
ALCATEL

TEMPORIS IP 200

ADMINISTRATOR'S GUIDE



Temporis IP 200 phone is a full-featured, qualitative SIP solution dedicated to business environment use. Its wide interoperability guarantees seamless integration with a large number of IP PBX and VoIP providers.

Please read through this guide for a better knowledge on available functionalities and configuration capabilities you can have access to with Temporis IP200.

Declaration of Conformity



Hereby, it's declared that this phone is in conformity with the essential requirements and other relevant provisions of the CE, FCC.

CE Mark Warning

This is a class B device, in a domestic environment; this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning



To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

Table of Contents

Getting Started - 7 -

Packing List.....	- 7 -
Assembling Temporis IP 200	- 7 -
Configuration and Registration	- 10 -
Basic Configuration via Web Page.....	- 10 -
Basic Configuration using phone keypad	- 12 -

Overview - 13 -

Keypad.....	- 13 -
LEDs	- 14 -
Icons	- 15 -
User Interface	- 15 -

Customizing Your Phone..... - 18 -

General Settings.....	- 18 -
Phone Status	- 18 -
Language	- 19 -
Time and Date	- 20 -
Maximum number of Calls.....	- 21 -
Keypad Lock.....	- 22 -
Logo Customization	- 25 -
Audio Settings.....	- 26 -
Volume	- 26 -
Ring Tones	- 26 -
Codec Selection	- 29 -
Tone Settings.....	- 30 -

Voice	- 31 -
Distinctive Ringing	- 32 -
Contact Management	- 33 -
Edit/Add/Delete Contacts	- 33 -
Import/Export Contact list.....	- 37 -

Basic Call Functions - 39 -

Outgoing calls	- 39 -
Call Devices.....	- 39 -
Call Methods.....	- 39 -
Live Dialpad	- 42 -
Send Key configuration	- 43 -
Password dial.....	- 44 -
Call Completion.....	- 46 -
Auto Redial.....	- 46 -
Hot Line.....	- 48 -
Anonymous call.....	- 49 -
Dial Plan	- 51 -
Replace Rule	- 52 -
Dial Now.....	- 53 -
Area Code	- 54 -
Forbidden Numbers (Block Out).....	- 54 -
Incoming calls	- 56 -
Answering an incoming call	- 56 -
Rejecting an incoming call.....	- 56 -
DND	- 56 -
Call Forward	- 57 -
DND/FWD Synchronisation	- 58 -
Intercom.....	- 59 -
Auto Answer.....	- 60 -

Call log	- 61 -
Accessing Call log	- 62 -
Accessing Missed Calls list	- 62 -
Accessing Dial Calls list	- 62 -
Call list items management	- 63 -
Missed call log enable/disable	- 63 -
During an Active Call	- 64 -
Mute	- 64 -
Call Hold	- 65 -
Call Waiting	- 65 -
Handling a waiting call	- 65 -
Call Waiting Configuration	- 65 -
Call Transfer	- 67 -
3-way Conference	- 67 -
Network Conference	- 68 -
Voicemail	- 69 -
Voicemail settings	- 69 -
Retrieving your Voicemail messages	- 70 -

Advanced Functions.....- 73 -

Account Setting	- 73 -
TLS	- 74 -
DNS-SRV	- 74 -
Network Settings	- 75 -
PC Port Setting	- 75 -
Advanced Network Settings	- 76 -
LLDP	- 76 -
VLAN	- 77 -
HTTPS	- 78 -
802.1X	- 79 -

Other Settings	- 79 -
Maintenance Tasks	- 80 -
User levels	- 80 -
Administrator Mode	- 81 -
Reboot	- 81 -
Reset to Factory	- 82 -
Configuration Export/Import	- 83 -
WatchDog	- 83 -
Action URL/URI	- 84 -
Firmware Update and Autoprovisioning	- 85 -
Decryption	- 86 -
Zero-Active Interface	- 88 -
System Log Export	- 89 -
PCAP Trace Export	- 89 -
Keys configuration	- 89 -
Send Key configuration	- 89 -
Programmable keys	- 90 -
Line keys Configuration	- 91 -
BLF	- 92 -
BLF List	- 93 -
Note:	- 93 -
Line	- 93 -
Speed Dial	- 93 -
Intercom	- 94 -
URL Record	- 94 -
Shared Line	- 95 -
Conference	- 95 -
Forward	- 96 -
Transfer	- 96 -
Hold	- 97 -

DND.....	- 97 -
Redial	- 97 -
Call Return	- 98 -
Pick Up	- 98 -
Call Park	- 98 -
Group Listening.....	- 99 -
Voice Mail	- 99 -
DTMF.....	- 99 -
Public Hold.....	- 100 -
Private Hold	- 100 -
Group Pick up.....	- 101 -
Paging	- 101 -
Record.....	- 101 -
Prefix.....	- 102 -
Zero-Active.....	- 102 -
ACD	- 102 -
Local PhoneBook	- 103 -

Trouble shooting- 104 -

Appendix A.- 106 -

Line Key configuration menu	- 106 -
Memory Key configuration menu.....	- 108 -

Getting Started



Packing List

The following components are included in your package:

TEMPORIS IP 200

Phone Foot Stand

Wall mount accessories

Handset & Handset Cord

Ethernet Cable

Quick Start Guide

Assembling Temporis IP 200

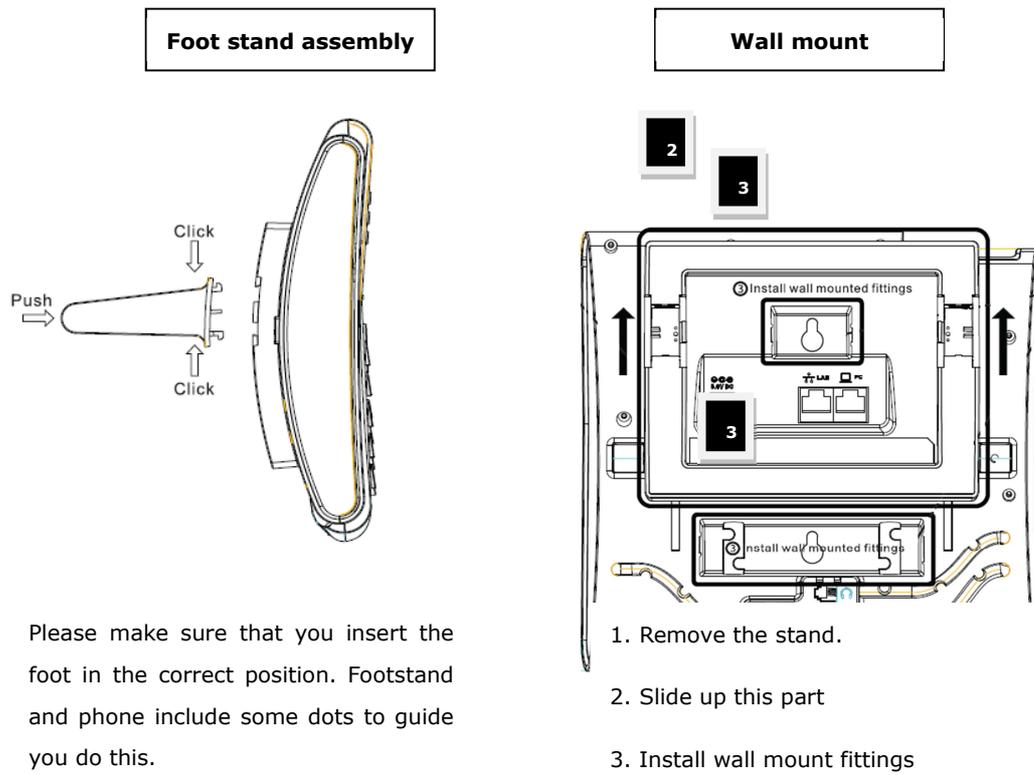
This section introduces how to assemble the phone with the components in the packing list:

Attach the foot stand or wall mount accessories;

Connect Handset;

Connect Network and Power.

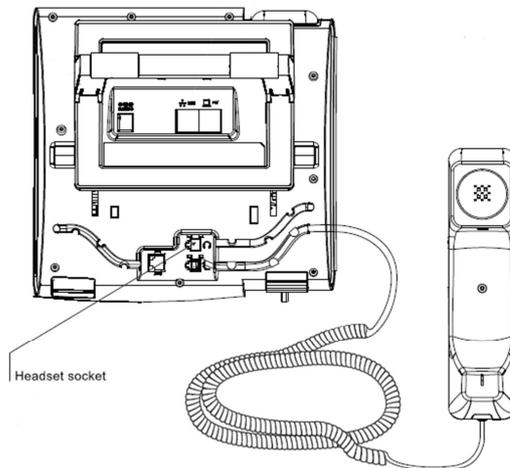
Attach the footstand or wall mount accessories, as shown below:



Please make sure that you insert the foot in the correct position. Footstand and phone include some dots to guide you do this.

1. Remove the stand.
2. Slide up this part
3. Install wall mount fittings

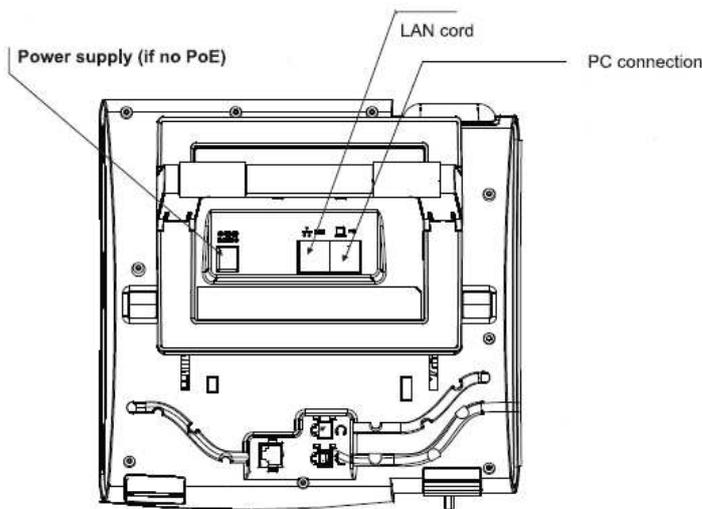
Connect Handset and Headset, as shown below:



Note: headset is not provided.

Connect Network and Power

You can either connect the phone to the AC Power directly using a power adapter (not supplied) or to a PoE compliant switch or hub. Your system administrator will advise you on which one to use.



Note:

If inline power is provided, do not install AC adapter (not provided). Make sure the Ethernet cable and switch/hub are PoE compliant.

The Internet Port can also be connected to Hub/Switch/IP PBX or other internet devices.

The phone can also share the network connection with other network devices such as PCs. Connect the phone's PC port and computer's network port together using an Ethernet cable.

Configuration and Registration

Some simple configurations are needed to get your phone ready for making calls: network and account information must be correct.

If you are not an administrator, please contact your internet administrator or service provider for more details.

Basic Configuration via Web Page

Temporis IP200 features an embedded web server to help administrators and users configure, manage and use their phone in an intuitive and simple manner. The phone's web server can be accessed with any web browser.

Press OK button on the keypad of the phone to enter the status page and find out your Temporis IP 200 IP address. Enter it (for example <http://192.168.1.100>) into the address bar of your web browser. The default administrator's login name and password are **admin/admin**.

Note:

Please locate your PC in the same network segment of your Temporis IP (192.168.1.X) to be able to access the web configuration page. Please consult your system administrator for help.

Network Settings

Choose Network->Internet Port (WAN) .

DHCP: By default the phone attempts to contact a DHCP Server in your network in order to obtain its valid network settings, e.g. IP address, subnet mask, gateway, DNS server, etc.

Static IP Address: If your phone cannot contact a DHCP Server for any reason, you need to enter the network settings manually via Static IP Address. Please contact your internet administrator for more details.

PPPoE: If you are using an xDSL Modem, you may connect your phone to the internet via PPPoE mode. Please contact your ISP for the **User Name** and **Password** for internet access.

Note: Using the wrong network parameters may result in inaccessibility of your phone and may also have an impact on your network performance. Please contact your network administrator.

Account Settings

The phone attempts to register to the SIP server using the account data provided by automatic or manual initialization.

Choose Account page. You will find the following parameters:

Field	Description
Register Status	It shows the register status of the phone (read only)
Account Active	You can choose on/off to enable/disable the account respectively.
Label	The name showing on the LCD of current device.
Display Name	The local phone name showing on the other phone when calling.
Register Name	Register name provided by ISP.
User Name	User account information, provided by ISP.
Password	Account password provided by ISP.
SIP Server	SIP server address provided by ISP.

When all accounts registering fail, the phone will display "No Service" by default.

When the phone reboots, it will register automatically. If many phones register at the same time, for example after a power outage, the SIP server may be affected; admins can set a register power up time so that the phones will register a random time after power up.

Setting power up time via web interface:

Choose network->Advanced-> Random Registration, enter the time in the field and validate with "Confirm".

Note:

Should the IP PBX (SIP registrar) require an authentication, a correct password is needed. Make sure you are using the appropriate input method or enter the password via the web user interface.

Basic Configuration using phone keypad

You will need admin privilege to perform this and other configurations. By default the phone will be in user mode, so you will need to perform a simple procedure to switch to admin mode. Phone will return to user mode after reboot.

Change phone to admin mode: Press MENU→7.**User type**. Press right navigation key until you see "admin" on the screen. Then press the down key to be prompted for the password. It is "admin" by default. Type it and press OK.

Network Settings: Press MENU->3.**Settings**->2.**Advanced**->OK. Enter the admin password, and choose **Network**->**WAN Port/PC Port /VLAN/Web Type/802.1x Settings** to enter the internet relating configuration page.

Account Settings: Press MENU->3.**Settings**->2.**Advanced**->OK. Enter the admin password, and choose **Accounts**->OK to configure the account settings.

You can refer to the above "Configuring via Web Page" for parameter details.

Note:

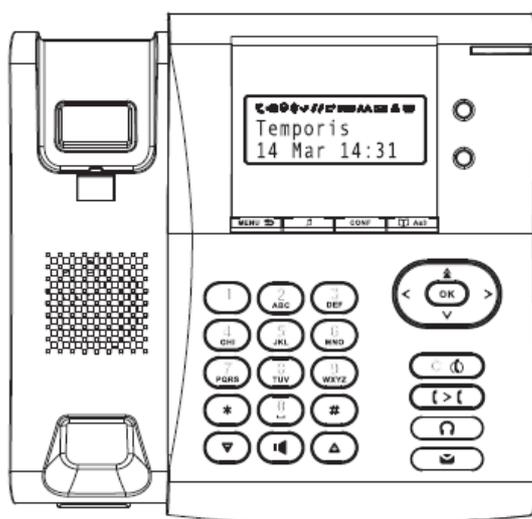
The active User Type is shown with an * on the left of display when you enter Menu→ 7. User Type menu, like this:

```
<*user >
```

Once you finish all admin related tasks you should go back to user mode with the same procedure, or simply reboot the phone.

Overview

Keypad



-  Hold key
-  Conference key
-  In idle mode : access to phonebook
Text input mode : toggle between letter/number
-  In idle mode : access to menu.
Back key and hang up
-  Line keys
-  Navigation keys (up,down, left, right) +
short cuts to other functions
-  In idle mode : access to redial list
-  Confirm key
-  In editing/pre-dialling mode :
erase character/digits
In communication mode :
mute/unmute a call
When ringing, press to reject call
-  Access to call transfer
-  Headset key
-  Access to voice mail
-  During ringing : increase ringer volume
In communication : increase earpiece,
headset or handsfree volume
-  Activate/deactivate handsfree
-  During ringing : decrease ringer volume
In communication : decrease earpiece,
headset or handsfree volume

LEDs

Table 1 Line Keys

LED Status	Description
Steady green	The account is active
Blinking green	There is an incoming call to the account, or there is a call on hold
Off	The phone is in idle status whatever registered /unregistered

Table 2 Line Keys set to BLF

LED Status	Description
Steady green	The monitored account is in idle status
Fast blinking green	There is an incoming call to the monitored account
Slow blinking green	The monitored account is on an conversation
Off	It is not active as BLF

Table 3 Power Indication LED

LED Status	Description
Steady amber	when new messages or new missed calls When boot process
Blinking amber	When there is an incoming call or when a call is muted
Slow blinking amber	When network is not available or sip server failure
Off	in idle mode

Icons

Temporis IP 200 displays different kinds of icons on its LCD, you can refer to the following table for their meanings:

Icon	Description
	Flashes when the internet is disconnected
	Missed calls
	Call in
	Call out
2aB	Input Method: all letters and numbers
123	Input Method: numbers
abc	Input Method: multi-lingual letters in lower case
ABC	Input Method: multi-lingual letters in upper case
	Call mute
	Voicemail
	Call forward
	Auto answer
DND	DND
	Keypad Lock
	In handset mode
	In headset mode
	In speaker mode

User Interface

There are two ways to customize specific options on your phone:

Using keypad and display of the phone.

Using Web user interface in an Internet browser from your PC; please refer to "Configuration and Registration" to get into the Web interface.

In many instances, it is possible to use either the phone user interface or the web interface to operate the phone and change settings. Some operations, however, are only possible via phone or web user interface. Please refer to the following table for differences:

Phone Options	Phone UI	Web UI
Status		
--IP		✓
--MAC		✓
--Firmware	✓	✓
--Network		✓
--Phone		
--Accounts		
Call Features		
--Forward	✓	
--Call Waiting	✓	
--Function Keys	✓	
--Send Key	✓	
--Hot Line	✓	
--Anonym Call	✓	✓
--Auto Redial	✓	
--Intercom	✓	
--DND	✓	
--Call Completion	✓	
--Logs Setting	✓	
Basic Phone Functions		
--Language	✓	
--Time & Date	✓	
--Ring Tone	✓	✓
--Phone Volume	✓	
Advanced Phone Functions		
--Accounts	✓	
--Network	✓	✓

--Keypad Lock	√	
--Reset Factory	√	
--Set Password	√	
--Set AES Key	√	
--Auto Provision		
Other Features		
--Messages		√
--History	√	√
--Directory		√
--User Mode		

Note:

- 1) The above table only indicates most of phone functions rather than all of them. Please refer to next chapters for more details.
- 2) Some functions are password protected. In this case when you are prompted for a password make sure to introduce one in accordance with active User Type

The default administrator password is **admin**.

Customizing Your Phone

General Settings

Phone Status

You can view the status of your phone using the Phone interface or the Web interface.

This option allows you to review:

IP

MAC

Firmware

Network: MAC, WAN, LAN, Gateway, DNS, etc

Phone: Model, Hardware, Firmware, Product ID and MAC

Accounts: The 2 SIP accounts status

To check the Phone Status via Phone interface:

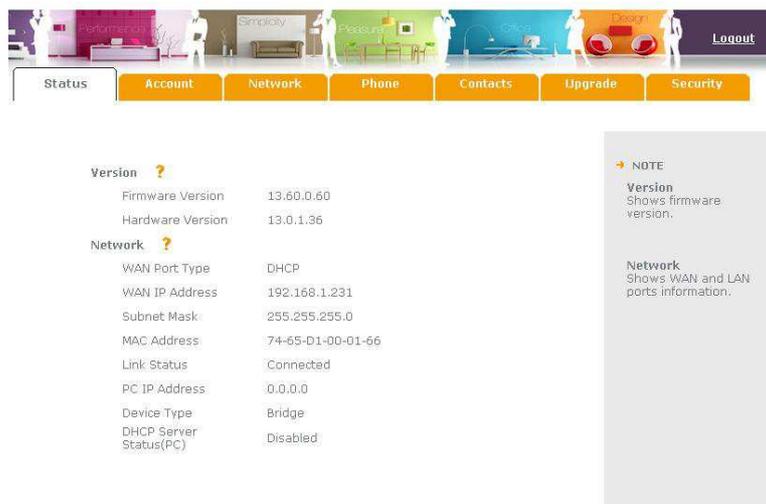
Press OK button directly to check the IP.

Use the navigation keys to check the other information.

```
IP:
192.168.0.120
```

To view the Phone Status via Web interface:

Open a web browser and input the IP Address of the phone. Enter login and password (default login and password are both "admin"). Status page will be shown directly.



Language

The default Phone interface language is **English**.

It also supports French, Spanish, Dutch, German, Italian, Polish, Portuguese, Turkish, , etc. You can change the language for the phone user interface and the web user interface independently from each other.

Note:

All languages may not be available for selection. The available languages depend on the language packs currently loaded on the IP phone. Please contact your system administrator or technical support for more information about language packs loading.

To change the language via Phone interface:

Press ->3.**Settings**->1.**Basic**->1.**Language**->OK.



Scroll through the list of available languages.

Press OK button when the desired language is highlighted.

Press  key to return to the previous screen.

Web interface language depends on your computer Operative System. It will automatically match the language with your computer and browser, and fall back to English if needed.

Time and Date

Time and date are displayed on Temporis IP200 idle screen. By default the phone will try to obtain them from a preconfigured SNTP server, but you can set them manually if needed. Please contact your system administrator if the time or date are incorrect.

To change Time and Date via the Phone interface:

Press ->3.**Settings**->1.**Basic**->2.**Time & Date**->OK.



1. SNTP
2. Manual

If SNTP is chosen, the phone will automatically get the time from an NTP Server. Use the up and down navigation keys to highlight the specific options and change parameters to meet your needs. You can set Time Zone, NTP Server1/Server2, and DST respectively. By default GMT+1 and automatic DST are configured.

If Manual is chosen, the time can be set manually. Use the up and down navigation keys to highlight the option and enter the specific date and time.

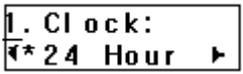


1. Date (D-M-Y) :
18-10-10

Press OK button to save your changes. Or press MENU key to return to the previous screen.

To set time format via the Phone interface:

Press ->3.Settings->1.Basic->2.Time & Date->3.Time & Date->OK.



1. Clock:
24 Hour

Use the Left/Right navigation keys to choose a preferred time format: 12 hour or 24 hour.

Press the OK button to save the changes and return to the previous screen.

To change Daylight Saving Time Settings via the Web interface:

- 1) Go to Choose Phone->Preference->Daylight Saving Time. Select between Enable/Disable/Automatic
- 2) If you select Enable , then you will need to set Daylight Saving Time period start and end dates.
- 3) If Automatic is chosen dates will be derived from the content of a table saved within your configuration. This table is named AutoDST.xml and can be changed via provisioning.

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Preference | Features | Function Keys | Action URL | Voice | Ring | Tones | Dial Plan

WEB Language: English ?
 DHCP Time: Disabled
 Time Zone: +1 Albania(Tirane) ?
 Primary NTP Server: europe.pool.ntp.org ?
 Secondary NTP Server: europe.pool.ntp.org ?
 Update Interval(seconds): 1000 ?
 Daylight Saving Time: Automatic ?
 Fixed Type: Enabled By Week
 Start Month: Automatic
 Start Day of Week: Sunday
 Start Week: Last In Month
 Start Hour of Day: 2
 End Month: October
 End Day of Week: Sunday
 End Week: Last In Month
 End Hour of Day: 3
 Offset(minutes): 60
 Manual Time: Disabled ?
 Time Format: 24 Hour ?
 Live Dialpad: Disabled ?
 Inter Digit Time(1~14)(seconds): 4 ?
 Flash Hook Time(<800ms): 1 ?
 Keyboard Lock: Disabled ?
 WatchDog: Enabled
 Ring Type: Ring2.wav Del ?
 Upload Ringtone: Parcourir...
 Upload Cancel
 Confirm Cancel

+ NOTE
Time Zone
 Choose the time zone you are in.
NTP Server
 Server from which time information will be retrieved.
Update Interval
 Time between time synchronizations with the NTP server.
Daylight Saving Time
 Activate summer time policy.
Manual Time
 Enable/disable to set time manually.
Ring Tone
 Ringtones to be uploaded must be in wav format (8K sampling rate, mono, 16-bit U-law compression)

Note:

By default the time zone is +1 France. Daylight Saving Time is Automatic.

Maximum number of Calls

In some situations service providers want to limit the number of calls a subscriber is able to send or receive.

Your IP200 has been configured by default to handle 2 simultaneous calls (either incoming, outgoing or a mixture). However you can set this limit to any number up to 20.

To set max number of calls using the web interface:

- 3) Go to Phone->Features->Max simultaneous calls
- 4) Enter the desired value and press Confirm button

Note:

This setting is independent from Call Waiting, which only limits incoming calls when disabled. See Call Waiting chapter for more details.

Keypad Lock

You can lock the keypad of your phone when you are temporarily not using it. This function helps you to protect your phone from unauthorized use. You can lock the following specific keys:

Menu Key: The Menu key cannot be used until unlocked. You cannot access the menu of the phone.

Function Keys: The hard function keys cannot be used until unlocked. You cannot access the MESSAGE, HEADSET, TRANSFER, REDIAL, Speaker, CONFERENCE, HOLD, Navigation Keys, OK button, Mute button, Volume Adjustment button, etc.

All Keys: None of the keys can be used until unlocked. You can only use the phone to answer incoming calls.

Lock&Answer: In this mode you cannot initiate or terminate calls. You will receive calls in auto answer mode. You will also have access to the Menu.

To enable keypad lock via Phone interface:

Press ->3.**Settings**->2.**Advanced**, enter the password, and then press OK button.

Choose 3.**Keypad Lock**->OK.

Use the left or right navigation keys to highlight the one you want to lock.



1. Keypad Lock:
* Disable

Press OK button to active the change, or MENU to return to the previous screen.

The icon  will be displayed on the top right corner of the idle screen.

If you choose **Lock&Answer**, it will display the icon  and AA on the LCD.

To unlock the phone via Phone interface:

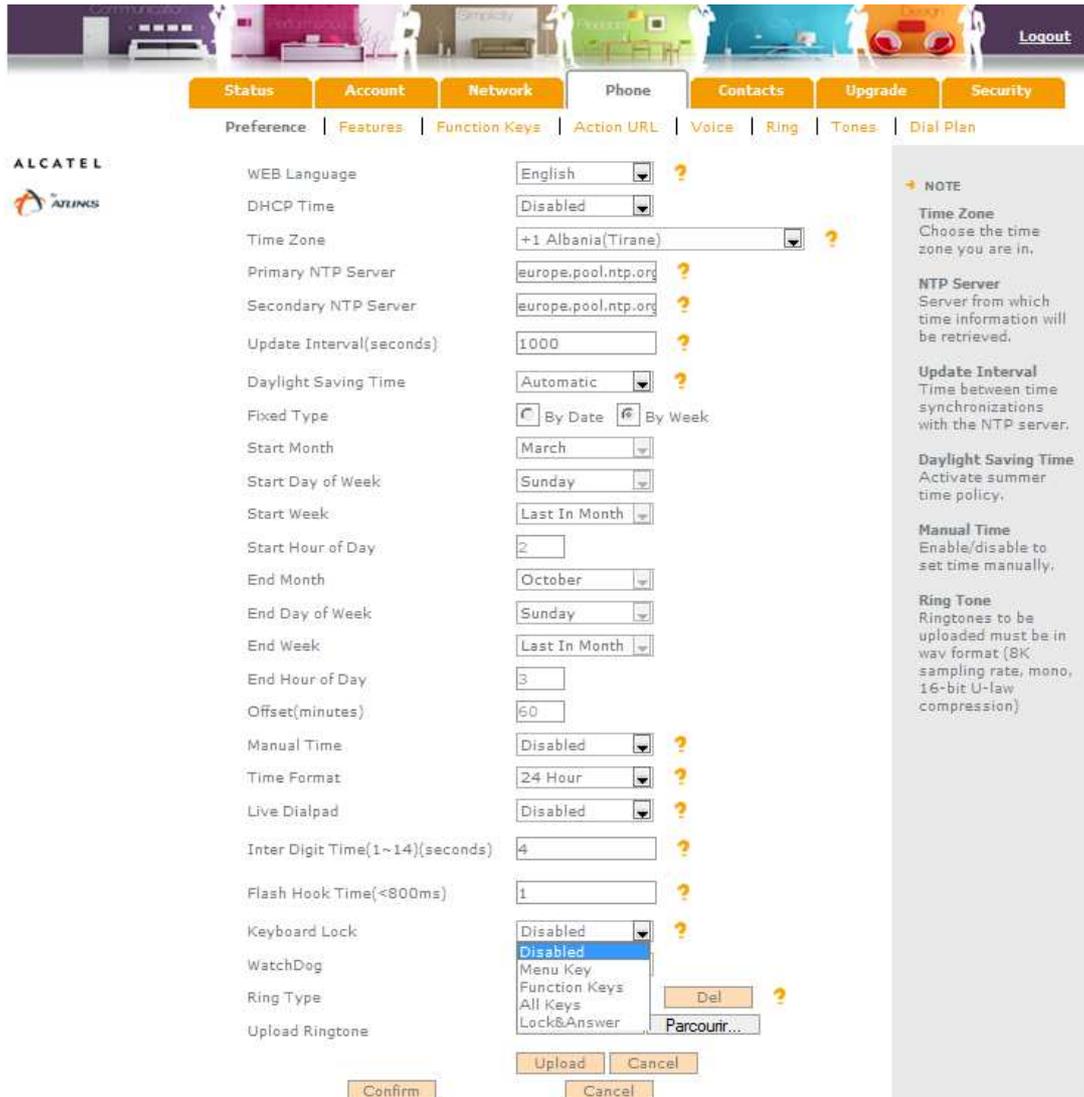
Press  key, you are prompted for the password.

Enter the password, and then press OK button, the phone is unlocked.

If you choose **Lock&Answer**, you have to enter ->3.**Settings**->2.**Advanced**->3.**Keypad Lock** interface to disable this option.

To enable keypad lock via Web interface:

Choose Phone->Preference-> Keyboard Lock. Please refer to the instructions above for the different options.



To configure a programmable key as shortcut for keypad lock via Web interface

You can use any of the programmable keys to provide a shortcut to Keypad Lock. Choose Phone->Function Keys->select a programmable key and in the scrolling menu select Keypad lock.

In idle mode, when pressing this key, you're prompted to enter the password and then you enter the keypad lock menu. You can use the same key to unlock.

Note:

1. The default password for lock/unlock is **admin** or **user** depending on the active User Type.
2. Users can make emergency calls when the phone is locked if an emergency number has been configured

Logo Customization

Temporis IP200 can display a textual logo on the idle screen. Maximum length is 15 characters.

Using the web interface choose Phone-> Features-> Use Logo. Select enable or disable in the pull-down menu.

Ringer Device for Headset	Use Speaker	?
Headset Sending Volume (1~53)	29	
Call rejected response code	486 (Busy here)	?
DND Response code	480 (Temporarily not ava	?
DND On Code		?
DND Off Code		?
Allow Intercom	Enabled	?
Intercom Mute	Disabled	?
Intercom Tone	Enabled	?
Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attended Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	1	
ACD Auto Available	Disabled	
ACD Auto Available Timer (0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
Process 180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength		
PswDial	Disabled	
SaveCallHistory	Enabled	
Use Logo	Enabled	?
Text Logo	Enabled	

Picture must be in dob format. Black and white or 2 gray scale are possible.

Confirm Cancel

If "enable" is selected you will be able to enter the text in the "Text Logo" field.

Click Confirm button to save the change.

Audio Settings

Volume

Handset, speaker, headset and ringer volumes can be adjusted independently.

To adjust the volume in idle mode:

Press ->3.**Settings**->1.**Basic**.

Scroll to 4.**Phone Volume**, and press OK button. Select the item you wish to adjust and use  and  or left and right navigation keys to increase or decrease volume.

Press OK button to save volume change or MENU key to cancel.

To adjust ring volume you can also press  and  when the phone is in idle status.

To adjust the volume when you are in an active call:

When Handset/Headset/Hands-free mode is activated, press  or  to adjust the volume.

Note:

The volume can only be adjusted via Phone interface.

Ring Tones

You can select the type of the ring tone.

To select Ring Tone Type via Phone interface:

Press ->3.**Settings**->1.**Basic**->OK.

Scroll to 3.**Ring Tone**, and press OK button.

Use the up and down navigation keys to highlight your selection. The tone will be played.

* 1. Ri ng1. wav
2. Ri ng2. wav

Press OK button to save the change or MENU to cancel.

To change Ring Tone Type via Web interface:

Choose Phone->Preference->Ring Type. Highlight your selection in the scroll-down menu and click confirm button to update the change. You can also delete a ring tone by clicking the Delete button.

The screenshot displays the Alcatel ATLINKS web interface for configuring a phone. The navigation menu includes Status, Account, Network, Phone, Contacts, Upgrade, and Security. Under the 'Phone' menu, the 'Ring' option is selected. The configuration page shows various settings for the phone, including time zone, NTP servers, and update intervals. The 'Ring Type' setting is currently set to 'Ring2.wav'. A dropdown menu is open, showing a list of ring tones: Ring1.wav, Ring2.wav (highlighted), Ring3.wav, Ring4.wav, Ring5.wav, Ring6.wav, Ring7.wav, and Ring8.wav. A 'Del' button is located next to the dropdown. A 'Confirm' button is at the bottom of the page. A 'NOTE' section on the right provides additional information about the 'Ring Tone' settings, stating that ring tones must be in wav format (8K sampling rate, mono, 16-bit U-law compression).

Note:

System ring tones cannot be deleted.

To upload a new Ring Tone via Web interface:

Choose Phone->Preference->Upload Ringtone.

Click Browse button to choose your ring tone file.

Click Upload button to upload the file.

Note: The ring tone file format must be in 16bits WAV format (via Ulaw Compression), 8K sample rate (monophony). Blank or other special characters cannot be included in the file name.

To specify ring tones for a specific account via Web interface:

Choose Account->Basic->Ring Type option, and highlight the preferred one for the chosen account in the scroll-down menu, then click confirm button to update the change.

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Account: Account 1

Basic >>

Register Status: Registered

Account Active: On Off

Label: Ext 511

Display Name: John

Register Name: 511

User Name: 511

Password: *****

SIP Server: 10.24.20.91 Port: 5060

Enable Outbound Proxy Server: Disabled Port: 5060

Outbound Proxy Server: Port: 5060

Transport: UDP

Backup Outbound Proxy Server: Port: 5060

NAT Traversal: Disabled

STUN Server: Port: 3478

Voice Mail: *2

Proxy Require:

Anonymous Call: Off

On Code:

Off Code:

Anonymous Call Rejection: Off

On Code:

Off Code:

Missed call log: Enabled

Auto Answer: Disabled

Ring Type: common

Codecs >>

Advanced >>

Confirm

NOTE

Display Name
SIP service subscriber name used for Caller ID display.

Register Name
SIP service subscriber ID used for authentication.

User Name
User account provided by VoIP service provider or PBX admin.

NAT Traversal
Control STUN server settings.

Proxy Require
Relevant for Nortel server only. If you wish to login to a Nortel server, value should be: com.nortelnetworks.firewall

Codecs
Select codec types and priority.

Advanced
Administrator oriented parameters.

Codec Selection

Temporis IP200 supports the following voice codecs:

G.722, G.723_53, G.723_63, G.726-16, G.726-24, G.726-32, G.726-40, G.729, PCMU and PCMA

You can enable/disable the desired codecs via Web interface. Please contact your System Administrator for more details about the codecs.

To enable/disable the codecs via web interface:

Choose Account->Codecs.

Use the navigation keys to highlight the desired one in the Enable/Disable Codecs list, and press the **>>** / **<<** to move to the other list.

Click Confirm button to save the change.

Note:
Codec Selection can only be done via Web interface.

Tone Settings

Your IP200 allows you to configure how the following tones used for subscriber signaling will sound like:

- Dial
- Ring Back
- Busy
- Congestion
- Call Waiting
- Dial Recall
- Record

- Info
- Stutter
- Message
- Auto Answer

IP200 includes preconfigured tones for a large number of countries. However you can generate a fully customized tone set by defining their frequency and duration.

the frequency and time period of all the following tones:

To configure your tones via Web interface:

- 1) Go to Phone->Tones.
- 2) Select your country or Custom
- 3) If Custom is selected, enter the frequency(Hz) and time period(in ms) as follows:

Frequency /Time Period (for example 400/200).

- 4) Press Confirm button to save the changes, cancel to cancel the changes.

Note:

1. Please contact your system administrator for more information about the frequency and time period parameters. You can enter up to 8 groups for each tone.
2. Frequency set to 0, is interpreted as a silent period.

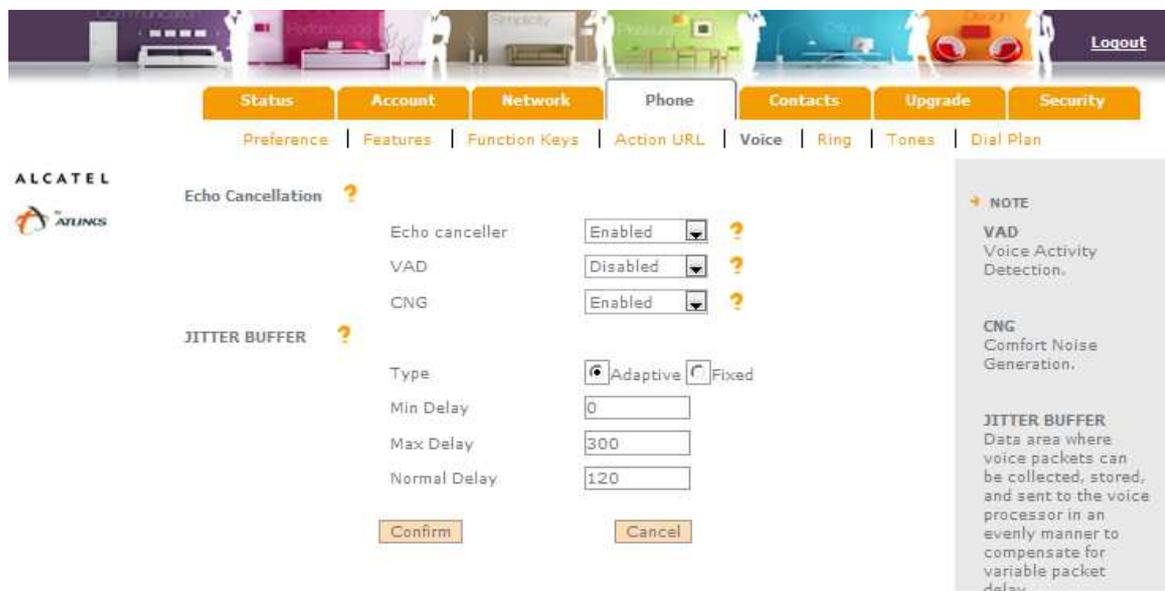
Voice

This section allows you to configure some parameters related with Voice management using the Web interface.

- 1) Go to Phone->Voice.
- 2) Set the following parameters:

Parameter	Description
Echo canceller	Defines whether to enable the echo canceller.
VAD	Voice activity detection (VAD), also known as speech activity detection or speech detection, is a technique used in speech processing in which the presence or absence of human speech is

	detected.
CNG	A comfort noise generator (CNG) is a program used to generate background noise for voice communications during periods of silence that occur during the course of conversation.
JITTER BUFFER	It is a shared data area where voice packets can be collected, stored, and sent to the voice processor evenly to overcome variable delay introduced by networks, known as jitter.
Type	To choose the type of JITTER BUFFER, Adaptive or Fixed.
Delay	To set Jitter Buffer Min Delay, Max Delay and Normal Delay parameters.



3) Press Confirm button to save your changes, or Cancel to delete them.

Distinctive Ringing

The Ring section in the web gui allows to configure how the phone will work with "Distinctive ringing" server feature.

You can set the values of the "Alert-info" header for an incoming call to determine which ring tone will be played.

Up to 10 distinctive ring tones can be set by configuring "Internal ringer text". This field will have to be filled in with the expected content of the Alert-info header in the received "INVITE".

To edit the Ring option via Web interface:

- 1) Go to Phone->Ring.

	Internal Ringer Text	Internal Ringer File
1		Ring1.wav
2		Ring1.wav
3		Ring1.wav
4		Ring1.wav
5		Ring1.wav
6		Ring1.wav
7		Ring1.wav
8		Ring1.wav
9		Ring1.wav
10		Ring1.wav

- 2) Set Internal Ringer Text to match the content of the Alert-Info header
- 3) Set Internal Ringer File: select ring tone to be used for that value of Alert-Info.
- 4) Click Confirm button to save the changes.

Contact Management

Edit/Add/Delete Contacts

Temporis IP200 features a 300 entries local phonebook and 10 additional speed dial keys. .

With the local directory you can:

- add,

-
- edit,
 - delete,
 - dial,
 - search for a contact in this directory.
 - create groups of contacts

Phonebook includes Directory and Blacklist.



To add a contact via Phone interface:

Press  or  ->6.**Directory**-> 1.**Directory**.

Press OK button.

Press OK button to enter into the directory, press UP navigation key to select ADD, Press OK button, enter Name, press down navigation key to enter phone number of the contact via the keypad. Use the  key to select between numeric and upper/lower case alphanumeric modes.

Press down navigation key to set and choose a special ring tone for the contact.

If you want to assign this contact to a group, use the down navigation key to select the group.

Press OK button to add the contact, or MENU key to cancel the change.

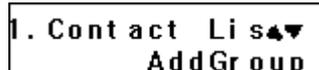
To add a Group via Phone interface:

Press  or ->6.**Directory**-> 1.**Directory**

Use left/right navigation key to choose **AddGroup**, press OK button to enter to the Add Group page.

Enter the group name and choose associated ring tone.

Press OK button to save.



To configure the speed dial keys via Phone Interface:

Press ->2. Features->2.Function Keys->2.Memory Keys

Use the up and down navigation keys to select the key on which you want to store a contact number, press OK key to confirm

Use the right and left navigation keys to select **Speed Dial**, then use the up and down keys to select **Account**, use the left or right navigation keys to if you want to assign this speed dial keys to a specific account

Then use the right and left navigation keys to select **Value**, enter the desired number and press OK key to confirm.

Note: in idle mode, with dial tone or during a conversation if you make a long press on the key, it will automatically call the contact stored on the key.

To edit/delete a contact via Phone interface:

Press  or ->6.**Directory**.

Press OK button twice.

Use the Up/Down navigation key to highlight the contact you want to edit/delete, then press left/right navigation keys to select Edit/Del page and press OK button.



Make the desired changes, press OK button, or press MENU return to previous menu.

To move a contact to the Blacklist via Phone interface:

If you add a contact to the blacklist your phone will automatically reject all calls from this contact.

Press  or ->6.**Directory**.

Press OK button twice.

Use the Up/Down navigation key to highlight the contact you want to move to blacklist,

then press left/right navigation keys to select M2B page.

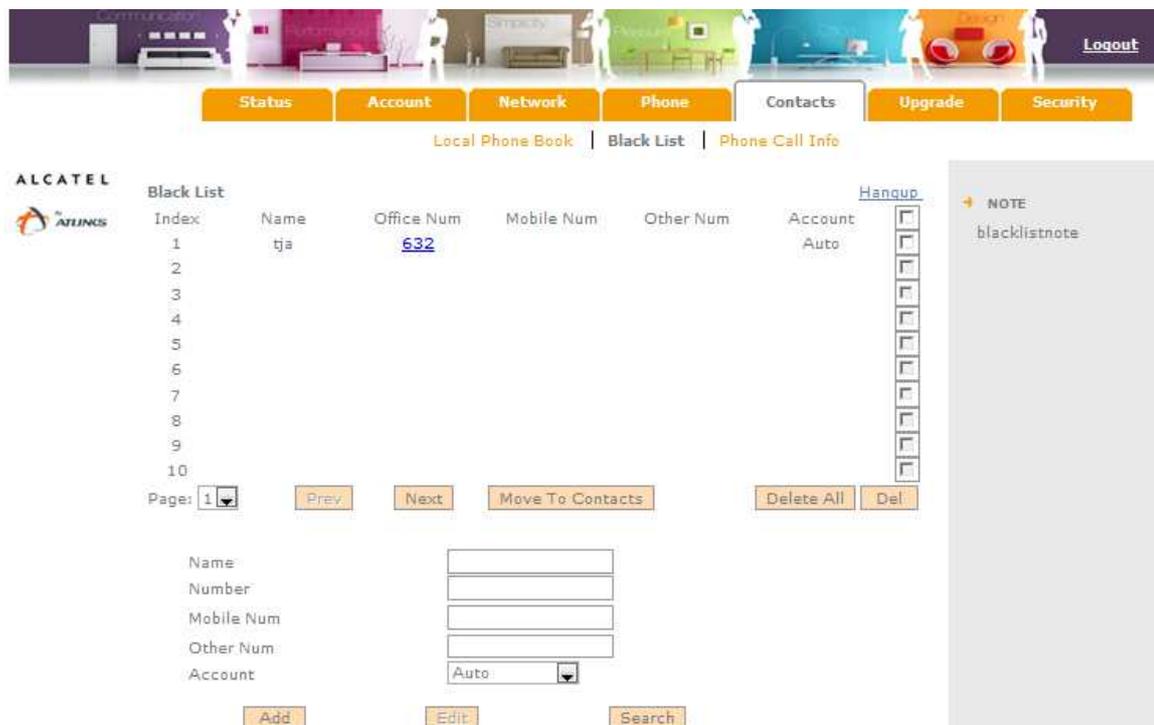
Press OK button, or press MENU to return to previous menu.

A warning message will ask for confirmation to move the contact.

Press OK button to confirm the operation, or press the MENU button to return to the directory

Manage Blacklist via Web interface:

Choose Contacts->Black List



To move a contact from Call Log to Contacts via Phone interface:

Press **MENU** ->History->OK, or use the down key shortcut

Use the navigation keys to highlight a record, then press the **Aa3** key to enter the edit page.

To add/delete/edit the Contact list via Web interface:

Go to Contacts→Local Phonebook. Fill in name and number and press Add

ALCATEL **Contacts** All Contacts ? Hangup

Index	Name	Office Num	Mobile Num	Other Num	Account	Groups
1	daniel	513			Auto	
2	George	514			Auto	
3	jose	123456			Auto	
4	tja	632			Auto	
5	wad				Auto	
6	wmt				Auto	
7						
8						
9						
10						

Page: 1 Prev Next Move To BlackList Delete All Del

Contacts

Name

Number

Mobile Num

Other Num

Account

Ring

Groups

Add Edit Search

Group Information

Groups

Ring

Add Edit Del Delete All

Please select Contacts file

Parcourir...

Import XML Export XML

Parcourir...

Import CSV Export CSV

Show title

NOTE

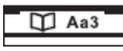
Add Contact
Fill in contact information. User should not leave contact name blank.

Delete Contact
Select the contact in the grid you want to delete, and then press the Delete button to submit.

Import
Browse and upload a file in XML format.

Export
Export button downloads Contacts into an xml file.

To search a contact via Phone interface:

Press the  key to enter the contact list.

Enter the 1st letter of the contact name and press OK button. Contact list will be searched for matching records.

Import/Export Contact list

Import/Export Contact List via Web interface:

Choose Contacts->Local Phone Book.

ALCATEL **CONTACTS** All Contacts ? Hangup

Index	Name	Office Num	Mobile Num	Other Num	Account	Groups
1	daniel	513			Auto	
2	George	514			Auto	
3	jose	123456			Auto	
4	tja	632			Auto	
5	wad				Auto	
6	wmt				Auto	
7						
8						
9						
10						

Page: 1 Prev Next Move To BlackList Delete All Del

Contacts
 Name:
 Number:
 Mobile Num:
 Other Num:
 Account:
 Ring:
 Groups:
Add Edit Search

Group Information
 Groups:
 Ring:
Add Edit Del Delete All

Please select Contacts file

Import XML Export XML

Import CSV Export CSV
 Show title

NOTE
Add Contact
 Fill in contact information. User should not leave contact name blank.
Delete Contact
 Select the contact in the grid you want to delete, and then press the Delete button to submit.
Import
 Browse and upload a file in XML format.
Export
 Export button downloads Contacts into an xml file.

Browse to select your contact list file in .XML or .csv format, and then click Import button. The imported contact lists will be shown in the directory.

Click the Export button to export the contact list.

Note:

- 5) Import/Export Contact List can be only set via Web interface.
- 6) You can export your contact list to get a template you will later use to import

Basic Call Functions

Outgoing calls

Call Devices

You can make a phone call via the following devices:

Pick up the handset,  icon will be shown on the idle screen.

Press  button, icon  will be shown on the idle screen.

Press  button if the headset is connected to the Headset Port. The icon  will be shown on the idle screen. This is a toggle key. Pressing it again will unselect headset as audio device.

Note:

- 1.-You can dial the number first, and then choose the device you will use for your call.
- 2.-If headset mode is on, you will need to press OK or end of dialing key to launch your calls

Call Methods

From phone interface:

There are several ways to initiate a call:

Press an available line button and dial the number you want to call, or

Dial a number and select the audio device (remember that if you have chosen headset, you will have to press OK)

Go to Directory, use the navigation keys to select the desired contact, or

Press  key to enter the call log interface, then use the navigation keys to make your selection (press Left/Right button to chose All Calls, Missed Calls, Dialed Calls, Received Calls and Forwarded Calls) or

Press  button to enter the Dialed Calls interface, then choose a record to dial out or

Press a Line key which has been set as speed dial button or

Long press on a digit key that has been set as speed dial key

Then press the  button to dial out if necessary.

To terminate a call you can:

- 7) press the  key or
- 8) If you are in handset mode, replace the handset on hook or
- 9) If you are in handsfree mode, press the SPEAKER key

Note:

Please see Live Dialpad, Dial plan and End of dialing key sections for more possibilities in terms of launching the call

Click- to-dial using web interface:

Go to Contact->Local Phone Book/BlackList and click on the number you want to call. . The phone will dial out automatically. Default account will be used.

ALCATEL ATLINKS

Contacts: All Contacts

Index	Name	Office Num	Mobile Num	Other Num	Account	Groups
1	daniel	513			Auto	
2	George	514			Auto	
3	jose	123456			Auto	
4	wad				Auto	
5	wmt				Auto	
6						
7						
8						
9						
10						

Page: 1

Buttons: Prev, Next, Move To BlackList, Delete All, Del

Contacts

Name:

Number:

Mobile Num:

Other Num:

Account: Auto

Ring: Auto

Groups: Not In Group

Buttons: Add, Edit, Search

Group Information

Groups:

Ring: Auto

Buttons: Add, Edit, Del, Delete All

Please select Contacts file

Buttons: Parcourir..., Import XML, Export XML, Import CSV, Export CSV

Show title

NOTE

Add Contact
Fill in contact information. User should not leave contact name blank.

Delete Contact
Select the contact in the grid you want to delete, and then press the Delete button to submit.

Import
Browse and upload a file in XML format.

Export
Export button downloads Contacts into an xml file.

Or choose Contact->Phone Call Info, enter the number in the Dial a Number box and select the line from the Outgoing Identity list. Then click the dial button to call out.

Or choose Contact->Phone Call Info, click the number which you want to dial out from the call list, the phone will dial out using default account.

The screenshot shows the Alcatel-Lucent web interface. At the top, there are navigation tabs: Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below these are links for Local Phone Book, Black List, and Phone Call Info. The main area is titled 'Call Panel' and includes a 'Dial a Number' input field, a 'Dial' button, and a 'Hangup' button. The 'Outgoing Identity' is set to '511@10.24.20.9'. Below this are four call lists: Dialed List, Missed List, Received List, and Forwarded List. Each list has columns for Index, Date, Time, Local Identity, Name, and Number. The Dialed List shows six entries with names like 'tja', 'George', 'jose', and 'George'. The Missed List shows three entries with names 'John', 'Ringo', and 'daniel'. The Received List shows three entries, all with the name 'George'. The Forwarded List is currently empty.

To terminate calls initiated with click-to-dial you can either use phone interface or click the Hang up button in the web page.

Live Dialpad

This setting defines whether dialed digits will be sent out automatically after a timeout or not. In that case it will be necessary to press OK or End of dialing key for dialing process to be completed.

This function can only be set via the Web interface, with admin/var rights:

- 1) Go to Phone->Preference->Live Dialpad.
- 2) Enable or disable it in the pull-down menu.
- 3) Click Confirm button to save the change.

Send Key configuration

Admins can set a specific button (“#” or “*”) as the send key. If this feature is enabled, once you’ve dialed the desired number, you can press this send key to call your correspondent immediately.

To set the send key via the IP phone interface:

- 1) Make sure active User Type is admin or var
- 2) Press ->2.**Features**->4.**Send Key**-and press OK button to enter the configuration page.

1. Send Key:

< * # >

- 3) Press the Left/Right navigation keys to choose a button that you want to use as the send key: "#", "*", or disable this option. Note that currently active option has an "*" on the left
- 4) Press the OK button to save the changes, or MENU to discard.

To set the send key via the Web interface:

- 1) Choose Phone->Features-> Send Key.
- 2) Highlight your desired option in the pull-down menu, then click confirm button to save your change, or cancel to discard.

The screenshot shows the Alcatel-Lucent web interface for configuring the 'Send Key' feature. The interface includes a navigation bar with tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. The 'Phone' tab is selected, and the 'Features' sub-tab is active. The 'Send Key' feature is configured with the following settings:

Feature	On	Off
Always	<input type="checkbox"/>	<input checked="" type="checkbox"/>
Target	<input type="text"/>	?
On Code	<input type="text"/>	?
Off Code	<input type="text"/>	?
Busy	<input type="checkbox"/>	<input checked="" type="checkbox"/>
Target	<input type="text"/>	?
On Code	<input type="text"/>	?
Off Code	<input type="text"/>	?
No Answer	<input type="checkbox"/>	<input checked="" type="checkbox"/>
After Ring Time(seconds)	10	?
Target	<input type="text"/>	?
On Code	<input type="text"/>	?
Off Code	<input type="text"/>	?

General Information:

Max Simultaneous Calls	3
Call Waiting	Enabled
Call Waiting Tone	Enabled
Auto Redial	Disabled
Send Key	#
Reserve # in User Name	#
Button Sound	Enabled
Send Sound	Enabled
Hotline Number	
Hotline Delay	4

NOTE:

- Forward:** Allows you to divert incoming calls to another phone number, using SIP responses or feature codes.
- Target:** Number to which incoming calls will be forwarded.
- On Code:** Code that will be sent to the sip server to activate this feature.
- Off Code:** Code that will be sent to the sip server to deactivate this feature.
- Call Waiting:** If Call Waiting is enabled your phone will be capable to accept other incoming calls during conversation.
- Send Key:** Select * or # as the end of dialing (send) key.
- Hotline Number:** Hot line number will be dialed.

Password dial

This feature is dedicated to environments in which a password or confidential billing code has to be dialed to access other extensions or external numbers.

In order to keep this code confidential it will be masked on the screen if dialed digits start with a certain prefix. When an entered number is beginning with the password prefix (PswPrefix field), the N subsequent numbers will be hidden as *, where N stands for the value entered in the PswLength field. For example: if PswPrefix is set to 3, and PswLength is 2, when you dial 34567 the display will show 3**67.

Set password dial via web interface:

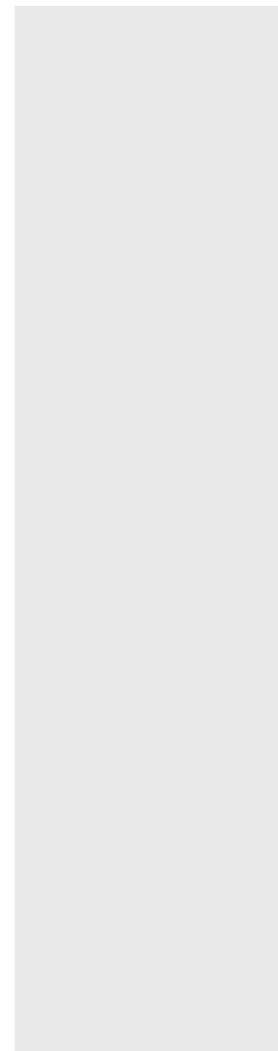
Choose phone->features->PswDial, in the pull-down menu, choose enable.

Enter the password prefix in the PswPrefix field

Enter the Length in the PswLength field.

Click the confirm box to save the changes.

Intercom Mute	Disabled	?
Intercom Tone	Enabled	?
Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attended Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	1	
ACD Auto Available	Disabled	
ACD Auto Available Timer (0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
Process 180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength		
PswDial	Disabled	
SaveCallHistory	Enabled	
Use Logo	Enabled	?
Text Logo	-Orange Group-	



Call Completion

Call completion on Busy Subscriber (CCBS) , is a supplementary service which allows a user to be notified when a busy called party has become available.

Should you enable this feature, it will be invoked whenever a called party returns a busy response code.

You need admin or VAR rights to enable/disable this feature.

Your IP PBX or server needs to allow this service.

To configure Call Completion via phone interface:

Make sure the active User Type is admin or var. If not, please change user mode as explained in "Configuration and Registration" chapter.

Press ->**2.Features**->**10.Call Completion** to enter the configuration page.

Use Left/Right navigation keys to choose Enable/disable. Note the active value will be marked with an "*".

Press the OK to save your changes.

To configure Call Completion via phone interface:

10)Go to Phone > Features > Call Completion

11)Select desired status (Enabled/disabled) and press Confirm to save your changes.

Auto Redial

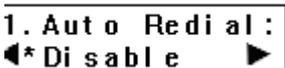
Auto redial is a feature that automatically redials a busy number a fixed number of times before giving up.

Purpose is similar to Call Completion, but the implementation is purely local and hence is available in all scenarios without server support.

To set auto redial via the IP phone interface:

1) Make sure active User Type is admin or var

2) Press ->**2.Features**->**6.Auto Redial**-> OK button to enter the configuration page.



3) Use left and right navigation keys to enable/disable the auto redial function.

4) Press down navigation key to set redial interval. It is measured in seconds.

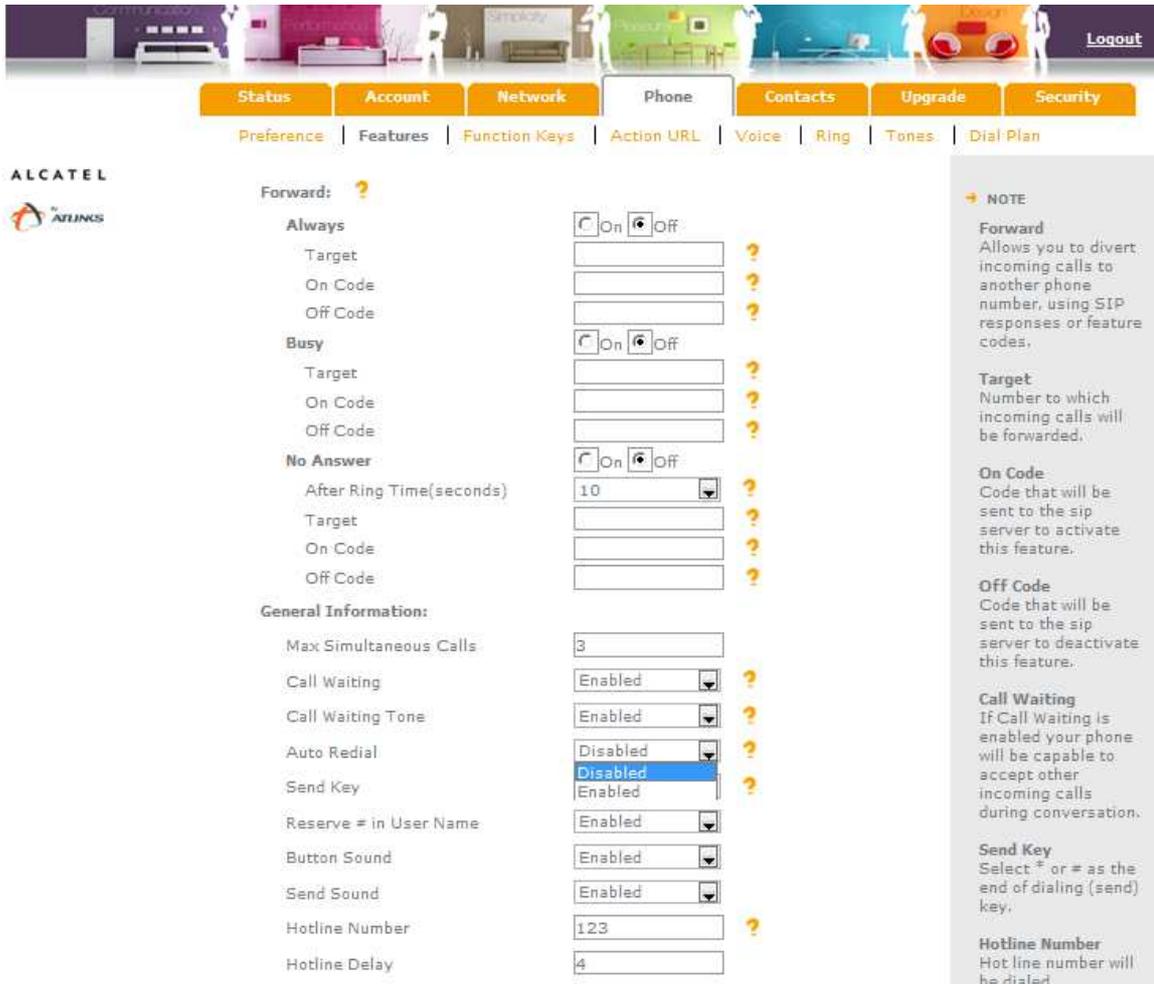
- 5) Press down navigation key to set the number of times redial will take place.
- 6) Press OK button to save the changes.

Note:

When Auto redial is enabled, if your phone receives a Busy response for an outgoing call, you will be prompted if you wish to Autore dial or not. If you press Cancel softkey or do not press any keys for 5 seconds the phone will return to idle interface and Auto redial will not take place.

To set auto redial via the Web interface:

- 1) Go to Phone->Features->Auto Redial.
- 2) Choose Enabled or Disabled in the pull-down menu
- 3) Click Confirm button to save your changes or Cancel to discard.



Hot Line

Hot Line allows your phone to dial a number automatically if it has been off hook for a specified time period and no keys have been pressed. Both destination number and timeout can be configured.

The service is enabled if Hot Line Number has been configured.

To set the hot line number via the IP phone interface:

Press ->2.**Features**->5.**Hot Line** OK button to enter the configuration page.

1. Hot Number : 123

Enter the hot line number and press up/down navigation key to select HotLine Delay. Set the Hot Line Delay (default is 4seconds), then press the OK button to save the changes.

To set the Hot Line via the Web interface:

Go to Phone->Features.

Input the Hotline Number and Hotline Delay, and then click Confirm button to save your changes.

The screenshot shows the Alcatel SIP configuration web interface. The 'Phone' tab is selected, and the 'Features' sub-tab is active. The 'Forward' section is expanded, showing settings for 'Always', 'Busy', and 'No Answer' call forwarding. The 'General Information' section is also visible, with 'Hotline Number' and 'Hotline Delay' fields highlighted by a blue box. A 'NOTE' sidebar on the right provides detailed explanations for various features like Forward, Target, On Code, Off Code, Call Waiting, Send Key, and Hotline Number.

Anonymous call

Anonymous call, also known as Caller ID restriction, allows you to request the network to hide your subscriber identity when you are performing outgoing calls.

Temporis IP200 supports Permanent Caller ID restriction, meaning that while enabled all calls will request to be anonymous and you need to disable if you want your identity to be transmitted. This service can be performed locally, or by sending an activation code to the network.

Additionally Temporis IP200 allows you to automatically reject anonymous incoming calls.

These services can be managed through both phone and Web interfaces as ar

To set Anonymous call via phone interface:

- 1) Press  -> **2.Features**->**5.Anonym Call**->OK button to enter the configuration page.

1. Line ID:
◀ Line 1 ▶

- 2) Use left or right navigation keys to select the account to be anonymous, you can choose the Line ID.
- 3) Press down navigation key to display Anonym call, then press left or right navigation keys to enable/disable the anonymous call function.
- 4) If you want to perform this function locally, press OK to save your changes.
- 5) If you would like to activate this function in your service provider please press down key and enter the Anonym On Code and Anonym Off Code. Whenever you enable/disable the anonymous call function on your phone, it will automatically send these codes to the server.
- 6) Press OK to save your changes, or MENU to cancel

To set Anonymous call rejection using phone interface:

- 1) Enter Anonymous call menu as explained above
- 2) Press the up or down navigation keys to enter the Rejection page
- 3) Press right or left navigation keys to enable/disable Anonymous Call Rejection function.
- 4) If you would like to use your service provider to get this service, please press the down key and enter the Reject On Code and Reject Off Code. Whenever you enable/disable the Rejection option on your IP phone, it will automatically send these codes to the server.

Note:

1. To manage Anonymous Calls and Rejection codes you need to have admin or var rights. Check active User Type before attempting to configure using phone interface.
2. Anonymous Call Rejection is only available for the current default account.

Press OK to save your changes, or MENU to cancel.

To set Anonymous Call Services using the Web interface:

- 1) Go to Account-> Basic-> Anonymous Call and perform necessary changes. Please refer to the instructions above for details.
- 2) Click the Confirm button to save the changes, or Cancel to discard.

ALCATEL
By **ATLINKS**

Account: Account 1

Basic >>

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	Ext 511	?
Display Name	John	?
Register Name	511	?
User Name	511	?
Password	*****	?
SIP Server	10.24.20.91	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060 ?
NAT Traversal	Disabled	?
STUN Server		Port 3478 ?
Voice Mail	*2	?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?
On Code		?
Off Code		?
Missed call log	Enabled	?
Auto Answer	Disabled	?
Ring Type	common	?

Codecs >> ?

Advanced >>

Confirm Cancel

NOTE

Display Name
SIP service subscriber name used for Caller ID display.

Register Name
SIP service subscriber ID used for authentication.

User Name
User account provided by VoIP service provider or PBX admin.

NAT Traversal
Control STUN server settings.

Proxy Require
Relevant for Nortel server only. If you wish to login to a Nortel server, value should be: com.nortelnetworks.firewall

Codex
Select codec types and priority.

Advanced
Administrator oriented parameters.

Dial Plan

A dial plan establishes the expected number and pattern of digits for a telephone number. This includes country codes, access codes, area codes and all combinations of dialed digits.

Dial plans can be used for least cost routing, dialing convenience or to prevent users from calling specific numbers or groups of numbers.

Temporis IP 200 supports various types of elements related to dial plan: replace rules, dial now entries, area code management and forbidden numbers. All of them require admn or var privilege.

Replace Rule

A replace rule is a dial plan item which enables automatic replacement of a digit string by another. For example if you set the *Prefix* as 0 and *Replace* as 0033 (France country code), when you dial 0670200030 out, the number will be replaced by 0033670200030 automatically.

To set a Replace Rule via the Web interface:

- 1) Choose Phone->Dial Plan->Replace Rule.



The screenshot shows the Alcatel web interface for configuring a Replace Rule. The navigation menu includes Status, Account, Network, Phone, Contacts, Upgrade, and Security. Under the Phone menu, there are sub-menus for Preference, Features, Function Keys, Action URL, Voice, Ring, Tones, and Dial Plan. The 'Replace Rule' page is displayed, showing a table with columns for Index, Prefix, Replace, and Account. Below the table are input fields for Prefix, Replace, and Account, and buttons for Add, Edit, and Del. A 'NOTE' section on the right explains the syntax for digits and variables.

Index	Prefix	Replace	Account
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			

Prefix Replace Account

NOTE

- Digit 0-9 *
Identifies a specific digit (do not use # if it is defined as send key).
- [digit-digit]
Identifies any digit dialed included in the range.
- [digit-digit,digit]
Specifies a range as a comma-separated list.
- x
Matches any single digit/character.
- *
Matches an arbitrary number of digits.

- 2) Enter the desired *Prefix*, *Replace* string and *Account* through which the call should be sent.
- 3) Press Add button to save the changes.
- 4) You can also delete a specific item from the list by pressing Del button.
- 5) You can select a record to modify, then click Edit button to submit.

Note:

If you need to replace unknown contents you can use (.) or (x) in Prefix field, where "." stands for a string of char, and "x" stands for any single char.

Additionally Prefix content in () will be considered as a variable.

In the Replace field, the first variable is expressed by \$1, the second variable is expressed as \$2, etc.

For example: if you want to replace any number of dialed digits with the same digits but pre-pending an 8, input (.) in Prefix box, and input 8\$1 in Replace box.

Dial Now

Dial-now entries define digit patterns which should be dialed out immediately, without the need for an end or dialing key press.

For example, if a dial now entry is 5xx, when a user dials 534 the call will be sent automatically to the server without waiting for further digits or for a send key or OK key press.

To set a Dial Now using the Web interface:

- 1) Choose Phone->Dial Plan->Dial Now.

The screenshot shows the Alcatel web interface for configuring Dial Now rules. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. Below this, there are sub-navigation tabs: 'Preference', 'Features', 'Function Keys', 'Action URL', 'Voice', 'Ring', 'Tones', and 'Dial Plan'. The main content area is titled 'Replace Rule >>' and 'Dial-now >>'. It features a table with columns for 'Index', 'Dial-now Rule', and 'Account'. The table has 10 rows, with the first row containing the number '1'. Below the table, there are input fields for 'Dial-now Rule' and 'Account', and buttons for 'Add', 'Edit', and 'Del'. A 'NOTE' section on the right provides definitions for symbols used in the rules: 'Digit 0-9 *' (identifies a specific digit), '[digit-digit]' (identifies any digit dialed included in the range), '[digit-digit,digit]' (specifies a range as a comma-separated list), 'x' (matches any single digit/character), and '*' (matches an arbitrary number of digits).

- 2) Enter the digit pattern Dial-now Rule. Fill in the account to be used to send the call.
- 3) Press Add button to save the changes.
- 4) You can select a record to modify, then click Edit button to submit.
- 5) You can also delete a selected entry from the Dial Now list by pressing Del button.

Note:

A timeout will be waited for before actually sending dialed digits matching the pattern. This timeout can be configured in Phone->Features->Timeout for dial now rule

Area Code

Area codes are also known as Numbering Plan Areas (NPAs). These are necessary (for the most part) only when dialed from outside the code area and from mobile phones. Area codes usually indicate geographical areas within one country, although the correlation to geographical area is becoming obsolete. For non-geographical numbers, as well as mobile telephones outside of the United States and Canada, the "area code" does not correlate to a particular geographic area.

To add the area code via the Web interface:

- 1) Choose Phone->Dial Plan->Area Code.

- 2) Enter the Code, set the Min Length and the Max Length options, the account to be used and then click the Confirm button to save.

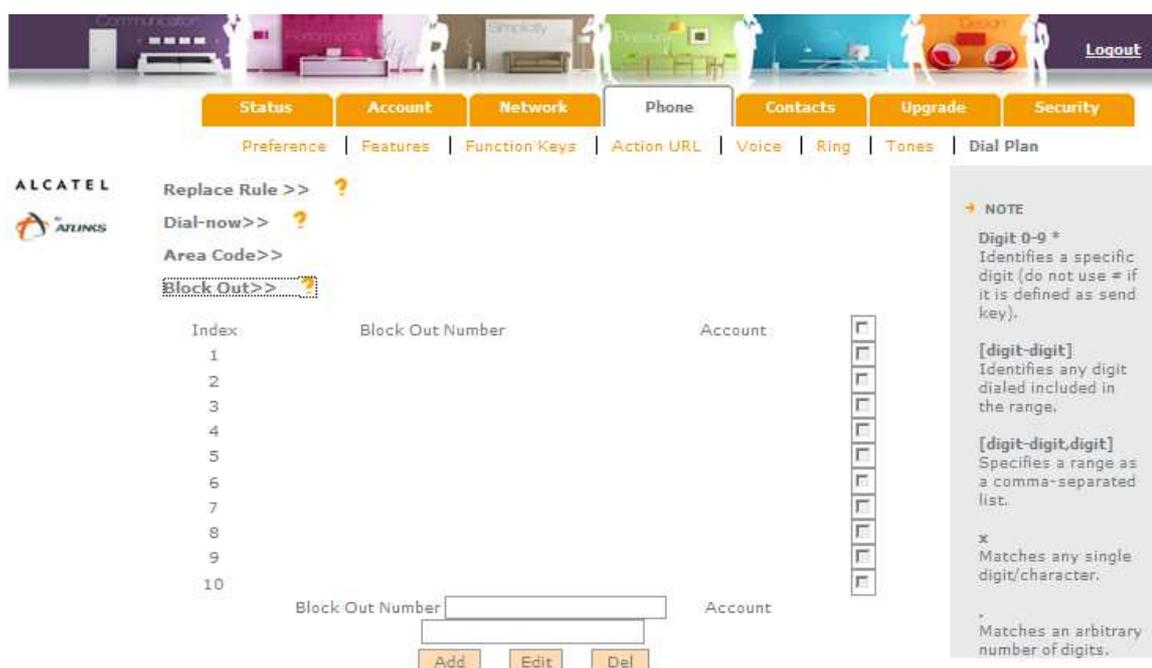
Forbidden Numbers (Block Out)

Specific phone numbers or digit patterns can be forbidden to be called out from your phone.

To set Block Out via the Web interface

- 1) Choose Phone->Dial Plan->Block Out.
- 2) Enter the specific digit pattern, and click Add button to save the changes, or choose a specific entry in the list, click Del button to delete the record.
- 3) You can select a record to modify, then click Edit button to submit.

Once a number or digit pattern is included in the Block out list, it will not be possible to dial out a number matching that entry or pattern.



Note:

1. In the Account field, you can enter 1,2,3..., where "1" represents Account 1, "2" represents Account, etc
2. 2, if the account box is empty, this rule will be applicable to all accounts .

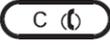
Incoming calls

Answering an incoming call

- 1) Lift the handset or press  button to answer using the speakerphone, or press  button to answer using the headset.
- 2) If you are already talking on the phone, press the corresponding line key or OK button to answer the new call.

During the conversation, you can alternate between Headset, Handset and Speaker phone by pressing the corresponding buttons or picking up the handset.

Rejecting an incoming call

Press the  key to reject an incoming call directly.

DND

When DND function is enabled, all the incoming calls will be rejected and the display shows **DND** icon; you can find the incoming call record in the Call History.

If you press CONF key in idle mode, you activate or deactivate the DND function.

To configure the DND function via Phone interface:

- 1) Go to  -> **2.Features**->**8.DND** and press OK button to enter the configuration interface.
- 2) Use the right or left navigation keys to choose Enable/Disable.
- 3) Press the OK button to save the changes.
- 4) Choose DND option again to deactivate DND mode.

You can also DND function using DND Codes if your IPPBX or server provide this function:

- 1) Go to MENU->**2.Features**->**8.DND** and press OK button to enter the configuration interface.
- 2) Set the DND On Code and the DND Off Code by pressing up or down navigation keys, then press the OK button to save the changes.
- 3) When you select to enable/disable the DND function on the phone, it will send these

codes to the server, and the server will turn on/off the DND function. In this case any calls addressed to the phone's extension will be rejected by the server automatically, and will not be included in the Call Log.

Note:

When DND is locally enabled (not using codes), the corresponding icon will be shown on the display

Call Forward

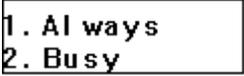
This feature allows you to divert an incoming call to another phone number e.g. a cell phone or voice mailbox.

The following call forwarding events can be configured:

- *Always:* Incoming calls are immediately forwarded.
- *Busy:* Incoming calls are immediately forwarded when the phone is busy.
- *No Answer:* Incoming calls are forwarded when the phone is not answered after a specific time period.

To configure Call Forward via Phone interface:

- 1) Go to ->2.**Features**->1.**Forward**. You can also press  button or a previously configured programmable key to enter the forward setting page directly when the phone is in idle mode.



1. **Always**
2. **Busy**

- 2) There are 3 options: **Always**, **Busy** and **No Answer**.
- 3) Select one of them with OK. Then enter the phone number to which calls must be forwarded. For No answer you can also select the No answer timeout pressing the down key.
- 4) If you prefer the server to perform this function, please enter the On Code and Off Code options, then when you choose to enable the call forward function via your IP phone, it will send the corresponding code to the server, which will take care of forwarding your calls. In this case forwarded calls will not be stored in the Call Log.
- 5) Press the OK button to save the changes.

Notes:

1. When you get an incoming call, if you press  during ringing you can dynamically forward the call to a desired number.
2. You can quickly enable/disable Forward by pressing down key or Transfer key from idle mode. Forward type and target number will be the last ones configured using the menu.
3. Only admin/var can set the On/off codes. Make sure the right User Type is active.
4. When Call forward is locally enabled (not through codes), the corresponding icon will be shown on the display

To configure Call Forward via Web interface:

Go to Phone->Features->Forward to do necessary changes. Please refer to above configuration information.



DND/FWD Synchronisation

This feature requires specific server support. Check with your service provider.

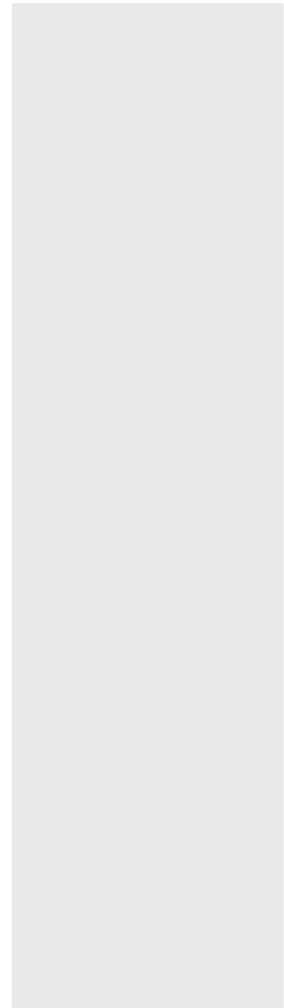
When this function is enabled, DND/FWD status on device and server will be in correspondence.

To set Feature Key Synchronisation via the Web interface:

- 1) Go to Phone->Features-> Feature Key Synchronisation
- 2) Choose whether to enable this function from the pull-down menu.

3) Click Confirm to save your change, or Cancel to discard.

Intercom Tone	Enabled	?
Intercom Barge	Enabled	?
Call Completion	Disabled	?
Enable Semi-Attended Transfer	Enabled	?
Blind Transfer OnHook	Enabled	
Attend Trans OnHook	Enabled	
Transfer on Conference Hang up	Disabled	
Feature Key Synchronisation	Disabled	
Time Out for Dial-now Rule	Disabled Enabled	
ACD Auto Available	Disabled	
ACD Auto Available Timer (0~120s)	60	
RFC 2543 Hold	Disabled	
Use Outbound Proxy In Dialog	Enabled	
Process 180	Enabled	
Logon Wizard	Disabled	
PswPrefix		
PswLength		
PswDial	Disabled	
SaveCallHistory	Enabled	
Use Logo	Enabled	?
Text Logo	-Orange Group-	



Intercom

Intercom mode is useful in an office environment as a quick access to connect to the operator or the secretary. In an intercom call the callee's phone will answer automatically without user intervention.

Since intercom is based on some headers in the sip messages you need your server to either support or at least allow (pass through) this feature .

Your set can be configured to accept different levels of Intercom, either through phone or Web interfaces.

To configure Intercom option via phone interface:

- 1) Go to ->2.**Features**->7.**Intercom**->OK to enter the configuration page.
Press up or down navigation keys to display :

-
- 2) Intercom Allow: To set whether to answer the incoming intercom calls.
 - 3) Intercom Mute: To set whether to mute the incoming intercom calls automatically.
 - 4) Intercom Tone: To set whether to play ring tones when there is incoming intercom calls to your extension.
 - 5) Intercom Barge: To set whether to answer the incoming intercom calls during a conversation. If the option is enabled, when there are incoming intercom calls to your extension, if you are on an intercom conversation, it will refuse the call automatically; or it will put the current call on hold and put the incoming intercom call through.
 - 6) Make your selection and use left or right navigation keys to enable or disable the feature.
 - 7) Press the OK button to save your changes.

To configure Intercom settings using the web interface:

- 1) Go to Phone->Features and set Intercom related parameters
- 2) Press Confirm to save your changes or Delete to discard.

Note:

To initiate an Intercom call you can either use a function key or speed dial key or directly type a code supported by your PBX. Please see Key Configuration chapter.

Auto Answer

Auto answer allows an incoming call to be answered without requiring any action by the user. This is a useful feature for people who have some kind of impairment. This function is set on a per account basis.

This function should be enabled/disabled by admins or vars.

To set Auto Answer via the IP phone interface:

- 1) Make sure active User type is var or admin.
- 2) Press ->3.**Settings**->2.**Advanced**, enter the password and press OK button
- 3) Then choose 1.**Accounts**->Line X to enter the configuration page, use the up and down navigation keys to choose Auto Answer option.



- 4) Use the left or right navigation keys to enable or disable the auto answer function. The default is **Disable**.

5) Press the OK button to save the changes.

To set Auto Answer via Web interface:

- 1) Go to Account->Basic->Auto Answer option.
- 2) Select Enabled or Disabled in the pull-down menu, click Confirm button to save the change.

ALCATEL
Account: Account 1

Basic >>

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	Ext 511	?
Display Name	John	?
Register Name	511	?
User Name	511	?
Password	*****	?
SIP Server	10.24.20.91	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server		Port 5060 ?
NAT Traversal	Disabled	?
STUN Server		Port 3478 ?
Voice Mail	*2	?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Rejection	Off	?
On Code		?
Off Code		?
Missed call log	Enabled	?
Auto Answer	Disabled	?
Ring Type	Enabled	?

Codecs >> ?

Advanced >>

Confirm Cancel

NOTE

- Display Name**
SIP service subscriber name used for Caller ID display.
- Register Name**
SIP service subscriber ID used for authentication.
- User Name**
User account provided by VoIP service provider or PBX admin.
- NAT Traversal**
Control STUN server settings.
- Proxy Require**
Relevant for Nortel server only. If you wish to login to a Nortel server, value should be: com.nortelnetworks.firewall
- Codecs**
Select codec types and priority.
- Advanced**
Administrator oriented parameters.

Call log

Temporis IP200 can store up to 100 calls in its Call log. There are 5 different lists:

- All calls

-
- Forwarded calls 
 - Missed calls 
 - Received calls 
 - Dialed calls 

They are accessible using the web interface as described in Contact management chapter. In the next paragraphs we will explain how to access them from the phone interface.

Accessing Call log

You can access the Call log by pressing the left key from idle mode. You will be in the "All calls" list, where calls are sorted sequentially, and each call is marked with the corresponding icon.

To change list use the right key: the sequence is All, Forwarded, Missed, Received, Dialed. Note the icons at the top of the screen which indicate the list you are on.

Accessing Missed Calls list

You can access the Missed calls list directly if, when you have a "Missed call" indication on the display, you press the OK key.

Of course you can also access Missed Calls list by pressing Left navigation key from idle mode to reach Call log and then right navigation key twice as explained above.

When you are in the Missed Calls list you should see  icon on the screen.

Accessing Dial Calls list

You can access the Dialed calls list directly by pressing the Up navigation key from idle mode.

You can also access this list by pressing Left navigation key to enter the Call Log, and then right navigation key four times as explained above.

When you are in the Missed Calls list you should see  icon on the screen.

Call list items management

When you are in a call list, you can:

- 1) Dial current entry by pressing OK key, or
- 2) Store it in Directory or Black list by pressing the Phonebook key (see Contacts Management chapter for details), or
- 3) Delete it using  key

Missed call log enable/disable

It can be defined whether issued calls will be saved in the call history record or not. This function can only be set via the Web interface:

- 1) Go to Account->Basic->Missed call log.
- 2) Choose Enabled or Disabled in the pull-down menu, click Confirm button to save the change.

During an Active Call

Mute

This function allows you to mute the microphone of phone during a call; you cannot be heard by the other party. You can still hear all other parties while mute is enabled.

To mute/resume the conversation:

Press  button during a conversation to mute the current call, the icon  will be shown on the LCD, and the power indication LED will blink. Press  again to

unmute the call. When you press  button all of the conversations will be muted.

Call Hold

This call function allows you to place an active call on hold. In this case your IP PBX might play a melody or message to the other party while waiting. Other calls can be received and made while having a call on hold.

To hold/resume a call:

Press  button to put your active call on hold.

During the call, there will be a warning tone every 30 seconds, suggesting that there is a current call in hold state.

If there is only one call on hold, press  button again to retrieve the call.

If there is more than one call on hold, press the Up/Down navigation keys to select the desired call, then press  button again to retrieve the call.

Note:

When you are under the call hold status, putting down the handset, the conversation will go on over the speaker instead of hanging up the call.

Call Waiting

This call feature allows your phone to accept other incoming calls when there are other active calls.

Handling a waiting call

By default Call Waiting is enabled, and when an incoming call is received during another active call a warning tone will be played periodically.

In this situation, user can:

- 1) Accept the second call using the OK key. The first one will be put on hold automatically. Or
- 2) Reject the call using  button

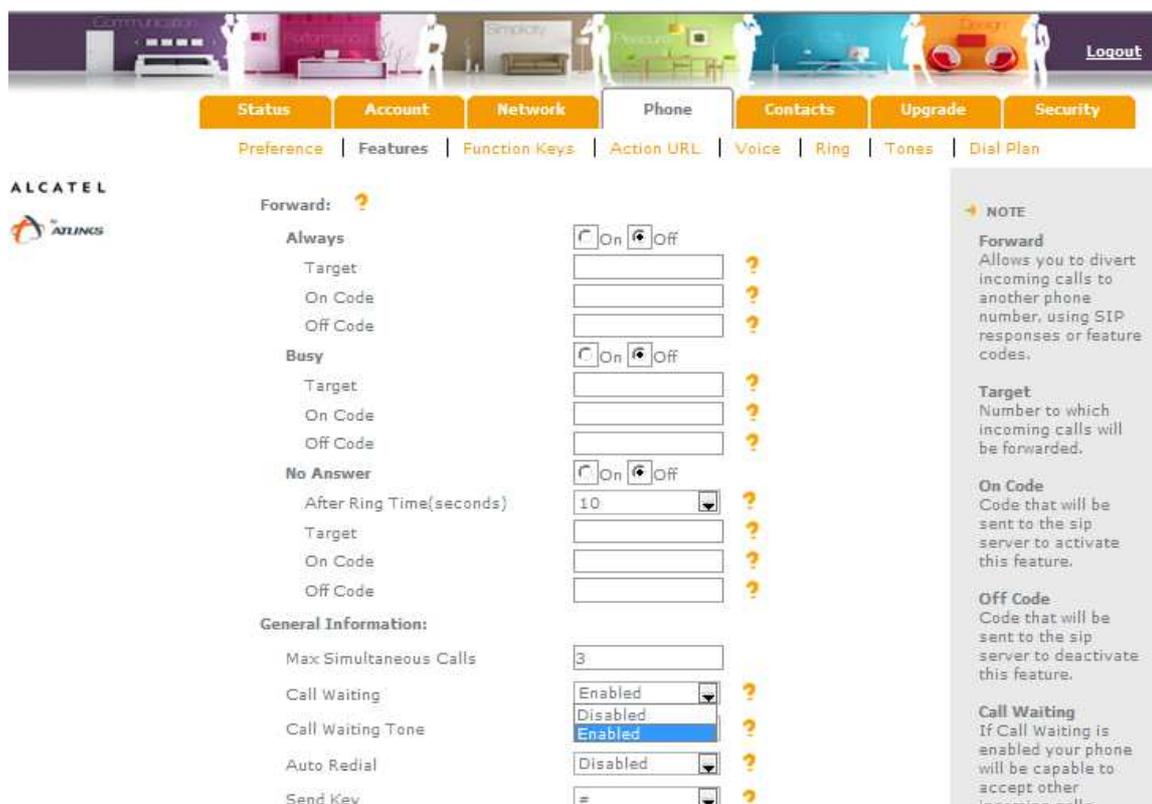
Call Waiting Configuration

To configure Call Waiting via Phone interface:

- 1) Make sure active User Type is admin or var
- 2) Press ->2.**Features**->2.**Call Waiting**.
- 3) Use the left or right navigation keys to enable/disable call waiting feature.
- 4) Use the navigation keys to enable/disable the **Play Tone** option. This option is used to define whether to play a warning tone when there is an incoming call during an active call.
- 5) Press OK button to save the changes, or MENU to return to the previous menu.

To configure Call Waiting via Web interface:

Go to Phone->Features->Call Waiting and perform necessary changes. Refer to previous paragraphs for parameter details.



ALCATEL
ATLINKS

Forward: ?

Always On Off

Target: ?

On Code: ?

Off Code: ?

Busy On Off

Target: ?

On Code: ?

Off Code: ?

No Answer On Off

After Ring Time(seconds): 10 ?

Target: ?

On Code: ?

Off Code: ?

General Information:

Max Simultaneous Calls: 3 ?

Call Waiting: Enabled ?

Call Waiting Tone: Enabled ?

Auto Redial: Disabled ?

Send Key: # ?

NOTE

Forward
Allows you to divert incoming calls to another phone number, using SIP responses or feature codes.

Target
Number to which incoming calls will be forwarded.

On Code
Code that will be sent to the sip server to activate this feature.

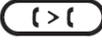
Off Code
Code that will be sent to the sip server to deactivate this feature.

Call Waiting
If Call Waiting is enabled your phone will be capable to accept other incoming calls.

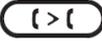
Call Transfer

This call function allows your phone to transfer an active call to another phone : Blind Transfer, Attended Transfer and Semi-Attend Transfer are supported.

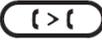
To make a Blind Transfer via phone interface (no prior announcement of the call):

- 1) Press  Button during an active call, the call is put on hold.
- 2) Dial the second phone number and press  Button or hang up to complete the transfer.

To make an Attended Transfer via phone interface (with prior announcement):

- 1) Press  button during an active call. The call is put on hold.
- 2) Dial the second phone number.
- 3) When the call is answered, press  button or hang up to complete the transfer.

To make a Semi-Attended Transfer via phone interface (no prior announcement of the call):

- 1) Press  button during an active call. The call is put on hold.
- 2) While third party's phone is ringing, press  button or hang up.

Note:

Make sure that the SIP server you have registered with supports this function.

3-way Conference

Your Temporis IP200 is able to establish a three-party conference by locally mixing the audio signals of your two correspondents.

To establish a conference via phone interface:

- 1) Once in line with 1st caller, press a line key, 1st caller is put on hold.

- 2) Dial 2nd party number.
- 3) When in line with 2nd party, press  button to establish the conference.
- 4) Hang up to disconnect all parties.

Network Conference

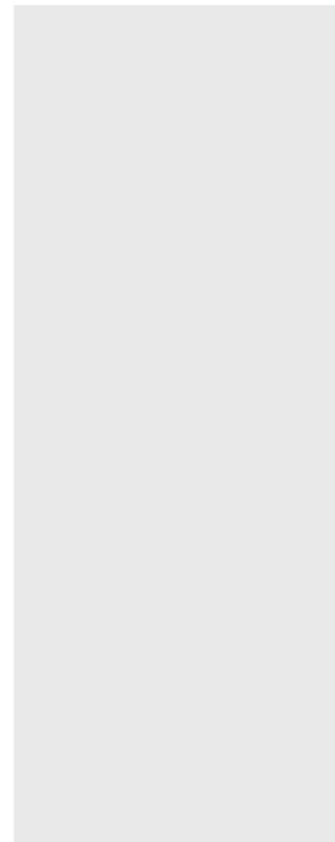
If you want to make a conference with more than three people, you can use a dial in conference server or use network conference if your service provider supports this function.

Note: server support is required. Please check with your service provider .

To enable network conference via web interface:

- 1) Go to Account->Account X->Advanced->Conference Type, there is a pull-down menu, choose network from the list.
- 2) Enter the Conference URI.
- 3) Press Confirm button to save the changes.

Session Timer(seconds)	<input type="text"/>	?
Refresher	Uac	?
Use user=phone	Disabled	?
Voice Encryption (SRTP)	<input type="checkbox"/> On <input checked="" type="checkbox"/> Off	?
ptime(ms)	20	?
BLF List URI	<input type="text"/>	?
BLF List Code	*97	
Shared Line	Disabled	?
Dialog-Info Call Pickup	Disabled	?
BLA Number	<input type="text"/>	?
BLA Subscription Period (Scope:60~7200)	300	?
SIP Send MAC	Disabled	?
SIP Send Line	Disabled	?
SIP Registration Retry Timer (Scope:0~1800)(seconds)	30	?
Enable Signal Encode	Disabled	
Signal Encode Key	<input type="text"/>	
Conference Type	Local	
Conference URI	Local Network	
ACD Subscription Period(120~3600)	3600	
Caller ID Header	FROM	?
Callee ID Header	PAI-RPID	
Trust SIP servers only	Disabled	



To establish a conference:

-
- 1) Press  key during an active call.
 - 2) Dial in the number to conference, then press the OK key
 - 3) When the call is answered, press the  button.
 - 4) After starting a three way conference, press  button to enter Conference dialing interface and invite another party to participate in teleconference.
 - 5) After starting conference, press Hold key to Hold local call without influencing others in the conference.

Voicemail

Your voice mailbox messages, which are usually stored in your local or hosted VoIP telephony system, can be accessed from your phone.

New voice messages can be indicated both acoustically and visually as described below:

- The idle screen will indicate the new voice messages.
-  button backlight will be on.
- The orange LED of the phone will be steady on
-

Voicemail settings

To configure the Voicemail code via Phone interface:

- 1) Press ->4.**Messages**-> 2.**Set Voice Mail**.
- 2) Use the navigation keys to highlight the Line you want to set, enter the number to connect to your system voicemail. You can press  to choose the proper input method.
- 3) Press OK button to save the change, or press MENU to return to the previous menu.

Note:

1. Please contact your system administrator for the connecting code. Different systems have different codes.
2. To benefit from the MWI service in function of your system you may need to subscribe to it. This setting can only be done using the web interface. Contact your system administrator or your service provider for more details.

To configure Voicemail settings using the Web interface:

- 1) Go to Account->Basic and set Voicemail number
- 2) If needed, go to Account->Advanced-> and set Subscribe for MWI and MWI Subscription Period.

Advanced >>

UDP Keep-alive Message	Enabled	?
UDP Keep-alive Interval(seconds)	30	
Login Expire(seconds)	3600	?
Local SIP Port	5060	?
RPort	Disabled	?
SIP Session Timer(seconds) T1	0,5	?
SIP Session Timer(seconds) T2	4	
SIP Session Timer(seconds) T4	5	
Subscribe Period(seconds)	1800	?
DTMF Type	RFC2833	?
SIP INFO DTMF Type	Disabled	?
DTMF Payload(Scope:96~255)	101	
100 reliable retransmission	Disabled	?
Enable Precondition	Disabled	?
Subscribe Register	Disabled	?
Subscribe for MWI	Disabled	?
MWI Subscription Period (Scope:0~84600)(seconds)	Enabled	?
Use Session Timer	Disabled	?
Session Timer(seconds)		?
Refresher	Uac	?

Register Name
SIP service subscriber ID used for authentication.

User Name
User account provided by VoIP service provider or PBX admin.

NAT Traversal
Control STUN server settings.

Proxy Require
Relevant for Nortel server only. If you wish to login to a Nortel server, value should be: com.nortelnetworks.firewall

Codecs
Select codec types and priority.

Advanced
Administrator oriented parameters.

Retrieving your Voicemail messages

To check your voicemail status via the Phone interface:

- 1) Press ->4.**Messages**->1. **View Voice Mail**->OK button.
- 2) You can view your messages (new and old messages).
- 3) Choose the account and press the Connect button , then you are able to listen to your new and old messages.

To listen to your new messages via the Phone interface:

- 1) Press  button directly.
- 2) You may be prompted to enter the password which is needed to connect to your VoIP telephony system. It depends on your system.

-
- 3) Your voice mailbox is called and you are able to listen to your new and old voicemails.

Note:

1. Before retrieving the new voicemail, please make sure that the connecting code has been set on the phone.
2. If you have more than one account set on the phone, when pressing the  key you will be calling the active account mailbox.



Advanced Functions

Account Setting

Please refer to "Configuration and Registration" chapter for the basic account setting information. The following table lists the parameters in Advanced Account Setting section of the web interface.

Field Name	Description
UDP Keep-alive Message	Defines whether to activate the phone UDP Keep-alive mechanism. The default is Enabled.
UDP Keep-alive Interval(seconds)	This parameter specifies how often the phone will send a packet to the SIP server. Default is 30 seconds.
Login Expire(seconds)	This parameter specifies the time frequency that phone refreshes its registration. The default interval is 3600 seconds.
Local SIP Port	Local SIP port. The default value is 5060.
RPort	The parameter allows you configuring the proxy to send responses back to a particular address and port. The default is disabled.
SIP Session Timer	This document defines an extension to the Session Initiation Protocol (SIP). This extension allows for a periodic refresh of SIP sessions through a re-INVITE or UPDATE request. The refresh allows both user agents and proxies to determine if the SIP session is still active.
Subscribe Period(seconds)	This parameter sets the period of the subscription. The default value is 3600.
DTMF Type	Select DTMF type.

You can only configure these settings via Web interface:

- 1) Choose Account.
- 2) Select desired account.
- 3) Choose Advanced to do necessary settings.

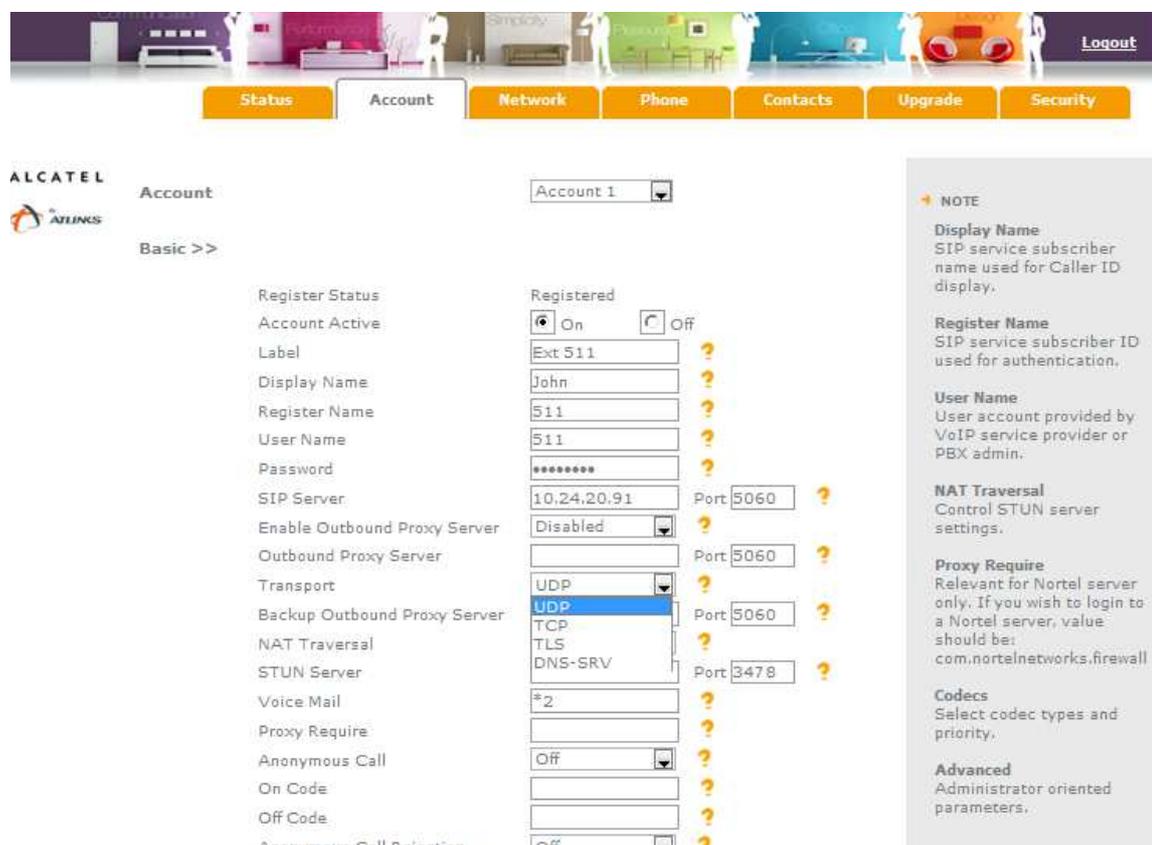
You can check with your system administrator or service provider for more information.

TLS

TLS (Transport Layer Security), an IETF standards track protocol(RFC 5246), was based on the earlier SSL specifications developed by Netscape Corporation.

If you make a call based on TLS and SRTP (Set the transport as TLS and the SRTP as On), the phone will display the connecting and ring back interface. If the other party accepts to set the transport as TLS, then the phone will display the icon .

Go to Account->Basic, choose Transport option, in the pull-down menu, you can choose the TLS option, and then click the Confirm button to save the change.



ALCATEL
Account 1

Basic >>

Register Status	Registered	
Account Active	<input checked="" type="radio"/> On <input type="radio"/> Off	
Label	Ext 511	?
Display Name	John	?
Register Name	511	?
User Name	511	?
Password	*****	?
SIP Server	10.24.20.91	Port 5060 ?
Enable Outbound Proxy Server	Disabled	?
Outbound Proxy Server		Port 5060 ?
Transport	UDP	?
Backup Outbound Proxy Server	UDP	Port 5060 ?
NAT Traversal	TCP	?
STUN Server	TLS	?
Voice Mail	*2	?
Proxy Require		?
Anonymous Call	Off	?
On Code		?
Off Code		?
Anonymous Call Protection	Off	?

NOTE

Display Name
SIP service subscriber name used for Caller ID display.

Register Name
SIP service subscriber ID used for authentication.

User Name
User account provided by VoIP service provider or PBX admin.

NAT Traversal
Control STUN server settings.

Proxy Require
Relevant for Nortel server only. If you wish to login to a Nortel server, value should be: com.nortelnetworks.firewall

Codecs
Select codec types and priority.

Advanced
Administrator oriented parameters.

DNS-SRV

DNS-SRV option allows the SIP phone to discover what are the available servers (IP and Transport and priority) for the specified SIP server domain.

To set DNS-SRV via web interface:

Go to Account->Basic-> choose Transport option, in the pull-down menu, you can choose the DNS-SRV option, and then click the Confirm button to save the change.

Network Settings

Please refer to chapter “Configuration and Registration” for the Basic Network WAN setting information. In the next paragraphs we will describe other Network related settings: PC Port and Advanced.

PC Port Setting

The following table lists PC Port settings.

Field Name	Description
As Bridge	If you select the Bridge mode, then the two Fast Ethernet ports will be transparent.
As Router	If you select the Router mode, the phone will work as a router.
IP address	Admin can configure the phone’s PC port IP address.
Enable DHCP Server	If you set the DHCP server on, the device connected to the PC port will get the IP address automatically between the start IP address and the end IP address. But if you select the bridge mode, the DHCP

	server cannot be enabled.
Start IP Address	DHCP server pool start IP address.
End IP Address	DHCP server pool end IP address.

To configure PC Port settings via Phone interface:

- 1) Make sure active User type is admin or var
- 2) Press ->3.**Settings**->2.**Advanced**->OK.
- 3) Enter the password required, scroll to **Network** option, press OK button and select **PC Port** option, then press OK button to enter.

1. Bridge * 2. Router
--

- 4) If you choose **Bridge**, it will return to the previous menu.
- 5) If you choose **Router**, you will be prompted to enter the IP, Subnet Mask, DHCP Server Disable/Enable.
- 6) Press OK button to save the changes, or MENU to return to the previous menu.

To configure PC Port via Web interface:

Choose Network->PC Port to do necessary changes. Please note start and end IP address can only be set via Web interface. Check with your system administrator for more information.

Advanced Network Settings

LLDP

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral Layer 2 protocol that allows a network device to advertise its identity and capabilities on the local network.

If LLDP function is enabled the phone will go to try a get related VLAN parameters automatically from the switch. (Synchronous with VLAN in switch)

To configure LLDP settings via Web interface:

Go to Network->Advanced->LLDP->Active option. In the pull-down menu, choose Enable.

Then enter the corresponding Packet Interval in Packet Interval field.

You can also disable this function when you choose disable in active field.

Click Confirm button to save the change.

The screenshot shows the configuration page for VLAN settings. The 'Active' dropdown menu is open, showing options: Disabled, Enabled, Disabled (Scope: 1~3600s). The 'VLAN' section includes settings for WAN Port and PC Port, both with Active set to Disabled, VID set to 0, and USR PRIORITY set to 0. The 'Port Link' section shows WAN Port Link and PC Port Link both set to auto negotia. The 'Voice QoS' section shows Voice QoS and SIP QoS both set to 40. The 'Local RTP Port' section shows MaxRTPPort set to 11800 and MinRTPPort set to 11780. The 'SNMP' section shows Port and Trusted Address fields. The 'WebServer' section is also visible. A 'NOTE' box on the right explains VLAN and QoS.

VLAN

VLAN is a group of hosts with a common set of requirements that communicate as if they were attached to the Broadcast domain, regardless of their physical location.

IP 200 supports 802.1p and q packet tagging. VLAN ID and Priority can be set independently for PC and WAN ports.

To configure VLAN settings via Phone interface:

- 1) Make sure the active User Type is admin or var
- 2) Press  -> 3. **Settings** -> 2. **Advanced**.
- 3) Enter the password required, scroll to **Network** option, press ok button and select **VLAN**, then press OK to enter.

- 4) Choose **WAN Port** and press OK button to enter.
- 5) Use the left or right navigation keys to choose and set the **VLAN Status**; input the VID Number, Priority.
- 6) Press OK button to save the settings, or MENU to return to VLAN menu.

Follow the same method to set the PC Port option.

To configure VLAN settings via Web interface:

Choose Network->Advanced. Perform necessary changes. Check with your system administrator for more information.

The screenshot displays the Alcatel-Lucent web interface for configuring VLAN settings. The 'Network' tab is selected, and the 'Advanced' sub-tab is active. The 'VLAN' section is highlighted with a blue box and contains the following configuration options:

Port	Active	VID	USR PRIORITY
WAN Port	Disabled	0 (0-4094)	0
PC Port	Disabled	0 (0-4094)	0

Other visible configuration sections include:

- LLDP:** Active (Disabled), Packet Interval (120)
- Port Link:** WAN Port Link (auto negotiate), PC Port Link (auto negotiate)
- Voice QoS:** Voice QoS (40), SIP QoS (40)
- Local RTP Port:** Max RTP Port (11800), Min RTP Port (11780)
- SNMP:** Port, Trusted Address
- WebServer:** (Type, Port, Trusted Address)

NOTE: VLAN is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations. QoS Packet tagging intended to provide priority to time-critical traffic. Local RTP Port Define the port to be used for voice transmission.

HTTPS

Temporis IP200 supports HTTPS (Hypertext Transfer Protocol over Secure Socket Layer). Adding SSL layer under HTTP, in short, adds a security layer to HTTP. Admins can set this transmission mode via web page.

To configure HTTPS settings via Web interface:

Go to Network->Advanced, choose WebServer option, in the pull-down menu of Type

field, choose the transmission mode, then click the Confirm button to save the changes.

802.1X

IEEE 802.1X is an IEEE Standard for port-based Network Access Control (PNAC). It is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN, either establishing a point-to-point connection or preventing it if authentication fails. It is used for securing wireless 802.1x access points and is based on the Extensible Authentication Protocol (EAP).

This IP phone supports 802.1X. For the details, please consult your system administrator or technical support.

Other Settings

The table below includes other parameters which can be configured in the Advanced section of Network settings web interface.

Field Name	Description
Voice QoS	Set the TOS field value to indicate the network desired priority for voice and signaling packets
Local RTP Port	Define the port range for voice transmission.
WebServer	Users can choose WebServer type: Disable, HTTP, HTTPS, or HTTPS & HTTP.

Voice QoS ?			
	Voice QoS	<input type="text" value="40"/>	(0~63)
	SIP QoS	<input type="text" value="40"/>	(0~63)
Local RTP Port ?			
	MaxRTPPort	<input type="text" value="11800"/>	(0~65535)
	MinRTPPort	<input type="text" value="11780"/>	(0~65535)
SNMP			
	Port	<input type="text"/>	(0~65535)
	Trusted Address	<input type="text"/>	
WebServer ?			
	HTTP port	<input type="text" value="80"/>	(1~65535)
	HTTPS port	<input type="text" value="443"/>	(1~65535)
	Type	<input type="text" value="HTTP&HTT"/> <input type="text" value="Disabled"/> <input type="text" value="HTTP&HTTPS"/> <input type="text" value="HTTP Only"/> <input type="text" value="HTTPS Only"/>	
802.1x ?			
	802.1X Mode		
	Identity	<input type="text"/>	
	MD5 Password	<input type="text"/>	
Span to PC port			
	Span to PC port	<input type="text" value="Disabled"/>	
Random Registration			
	Random Registration	<input type="text" value="0"/>	(0~60s)
		<input type="button" value="Confirm"/>	<input type="button" value="Cancel"/>

used for voice transmission.

Note:

For more details of the HTTPS, you can consult with your system administrator.

IP phone also support Internet Protocol Version 6.

Maintenance Tasks

User levels

The phone allows three modes to configure the phone:

- User Mode
- VAR mode
- Administrator Mode

Administrator mode grants unlimited access to the phone configuration on both Web and Phone interface. User Mode is not able to access the settings on the Phone



interface such as: Accounts, Network, Reset to Factory, other advance phone settings. VAR mode grants access to subset of Administrator functions (Login and password are var/var).

Administrator Mode

Administrator mode grants unlimited access to the phone configuration on both web and phone user interface. The administrator/user password is used to access:

Web interface.

The advanced settings of the phone such as Network, Account, Reset to Factory Settings via the Phone interface.

The default administrator password is **admin**. Meanwhile the user name for Web interface access is **admin**.

To change the administrator password via Phone interface:

- 1) Make sure active User Type is admin
- 2) Press ->3.**Settings**->2.**Advanced**->OK->5.**Set Password**->OK.
- 3) You are prompted to enter the Current PWD, New PWD and Confirm PWD, Press  to change the input method.
- 4) Press OK button to confirm the change, or MENU to return to previous menu.

To change the administrator password via Web interface:

Choose Security->Password->admin, enter the Current, New and Confirm password, click Confirm button to save the changes, or Cancel button to cancel the changes.

To logout via Web interface:

Click the Logout button in the top right corner.

Reboot

You should reboot the phone when you are asked to, e.g. after applying specific changes to phone configuration.

To reboot via Web interface:

- 1) Go to Upgrade->Basic.
- 2) Click Reboot button.

-
- 3) You are prompted to confirm your request. Press OK to confirm, press Cancel to cancel the operation.

Note:

Please do not power off during reboot, or it will cause the flash memory error.

Reset to Factory

You should reset the phone only in this case: the phone configuration was changed and the phone is not functioning anymore. To maintain the configuration of the phone, you need your system administrator or service provider's advice.

To reset to factory via phone interface:

- 1) Make sure active User Type is admin or var
- 2) Press ->3.**Settings**->2.**Advanced**.
- 3) You are prompted to enter the required password.
- 4) Scroll to **Reset Factory** option, then press OK button.
- 5) You are prompted to confirm the change, press OK to reset to factory settings, or MENU to return to previous menu.

It will take a few minutes to reset, please do not power off during resetting, or it will cause flash memory error.

To reset to factory via Web interface:

Choose Upgrade->Basic.

Click Reset button.

You are prompted to confirm the change, press OK to confirm the changes, press Cancel to cancel the operation.

Note:

If you press Confirm all current settings including contact list, call history, account settings, etc will be lost. You need to export the configuration first if you still want to import the old configurations after reset.

Otherwise your phone will have to be configured manually unless mass provisioning is used

Configuration Export/Import

To Export/Import your configuration via Web interface:

- 1) Go to Upgrade->Advanced, select Import/Export Config, click Export button to export the file to your local computer.
- 2) Choose Upgrade->Advanced, select Import/Export Config, click Browse button, select the specific configuration file in your local computer, click Import button.
- 3) It will take a few minutes to reset, please do not power off during resetting, or it will cause flash memory error.

WatchDog

When 'WatchDog' function is 'Enabled', phone will auto reboot after 10 seconds if any process in the phone crashes. When the function is disabled, the phone will not reboot.

Configure watchdog via web interface:

Choose phone-> Preference->WatchDog, in the pull-down menu, choose enable or disable this function.

Action URL/URI

Action URL: when an operation takes place, the phone will send an http GET request to configured URL.

The events that can be recorded include: Setup Completed, Log On, Log Off, Register Failed, Off hook, On hook...etc.

To set Action URL via web interface:

Choose Phone->Action URL

Enter the Corresponding contents

Click Confirm to save the changes.

Action URI: Remote control commands which can be used to generate CTI applications using Temporis IP200.

Please ask your Technical Support for more details on these features.

Firmware Update and Autoprovisioning

The phone is delivered with pre-installed firmware. If you require updating the phone's firmware, please contact your system administrator. You can only update the firmware via Web interface.

To update the firmware manually via Web interface:

Choose Upgrade->Basic->Browse; select the firmware file in your local computer.

Click Upgrade button to update the new firmware.

Please do not power off during the updating.

To update the firmware and configuration automatically via Web interface or using the phone GUI you will need to set up an Autoprovision environment

Note:

Please check with your Technical Support for the Autoprovisioning guide.

To launch Autoprovisioning using web interface:.

- 1) Choose Upgrade->Advanced. Configure necessary settings: Custom Option, Custom Option Type, URL, Account, Password, Common AES Key and MAC-Oriented AES Key, PNP config and Check New Config.
- 2) Click Confirm button, the phone will check the server for a new firmware in a specific time, and will update automatically if there is new firmware.
- 3) You can also update the firmware immediately by pressing Autoprovision button.

Set Auto Provision via phone interface:

Go to ->3.Settings->2.Advanced ->7. Auto Provision->OK.

Enter the URL, User Name and Password (optional).

Click OK key to save the changes and launch Autoprovision.

Note:

1. Any power interruption during the subsequent process will most likely lead to a flash memory error. As a result the system cannot boot up anymore. The procedure can be only conducted in Administrator Mode.
2. If the updating is failed, you can re-plug the power adaptor, and press the Speaker key for 3 seconds, the IP phone will enter the account configuration interface. Users can reconfigure the IP phone account information here, and then press the OK button to save the changes.
3. The upgrade priority is PNP, Custom Option, DHCP options, preconfigured URL .
- 4.

Decryption

Your Temporis IP 200 supports configuration file encryption for provisioning. In case encryption is used, AES key needs to be configured on the phone.

Set AES key via Web interface:

Choose Upgrade->Advanced->Fill in either common AES Key or MAC-oriented AES Key. The parameters to fill in depend on the files you're using for provisioning. Contact your service provider or check with your technical support. Then click the Confirm button as shown below.

ALCATEL

Custom DHCP Option(128 ~ 254) ?

Option Type ?

URL ?

Account ?

Password ?

Common AES Key ?

MAC-Oriented AES Key ?

Zero Active ?

WaitTime ?

PNP Config ?

Check New Config ?

Click this button to auto provision immediately ?

Export / Import Config ?

Export System Log ?

PCAP Trace ?

NOTE

Custom DHCP Option
Specify DHCP Option to use for provisioning. Refer to Auto Provision Manual for details about provisioning.

AES Key
Provided by your ISP.

Click this button to launch autoprovisioning immediately
Click this button to launch autoprovision immediately.

Export/Import Config
Backup your settings in a configuration file with Export. You can retrieve these settings later with Import.

System Log
Two methods are supported for the system log, Local or External Server.

Note:

You can ask your system administrator for the decrypt password.

Set AES Key via phone interface:

Go to ->3.**Settings**->2.**Advanced**->6.**Set AES Key**->OK.

1. Common:

Enter the Common AES and the MAC-oriented option.

Press the OK button to save the changes.

Zero-Active Interface

Zero-sp-touch function can help users to configure Auto provisioning and network parameters quickly.

If this feature is enabled, whenever the set powers on or a pre-assigned Function Key is pressed, the phone will turn to the Zero-Active interface.

Turn on Zero-Active via web interface:

- 1) Choose Upgrade->Advanced-> Zero Active, in the pull-down menu, choose enable to turn on this function.
- 2) Click Confirm to save.

The screenshot displays the Alcatel-Lucent web interface for configuring a phone. The top navigation bar includes tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below this, there are sub-tabs for Basic, Advanced, and TR069. The main content area is titled 'ALCATEL' and 'ATLINS'. It contains a list of configuration options for Custom DHCP Option (128 ~ 254):

- Option Type: String
- URL: [Text Input]
- Account: [Text Input]
- Password: [Text Input]
- Common AES Key: [Text Input]
- MAC-Oriented AES Key: [Text Input]
- Zero Active: Enabled (selected)
- WaitTime: [Text Input]
- PNP Config: Disabled
- Check New Config: Disabled

Below these options are buttons for 'Auto provision', 'Export / Import Config' (with 'Parcourir...' button), 'Export System Log' (with 'Local' dropdown and 'Export' button), and 'PCAP Trace' (with 'Start', 'Stop', and 'Export' buttons). At the bottom are 'Confirm' and 'Cancel' buttons.

NOTE
Custom DHCP Option
Specify DHCP Option to use for provisioning. Refer to Auto Provision Manual for details about provisioning.

AES Key
Provided by your ISP.

Click this button to launch autoprovisioning immediately
Click this button to launch autoprovision immediately.

Export/Import Config
Backup your settings in a configuration file with Export. You can retrieve these settings later with Import.

System Log
Two methods are supported for the system log, Local or External Server.

Once the phone enters into Zero-Active interface (at power on or due to a specific function key press), a countdown screen will be shown first.

If no operation is done nor Cancel key is pressed the phone will revert into normal idle interface at the end of the countdown.

If OK key is pressed prior to the end of the countdown, the phone will enter into a

network setting interface.

Press OK key again to enter an AutoP setting interface. Enter the corresponding contents and press OK key to save the settings or press MENU key return to previous menu.

If you have pressed OK, autoprovisioning will be launched with the parameters you just configured.

System Log Export

If some errors happen in your phone, you can export the system log and send it to your system administrator for diagnosis.

To export the System Log via web interface:

Choose Upgrade->Advanced, select Export System Log, if the type is Local, it will export the syslog directly; if the type is server, it will export the syslog to the specified server.

Set the desired debug level for the System Log

Click Export button to export the file

PCAP Trace Export

The PCAP Trace is used to record the data packets at the WAN port of your phone. If some errors happened in your phone, you can export the PCAP trace and send to your system administrator for diagnosis.

To export the PCAP Trace via web interface:

Choose Upgrade->Advanced. Select PCAP Trace option, click Start button to launch Wireshark capture. Click Stop to stop Wireshark capture, and then click Export to export the file to your local computer.

Keys configuration

Send Key configuration

Admins can set a specific button (“#” or “*”) as the send key. If this feature is enabled, once you’ve dialed the desired number, you can press this send key to call your correspondent immediately.

Please check Send Key configuration in Basic Calls chapter for a complete description.

Programmable keys

The keys from 0-9 (memory keys), navigation keys and function keys on the keypad are editable. Users can customize specific features for these keys according to their needs. **(The programmable keys are only available when the phone is in idle mode).**

Default configuration for idle mode is:

Up key: access to redial list

Down key: access to call forward

Left key: access to call log

Right key: access to account switch

Ok key: access to status

Conf key: access to DND

Transfer key: access to call forward

This function can only be set via the Web interface:

Choose Phone->Function Keys->Programmable Key.

Key	Type	Line	Extension
Up	Redial	Auto	
Down	N/A	Auto	
Left	History	Auto	
Right	Menu	Auto	
OK	Forward	Auto	
Cancel	Call Return	Auto	
CONF	Pick Up	Auto	
Hold	Speed Dial	Auto	
TRAN	Local PhoneBook	Auto	
	Local Group	Auto	
	Keypad Lock	Auto	

Choose and customize specific features for these keys.

Click Confirm button to save the changes.

Line keys Configuration

The phone has 2 line keys to which functions can be assigned out of a list of 28 functions. The following list shows the functions you can set on the Line keys. The next paragraphs provide a description for each function. The default configuration for each key is Line.

BLF

Line

Speed Dial

Intercom

URL Record

Shared Line

Conference

Forward

Transfer

Hold

DND

Redial

Call Return

Pick Up

Call Park

Group Listening

Voice Mail

DTMF

Public Hold

Private Hold

Group Pick up

Paging

Record

BLF List
Prefix
Zero-Active
ACD
Local PhoneBook

Note:

Many of those features require IP PBX or service provider support and hence must be configured in the server first in order to work on your phone. Contact your system administrator or Technical support for more details.

In addition, please note that not all the servers support the just above mentioned features

BLF

Busy Lamp Field (BLF) allows you to monitor the status (idle, ringing, or busy) of other SIP accounts. User can dial out on a BLF configured key.

To assign the key as BLF via web interface:

- 1) Choose Phone->Function Keys->Line Key. Choose one of the keys on which you want to make the assignment. , Note there is a pull-down menu in the Type field. Choose BLF from the list.
- 2) Enter the number you want to monitor in the Value field,
- 3) In the "Line" field, select a line for which to apply this key.
- 4) And then enter the feature code for call capture in the extension field.
- 5) Press Confirm button to save the changes.

Please refer to "LEDs" chapter for more details about the LED status in different situation.

Note:

1. In the Web interface, you can also set the pickup number to activate the pickup function. For example, if you set the BLF number as 212, and the pickup number is *83, then when there is an incoming call to 212, press the BLF key, it will call out the *83 automatically to pick up the incoming call on 212.
2. Not all servers support BLF. Please check with your service provider or system administrator

BLF List

BLF list is a function which can monitor group status versus one to one monitoring in conventional BLF. Information feedback from the server will indicate which BLF list will monitor which account.

To set BLF List via web interface:

- 1) Choose Account->Advanced-> BLF List URI, enter the BLF List URI.
- 2) Then enter the BLF List Code in the BLF List Code field.
- 3) Click the Confirm button to save.

To assign BLF List to a function key via web interface:

- 1) Choose Phone->Function Keys->Line Key, choose one of the keys on which you want to make the assignment. Note there is a pull-down menu in the Type field, choose BLF List from the list.
- 2) In the "Line" field, select a line for which to apply this key.
- 3) Press Confirm button to save the changes.

Note:

This feature is not supported by all servers. Please check with your service provider or system administrator

Line

The most frequent usage for line keys is assigning them one of the two possible user accounts for call management.

To assign the key as Line via web interface:

Choose Phone->Function Keys->Line Key, choose one of the line key on which you want to make the assignment. Note there is a pull-down menu in the Type field: choose Line from the list, and press Confirm button to save the changes.

Speed Dial

You can configure the key as a simplified speed dial key. This key function allows you to easily access the most frequently dialed numbers.

To assign the key as Speed Dial via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose

Speed Dial from the list.

- 2) Enter the number you want to perform Speed Dial in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

Intercom

Intercom is useful in office environments as a quick access to connect to the operator or the secretary.

To configure a function key as Intercom via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, and choose Intercom from the list.
- 2) Enter the extension number you want to intercom in the Value field.
- 3) In the "Line" field, select a line for which to apply this key, the default one is Line 1.
- 4) Press Confirm button to save the changes.

Note:

1. Your VoIP PBX must support this feature.
2. Please make sure the Intercom Allow is enabled.

URL Record

This is a server specific function that allows you to record your conversation by pressing a preconfigured function key.

When you are in conversation, press function key to start the recording process of the current call.

Press the function key again to disable the recording function

Follow the voice prompts to listen to the recording.

To assign a Line key as URL Recorder via web interface:

Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose URL Record from the list.

Enter the url and condition code in the Value field.

Press Confirm button to save the changes.

Note:

1. During a conversation, press the dedicated line key to start the recording process; if the other party hangs up, your phone will turn to the idle status.
2. This feature is supported by specific servers only . Please check with your service provider.

Shared Line

The server-specific Shared Line Appearances (SLA, which is also named as BLA) feature allows subscribers to share SIP lines and also provides status monitoring of the shared line. When a user places an outgoing call using such an appearance, all members belonging to that particular SLA group are notified of this usage and are blocked from using this line appearance until the line goes back to idle state or when the call is placed on hold.

Similarly all members of the SLA group are notified of an incoming call and the call can be picked up on a line appearance associated with the SLA extension.

To assign the key as Shared Line via web interface:

Choose Phone->Function Keys->Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose Shared Line from the list.

Enter the condition code in the Value field.

In the "Line" field, select a line for which to apply this key, the default one is Line 1.

Press Confirm button to save the changes.

Note:

This service is supported by specific servers only. Please check with your service provider

Conference

You are allowed to configure the Line key to be used as a conference key while remaining in the current call. This key allows a user on a call to conference another party while remaining in the conference.

To assign the key as Conference via web interface:

Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose Conference from the list.

Press Confirm button to save the changes.

Forward

Two use cases can be considered for this function. If a key is configured as Forward key:

- When you press this key during an incoming call, you will be able to divert this call directly. Depending on your configuration you will be prompted to enter target number, or stored value will be used. This function can be used, for example, to divert certain calls to your voicemail
- When you press it in idle status the IP phone will enter the forward interface. In this case you will be able to enable CF and set the phone number to which calls must be forwarded if it had not been set previously. Forward type will be the latest used.. By pressing the key again CF will be disabled.

To assign the key as Forward via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose Forward from the list.
- 2) Enter the extension number you want to forward to in the Extension field. (For forward on ringing scenarios)
- 3) Press Confirm button to save the changes.

Transfer

You are able to configure a key as a transfer key to perform the Blind/Attended/Semi-Attended Transfer. The main difference with Transfer hard key is in this case you can pre-configure the number to which transfer will be performed. This is very useful when there is a limited number of numbers to which you usually transfer your calls.

To assign the key as Transfer via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment. From the pull-down menu in the Type field select Transfer.
- 2) Enter the Number in the "Value" field. In this case, when you are on a conversation,

if you press this key the phone will blind transfer to that number. If you leave it blank the key will behave as transfer hard key.

- 3) Press Confirm button to save the changes.

Hold

The key can be configured as a hold key. You can use this key to hold and retrieve a call during the conversation.

To assign the key as Hold via web interface:

Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose Hold from the list.

Press Confirm button to save the changes.

DND

If the key is configured as DND key, it will work as a toggle key to enable/disable this service.

Please check DND chapter in the Incoming calls section of Basic Call Functions chapter for more information about this function.

To assign DND to a key via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment. From the pull-down menu in the Type field select DND.
- 2) Press Confirm button to save the changes.

Redial

If a key is configured as Redial key:

- If you press this key in idle status, it will enter the Dialed Calls interface
- If you press this key in off hook mode, last dialed number will be sent directly
- If you press this key when there is an active call, it will be put on hold and a new call sent to the last dialed number

To assign Redial to a key via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Choose one of the keys on which you want to make the assignment. From the pull-down menu in the Type field select

Redial.

- 2) Press Confirm button to save the changes.

Call Return

Call Return function dials out the last number from which you received a call.

If there is already an active call, when pressing Call Return key the first call will be put on hold and a new one will be sent to the Call Return number

To assign Call Return to a key via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Call Return.
- 2) Press Confirm button to save the changes.

Pick Up

Pickup key allows you to perform call capture from another ringing station with a single key press. This feature is not supported by all servers. Please check with your service or IP PBX provider for more details.

To assign Pick Up to a key via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Pick Up option.
- 2) Enter the feature code for call capture (for example, input *78345, *78 is the feature code and the 345 is the extension number you want to pickup,) in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

Call Park

Call Park is a feature that allows a person to put a call on hold at one telephone set and continue the conversation from any other telephone set. Your server needs to support this feature. Please check with your service provider or IP PBX vendor for more details.

The "call park" feature is activated by pressing a preprogrammed button or a special sequence of buttons. When the conversation which is monitored was transferred to an unused extension number, you can press this key to retrieve the call.

To assign the key as Call Park via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, and choose Call Park from the list.
- 2) Enter the number you want to park in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

Group Listening

When the key is configured as Group Listening key, you are allowed to enable the Speakerphone and Handset/Headset mode at the same time. It is suitable for a group conversation which has more than one person at one side. You are able to speak and listen using handset/headset; meanwhile the others nearby can listen using speakerphone. You can get back to the previous mode by pressing the key again. (If the current mode is handset or headset, users can press the speaker button to open or close the group listening function)

To assign the key as Group Listening via web interface:

- 1) Choose Phone->Function Keys-> Line Key, choose one of the keys you want to make the assignment, there is a pull-down menu in the Type field, and choose Group Listening from the list.
- 2) Press Confirm button to save the changes.

Voice Mail

When the key is configured as Voicemail key you are allowed to access voicemail quickly by pressing this key.

To assign the key as Voice Mail via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, choose Voice Mail from the list.
- 2) Enter the number you want to set as the voice mail box in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

DTMF

With this function you will be able to send out the desired DTMF number during the conversation. The number needs to be set in advance. You can use it for example to provide some inputs to frequently used IVRs, or to launch voice recording in Asterisk environments

To assign the key as DTMF via web interface:

- 1) Choose Phone-> Function Keys -> Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, and choose DTMF from the list.
- 2) In the "Value" field, enter the specific number.
- 3) Press Confirm button to save the changes.

Public Hold

This is a feature supported by some specific servers.

You can configure a key for Public Hold. During a conversation, all members belonging to that particular BLA group can use this key to hold or retrieve a call.

To assign the key as Public Hold via web interface:

- 1) Choose Phone-> Function Keys ->Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Public Hold.
- 2) Press Confirm button to save the changes.

Private Hold

This is a feature supported by some specific servers.

You can configure a key for Private Hold. During a conversation, all members belonging to a particular BLA group can use this key to hold the call, but only the initiator can retrieve the call.

Please check with your service provider.

To assign the key as Private Hold via web interface:

- 1) Choose Phone-> Function Keys ->Line Key, choose one of the keys on which you want to make the assignment, there is a pull-down menu in the Type field, and choose Private Hold from the list.
- 2) Press Confirm button to save the changes.

Group Pick up

Group Pickup key allows you to perform call capture from any ringing station which belongs to a Capture Group, with a single key press. This feature is not supported by all servers. Please check with your service or IP PBX provider for more details.

To assign the key as Group Pick Up via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Group Pick Up option.
- 2) Enter the feature code (for example,*78) in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

Paging

This feature has to be supported by the server. Check with your service provider or IP PBX vendor.

When you configure a key as Paging key and press this key, the phone will dial configured code or number to invoke Paging.

To assign the key as Paging via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Paging option.
- 2) Enter the number or feature code you want to dial out directly in the Value field.
- 3) In the "Line" field, select a line for which to apply this key.
- 4) Press Confirm button to save the changes.

Record

Call recording is a server function invoked by the phone when the Record key is pressed. The phone requests the server to record the active conversation. It is supported by specific servers and is implemented using SIP INFO messages. Please check with your service provider or IP PBX vendors.

By pressing Record key the server should start recording, and by pressing it again recording should stop. Please pay attention to the maximum recording time and frequency your server is able to handle..

To assign the key as Record via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Record option.
- 2) Press Confirm button to save the changes.

Prefix

When you press a key in which prefix function has been set up, the phone will automatically dial the prefix and you will only have to dial the rest of your correspondent number to call him out. Prefix can only be assigned to the line keys.

To assign the key as Prefix via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Prefix, and enter the number you want to show up on the dial interface in the Value field.
- 2) Press Confirm button to save the changes

Zero-Active

When you press a Function Key which has been set as Zero-Active, the phone will launch Zero-Active interface. Please check section "Zero-Active" section in Maintenance Tasks chapter for more details.

To assign a key as Zero-Active via web interface:

- 1) Choose Phone-> Function Keys -> Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Zero-Active.
- 2) Press Confirm button to save the changes

ACD

ACD (Automatic Call Distribution) aims at having incoming calls distributed automatically to different extensions according to their availability stages. To perform this function, you have to assign previously this feature to a Line key. This feature is supported by some specific servers. Please check with your service provider or PBX vendor for details.

To use ACD:

- 1) Press line Key. A login box will pop up. Enter the User ID and Password and click

OK button to login.

- 2) Press line Key again to enter in the ACD page, press OK button to change the Available/Unavailable status.
- 3) You can press  button to logout.
- 4) Press MENU key to cancel the operation
- 5) When your status is Available, calls will be directed to your phone. When the status is unavailable, calls will not be directed to your phone.

To assign the key as ACD via web interface:

- 1) Choose Phone-> Function Keys ->Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose ACD.
- 2) In the "Line" field, select a line for which to apply this key.
- 3) Press Confirm button to save the changes

Note:

ACD is not available on all call servers. For more information, contact your system administrator.

Local PhoneBook

The Line keys can be configured as Local PhoneBook key. Then pressing this key when in idle status, you can enter the Local PhoneBook interface, and more in particular you can choose to access directly one of your Groups.

To assign the key as Local PhoneBook via web interface:

- 1) Choose Phone-> Function Keys ->Line Key. Select the key on which you want to make the assignment. From the pull-down menu in the Type field choose Local PhoneBook.
- 2) In the Line field, choose a Group.
- 3) Press Confirm button to save the changes.

Trouble shooting

I cannot register to the server

Check the IP address (press OK key for status). If you set your WAN port in DHCP mode, please make sure there is an available DHCP server.

Check your gateway.

Check your DNS server.

Make sure your account information is the same as you got from your service provider.

Check whether the SIP server is on.

Check the SIP register port, the default value is 5060.

I can't get an IP address

Make sure you have plugged the Ethernet cable into the WAN port.

Make sure there is an available DHCP server on, and there are available IP addresses in the server.

Try to set your WAN port to static IP client mode.

During a call, I cannot hear any voice

Make sure your handset is tightly connected with the phone.

Check whether you have muted the conversation or not.

Consult the outbound server details with your service provider.

I have problems to send DTMF digits during calls

Check which kind of DTMF you are using, and whether it is compatible with the server.

Consult the payload value with your service provider.

Time and date are not correct

Select the right time zone or enter the time information manually on the webpage or the phone.

Check your DST settings.

How to answer the incoming calls during a call?

If a call comes in when you are in a conversation, press OK key to answer the incoming call.

How to refuse incoming calls during a call?

To refuse current incoming call press  key.

To refuse all incoming calls that may arrive when you are in a conversation you can turn off call waiting function.

How to update the firmware?

Enter the webpage of your phone, go to Upgrade, then you can find the option "Select and Upgrade Firmware" at the bottom of the page.

1) Select the file to update, and then click the Upgrade button.

Note:

Make sure the firmware you choose is provided by your service provider, or the device will probably crash after the update.

How to auto provision?

Consult auto provision server address with your service provider and ask your technical support for details.

Appendix A.

Line Key configuration menu

1.Type: Speed dial	→	1.Type: Intercom	→	1.Type: Line	→	1.Type: BLF	→	1.Type: BLF List	→	1.Type: Shared LIne	→	1.Type: URL Rec	→	1.Type: ACD	→
↓		↓		↓		↓		↓		↓		↓		↓	
2.Account ID		2.Account ID		2.Account ID		2.Account ID		2.Account ID		2.Account ID		2.URL		2.Account ID	
↓		↓		↓		↓		↓		↓					
3.Value		3.Value		3.User (R/O)		3.Value				3.Value					
				↓		↓									
				4.Server (R/O)		4.Extension									



1.Type: Key Event																	
↓																	
2.Key type: Call Return	→	2.Key type: Pickup	→	2.Key type: Call Park	→	2.Key type: DTMF	→	2.Key type: Voicemail	→	2.Key type: Group Listening	→	2.Key type: Zero-Active	→	2.Key type: Prefix	→		
		↓		↓		↓		↓						↓			
		3.Account		3.Account		3.Value		3.Account									3.Value
		↓		↓				↓									
		4.Value		4.Value				4.Value									

2.K type: group Pickup	→	2.K type: Paging	→	2.K type: Record	→	2.Key type: Local Group	→	2.K type: Conf	→	2.K type: FWD	→	2.K type: Transfer	→	2.K type: Hold	→	2.Key type: DND	→	2.Key type: Redial	→
↓		↓		↓		↓				↓		↓							
3.Account		3.Account				3.Local Group				3.Value		3.Value							
↓		↓				↓													
4.Value		4.Value																	

Memory Key configuration menu

1.Type: Speed dial	→	1.Type: Intercom	→	N/A	→	1.Type: Key Event			
↓		↓				↓			
2.Account		2.Account				2.Key type: Call Return	→	2.Key type: Paging	→
↓		↓						↓	
3.Value		3.Value						3.Account	
								↓	
								4.Value	

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