OptiMobile UniPhone for Nokia Series 60 devices

Manual 2.2.5

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1. Introduction to UniPhone for Nokia Series 60 devices

OptiMobile UniPhone is a dual mode 3G/GSM and VoIP software client for Nokia Series 60 devices. The user experience is very close to the telephony application already present in your mobile phone. With UniPhone you can dial and receive phone calls, and make phone calls from the built-in phone book and call log.

The difference with UniPhone is that it is a Voice over IP (VoIP) client using the common SIP standard. This allows UniPhone to be used for communicating with desktops, laptops, PDAs and standalone IP-phones. You can also reach any traditional wired or mobile phone number available through a SIP-PSTN gateway, which typically is available via a VoIP operator or a corporate IP-PBX.

UniPhone together with a VoIP subscription from an operator converts your mobile phone into a perfect replacement for DECT phones. As a bonus you can close down your fixed line subscription. UniPhone is also ideal for enterprises that have an IP-PBX. With UniPhone enhanced mobile phones there will be no need for fixed IP desk phones and the office will be completely mobile.

UniPhone is a full featured SIP client enabling VoIP telephony towards SIP infrastructure, either pure peer-to-peer VoIP or VoIP to PSTN calls. By default UniPhone uses VoIP line for calling within Wi-Fi coverage and circuit switched 3G/GSM for calling out of Wi-Fi coverage. IP/Internet connection is a necessity to use UniPhone and it is mainly delivered via wireless local area networks (Wi-Fi) but also on 3G cellular network. GPRS may or may not work and is not supported in any way.

Note*: Running VoIP over 3G cellular networks may induce heavy data transfers that may lead to substantial bills from your cellular operator, please consult your billing plan before using VoIP over cellular networks. VoIP may also use additional battery/power resources in your device.*

VoIP calls over the Internet are subjected to a multitude of factors that affect the voice quality. In managed environments, the voice quality offered by VoIP is better than that with a fixed or a mobile phone. When using the unmanaged Internet for voice calls, the quality is typically very good but quality may occasionally fluctuate substantially.

Please note that all screen dumps in this manual are from Nokia 5800 and they may differ slightly from other Nokia Series 60 phones.

1.1 OptiMobile UniPhone features

- SIP compatibility allows you to:
  - communicate with other SIP compatible VoIP applications and systems.
  - call a regular phone using a SIP-PSTN gateway typically through a VoIP operator or corporate IP-PBX.
  - call a computer or IP-phone in a private network through a SIP proxy.
  - handle multiple calls: put call on hold and switch between two active calls.
- Support for sending DTMF signals.
- Configurable number handling with support for a wide variation of systems, from generic operators to more advanced enterprise PBX systems. The number handling settings are:
  - ***Country/Region code***
  - ***Convert plus to***
  - ***National prefix***
  - ***External line prefix***

    - ***Max internal num len***
- Selectable preference of early media to support custom streamed ring tones and other feedback tones with early media.
- DTMF modes: **RTP out-of-band** and **SIP INFO**.
- Selectable autostart of UniPhone at phone start-up.
- Several VoIP profiles with per profile selectable settings.
- AP handling with dynamic VoIP profile attachments.
- Integration of platform phone **Contacts**, **Calls Log** and **Home screen dialler** makes the phone easy to use.
- 3G/GSM compatibility allows you to use 3G/GSM networks for dialling when *3G/GSM Line used*. Within Wi-Fi coverage VoIP line is used for dialling, otherwise the circuit switched 3G/GSM line is automatically selected.
- Support for DTMF based and operator based routing of outgoing 3G/GSM calls, for call routing through e.g. a corporate PBX.
- Handling emergency calls using cellular 3G/GSM network.
- Seamless handover of active calls from VoIP to 3G/GSM and inversely using *OptiMobile VCC Enabled Services*.
- Supported languages: UK English, Svenska.

1.2 System requirements

OptiMobile UniPhone application is compatible with Nokia phones running S60 3rd Edition Feature Pack 1 based on Symbian OS v9.2, phones running S60 3rd Edition Feature Pack 2 based on Symbian OS v9.3 and phones running S60 5th Edition based on Symbian OS v9.4.

1.3 Installation

Install the OptiMobile UniPhone application following the device manual on how to install applications. UniPhone application is bundled with Qt smart installer. If Qt is not present on the device, the smart installer downloads and installs Qt (on user acceptance to use a network connection) before installing the application. Ensure that your phone is prepared with configuration for IP connectivity over either Wi-Fi or on a cellular 3G network. Please consult your device manual on how to configure this. Thereafter start UniPhone from **Menu->Applications->UniPhone**.

2. Setup

The UniPhone application can have several VoIP profiles but only one can be used at a time. By default there is one empty profile called *UniPhone*.

UniPhone Settings is a separate application that can be left running without problems, if one desires to exit it anyway it can be performed through *UniPhone Settings* **Exit** option or through device task manager.

The default settings mentioned in this section may differ depending on your UniPhone distribution channel.

2.1 VoIP profiles

The UniPhone application can handle several VoIP profiles that can be manually or automatically selected. This is useful in many cases, e.g. when the user has both private and office SIP accounts and needs to switch between them regularly. Each profile has three tabs: **SIP** for basic SIP settings, **VCC** for VCC handover settings and **Adv** for less common SIP settings and other Advanced settings that are unique for this profile.

- **Add new profile**

  To add another VoIP profile, select **Options->New profile** in the settings menu and give your profile a name.

- **Edit profile**

  Select a profile from the VoIP profile list and select **Edit** submenu. Then insert the desired settings and select **Back**.

- **Select which profile to use**

  Select this profile in the VoIP profile list and select **Use** submenu.

- **Delete profile**

  To delete a profile, select it from the VoIP profile list and select **Delete** submenu.

- **Rename profile**

  To rename a profile select it from the VoIP profile list and select **Rename** submenu.

- **Use SIP proxy address as profile name**

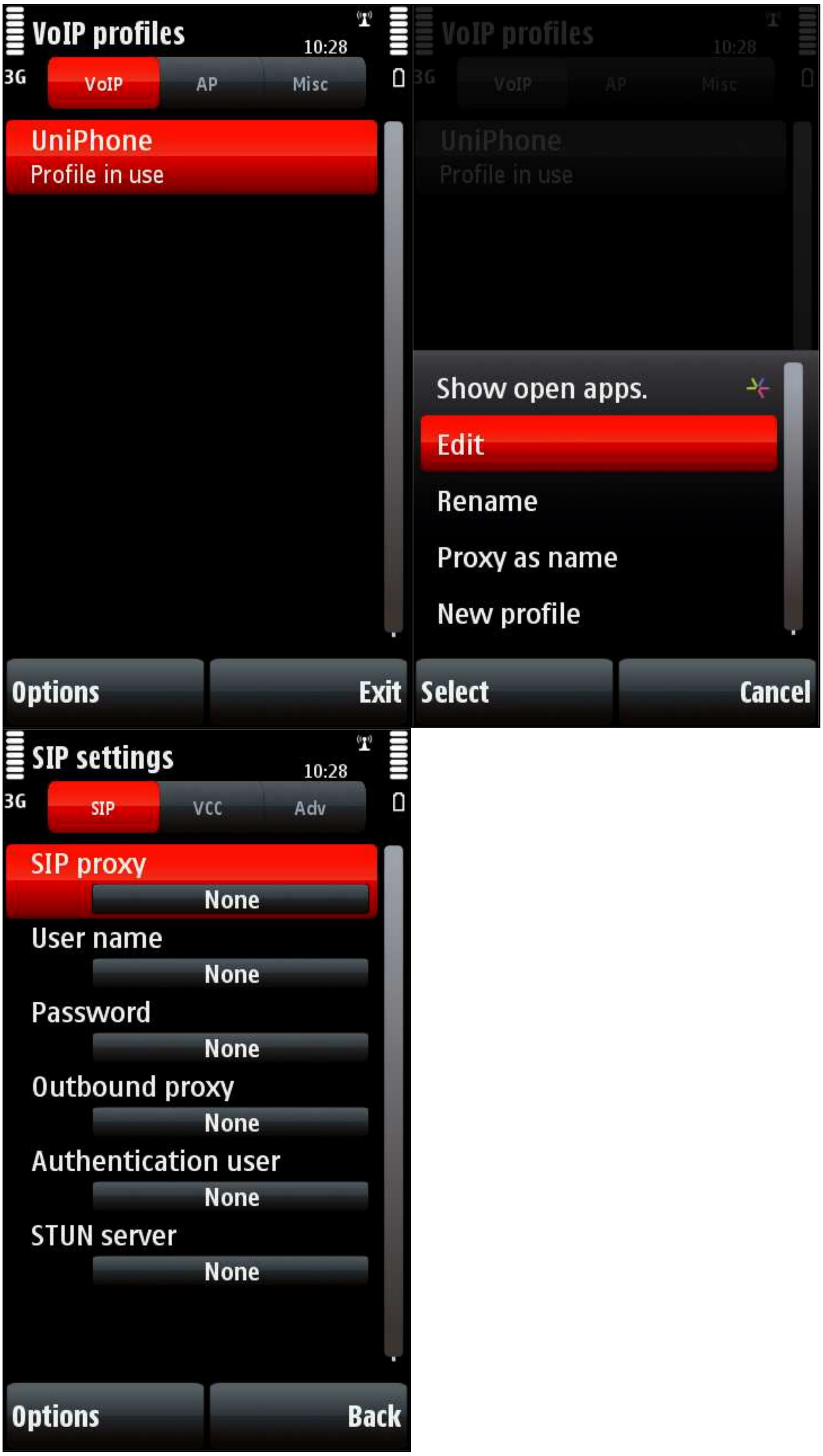
  To rename a profile to the defined ***SIP proxy***, select it from the VoIP profile list and select **Proxy as name** submenu.

2.1.1 SIP settings

The UniPhone application is a SIP (Session Initiation Protocol) client. To be able to call and receive VoIP calls you must have a SIP account from a VoIP service provider. When the SIP settings are configured and a Wi-Fi connection is active, the application registers itself to the SIP proxy.

Generally, SIP settings point your client to a SIP proxy server that keeps track of other SIP clients and enables them to find each other to communicate directly or to connect to the fixed and mobile telephony networks. SIP settings are commonly delivered to you by a VoIP operator or system administrator of a corporate IP-PBX or other SIP proxy server.

To configure **SIP** tab settings: start UniPhone application and select **Options->Settings**, then select a VoIP profile and select **Edit** submenu.



Fill your SIP account information which consists of several parameters:

- ***SIP proxy*** `<host[:port]>`

  This is the SIP proxy that UniPhone registers towards. It is usually a host name such as `sip.myoperator.com` but can also be an IP address such as `10.11.12.13`.

In some cases an optional port number is supplied, if not the standard 5060 port is to be used. It is then directly appended after a `:` sign and for port `55060` it would look like `sip.testproxy.com:55060`.

- ***User name*** `<user>`

  This is the unique identifier for an account on the SIP proxy. It can be a plain name such as `martin`, but also a number string such as `08123456`.

- ***Password*** `<password>`

  The secret password string, please ensure to keep this private.

- ***Outbound proxy*** `<host[:port]>`

  An outbound proxy is sometimes used and is then supplied to be used in the configuration. The syntax is the same as for SIP proxy.

- ***Authentication user*** `<[user]>`

  In some systems the authentication user differs from user name. If ***Authentication user*** and ***User name*** are the same then this field can be left blank.

- ***STUN server*** `<[host]>`

  Some operators offer an STUN server to assist NAT traversal. If one is supplied then enter it here. See section 4.2 Networking issues for more information.

When the SIP settings are configured select **Back** to save changes. If a Wi-Fi connection is active, the UniPhone application registers itself to the SIP proxy or registrar and when status is *VoIP Line used* then UniPhone is ready to make VoIP calls.

2.1.2 VCC settings

In case you have a subscription that includes *OptiMobile VCC Enabled Services* you will be able to enjoy automatic and seamless handover of ongoing voice calls between VoIP and 3G/GSM.

To edit handover settings for each profile, select **Edit** to edit a profile and then select **VCC** tab.

Note*: The **VCC service level** setting affects UniPhone behaviour even when **Use VCC services** is disabled. If VoIP line is regularly disabled even though you have Wi-Fi connectivity, try to select a more restrictive mode, or disable this feature by selecting **Manual only** mode.*

Description of the fields:

- ***Use VCC services*** `<enable|disable>`

  Enable ***Use VCC services*** if you have an *OptiMobile VCC Enabled Services* account.

- ***VCC type***

  Select voice call continuity service type:

  - **DMS VCC** is the corporate IP-PBX type of *OptiMobile VCC Enabled Services*.
  - **OM VCC** is the IMS operator type of *OptiMobile VCC Enabled Services*.
- ***VCC number*** `<[number]>`

  Optionally enter the caller number of the VCC server during handover to 3G/GSM. If ***VCC number*** is left empty, it will be automatically set after the first handover. The first handover is done manually by selecting **Options->Handover to 3G/GSM** during an active VoIP call. You might have to answer the first incoming handover call to 3G/GSM manually, since UniPhone is not automatically answering the handover call, if the incoming caller ID differs from the value in ***VCC number*** field.

- ***VCC service level***

  Select voice call continuity service level:

  - **Seamless** is the highest service level mode that performs handover in order to continuously maximize voice quality and end-user impression. Its goal is to provide uninterrupted voice conversations at all times.
  - **Proactive** is a medium service level mode that performs handover when deemed necessary, with risk of short gaps in voice conversations during handover.

- **Restrictive** is a more restrictive mode that performs handover only when absolutely necessary. Voice conversations will mostly experience gaps of a few seconds during handover, but the voice session is maintained over networks and continues after handover.
- **Manual only** means that no automatic handovers are performed. Manual handover can only be performed while in Wi-Fi coverage, allowing user to choose between the 3G/GSM and Wi-Fi as it suits him.

In radio frequency, fading is caused by changes in the environment between the transmitter and the receiver. For example, a person might walk between you and the AP, you might accidentally put your hand over the Wi-Fi antenna of your phone, you might turn in your chair so that you are between the antenna and the access point, and so on. The environment between a transmitter (an access point) and a client is complex and constantly changing. Radio frequency interactions like reflection, refraction and scattering mean that in indoor environments, there is never a simple linear fading of Wi-Fi quality between a transmitter and a receiver.

Therefore, the signal strength that a client perceives can fade unpredictably, even when the client is sitting still and then a handover may still happen automatically in all service levels, except in the **Manual only** mode, to ensure a continuous voice session at some level. This can happen because VoIP is a real time service and has higher requirements on uninterrupted data service than many other services that can be without data transmission for several seconds without the end-user notices this.

VoIP line is automatically disabled when the Wi-Fi connection becomes bad in all service levels, except in the **Manual only** mode.

2.1.3 Advanced settings

To edit Advanced settings for each profile, select **Edit** to edit a profile and then select the **Adv** tab. In Advanced settings less common SIP settings are handled together with other settings unique for that profile. The appendix B. OptiMobile UniPhone number handling presents several different configuration samples.

- ***DTMF mode***

  In SIP VoIP calls DTMF tones can be transferred in two different ways: **RTP out-of-band** or **SIP INFO**.

- ***Country/Region code*** `<[number]>`

  Sets your country calling code so the UniPhone application knows this internally in order to be able to distinguish between home and foreign country phone numbers. Default is `46` for Sweden.

- ***Convert plus to*** `<[number]>`

  Sets the digits that the plus sign are replaced with on outgoing VoIP calls. Empty setting means that plus sign is not replaced. Default is `00`.

- ***National prefix*** `<[number]>`

  Sets the national prefix which is used to replace **+** and the ***Country/Region code***, e.g. replace **`+46`** with **`0`** in Sweden. Default is `0`.

- ***External line prefix*** `<[number]>`

  This optional setting is used when VoIP calls are routed through a PBX that requires a prefix, e.g. `00`, for outbound calls. This is common in enterprise IP-PBX environments.

- ***Max internal num len*** `<0:20>`

  When number of digits are larger than maximum internal number length, the ***External line prefix*** is automatically added to the dialled number for VoIP calls, this is common in enterprise IP-PBX environments. When ***Max internal num len*** is 0, then the ***External line prefix*** is never added to the dialled number.

- ***Use operator routing*** <enable|disable>

  Enable ***Use operator routing*** if your 3G/GSM operator routes calls through e.g. a corporate PBX.

- ***Use DTMF routing*** <enable|disable>

  Enable ***Use DTMF routing*** if DTMF based 3G/GSM call routing is wanted and supported by e.g. a corporate PBX.

- ***PBX access number*** <[number]>

  If DTMF based routing is used, this field must be set to the PBX access number. If the ***PBX access number*** shall begin with +, it can be entered pressing **1** key several times in a row until the + symbol is displayed.

- ***End number with*** <#|A|B|C|D>

  When DTMF based routing is used, the destination number sent by DTMF tones and ended with selected tone. Default is **#**.

- ***SIP register timeout*** <number of seconds>

  Sets timeout in seconds before a re-registration towards SIP proxy is performed. Default is 600 seconds. Note that the SIP proxy still decides if it accepts this value or not, it may return another value to use and then UniPhone will use that.

- ***Prefer early media*** <enable|disable>

  Enable ***Prefer early media*** to set the preference of playing the early media stream, before the dialled peer has answered the call, over an internally generated ringing tone. Early media refers to the audio that is received before an outgoing call is connected, i.e. the ringing tone. This option is enabled by default, even though not all SIP systems send early media. If you have problems with the ringing tones on your system, try disabling this option.

- ***Use SIP display name***

  Select display name mode for incoming VoIP calls. **No** means: never display SIP display info, but display local Contacts name, if available. **Yes (prefer contact)** means: display local Contacts name, if available; otherwise display SIP display info. **Yes** means: display SIP display info, if not equal to remote number; otherwise display local Contacts name, if available.

- ***Audio codec***

  Lock all VoIP calls to use the specified audio codec. **Auto** means that all available audio codecs can be used. Please note that all operators do not support all audio codecs, hence changing this setting to an unsupported audio codec might result in that it is not possible to make calls.

2.2 AP settings

In the **AP** tab a list of all the device Internet *Access Points* are shown. At least one working Access Point (AP) has to be defined before using UniPhone application for VoIP calls. For more information on how to manage the phone Internet connectivity and access points priorities, please see the phone manual.

In UniPhone application each AP can have one of three modes:

- **Use any profile for an AP**

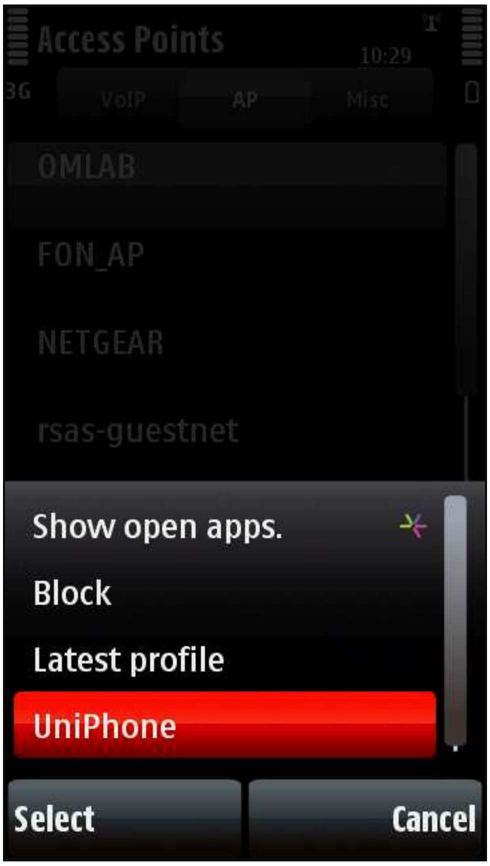
  To let a AP use any VoIP profile, select the AP and select **Options->Latest profile**. When connecting to this AP the latest used profile will then be used. This is the default mode for all defined Wi-Fi access points.

- **Block an AP for VoIP**

  To ensure that an AP will never be used for a VoIP connection by UniPhone, select the AP and select **Options->Block**. This is the default mode for all 3G/2G data bearers; they are blocked and need to be unblocked manually in order to be able to be used for VoIP.

- **Attach an AP to a specific VoIP profile**

  In order to attach an AP to a specific VoIP profile, select the AP and then **Options** and any of the listed VoIP profiles available. UniPhone then automatically selects this VoIP profile when connecting to the AP. For example, UniPhone could be configured to use the office VoIP profile when the phone is connected to the corporate Wi-Fi at the office, and to use the private VoIP profile when the phone is connected to Wi-Fi at home.

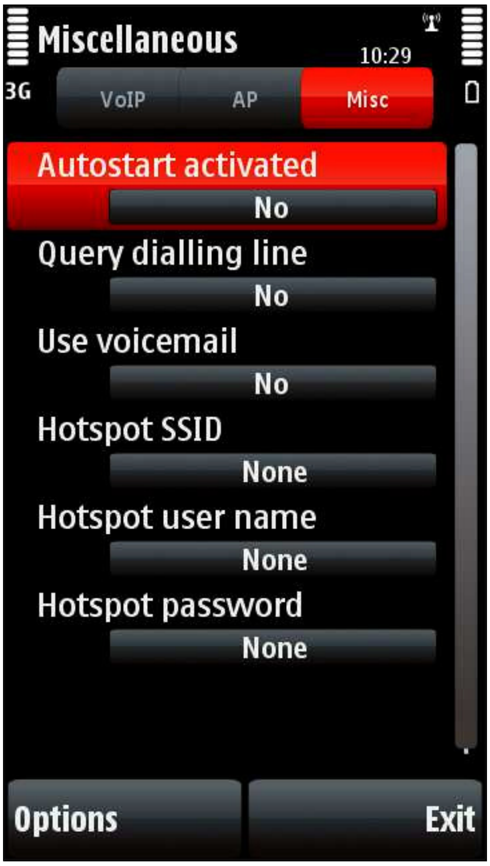


Attaching the AP *OMLAB* to the VoIP profile *UniPhone*.

***Note**: The **AP** tab settings are only used when connecting to an AP. Changes in the AP tab will not take effect until a new connection is set up. This means that changes made to the AP in use will not take effect until a reconnection to that AP is made.*

2.3 Miscellaneous settings

In UniPhone settings, select **Misc** tab to edit Miscellaneous settings.



- ***Autostart activated*** `<enable|disable>`

  Enable ***Autostart activated*** to automatically start UniPhone when the phone is started.

- ***Query dialling line*** `<enable|disable>`

  Enable ***Query dialling line*** to when dialling, and when both VoIP and 3G/GSM lines are available, always query which of the lines that should be used. When this setting is disabled then VoIP line is preferred automatically without prompting the user.

- ***Use voicemail*** `<enable|disable>`

  Enable ***Use voicemail*** to use 3G/GSM line when dialling voicemail number. If the voicemail number is not set in device then the device will prompt about the number when enabling this setting.

- ***Hotspot SSID*** `<[SSID]>`

  This is the Wi-Fi hotspot SSID of your Wireless Internet Service Provider (WISP) that are Wi-Fi Alliance/WISP Roaming (WISPr) compliant, such as Telia Surfzon/Homerun. If you enter the ***Hotspot SSID***, e.g. `telia` or `homerun`, and the WISPr credentials (***Hotspot user name*** and ***Hotspot password***) then UniPhone will automatically login to the Wi-Fi hotspot when the device moves into Wi-Fi coverage with SSID that matches the entered ***Hotspot SSID***.

- ***Hotspot user name*** `<[user]>`

  This is the unique identifier for an account for Wi-Fi hotspot access from the service provider.

- ***Hotspot password*** `<[password]>`

  The secret password string, please ensure to keep this private.

3. UniPhone

3.1 Making a call

With the UniPhone application you can make calls from the **UniPhone dialler**, and the devices **Home screen dialler**, **Contacts** and **Calls Log**.

Call from Home screen dialler or UniPhone dialler:

The most direct method is by entering the number using the keypad from the devices **Home screen dialler**, when UniPhone is running in background, or from **UniPhone dialler** when UniPhone application is displayed in foreground.

***Note**: If the dialled number contains * or #, and is not a 3G/GSM operator service request, then **UniPhone dialler** has to be used.*

- Call from **Home screen dialler**: Enter the phone number and press the **GREEN** key.

  UniPhone displays a status icon on the top of the **Home screen**. GREEN icon means that VoIP line is available, YELLOW icon means that only 3G/GSM line is available and RED icon means that none of the lines are available.

  Some Nokia S60 5th Edition devices, such as Nokia N97 supports **Home screen** widgets. By adding UniPhone widget to **Home screen** it is possible to see actual line status in widget.

- Call from **UniPhone dialler**: Enter the phone number and press the **GREEN** key or select the **Call** button.

Call from Contacts:

The UniPhone application uses the same **Contacts** application as **Home screen dialler**.

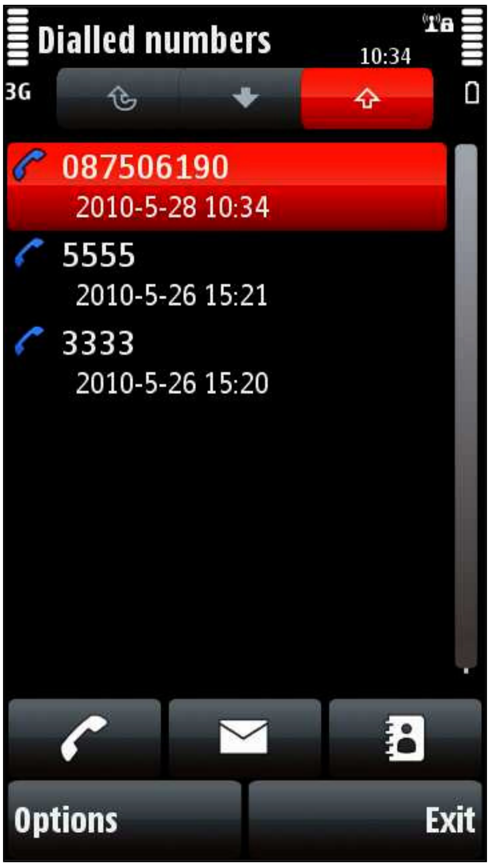
1. Activate **Contacts** application from UniPhone by selecting **Contacts** button or **Options->Contacts**.
2. Find the contact, select it and press the **GREEN** key.

Call speed contacts:

UniPhone uses the same speed contacts as **Home screen dialler**. To make a call via speed contacts from **UniPhone dialler** enter the speed number, 1-9, and then press the **GREEN** key or select the **Call** button.

Call from Calls Log:

1. Activate **Calls Log** application from UniPhone by pressing the **GREEN** key, with no number entered in **UniPhone dialler**, or selecting **Calls** button or **Options->Recent calls**.
2. Select to open *missed*, *received* or *dialled* calls view. Select the call you want to use and press the **GREEN** key.



3.2 Answering a call

You need to have UniPhone running and connected to a proxy to answer an incoming call.

When UniPhone receives a call you will have the same alert notification you have selected for cellular 3G/GSM calls, typically some ring signal and/or vibration. To answer the call just press the **GREEN** key or select **Options->Answer** or **Answer** button. To reject the call press the **RED** key or select **Options->Ignore** or **Ignore** button.

In order to stop call alert without answering or rejecting an incoming call, select **Silence** button.

3.3 Ending a call

Press the **RED** key or select the **End call** button to terminate the call.

UniPhone goes automatically to background when the call is ended.

3.4 Mid call handling

In order to mute microphone during an active call, select **Mute** button or **Options->Mute**. Unmute microphone by selecting **Unmute** button or **Options->Unmute**.

To enable speaker phone during an active call select **Speaker** button or **Options->Activate loudspeaker**. Speaker phone is turned off by selecting **Handset** button or **Options->Activate handset**.

3.4.1 Holding a call

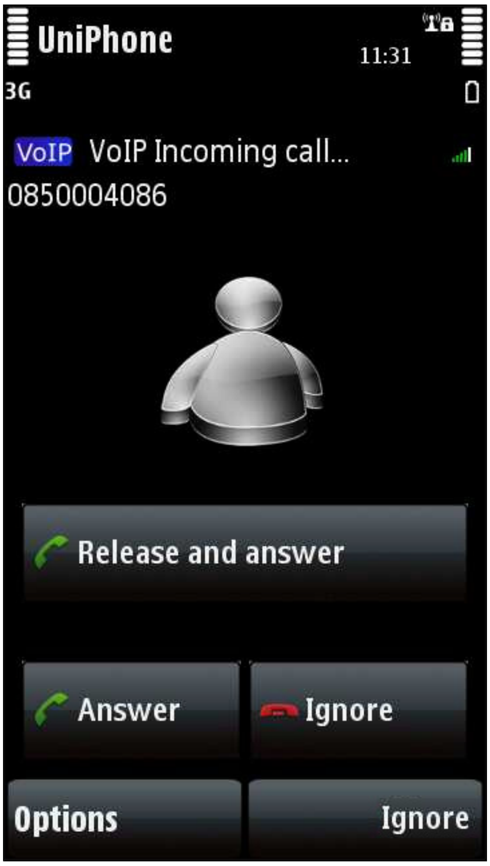
Select **Options->Hold** to put a call on hold.

Now it is possible to issue a new call by dialling it.

Retrieve the previously on-hold call by selecting **Options->Retrieve** or **Retrieve** button.

3.4.2 Incoming call during active call

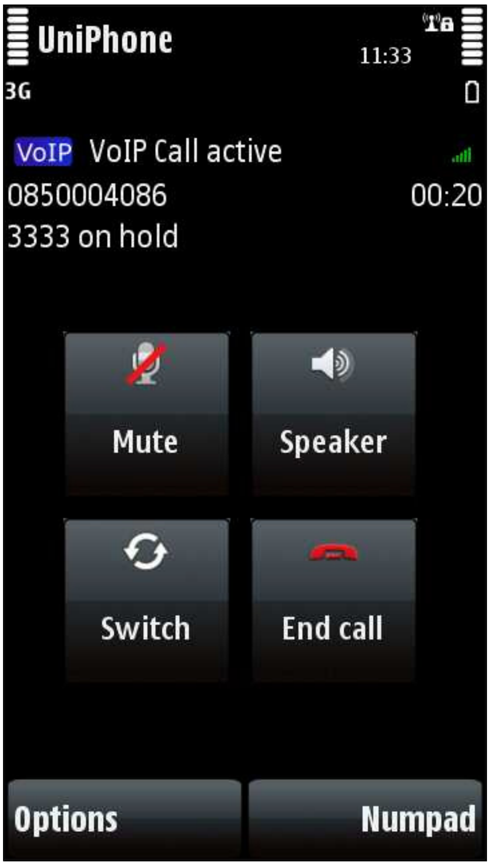
During ongoing active VoIP call and when another incoming VoIP call is received users will have the options of:



- **Release and answer** button or **Options->Release and answer** selection: ends the active call and answers the incoming call.
- **GREEN** key or **Answer** button: puts the active call on hold and answers the incoming call.
- **RED** key or **Options->Answer** or **Ignore** button: rejects the incoming call, while the active call continues as normal.

3.4.3 Incoming call during multiple active calls

If two VoIP calls already are in progress, one active and one on-hold then the user can swap between these two calls by pressing **GREEN** key or selecting **Options->Switch** or **Switch** button.



When two VoIP calls already are in progress, then the following incoming VoIP calls will be automatically rejected with busy signal.

3.4.4 Incoming VoIP call during active cellular call

When a VoIP call is received during active or pending cellular 3G/GSM call, or another pending VoIP call, then the VoIP call is rejected. When all lines becomes idle, the number of missed calls is displayed in a notification bubble.

3.4.5 Incoming cellular call during active VoIP call

When a cellular 3G/GSM call is received, there are three options:

- **GREEN** key: ends all VoIP calls and answers the cellular 3G/GSM call.
- **RED** key: rejects the 3G/GSM cellular call, while VoIP calls continue unaffected.
- **Silence** button: stops call alert, while VoIP calls continue unaffected. After the muting it is still possible to press the **GREEN** or **RED** key.

When a cellular 3G/GSM call is received during pending VoIP call, then the 3G/GSM call is rejected. When all lines becomes idle, the number of missed calls is displayed in a notification bubble.

3.4.6 Transferring a call

Select **Options->Transfer call** to transfer an active call.

Enter the 3rd party destination number or a speed contact number in the displayed dialog and transfer the active call by selecting the **Call** button.

3.5 Enable and disable VoIP

UniPhone automatically registers to the configured SIP proxy, if it has or when it gets an IP address on the Wi-Fi interface. It also switches to *3G/GSM Line used* or *Not ready* state, if the device loses its IP address, i.e. its network connection, or if Wi-Fi radio signal is bad and ***Use auto disabler*** is configured. connection. Note that UniPhone switches to unregistered state immediately, but that the SIP proxy/registrar will not switch to unregistered state before the last registration times out, since it cannot be updated if the network is down.

It is also possible to force register or unregister depending on actual state.

- To register, select **Options->Connect VoIP**.
- To unregister, select **Options->Disconnect VoIP**.

3.6 Additional dialling information

3.6.1 Missed calls

When the user has missed incoming VoIP calls while idle or active, a notification bubble is displayed to notify the user. The notification bubble shows the number of missed calls and is displayed when the VoIP line is idle.

3.6.2 Emergency calls

For every call made from UniPhone it checks whether the dialled number is an emergency number or not. If it is, then UniPhone will make a cellular 3G/GSM network emergency call instead of establishing a VoIP call.

Existing 3G/GSM network services for emergency calls are currently preferred over VoIP for several reasons. One reason is location services - VoIP services sometimes don't show exactly where the emergency call physically is made from.

3.6.3 Bluetooth headset support

A Bluetooth headset might work with UniPhone but it is not possible to answer and hang up the call via the Bluetooth device. When a Bluetooth headset is connected then the VoIP sound is automatically routed though it.

3.6.4 Roaming support

Currently OptiMobile UniPhone number handling algorithms have some limitations using DTMF or operator supported routing when *3G/GSM Line used*. Thus dialling when roaming abroad, i.e. outside home country, is not supported and the user is recommended to exit the UniPhone application and make outgoing calls without UniPhone running in background.

Being aware of this limitation it is still possible to make VoIP calls when *VoIP Line used*, when staying in Wi-Fi coverage during the call. Also see configuration samples in appendix B. OptiMobile UniPhone number handling.

3.7 Making handover

This section is relevant in case you have a subscription that includes *OptiMobile VCC Enabled Services*.

When you have an active VoIP call and are about to leave the wireless area, without wanting to end the call, you can manually switch to 3G/GSM by selecting **Options->Handover to 3G/GSM**. The call will then be seamlessly switched to 3G/GSM. If there is a **Options->Handover to VoIP** option while you have an 3G/GSM call active, then it is possible to manually switch the call back to VoIP, in the same manner.

Note*: When making handover from 3G/GSM to VoIP on Nokia phones running S60 3rd Edition Feature Pack 1 or S60 5th Edition, there is an audible tone played.*

4. Troubleshooting

The following is a guide to troubleshooting various issues that may arise when using the UniPhone application. You may find that the problem you are experiencing is explained within.

4.1 Registration errors

If VoIP registration fails it might be solved like this:

- You may have entered your ***User name*** or ***Password*** incorrectly when configuring the **SIP** settings. Double-check these values and your account status with your VoIP service provider.
- This may also occur due to network problems. Check the network connection of your device, for example by browsing the Internet.
- Check that other SIP software is not running, and blocking the local SIP port.
- Also make sure that the Internet Access Point (AP) you are connected to now is allowed to use that VoIP profile.

4.2 Networking issues

This guide assumes that your Internet connection is working normally. If this is not the case, please contact your Internet service provider.

Symptoms that you have problems with firewall or NAT traversal:

- Audio goes only one way, either inbound or outbound.
- Incoming calls cannot reach you.

If your device is behind a firewall, certain ports must be open for UniPhone to be able to communicate with your VoIP service provider. UniPhone uses UDP ports 5060-5160 for SIP and UDP ports 5700-5800 for RTP. Some firewalls, with support for SIP, can open the currently used media ports in real time.

If your device is behind a NAT, i.e. is on a LAN and has a private IP address, which is very common, there are two options: use a SIP proxy with support for Symmetric Response Routing (SIP RFC 3581), or use STUN. STUN is a service provider independent way of finding out the external IP and port of your NAT, and makes the NAT transparent for the SIP communications. Unfortunately there are some limitations to STUN, most notably the inability to use it behind a *Symmetric NAT*. If your VoIP service provider does not specifically state that STUN must be used, please test your account without STUN and make sure both incoming and outgoing calls work flawlessly. If there is a problem with the setup, try to enable STUN. There are a number of public STUN servers available if your VoIP service provider does not provide one.

4.3 Audio issues

There are several factors that may affect sound quality when using UniPhone. These include the quality of the network connection, network device, and headset. The most sensitive part of the network connection is the connection to the mobile phone over either the cellular 3G network or Wi-Fi that can change rapidly when the user is moving around and which is also affected by other users on the same base station or access point.

- **Poor sound quality**

Poor or scratchy sound quality is a symptom of a problem with your network connection, or network adapter. There may be packet loss on the IP network. Check the network performance, latency, and throughput. A 200 milliseconds round trip latency as determined from a ping should deliver good quality audio. Wi-Fi networks are sensitive and their quality may fluctuate rapidly when moving, try move closer to the Wi-Fi access point or move away from objects such as thick walls.

If poor sound quality is primarily related to when one is using 3G for VoIP and the sound is poor on the uplink, i.e. from UniPhone to the other party, but OK on the downlink, i.e. to UniPhone, then it may have to do with insufficient bandwidth using the default G.711 audio codec. Try to use an audio codec that uses lower bitrate, e.g. G.729, instead by locking to it using ***Audio codec*** in **Adv** tab in the VoIP profile you are using. If it is not possible to make VoIP calls with this configuration then please contact your VoIP operator in order to enable G.729 support in the system.

- **Echo**

  If you experience echo during the conversation, use a wired handsfree with your device. You could also have the other party use a handsfree or headset, if appropriate, or try to lower the volume slightly.

- **Audio goes only one way**

  See 4.2 Networking issues section for more information.

4.4 Standby issues

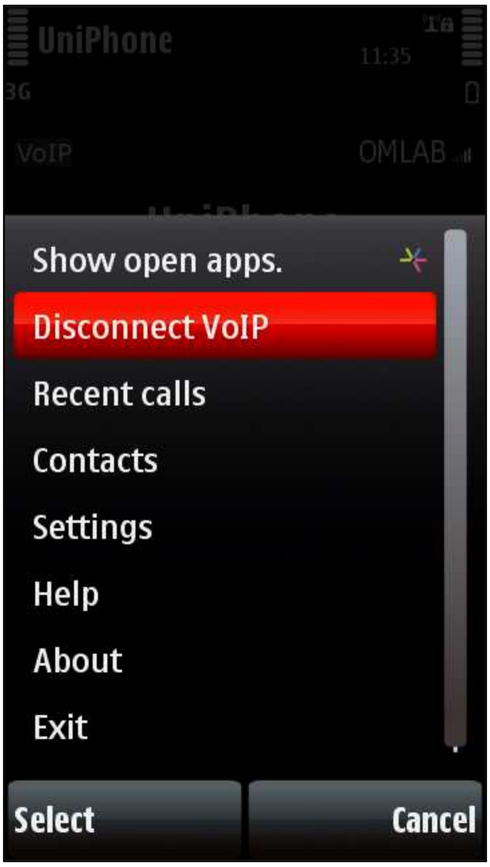
If UniPhone does not have a network connection it will periodically try to set up one. If there has been user activity during the last minute, a new connection attempt will be made every fifth second. If there has been no user activity during the last minute, then UniPhone goes into power save mode, and new connection attempts will be made with an increasing time interval with a maximum interval of 10 minutes (600 seconds). The periodic time between connection attempts in power save mode are: 10, 20, 40, 80, 160, 320 and 600 seconds. This is to minimise power consumption while maintaining reasonable quick re-connection to networks when they become available. If the phone is brought up from power save mode the sequence is reset.

4.5 Power consumption issues

Power consumption in the device varies depending on how much functionality is used and affects how often the device batteries need to be charged. Among things that consume much power are the processor, display, cellular 3G/GSM radio, Wi-Fi radio and Bluetooth radio. The device consume more battery power while being connected to the cellular 3G/GSM network and while having active telephone calls.

Wi-Fi interface consumes even more power on some devices. Power consumption increases when the Wi-Fi interface is turned on and it is even higher when a VoIP call is active. For the VoIP calls the power consumption is fairly constant but higher than 3G/GSM calls because VoIP calls are routed through the Wi-Fi interface. However, for the case when the phone is connected to the VoIP (SIP) server and is also in the standby mode, the additional power consumed due to the Wi-Fi interface is typically quite small but a slight variation may be observed depending on the Wi-Fi network quality. Disconnecting the application from the VoIP server would save negligible amount of power. Disabling the Wi-Fi interface saves maximum power. Even if the Wi-Fi interface is enabled and UniPhone application is closed, it would still save some although little amount of power.

It is possible to save significant amount of power if the Wi-Fi interface is disabled and the UniPhone application is still running in the background. In that case, later when the Wi-Fi interface is enabled, the application would still function in a normal way and does not need to be restarted.



4.6 Video call/recording issues

Unexpected behaviour may be observed if an active video call, or video recording session, on your device is interrupted by an incoming VoIP call, or if an active VoIP call is interrupted by an incoming video call. However, the video call would not be terminated, while the VoIP call would, since the video call occupies the audio path. Video recording is not supported during VoIP call even though it might work.

A. OptiMobile UniPhone functionality specifications

OptiMobile UniPhone features

- SIP compatibility allows you to:
    - communicate with other SIP compatible VoIP applications and systems.
    - call a regular phone using a SIP-PSTN gateway typically through a VoIP operator or corporate IP-PBX.
    - call a computer or IP-phone in a private network through a SIP proxy.
    - handle multiple calls: put call on hold and switch between two active calls.
- Support for sending DTMF signals.
- Configurable number handling with support for a wide variation of systems, from generic operators to more advanced enterprise PBX systems. The number handling settings are:
    - ***Country/Region code***
    - ***Convert plus to***
    - ***National prefix***
    - ***External line prefix***
    - ***Max internal num len***
- Selectable preference of early media to support custom streamed ring tones and other feedback tones with early media.
- DTMF modes: **RTP out-of-band** and **SIP INFO**.
- Selectable autostart of UniPhone at phone start-up.
- Several VoIP profiles with per profile selectable settings.
- AP handling with dynamic VoIP profile attachments.
- Integration of platform phone **Contacts**, **Calls Log** and **Home screen dialler** makes the phone easy to use.
- 3G/GSM compatibility allows you to use 3G/GSM networks for dialling when *3G/GSM Line used*. Within Wi-Fi coverage VoIP line is used for dialling, otherwise the circuit switched 3G/GSM line is automatically selected.
- Support for DTMF based and operator based routing of outgoing 3G/GSM calls, for call routing through e.g. a corporate PBX.
- Handling emergency calls using cellular 3G/GSM network.
- Seamless handover of active calls from VoIP to 3G/GSM and inversely using *OptiMobile VCC Enabled Services*.
- Supported languages: UK English, Svenska.

RFCs at least supported

- RFC 3261: SIP: Session Initiation Protocol
- RFC 2327/4566: SDP: Session Description Protocol
- RFC 3264: An Offer/Answer Model with Session Description Protocol (SDP)
- RFC 4317: Session Description Protocol (SDP) Offer/Answer Examples
- RFC 2617: HTTP Authentication: Basic and Digest Access Authentication
- RFC 3489: STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
- RFC 3581: An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
- RFC 3608: Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration
- RFC 4320: Actions Addressing Identified Issues with the Session Initiation Protocol's (SIP) Non-INVITE Transaction
- RFC 4321: Problems Identified Associated with the Session Initiation Protocol's (SIP) Non-INVITE Transaction
- RFC 2976: The SIP INFO Method

- RFC 3550: RTP: A Transport Protocol for Real-Time Applications
- RFC 3951: Internet Low Bit Rate Codec (iLBC)
- RFC 3952: Real-time Transport Protocol (RTP) Payload Format for Internet Low Bit Rate Codec (iLBC) Speech
- RFC 4733: RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals.

Supported media types

G.711 (u-law and a-law) and G.729. Note that the SDP codec priority list of the SIP proxy/media gateway is used when no audio codec is locked in VoIP profile Advanced settings.

3G/GSM call interworking

Interworking with native telephony application with missed incoming VoIP call notifications. See 3.4.4 Incoming VoIP call during active cellular call and 3.4.5 Incoming cellular call during active VoIP call for more information.

Interoperability

SIP interoperability to at least these SIP platforms - Asterisk, SER and HotSIP/Oracle.

Compatible devices

Nokia phones running S60 3rd Edition Feature Pack 1 based on Symbian OS v9.2, phones running S60 3rd Edition Feature Pack 2 based on Symbian OS v9.3 and phones running S60 5th Edition based on Symbian OS v9.4.

B. OptiMobile UniPhone number handling

This appendix is intended to describe how UniPhone converts numbers from **UniPhone dialler**, and the devices **Home screen dialler**, **Contacts** and **Calls Log** before dialling.

- **Input number**: what UniPhone gets from **UniPhone dialler**, and the devices **Home screen dialler**, **Contacts** and **Calls Log**.
- **Output number**: what UniPhone dials when *VoIP Line used*, **and** when DTMF or operator supported routing is used also when *3G/GSM Line used.*

Sample configuration 1

Country/Region code: `empty`
Convert plus to: `empty`
National prefix: `empty`
External line prefix: `empty`
Max internal num len: `0`

With all number conversion fields empty, the output number equals to input number. The only change is that visual separators is removed from input number.

Input number	Output number
+46 (70) 12345	+467012345
+46 7012345	+467012345
4670123	4670123
070-12345	07012345
0007012345	0007012345
+477012345	+477012345
4770123	4770123
0000477012345	0000477012345

Sample configuration 2

Country/Region code: `46`
Convert plus to: `00`
National prefix: `0`
External line prefix: `empty`
Max internal num len: `7`

This is the default configuration for Sweden.

Input number	Output number
+46 (70) 12345	07012345
+46 7012345	07012345
4670123	4670123
070-12345	07012345
0007012345	0007012345
+477012345	00477012345
4770123	4770123

0000477012345	0000477012345

Sample configuration 3

Country/Region code: `46`
Convert plus to: `00`
National prefix: `0`
External line prefix: `00`
Max internal num len: `7`

When using this configuration, `00` is added as prefix to all numbers longer than 7 digits, i.e. ***Max internal num len***. This prefix is also added in the same manner for numbers dialled from **UniPhone dialler**, and the devices **Home screen dialler**, **Contacts** and **Calls Log**.

Input number	Output number
+46 (70) 12345	0007012345
+46 7012345	0007012345
4670123	4670123
070-12345	0007012345
0007012345	000007012345
+477012345	0000477012345
4770123	4770123
0000477012345	000000477012345

Sample configuration 4

Country/Region code: `46`
Convert plus to: `00`
National prefix: `0`
External line prefix: `00`
Max internal num len: `0`

This is a typical enterprise configuration. All numbers in **Contacts** is expected to be entered in international format, beginning with +. When entering an external number in **UniPhone dialler** or the devices **Home screen dialler** the user is expected to add the external line prefix herself.

Input number	Output number
+46 (70) 12345	0007012345
+46 7012345	0007012345
4670123	4670123
070-12345	07012345
0007012345	0007012345
+477012345	0000477012345
4770123	4770123
0000477012345	0000477012345

Sample configuration 5

Country/Region code: `empty`
Convert plus to: `00`
National prefix: `0`
External line prefix: `00`
Max internal num len: `0`

This configuration is basically the same as sample configuration 4 with the difference that country/region code is empty.

Input number	Output number
`+46 (70) 12345`	`0000467012345`
`+46 7012345`	`0000467012345`
`4670123`	`4670123`
`070-12345`	`07012345`
`0007012345`	`0007012345`
`+477012345`	`0000477012345`
`4770123`	`4770123`
`0000477012345`	`0000477012345`

Sample configuration 6

Country/Region code: `46`
Convert plus to: `00`
National prefix: `empty`
External line prefix: `00`
Max internal num len: `0`

This configuration is basically the same as sample configuration 4 with the difference that national prefix is empty.

Input number	Output number
`+46 (70) 12345`	`007012345`
`+46 7012345`	`007012345`
`4670123`	`4670123`
`070-12345`	`07012345`
`0007012345`	`0007012345`
`+477012345`	`0000477012345`
`4770123`	`4770123`
`0000477012345`	`0000477012345`

C. OptiMobile UniPhone license conditions

License conditions for OptiMobile UniPhone are regulated by separate agreements.