FORUM NOKIA

Nokia S60 VoIP Release 2.0 Configuration Tutorial

Version 1.0; October 27, 2006

VoIP

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Change history

October 27, 2006	Version 1.0	Initial document release

1 Introduction

This tutorial describes the configuration of Nokia S60 Voice over IP (VoIP) Release 2.0.

Nokia S60 VoIP Release 2.0 can be configured with five separate setting groups:

1. VoIP profile settings
   a. Codec settings
   b. SIP profile-specific VoIP settings
2. VoIP generic settings
3. SIP profile settings
4. Access point settings
5. NAT/Firewall traversal settings

The following figure presents the relations between the settings.

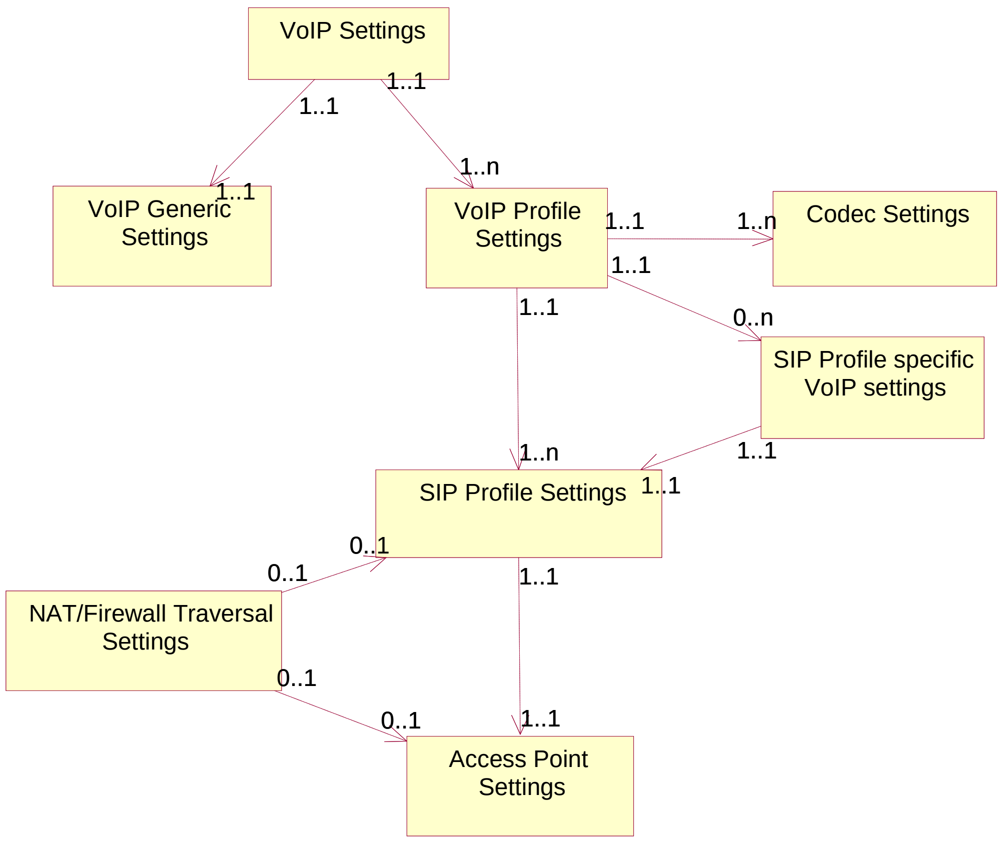

Figure 1: Relations between the VoIP settings

Table 1 explains the arrow numbers displayed in Figure 1.

Item	Meaning
1..1	One setting group
1..n	One or any number of setting groups
0..1	No setting group or one setting group
0..n	No setting group or any number of setting groups

Table 1: Explanation of the arrow numbers

The terminal user interface (UI) has only one group of VoIP terminal settings which affects all VoIP services. For VoIP terminal settings, see Chapter 2.

The VoIP profile includes settings for one or multiple speech codecs. For codec order, see Section 3.1. The VoIP profile may also have SIP profile-specific VoIP settings defined for each SIP profile. VoIP Rel 2.0 supports only SIP protocol.

Each VoIP profile can refer to one or multiple SIP profiles, but each SIP profile can only refer to one Internet Access Point (IAP).

The NAT/Firewall traversal settings can either refer to a SIP domain or access point, or both. For SIP profile settings, see Chapter 4.

Many of these settings cannot be edited from the UI but must be provisioned by the service provider.

Note: For documentation on the supported OTA provisioning, see www.forum.nokia.com. When using OMA Client Provisioning for Nokia S60 VoIP Rel 2.0 configuration, the message encoder must support APPREF and TO-APPREF parameters to make the linking between the setting groups work. The parameters are defined in the Open Mobile Alliance document OMA-DM-2004-0211R01-CR_AC.doc available at http://member.openmobilealliance.org/ftp/Public_documents/DM/2004/OMA-DM-2004-0211R01-CR_AC.zip.

2 VoIP generic settings

Generic or terminal-specific VoIP settings affect all VoIP profiles and the generic behaviour of the S60 telephone applications. Choosing between a cellular and an Internet telephone call, as well as modifying any other terminal-controlled supplementary settings, such as call waiting and anonymous call blocking, are matters of personal user preference. But these settings can also be pre-adjusted at the service provisioning; the latest group of settings provisioned overdrives the previous one.

For terminal-specific settings on the UI, select **Menu > Tools > Settings > Call settings**.

For information on provisioning terminal-specific VoIP settings, see References [2] and [3].

2.1 Terminal-specific VoIP settings

- **Internet call waiting:**
  - '0': Disabled. The called party is not indicated about an incoming call. The calling party receives a busy tone.
  - '1': Enabled. The called party is indicated about a waiting call. The calling party receives a waiting indication and/or the phone continues ringing.
  - Default value: '0'
- **Anonymous call block rule:**
  - The value determines the rule for the 'Anonymous call block' feature.
  - '0': Anonymous calls are received
  - '1': Anonymous calls are rejected
  - Default value: '0'
- **Preferred telephony:**
  - '0': Cellular telephony
  - '1': VoIP telephony (Internet)
  - Default value: '0'
- **VoIP CLIP/CLIR:**
  - '0': CLIR disabled and CLIP enabled. The caller ID is sent.
  - '1': CLIR enabled and CLIP disabled
  - Default value: '0'
- **Do not disturb setting (DND):**
  - '0': The 'Do not disturb' feature is disabled
  - '1': The 'Do not disturb' feature is enabled. The calling party receives a busy tone, but the call event is registered in the log.
  - Default value: '0'

3 VoIP profile settings

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. The following is an example of a VoIP service divided into two different VoIP profiles:

- 'ServiceName Home'
- 'ServiceName City WLAN'

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 from the Internet telephone application.

The VoIP services work in automatic or manual mode:

- In automatic mode, the VoIP service is automatically registered if a network is available. The WLAN network availability is detected when the WLAN scan is triggered (the scanning frequency can be set in the WLAN settings).
- In manual mode, the user connects by an active idle shortcut or from the Internet telephone application. It is also possible to register when making an Internet call from the phonebook logs or idle view.

Changing the registration mode from the Internet telephone application also changes the registration mode of the SIP profiles used.

For information on provisioning VoIP profile settings, see References [1] and [3].

3.1 Profile-specific VoIP settings

- **Id of the VoIP settings:**
  - Settings ID. The ID value is also the priority value of the VoIP profile. 0 = first profile.
- **Provider of the settings:**
  - 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. If the VoIP service has a customer brand name, put it here.
- **Name of the settings:**
  - 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.

- **VoIP codecs in preferred order**
  - The following default order of speech codecs is used when the settings have been created manually:
    i. AMR NB
    ii. PCMU (G.711 μ-law)
    iii. PCMA (G.711 A-law)
    iv. iLBC
    v. G.729
    vi. Comfort Noise for PCMU and PCMA (not a separate codec, but handled as such)
  - Different order can also be defined, or some of the codecs can be entirely disabled, but this is not recommended due to possible problem scenarios. See also Section 3.2, "Speech codec settings."
  - The Comfort Noise is not an actual codec by itself, but is handled as such. This codec enables the usage of generic comfort noise RTP payload format with iLBC, PCMU, and PCMA codecs. The RTP payload format is defined in RFC 3389.
  - If the Comfort Noise codec is disabled from the VoIP profile, the generic comfort noise RTP payload format is neither supported nor used.
- **SIP profiles:**
  - IDs of the SIP profiles used by this VoIP profile. The VoIP profile can refer to one or many SIP profiles.
- **SIP profile-specific VoIP settings IDs:**
  - IDs of the used SIP profile-specific VoIP settings.
- **Start media port number:**
  - A number in range of 1024–65535.
  - The lower limit of the RTP port range.
  - Default value: '16384'
- **End media port number:**
  - A number in range of 1024–65535.
  - 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: '32766'
- **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.
  - Default value: '46'
- **Whether to generate DTMF in-band signaling:**
  - 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.
  - ‘0’: Disabled
  - ‘1’: Enabled
  - Default value: ‘1’

- **Whether to generate DTMF out-band signaling:**
  - 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.
  - ‘0’: DTMF digits out-band are not generated.
  - ‘1’: DTMF digits out-band are generated, if requested by the remote side.
  - Default value: ‘1’

- **VoIP Profile locked to pre-defined IAPs:**
  - ‘0’: Dynamic IAP creation or using other IAPs is allowed for this VoIP profile.
  - ‘1’: VoIP profile can be used from a pre-defined IAP only. If this setting is enabled, the Internet telephone application shows only the pre-defined IAPs of the VoIP service.
  - Default value: ‘0’

- **Allow VoIP over WCDMA:**
  - If this setting is enabled, the Internet telephone application shows also the available WCDMA access points.
  - ‘0’: VoIP over WCDMA is not allowed
  - ‘1’: VoIP over WCDMA is allowed
  - Default value: ‘0’

- **RTCP reporting:**
  - This setting enables the Real-Time Transport Control Protocol (RTCP) reports defined in RFC 3550.
  - ‘0’: RTCP reporting is disabled.
  - ‘1’: RTCP reporting is enabled.
  - Default value: ’0’

- **SIP VoIP User Agent header: terminal type:**
  - ‘0’: The terminal type is not appended to the UA header.
  - ‘1’: The terminal type is appended to the UA header.
  - Default value: ‘1’

- **SIP VoIP User Agent header: WLAN MAC address display:**
  - ‘0’: The MAC address is not appended to the UA header.
  - ‘1’: The MAC address is appended to the UA header.
  - Default value: ‘0’

- **SIP VoIP User Agent 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.
    - Samples of the User Agent header:

Terminal type 1, WLAN MAC 0, no free text (default)	User-Agent: Nokia RM-92 V 4.0632.0.37
Terminal type 1, MAC 1, no free text	User-Agent: Nokia RM-92 V 4.0632.0.37 00-16-bc-7a-14-f6
Terminal type 0, MAC 1 and free text	User-Agent: 00-16-bc-7a-14-f6 MyVoIPconnection
Terminal type 1, MAC 0 and free text (recommended)	User-Agent: Nokia RM-92 V 4.0632.0.37 MyVoIPconnection

3.2 Speech codec settings

3.2.1 AMR NB codec

- **VoIP codec ID:**
    - VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
    - Value: 'audio'
- **Media subtype name:**
    - Value: 'AMR'
- **Jitter buffer size:**
    - A positive integer (milliseconds) in range of 20–200.
    - Default value: '200'
- **octet-align:**
    - '0': Disabled. Bandwidth efficient framing is employed.
    - '1': Enabled. Octet-aligned framing used according to RFC 3267.
    - Default value: '0'
- **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'

- **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.
   - '0': Disabled
   - '1': Enabled
   - Default value: '0'

3.2.2 PCMU (G.711 µ-law) codec

- **VoIP codec ID:**
   - VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
   - Value: 'audio'
- **Media subtype name:**
   - Value: 'PCMU'
- **Jitter buffer size:**
   - A positive integer (milliseconds) in range of 20–200.
   - Default value: '200'
- **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'
- **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.

- ‘0’: Disabled
- ‘1’: Enabled
- Default value: ‘0’

3.2.3 PCMA (G.711 A-law) codec

- **VoIP codec ID:**
  - VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
  - Value: ‘audio’
- **Media subtype name:**
  - Value: ‘PCMA’
- **Jitter buffer size:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: ‘200’
- **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’
- **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.
  - ‘0’: Disabled
  - ‘1’: Enabled
  - Default value: ‘0’

3.2.4 iLBC codec

- **VoIP codec ID:**
  - 0 or a positive integer. VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
  - Value: 'audio'
- **Media subtype name:**
  - Value: 'iLBC'
- **Jitter buffer size:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: '200'
- **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'
- **Voice Activation Detection (VAD):**
  - '0': Disabled
  - '1': Enabled
  - Default value: '0'

3.2.5 G.729 codec

- **VoIP codec ID:**
  - VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
  - Default value: 'audio'
- **Media subtype name:**
  - Default value: 'G729'

- **Jitter buffer size:**
  - A positive integer (milliseconds) in range of 20–200.
  - Default value: ‘200’
- **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’
- **Voice Activation Detection (VAD):**
  - ‘0’: Disabled
  - ‘1’: Enabled
  - Default value: ‘0’
- **Annex b:**
  - A number. Enable enhancement according to IETF RFC 3555 annex-b.
  - ‘1’: Yes
  - ‘0’: No
  - Default value: ‘0’

3.2.6 Comfort Noise codec

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

- **VoIP codec ID:**
  - VoIP codec ID is referred to by the VoIP settings. A generated, positive decimal number.
- **Media type name:**
  - Value: ‘audio’
- **Media subtype name:**
  - Value: ‘CN’

3.3 SIP profile-specific VoIP settings

- **IDs of the used SIP profiles:**
    - A number, 0 or a positive integer.
- **IP Voice Mailbox URI:**
    - A SIP or TEL URI defining the IP voice mailbox address (IETF RFC 3842-compliant server) of the user. For example, *sip:alice@voicemailbox.example.com*.

4 SIP profile settings

Session Initiation Protocol (SIP) profile settings include items such as:

- Public user name (SIP AOR)
- SIP registrar server address and authentication credentials
- Proxy server address and authentication credentials

Multiple S60 client applications can use a shared SIP profile simultaneously, but it is recommended to configure VoIP to use unique SIP profiles, to avoid conflict situations.

As each SIP profile can only have one Internet Access Point (IAP), using many IAPs for VoIP service requires separate SIP profiles. The SIP profiles that do not have a default access point defined should not be configured to 'Always on' mode. The SIP profiles under a single VoIP service should not either be configured to 'Always on' mode for different types of access points that may have overlapping coverage, such as WLAN network AP and WCDMA network AP. Nokia S60 VoIP Rel 2.0 is intended for WLAN use and configuration to 'Always on' mode for different types of access points may produce usability problems. Also, it is not recommended to configure the SIP profiles in 'Always on' mode or other applications to compete for the resources of WLAN networks with overlapping coverage.

Multiple SIP profiles shall not have different user names for the same SIP authentication realm.

For information on provisioning SIP profile settings, see References [4] and [3].

4.1 Creating a SIP profile

To create a SIP profile for VoIP use, the following SIP settings are needed:

- **Profile name:**
  - Give a name to describe the settings, identify this particular set of parameters. Using the same name as for the VoIP profile settings is recommended.
- **Service profile:**
  - A selection between IETF and 3GPP SIP dialects.
  - Default value: 'IETF' (use 3GPP with IMS)
- **Default access point:**
  - WLAN access point should be used, select the one created for VoIP use in provisioning.
- **Public user name:**
  - A SIP URI for the user including the hostname part, that is, the SIP domain name provided by the service provider. For example, *sip:john.doe@example.com* or *sip:+12345678@john.doe@example.com*.
- **Use compression:**
  - Signalling compression can be used with a cellular radio to reduce the data generated by the SIP signalling, requires an outbound proxy support, may cause error situations in poor WLAN coverage, and is thus not recommended with WLAN.
  - Default value: 'No'
- **Registration:**
  - 'Always on' registration when the terminal is started and a configured WLAN access point is available.

- ‘When needed’ registration manually via Internet telephone application.

- **Use security:**
  - Requires a sec-agree support from the SIP server side. Used mainly with IMS in 3GPP mode.
  - Default value: ‘No’
- **Proxy server:**
  - Needed if an outbound proxy is used.
- **Registrar server:**
  - Always needed.

4.1.1 Proxy server

- **Proxy server address:**
  - The address of the SIP outbound proxy, leave empty if the outbound proxy is not used.
  - If the proxy server is defined as FQDN, resolving the related IP address will be done with the procedures specified within RFC 3263. 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 NAPTR, SRV, and finally the A and AAAA queries.
    - If the FQDN is supplied without the port but with the transport parameter, the terminal will try to resolve it using the DNS SRV and finally the A and AAAA queries.
    - If the FQDN is supplied with the port parameter, the terminal will try to resolve it using only the A and AAAA queries.
  - This setting creates a pre-route set according to RFC 3261; 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.
- **Realm:**
  - The realm parameter sent by the proxy in the authentication challenge in the 407 response. The recommended value is the SIP domain. The value must be exactly the same as the proxy is configured to use, case sensitive.
- **User name:**
  - Needed for 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.
- **Password:**
  - Needed if proxy authentication is used (provided by the service provider).
- **Allow loose routing:**
  - ‘Yes’ for RFC 3261-compliant SIP proxy

- ‘No’ for SIP 1.0 specification-compatible strict routing mode

- **Transport type:**
  - 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. Set if the proxy does not support TCP.
  - ‘TCP’ - TCP transport forced to be used. Set if a persistent TCP is used for NAT traversal.
  - ‘Auto’ - transport selected according to RFC 3261, that is, UDP is used for ≤1300 bytes, and TCP for >1300 bytes long initial requests.
- **Port:**
  - TCP and/or UDP port the SIP proxy is listening to, the default value of 5060 is typically used.

4.1.2 Registrar server

- **Registrar server address:**
  - FQDN of the registrar server or the SIP domain, that is, the hostname part of the user’s SIP URI.
- **Realm:**
  - The realm parameter sent by the registrar in the authentication challenge in the 401 response. The recommended value is the SIP domain. The value must be exactly the same as the registrar is configured to use, case sensitive.
- **User name:**
  - Needed for user authentication. Often, but not necessarily always the same as the public user name. This parameter is often called a private user ID. Normally, the same value as for the proxy authentication, see the proxy section above.
- **Password:**
  - Needed for registrar authentication. Typically the same as for the proxy authentication.
- **Transport type:**
  - ‘UDP’ - UDP transport forced to be used. Set if the registrar does not support TCP or if UDP is always to be used in case an outbound proxy is not defined.
  - ‘TCP’ - TCP transport used for the REGISTER request sent to the registrar and forces to use TCP transport for all initial requests in case an outbound proxy is not defined.
  - ‘Auto’ - transport selected according to RFC 3261, that is, UDP is used for <1300 bytes, and TCP for >1300 bytes long initial requests.
  - This setting is used only if no proxy has been defined. In that case it affects all initial SIP requests.
- **Port:**
  - TCP and/or UDP port the SIP registrar is listening to, the default value of 5060 is typically used.

5 Access point and WLAN settings

By default, the Internet telephone application  shows only WLAN access points for VoIP services, but it is possible to use any type of access point for VoIP services if it is configured to the SIP profile by provisioning, or manually from the SIP settings. Two provisionable profile-specific VoIP settings define the visibility of available networks or IAPs in the Internet telephone application:

- With 'VoIP Profile locked to pre-defined IAPs' setting the user only sees the pre-configured access points in the Internet telephone application.
- With 'Allow VoIP over WCDMA' setting the user can use any available WCDMA IAP without pre-configuration (the default is 'off').

The Internet telephone application creates an access point if the user selects a WLAN network that has no access point defined. To activate the scanning of WLAN networks, set the WLAN access point setting 'Show WLAN availability' on. This option is switched on automatically, if the VoIP provisioning message contains at least one 'Always on' type SIP profile that uses a WLAN access point. The user can also activate the WLAN scanning from the SIP settings or the Internet telephone application.

Note that hidden WLAN networks are not scanned if the terminal is in 'Offline' mode.

To create access points manually, select **Menu > Tools > Settings > Connection > Access points**.

6 NAT/Firewall traversal settings

Nokia S60 VoIP Release 2.0 has STUN protocol support for NAT traversal and NAT binding refresh features. The NAT/Firewall traversal features enable the VoIP functionality behind NATs of certain type.

The STUN/firewall settings cannot be edited from the terminal UI, but must be provisioned by the service provider.

If the STUN server address is not configured, the terminal tries to find one with a SIP SRV query using the Public User Identity domain of the used SIP profile.

Additional dummy packet (CRLF) refresh may be enabled by provisioning. The CRLF refresh is also enabled automatically if the STUN server is situated in a different address from the SIP proxy/registrar. The CRLF refresh is also enabled if there is no STUN server configured, and the user's own IP address is in a private address space, since the terminal is then presumably behind a NAT.

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

For information on provisioning NAT/FW settings, see References [5] and [3].

6.1 NAT/Firewall settings

6.1.1 Domain-specific settings

- **Domain:**
  - SIP domain to specify domain-specific NAT/FW traversal values.
- **STUN Server Address:**
  - 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.
- **STUN Server Port:**
  - This parameter defines the STUN server port in the domain-specific NAT-FW settings. Optional.
  - Default value: '3478'
- **NAT Refresh TCP:**
  - 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'
- **NAT Refresh UDP:**
  - 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'

- **Enable 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.

6.1.2 IAP-specific settings

- **Access point ID:**
    - Access Point ID of the Access Point-specific NAT/FW traversal settings. Optional.
- **NAT Refresh TCP:**
    - 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'
- **NAT Refresh UDP:**
    - 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 Retransmit:**
    - This parameter defines the STUN request retransmit timer (time in milliseconds) in the IAP-specific NAT-FW settings. Optional.
    - Default value: '250'

7 VoIP settings not visible from the UI

The following VoIP settings are neither visible nor can they be edited from the terminal user interface.

- Speech codec selection, order, and codec-specific settings
- NAT/Firewall settings
- The following VoIP profile settings:
  - Provider of the settings
  - Start media port number
  - End media port number
  - Media QOS
  - Whether to generate DTMF in-band signalling
  - Whether to generate DTMF out-band signalling
  - VoIP profile locked to pre-defined IAPs
  - Allow VoIP over WCDMA
  - RTCP reporting
  - SIP VoIP User Agent header: terminal type
  - SIP VoIP User Agent header: WLAN MAC address
  - SIP VoIP User Agent header: free string

8 Usage

When the settings have been provisioned or set up manually, the user may still need to set up the access network:

- If the service provider has not provisioned the access points, the user can add them in the Internet telephone application.
- The user can use the Internet telephone application to connect to service via network, if allowed by the profile settings.
- The Internet telephone application creates copies of the VoIP profile's first SIP profile if the user selects a new network unknown to the VoIP service.

The connectivity-related SIP settings can be modified in the Internet telephone application.

Nokia S60 VoIP Release 2.0 terminal may be registered simultaneously to cellular and multiple VoIP services. Mobile originated (MO) calls are made with the following rules:

1. Preferred telephone switch:
   a. Cellular call mode prefers cellular network
   b. Internet call mode prefers VoIP

   The preferred telephone switch first determines the call preference — cellular or Internet. If the user makes an Internet call, the switch in rule no. 2 and the availability of networks determine the MO call.
2. If multiple VoIP services are available, the call is made to the preferred one. The preference is defined by the order of the stored VoIP profiles. The user can change the preferred VoIP service with the **Change service** option in the Internet telephone application.
3. If the user specifically selects the Internet or Voice call from the phonebook or log, the call is made respectively as VoIP or cellular.
4. If the user calls to a SIP URI, the call is made as VoIP.
5. If the user is not registered to a VoIP service and makes a call to a number containing only 0...9, *, and #, the call will be made as cellular, although an Internet call would be the preferred mode.
6. If the user tries to make an Internet call without an active VoIP service registration, the registration is performed if a known VoIP-capable access point is available.

9 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	Caller Line Identification Presentation
CLIR	Caller Line Identification Restriction
CN	Comfort Noise
CP	Client Provisioning
DM	Device Management
DND	Do Not Disturb
DSCP	DiffServ Code Point
DTMF	Dial-Tone Multi Frequency
DTX	Discontinuous Transmission
FW	Firewall
HTTP	Hyper Text Transport Protocol
IAP	Internet Access Point
ID	Identity
IETF	The Internet Engineering Task Force (www.ietf.org)
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 (www.openmobilealliance.org)
PCMA	Pulse Code Modulation a-law
PCMU	Pulse Code Modulation µ-law
Ptime	Packetization interval
PHB	Per-Hop forwarding Behaviour
QoS	Quality of Service
RFC	Request For Comments
RTP	Real-Time Transport Protocol
SBC	Session Border Controller

Term or abbreviation	Meaning
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
TOS	Type of Service
URI	Uniform Resource Identifier
VAD	Voice Activation Detection
VoIP	Voice over IP
WLAN	Wireless LAN, Wireless Local Area Network
μ-law	Name of G.711 PCMU algorithm (North American)

10 References

[1] OMA DM: Device Management Object for Nokia VoIP Release 2.0

[2] OMA DM: Device Management Object for Generic VoIP Settings

[3] Client Provisioning Registration available at www.forum.nokia.com

[4] OMA DM: Device Management Object for SIP

[5] OMA DM: Device Management Object for NATFW

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