

**THOMSON**

## ST 2030 SIP Administrator Guide

VoIP Business Phone



**Thomson Telecom**

S.A.S with a capital of 130 037 460 €

RCS 390 546 059

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### **European Community Declaration of Conformity**

This equipment complies with the requirements relating to electromagnetic compatibility,  
EN55022 Class B for ITE and EN55024.

This meets the essential protection requirements of the European Council Directive 89/336/EEC  
on the approximation of the laws of the Member States relating to electromagnetic compatibility.

Thomson Telecom declares that this ST2030 IP Phone is in compliance with the essential  
requirements and other relevant provisions of Directive 1999/5/EC.

You can download the declaration of conformity on [www.speedtouchpartner.com](http://www.speedtouchpartner.com).

The CE logo involves the conformity of the product with the essential requirements of  
implemented directives.

### **Northern America Federal Communications Commission (FCC) Statement**

This equipment generates uses and can radiate radio frequency energy and, if not installed and  
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reasonable protection against radio interference when operated in a commercial environment.  
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# Chapter 1 Safety Information

## 1. Operating conditions

This telephone is to be operated on a local area network. The telephone must be equipped with the appropriate software version. The actual version of firmware is v.1.30, you can download the latest on:

[www.speedtouchpartner.com](http://www.speedtouchpartner.com)



Read these instructions carefully before connecting the SIP phone to its power source.

**Caution**

## 2. Location safety instructions

Do not expose the IP Phone to fire, direct sunlight or excessive heat.

Do not expose the IP Phone to rain or moisture and do not allow it to come into contact with water.

Do not install the IP phone in an environment likely to present a THREAT OF IMPACT.

The IP phone must be installed at least 1 meter from radio frequency equipment, such as TVs, radios, hi-fi or video equipment (which radiate electromagnetic fields).

The IP phone is designed to work in temperatures from 0°C to 45°C.

## 3. Care

You may clean the IP phone using a fine damp cloth. Never use solvents (such as trichloroethylene or acetone), which may damage the phone's plastic surface and LCD screen. Never spray the phone with any cleaning product whatsoever.

## 4. Connections

Equally, incorrect reassembly could cause electric shock on re-use of the appliance.

The IP Phone must be powered using the power adaptor provided with the package.

If you do use an alternative power adaptor, it must comply with the following standards:

- EN60950, CE mark, U/L
- Output: 5VDC /600mA.
- Input: 100V/60Hz to 230V/50Hz.

Any damage caused to the IP Phone as a result of using unsupported power adaptors will not be covered by the manufacturer's warranty.

Do not connect the LAN/PC ports to any network other than an Ethernet network.

Do not work on the system or connect or disconnect cables during lightning storms.

Before working on any system fitted with an ON/OFF switch, turn OFF the power and unplug the power cord.

## **5. Qualified service**

No repair can be performed by the customer, if you experience trouble with this equipment for repair or warranty information, please contact your administrator.

Thomson disclaims all responsibility in the event of use that does not comply with the present instructions.

Product disposal warning:

Ultimate disposal of this product should be handled in accordance with national laws and regulations.

Information regarding the products in this manual is subject to change without notice. This guide is believed to be accurate but is presented without warranty of any kind, express or implied. The usable services and features on the phone depend on the installed software release and on call manager. Therefore, the conformity of the admin guide cannot be guaranteed.

# Chapter 2 Product overview and connections

## 1. Introduction

This administrator guide describes how to set up, connect cables, and configure your ST2030 SIP Phone. It also provides information on how to configure the Network settings and change the settings of your IP Phone. The administrator guide also includes the way to view and upgrade the firmware.

## 2. Note and Caution

**Note** and **Caution** in this manual are highlighted with graphics as below to indicate important information.



*Contains related information that corresponds to a topic.*

**Note** \_\_\_\_\_

---



*Represents essential steps, actions, or messages that should not be ignored.*

**Caution** \_\_\_\_\_

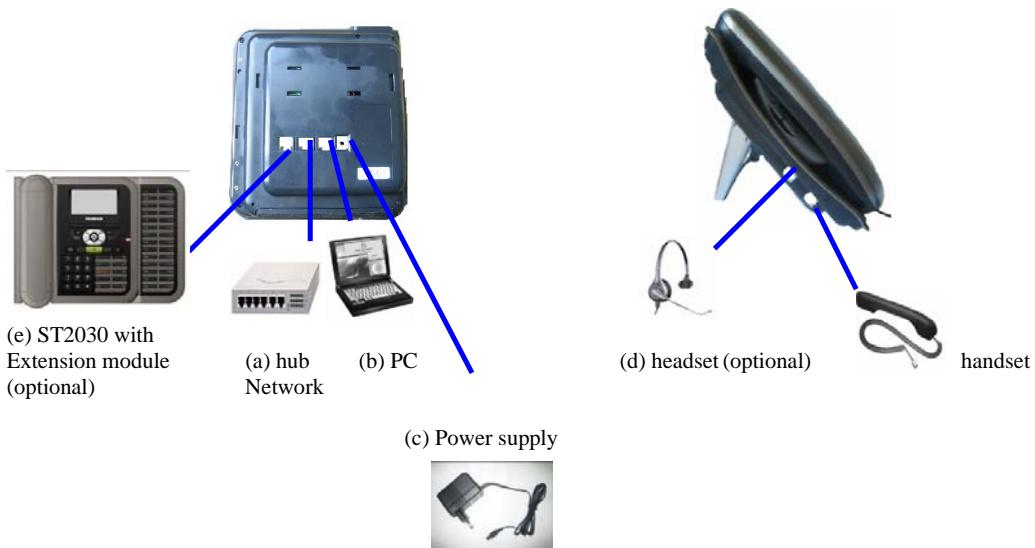
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## 3. Package contents

1. IP phone base unit
2. Handset
3. Coiled handset connecting cord
4. Quick installation and user guide
5. 1 Ethernet cable
6. 1 Power supply

Not included: Ethernet cable to connect any PC to the telephone through its PC switch

Optional: Extension module with 20 memory keys and headset



**Figure 2.1 IP Phone Cable Connections**

## 4. Connecting to the Network and the PC

The SIP Phone has 2 RJ-45 ports that each support 10/100 Mbps full duplex Ethernet connection to external devices- Network port and access port (one for PC and one for LAN).

### ➤ Network Port (10/100LAN)

Use the Ethernet cable to connect the LAN port to an Internet equipment, such as a hub, switch. Or directly to the Network. In Power over Ethernet (PoE) office environment, the IP phone can be powered from a switch via Ethernet cable, in which case the external power adaptor is not needed.

### ➤ Access port (10/100PC)

Use the Ethernet cable to connect a network device, such as a computer to the PC port on the back of your phone.

## **5. Powering up the Phone**

The ST2030 could be powered by either a main power unit, or PoE.  
Its power consumption is under 6W.

➤ Power Plug

The power plug is fed with 9VDC, 800mA, linear.

The power plug unit will be adapted specifically to the target country requirements:

- EU plug for Europe (ST2030 EU)
- « American » plug for US (ST2030 US)
- “UK” plug for UK (ST2030 UK)

➤ Power Over Ethernet

The Speed Touch 2030 supports PoE as defined by 802.3af Standard

It is class 2, and supports class negotiation (in order to plug as many devices as possible on a PoE hub). It can be powered with PoE up to 100 meters with category 5,5e or 6 cables.

## **6. Connected a headset**

If you will use a headset, plug the RJ9 headset connector into the headset jack port on the left side of your IP Phone (see the figure 2.1)

## **7. Overview**

The SpeedTouch 2030 SIP is a VoIP Phone that can be plugged directly into an IP Network and used very much like a standard private branch exchange (PBX) telephone. The SpeedTouch 2030 is an IP telephony instrument that can be used in a VoIP environment.

The ST2030 IP phone is compliant with SIP protocol.

## **8. Main Features**

- 6-line LCD display
- Connectivity: Integrated 2 ports 10/100 Ethernet switch
- Phone services:
  - Multilines (up to 5), Call Forward, Call Transfer (attended), Call Hold, Redial
  - Group listening, Hands free
  - Message Waiting Indicator
  - Speed dial, Conference call
  - Phonebook, Call logs
  - Caller ID display
- Audio extension connection: Integrated headset RJ9 port
- Multiple power options: power over Ethernet 802.3af and external power supply adaptor
- VoIP Standard: SIP V2 (RFC 3261)
- Web browser interface for configuration



This administrator guide is based on firmware v.1.30, you can download the latest administrator guide on:  
[www.speedtouchpartner.com](http://www.speedtouchpartner.com)

Note

## 9. Supported Features

In addition to the physical features illustrated in Figure 3.1 and table 3.1, your Thomson Speed Touch 2030 also provides the following:

### Technical specifications:

- An integrated 2 Ethernet ports switch that allows the telephone and a computer to share a single Ethernet jack.
- A direct connection to a 10BaseT or 100BaseT Ethernet (RJ45) network (half or full duplex connections are supported).
- G.711, G.723 and G.729ab voice compression standards
- In band Dual-Tone Multi Frequency (DTMF) support for touch-tone dialing
- Out-of-band DTMF signalling for codecs that do not transport the DTMF signalling correctly (for example, G729 or G729a)

### Configuration:

- IP Number addressing: manually configured via local setup menu, static or dynamic IP configuration (integrated Dynamic Host Configuration Protocol DHCP)
- Configuration support:
  - Local & remote warm reboot
  - Network start up via DHCP and TFTP (Trivial File Transfer Protocol)
  - TFTP/HTTP sever download
  - Web browser management
  - Password protection for configuration

### Phone set function support and Call Options:

- Call hold, Call Transfer, Call Forward, Conference Call, Call Park and Call Pick-up.
- On-hook dialing, Dial from call log, multi line (5 lines)
- Redial, Mute, Call log, phone book (30 entries), 10 Speed dial memory keys
- Hands free (full-duplex)

## 10. Supported Protocols

Your Thomson Speed Touch 2030 SIP Phone supports the following standard Internet protocols:

### **• Internet Protocol (IP)**

IP is a network layer protocol that sends datagram packets between nodes on the Internet. IP also provides features for addressing, type-of-service (ToS) specification, fragmentation and reassembly, and security.

- **User Datagram Protocol (UDP)**

UDP is a simple protocol that exchanges data packets without acknowledgments or guaranteed delivery. SIP can use UDP as the underlying transport protocol. If UDP is used, retransmissions are used to ensure reliability.

- **Trivial File Transfer Protocol (TFTP)**

TFTP allows files to be transferred from one computer to another over a network.

- **Dynamic Host Control Protocol (DHCP)**

DHCP is used to dynamically allocate and assign IP addresses. DHCP allows you to move network devices from one subnet to another without administrative attention. If using DHCP, you can connect ST2030 IP phone to the network and become operational without having to manually assign an IP address and additional network parameters.

- **Domain Name System (DNS)**

DNS is used in the Internet for translating names of network nodes into addresses. Sip uses DNS to resolve the host names of end points to IP addresses.

- **Hyper Text Transfer Protocol (HTTP)**

HTTP is the underlying protocol used by the World Wide Web. It defines how messages are formatted and transmitted, and what actions Web servers and browsers should take in response to various commands.

- **Simple Network Time Protocol (SNTP)**

SNTP is a simplified version of NTP. SNTP can be used when the ultimate performance of the full NTP implementation described in RFC 1305 is not needed or justified.

- **Network Time Protocol (NTP)**

NTP is an Internet standard protocol that assures accurate synchronization to the millisecond of computer clock times in a network of computers. NTP sends periodic time requests to servers, obtaining server time stamps and using them to adjust the client's clock.

- **Simple Network Management Protocol (SNMP)**

SNMP is a set of protocols for managing complex networks. It works by sending messages, called protocol data units (PDUs), to different parts of a network. SNMP-compliant devices, called agents, store data about themselves in Management Information Bases (MIBs) and return this data to the SNMP requesters.

- **Address Resolution Protocol (ARP)**

ARP is a network layer protocol used to convert an IP address into a physical address (called a DLC address), such as an Ethernet address. A host wishing to obtain a physical address broadcasts an ARP request onto the TCP/IP network. The host on the network that has the IP address in the request then replies with its physical hardware address

- **Transmission Control Protocol (TCP)**

TCP is one of the main protocols in TCP/IP networks. Whereas the IP protocol deals only with packets, TCP enables two hosts to establish a connection and exchange streams of data. TCP guarantees delivery of data and also guarantees that packets will be delivered in the same order in which they were sent.

- **Session Description Protocol (SDP)**

SDP is a protocol that defines a text-based format for describing streaming media sessions and multicast transmissions. SDP is not a transport protocol but a method of describing the details of the transmission. For example, a SDP file contains information about the format, timing and authorship of the transmission, name and purpose of the session, any media, protocols or codec formats, the version number, contact information and broadcast times.

## 11. Abbreviations

<b>SIP</b>	Session Initiation Protocol. An IP telephony signalling text-based protocol developed by the IETF
<b>DNS</b>	Domain Name Server
<b>DHCP</b>	Dynamic Host Control Protocol
<b>FTP</b>	File Transfer Protocol
<b>H323</b>	An ITU standard for realtime voice and videoconferencing over packet networks, including LANs, WANs and the Internet
<b>LAN</b>	Local Area Network
<b>MGCP</b>	Media Gateway Control Protocol
<b>RFC 3261</b>	Request For Comments. This document describes the specifications for business extended services under the MGCP protocol.
<b>TFTP</b>	Trivial File Transfer Protocol
<b>XML</b>	EXtensible Markup Language
<b>DTMF</b>	<b>Dual Tone Multi-Frequency</b> The system used by touch-tone telephones. DTMF assigns a specific frequency (consisting of two separate tones) to each key so that it can easily be identified by a microprocessor.

## Chapter 3 Phone operations

### 1. General appearance

Mise en forme : Puces et numéros

The general appearance of the ST2030 is as below:



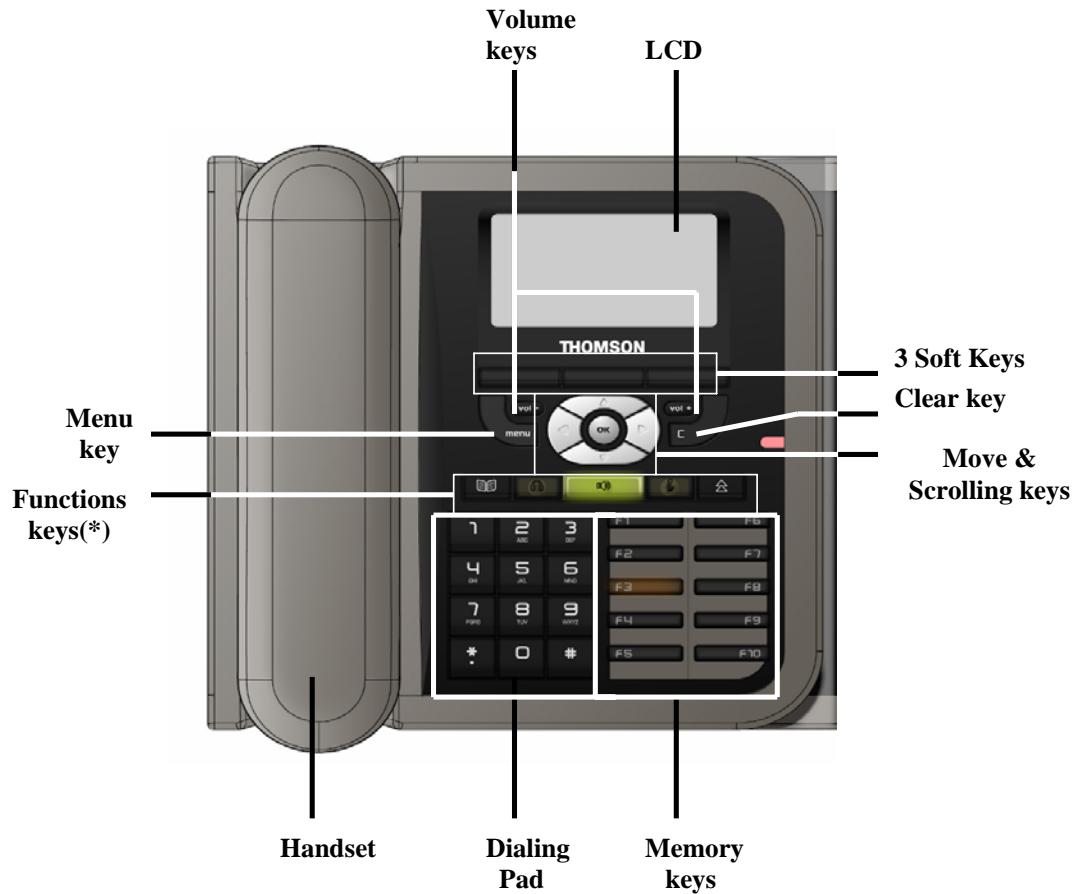
ST2030 front view



ST2030 with his extension module

## 2. Physical Features

Figure 3.1 and Table 3.1 illustrate physical features of the ST2030 SIP Phone:



\* Keys from left to right: phone book/ headset/ speaker/ mute and redial.

**Figure 3.1 The IP Phone User panel**

### a) Display keys



**Confirmation key:**



**Clear key:**

Clear characters in editing mode or exit to standby display (long click) or return back to the previous page.



**Menu key:**

Enables access to menu



**Phone book key:**

Enables to access the phonebook and view the names and phone numbers the user wants to dial.



**Scrolling and move keys:**

Enable browsing setting options, display the latest 30 call numbers in standby mode and move among soft keys options in editing mode.



**3 Soft keys:**

Activate the features described by the text message directly above on the LCD screen.



**LCD Screen:**

Displays information about the phone settings, such as the number dialing out or calling in, date, time, calls status, call duration timer etc. It is a 6-line of 20 characters LCD screen.

### b) Audio keys

#### **Volume keys:**



Adjust the volume of the handset, headset, speaker phone, ringer phone.



#### **Speaker key:**



Activates/disables hands free or group-listing function and switch on/off the microphone



#### **Headset key:**

Activates headset mode during a call



#### **Mute key:**

Activates/deactivates mute function.

### c) Dialing keys



#### **Dialing pad**

Press the dial pad buttons to dial a number. Dial pad buttons work exactly like those on your existing telephone. In the editing mode, it can be used to input characters.



#### **Redial key:**

Calls the last number dialed.



#### **Memory keys:**

Are used as Speed dial keys.

#### **Handset:**

Makes and receives calls

**Table 3.1    The IP Phone physical features**

### 3. Display

The product has a full dot matrix LCD screen with a resolution of 128\*64 pixels. Its screen is monochrome and not backlit.

The viewing area of the screen is 70.7x38.8 mm (active area: 66.52x33.24 mm).

It is able to display characters defined by “Thomson-CharDisplayRequirement-Ed01-03November2004.doc”.

### 4. Definitions of Soft Keys

The table 3.2 describes the main functions of each soft keys you can use on the SIP phone.

<b>Soft Key</b>	<b>Brief Description</b>
<b>A/a/1</b>	Input mode switch in edit state
<b>Active</b>	Activate option/service
<b>Admin</b>	Enter to administration sub-menu
<b>Answer</b>	Answer an incoming call
<b>Anym</b>	Shortcut to "Anonymous"
<b>Autoan</b>	Shortcut to "Auto Answer"
<b>Back</b>	Return to previous menu
<b>CalLog</b>	Shortcut to Call log
<b>Cancel</b>	Cancel an action or exit to previous menu without applying changes
<b>CBack</b>	Perform Callback function
<b>CBlock</b>	Shortcut to "Call Block"
<b>CFwd</b>	Shortcut to "call forwarding"
<b>CidDis</b>	Shortcut to "Number Display"
<b>Change</b>	Change phone password
<b>Conf</b>	Create a conference call. During a call/conference and another call is coming, press the soft key will join a conference
<b>DelChr</b>	Delete character in edit mode
<b>Delete</b>	Delete specific entry
<b>Detail</b>	Show detailed information
<b>Dial</b>	Dial an entered phone number
<b>DNDst</b>	Do Not Disturb
<b>Down</b>	Contrast adjustment
<b>Dsub</b>	Shortcut to "Dial Subscriber"
<b>EndCal</b>	Terminate active call
<b>Edit</b>	Edit parameter content
<b>Exit</b>	Return to previous menu
<b>Format</b>	Change display format of specific parameter, such like Date and Time etc.
<b>Hold</b>	Hold current active line

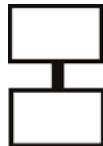
<b>Join</b>	Join several connected calls to create a conference call
<b>Lock</b>	Shortcut to lock phone
<b>Mail</b>	Dial to voice mail server
<b>MsCall</b>	Shortcut to missed call list
<b>NewCal</b>	Make a new call
<b>OFF</b>	Set specific service OFF
<b>OK</b>	Confirm setting
<b>ON</b>	Set specific Service ON
<b>Option</b>	Shortcut to phone services
<b>Park</b>	Store a call using Call Park
<b>PhBook</b>	Shortcut to phone book
<b>PickUp</b>	Answer a call on another phone
<b>Play</b>	Play music or melody
<b>Reject</b>	Reject a call
<b>Remove</b>	Remove a conference participant
<b>Resume</b>	Resume to original call from call transfer
<b>RtPark</b>	Retrieve Park
<b>Save</b>	Save the chosen setting
<b>Select</b>	Select current item on the screen
<b>Stop</b>	Stop music or melody
<b>Transf</b>	Transfer a call
<b>TrMail</b>	Transfer call to voice mail system
<b>UnHold</b>	Unhold a held line
<b>Up</b>	Contrast adjustment
<b>User</b>	Enter to user submenu
<b>View</b>	Show details when data item content is more than LCD can display in one line

**Table 3.2 Definitions of soft keys**

## 5. Icons & Indicators

The SIP Phone has a 6-line of 20 characters LCD display

- Line 1 : date/time and icons information
- Line 2 : line 5: operation information
- Line 6 : Soft keys display

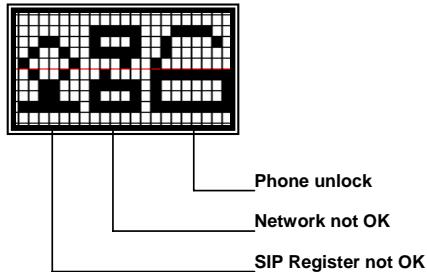


→ Network status is okay



→ SIP register that is okay

→ “Alarm” setting is enabled

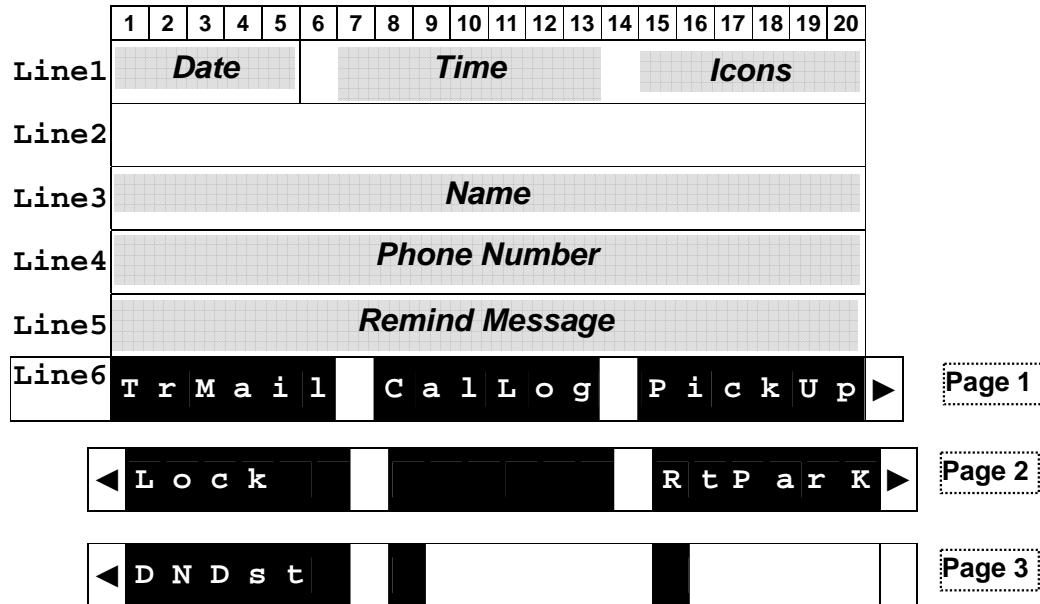


One or more of the following icons may be shown when a service or function is activated.

## 6. Main Displays

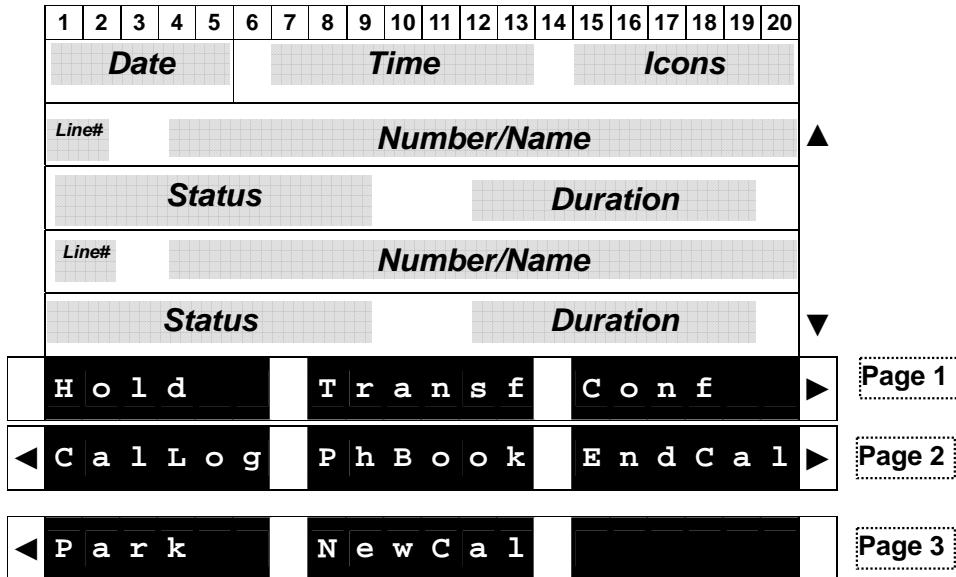
### a) Display in idle mode

#### Home page



*b) Display during a call*

Phonebook and call log can be displayed during a call.



## 7. Definitions of LED

LED Indicator	Condition	Color	Status
Speaker	Default	Red	OFF
	During FW boot		ON
	Loudspeaker is activated during a call		ON
Headset	Default	Red	OFF
	During FW boot		ON
	Headset or Headset group-listing mode is activated during call		ON
Mute	Default	Red	OFF
	During FW boot		ON
	Mute is active		ON
Feature keys	Default	Green	OFF
	During FW boot		ON
	Used for speed dial keys in standby mode		OFF
	Line appearance: standby mode		OFF
	Line appearance: a call is incoming		Fast blinking
	Line appearance: is active		ON
	Line appearance: hold or remote hold		Slow blinking
System	Boot: during FW boot	Red	Red ON
	Boot: during DHCP process (if any)		Red Slow blinking
	Boot: during SIP registration		Red Fast blinking
	Running mode: Ethernet connection is down		Red Slow blinking
	Running mode: IP connection is down (e.g. no IP address allocated)		Red Slow blinking
	Running mode: SIP connection is down (e.g. SIP registration or registration refresh failure)		Red Fast blinking
	Auto-configuration: during configuration file download		Red Fast blinking
	Auto-configuration: during firmware file download		Red Fast blinking
	System is programming FLASH		Red ON
	Ringing state		Green Slow blinking
	Alert messages and missed calls		Green Slow blinking

Table 3.3 Definitions of LED

## **8. Extension module**



The extension module features 28 keys. One can plug up to 2 extension modules in a row, leading to a total of  $10 + 2 \times 28 = 66$  possible “multiline” keys.

## **9. Memory**

The ST2030 has 4MB of Flash, and 16MB of RAM

## **10. Power supply**

The ST2030 could be powered by either a main power unit or PoE.

The power plug and PoE can be plugged at the same time. The power plug has priority, and PoE takes over in case of power failure.

Its power consumption is under 6W.

### 1. Talking Mode & Operations

#### a) 5 kinds of communication mode

- Handset mode
- Hands free mode
- Headset mode
- Handset group listening
- Headset group listening

#### b) Operation of off hook

There are four ways supported for Off Hook operation

- Picking up the handset: Handset mode
- Press **Speaker** key: Hands free mode
- Press **Headset** key: Headset mode
- Press **F1** when phone is ringing

#### c) Operation of on hook

- Handset mode: replacing the handset
- Handset -Group-Listening mode: press Speaker key and replace the handset
- Hands free mode: press Speaker key
- Headset/Headset-Group-Listening mode: press Headset key

#### d) Operation of mode switch during call

	<b>Handset</b>	<b>Hands free</b>	<b>Headset</b>	<b>Handset -GL</b>	<b>Headset-GL</b>
<b>Handset</b>		1. Press Speaker key 2. Replace handset	Press Headset key	Press Speaker key	N/A
<b>Hands free</b>	Pick up handset		Press Headset key	N/A	N/A
<b>Headset</b>	Pick up Handset	1. Press Speaker key 2. Press headset key		N/A	Press Speaker key
<b>Handset -GL</b>	Press Speaker key	Replace handset	N/A		Press headset key
<b>Headset-GL</b>	N/A	Press headset key	Press Speaker key	Pick up Handset	

**Table 4.1 Operations of mode switch during call**

## 2. Main Call Functions

### a) Make a call

#### On-hook dialing

In standby mode, there are several ways to show dial number.

- Press  after / before pressing digital keys directly
- In standby mode, press “

<b>Key Pressed</b>	<b>LCD display</b>
“  	
If number is stored in phone book, the associated name will be displayed.	

- Press one of speed dial keys (i.e. feature keys) **F1 – F10** to display number
- Query phone book: press 
- Query call log (**dialed, received, missed**)
- Query memory key (**F1 - F10**)

And then you can trigger dialing by any one of following.

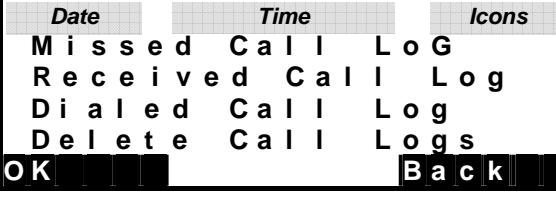
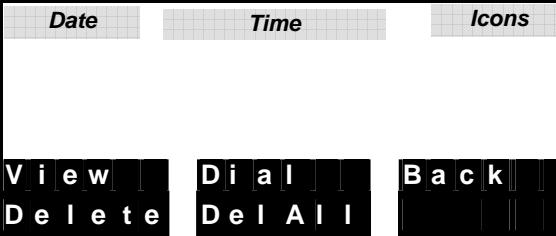
- Press soft key **Dial**
- Press **OK** key
- Go off-hook

### Off-hook dialing

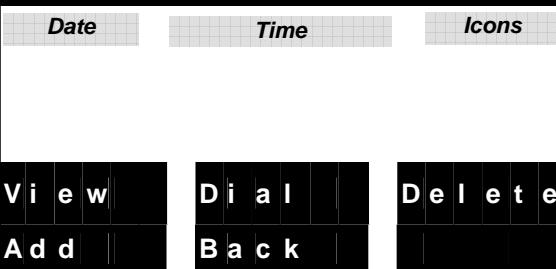
Performing off-hook operation first, and then you can dial out by any one of following ways.

- Press digital keys
- Press Redial key
- Press any one of memory keys F1 to F10

#### ➤ Use the call log

Key Pressed	LCD display
Press <b>CalLog</b> to show Call Log messages Scrolling and confirm by pressing <b>Ok</b>	 <p>Date Time Icons</p> <p>M i s s e d C a l l L o g</p> <p>R e c e i v e d C a l l L o g</p> <p>D i a l e d C a l l L o g</p> <p>D e l e t e C a l l L o g s</p> <p>O K      Back</p>
“↑”, “↓”: Scrolling and dial number you desired and press “Dial» or <b>Ok</b>	 <p>Date Time Icons</p> <p>V i e w      D i a l      B a c k</p> <p>D e l e t e      D e l A l l</p>

#### ➤ Use the phonebook

Key Pressed	LCD display
Press <b>Book</b> or <b>PhBook</b> to show phone book information. “↑”, “↓”: Scrolling and press “Dial» or <b>Ok</b>	 <p>Date Time Icons</p> <p>V i e w      D i a l      D e l e t e</p> <p>A d d      B a c k</p>

When making a new call (by going off hook or pressing )

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date	Time	Icons																	
Enter a Number																			
<i>Dialed number display</i>																			
  																			

Dial phone number display

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date	Time	Icons																	
Enter a Number																			
<i>Dialed number</i>																			
																			

When a call is connecting

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date	Time	Icons																	
Connecting...																			
																			

### When a call is connected

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date	Time		Icons																
	R i n g i n g . . .																		

### **When a call is failed due to party (number) does not exist**

### **When a call is failed due to other reasons**

## When a call is put on hold

## Last number Redial:

- Pick up the handset and press 
- Press  then 

## b) Speed dial

### Save memory keys

- Press any one of **F1 – F10** in standby mode
- Press **Edit**
- Key in number you want to save
- Press **Save** to store the entry

The memory keys could be modified too in a submenu.

### Speed dialing – on-hook dialing

- Press any one of **F1 – F10** in standby mode
- On-hook dialing : go off-hook, re-press the feature key, or press **OK** key

### Speed dialing – off-hook dialing

- Perform off-hook action
- Press any one of **F1 – F10** in dialing mode

## c) Answer a call

When a call is coming, the caller number and alias are displayed, if the caller number is stored in the phonebook, the associated name is displayed.

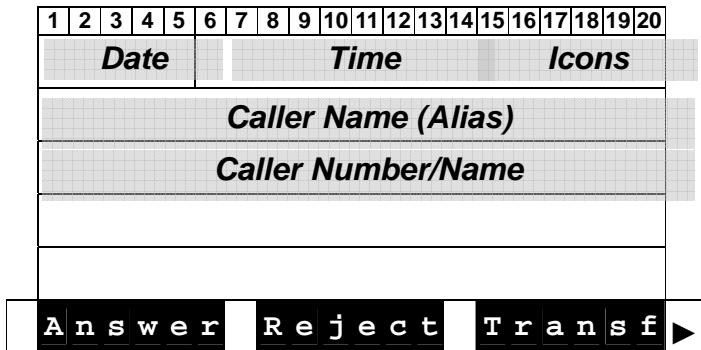
While the phone is ringing, you can perform off-hook action or press **Answer** to answer the call



Incoming call display is visible only when phone is in talking state or in standby mode.

Note

## When the phone is ringing



### **d) Reject a call**

- Press **Reject** when the phone is ringing

### **e) Call back**

Executes Auto Call Back when the called party is busy or does not reply.

ringing on any other station within his own predefined “pickup group”

When you fail to make a call due to destination is “Busy”, you can activate callback function to complete the call automatically in standby mode.

Press **Menu** and **Option** then select “DialSubscribe” to view the status.

You can do modifications by pressing **Change** to enter the phone number you want to auto call back, the time of the call and the status.

## Operation of Call Completion to Busy Subscriber (CCBS)

- Place a call and you hear busy tone
- Press soft key **CBack**
- Replace handset (i.e. Go on-hook, and CCBS is activated)

### When CCBS is successful

- The phone is ringing
- Go off-hook.
- You hear ring-back tone if party doesn't answer yet. Or start conversation if part answers the call.
- The party will hear ring-back tone, if you don't go off-hook yet.

### You can cancel CCBS function by any one of following ways.

- Go off-hook
- Press any digital key

### *f) Call forward*

The function of the Call Forward processed the line forwarding to another phone number. The user can set the system to another phone number. When a call is coming, the system will forward the line to another phone number that is set previously. The system supports three kinds of ways to implement this function including “Unconditional”, “Busy”, and “No Answer”.

**Unconditional:** Provided that a call is coming, the system will forward the line immediately.

**Busy:** A call is coming and the user is on a busy line. Then, the system will forward the calling line.

**No Answer:** On condition that the user does not respond to the calling for 30 seconds, the system will forward the line to another phone number.

Press **Menu** and **Option** and select “Call Forward” to view the status of Call Forward you have. You can change this status by pressing **Change**. Enter the phone number towards which you want to forward your calls. Once this process is completed, press up/down of the cursor to the Call forward type. Finally, press **Save**.

### *g) Transfer Message to Voice Mail*

You can transfer all the calls you receive to your Voice Mail.

Press **TrMail**, then activate the transfert of your messages to your Voice Mail Box.

Press **Change** until obtaining “TrToVoiceMail-ON”, finally validate it by pressing **OK**

### 3. Call functions during conversation

#### a) Hold and Retrieve a call

- During a call, press **Hold** to put a call on hold.
- Press **Unhold** to retrieve a held call.

#### b) Call Transfer

During conversation mode, you can press the soft key **Transf** to activate Call-Transfer service.

##### Operation of “Blind” transfer

- Press **Transf**
- Dial the desired phone number to which you want to transfer the current call
- Press **Transf**
- Hang Up

##### Operation of “Attended” transfer

- Press **Transf**
- Dial the desired phone number to which you want to transfer the current call
- Wait for the transfer recipient to answer.
- Press **Transf**, if the recipient accepts the transferred call.
- Press **Back**, if the recipient refuses the transferred call.

After pressing once **Transf**

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20															
Date	Time	Icons																																
Enter A Number																																		
<i>Dialed number display</i>																																		
Ph Book	Call Log	Back																																

When the callee picks up

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20															
Date	Time	Icons																																
Talking . . .																																		
Transf		Back																																

*c) Conference call*

The Speed Touch 2030 SIP phone allows a 3-way conference.

Start a conference by calling other party:

- During conversation, Press **Conf**
- Dial the desired phone number (see the LCD display below)
- When the callee picks up, press **Conf** to add the first and second calls into the calling party.

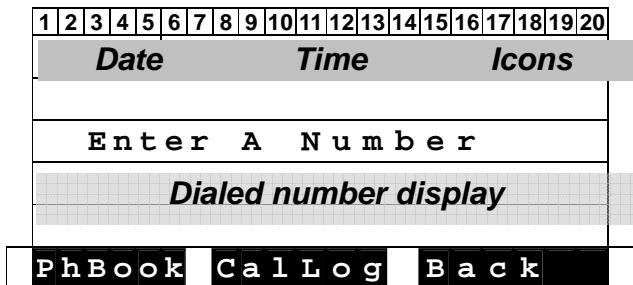
Drop any one participant of the conference call:

- Scrolling to the line
- Press **Remove** to drop the held participant.

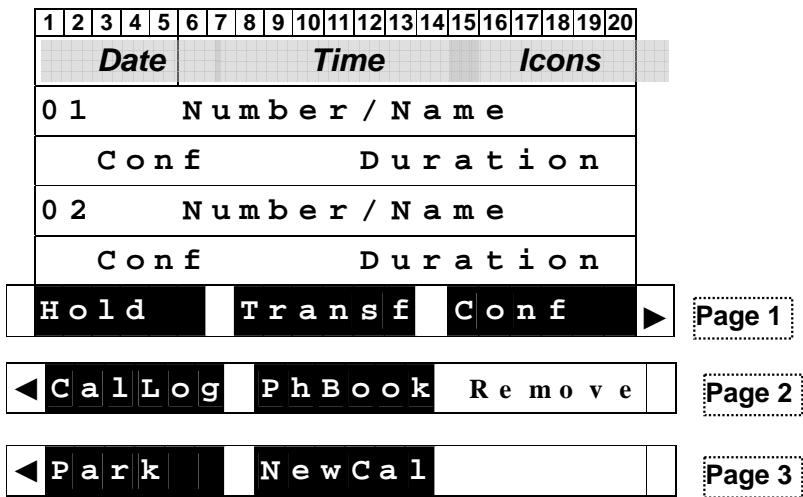
End the conference call:

- Clear each participant one by one.

When pressing [Conf] once.



When a call/conference is active



## When the callee picks up

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20				
<b>Date</b>		<b>Time</b>				<b>Icons</b>																	
T a l k i n g . . .																							
<b>Conf</b>																	<b>Back</b>						

## *d) Volume adjustment*



While the phone is ringing you can adjust ring volume by pressing the volume keys.  
During conversation, you can also adjust volume of any of 5 modes by pressing volume keys.

## *e) Mute:*



During conversation, press mute key to prevent the callee from hearing what you say.

## *f) Multi line*

### Display of Line/Call status

The IP phone supports up to 10 multi-lines that is configurable by administrator via Web-Page access. Each line can be separately operated various services, including answer, reject, hold, transfer and conference call etc. When the phone is in conversation mode, following is the Line/Call status displayed on LCD.

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date					Time					Icons									
Line#		Number/Name													▲				
Status					Duration														
Line#		Number/Name														▼			
		Status					Duration												
		H o l d		T r a n s f		C o n f										▶	Page 1		
◀	C a l L o g		P h B o o k		E n d C a l												Page 2		
◀	P a r k		N e w C a l														Page 3		



If we have more than 2 lines connected, up and down arrows to display other lines.

Note \_\_\_\_\_

#### Definition of Status

- Talking
- Hold: hold line
- R-Bye: remote terminates the line
- R-Hold: remote side holds the line
- Conf: conference call

#### **g) Switch Between Line/Call**

The IP phone supports up to 10 multi-lines that is configurable by administrator. Each line can be switched by pressing line keys (i.e. keys **F1 – F10**). Besides the current active line, all others are held by IP phone and can be resumed by pressing associated line key or pressing proper soft key. And the first call is always assigned to line 1.

Press active line key will terminate the call.

<For example >

When you are communicating with one party and wish to make a brief to another party.

- Press a free line key (then you will hear dial tone, and the original call is held)
  - Key in number you want to call
- Or
- Press **NewCal** (then you will hear dial tone, and the original call is held)
  - Key in number you want to call

### **Answer an incoming call when another call is active.**

- The phone will assign a free line dedicated to the incoming call, and the associated LED will blink fast.
  - Press the line key, and the original line is held by phone automatically
- Or
- Scrolling the ringing line
  - Press **Answer** to answer the call

### **Switch between lines.**

- Press the line key you desire, and the original line is held by phone automatically
- Or
- Scrolling to the line
  - Press **UnHold** to resume the call

## ***h) Pick up***

A station user may dial a special code to answer any incoming calls

- Use menu (table, arborescence)
- Use HD, SP, HD, Adjust volume

### **To pick up call on another phone.**

- Press **PickUp** key in standby mode
- Key in the phone number. You are connected with the caller.
- Press OK

## ***i) Park up***

Somewhat similar to the Call Hold feature; however, once a call is placed in the “park” condition, any station within the system may retrieve it by either dialing the appropriate access code.

Park a call at the phone, and resume the call at another phone.

### **Set Call park**

- Press **Park**
- Replace the handset

### **Retrieve Call park on an other phone**

- Press **RtPark** in standby mode
- Key in the parked phone number
- Press OK

## 4. Phone book & Call log

### 1 Call logs

Press **CalLog** to view, add, clear call logs and dial from call logs.

There are 3 kinds of call logs: Missed call logs, Received call logs and dialed call logs.

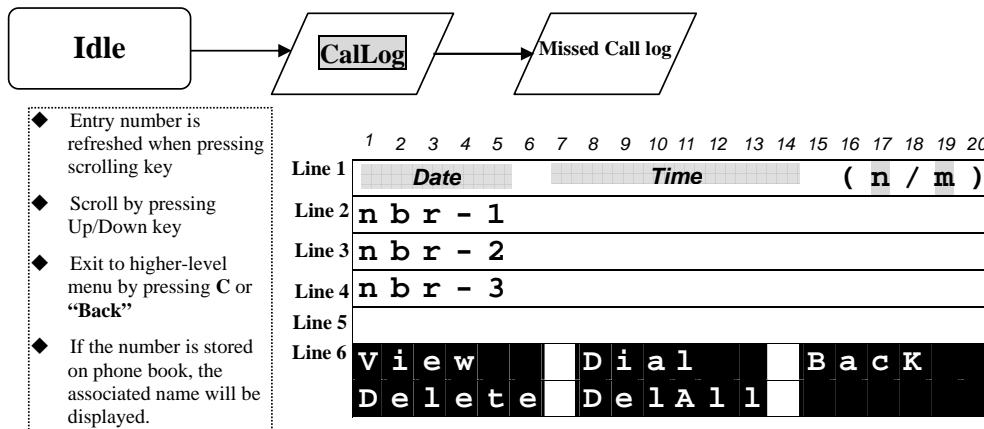
You can also dial from call logs by pressing **OK** on selected call.

Key Pressed	LCD display															
Press <b>CalLog</b> to show Call Log messages	<table border="1"><thead><tr><th>Date</th><th>Time</th><th>Icons</th></tr></thead><tbody><tr><td>M i s s e d</td><td>C a l l</td><td>L o g</td></tr><tr><td>R e c e i v e d</td><td>C a l l</td><td>L o g</td></tr><tr><td>D i a l e d</td><td>C a l l</td><td>L o g</td></tr><tr><td>D e l e t e</td><td>C A L L</td><td>L o g</td></tr></tbody></table>	Date	Time	Icons	M i s s e d	C a l l	L o g	R e c e i v e d	C a l l	L o g	D i a l e d	C a l l	L o g	D e l e t e	C A L L	L o g
Date	Time	Icons														
M i s s e d	C a l l	L o g														
R e c e i v e d	C a l l	L o g														
D i a l e d	C a l l	L o g														
D e l e t e	C A L L	L o g														
Scrolling and confirm by pressing <b>Select</b>	<table border="1"><tr><td><b>Select</b></td><td><b>Back</b></td></tr></table>	<b>Select</b>	<b>Back</b>													
<b>Select</b>	<b>Back</b>															

#### Options list

No.	Option Message	Comment
(1)	<b>Missed Call Log</b>	30 entries supported
(2)	<b>Received Call Log</b>	30 entries supported
(3)	<b>Dialed Call Log</b>	30 entries supported
(4)	<b>Delete all Logs</b>	

### 2 Missed Calls

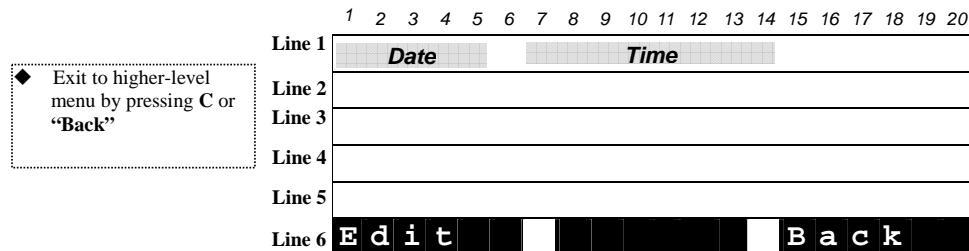


When pressing feature key, the behavior is the same as in Phone book.

During the Query, you can press **Delete** to delete the displayed entry..

Press **View** to entry sub-menu as following.

When you have pressed **View**

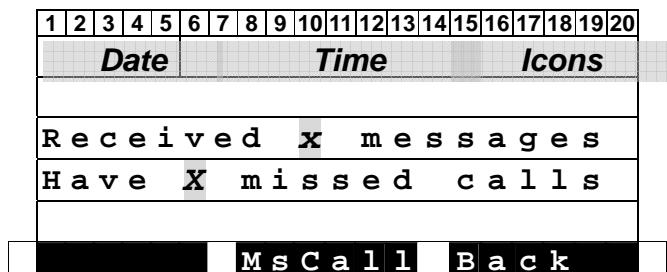


When pressing feature key, the behavior is the same as in Phone book.

## **Content of missed call Parameters**

Item Name	Data Format	Max. Length	Comment
Number	Numeric	24	While length of number is more than that LCD can display, press “ <b>←</b> ” or “ <b>→</b> ” to get remainder.
Alias (Phone name)	Alphanumeric	20	
Date Time	YY/MM/DD hh:mm		

Display when there are any message or missed calls



- The reminding message “**Received X messages**” is always displayed in standby mode till you press softkey **Back** or **Cancel** key.
  - The reminding message “**Have X missed calls**” is always displayed in standby mode till you press softkey **MsCall**, **Back** or **Cancel** key.

### 3 Received Call & Dialed Calls

Be the same as operation of Missed Calls

### 4 Phone book



Press to access the phone book.

You can view, delete and add phone numbers in the phonebook.

You can also dial from phonebook.

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1																				
Line 2																				
Line 3																				
Line 4																				
Line 5																				
Line 6	V	i	e	w			D	i	a	l			D	e	l	e	t	e		
	A	d	d				B	a	c	k										

The user is able to search and retrieve a phone number without having to scroll between all the entries by typing on a letter. For instance, if the user presses quickly 2 times on key '2' (corresponding to "b" letter), the first name beginning with "b" will be displayed on top of screen.

When pressing feature key, then replace current memory display by item to be memorized is displayed. And corresponding softkeys **Replace**, **Edit** and **Back** are displayed.

Edit function is for editing the feature key.

#### Content of Phone book entry

Item Name	Data Format	Options List	Max. Length	Comment
Name	Alphanumeric		20	
Number (TelNbr)	numeric		24	While length of number is more than that LCD can display, press "←" "→" to get rest.
RingType (Bell)	Options	Ringer-1 Ringer-2... Ringer-n Music-1 Music-2		

## 5. Character Input- Edit Mode

The editor is automatically activated whenever you are modifying content of configurable parameters. Apart from entering characters, you can use this editor to navigate through text or delete characters. Even though the maximal text length is not limited by the number of characters on the display because the text can be shifted to the left and right, there are limits for the certain scenarios. E.g. for phone book name, the limit is 15. For number, it is 31. For IP address, it is 15. The following overview shows all characters that can be entered. Number keys must be repeatedly pressed until the required letter appears; the cursor advances after a short delay (one second) or pressing any other key(0..9,\*,#, up, down etc.) indicating that the required letter has been accepted.

Key	Lower case	Upper case
1	1 . , = + - & ^	
2	a b c 2 à á â á ç	A B C 2 Ä Å Å Æ
3	d e f 3 è é ê ë	D E F 3 É
4	g h i 4 ï í î ï	G H I 4
5	j k l 5	J K L 5
6	m n o 6 ñ ò ó ô õ ö ø	M N O 6 Ñ Ø
7	p q r s 7	P Q R S 7
8	t u v 8 ù ú û	T U V 8 Ü
9	w x y z 9 ÿ	W X Y Z 9
0	0 Space @ % () [] < >	
*	* # ? ! : ; ‘ “ _ /	
#	#	



Softkey **ABab12** : Switch uppercase (with indicator “>ABC”), lowercase (with indicator “>abc”) and numeric (with indicator “>123”) when entering letters.

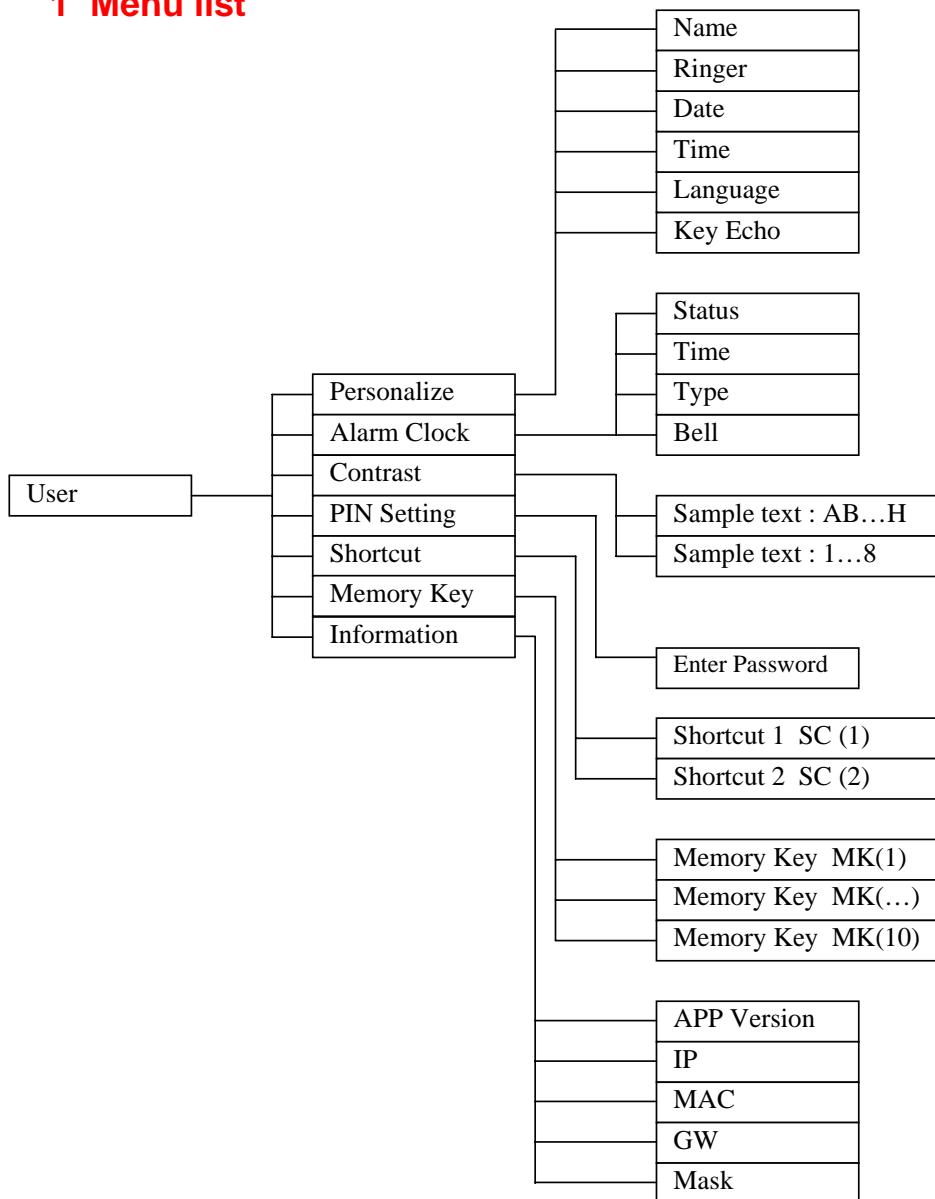
Note

All other symbol needed will be appended to '\*' key.

**Table 4.2      Character input method table**

# Chapter 5 Configuration through LCD

## 1 Menu list



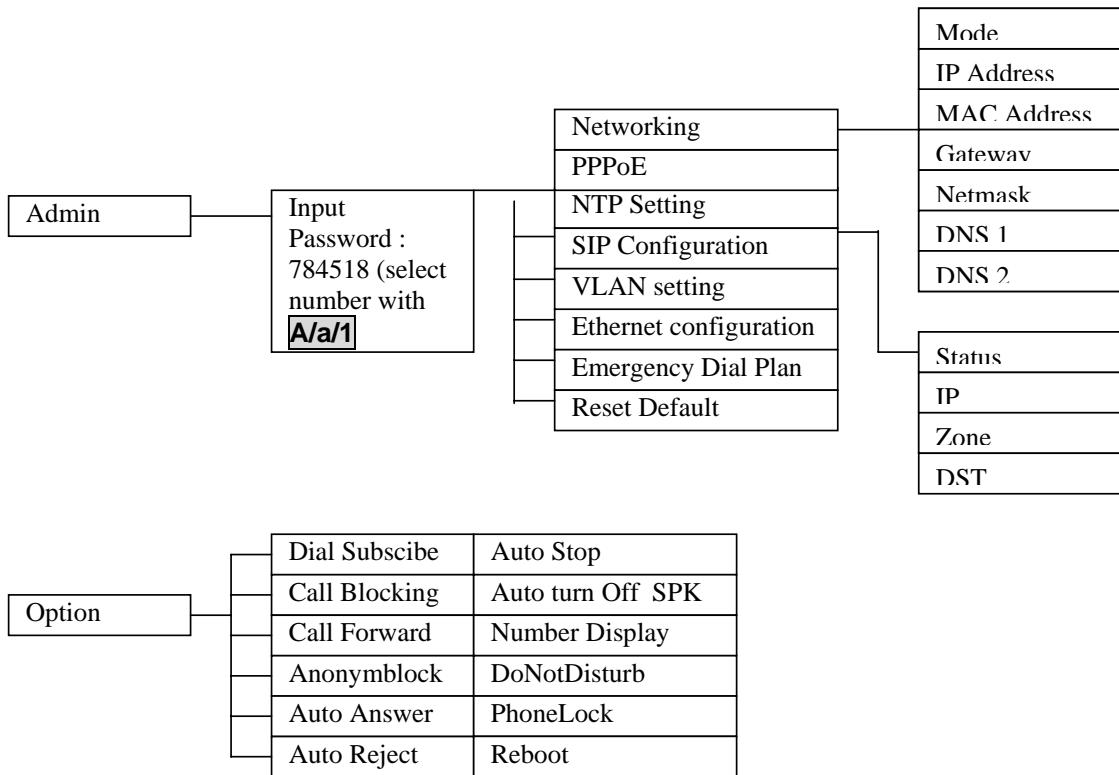
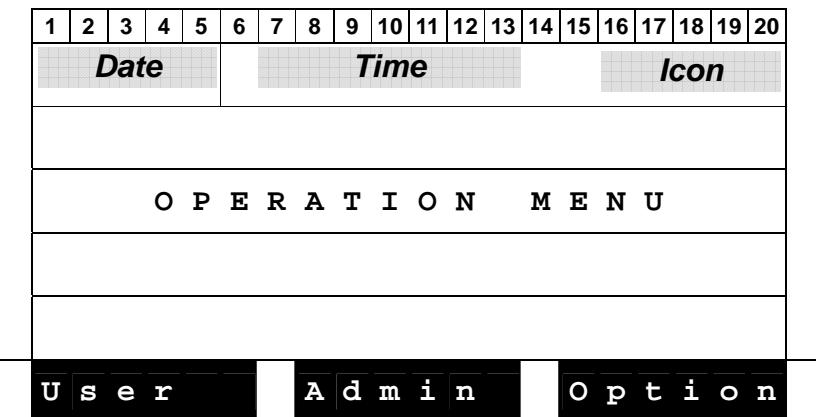


Figure 5.1 Operation Menu LCD Display

## 2 Operation menu display

### a) Home operation menu display



## b) User Menu Display

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1																			
Line 3																			
Line 4																			
Line 5																			
Line 6	S	e	l	e	c	t									B	a	c	k	

### Sub-menu list of User Setting

No.	Option Message	Comment
(1)	<b>Personalize</b>	Set date/time format, Key-echo, language and Ringer type
(2)	<b>Alarm Clock</b>	Activate Alarm clock, set bell and time
(3)	<b>Contrast</b>	
(4)	<b>PIN Setting</b>	Change personal phone password
(5)	<b>Shortcut</b>	Shortcut setting
(6)	<b>Memory Keys</b>	Configure Speed dial keys
(7)	<b>Information</b>	Software Version, IP, MAC, Gateway , Mask

## c) Admin Menu Display



You have to input a password to access Admin settings: **784518**  
Don't forget to switch from uppercase to numeric ABC to 123

Caution

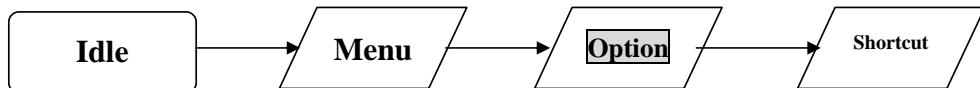
- ◆ Scroll by pressing Up/Down key
- ◆ Select by pressing **OK** or “**Select**”
- ◆ Return to idle state by pressing C or “**Back**”

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20																		
Line 1	Date		Time																Icon																			
Line 2																																						
Line 3																																						
Line 4																																						
Line 5																																						
Line 6	S	e	l	e	c	t			B	a	c	k																										

#### Sub-menu list of admin setting

No.	Option Message	Comment
(1)	<b>Networking</b>	
(2)	<b>PPPoE</b>	Invisible when <b>PPPoE</b> function is disabled
(3)	<b>NTP Setting</b>	
(4)	<b>SIP Configuration</b>	
(5)	<b>VLAN setting</b>	
(6)	<b>Ethernet configuration</b>	.
(7)	<b>Emergency Dial Plan</b>	
(8)	<b>Reset Default</b>	

#### *d) Option Menu Display*



- ◆ Scroll by pressing Up/Down key
- ◆ Select by pressing **OK** or “**Select**”
- ◆ Return to idle state by pressing C or “**Back**”

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20																		
Line 1	Date		Time																Icon																			
Line 2																																						
Line 3																																						
Line 4																																						
Line 5																																						
Line 6	S	e	l	e	c	t			B	a	c	k																										

#### Sub-menu list of admin setting

No.	Option Message	Comment
(1)	Dial Subscribe	
(2)	Call Blocking	
(3)	Call Forward	
(4)	Call waiting	
(5)	Anonymblock	
(6)	Auto Answer	
(7)	Auto Reject	
(8)	Auto Stop	
(9)	Auto turn Off SPK	
(10)	Number Display	
(11)	DoNot Disturb	
(12)	PhoneLock	
(13)	Reboot	You need to reboot for saving your modifications

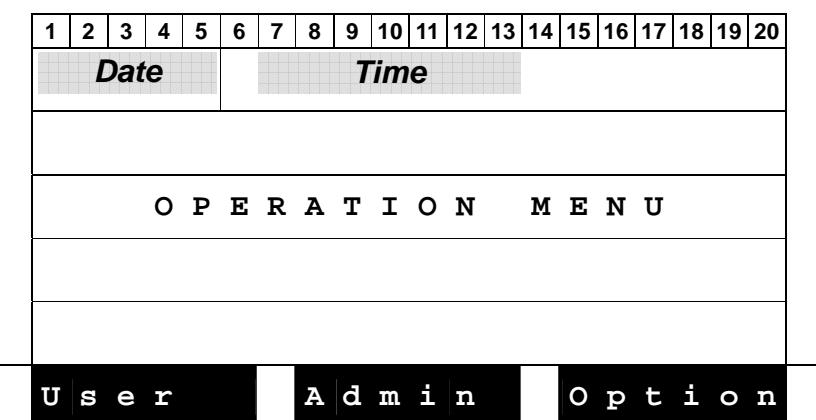
#### Content of Shortcut Parameters

Item Name	Data Format	Comment
X	options	Dial Subscriber -> <b>Dsub</b> Call Blocking -> <b>CBlock</b> Call Forward -> <b>CFwd</b> Anonymous -> <b>Anym</b> Auto Answer -> <b>AutoAn</b> Number Display -> <b>CidDis</b> <b>Else . . .</b>

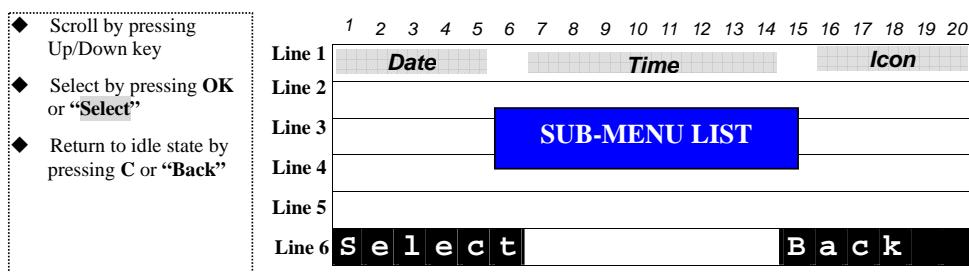
#### e) Detailed LCD menu settings



## Home operation menu display



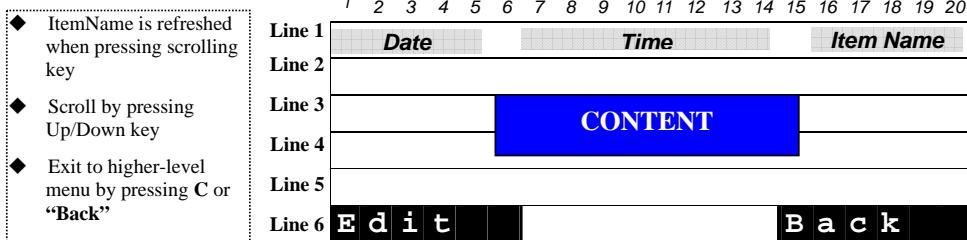
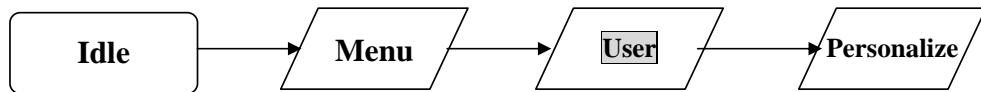
## User Menu Display



### Sub-menu list of User Setting

No.	Option Message	Comment
(8)	<b>Personalize</b>	Set date/time format, Key-echo, language and Ringer type
(9)	<b>Alarm Clock</b>	
(10)	<b>Contrast</b>	
(11)	<b>PIN Setting</b>	Change personal phone password
(12)	<b>Shortcut</b>	Shortcut setting
(13)	<b>Memory Keys</b>	
(14)	<b>Information</b>	Software Version, IP, MAC, Gateway , Mask

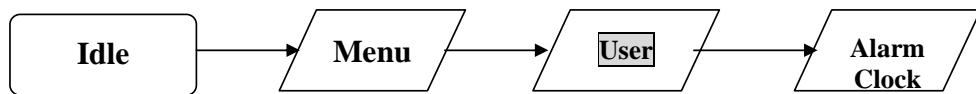
## Personalize Setting



## Content of Phone Option/Parameters

Item Name	Configurable	Data Format	Options List	Comment
Name	Yes	Alphanumeric		Max. length: 15
Ringer	Yes	Options	Ringer-1 ... Ringer-4 Canon Shuffle	You can press soft key <b>Play</b> when editing the content.
Date	Yes (Configurable when NTP is disabled)	MM/DD/YY DD/MM/YY YY/MM/DD		You can press soft key <b>Format</b> to change format. <b>MM/DD/YY</b> <b>DD/MM/YY</b> <b>YY/MM/DD</b>
Time	Yes (Configurable when NTP is disabled)	hh:mm	hh:mm am hh:mm	You can press soft key <b>Format</b> to change format. <b>12 Hours</b> <b>24 Hours</b>
Language (Lang)	Yes	Options	English French Spanish German Italian Portuguese Deutsch	
Key Echo (KeyEch)	Yes	Options	ON OFF	

## Alarm Clock



1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
		Date				Time													
Line 1	Line 2																		
Line 3																			
Line 4																			
Line 5																			
Line 6	E	d	i	t												B	a	c	k

### Content of Alarm Parameters

Item Name	Configurable	Data Format	Options List	Comment
Startup (Status)	Yes	Options	ON OFF	
Time	Yes	hh:mm		
Type	Yes	Options	One Shot Periodic	
Bell Type (Bell)	Yes	Options	Ringer-1 ... Ringer-4 Canon Shuffle	You can press soft key <b>Play</b> to play option when editing the content of <b>Bell Type</b> .

First activate alarm clock then change time.

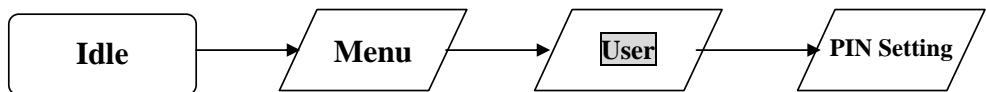
## Contrast



- ◆ Scroll by pressing Up/Down key
- ◆ Exit to higher-level menu by pressing C
- ◆ Press “**Down**” or “**Up**” to adjust contrast
- ◆ Press “**OK**” to confirm and exit to higher-level menu.
- ◆ Press “**Cancel**” to cancel the setting.

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1 <b>Date</b> <b>Time</b>																			
Line 2 <b>S a m p l e   T e x t : A B C D E F G H</b>																			
Line 3 <b>S a m p l e   T e x t : 1 2 3 4 5 6 7 8</b>																			
Line 4																			
Line 5																			
Line 6 <b>D o w n</b>   <b>U p</b>   <b>O K</b>																			

## PIN Setting



1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1 <b>Date</b> <b>Time</b>																			
Line 2																			
Line 3 <b>E n t e r   P a s s w o r d</b>																			
Line 4 <b>* * * *</b>																			
Line 5																			
Line 6 <b>B a c k</b>																			



The password is : 0000

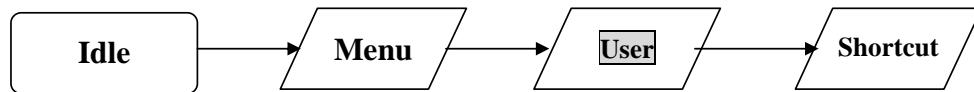
Caution \_\_\_\_\_

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20																
Line 1	Date				Time				Icon																											
Line 2																																				
Line 3	Options list																																			
Line 4																																				
Line 5																																				
Line 6	C	h	a	n	g	e										B	a	c	k																	

### Options list

No.	Option Message	Data Format	Comment
(1)	Flag	ON OFF	<ul style="list-style-type: none"> <li>◆ When Status is ON, the phone lock setting will be authorized by checking personal code.</li> <li>◆ Press <b>Change</b> to enable/disable the item.</li> </ul>
(2)	Change Code		<ul style="list-style-type: none"> <li>◆ Press soft key “<b>Change</b>” to change PSW</li> <li>◆ The default password is <b>0000</b>.</li> </ul>

### Shortcut Setting

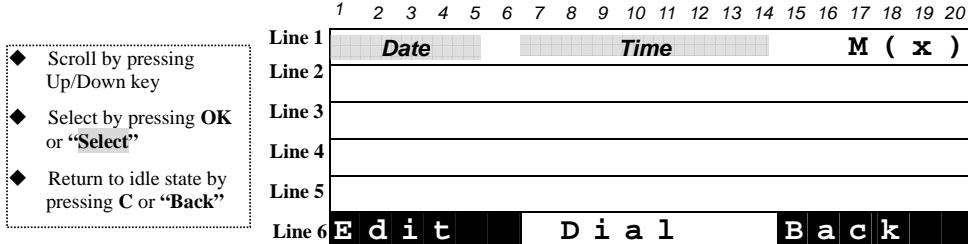
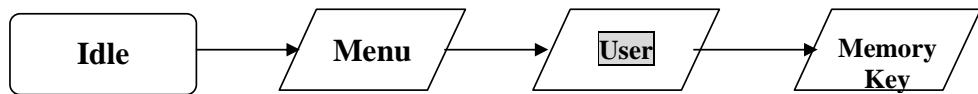


You can edit two shortcuts

### Information



## Memory Keys Setting



## Menu of Admin Settings



You have to input a password to access Admin settings: **784518**  
Don't forget to switch from uppercase to numeric ABC to 123

Caution

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1										Date				Time						
Line 2																				
Line 3																				
Line 4																				SUB-MENU LIST
Line 5																				
Line 6										Select				Back						

### Sub-menu list of admin setting

No.	Option Message	Comment
(7)	<b>Networking</b>	
(8)	<b>PPPoE</b>	Invisible when PPPoE function is disabled
(9)	<b>NTP Setting</b>	
(10)	<b>SIP Configuration</b>	
(11)	<b>VLAN setting</b>	
(12)	<b>Ethernet configuration</b>	.
(13)	<b>Emergency Dial Plan</b>	
(14)	<b>Reset Default</b>	

## Networking Configuration



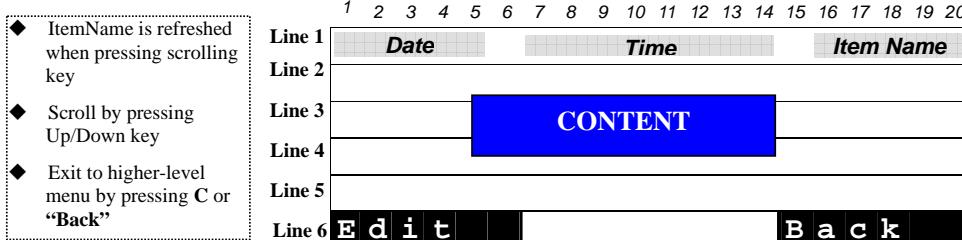
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1										Date				Time						Item Name
Line 2																				
Line 3																				
Line 4																				CONTENT
Line 5																				
Line 6										Edit				Back						

## Content of Network Parameters

Item Name	Configurable	Data Format	Options List	Max. Length
Mode	Yes	Options	Fix IP DHCP PPPoE	
IP	Yes (For Fix-IP only)	xxx.xxx.xxx.xxx		15
MAC	No	xx:xx:xx:xx:xx:xx		
Gateway (GW)	Yes (For Fix-IP only)	xxx.xxx.xxx.xxx		15
Net mask (Mask)	Yes (For Fix-IP only)	xxx.xxx.xxx.xxx		15
Pri DNS	Yes (For Fix-IP only)	xxx.xxx.xxx.xxx		15
Sec DNS	Yes (For Fix-IP only)	xxx.xxx.xxx.xxx		15

## PPPoE Setting

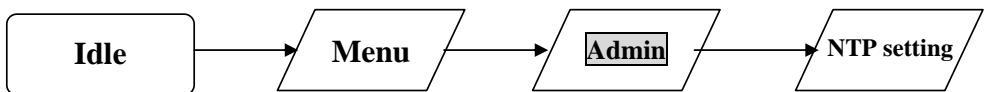
Press Menu key in idle state triggers menu operation.



## Content of PPPoE Parameters

Item Name	Configurable	Data Format	Options List	Max. Length
ACC Name (Acc)	Yes	Alphanumeric		63
PASSWORD (Pwd)	Yes	Alphanumeric		31

## NTP Setting

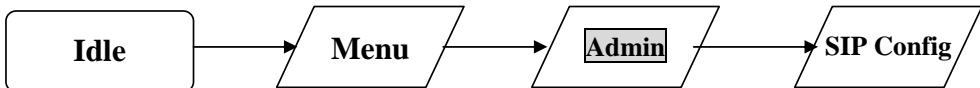


- ◆ ItemName is refreshed when pressing scrolling key
  - ◆ Scroll by pressing Up/Down key
  - ◆ Exit to higher-level menu by pressing **C** or “Back”

## Content of NTP Parameters

Item Name	Configurable	Data Format	Options List	Max. Length
Status	Yes	Options	ON OFF	
IP	Yes	xxx.xxx.xxx.xxx Domain Name		15 63
Zone	Yes	Options	- 12:00 ~ +12:00	
DST	Yes	Options	ON OFF	

## SIP Configuration Setting



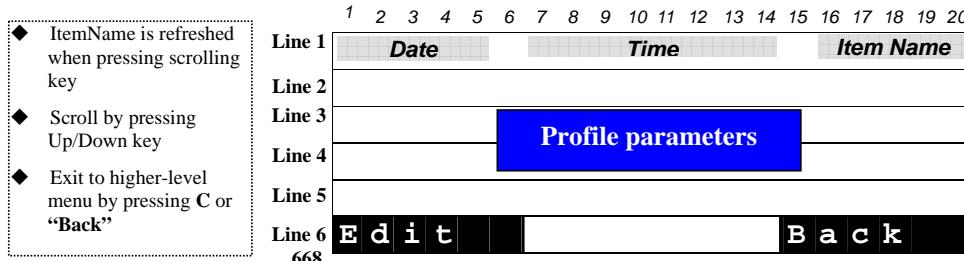
- ◆ Scroll by pressing Up/Down key
  - ◆ Exit to higher-level menu by pressing **C** or “Back”

	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Line 1																		<b>Date</b>	<b>Time</b>	<b>Prof (n)</b>
Line 2																				
Line 3																		<b>Profile list</b>		
Line 4																				
Line 5																				
Line 6	S	e	l	e	c	t		A	c	t	i	v	e		B	a	c	k		

## Profile List

No.	Profile Name	Comment
(1)	<b>Profile 1</b>	◆ Profile name is configurable (refer to "3.4.1 SIP Profile Parameter") ◆ Only one of profiles is active, and the active profile will be marked with "(ON)"
(2)	<b>Profile 2</b>	
(3)	<b>Profile 3</b>	
(4)	<b>Profile 4</b>	

SIP Profile Parameters



## Content of SIP Profile

Item Name	Configurable	Data Format	Options List	Max. Length
ID	Yes	Alphanumeric		15
PhoneName (Name)	Yes	Alphanumeric		20
PxySrv	Yes	xxx.xxx.xxx.xxx DomainName		15 127
RegSrv	Yes	xxx.xxx.xxx.xxx DomainName		15 127
RegID	Yes	Alphanumeric		63
RegPwd	Yes	Alphanumeric		63
DomainName (Dname)	Yes	Alphanumeric		127
TelNbr	Yes	Numeric		24
BackupPxySrv (BPSrv)	Yes	xxx.xxx.xxx.xxx DomainName		15 127
BackupRegSrv (BRSrv)	Yes	xxx.xxx.xxx.xxx DomainName		15 127

# **Chapter 6 Configuration via Web Interface**

## **1 Configuration through the Web interface**

The phone can also be configured using the web browser. It has a friendly web interface to set and modify parameters.

### **Step 1 – Before starting**

The IP phone should be connected to the Network and then be powered it up before the connection to the web server.

### **Step 2 – Connect to the IP Phone web server**

- 1) Start a web browser session
- 2) Input the phone's IP address into the address field.
  - a. For example: <http://10.0.0.139/admin.html>
- 3) Enter the username (by default: administrator)
- 4) Enter the password (by default: 784518)
- 5) Once the process is completed, you can change the settings values to your convenience.

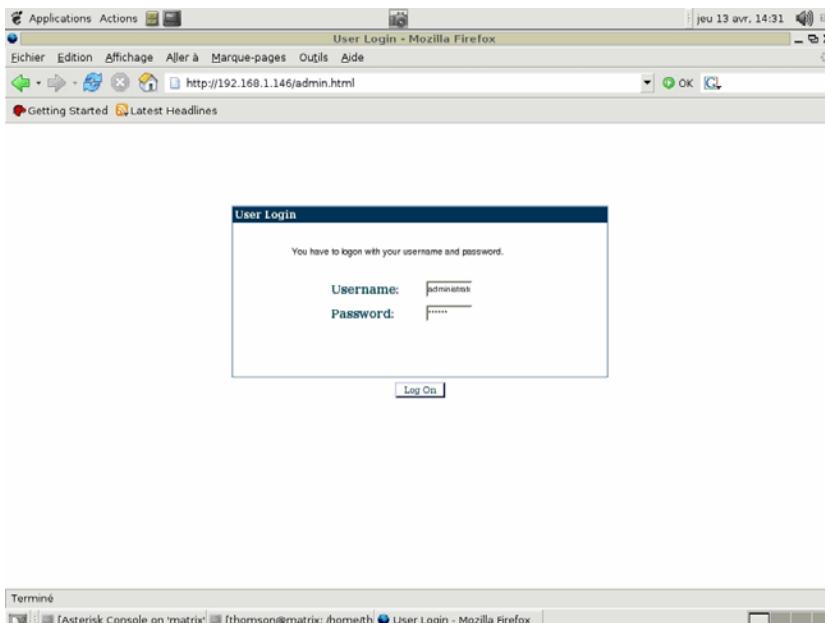
### **Step 3 – Reboot the phone**

Some parameters will require the phone to be rebooted for changes to be operated. The web pages below will be displayed once you have input the phone's IP address into the address field.

## **2 Detailed Web pages**

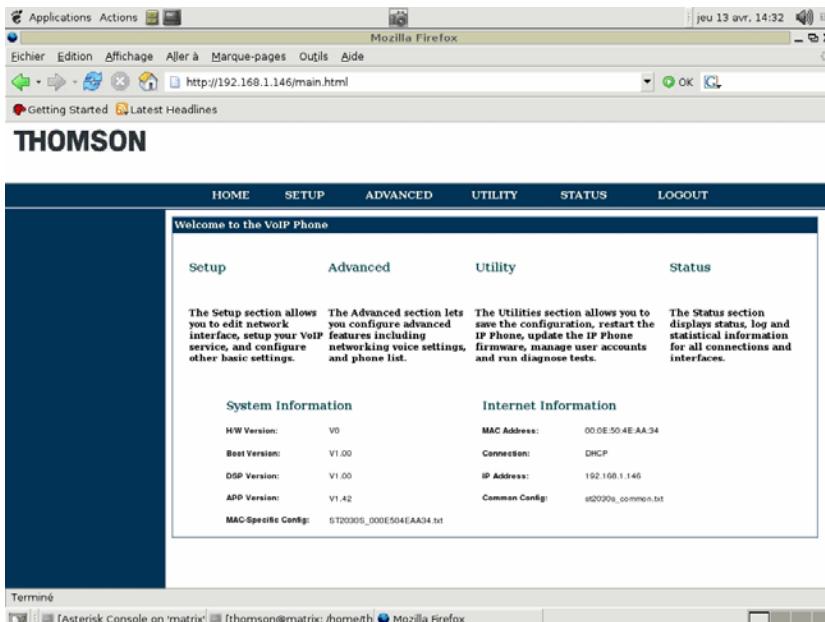
Please find below the web pages related to the configuration of phone settings via web interface

## CONNECTION PAGE

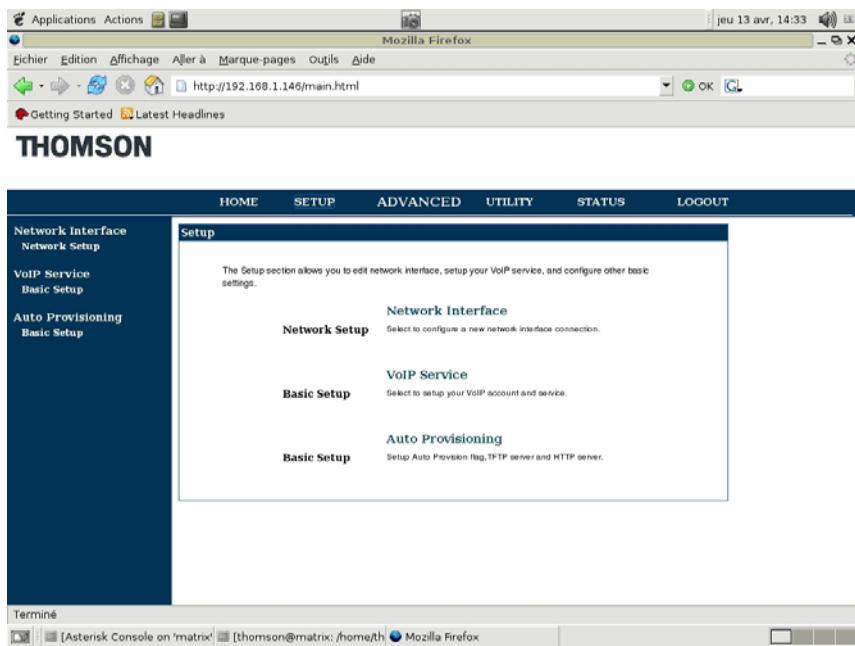


The username is: administrator  
The password is: 784518

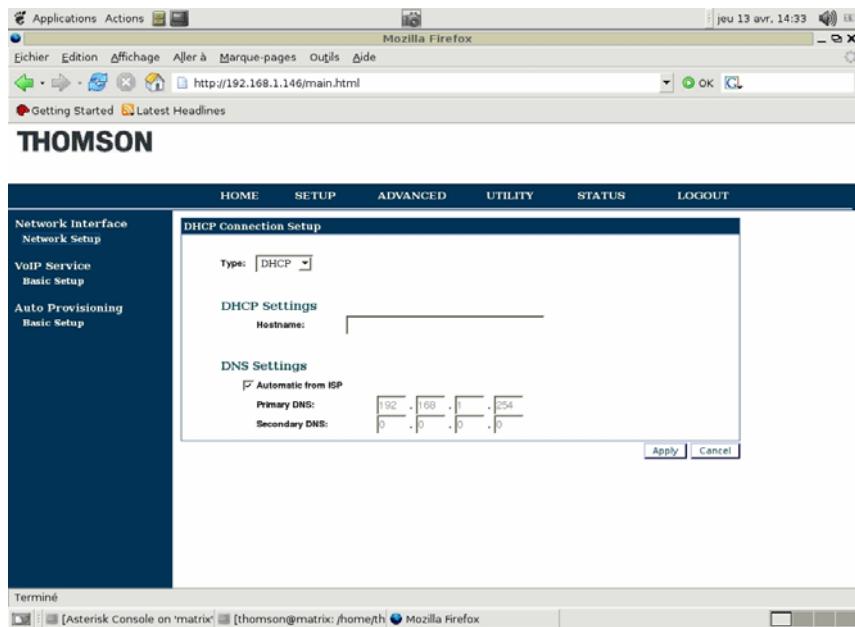
## HOME PAGE



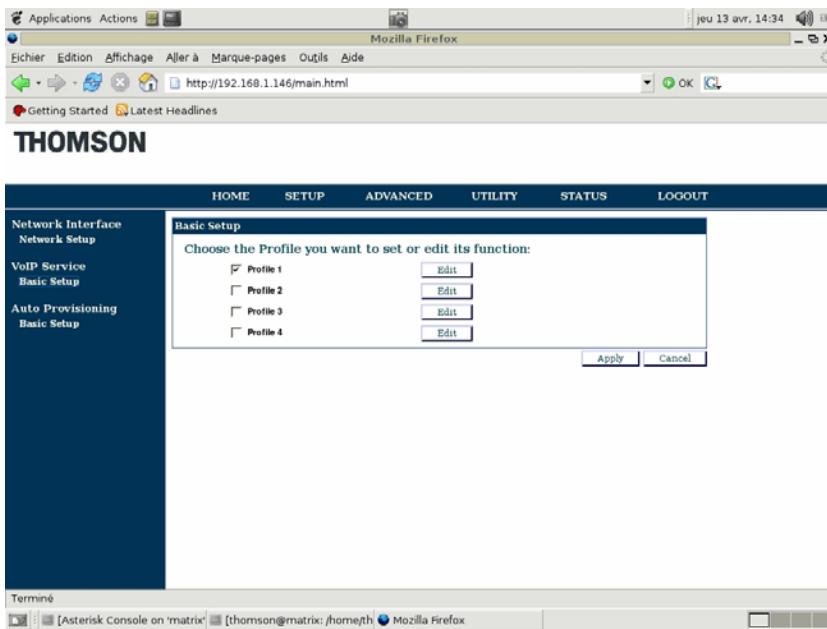
## **SETUP PAGE**



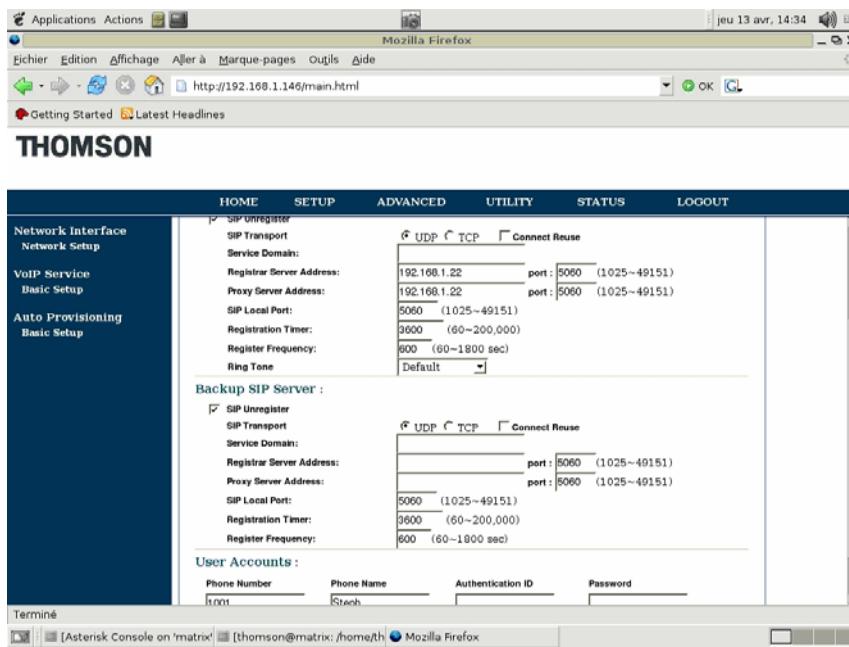
**SETUP: Network setup page (DHCP):** Page uses to DHCP connection setup.



## SETUP: Basic setup page: Page for edit a profile



## Profile edit



Here, you must indicate the address of the proxy, the login of the user like his password and the associated telephone number.

SIP local port is by default 5060

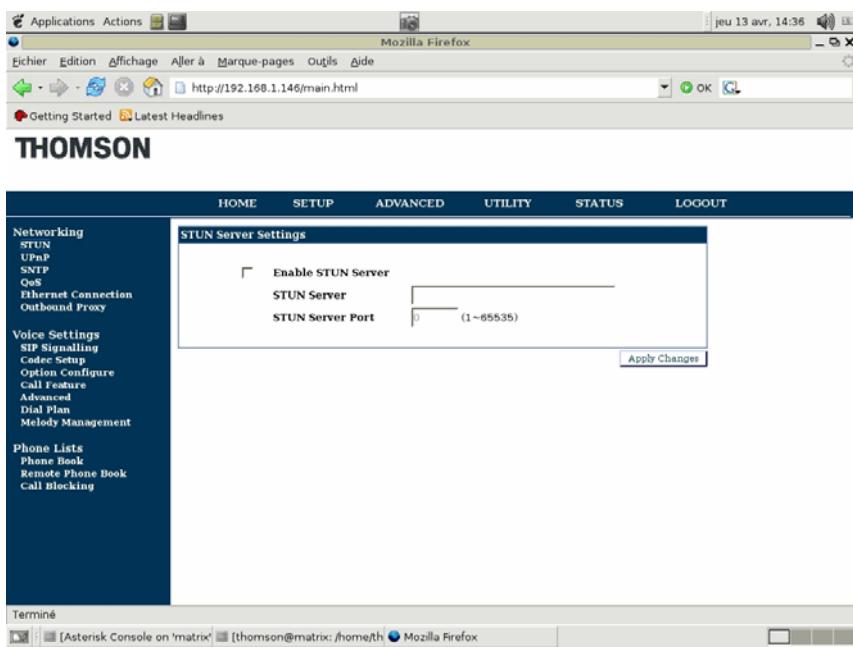
**APS basic setup:** It is the page to configure APS

The screenshot shows a Mozilla Firefox browser window with the address bar set to `http://192.168.1.146/main.html`. The main content area displays the 'Auto Provisioning System Configuration' page. On the left, a sidebar menu includes 'Network Interface', 'Network Setup', 'VoIP Service', 'Basic Setup', 'Auto Provisioning', and 'Basic Setup'. The right panel is titled 'Auto Provisioning Setting' and contains four configuration options: 'DHCP' (selected), 'HTTP preset', 'TFTP preset', and 'No Provisioning'. Each option has fields for 'Server IP Address' and 'File Name'. At the bottom of the panel are 'Apply' and 'Cancel' buttons.

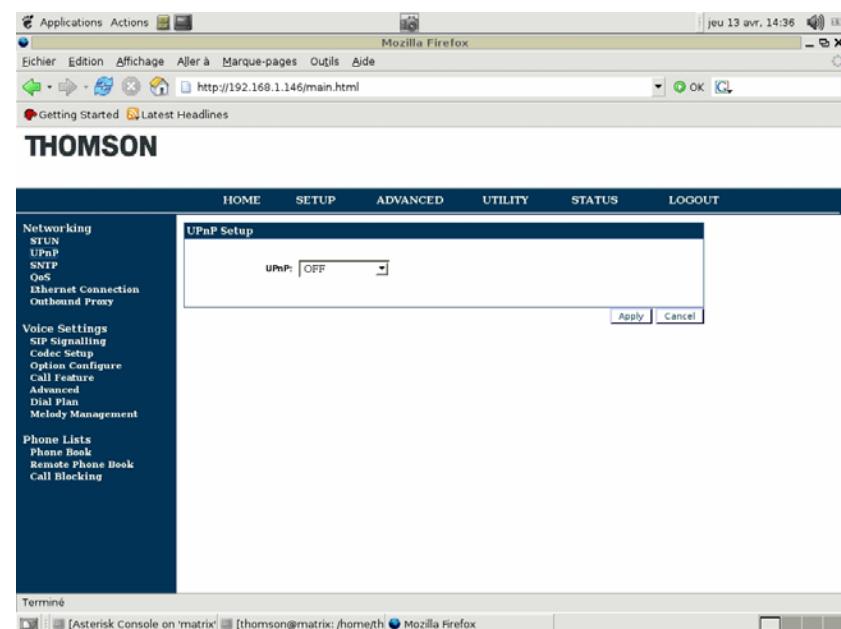
## ADVANCED PAGE

The screenshot shows a Mozilla Firefox browser window with the address bar set to `http://192.168.1.146/main.html`. The main content area displays the 'Advanced' configuration page. On the left, a sidebar menu includes 'Networking' (STUN, UPnP, SNTP, QoS, Ethernet Connection, Outbound Proxy), 'Voice Settings' (SIP Signalling, Codec Setup, Option Configure, Call Feature, Advanced, Dial Plan, Melody Management), and 'Phone Lists' (Phone Book, Remote Phone Book, Call Blocking). The right panel is titled 'Advanced' and contains sections for 'Networking' (Configuring STUN server settings, UPnP settings, SNTP to configure time server internet, QoS to configure quality of service settings, Ethernet connection to configure LAN's and PC's transmission, and Outbound Proxy to configure Outbound Proxy settings) and 'Voice Settings' (Configuring SIP signalling parameters, Codec Setup to select preferred Codec, Option Configure to configure call feature function to visible or invisible, Call Feature to select telephone operations of call feature, and Advanced to configure advanced telephone settings).

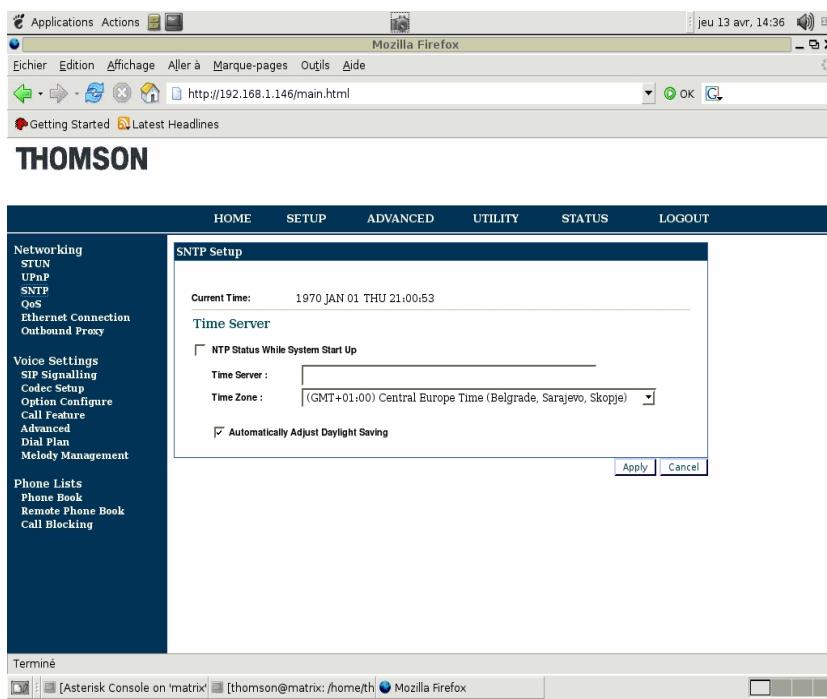
## ADVANCED: STUN page: STUN server settings



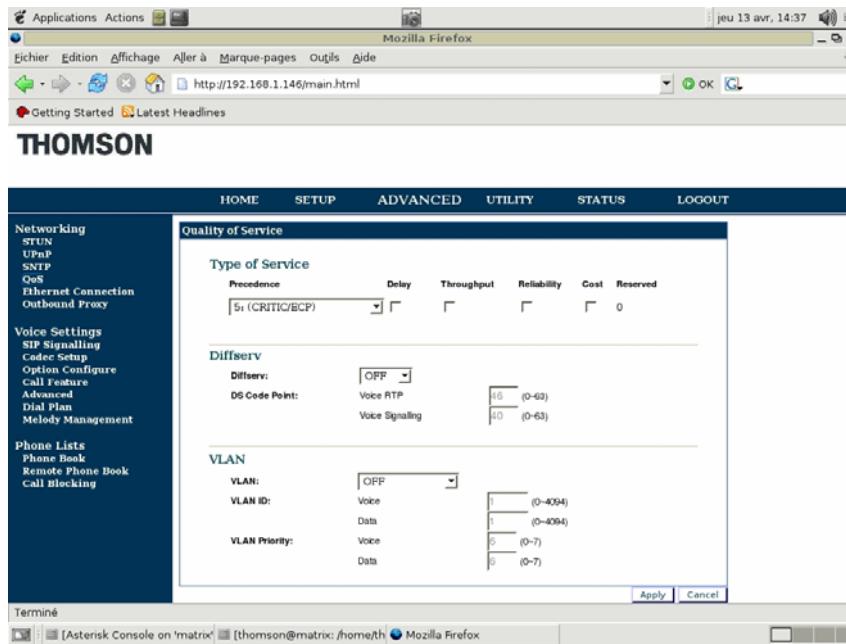
## ADVANCED: UPnP: UPnP setup



## ADVANCED: SNTP: SNTP setup to regulate the hour of the phone



## ADVANCED: QoS: Quality of service configuration



## ADVANCED: Ethernet connection

The screenshot shows the Thomson web interface for managing networking settings. The left sidebar contains links for Networking (STUN, UPnP, SNTP, QoS, Ethernet Connection, Outbound Proxy), Voice Settings (SIP Signalling, Codec Setup, Option Configure, Call Feature, Advanced, Dial Plan, Melody Management), and Phone Lists (Phone Book, Remote Phone Book, Call Blocking). The main content area is titled "Ethernet Connection" and displays a list of options: Auto - Negotiation, 100M Half - Duplex, 100M Full - Duplex, 10M Half - Duplex, and 10M Full - Duplex. At the bottom right of the content area are "Apply" and "Cancel" buttons.

## ADVANCED: Outbound proxy

The screenshot shows the Thomson web interface for managing networking settings. The left sidebar contains links for Networking (STUN, UPnP, SNTP, QoS, Ethernet Connection, Outbound Proxy), Voice Settings (SIP Signalling, Codec Setup, Option Configure, Call Feature, Advanced, Dial Plan, Melody Management), and Phone Lists (Phone Book, Remote Phone Book, Call Blocking). The main content area is titled "Outbound Proxy Settings" and displays configuration fields: Flag (set to Disable), Server (empty input field), and Port (set to 5060, with a note "(1~65535)". At the bottom right of the content area is an "Apply Changes" button.

## ADVANCED: SIP signalling

The screenshot shows the 'SIP Signalling' configuration page. The left sidebar includes options like Networking, Voice Settings, and Phone Lists. The main panel displays settings for SIP Signalling, including RTP Starting Port number (set to 41000), Session Timer (1800 sec), Minimum Session Timer (100 sec), and Session Refresh Method (INVITE). There are also checkboxes for Header Compact and PRACK Support. Buttons for 'Apply' and 'Cancel' are at the bottom right.

**ADVANCED: Codec setup:** you can choose the priority of the codecs.

The screenshot shows the 'Codec Setup' configuration page. The left sidebar includes options like Networking, Voice Settings, and Phone Lists. The main panel displays a table for Codec support priorities. It lists four codecs: G.711U, G.711A, G.723\_63, and G.729AB. For each codec, priority levels 1st through 4th are assigned. Under 'Priority', the first three levels are marked with a checked checkbox icon, while the 4th level is marked with a radio button icon. The table also includes columns for Packetization, Jitter Buffer Length, and Silence Suppression. Buttons for 'Apply' and 'Cancel' are at the bottom right.

## ADVANCED: Option configure

The screenshot shows a Mozilla Firefox browser window with the address bar set to `http://192.168.1.146/main.html`. The main content area displays the 'Option Configure' page under the 'ADVANCED' tab. On the left, a sidebar lists various configuration categories: Networking (STUN, UPnP, SNTF, QoS, Ethernet Connection, Outbound Proxy), Voice Settings (SIP Signalling, Codec Setup, Option Configure, Call Feature, Advanced, Dial Plan, Melody Management), and Phone Lists (Phone Book, Remote Phone Book, Call Blocking). The 'Option Configure' section contains a list of checkboxes for various call features:

- DialSubscribe
- CallBlocking
- CallForward
- CallWaiting
- AnonymBlock
- AutoAnswer
- AutoReject
- AutoStop
- AutoTurnOffSPK
- NumberDisplay
- DoNotDisturb
- PhoneLock
- Reboot

## ADVANCED: Call Features

The screenshot shows a Mozilla Firefox browser window with the address bar set to `http://192.168.1.146/main.html`. The main content area displays the 'Call Features' page under the 'ADVANCED' tab. On the left, a sidebar lists various configuration categories: Networking (STUN, UPnP, SNTF, QoS, Ethernet Connection, Outbound Proxy), Voice Settings (SIP Signalling, Codec Setup, Option Configure, Call Feature, Advanced, Dial Plan, Melody Management), and Phone Lists (Phone Book, Remote Phone Book, Call Blocking). The 'Call Features' section contains a form for 'Phone Operation' with the following settings:

- Privacy Call
- Call Waiting
- Anonymous Reject
- Hide Domain Name
- Call Log Prefix: [ ]
- Call Park
- Do Not Disturb
  - Permanent
  - Relative DD, HH : MM : 00 : 00 : 00
  - Absolute HH : MM : 00 : 00 : 00
- DND Response
  - 486
  - 486
  - 603
- Call Forward
  - >Forwarding Number [ ]
  - >Speed Dial Table [ ]
- Speed Dialling
  - >Function Key Table [ ]

**ADVANCED: Advanced:** Page for the telephone settings. You can configure your voicemail and the number of line

The screenshot shows the 'Advanced' tab selected in the top navigation bar. The left sidebar contains a tree view of configuration categories. The main panel is titled 'Telephone Settings' and includes the following configuration:

- DTMF: Out of Band (RFC2833)
- RTP Payload Type: 07 (97-127)
- Checkboxes:
  - Voice Activity Detection (VAD)
  - Acoustic Echo Cancellation (AEC)
  - Packet loss compensation
  - 'e' will be processed as normal digits
- Multiline: 10
- Subscribe to MWI:
  - Voice Mail Server Address: [ ]
  - Voice Mail Server Port: 5060
  - Telephone Number: [ ]
- On Hold:
  - Local music on Hold
  - Server music on Hold
- Dial-out timeout: Lease: 4 seconds
- Automatic answer: Lease: 0 seconds

**ADVANCED: Dial plan**

The screenshot shows the 'Dial Plan' tab selected in the top navigation bar. The left sidebar contains a tree view of configuration categories. The main panel is titled 'Dial Plan' and includes the following configuration:

- VoIP Dial Plan: [ ]
- Emergency Dial Plan: 911|11x|1[2-8]

Below the fields are 'Apply', 'Cancel', and 'Help' buttons.

**ADVANCED: Melody management:** You can update new melodies. To make a new melody you need to refer at chapter 8

The screenshot shows a Mozilla Firefox browser window with the URL <http://192.168.1.146/main.html>. The page title is "THOMSON". The navigation menu includes HOME, SETUP, ADVANCED, UTILITY, STATUS, and LOGOUT. The left sidebar lists various configuration options under Networking, Voice Settings, and Phone Lists. The main content area is titled "Melody Management" and contains two forms: "Add Ringer" and "Remove Ringer". The "Add Ringer" form has fields for "HTTP:" (with a browse button) and "TFTP:" (IP and File name fields). The "Remove Ringer" form has tabs for "Index", "Ringer", and "Delete". A message "Terminé" is displayed at the bottom.

**ADVANCED: Phone book:** To add new contacts in the local phonebook.

The screenshot shows a Mozilla Firefox browser window with the URL <http://192.168.1.146/main.html>. The page title is "THOMSON". The navigation menu includes HOME, SETUP, ADVANCED, UTILITY, STATUS, and LOGOUT. The left sidebar lists various configuration options under Networking, Voice Settings, and Phone Lists. The main content area is titled "Phone Book" and displays a table with one entry: Index 1, Name TOTO, Phone Number 0141065190. There are "Add", "OK", "Modify", and "Delete" buttons. A message "Terminé" is displayed at the bottom.

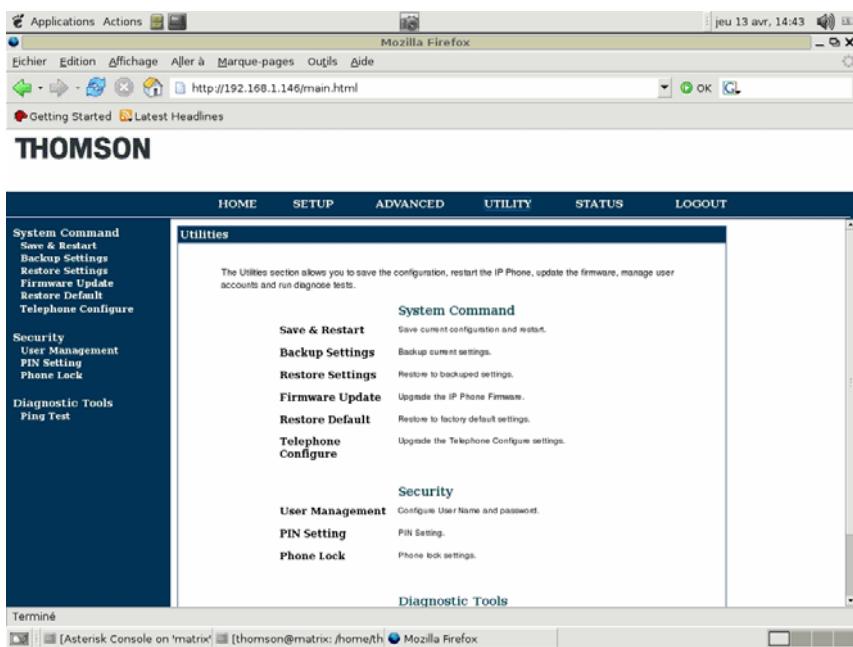
**ADVANCED: Remote phone book:** To add new contacts in remote XML phonebook.

The screenshot shows a Mozilla Firefox browser window with the URL <http://192.168.1.146/main.html>. The main content area displays the 'Remote Phone Book' configuration page. On the left, a sidebar menu includes options like Networking, Voice Settings, and Phone Lists. The central panel has a table titled 'Phone Book URL' with columns for Index, Phone Book URL, and Phone Book Name. There are five rows in the table, each with input fields for these three columns. At the bottom right of the panel are 'Apply' and 'Cancel' buttons. The status bar at the bottom of the browser window shows 'Terminé'.

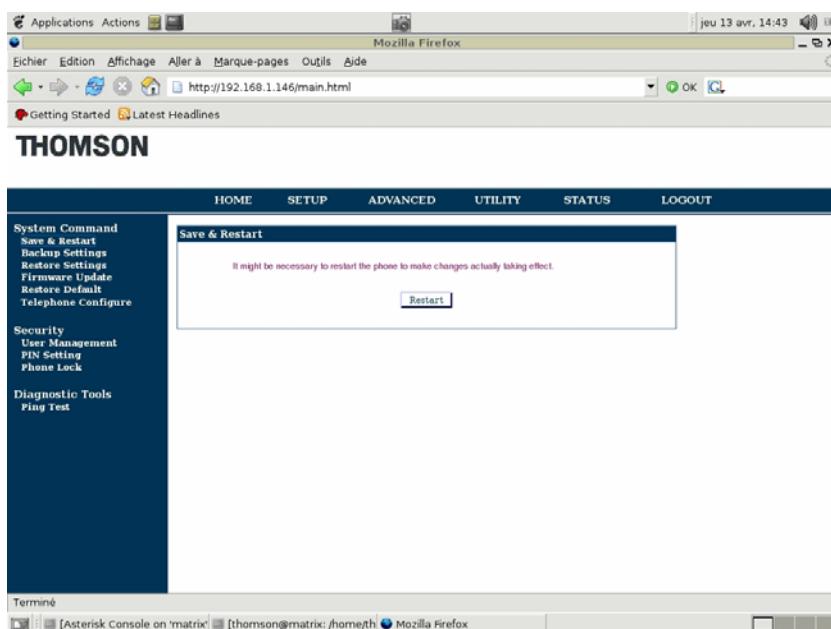
**ADVANCED: Call blocking:** to authorize or reject calls

The screenshot shows a Mozilla Firefox browser window with the URL <http://192.168.1.146/main.html>. The main content area displays the 'Call Blocking' configuration page. On the left, a sidebar menu includes options like Networking, Voice Settings, and Phone Lists. The central panel has a section titled 'Call Blocking' with a dropdown menu labeled 'Type' set to 'Disable'. Below this are two buttons: '>Edit Allow List' and '>Edit Reject List'. At the bottom right of the panel are 'Apply' and 'Cancel' buttons. The status bar at the bottom of the browser window shows 'Terminé'.

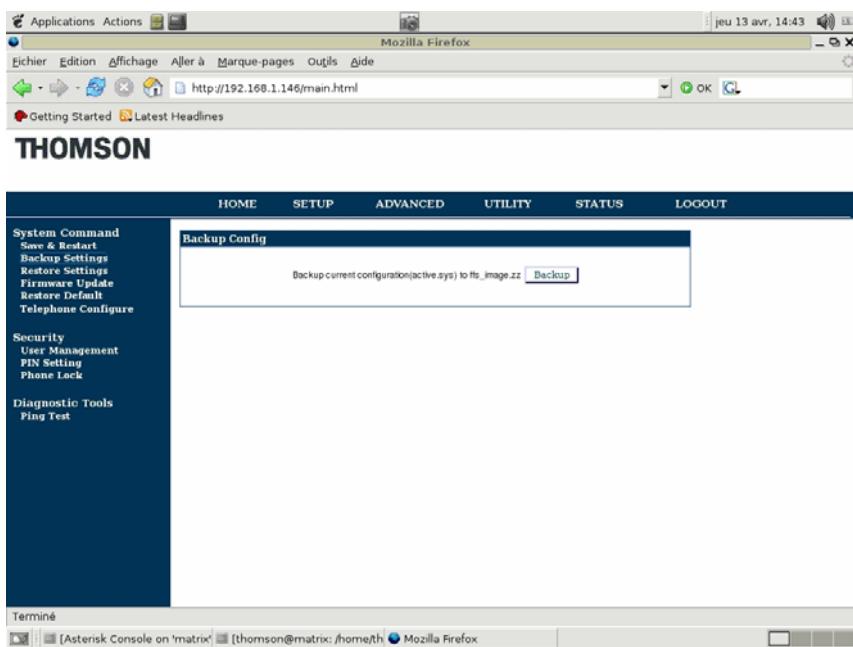
## UTILITY PAGE



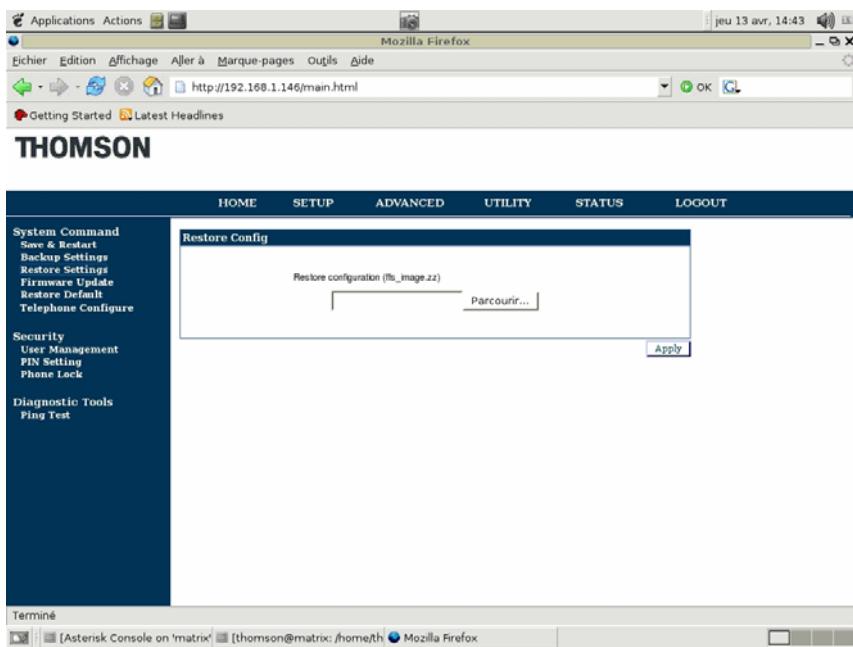
## UTILITY: Save & restart



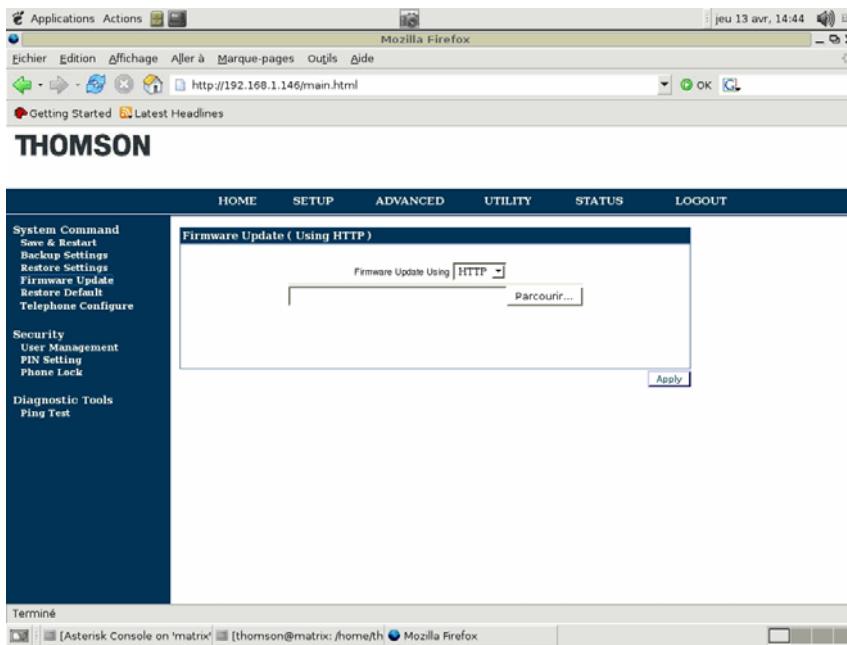
## UTILITY: Backup setting: backup current configuration (active.sys) to ffs\_image.zz



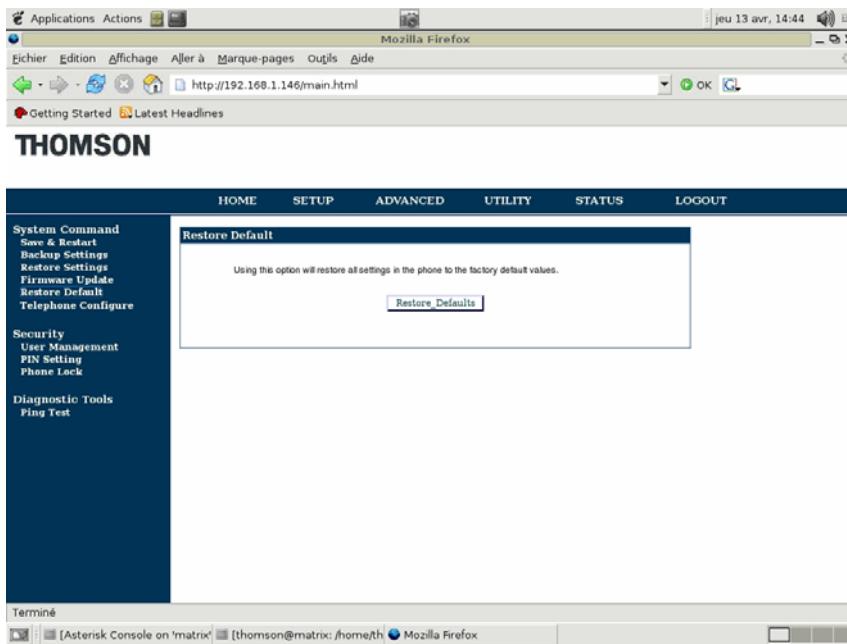
## UTILITY: Restore setting: Restore configuration (ffs\_image.zz)



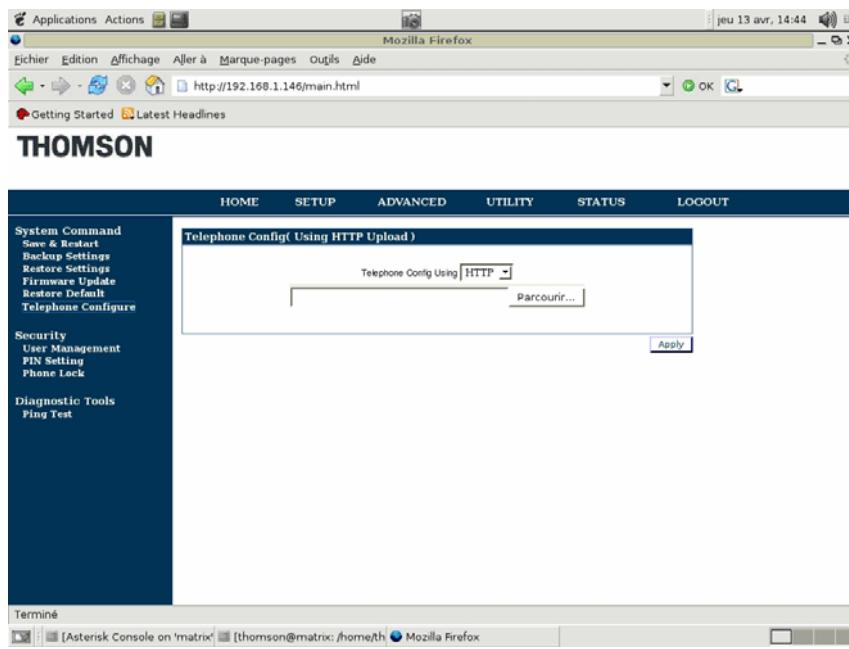
**UTILITY: Firmware Update:** It is here you can update your firmware. When you upload it, next you need to upload the Telconf file.



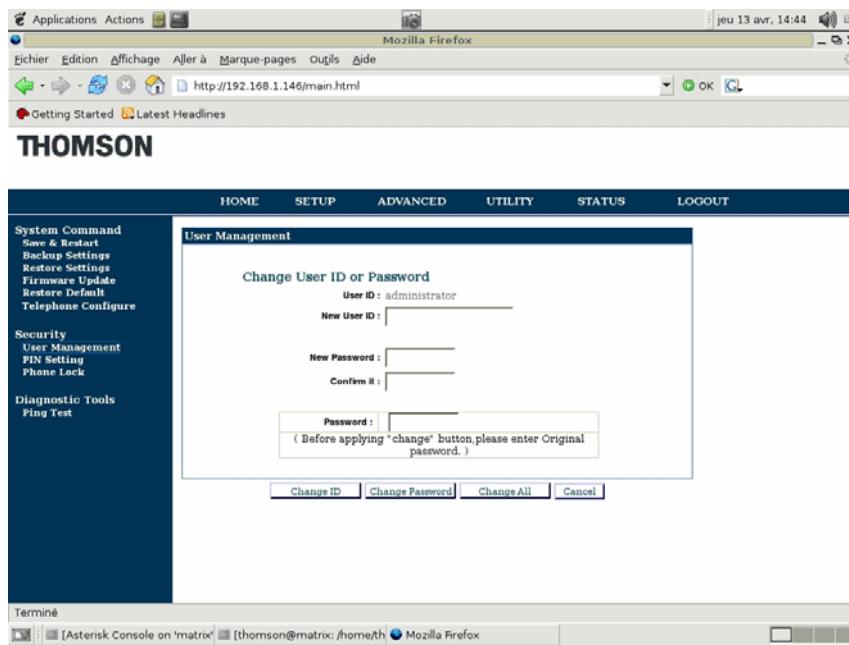
### UTILITY: Restore default



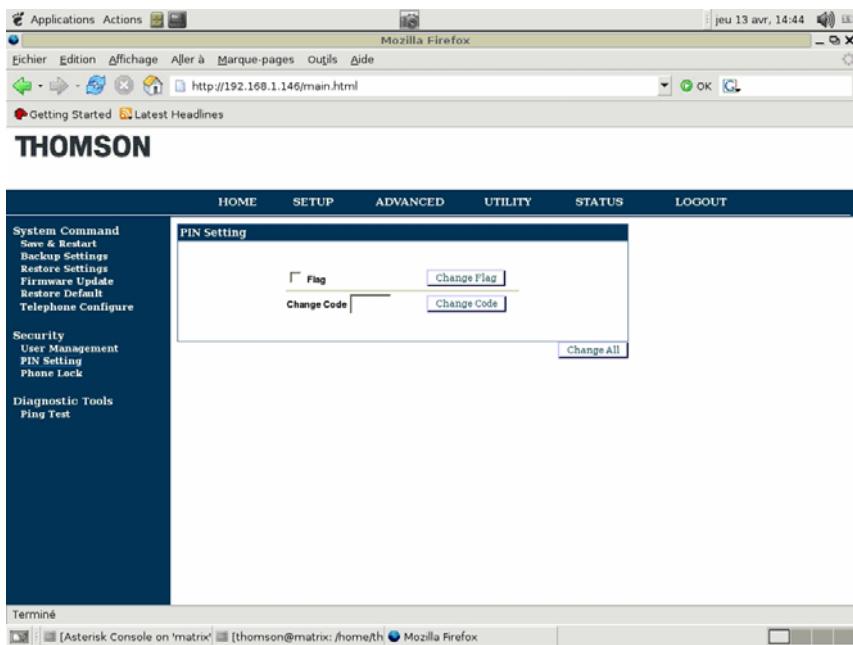
**UTILITY: Telephone configure:** It is here you upload the Telconf file.



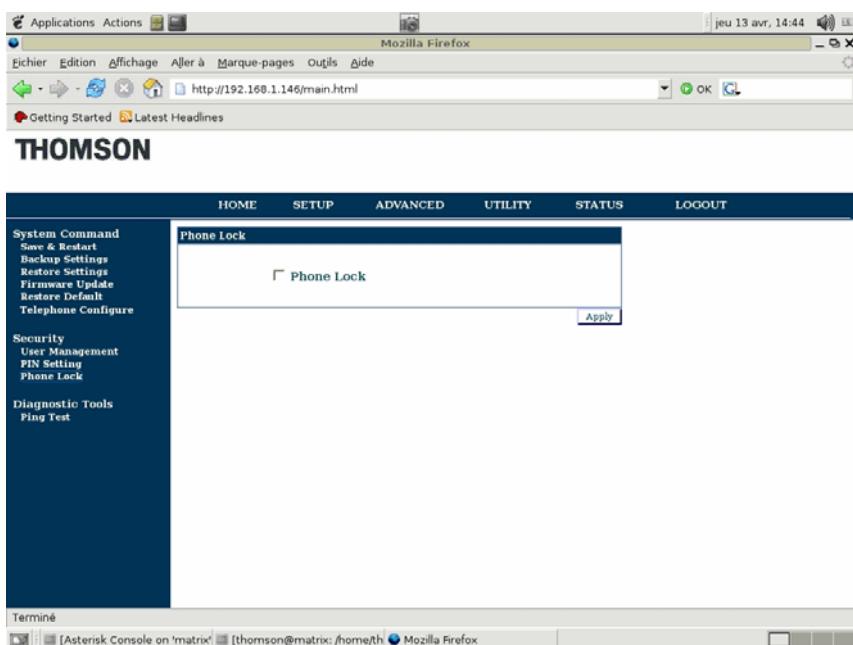
**UTILITY: User management:** You can change the parameters of the connection page.



## UTILITY: PIN setting



## UTILITY: Phone lock



## UTILITY: Ping test

The screenshot shows a Mozilla Firefox browser window with the address bar set to <http://192.168.1.146/main.html>. The main content area displays the 'Ping Test' configuration page. On the left, a sidebar lists various utility options like System Command, Security, and Diagnostic Tools. The central panel has a form titled 'Ping Test' with a 'Ping Destination IP' field containing a placeholder IP address. Below it is a table with columns 'Index' and 'IP Address', showing three entries: 'Packet1', 'Packet2', and 'Packet3'. At the bottom right of the panel are 'Apply' and 'Cancel' buttons.

## STATUS PAGE

The screenshot shows a Mozilla Firefox browser window with the address bar set to <http://192.168.1.146/main.html>. The main content area displays the 'Status' page. The left sidebar includes links for General Info, Network Status, VoIP Status, and Phone Status. The central panel contains sections for 'General Info' (with a link to 'Product Info'), 'Network Status' (with a link to 'Interface Status'), 'VoIP Status' (with a link to 'Phone Status'), and a descriptive text about viewing VoIP status. The bottom of the page includes a 'Terminé' message and a taskbar with icons for Asterisk Console and Mozilla Firefox.

## STATUS : Product info

The screenshot shows a Mozilla Firefox browser window with the address bar set to <http://192.168.1.146/main.html>. The main content area displays the 'Product Info' section of the THOMSON interface. On the left, a sidebar lists 'General Info', 'Product Info', 'Network Status', 'Interface Status', 'VoIP Status', and 'Phone Status'. The right panel shows 'Hardware Information' with 'HW version V0' and 'Software Information' with 'Boot Code version V1.00', 'DSP version V1.00 (4 way.)', and 'App version V1.42'.

## STATUS : Interface Status

The screenshot shows a Mozilla Firefox browser window with the address bar set to <http://192.168.1.146/main.html>. The main content area displays the 'Interface Status' section of the THOMSON interface. On the left, a sidebar lists 'General Info', 'Product Info', 'Network Status', 'Interface Status', 'VoIP Status', and 'Phone Status'. The right panel shows 'Network Interface' settings: Mode (DHCP), IP address (192.168.1.146), Net mask (255.255.255.0), and Gateway (192.168.1.254).

## STATUS : Phone Status

The screenshot shows a Mozilla Firefox browser window with the address bar set to <http://192.168.1.146/main.html>. The main content area displays the 'Phone Status' section of the THOMSON device's configuration interface. The left sidebar includes links for General Info, Product Info, Network Status, Interface Status, VoIP Status, and Phone Status. The right panel shows 'Service status' with registration and proxy server addresses both listed as 192.168.1.22, and a service domain listed as 'i'. Below this is a 'Phone status' section with four profiles: Profile 1 (Phone number 1001, Caller ID Name Steph), Profile 2, Profile 3, and Profile 4.

Profile	Phone number	Caller ID Name
Profile 1	1001	Steph
Profile 2		
Profile 3		
Profile 4		

# Chapter 7 Auto-Provisioning

The **automatic provisioning** requires one DHCP server and one TFTP server.

Please find below the way to provision your IP Phone.

## 1 Automatic phone provisioning with TFTP

The automatic phone provisioning is particularly adapted to the provision of several phones at the same time, of any large-scale deployment environment. Auto-provisioning is achieved using 2 well-known protocols: Dynamic Host Control Protocol (DHCP) and Trivial File Transfer Protocol (TFTP).

Simply follow next steps to provision your IP Phone:

### Needed:

**1 DHCP server which one can manage the options**

**1 TFTP server**

**1 ST2030 (SIP or MGCP)**

**Several files which one can need (correctly filled) are:**

SIP: File INF

Firmware

File « common\_config »

File MAC specific

File « Telconf »

MGCP: File INF

Firmware

Files Deck

File« common\_config »

File MAC specific

File « TelConf »

(See examples of these files in Appendix.)

## **The files:**

File INF: (extension .txt or .inf) inform about the place of the various files to download (relative way or absolute)

Firmware: Version's code which needs to be upgrade by the phone

File “common config”: (extension.txt) informs the phone about configurations which need to be upgrade on the level network, sip (or mgcp), auto provisioning, etc...

File MAC: (extension.txt). Same that the file “common\_config” but its name is based on phone's MAC address which needs to be upgrade and the configuration which is made inside will concern only the phone which has this MAC address. It has priority if the two files are on TFTP server.

File “TelConf”: (extension.txt), a continuation of Telnet command, which configure the phone on many parameters.

Files Deck: (ONLY MGCP) (extension .thd), allow phone to know how to display some flows (xml) on screen. Generally, it's files which are sent via the operator chosen to connection to the service.

We provide often to the customer only the upgrade firmware. The other files are normally managed by the administrator who takes care of the procedure of auto provisioning.

## **DHCP option:**

Option 66: IP address of TFTP server where are the files which need to be downloaded.

Option 67: File INF's name on TFTP server

Option 150: IP address of TFTP server used if this option (optional) is in DHCP options

Option 43: Used to http upgrade. If options 66 or 150 are not present, option 43 will be selected.

Option Next server Address: IP address of TFTP server used in some case when the other options are not manageable but this one yes.

Put in the following way in the file of configuration of DHCP (dhcpcd.conf)

```

option tftp-server-name "@IP serveur TFTP";           -----> option 66
option bootfile-name "nomDeFichierINF.txt";          -----> option 67
option provision-server-ip code 150 = text;           -----> option 150 is text, not binary mode
option provision-server-ip "@IP serveur TFTP ";       -----> next server address
option vendor-encapsulated-options "http://192.168.70.10/swupgrade_st2030m.txt" --->option 43

```

## 2 Installation of the platform

### Installation of the DHCP server

To set up all the necessary options (option 66 and 67)

### Installation of the TFTP server

To put in the repertory of the TFTP server the last version of the firmware

### Provisioning process

#### **Process of auto provisioning in the theory:**

At the time of the connection of a phone to the network, it will ask an IP address to DHCP server. This one will provide him an IP address but also the address of TFTP server where file INF is. This file will indicate on the phone the configuration file names which it must upgrade.

#### **Process of auto provisioning in the practice:**

1. Do not connect the phone with electricity. To put the cable (RJ45) in the Ethernet LAN port at the back of the phone.
2. Restore the default parameters. With this intention, it is necessary to maintain inserted the keys "headset" (ear-phone) and "transfers" (dumb) of the phone.
3. Connect phone with electricity.  
Maintain the keys "headset" (ear-phone) and "transfers" (dumb) of the phone inserted, until seeing that the provisioning is carried out.
4. The phone will start (it will recover an IP address by DHCP server, the IP address of TFTP server and the name of file INF which will be downloaded) and will begin the process of auto provisioning.

1. In the order:
  - Remote loading of file INF
  - Remote loading of the firmware  
(Remote loading of the files Deck for MGCP)
  - Reboot
  - 2<sup>nd</sup> Remote loading of file INF  
(2<sup>nd</sup> Remote loading of the Deck files for MGCP)
  - Remote loading of the file TelConf
  - Remote loading of the file Common\_config
  - Attempt of remote loading of the specific file MAC if he is
  - Reboot
  - Remote loading of file INF  
(2<sup>nd</sup> Remote loading of the files Deck for MGCP)

2. The phone is connected to the service of the operator.

#### Some precise details:

If phone doesn't receive an IP address, you need to check up that the parameter Mode→DHCP is activated

The file "Common\_config" will be downloaded only if the file's name has been change compared to the last upgrade.

The specific file MAC is downloaded (if there exists) if his parameter "config\_sg" was incremented.

It is imperatively necessary to download the file TelConf before returning it to the customer.

It is possible to pass from a phone SIP and quite simply to transform it into MGCP (but not the reverse) by providing on phone SIP a firmware corresponding to MGCP protocol as well as the good files which go with.

## 3 Automatic phone provisioning with HTTP

Also the provisioning can be carried out via HTTP. For that, it is necessary to follow following steps

#### Needed :

**1 router DHCP**  
**1 ST2030 (SIP or MGCP)**  
**The last version of the firmware**

### **Process of auto provisioning in the theory:**

At the time of the connection of a phone to the network, it will ask an IP address to DHCP server. This one will provide him an IP address. While going on the WEB interface of the phone in mode *admin*, you can upgrade the firmware.

### **Process of auto provisioning in the practice:**

1. Do not connect the phone with electricity. Connect an end of the cable (RJ45) in the Ethernet LAN port at the back of the phone and the other end is plugged into the router DHCP.
2. Restore the default parameters. With this intention, it is necessary to maintain inserted the keys "headset" (ear-phone) and "transfers" (dumb) of the phone
3. Connect phone with electricity.  
Maintain the keys "headset" (ear-phone) and "transfers" (dumb) of the phone inserted, until seeing that the provisioning is carried out.
4. The phone not having for the moment anything with provision and no gateway to connect itself, it will continue to blink.
5. Find the phone IP address (provided by router DHCP). So :

#### **If MGCP phone :**

- Press on the key *menu* of the phone
- Press on the softkey *admin*
- Press on the button *OK* of the phone
- Select *Network configuration* while going down with the arrows
- Select *IP address* and note the address indicated

#### **If SIP phone:**

- Press on the key *menu* of the phone
  - descend the cursor using the low button until *Information*
  - Select *Information* and note the IP address indicated
6. Open a WEB browser (Internet Explorer, Mozilla,...)
  7. Enter the bar of address : [IPaddressofphone/admin.html](http://IPaddressofphone/admin.html)

The IP address of the phone being noted in (5). The Web interface appears.

8. Login is : administrator
- Password is: 784518

9. This page is displayed:

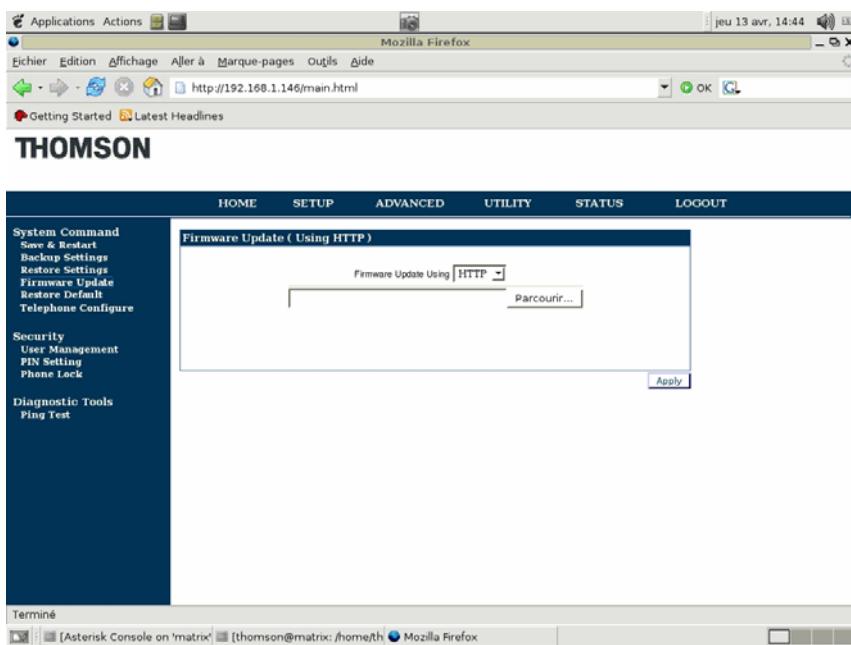
The screenshot shows the Mozilla Firefox browser window displaying the Thomson VoIP Phone web interface. The title bar reads "Mozilla Firefox" and the address bar shows "http://192.168.1.146/main.html". The main content area is titled "Welcome to the VoIP Phone". It features a navigation menu at the top with links for HOME, SETUP, ADVANCED, UTILITY, STATUS, and LOGOUT. Below the menu, there are four tabs: Setup, Advanced, Utility, and Status. The "Utility" tab is selected. The Utility section contains four columns of text and configuration details. The first column describes the Setup section. The second column describes the Advanced section. The third column describes the Utilities section. The fourth column describes the Status section. Below these sections, there are two tables: "System Information" and "Internet Information". The System Information table lists HW Version (V0), Boot Version (V1.00), DSP Version (V1.00), APP Version (V1.42), and MAC-Specific Config (st2030s\_000E504EA34.txt). The Internet Information table lists MAC Address (00:0E:50:4E:AA:34), Connection (DHCP), IP Address (192.168.1.146), and Common Config (st2030s\_common.txt).

10. Select « UTILITY ».

The screenshot shows the Mozilla Firefox browser window displaying the Thomson VoIP Phone web interface. The title bar reads "Mozilla Firefox" and the address bar shows "http://192.168.1.146/main.html". The main content area is titled "Utilities". On the left side, there is a sidebar with several categories: System Command (Save & Restart, Backup Settings, Restore Settings, Firmware Update, Restore Default, Telephone Configure), Security (User Management, PIN Setting, Phone Lock), Diagnostic Tools (Ping Test), and a general Utilities section. The Utilities section is currently active. It contains a "System Command" section with links for Save & Restart, Backup Settings, Restore Settings, Firmware Update, Restore Default, and Telephone Configure. It also contains a "Security" section with links for User Management, PIN Setting, and Phone Lock. At the bottom, there is a "Diagnostic Tools" section.



11. Select « Firmware Update »

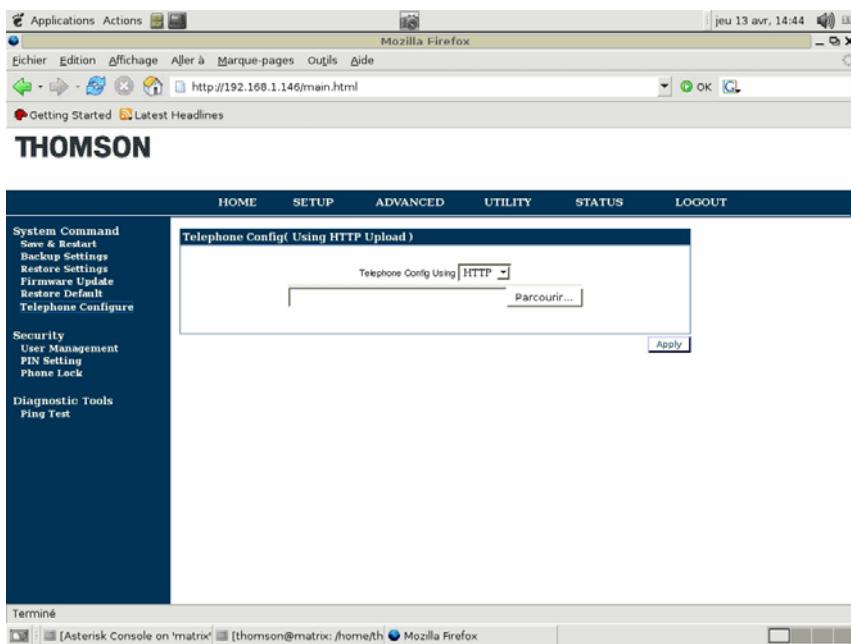


12. In the box Firmware Update Using leave "HTTP" and click on the button "Browse..."  
In the window which opens, select the version of the firmware to be installed.

13. Press « Apply »

14. The phone will normally reboot.

15. Startain from step 7) to step 11). Select « Telephone Configure »



16. In the box Firmware Update Using leave "HTTP" and click on the button "Browse..."  
In the window which opens, select the file TelConf to be installed.

17. Press « Apply »

18. The phone will normally reboot.

19. Start again the process (find IP address, open WEB interface as admin). On the banner page of the WEB interface, you should see the firmware's version of the phone. Look in « System Information », at the line « APP Version ».

**Caution: the main power supply must not be interrupted during the booting session. The phone could be damaged.**

**[Note]: when you want to upgrade the firmware or download a new configuration, you have to follow steps 1 and 2 and reboot the phone from its web-based interface.**

## Chapter 8 Ringtones configuration

## 1 Description

The ringtones which are integrated into the ST2030 use a RTTTL format. A ringing with RTTTL format looks like the following example:

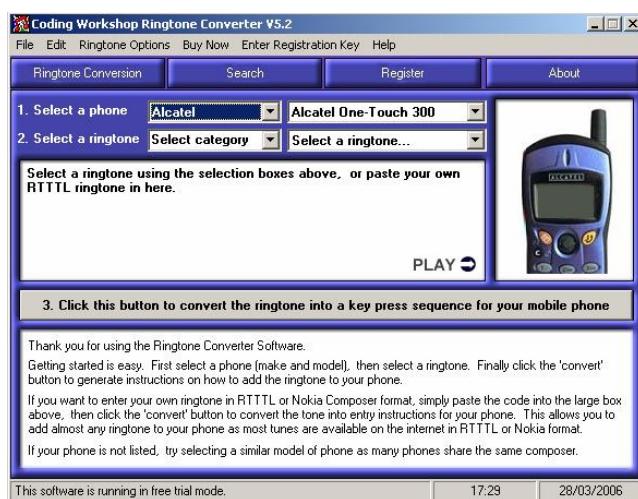
These ringtones can be created by the user and they can be also downloaded for free on Internet.

We give a series of specialized websites where you can find example of tones:

- [didier.elo.free.fr/ringtones.html](http://didier.elo.free.fr/ringtones.html)
  - [www.xgsmonline.com/smartsmslogos.shtml](http://www.xgsmonline.com/smartsmslogos.shtml)
  - [www.kortable.com/sonneries/sonneries-nokia-rtttl.asp](http://www.kortable.com/sonneries/sonneries-nokia-rtttl.asp)

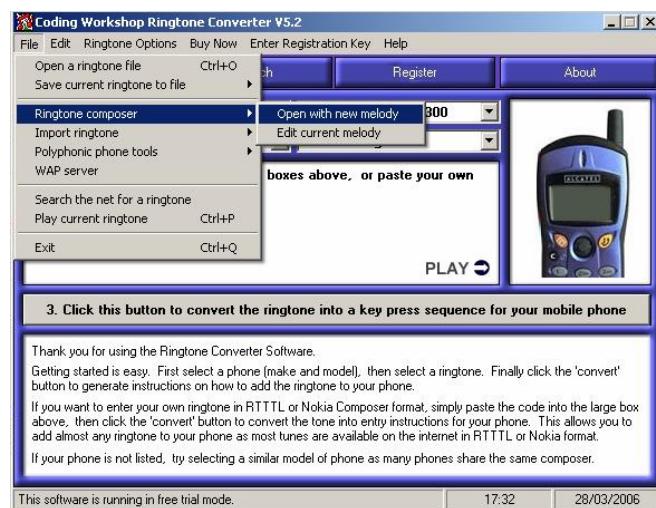
## 2 Make a melody

There is the software *Ringtone Converter* which is free and simple of use that we advise you.

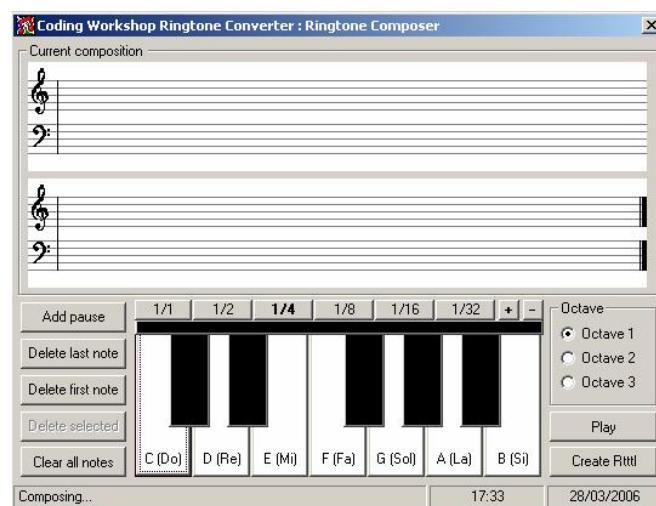


You can download this software here:  
[www.codingworkshop.com/ringtones/](http://www.codingworkshop.com/ringtones/)

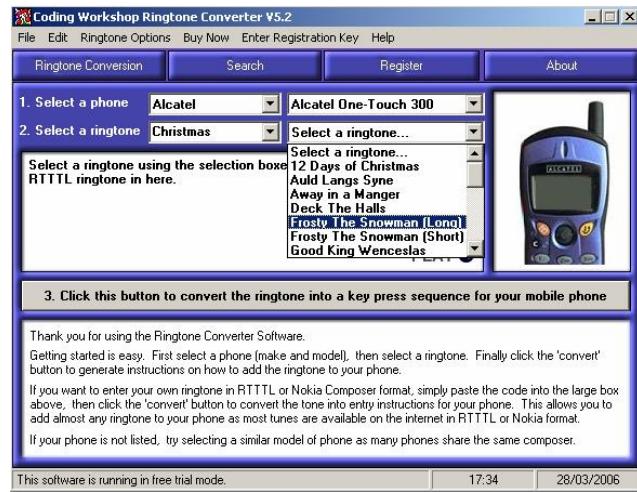
This software makes it possible to the user to create their own melodies directly with RTTTL format. For that, it is necessary to go in "File", "Ringtone composer" and finally "open with new melody"



The following screen is displayed and will make it possible to the user to compose the melodies according to its choice.

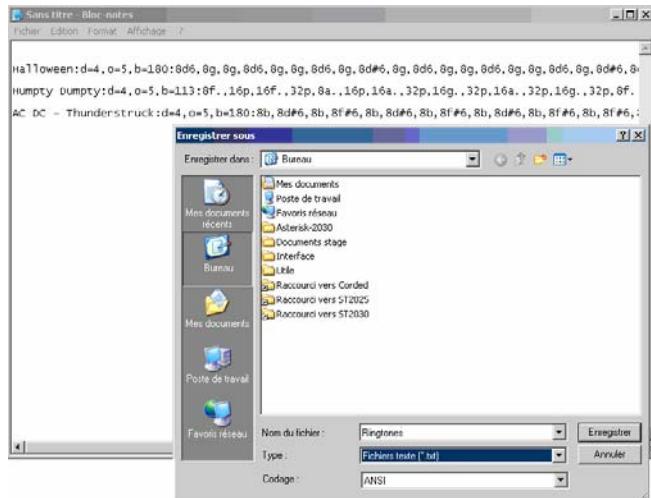


Moreover, a data base, already comprising a great number of melodies to RTTTL format, is integrated into this software. To reach these rings, it is necessary to carry out the handling which is presented to you on the screenshot below.



### 3 Integration of the rings with ST2030

In order to be able to integrate new ringtones, with RTTTL format, in his phone, the user must start by recopying all these rings with RTTTL format in a file with a txt extension. It is thus necessary to use software like *NotePad*.



Next we need to find the phone IP address. So:

If MGCP phone:

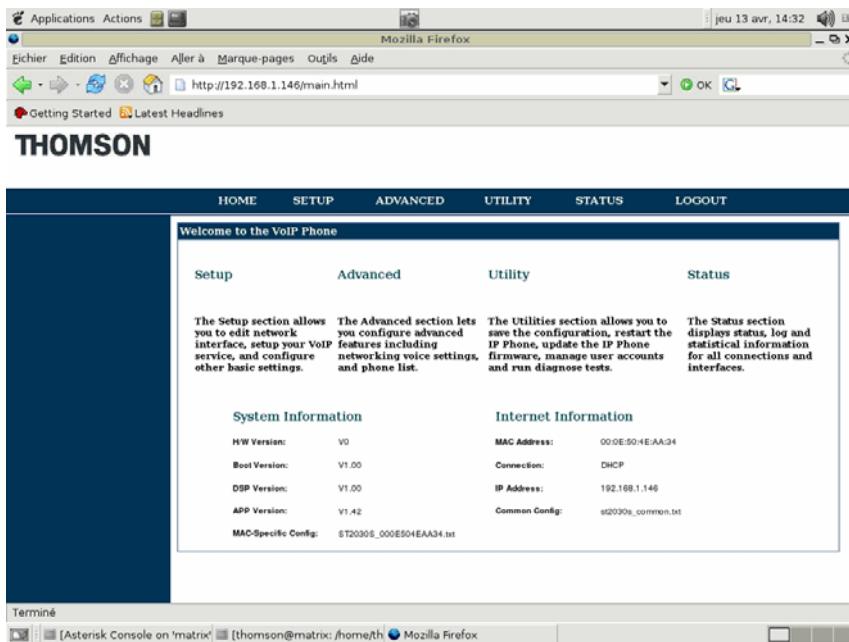
- Press on the key menu of the phone
- Press on the softkey admin
- Press on the button OK of the phone
- Select Network configuration while going down with the arrows
- Select IP address and note the address indicated

If SIP phone:

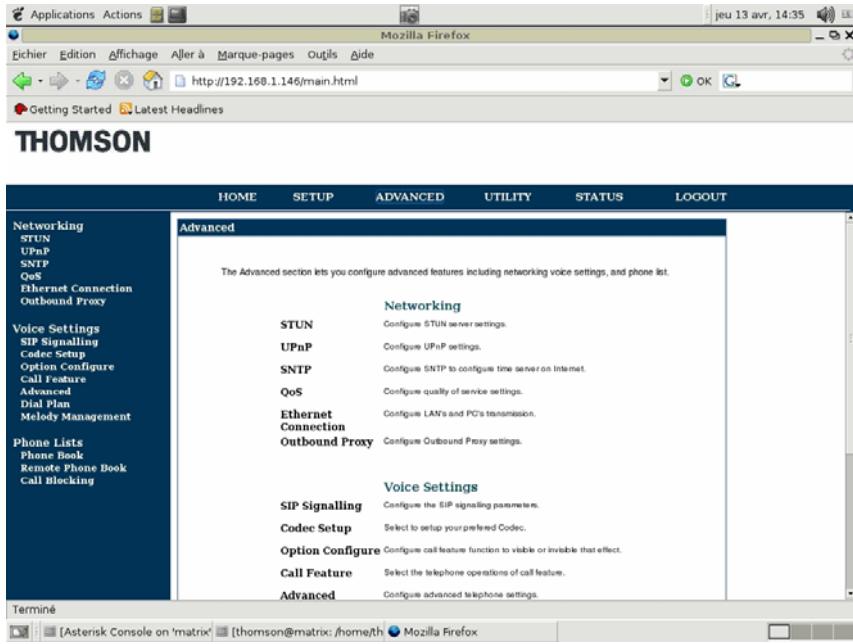
- Press on the key menu of the phone
- descend the cursor using the low button until Information
- Select Information and note the IP address indicated

IP address, that we recovered, must have returned in an Internet navigator (Mozilla, Internet explorer, etc.) in the following way: [adresseIPduTelephone/admin.html](http://adresseIPduTelephone/admin.html)

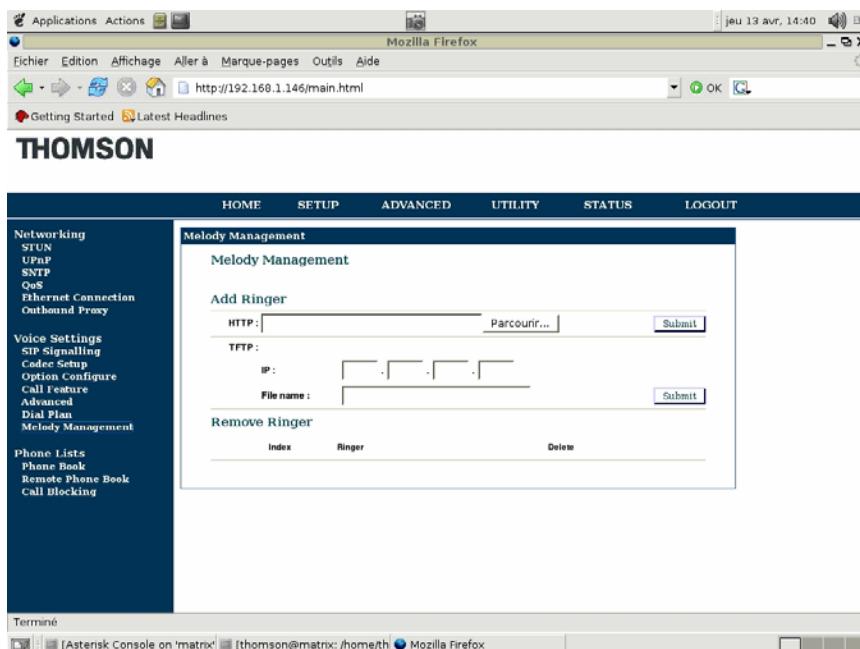
So this screen is displayed:



Next, we must click on « Advanced »:



Lastly, it is necessary to select "Melody management" and to enter the name of the file (txt extension) which contains the rings while clicking on browser.



**Note:** Soon the APS will be operational to update the rings with TFTP or HTTP.

## Chapter 9 Troubleshooting

If your phone is not operating as it should, please refer to the list of the problems in the following table. It should help you to solve the issue.

<b>Problem</b>	<b>Possible cause</b>	<b>Remedy</b>
<b>No dialing tone</b>	No connection to the LAN	Ensure that the Ethernet cable is correctly connected to the LAN port and the hub.
<b>No display</b>	No power.	Ensure that the power supply is correctly plugged in.
<b>"Waiting for CA..."</b> <b>Message and no dialing tone</b>	One of the following parameters is not set correctly: - Phone IP address - Call agent (CA) address - Subnet mask - Gateway address	Check these parameters with your system administrator.
<b>Your call partner cannot hear you OR You cannot hear him.</b>	The handset cable is not correctly plugged in.	Ensure that the handset cable is correctly plugged into the handset and the telephone
<b>Time is not the one you've programmed.</b>	Time is automatically specified by the network.	Ask your system administrator.
<b>"Phone needs to be rebooted" message.</b>	Following changes in the configuration of the phone, a reboot is needed to take effect.	Press the "OK" key to reboot the phone.
<b>Phone does not always ring.</b>	Call forwarding active or Do not disturb facility set.	Read the relevant section of the user guide. Deactivate the Call forward function by pressing <b>Fwd</b> and "enable"
<b>LED is blinking</b>	Incorrect Network connection	Unplug the Ethernet port and re-plug it.

**Table 8.1      Troubleshooting**

# Chapter 10 XML Phonebook

## 1 Description of the service

The aim of this service is to provide an easy access to the company's phonebook through the Thomson ST2030 MGCP Phone. The final user enters the name of the person he wishes to call on the telephone, using the DTMF keys. The telephone then consults an external server (using HTTP/XML), and displays the entries that matches the name entered. The user can then select one of them, and initiate a call. The user can consult the phonebook when the telephone is idle, but also when he is engaged in a conversation.

### Example:

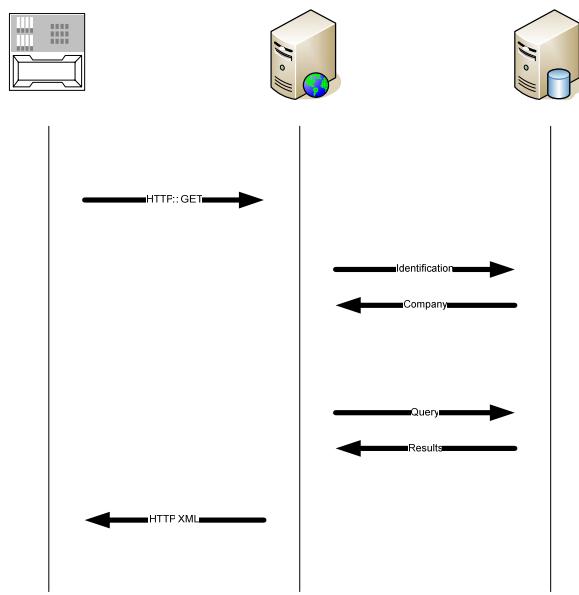
1. The User enters "Dup" as a search string
2. The phone displays 2 results "Dupond" and "Dupont"
3. The user selects Dupont
4. The phone displays the telephone number of Dupont
5. The user chooses to call Dupont
6. The phone places the call

The service described here is highly inspired by the Cisco IP Phone PhoneBook service.

## 2 Server specifications

### a) General structure

The phone sends the search string to the *HTTP* server by sending a *HTTP "Get"* request. The server identifies the phone, and then searches in a database all the matches to this query, in the companies in which the telephone is. The server then sends back a *XML* page to the telephone, which will contain the results of the query.



### b) Identification of the phone

The HTTP server will identify the phone by using one, two or three arguments sent by the phone in the *GET* request:

HTTP Name	Description
IP_ADDR	IP address of the phone
MAC_ADDR	MAC address of the phone
SERIAL_NB	Serial Number of the phone

The HTTP server will identify the searched string with the following parameter:

HTTP Name	Description
NAME	The string to search (most likely the first letters of the last name of the person we want to lookup the telephone number)

Thoses parameters will be sent by the telephone in query request. Here is an example of an HTTP request that would be sent by the phone in order to lookup in the phonebook all the names that start by "Dup".

[http://www.server.com/search.php?IP\\_ADDR=192.168.1.23&MAC\\_ADDR=000590024f3e&NAME=Dup](http://www.server.com/search.php?IP_ADDR=192.168.1.23&MAC_ADDR=000590024f3e&NAME=Dup)

No results to the query

If the server doesn't find any answers to the result of the query, it will send back to the phone this XML page:

[ThomsonPhoneBook XML page]

```
<ThomsonPhoneBook>
</ ThomsonPhoneBook>
```

### **Less than 32 results to the query**

If the server finds less than 32 answers to the query, it will send to the telephone this XML page:

[ThomsonPhoneBook XML page]

```
<ThomsonPhoneBook>
  <DirectoryEntry>
    <Name>Dupad André</Name>
    <Telephone>0175008348</Telephone>
  </DirectoryEntry>
  ....
  <DirectoryEntry>
    <Name>Dupont Antoine</Name>
    <Telephone>0175008338</Telephone>
  </DirectoryEntry>
</ ThomsonPhoneBook>
```

### **More than 32 results to the query**

If the server finds more than 32 answers, the phone won't be able to displays all the results at one. This is why the server won't send all the answers back to the telephone. He will instead send a list of URLs, where the phone will be able to fetch the results by block of 32 entries.

[ThomsonPhoneMenu XML page]

```
<ThomsonPhoneMenu>
  <MenuItem>
    <Name>Arrh → Foulard</Name>
    <URL>http://www.server.com/get32results\_1.php</URL>
  </MenuItem>
  ....
  <MenuItem>
    <Name>Roger → Ziad </Name>
    <URL>http://www.server.com/get32results\_7.php</URL>
  </MenuItem>
</ ThomsonPhoneMenu>
```

Each URL in the menu will send back a *ThomsonPhoneBook* XML page as described in the previous section.

## Chapter 11 Dial plan syntax

The syntax used to identify a *dial plan* in a digit map is adapted from [RFC 2705](#).

Dial Plan Syntax		
To specify a	Enter the following	Result
Digit	0 1 2 3 4 5 6 7 8 9 *	Identifies a specific digit (do not use #)
Range	[digit-digit]	Identifies any digit dialed that is included in the range
Range	[digit-digit, digit]	Specifies a range as a comma separated list
Wildcard	x	x matches any single digit that is dialed
Wildcard	.	. matches an arbitrary number of digits
Timer	T	Indicates that an additional time out period of 4 seconds should take place before automatic dialing starts

Some *dial plan* examples using the above syntax look as follows:

Dial Plan Syntax Examples		
For calls to	Users dial	Dial plan
Internal Extension	a two digit number	xx
Local Number	9 (if required for an outside line) and then a seven digit number	9xxxxxxxxT
Emergency	911	911
Local Operators	9 (if required for an outside line) then 0	90T
Long Distance	9 (if required), 1, area code and local number	91xxxxxxxxxx
International	9 (if required), 011, any number of additional digits	9011x.T

Different patterns in the dial plan are separated by "|", hence, a list of patterns would look like this:

|xx|9xxxxxxxxT|911|90T| etc

ST2030 SIP will use the dial plan to make your life easier by sending the digits to the sip proxy once a pattern is matched. But it does not forbid you to dial any number. I mean, if a SIP user presses OK the digits will be sent regardless the dial plan.

## Chapter 12 Appendix

### 1 Configuration and Commands

Table 9.1 lists a minimum set of configurable parameters and commands.

Parameters shall be configurable either via the auto-provisioning process, via the web-page, or via LCD menu. The Administrator will have access to all parameters, either via the web-page or LCD menu. The User will have access to a limited subset, via the web-page or LCD menu.

Configurable Parameter Or Command	Default value	Config file*	Web page		LCD menu		Comments
Display			Admin	User	Admin	User	
Adjustable contrast	3	x			x	x	In 1-5 range
Adjustable handset-speaker/headset/handsfree volume	3	x			x	x	In 1-5 range
Adjustable ringer volume	3	x			x	x	In 1-5 range
DTMF							
In Band	Deactivated	x	x				
Out of Band (RFC 2833)	Activated	x	x				
Ring tone number	1	x	x	x	x	x	In 1-4 range
Audio / video quality							
Voice Quality							
Voice Activity Detection (VAD)	Deactivated	x	x				
Comfort Noise Generation (CNG)	Deactivated	x	x				
Acoustic Echo Cancellation (AEC)	Activated	x	x				
Packet Loss Compensation	Activated	x	x				
Jitter buffer							
type	Adaptive	x	x				Can be Fixed or Adaptive
configurable jitter buffer length	40ms	x	x				0 to 200ms, used for both Fixed and Adaptive
Audio codecs							
g711a/μ	Choice #1; 10ms	x	x				10/20/30ms
g723.1	Choice #3; 30ms	x	x				30/60/90ms
g729ab	Choice #2; 10ms	x	x				10/20/30/40/50/60ms
Services							
User services							
Dial-out timeout	4 sec	x	x				0 sec means never dial out (only if user presses OK)
Menu language	English	x	x	x	x	x	
SNTP server	Blank	x	x		x		
Daylight saving time	Yes	x	x		x		
Selectable Time Zone	Central Europe	x	x		x		
NTP recycle timer	1 hour	x	x		x		
Maintenance							

Configuration							
Administrator Name	administrator	x	x		x		
Administrator Password	784518	x	x		x		6 Digits
User Name	user	x	x	x	x	x	
User Password	blank	x	x	x	x	x	4 Digits
Reset to factory defaults	N/A			x	x		Command
Save (upload) configuration on server	N/A			x	x		Command Saves all parameters
Protocol stacks							
Signalling							
port for sending signalling packets	2427	x	x				
port for receiving signalling packets	2427	x	x				
Addresses & names management							
Static IP	Deactivated	x	x		x		
DHCP client	Activated	x	x		x		
Subnet mask	blank	x	x		x		
Default IP gateway	blank	x	x		x		IP address or Domain Name
CA Address	blank	x	x		x		IP address or Domain Name
DNS server	blank	x	x		x		IP address or Domain Name
Backup DNS server	blank	x	x		x		IP address or Domain Name
RTP port	41000	x	x				
Same port for sending and receiving voice packets	Yes	x	x				
RTCP port	41001	x	x				
NAT							
UPnP IGD	Deactivated	x	x				
STUN server	blank	x	x				IP address or Domain Name
Remote information access							
TFTP server	blank	x	x				
FTP server	blank	x	x				
User name	blank	x	x				
Password	blank	x	x				
Quality of Service							
802.1p	5	x	x				
VLAN (802.1Q)	Activated	x	x				
ToS	5	x	x				
Diffserv	Activated	x	x				
Security							
Encryption	??	x	x				

**Table 9.1      Configurable Parameters and Commands**

## 2 Examples of files to configure auto provisioning via TFTP

### Example of file INF SIP:

```
[provision_mode]
provision_mode=auto_provisioning

[application]
fw_filename=st2030SEG.051118.1.38.zz

[config]
telcfg=TelConf2030SEG_051118.txt
common_config=st2030s_common.txt
```

### Example of file INF MGCP:

```
[provision_mode]
provision_mode=auto_provisioning

[application]
fw_filename=v2030MEC.060104.1.20.zz

[deckfile]
deckfile=MEC.060104.abrege.thd
deckfile=MEC.060104.call.thd
deckfile=MEC.060104.config.thd
deckfile=MEC.060104.default.thd
deckfile=MEC.060104.filtrage.thd
deckfile=MEC.060104.local.thd
deckfile=MEC.060104.log.thd
deckfile=MEC.060104.main.thd
deckfile=MEC.060104.phonebook.thd
deckfile=MEC.060104.register.thd
deckfile=MEC.060104.renvoi.thd
deckfile=MEC.060104.supervision.thd

[config]
telcfg=st2030m_v111_cfg.txt
common_config=st2030s_common.txt
```

**Example of file Common\_config SIP:**

```
[ipp]
LCDvoltLevel=41
RINGER_Flag=0
RingToneMP0=0
LocalMusicOnHold=1
LanguageType=0
AutoAnsTimer=0
PhoneLock=0
PasswordString=0000
AutoStopTimer=0
AutoRejectTimer=0
AutoTurnOffSPKTimer=0
DoNotDisturb_Type=0
DoNotDisturb_Duration=0
DoNotDisturb_AbHour=0
DoNotDisturb_AbMin=0
DoNotDisturb_Year=0
DoNotDisturb_Month=0
DoNotDisturb_Hour=0
DoNotDisturb_Mday=0
DoNotDisturb_Min=0
CaLogPrefix=
Suppress-DomainName-Flag=1

[net]
WanModFlag=1
ipwan=10.1.24.88
MaskWan=255.255.255.0
Gateway=10.1.1.1
DNSSrv1=192.168.1.1
DNSSrv2=192.168.1.2
DSCPflag=0
VLAN=0
ESWITCH_LAN=1

[sip]
TEL01Use=1
TEL02Use=0
TEL03Use=0
TEL04Use=0
RegisterServerMP1=10.1.25.105
ProxyServerMP1=10.1.25.105
ServiceDomainMP1=oxe.thomson.net
ExpireTimeMP1=3600
RegisterServerBK1=10.1.25.105
ProxyServerBK1=10.1.25.105
ServiceDomainBK1=oxe.thomson.net
ExpireTimeBK1=3600
TEL1Number=2208
DisplayName1=2208
```

```
regid1=2208
regpwd1=2208
SessionExpire=1800
SessionExpireMin=100
SessionMethod=0
CodecPriority_G711mu=2
CodecPriority_G711a=1
CodecPriority_G729=3
CodecPriority_G723=4
subscribe_event=0
sw_tr_vmail_flag=0
sw_tr_vmail_address1=
PrivacyFlag1=4
sw_anon_reject=0
CallBlockType=0
sw_CF=0
CF_Address1=
CF_domain=
sw_CFA=0
sw_CFB=0
sw_CFNA=0
sw_call_wait=0
CFNA_Time1=32
sw_not_disturb=0
USRPhoneFlg=1
sip_reg_srv_portMP1=5060
sip_reg_srv_portBK1=5060
sip_pxy_srv_portMP1=5060
sip_pxy_srv_portBK1=5060
sip_portMP1=5060
sip_portBK1=5060
RtpPort=41000
VoiceMailAddr=
VoiceMailTelNum=
VoiceMailPort=5060
sw_notify_autoprovision=1
DNDResp=2
UnRegister_priorFlg=1
UnRegister_priorFlgMP0=1
UnRegister_priorFlgMP1=1
UnRegister_priorFlgMP2=1
UnRegister_priorFlgMP3=1
UnRegister_priorFlgMP4=1
UnRegister_priorFlgBK0=1
UnRegister_priorFlgBK1=1
UnRegister_priorFlgBK2=1
UnRegister_priorFlgBK3=1
UnRegister_priorFlgBK4=1
```

```
[sys]
NormalDigitFlag=1
Dialednum_timeout=5
```

Current\_Max\_Multiline=10  
EmergencyDialPlan=|911|11x|1[2-8]|  
VOIPDialPlan=  
TelnetID=administrator  
TelnetPWD=784518  
UserID=user  
UserPWD=0000

[autoprovision]  
AutoprovisionFlag=1  
AutoprovisionTimeDays=0  
Autoprovisionstarttime=00:00  
AutoprovisionTimeSpan=0  
AutoprovisionRetryPeriod=30  
AutoprovisionServer=  
AutoprovisionFilename=  
AutoprovisionConfigname=

[qos]  
DSCH1=46  
DSCH2=40  
TOS=160  
VLANid1=1  
VLANid2=1  
VLANTag1=6  
VLANTag2=6  
VLANTagdata=6

[ftp]  
FTPID=  
FTPPWD=

[ntp]  
NTPFlag=1  
NtpIP=192.43.244.18  
NtpDaylight=0  
NtpZoneNum=50  
NtpSyncTime=1

[pppoe]  
PPPoEID=  
PPPoEPWD=

[stun]  
STUNFlag=0  
STUNSrv=

[upnp]  
UPnPFlag=1

**Example of file Common\_config MGCP:**

```
[mgcp]
ActivatedCAProfile=1
caip=ca@[213.176.148.98]:2727
```

```
[autoprovision]
autoprovisionTimeDays=0
autoprovisionstarttime=00:00
autoprovisionTimeSpan=0
```

```
[net]
DNSSrv1=10.22.1.1
DNSSrv2=10.22.1.1
WanModFlag=1
```

```
[sys]
UserID=user
UserPWD=000000
```

```
[autoprovision]
AutoprovisionFlag=1
AutoprovisionTFTPServer=213.213.133.161
AutoprovisionTimeDays=0
Autoprovisionstarttime=00:00
AutoprovisionTimeSpan=0
```

```
[qos]
TOS=160
```

```
[ntp]
NtpDaylight=1
NTPFlag=1
NtpIP=193.4.194.10
NtpZoneNum=22
```

### **Example of file TelConf: (Don't touch)**

It is not necessary to put it...

### **Specific file MAC:**

I have no example but it look like common\_config.

### **Files Deck: (only MGCP)**

#### **Until version 1.20 (11 files):**

MEC.060104.abrege.thd  
MEC.060104.call.thd  
MEC.060104.config.thd  
MEC.060104.filtrage.thd  
**MEC.060104.local.thd**  
MEC.060104.log.thd  
MEC.060104.main.thd  
MEC.060104.phonebook.thd  
MEC.060104.register.thd  
MEC.060104.renvoi.thd  
MEC.060104.supervision.thd

#### **Version 1.21 (10 files) :**

MEC.060127.abrege.thd  
MEC.060127.call.thd  
MEC.060127.config.thd  
MEC.060127.filtrage.thd  
MEC.060127.log.thd  
MEC.060127.main.thd  
MEC.060127.phonebook.thd  
MEC.060127.register.thd  
MEC.060127.renvoi.thd  
MEC.060127.supervision.thd

#### **Version 1.22 (11 files) :**

MEC.060224.abrege.thd  
MEC.060224.call.thd  
MEC.060224.config.thd  
MEC.060224.filtrage.thd  
**MEC.060224.default.thd**  
MEC.060224.log.thd  
MEC.060224.main.thd  
MEC.060224.phonebook.thd  
MEC.060224.register.thd  
MEC.060224.renvoi.thd  
MEC.060224.supervision.thd

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