism is not used.
 - 'On': STUN shared secret mechanism is used.
 - Default value: 'Off'.

- **Start port number:**
 - A number in the range of 49152–65535.
 - The lower limit of the domain port range. Do not edit this value unless for a specific reason. In any case, do not set the port range to zero because using the same value for start and end port numbers will prevent all VoIP calls.
 - Default value: '49152'.
- **End port number:**
 - A number in the range of 49152–65535.
 - The upper limit of the domain port range. Do not edit this value unless for a specific reason. In any case, do not set the port range to zero because using the same value for start and end port numbers will prevent all VoIP calls.
 - Default value: '65535'.

4.2 IAP parameters

As the default, only those IAPs with defined parameters are available.

To create IAP parameters or to modify the IAP-specific settings, select **IAP parameters > X access point > X parameter > Options > Create parameters/Open** (see Figure 60 – Figure 65).



Figure 60: Select the IAP parameters



Figure 61: Create new IAP-specific NAT-FW parameters



Figure 62: Select the access point for which settings are to be created



Figure 63: Open the settings to be modified



Figure 64: Select the IAP parameter to be modified

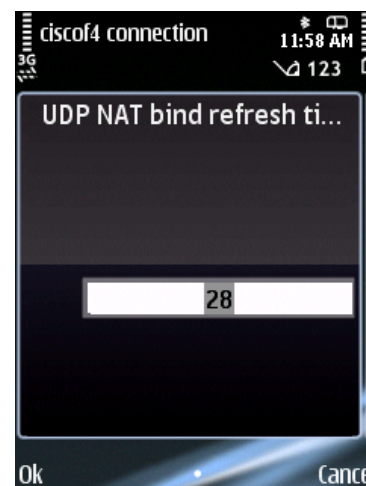


Figure 65: Modify the IAP parameter value

- **TCP NAT bind. refresh:**
 - A number in the range of 0–9999.
 - Parameter that 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.
 - Default value: '1200'.
- **UDP NAT bind. refresh:**
 - A number in the range of 0–9999.
 - Parameter that 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.
 - Default value: '28'.
- **STUN retransmission:**
 - A number in the range of 0–9999.
 - Parameter that defines the STUN request retransmit timer (time in milliseconds) in the IAP-specific NAT-FW settings.
 - Default value: '250'.

5 VCC settings

Nokia S60 VoIP Release 3.1 introduces VCC (Voice Call Continuity) settings. In devices that have VCC enabled, these settings can be edited using SIP VoIP Settings for the Nokia S60 VoIP Release 3 plug-in.

To access VCC settings, select **Advanced VoIP settings > VCC settings** (see Figure 66).

If there are no VCC settings available, the plug-in asks to create settings with default values when the VCC settings view is opened (see Figure 66 – Figure 67).

To edit a single setting, select **Options > Change** or hit the middle softkey (see Figure 68 – Figure 69).



Figure 66: Select the VCC settings

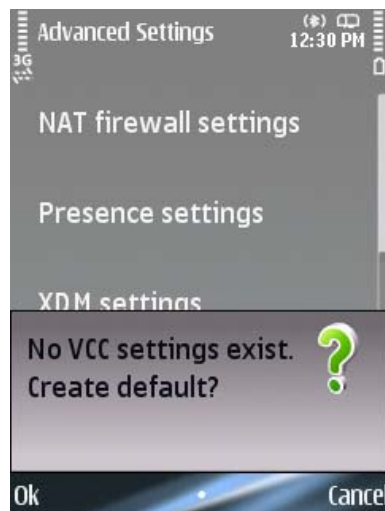


Figure 67: Create new VCC settings



Figure 68: Select editable setting



Figure 69: Edit a setting

5.1 VCC parameters

- VoIP service:
 - Name of the VoIP service. One VoIP service at a time can use VCC settings, and the selected one is shown here.
- VDI:
 - Current VCC transfer URI that is included in the SIP INVITE request to initiate domain transfer.
- VDN:
 - Current VCC transfer number that the user includes in a circuit switched call setup to initiate domain transfer.
- Preferred domain:
 - The operator's preferred domain for UE-originated calls/sessions.
 - '0' = Preference for CS domain.
 - '1' = Preference for IM CN subsystem.
 - '2' = Preference for CS domain only.
 - '3' = Preference for IM CN subsystem only.
 - Default value: '1'.
- Immediate DT:
 - Indicates whether to initiate a VCC domain transfer immediately to the operator's preferred domain when that domain becomes available. This policy only affects ongoing sessions.
 - If the Immediate Domain Transfer operator policy indicates that the domain transfer is not immediately required, then it is up to the VCC UE to decide when to perform the domain transfer if the preferred domain is available.
 - 'On' = Initiate domain transfer immediately to the preferred domain when that domain becomes available.
 - 'Off' = Do not initiate domain transfer immediately to the preferred domain when that domain becomes available.
 - Default value: 'On'.
- CS to PS allowed:
 - Tells whether VCC domain transfer from CS to PS is allowed or not.
 - 'On' = Allowed.
 - 'Off' = Not allowed.
 - Default value: 'On'.
- PS to CS allowed:
 - Tells whether VCC domain transfer from PS to CS is allowed or not.
 - 'On' = Allowed.
 - 'Off' = Not allowed.
 - Default value: 'On'.

- **Waiting calls allowed:**
 - Indicates whether VCC domain transfer is allowed during a waiting call.
 - 'On' = Allowed
 - 'Off' = Not allowed.
 - Default value: 'Off'.
- **WLAN HO threshold:**
 - Indicates the threshold value in dBm for WLAN signal strength from which domain transfer starts.
- **WLAN HO hysteresis:**
 - Defines the hysteresis value in dBm for WLAN signal.
- **WLAN HO hysteresis low:**
 - The amount of time in microseconds WLAN signal must be below WLAN HO threshold for the signal to be interpreted as 'bad'; DT will be committed.
- **WLAN HO hysteresis high:**
 - The amount of time in microseconds WLAN signal must be above WLAN HO threshold for the signal to be interpreted as 'good'; DT will not be committed.
- **CS HO threshold:**
 - Indicates the threshold value in dBm for CS signal strength from which DT starts.
- **CS HO hysteresis:**
 - Hysteresis value in dBm for CS signal.
- **CS HO hysteresis low:**
 - The amount of time in microseconds GSM signal must be below CS HO threshold for the signal to be interpreted as 'bad'; DT will be committed.
- **CS HO hysteresis high:**
 - The amount of time in microseconds GSM signal must be above CS HO threshold for the signal to be interpreted as 'good'; DT will not be committed.
- **Allow CS originated:**
 - Indicates whether domain transfer is allowed when the call is CS originated.
 - 'On' = DT is allowed.
 - 'Off' = DT is not allowed.
 - Default value: 'On'.

6 SIP settings

Below is a list of configurable SIP settings. More information about configuring SIP can be found in [1].

6.1 SIP profile settings

- Profile name:
 - Name of the SIP profile.
- Service profile:
 - A selection between IETF and 3GPP SIP dialects.
 - Default value: 'Internet' (IETF). With 3GPP, use 'IMS'.
- Default destination:
 - Destination network used with this SIP profile. A destination network can contain multiple access points. If used, this setting overrides the default access point setting.
- Default access point:
 - Access point used with this SIP profile. If a default destination is defined, this setting is not in use.
- Public username:
 - A SIP URI for the user including the host-name part, that is, the SIP domain part. For example, sip:john.doe@example.com.
 - To enable registering a SIPS URI, use the prefix 'sips:'. Note that always when registering a SIPS URI, a persistent TLS connection will be created towards the proxy (or registrar if no proxy is defined).
- Use compression:
 - Signalling compression can be used with a cellular radio to reduce the data generated by SIP signalling. It requires an outbound proxy support and may cause error situations in poor WLAN coverage, and is thus not recommended with WLAN.
- Registration mode:
 - Tells the registration mode. 'Always on' is highly recommended with VoIP service because using 'Always on' affects only SIP registration and not VoIP service activation.
- Security negotiation:
 - Requires a sec-agree support from the SIP server side. Used mainly with IMS in 3GPP mode.
 - Default value: 'Inactive'.
- Proxy server:
 - Configuration is needed if an outbound proxy is used.
- Registrar server:
 - Configuration is always needed.

6.2 SIP proxy settings

- Proxy address:
 - Address of the SIP outbound proxy. If outbound proxy is not used, this setting must be left empty.
 - If the proxy server is defined as FQDN, resolving the related IP address will be done with the procedures specified within RFC 3263 [4]. A short summary is as follows:
 - If the FQDN is supplied without port and transport parameters, the terminal will try to resolve it using the DNS NATRP, SRV, and, finally, A and AAAA queries.
 - If the FQDN is supplied without a port parameter but with a transport parameter, the terminal will try to resolve it using the DNS SRV and, finally, A and AAAA queries.
 - If the FQDN is supplied with the port parameter, the terminal will try to resolve it using only A and AAAA queries.
 - Apply value 0.0.0.0 to enable the discovery of SIP proxy using DHCP option 120.
 - This setting creates a preroute set according to RFC 3261 [5]; the Route header is inserted in the initial request.
 - The proxy has to insert Record-Route headers to keep itself in the route set on later requests inside a SIP session.
 - To enable SIP over TLS towards the proxy, use the prefix 'sips:'.
- Username:
 - Used in proxy authentication (not necessarily the same as the public user name). The authentication user name may simply be the user name part of the public SIP URI, but it may also contain the SIP domain or the SIP scheme as a prefix, that is:


```
john.doe
john.doe@example.com
sip:john.doe@example.com
```
 - Depends on the proxy vendor and configuration. This value is only needed if the proxy and registrar require different credentials.
- Password:
 - Needed if proxy authentication is used.
 - This value is only needed if the proxy and registrar require different credentials.
- Loose routing:
 - 'Yes' for RFC 3261-compliant SIP proxy.
 - 'No' for SIP 1.0-specification-compatible strict routing mode.
- Transport protocol:
 - This setting affects all initial requests. A possible transport parameter on the next hop in the Record-Route or Contact overdrives this setting.
 - 'UDP' – UDP transport forced to be used.
 - 'TCP' – TCP transport forced to be used.
 - 'Auto' – Transport selected according to RFC 3261 [5], that is, UDP is used for ≤ 1300 bytes, and TCP for > 1300-byte-long initial requests. The transport is TLS (RFC 3263 [4]) for SIPS.

- Port:
 - TCP and/or UDP port to which the SIP proxy is listening; the default value of 5060 is typically used.

6.3 SIP registrar server settings

- Registrar address:
 - IP address or FQDN of the registrar server on the SIP domain, that is, the host-name part of the user's SIP URI.
 - To enable SIP over TLS towards the registrar, use the prefix 'sips:'.
- User name:
 - Used in user authentication. Often, but not necessarily always, the same as the user name of the public SIP URI. This parameter is often called a private user ID. Normally, the same value is also used for the proxy authentication.
- Password:
 - Used in user authentication. Typically the same value is also used for the proxy authentication.
- Transport protocol:
 - 'UDP' – UDP transport is forced to be used if selected and if an outbound proxy is not defined.
 - 'TCP' – TCP transport used for the REGISTER request sent to the registrar and forces the use of TCP transport for all initial requests in cases where an outbound proxy is not defined.
 - 'Auto' – Transport selected according to RFC 3261 [5], that is, UDP is used for ≤ 1300 bytes, and TCP for > 1300 -byte-long initial requests. The transport may also be TLS (RFC 3263 [4]).
- Port:
 - TCP and/or UDP port to which the SIP registrar is listening; the default value of 5060 is typically used.

7 SIMPLE Presence settings

Below is a list of configurable SIMPLE Presence settings. More information about configuring SIMPLE Presence can be found in [1].

- Server name:
 - Name of the presence server.
- SIP profile:
 - The SIP profile used for the presence settings. The available profiles are presented on a list.
- XDM settings:
 - The XDM settings used for the presence settings. The available XDM settings are presented on a list.
- Object size:
 - The maximum size of the MIME object data in bytes. A value in the range of 0–999999999.
 - Default value of 5120 is recommended.
- Publish interval (sec):
 - The minimum time interval in seconds between two consecutive publications. A value in the range of 0–999999.
 - Default value of 3600 is recommended.
- Maximum subscriptions:
 - The maximum number of presence subscriptions. A value in the range of 0–9999. If this value is left empty, the number of subscriptions is unlimited.
- Max. contacts in list:
 - The maximum number of contacts in the subscription list. A value in the range of 0–9999. If this value is left empty, the number of contacts is unlimited.
- Domain syntax:
 - The syntax of the service URI, for example, @domain.

8 XDM settings

Below is a list of configurable XDM settings. More information about configuring XDM can be found in [1].

- XDM set name:
 - The name of the XDM setting set.
- Access point:
 - Access point to be used with these XDM settings. It is recommended that this be left empty and that the VoIP service be linked to a destination network.
- XCAP service address:
 - URI of the XDM server.
- HTTP user name:
 - Required for the XDM settings authentication.

9 Terms and abbreviations

Term or abbreviation	Meaning
a-law	Name of G.711 PCMU algorithm (European)
AMR	Adaptive Multi Rate
CRLF	Carrier Line Feed
CN	Comfort Noise (in codec settings)
CN	Core Network (in VCC settings)
CRLF	Carrier Line Feed
CS	Circuit Switched
dBm	Power ratio in decibels (dB) of the measured power referenced to one milliwatt (mW).
DHCP	Dynamic Host Configuration Protocol
DiffServ	Differentiated Service
DNS	Domain Name System
DSCP	DiffServ Code Point
DT	Domain Transfer
DTMF	Dual-Tone Multi Frequency
DTX	Discontinuous Transmission
FQDN	Fully Qualified Domain Name
FW	Firewall
GSM	Global System for Mobile communications
GSM-EFR	Global System for Mobile communications – Enhanced Full Rate
HO	Handover
HTTP	Hypertext Transport Protocol
IAP	Internet Access Point
ID	Identity
IEEE	Institute of Electrical and Electronics Engineers (www.ieee.org)
IETF	The Internet Engineering Task Force (www.ietf.org)
IM CN	IP Multimedia Core Network
iLBC	Internet Low Bitrate Codec
IMS	IP Multimedia System
IP	Internet Protocol
IPv4	Internet Protocol, version 4
IPv6	Internet Protocol, version 6
kbit/s	Kilobits per second

Term or abbreviation	Meaning
Maxptime	Maximum amount of media that can be encapsulated in a payload packet
MIME	Multipurpose Internet Mail Extensions
NAT	Network Address Translation / Network Address Translator
NB	Narrow Band
OMA	Open Mobile Alliance (www.openmobilealliance.org)
PCMA	Pulse Code Modulation a-law
PCMU	Pulse Code Modulation μ -law
PDC-EFR	Personal Digital Cellular – Enhanced Full Rate
Ptime	Packetization interval
PHB	Per-Hop forwarding Behaviour
PS	Packet Switched
QoS	Quality of Service
RFC	Request For Comments
RTCP	Real-time Transport Control Protocol
RTP	Real-Time Transport Protocol
SBC	Session Border Controller
SID	Silence Description
SIMPLE	SIP Instant Message and Presence Leveraging Extensions
SIP	Session Initiation Protocol
STUN	Session Traversal Utilities for NAT; a protocol that allows applications to detect that network address translation (NAT) is being used.
tel	Telephony
TC	Traffic Class
TCP	Transport Control Protocol
TLS	Transport Layer Security
TOS	Type of Service
U-APSD	Unsolicited Automatic Power Save Delivery
UDP	User Datagram Protocol
UE	User Equipment
UI	User Interface
URI	Uniform Resource Identifier
VAD	Voice Activation Detection
VCC	Voice Call Continuity
VDI	VCC Domain transfer URI

Term or abbreviation	Meaning
VDN	VCC Domain transfer Number
VoIP	Voice over IP
WB	Wide Band
WLAN	Wireless LAN, Wireless Local Area Network
WMM	Wireless Multimedia
XCAP	XML Configuration Access Protocol
XDM	XML Document Management
μ -law	Name of G.711 PCMU algorithm (North American)

10 References

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