THOMSON

ST2030 SIP

VoIP Business Phone

Administrator Guide

Thomson Telecom
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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.thomsonbroadbandpartner.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 used in accordance with the instructions in this manual, may cause interference to radio communications. This equipment has been tested and found to comply with the limits for a Class B digital device pursuant to Subpart J of Part 15 of FCC rules, which are designed to provide reasonable protection against radio interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, in which case the user, at his own expense, will be required to take whatever measures are necessary to correct the interface.

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Part 1 Safety Information

Operating conditions

This telephone is to be operated on a local area network. The telephone must be equipped with the appropriate software version. This guide is written for the actual version of firmware v2.68, you can download the latest on:

http://www.thomsonbroadbandpartner.com

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

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.

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.

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: EU/AUS 9VDC/1000mA or 800 mA, UK/US 12VDC /1000mA.

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.

Qualified service

No repair can by 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.

Part 2 Product overview and connections

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.

Note and Caution

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

Note

Contains related information that corresponds to a topic.

Caution

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

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 28 memory keys and headset

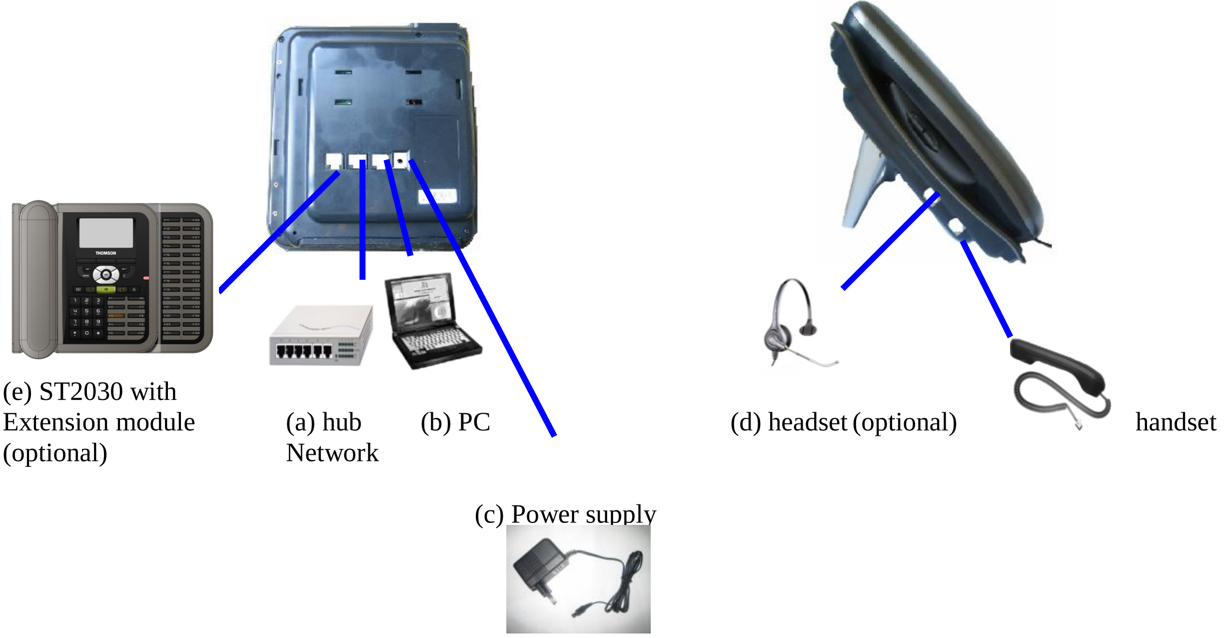


Figure 2.1 IP Phone Cable Connections

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.

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 or 12VDC /1000mA liner/switching.

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 ST 2030 supports PoE as defined by 802.3af Standard
It is class 2, and supports class negociation (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.

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)

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.

Main Features

- 6-line LCD display
- Font supports ISO8859-2, ISO8859-5, ISO8859-8
- Connectivity: Integrated 2 ports 10/100 Ethernet switch
- Phone services:
  - Multilines (up to 9), 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

Note

This administrator guide is based on firmware v.1.66, you can download the latest administrator guide on:

www.thomsonbroadbandpartner.com

Supported Features

In addition to the physical features illustrated in Figure 3.1 and table 3.1, your Thomson ST 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 100Base100BAseT 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 orG729a)

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 (9 lines)
- Redial, Mute, Call log, phone book (30 entries), 10 Speed dial memory keys
- Hands free (full-duplex)

Supported Protocols

Your Thomson ST 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.

Abbreviations

SIP	**S**ession **I**nitiation **P**rotocol. An IP telephony signalling text-based protocol developed by the IETF
DNS	**D**omaine **N**ame **S**erver
DHCP	**D**ynamic **H**ost **C**ontrol **P**rotocol
FTP	**F**ile **T**ransfer **P**rotocol
H323	An ITU standard for realtime voice and videoconferencing over packet networks, including LANs, WANs and the Internet
LAN	**L**ocal **A**rea **N**etwork
MGCP	**M**edia **G**ateway **C**ontrol **P**rotocol
RFC 3261	**R**equest **F**or **C**omments. This document describes the specifications for business extended services under the MGCP protocol.
TFTP	**T**rivial **F**ile **T**ransfer **P**rotocol
XML	E**X**tensible **M**arkup **L**anguage
DTMF	**D**ual **T**one **M**ulti-**F**requency 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.

Part 3 Phone operations

General appearance

The general appearance of the ST2030 is as below:

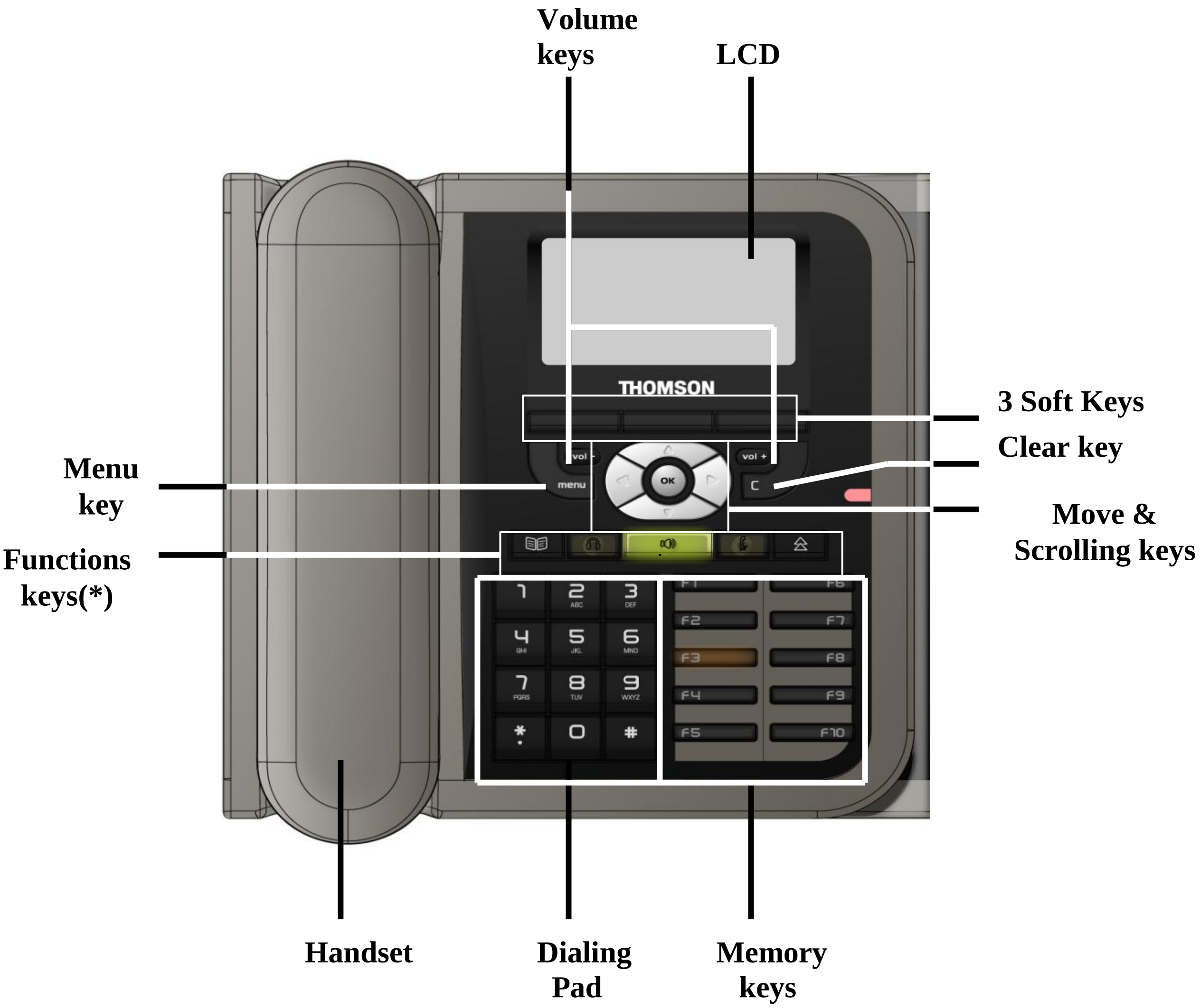
ST2030 front view

ST2030 side view

ST2030 with his extension module

Physical Features

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

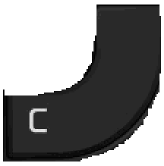


* Keys from left to right: phone book/ headset/ speaker/ mute and redial.

Figure 3.1 The IP Phone User panel

Display keys

Confirmation key:

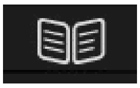


Clear key:

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.

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.

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

Display

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

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”.

Definitions of Soft Keys

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

Soft Key	Brief Description
A>a>1	Input mode switch in edit state
Active	Activate option/service
Admin	Enter to administration sub-menu
Answer	Answer an incoming call
Anym	Shortcut to “Anonymous”
Autoan	Shortcut to “Auto Answer”
Back	Return to previous menu
CalLog	Shortcut to Call log
Cancel	Cancel an action or exit to previous menu without applying changes
CBack	Perform Callback function
CBlock	Shortcut to “Call Block”
CFwd	Shortcut to “call forwarding”
CidDis	Shortcut to “Number Display”
Change	Change phone password
Conf	Create a conference call. During a call/conference and another call is coming, press the soft key will join a conference
DelChr	Delete character in edit mode
Delete	Delete specific entry
Detail	Show detailed information
Dial	Dial an entered phone number

DNDst	Do Not Disturb
Down	Contrast adjustment
Dsub	Shortcut to “Dial Subscriber”
EndCal	Terminate active call
Edit	Edit parameter content
Exit	Return to previous menu
Format	Change display format of specific parameter, such like Date and Time etc.
Hold	Hold current active line
Join	Join several connected calls to create a conference call
Lock	Shortcut to lock phone
Mail	Dial to voice mail server
MsCall	Shortcut to missed call list
NewCal	Make a new call
OFF	Set specific service OFF
OK	Confirm setting
ON	Set specific Service ON
Option	Shortcut to phone services
Park	Store a call using Call Park
PhBook	Shortcut to phone book
PickUp	Answer a call on another phone
Play	Play music or melody
Reject	Reject a call
Remove	Remove a conference participant
Resume	Resume to original call from call transfer
RtPark	Retrieve Park
Save	Save the chosen setting
Select	Select current item on the screen
Stop	Stop music or melody

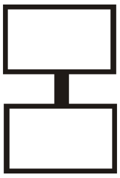
Transf	Transfer a call
TrMail	Transfer call to voice mail system
UnHold	Unhold a held line
Up	Contrast adjustment
User	Enter to user submenu
View	Show details when data item content is more than LCD can display in one line

Table 3.2 Definitions of soft keys

Icons & Indicators

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

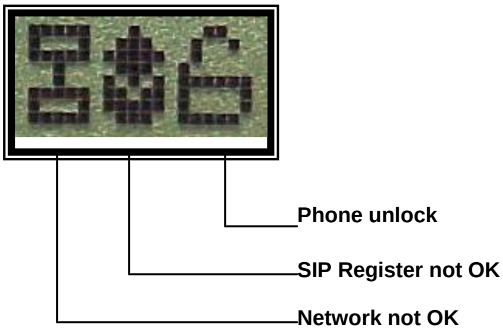
- Line 1 : date/time and icons information
- Line 2 to Line 5: operation information
- Line 6 : Soft keys display



➜ 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.

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** **Green**	**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

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 + 2x28 = 66 possible “multiline” keys.

Memory

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

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.

Part 4 Call Services

Talking Mode & Operations

5 kinds of communication mode

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

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

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

Operation of mode switch during call

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

Table 4.1 Operations of mode switch during call

Main Call Functions

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 “↑”, ”↓” or “Redial” key to display the last 30 phone numbers you called

Key Pressed	LCD display		
“↑”, ”↓” : Scrolling and select number you desired. “←” or “→”: scrolling to view rest. Press **View** to get details (Date, Time)	***Date*** **View**	***Time*** **Dial**	***Icons*** **Back**
	Delete	**Del All**	
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 **CallLog** to show Call Log messages Scrolling and confirm by pressing **Ok**	***Date*** ***Time*** ***Icons*** **Missed Call LoG** **Received Call Log** **Dialed Call Log** **Delete Call Logs** **Select** **Back**
“↑”,”↓”: Scrolling and dial number you desired and press “**Dial**» or **Ok**	***Date*** ***Time*** ***Icons*** **View** **Dial** **Back** **Delete** **Del All**

- Use the phonebook

Key Pressed	LCD display
Press 🕮 or **PhBook** to show phone book information. “↑”,”↓”: Scrolling and press “**Dial**» or **Ok**	***Date*** ***Time*** ***Icons*** **View** **Dial** **Delete** **Add** **Back**

Last number Redial:

⇮

- Pick up the handset and press ⇮ .
- Press 🔈 then ⇮

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

Answer a call

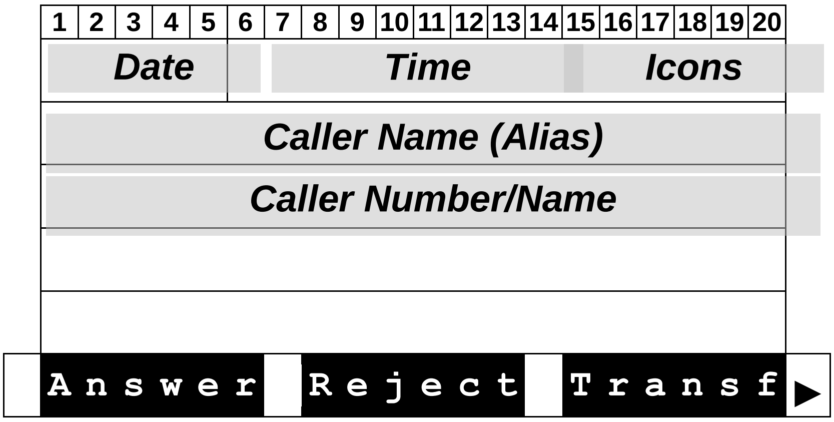
When a call is coming, the caller number and alias are displayed, if the caller number is stored in the phonebook, the associated name is displayed.

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

Note

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

When the phone is ringing



Reject a call

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

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

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**.

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**

Call functions during conversation

Hold and Retrieve a call

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

Call Transfer

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

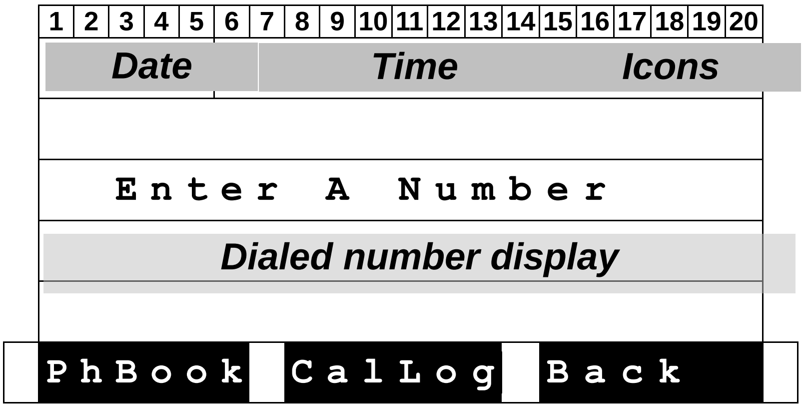
Operation of "Blind" transfer

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

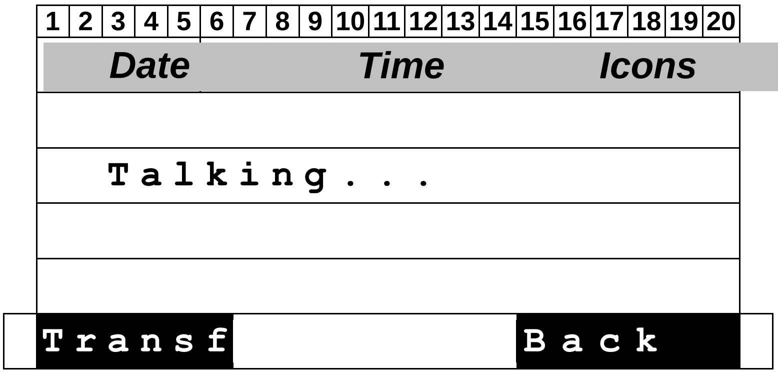
Operation of "Attended" transfer

- 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



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

Conference call

The ST 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.

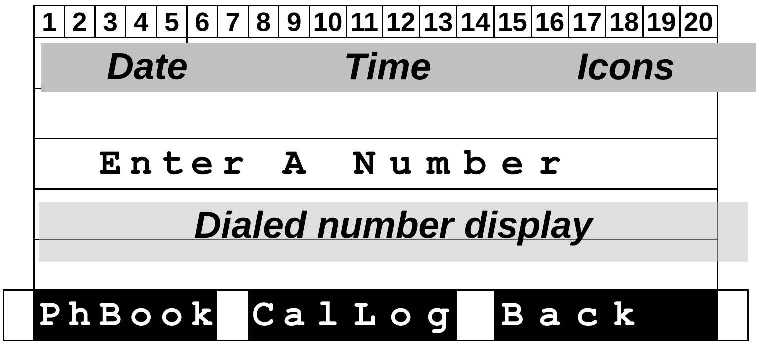
Drop any one participant of the conference call:

- 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.



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

Dialed number display

PhBook CalLog Back

When a call/conference is active

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

Date	Time	Icons	
01 Number/Name			
Conf	Duration		
02 Number/Name			
Conf	Duration		
Hold	Transf	Conf ▶	Page 1
◀ CalLog	PhBook	Remove	Page 2
◀ Park	NewCal		Page 3

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...		
Conf		Back

Options during a call

Group Listening
Hands free

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.

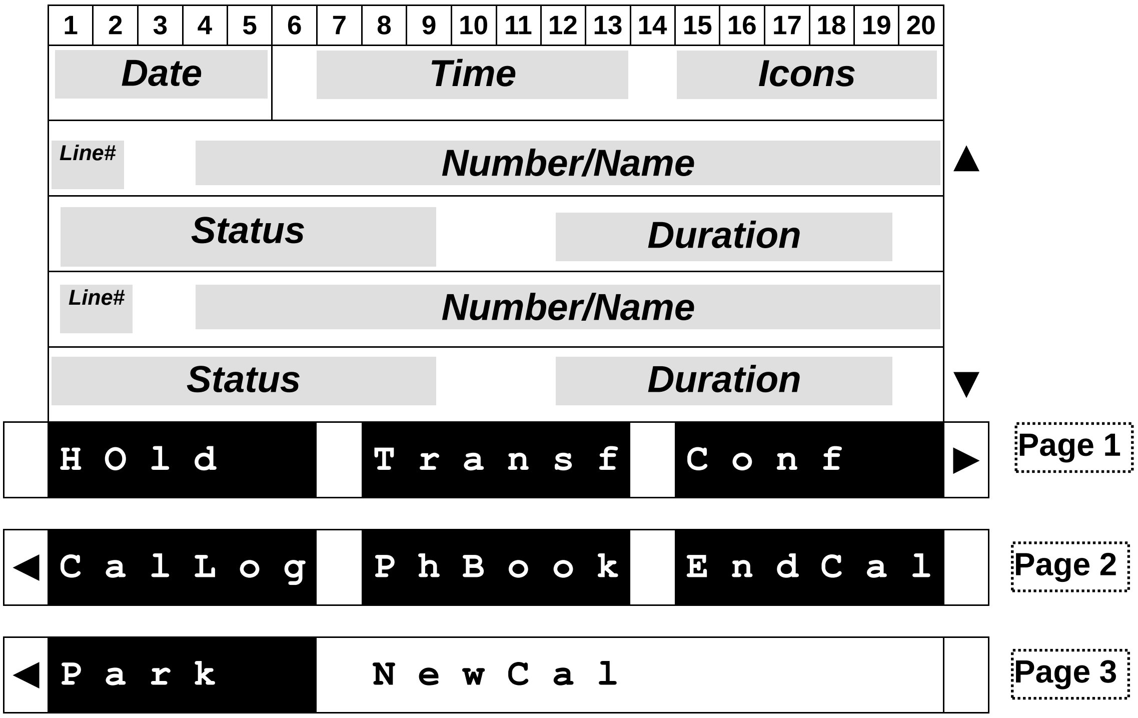
Mute:

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

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.



Note

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

Definition of Status

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

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

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

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

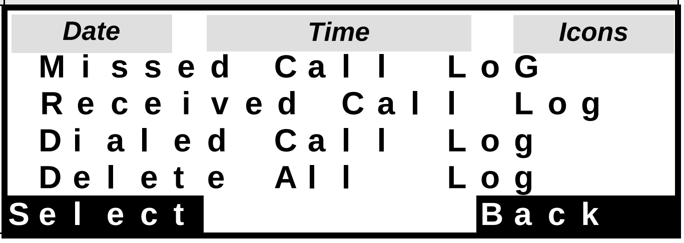
Retrieve Call park on an other phone

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

Phone book & Call log

Call logs

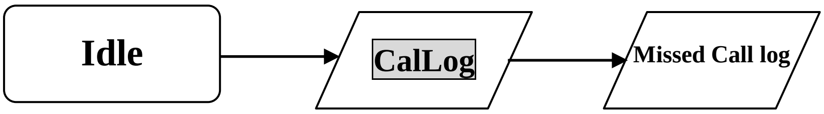
Press **CalLog** to view, add, clear call logs and dial from call logs.
There are 3 kinds of call logs: Missed call logs, Received call logs and dialed call logs.
You can also dial from call logs by pressing **OK** on selected call.

Key Pressed	LCD display
Press **CalLog** to show Call Log messages Scrolling and confirm by pressing **Select**	 Date Time Icons Missed Call LoG Received Call Log Dialed Call Log Delete All Log Select Back

Options list

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

Missed Calls



- Entry number is refreshed when pressing scrolling key
- Scroll by pressing Up/Down key
- Exit to higher-level menu by pressing **C** or **"Back"**
- If the number is stored on phone book, the associated name will be displayed.

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

Line 1	Date	Time	(n / m)
Line 2	nbr-1		
Line 3	nbr-2		
Line 4	nbr-3		
Line 5			
Line 6	View Delete	Dial DelAll	BacK

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**

- 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

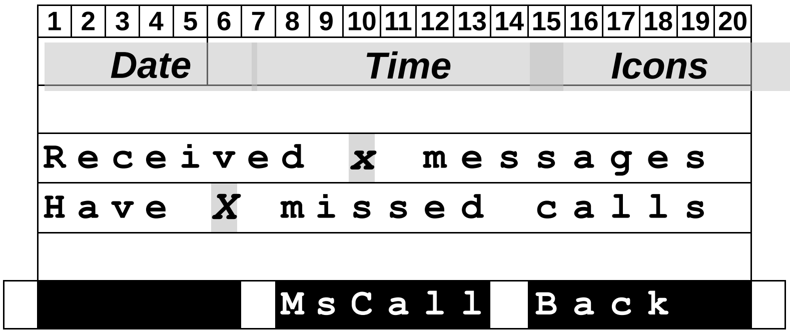
Line 1	Date	Time	
Line 2			
Line 3			
Line 4			
Line 5			
Line 6	Edit		Back

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 “←” or “→” to get remainder.
Alias (Phone name)	Alphanumeric	20	
Date Time	YY/MM/DD hh:mm		

Display when there are any message or missed calls



Received x messages

Have X missed calls

MsCall

Back

Note

- 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.

Received Call & Dialed Calls

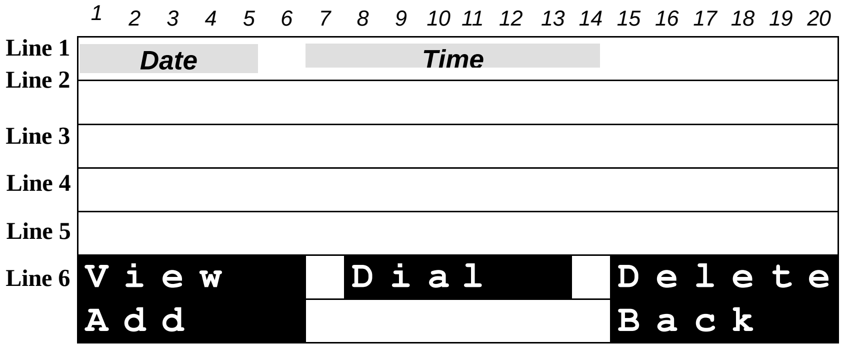
Be the same as operation of Missed Calls

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.

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



View

Dial

Delete

Add

Back

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**		

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 å ã â ç	A B C 2 Ã Ä Å Æ
3	d e f 3 è é ê ë	D E F 3 É
4	g h i 4 ì 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* @ % ( ) [ ] < >	
*****	* # ? ! : ; ‘ “ _ /	
#	#	

Note

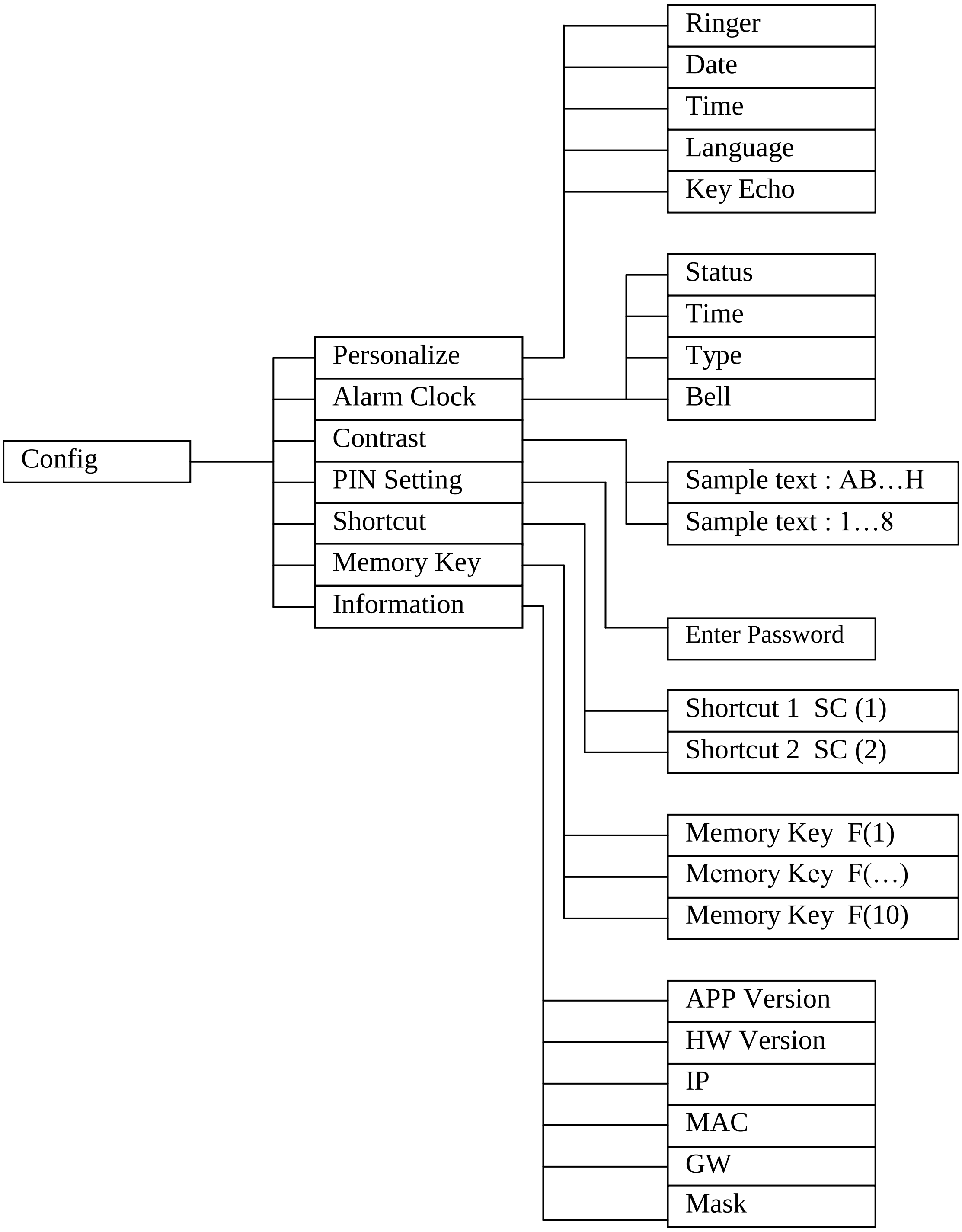
Softkey **ABab12** : Switch uppercase (with indicator “>ABC”), lowercase (with indicator “>abc”) and numeric (with indicator “>123”) when entering letters.

All other symbol needed will be appended to ‘*’ key.

Table 4.2 Character input method table

Part 5 Configuration through LCD

Menu list



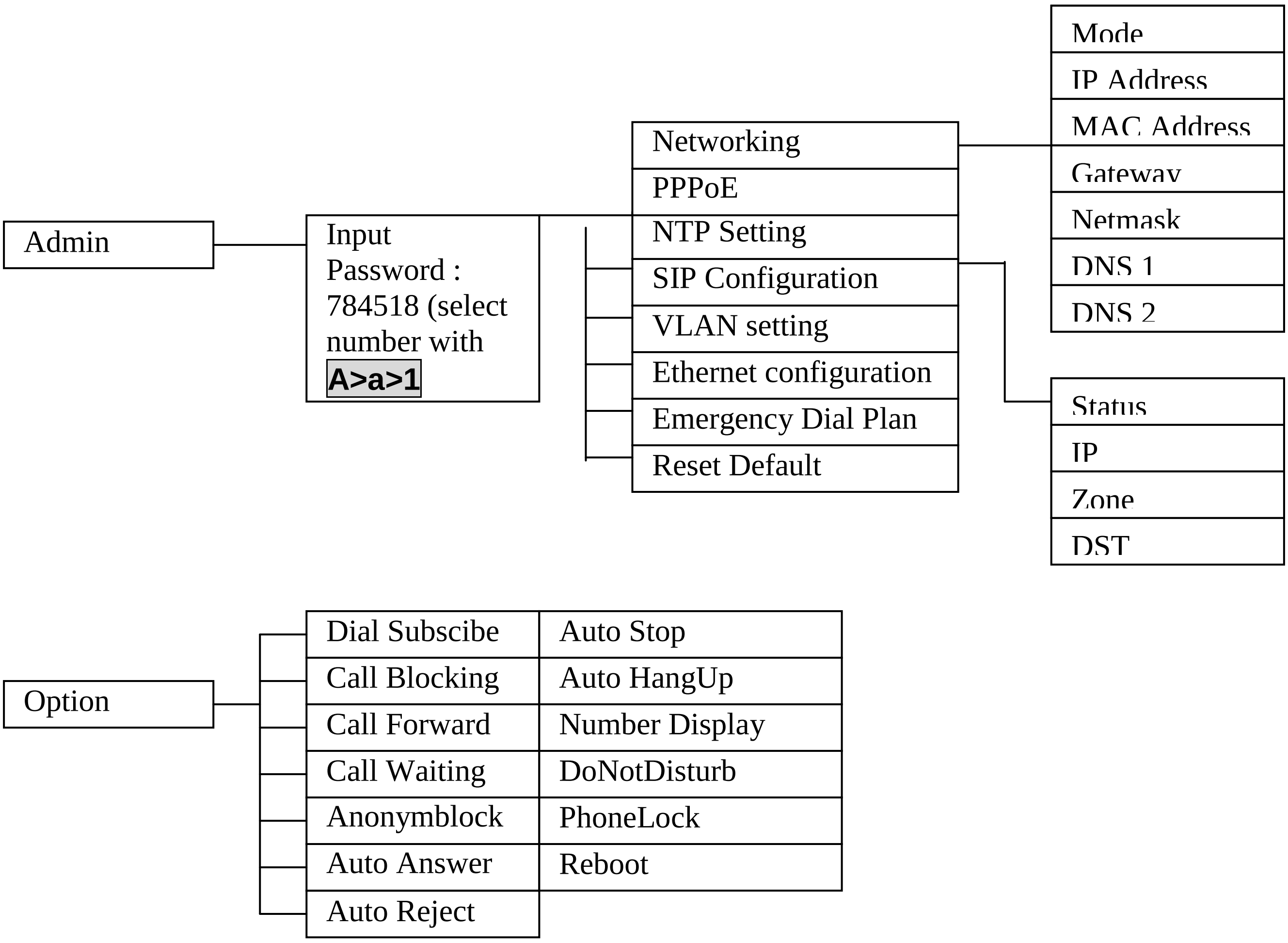
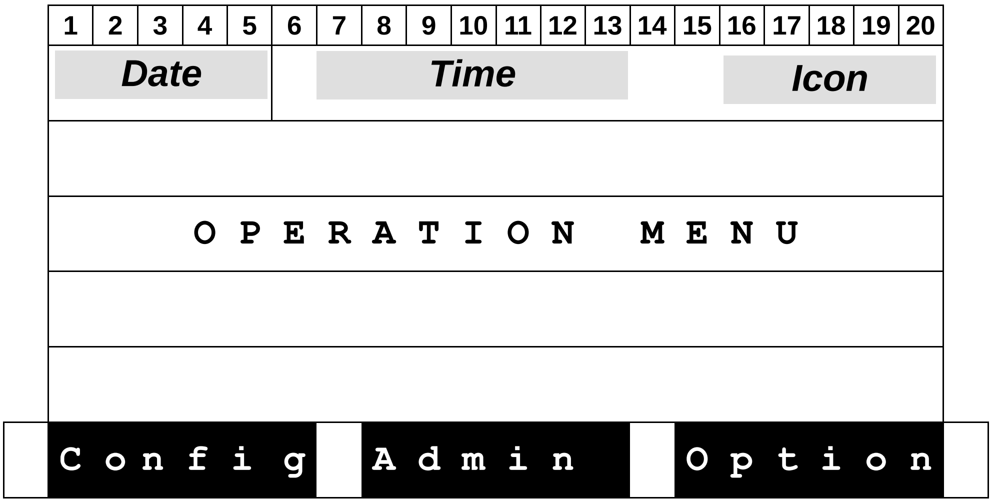


Figure 5.1 Operation Menu LCD Display

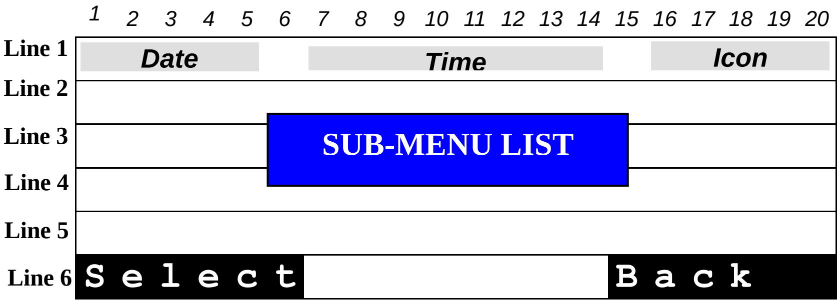
Operation menu display

Home operation menu display



Config Menu Display

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



Sub-menu list of User Setting

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

Admin Menu Display

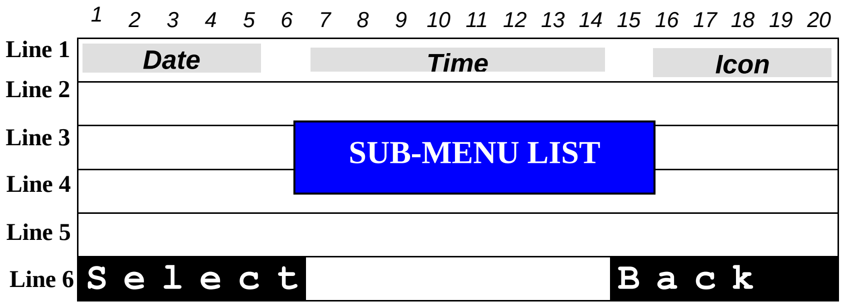
Caution

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

◆ Scroll by pressing Up/Down key

◆ Select by pressing **OK** or **"Select"**

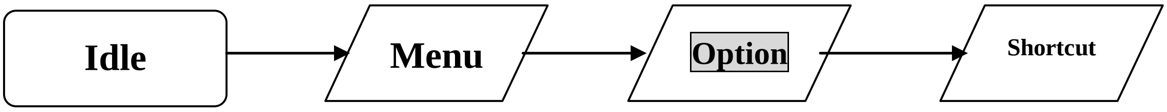
◆ Return to idle state by pressing **C** or **"Back"**



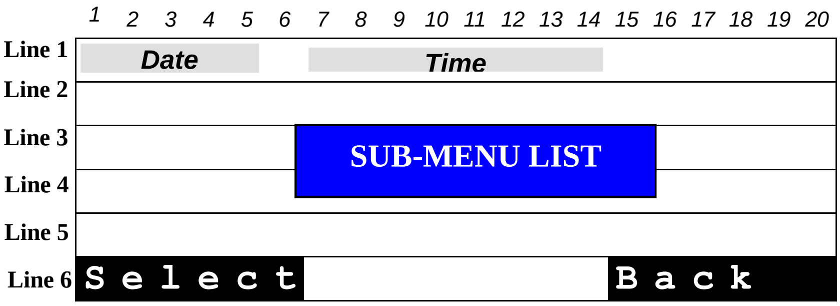
Sub-menu list of admin setting

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

Option Menu Display



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



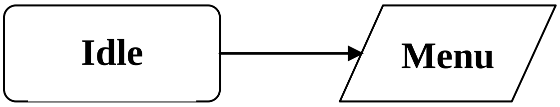
Sub-menu list of admin setting

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 HangUp**	
(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 -> **Dsub** Call Blocking -> **CBlock** Call Forward -> **CFwd** Anonymous -> **Anym** Auto Answer -> **AutoAn** Number Display -> **CidDis** ***Else . . .***

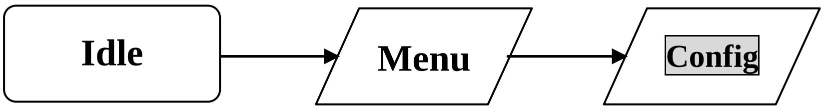
Detailed LCD menu settings



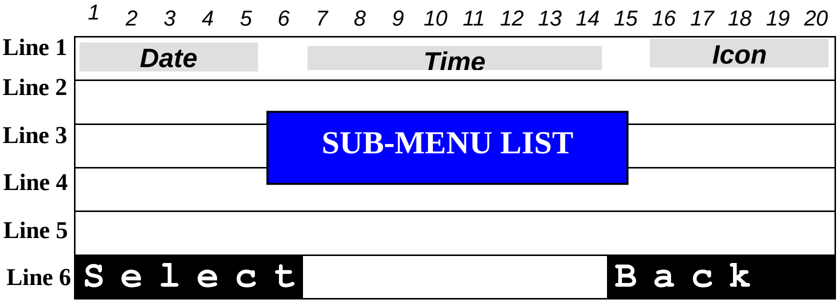
Home operation menu display

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
Date						*Time*													
			O	P	E	R	A	T	I	O	N		M	E	N	U			
U	s	e	r				A	d	m	i	n			O	p	t	i	o	n

User Menu Display



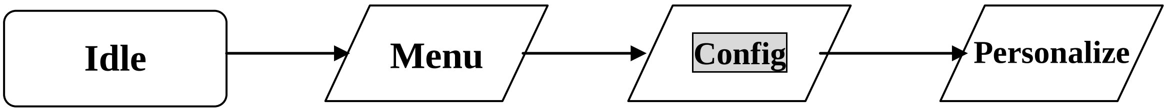
- Scroll by pressing Up/Down key
- Select by pressing **OK** or **"Select"**
- Return to idle state by pressing **C** or **"Back"**



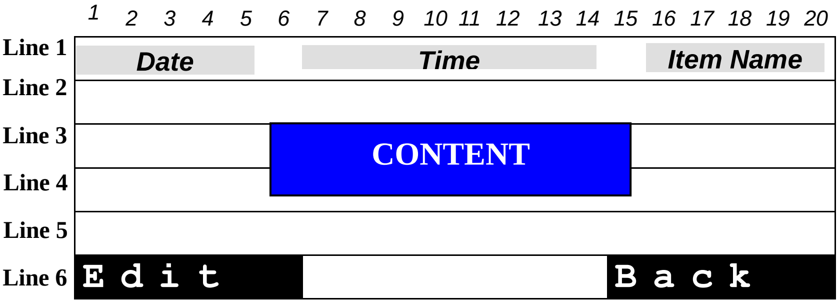
Sub-menu list of User Setting

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

Personalize Setting



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

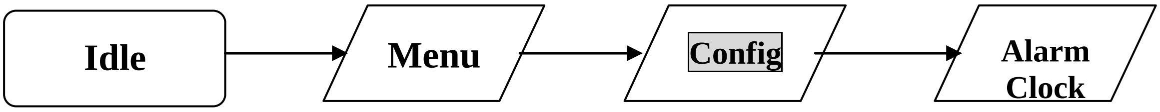


Content of Phone Option/Parameters

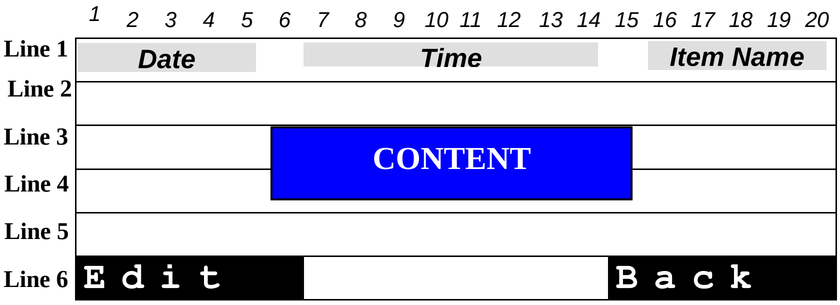
Item Name	Configurable	Data Format	Options List	Comment
Name	**Yes**	Alphanumeric		Max. length: 15
Ringer	**Yes**	Options	**Ringer-1** **…** **Ringer-4** **Canon** **Shuffle**	You can press soft key **Play** when editing the content.

Date	**Yes** (Configurable when NTP is disabled)	MM/DD/YY DD/MM/YY YY/MM/DD		You can press soft key **Format** to change format. ***MM/DD/YY*** ***DD/MM/YY*** ***YY/MM/DD***
Time	**Yes** (Configurable when NTP is disabled)	hh:mm	hh:mm am hh:mm	You can press soft key **Format** to change format. ***12 Hours*** ***24 Hours***
Language (Lang)	**Yes**	Options	**English** **French** **Spanish** **German** **Italian** **Portuguese** **Deutsch** **Norsk** **Russian** **Nederlands**	
Key Echo (KeyEch)	**Yes**	Options	**ON** **OFF**	

Alarm Clock



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

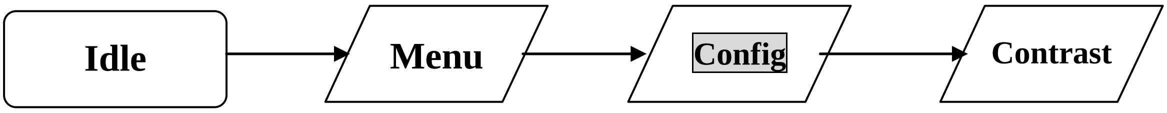


Content of Alarm Parameters

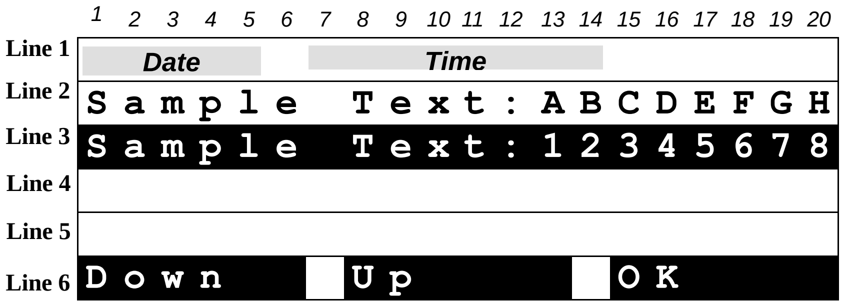
Item Name	Configurable	Data Format	Options List	Comment
Startup (Status)	**Yes**	Options	**ON** **OFF**	
Time	**Yes**	hh:mm		
Type	**Yes**	Options	**One Shot** **Periodic**	
Bell Type (Bell)	**Yes**	Options	**Ringer-1** **…** **Ringer-4** **Canon** **Shuffle**	You can press soft key **Play** to play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

First activate alarm clock then change time.

Contrast

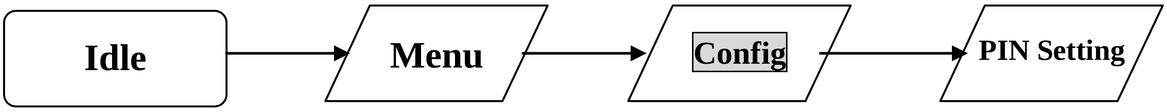


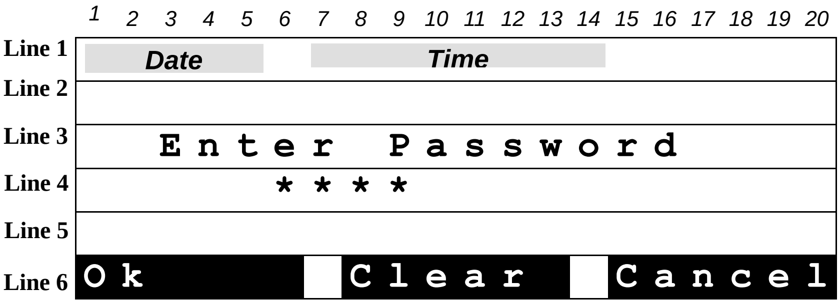
- Scroll by pressing Up/Down key
- Exit to higher-level menu by pressing **C**
- Press "**Down**" or "**Up**" to adjust contrast
- Press "**OK**" to confirm and exit to higher-level menu.
- Press "Cancel" to cancel the setting.



Line 1 Date Time
Line 2 Sample Text:ABCDEFGH
Line 3 Sample Text:12345678
Line 4
Line 5
Line 6 Down Up OK

PIN Setting



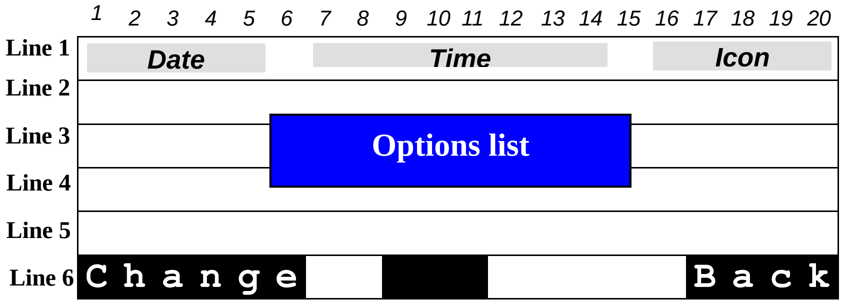


Line 1 Date Time
Line 2
Line 3 Enter Password
Line 4 ****
Line 5
Line 6 Ok Clear Cancel

Caution

The password is : 0000

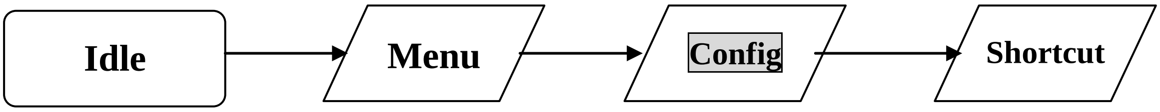
- Scroll by pressing Up/Down key
- Select by pressing **OK** or **“Select”**
- Return to idle state by pressing **C** or **“Back”**



Options list

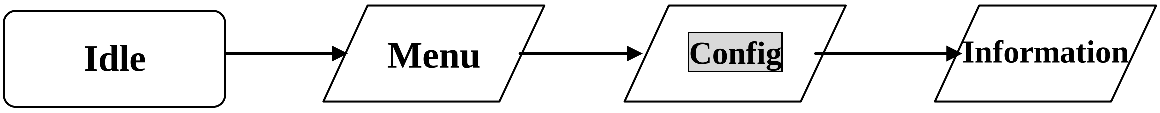
No.	Option Message	Data Format	Comment
(1)	**Flag**	**ON** **OFF**	◆ When Status is ON, the phone lock setting will be authorized by checking personal code. ◆ Press **Change** to enable/disable the item.
(2)	**Change Code**		◆ Press soft key “**Change**” to change PSW ◆ The default password is **0000**.

Shortcut Setting



You can edit two shortcuts

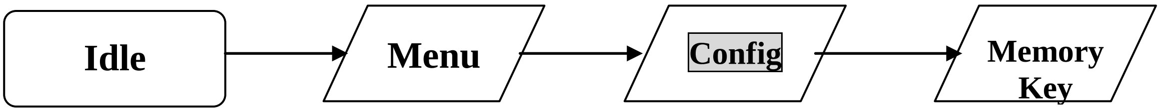
Information



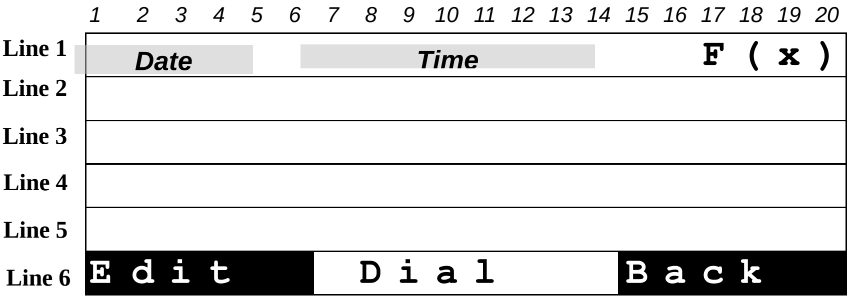
Content

Item Name	Configurable	Comment
FW version	**No**	Indicate to the user the firmware's version
HW version	**No**	Indicate to the use r the hardware's version
IP address	**No**	Indicate to the user the IP address
MAC address	**No**	Indicate to the user the MAC address
Gateway	**No**	Indicate to the user the gateway address
Mask	**No**	Indicate to the user the mask address

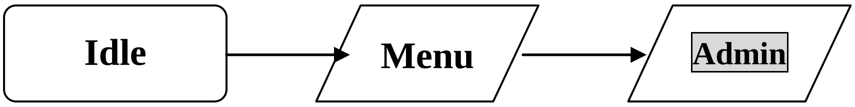
Memory Keys Setting



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



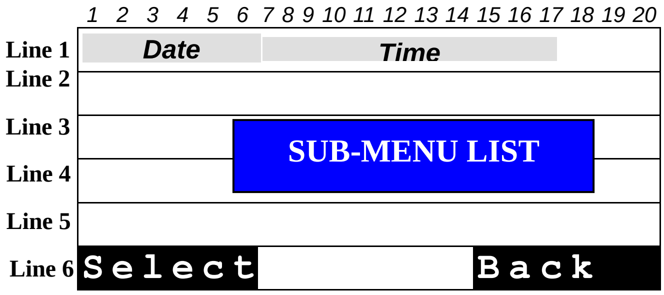
Menu of Admin Settings



Caution

You have to input a password to access Admin settings: **784518**

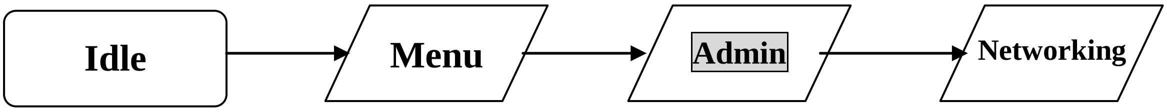
- Scroll by pressing Up/Down key
- Select by pressing **OK** or **"Select"**
- Return to idle state by pressing **C** or **"Back"**



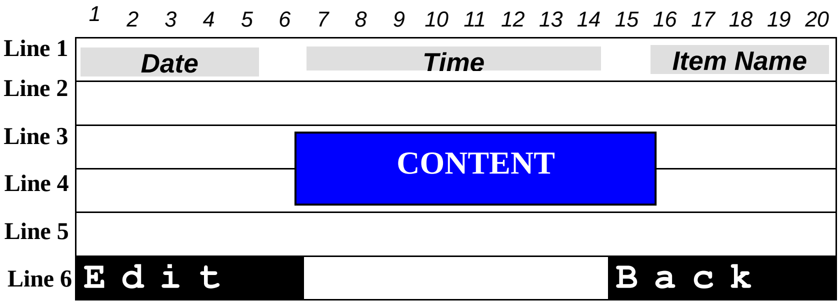
Sub-menu list of admin setting

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

Networking Configuration



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

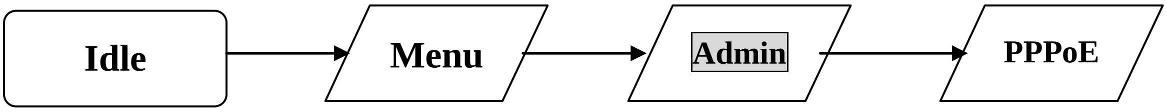


Content of Network Parameters

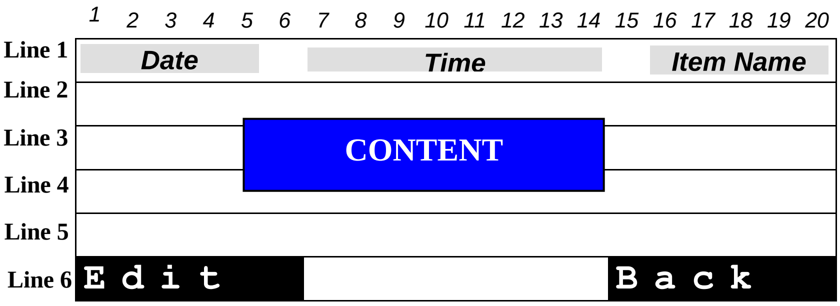
Item Name	Configurable	Data Format	Options List	Max. Length
Mode	**Yes**	Options	**Fix IP** **DHCP** **PPPoE**	
IP	**Yes (For Fix-IP only)**	xxx.xxx.xxx.xxx		15
MAC	**No**	xx:xx:xx:xx:xx:xx		
Gateway (GW)	**Yes (For Fix-IP only)**	xxx.xxx.xxx.xxx		15
Net mask (Mask)	**Yes (For Fix-IP only)**	xxx.xxx.xxx.xxx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



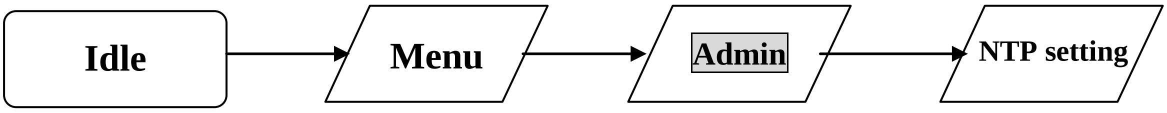
- ItemName is refreshed when pressing scrolling key
- Scroll by pressing Up/Down key
- Exit to higher-level menu by pressing **C** or **"Back"**



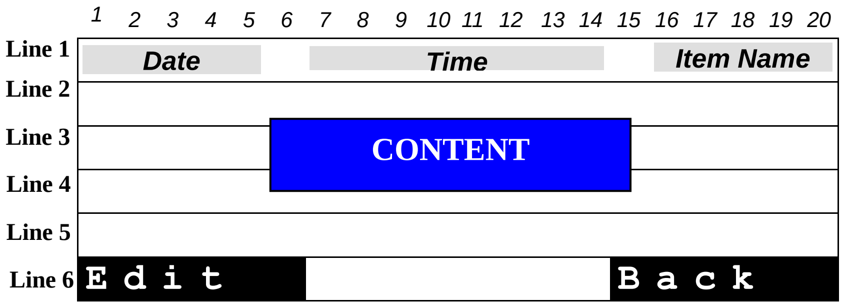
Content of PPPoE Parameters

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

NTP Setting



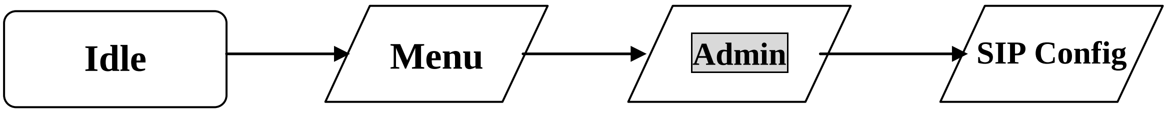
- ItemName is refreshed when pressing scrolling key
- Scroll by pressing Up/Down key
- Exit to higher-level menu by pressing **C** or **"Back"**



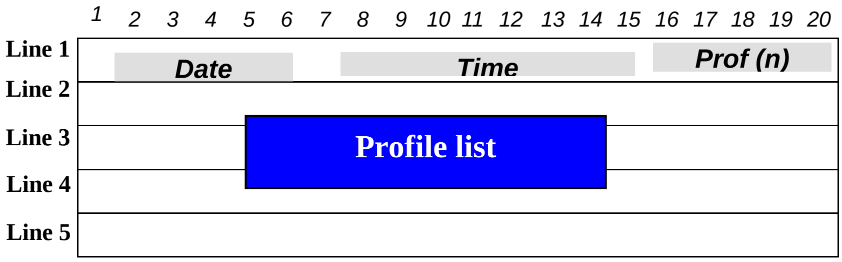
Content of NTP Parameters

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"**

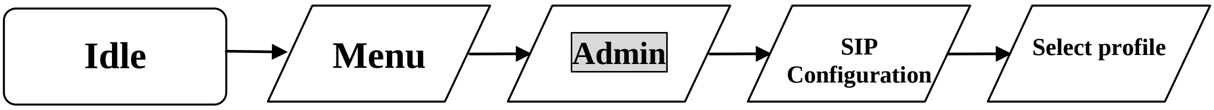


Line 6 Select Active Back

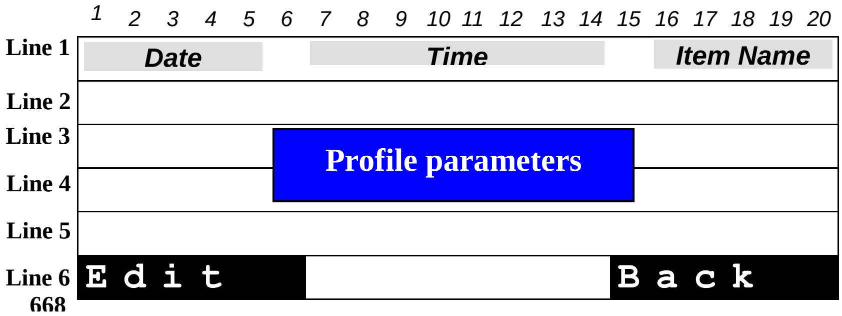
Profile List

No.	Profile Name	Comment
(1)	**Profile 1**	◆ 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)	**Profile 2**	
(3)	**Profile 3**	
(4)	**Profile 4**	

SIP Profile Parameters



- 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 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
2ndPxy	**Yes**	xxx.xxx.xxx.xxx DomainName		15 127
2ndReg	**Yes**	xxx.xxx.xxx.xxx DomainName		15 127

Part 6 Configuration via Web Interface

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.

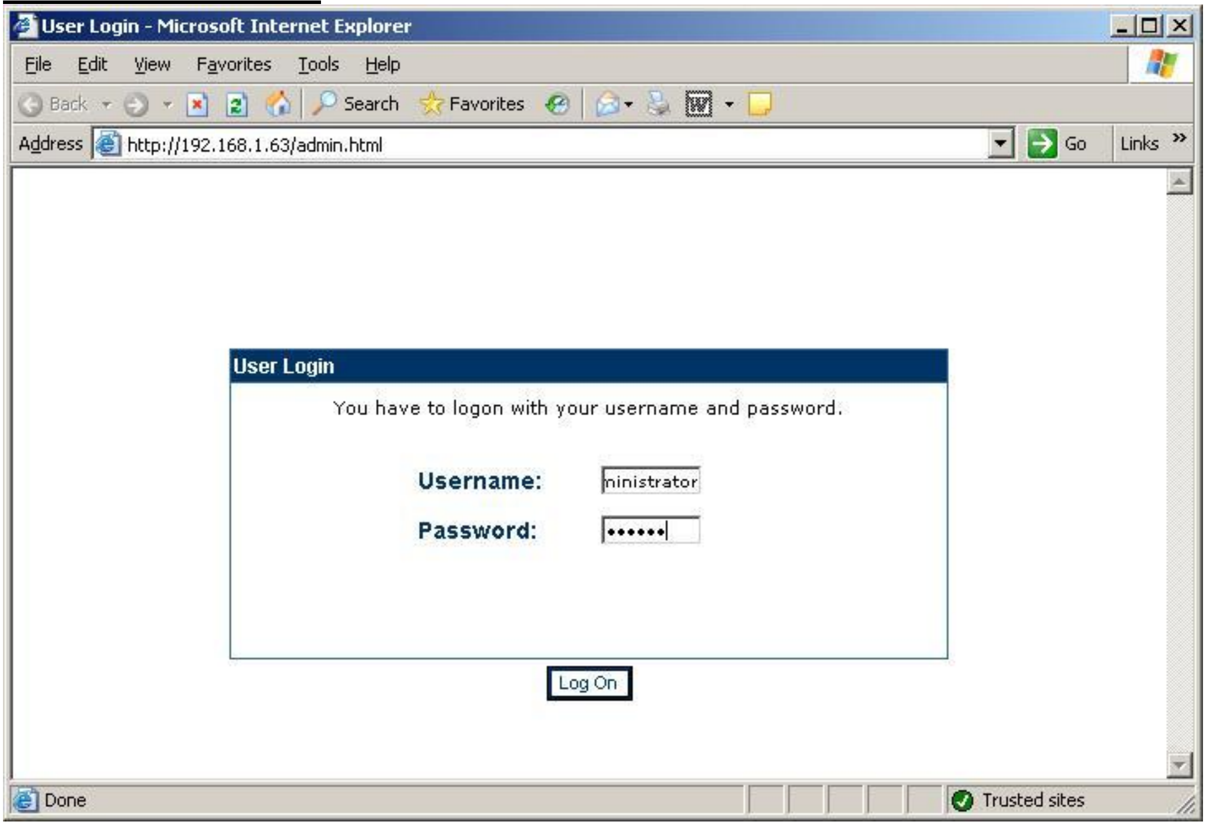
Step 3 – Reboot the phone

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.

Detailed Web pages

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

CONNECTION PAGE



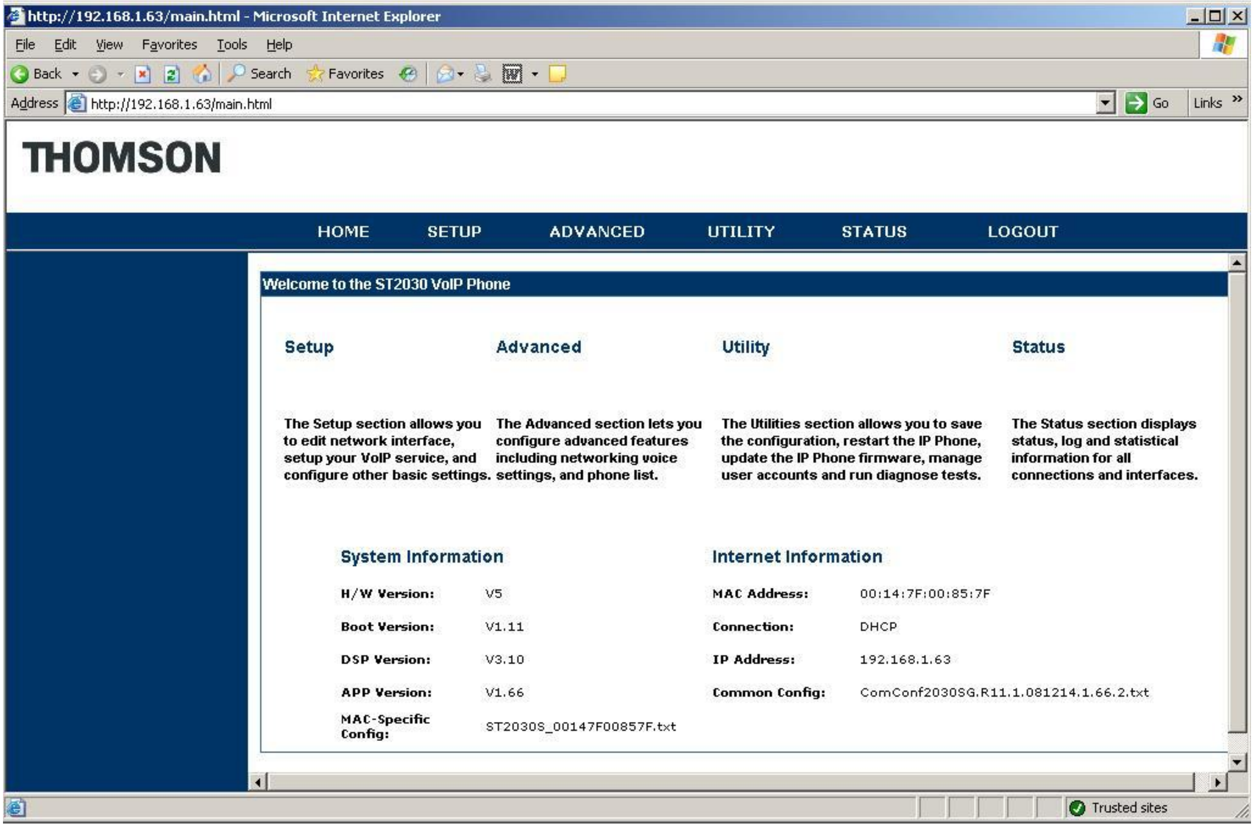
User Login

You have to logon with your username and password.

Username: ministrator

Password: ••••••

HOME PAGE



Welcome to the ST2030 VoIP Phone

Setup	Advanced	Utility	Status
The Setup section allows you to edit network interface, setup your VoIP service, and configure other basic settings.	The Advanced section lets you configure advanced features including networking voice settings, and phone list.	The Utilities section allows you to save the configuration, restart the IP Phone, update the IP Phone firmware, manage user accounts and run diagnose tests.	The Status section displays status, log and statistical information for all connections and interfaces.

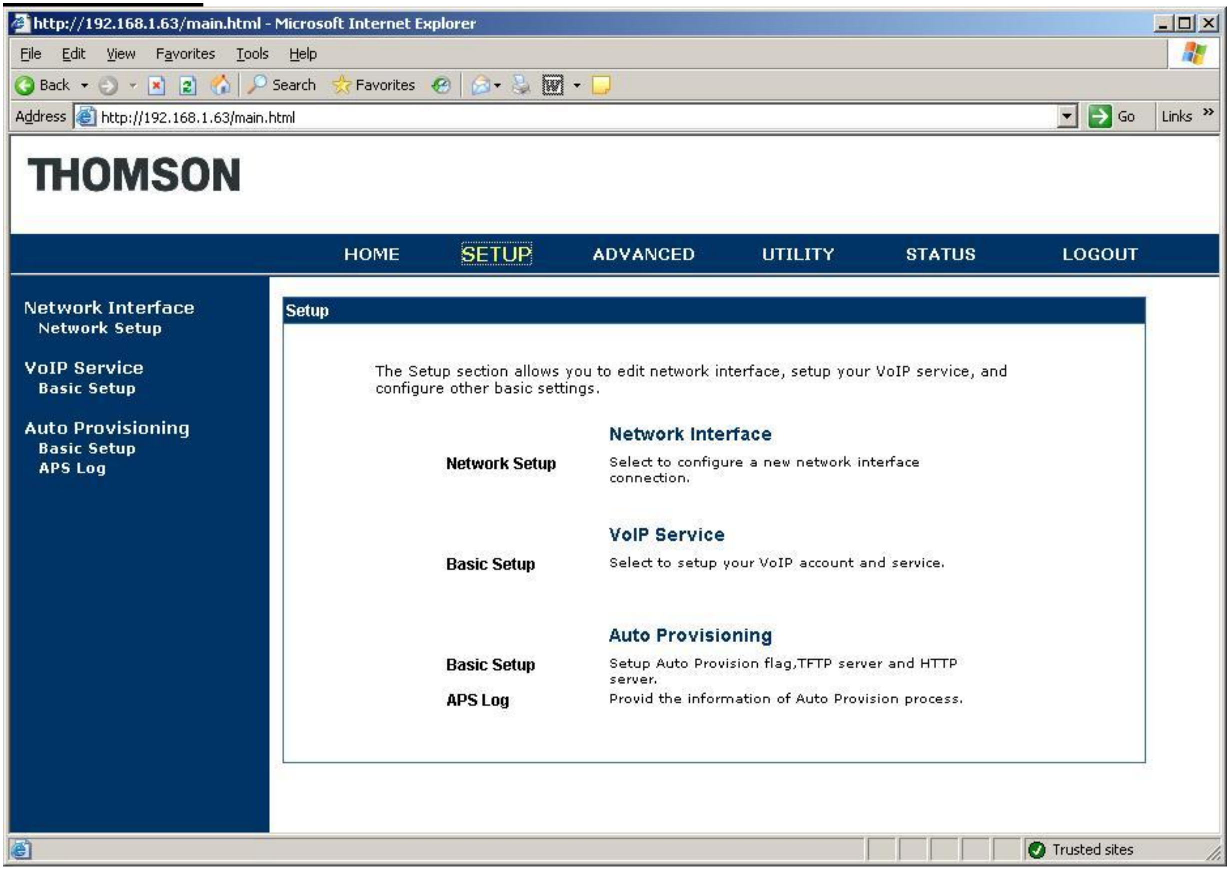
System Information

H/W Version:	V5
Boot Version:	V1.11
DSP Version:	V3.10
APP Version:	V1.66
MAC-Specific Config:	ST2030S_00147F00857F.txt

Internet Information

MAC Address:	00:14:7F:00:85:7F
Connection:	DHCP
IP Address:	192.168.1.63
Common Config:	ComConf2030SG.R11.1.081214.1.66.2.txt

SETUP PAGE

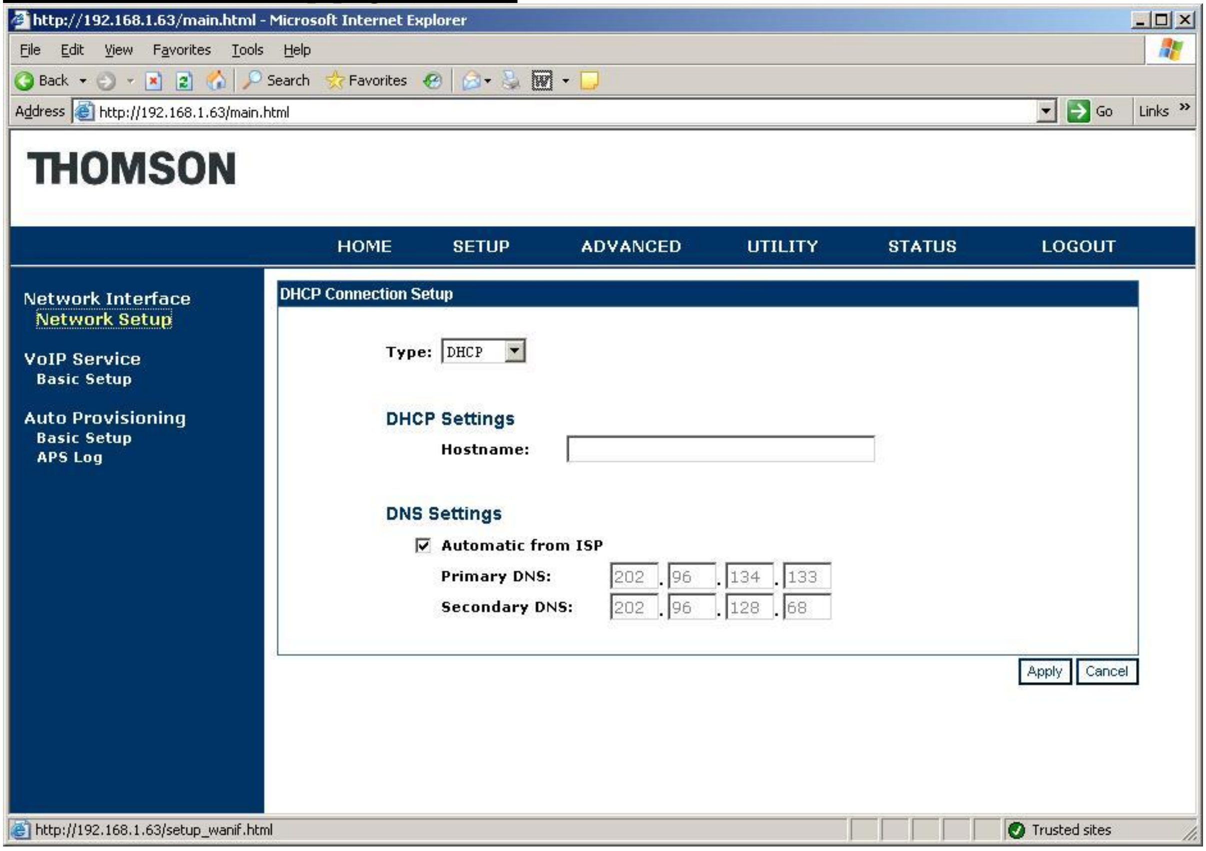


Setup

The Setup section allows you to edit network interface, setup your VoIP service, and configure other basic settings.

	Network Interface
Network Setup	Select to configure a new network interface connection.
	VoIP Service
Basic Setup	Select to setup your VoIP account and service.
	Auto Provisioning
Basic Setup	Setup Auto Provision flag,TFTP server and HTTP server.
APS Log	Provid the information of Auto Provision process.

SETUP: Network setup page (DHCP)



DHCP Connection Setup

Type: DHCP

DHCP Settings

Hostname:

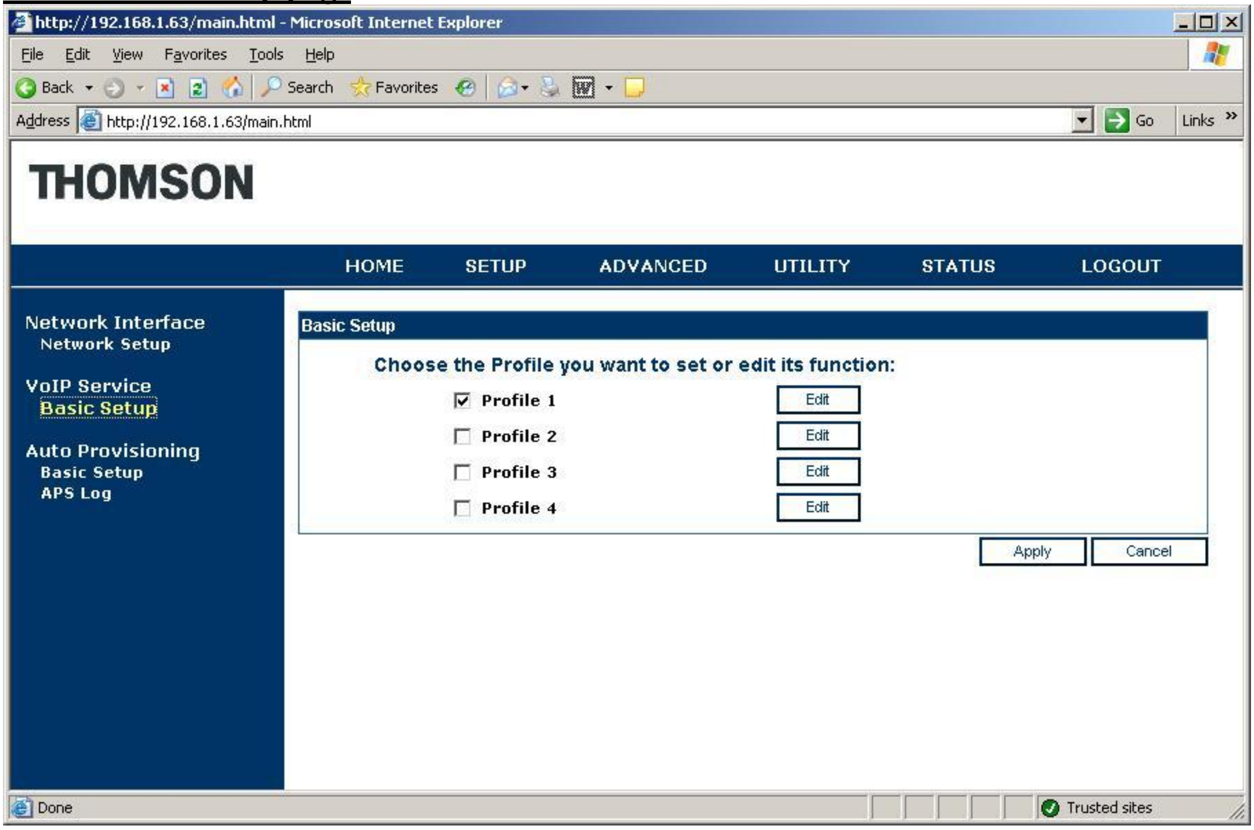
DNS Settings

☑ **Automatic from ISP**

Primary DNS: 202 . 96 . 134 . 133

Secondary DNS: 202 . 96 . 128 . 68

SETUP: Basic setup page

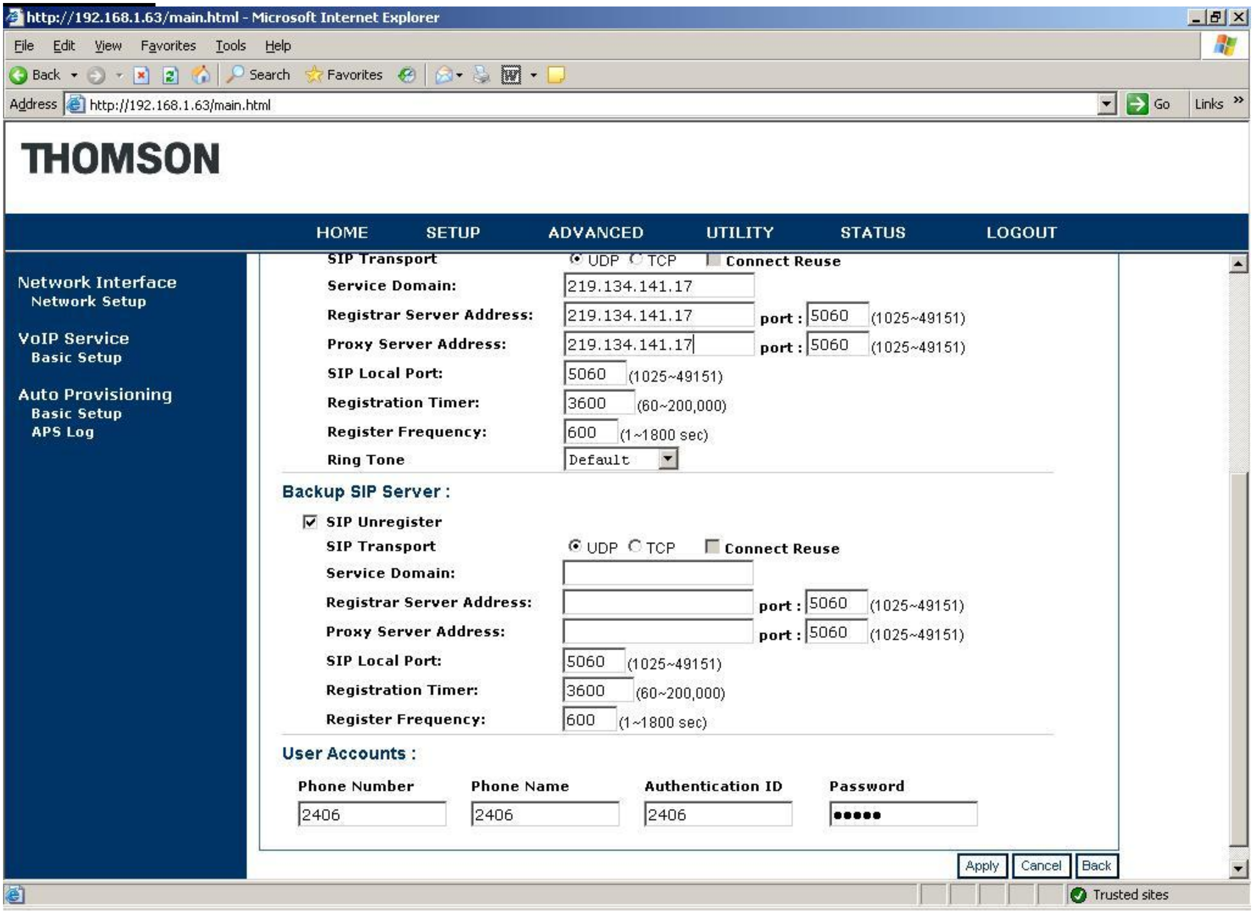


Basic Setup

Choose the Profile you want to set or edit its function:

☑ Profile 1
☐ Profile 2
☐ Profile 3
☐ Profile 4

Profile edit



SIP Transport ● UDP ○ TCP ☐ Connect Reuse
Service Domain: 219.134.141.17
Registrar Server Address: 219.134.141.17 port : 5060 (1025~49151)
Proxy Server Address: 219.134.141.17 port : 5060 (1025~49151)
SIP Local Port: 5060 (1025~49151)
Registration Timer: 3600 (60~200,000)
Register Frequency: 600 (1~1800 sec)
Ring Tone Default

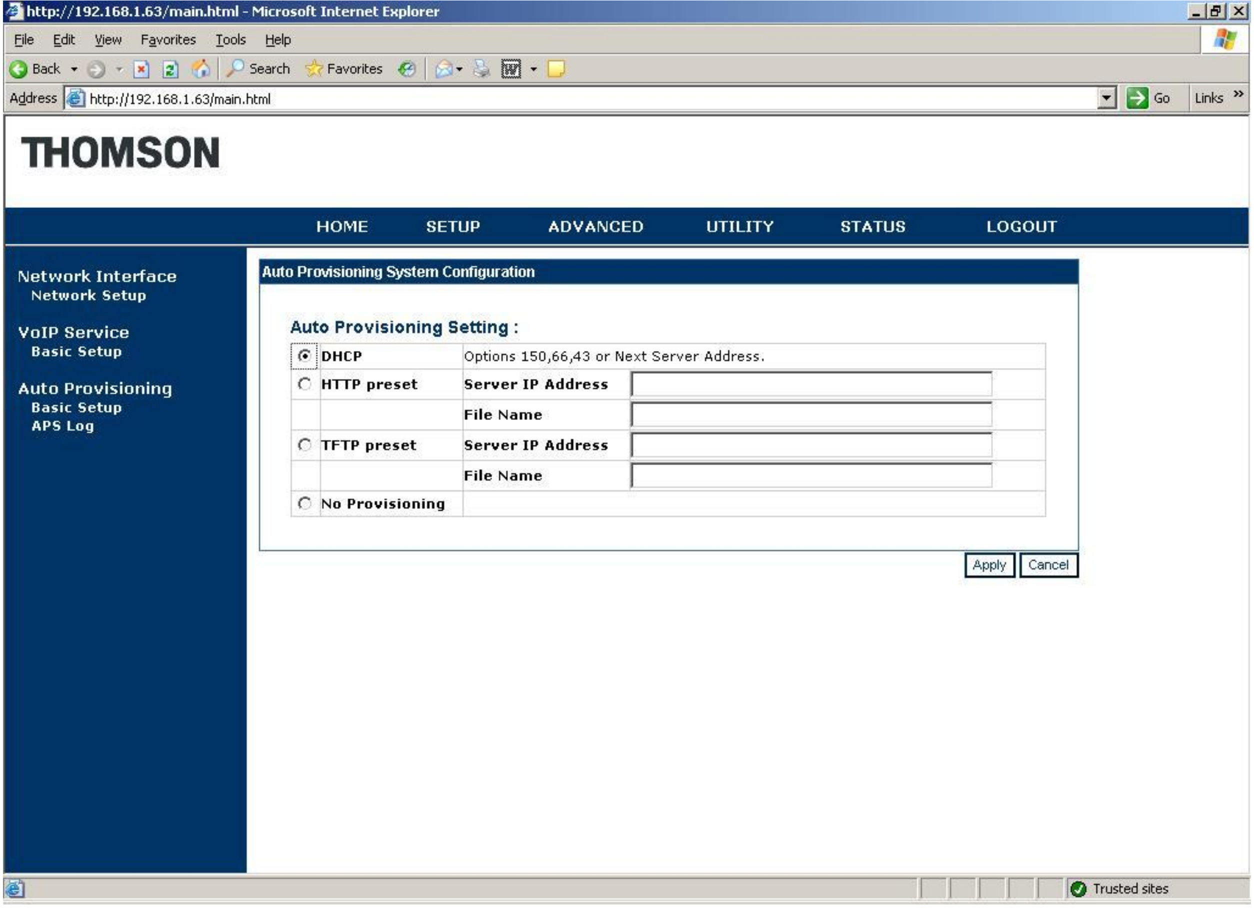
Backup SIP Server :

☑ SIP Unregister
SIP Transport ● UDP ○ TCP ☐ Connect Reuse
Service Domain:
Registrar Server Address: port : 5060 (1025~49151)
Proxy Server Address: port : 5060 (1025~49151)
SIP Local Port: 5060 (1025~49151)
Registration Timer: 3600 (60~200,000)
Register Frequency: 600 (1~1800 sec)

User Accounts :

Phone Number	Phone Name	Authentication ID	Password
2406	2406	2406	●●●●●

APS basic setup

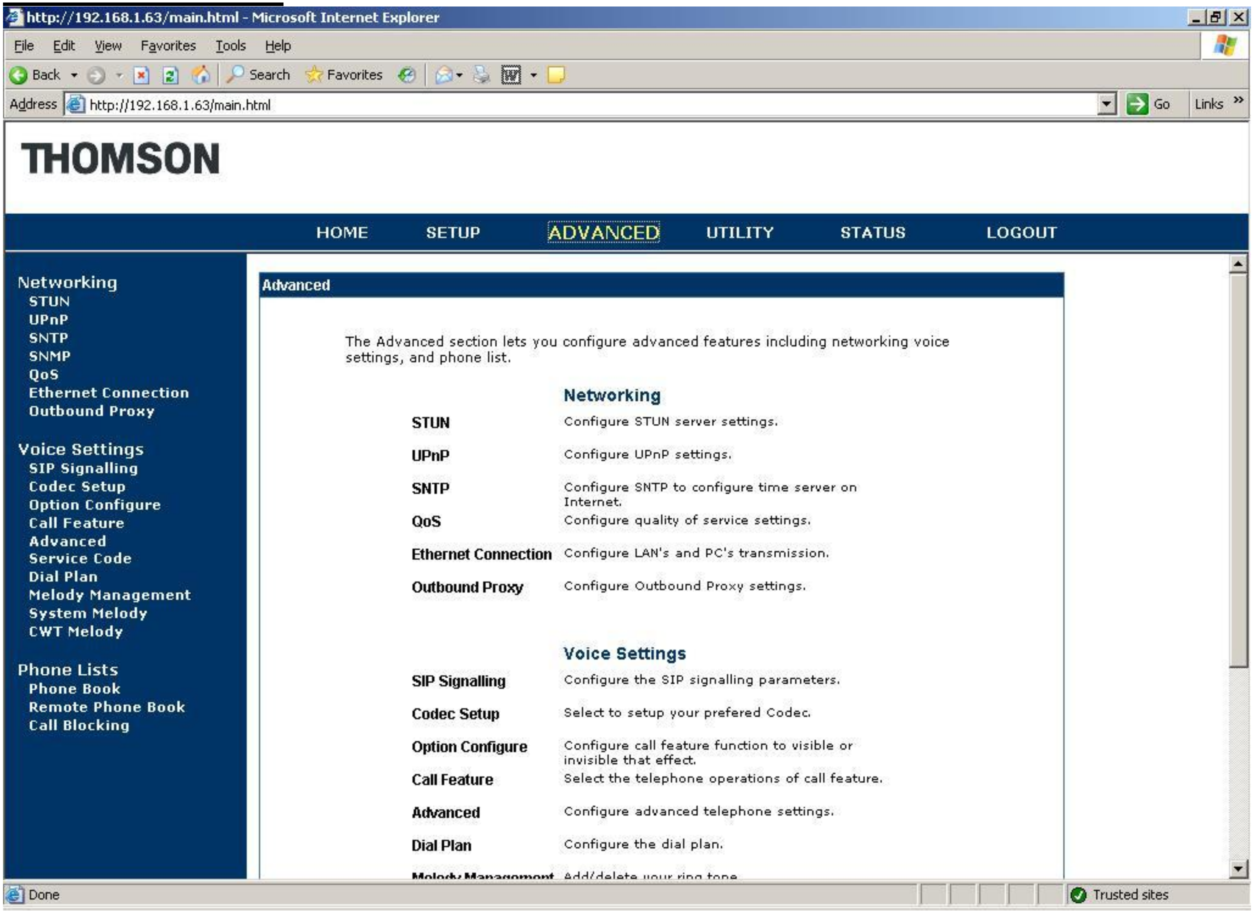


Auto Provisioning System Configuration

Auto Provisioning Setting :

●	DHCP	Options 150,66,43 or Next Server Address.	
○	HTTP preset	Server IP Address	
		File Name	
○	TFTP preset	Server IP Address	
		File Name	
○	No Provisioning		

ADVANCED PAGE

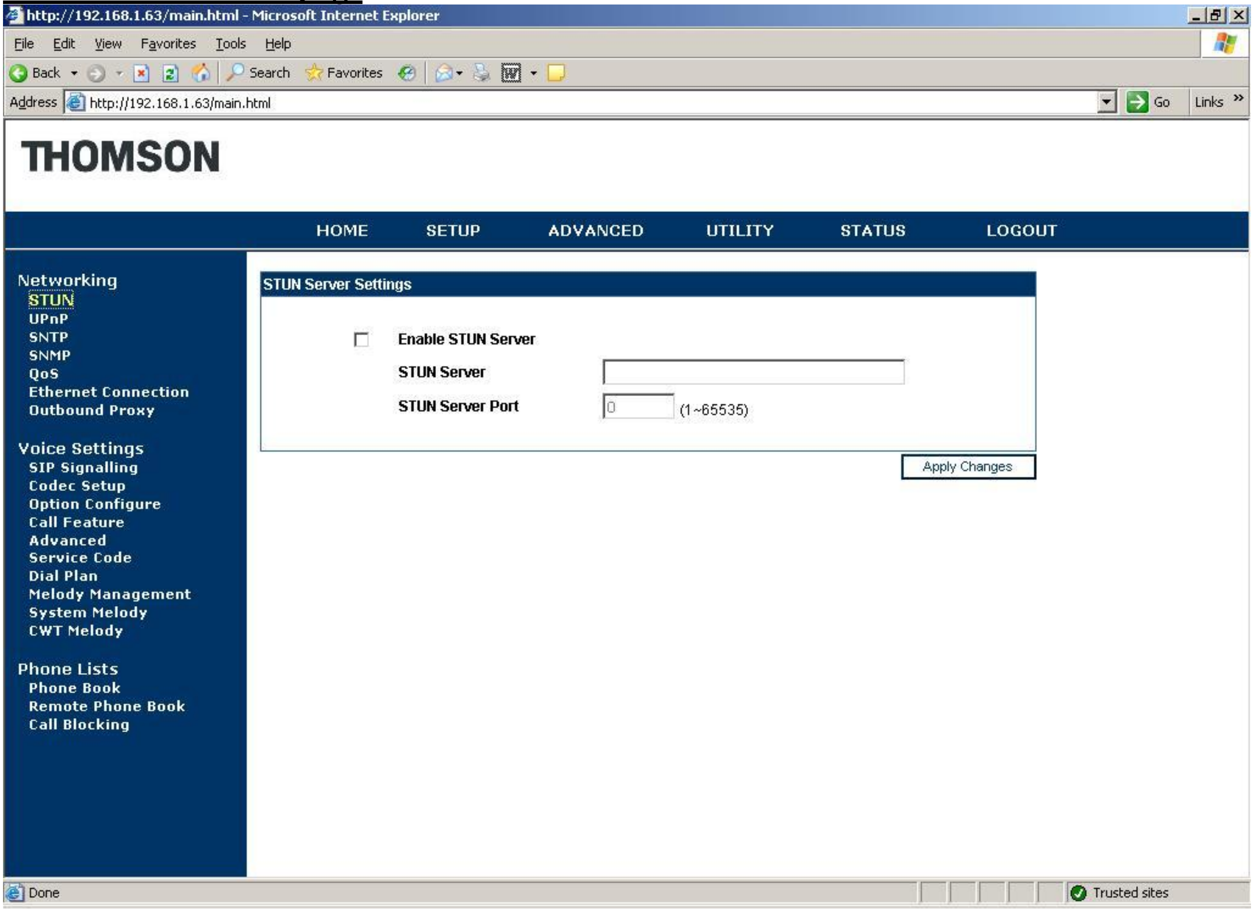


Advanced

The Advanced section lets you configure advanced features including networking voice settings, and phone list.

	Networking
STUN	Configure STUN server settings.
UPnP	Configure UPnP settings.
SNTP	Configure SNTP to configure time server on Internet.
QoS	Configure quality of service settings.
Ethernet Connection	Configure LAN's and PC's transmission.
Outbound Proxy	Configure Outbound Proxy settings.
	Voice Settings
SIP Signalling	Configure the SIP signalling parameters.
Codec Setup	Select to setup your prefered Codec.
Option Configure	Configure call feature function to visible or invisible that effect.
Call Feature	Select the telephone operations of call feature.
Advanced	Configure advanced telephone settings.
Dial Plan	Configure the dial plan.
Melody Management	Add/delete your ring tone

ADVANCED: STUN page



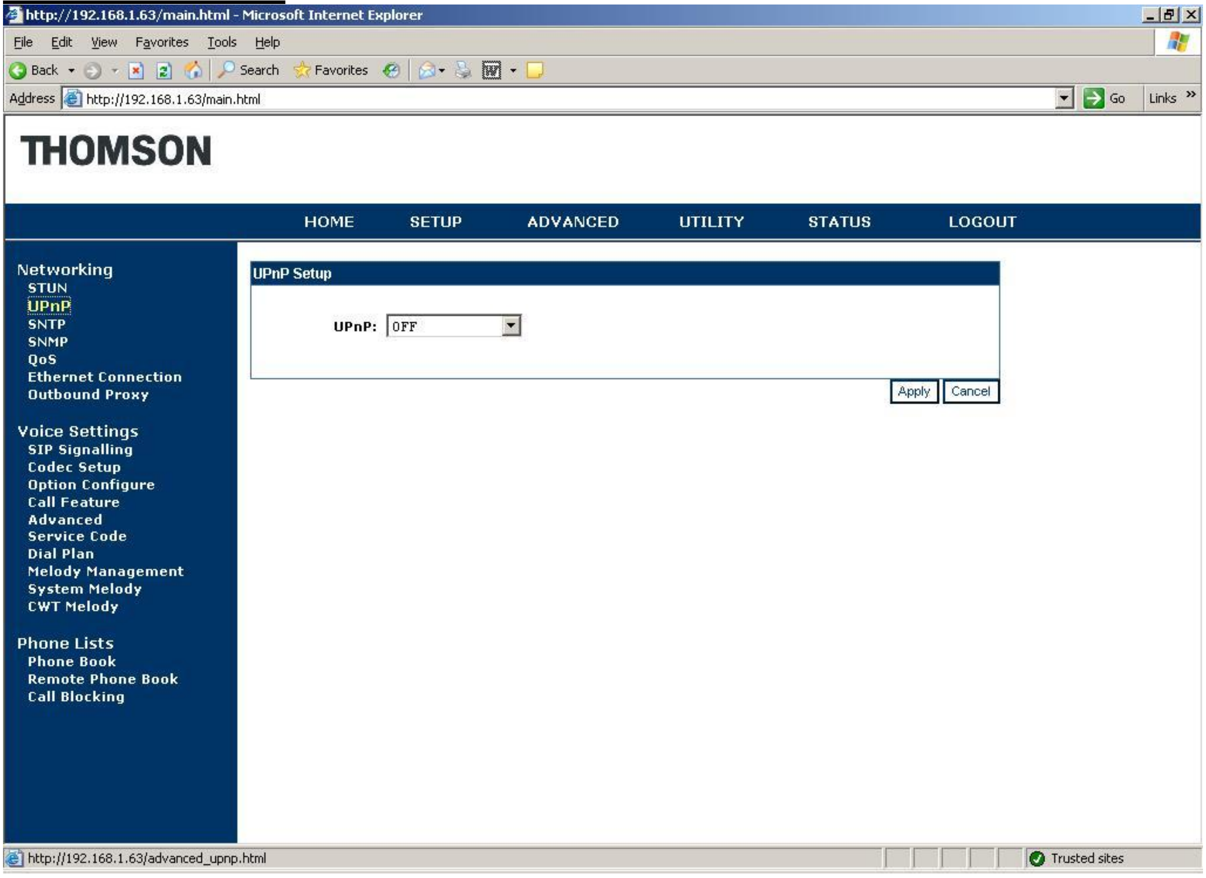
STUN Server Settings

☐ Enable STUN Server

STUN Server

STUN Server Port 0 (1~65535)

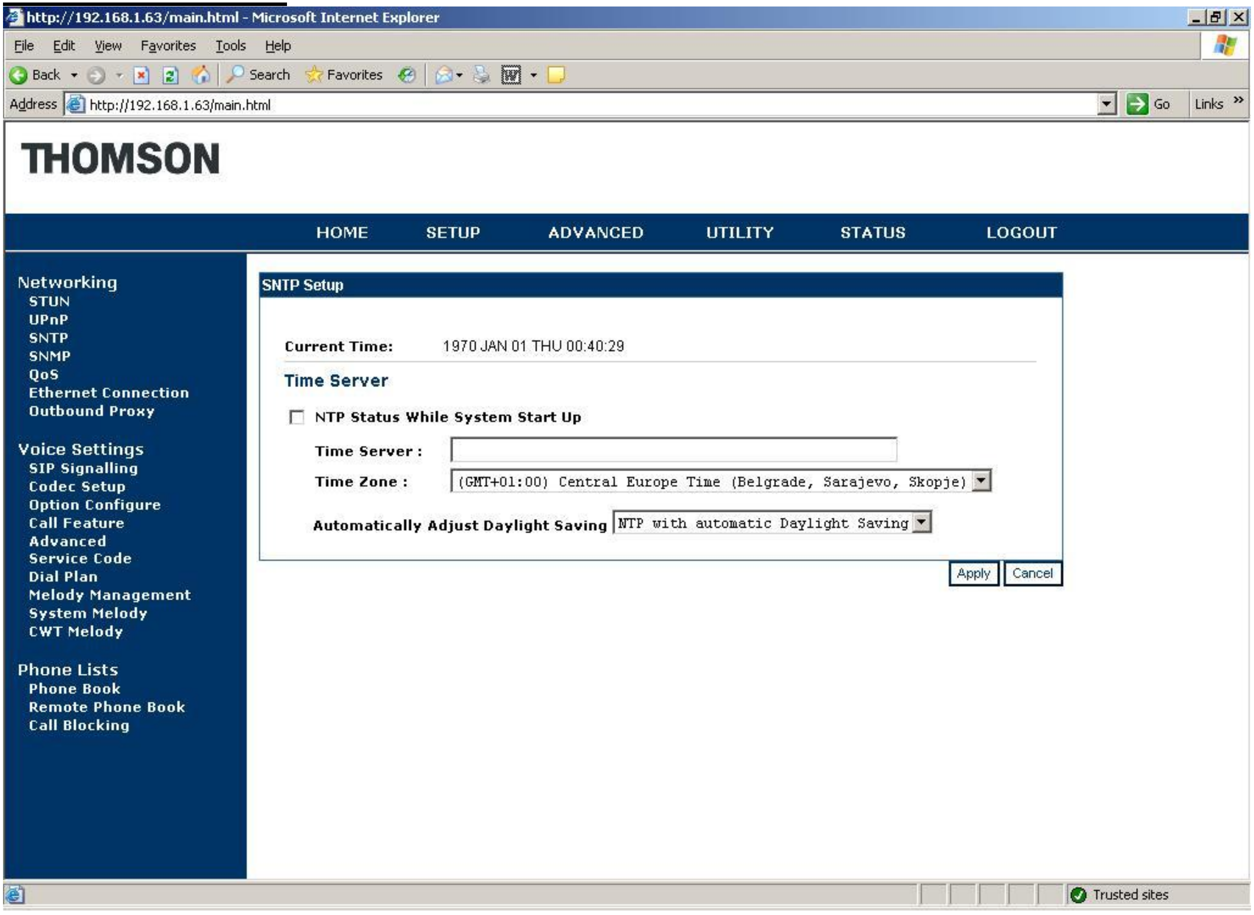
ADVANCED: UPnP



UPnP Setup

UPnP: OFF

ADVANCED: SNTP



SNTP Setup

Current Time: 1970 JAN 01 THU 00:40:29

Time Server

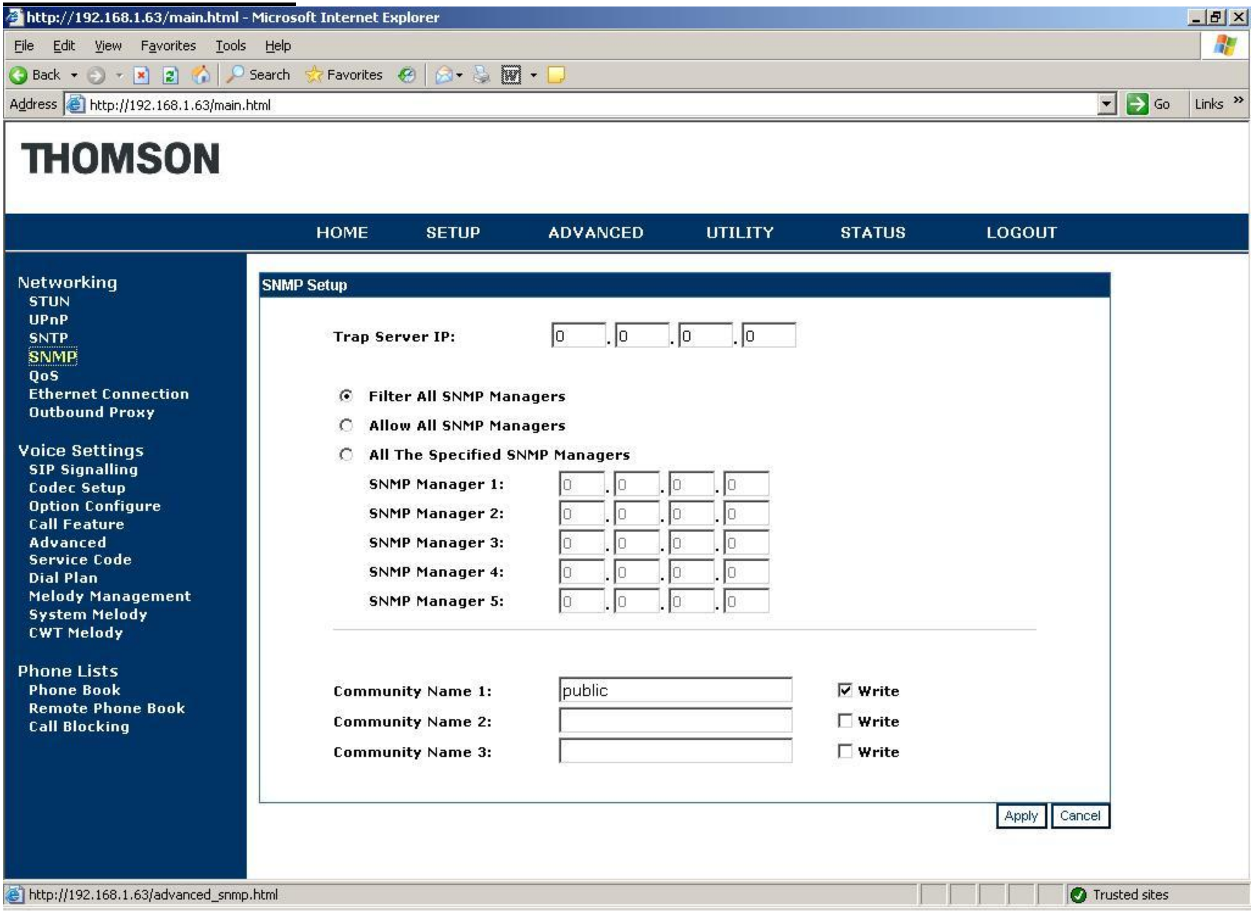
☐ **NTP Status While System Start Up**

Time Server :

Time Zone : (GMT+01:00) Central Europe Time (Belgrade, Sarajevo, Skopje)

Automatically Adjust Daylight Saving NTP with automatic Daylight Saving

ADVANCED: SNMP



SNMP Setup

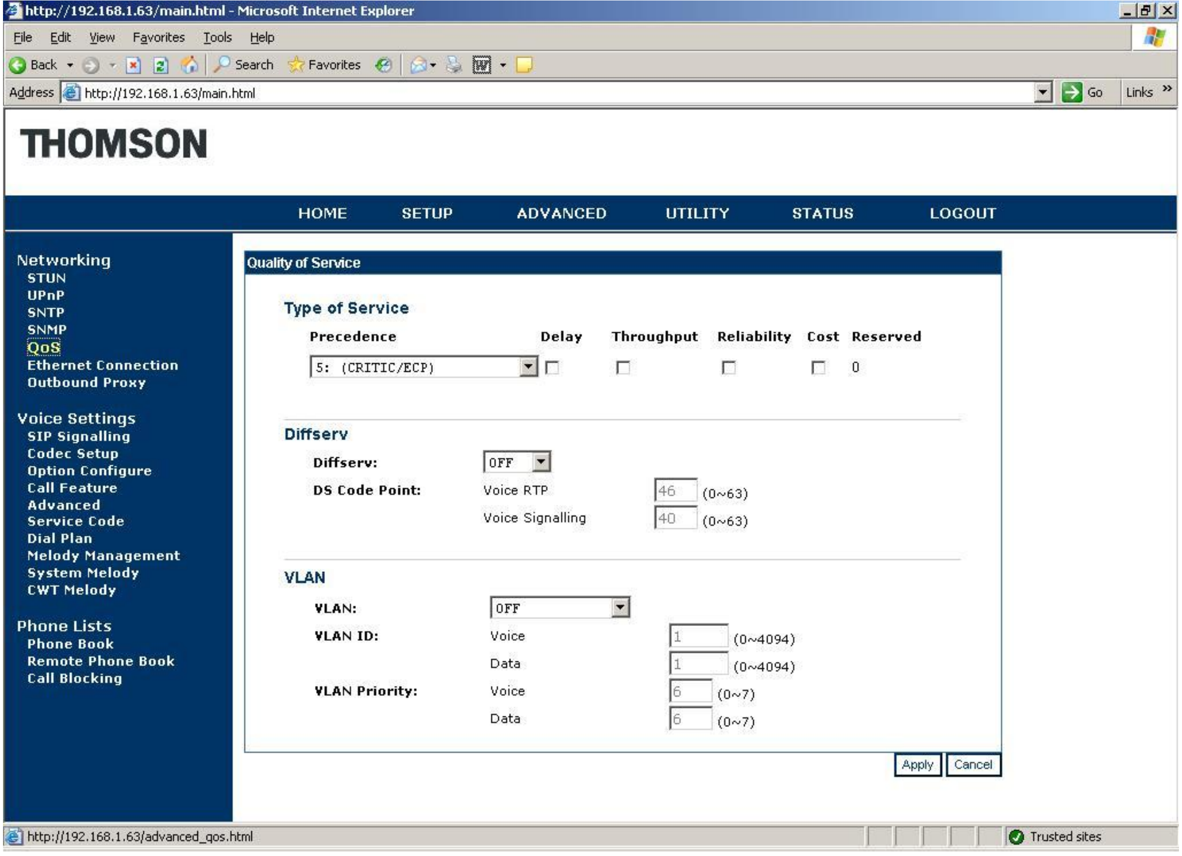
Trap Server IP: 0 . 0 . 0 . 0

● **Filter All SNMP Managers**
○ **Allow All SNMP Managers**
○ **All The Specified SNMP Managers**

SNMP Manager 1: 0 . 0 . 0 . 0
SNMP Manager 2: 0 . 0 . 0 . 0
SNMP Manager 3: 0 . 0 . 0 . 0
SNMP Manager 4: 0 . 0 . 0 . 0
SNMP Manager 5: 0 . 0 . 0 . 0

Community Name 1: public ☑ **Write**
Community Name 2: ☐ **Write**
Community Name 3: ☐ **Write**

ADVANCED: QoS



Quality of Service

Type of Service

Precedence	Delay	Throughput	Reliability	Cost	Reserved
5: (CRITIC/ECP)	☐	☐	☐	☐	0

Diffserv

Diffserv: OFF

DS Code Point: Voice RTP 46 (0~63)

Voice Signalling 40 (0~63)

VLAN

VLAN: OFF

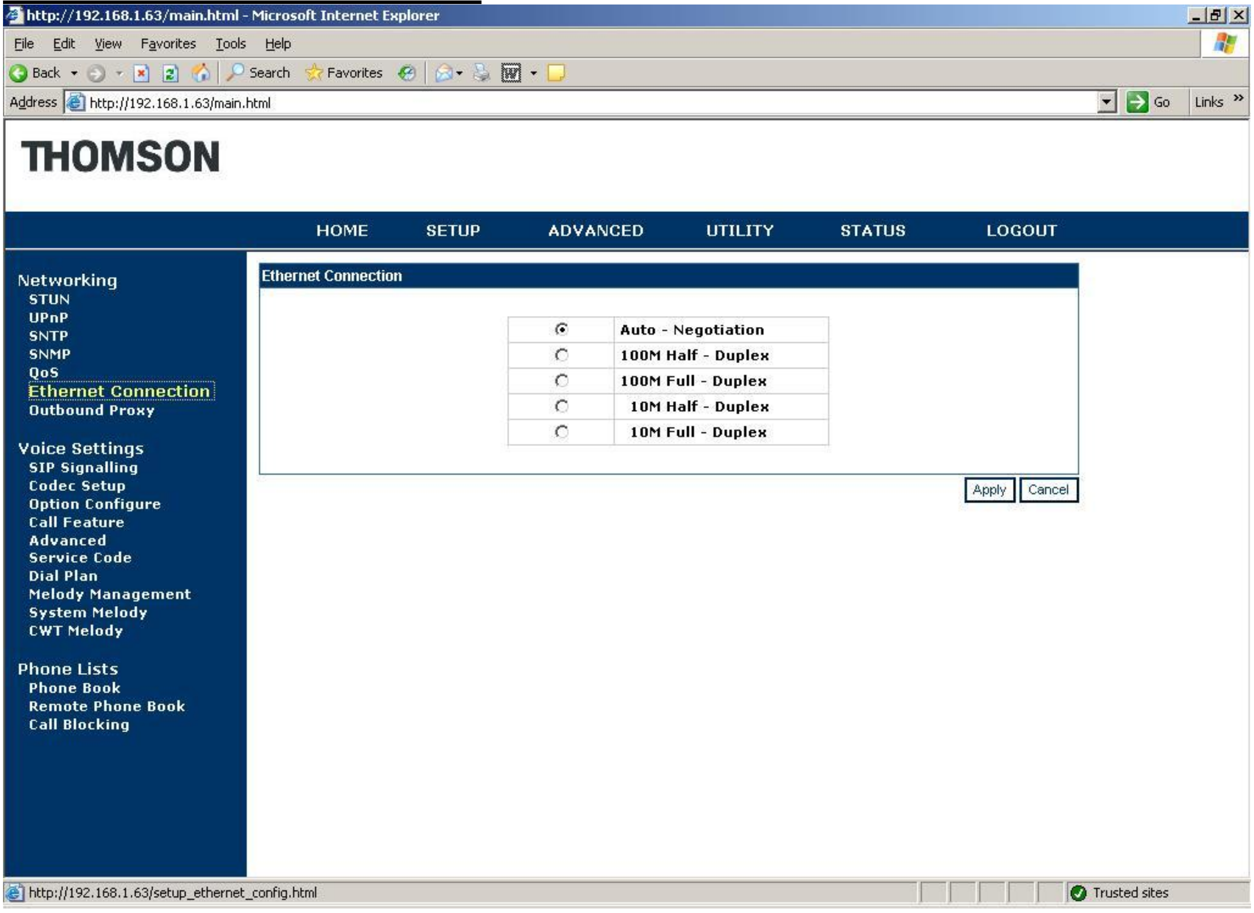
VLAN ID: Voice 1 (0~4094)

Data 1 (0~4094)

VLAN Priority: Voice 6 (0~7)

Data 6 (0~7)

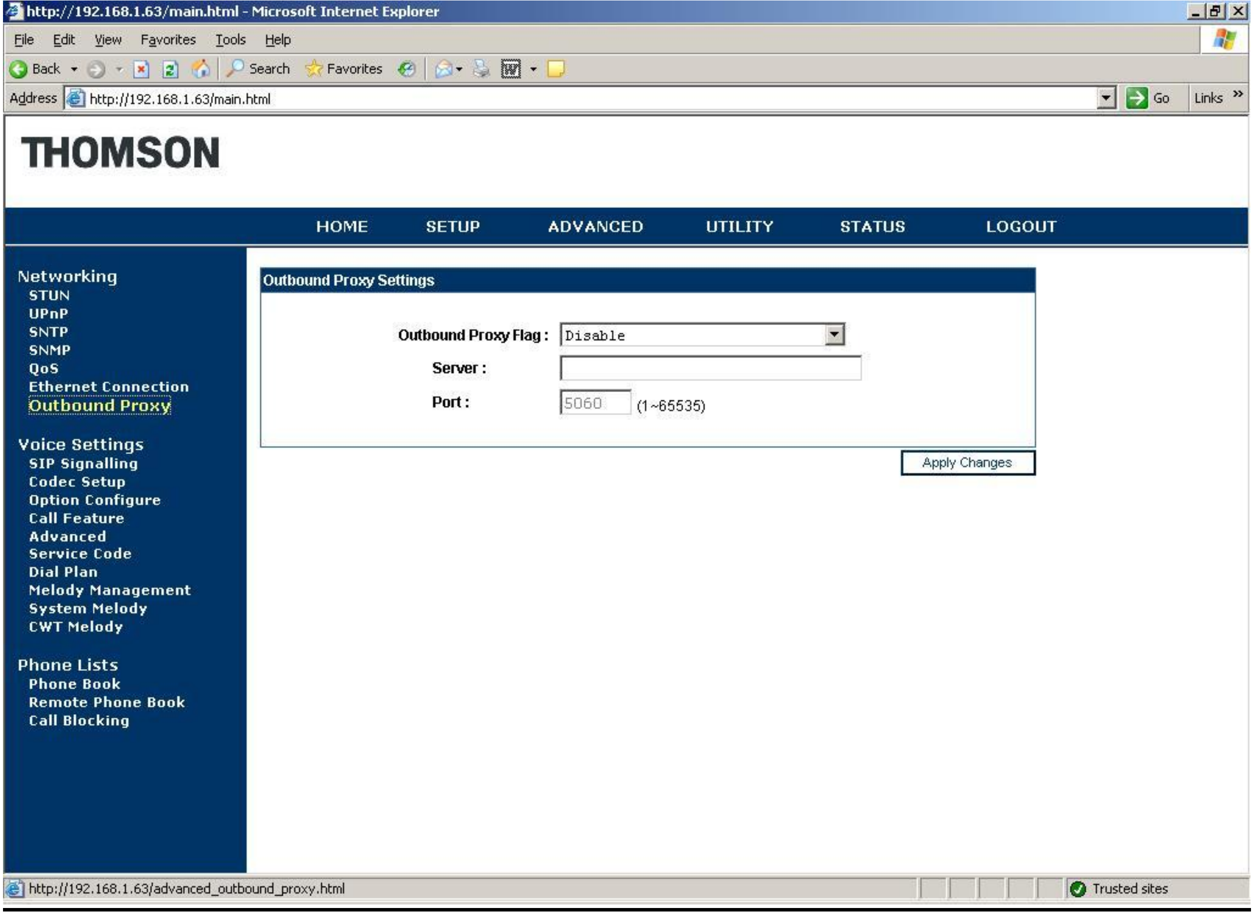
ADVANCED: Ethernet connection



Ethernet Connection

- ● **Auto - Negotiation**
- ○ **100M Half - Duplex**
- ○ **100M Full - Duplex**
- ○ **10M Half - Duplex**
- ○ **10M Full - Duplex**

ADVANCED: Outband proxy



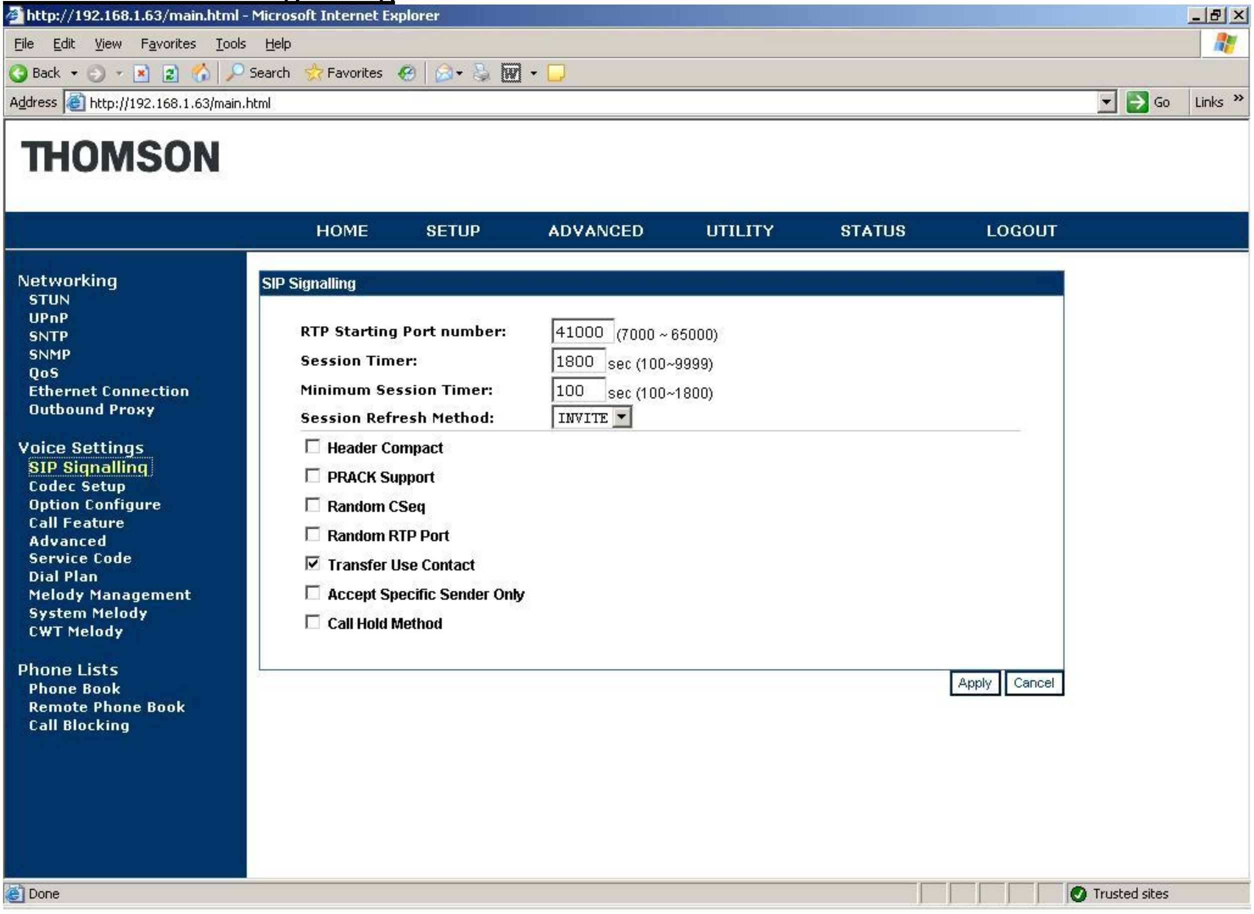
Outbound Proxy Settings

Outbound Proxy Flag : Disable

Server :

Port : 5060 (1~65535)

ADVANCED: SIP signalling



SIP Signalling

RTP Starting Port number: 41000 (7000 ~ 65000)

Session Timer: 1800 sec (100~9999)

Minimum Session Timer: 100 sec (100~1800)

Session Refresh Method: INVITE

☐ Header Compact

☐ PRACK Support

☐ Random CSeq

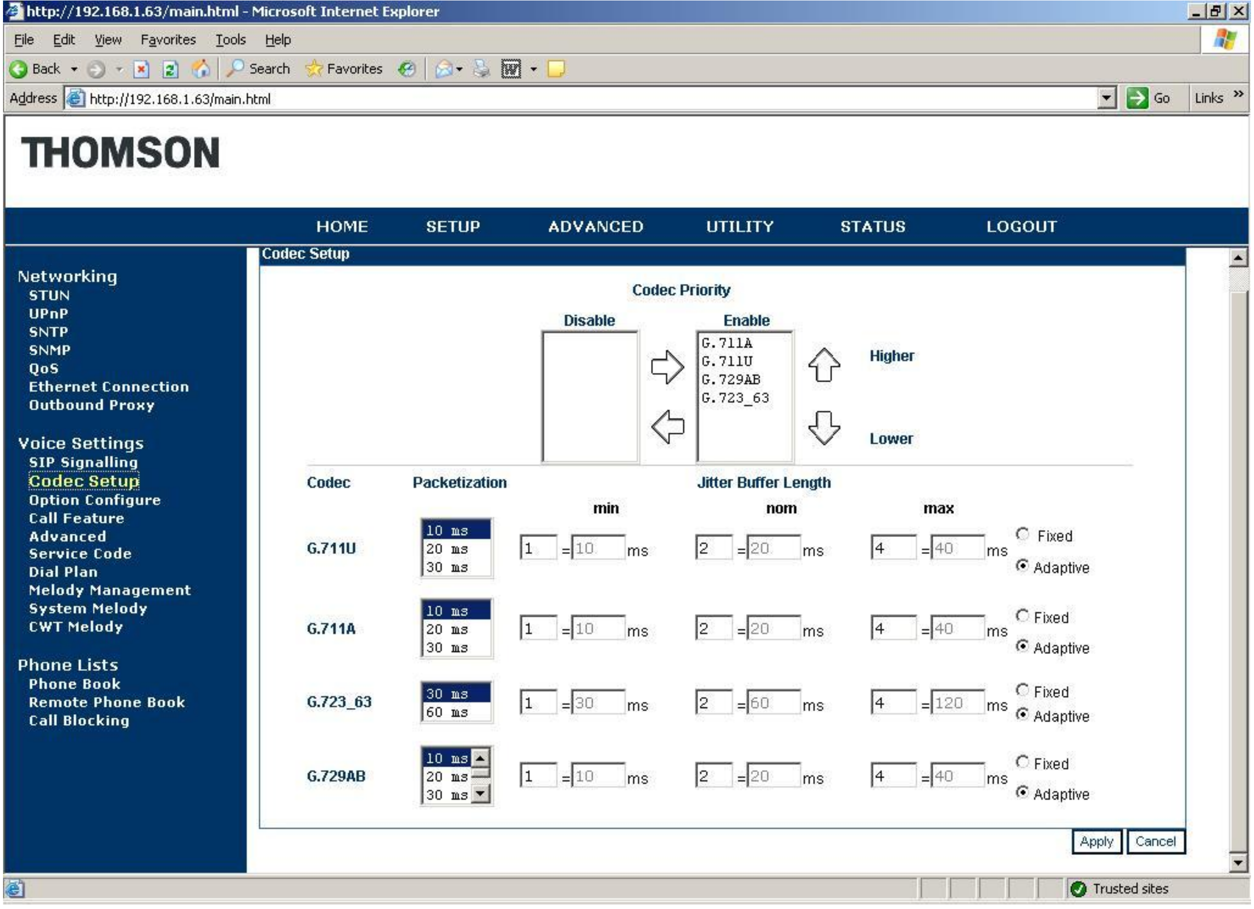
☐ Random RTP Port

☑ Transfer Use Contact

☐ Accept Specific Sender Only

☐ Call Hold Method

ADVANCED: Codec setup



Codec Setup

Codec Priority

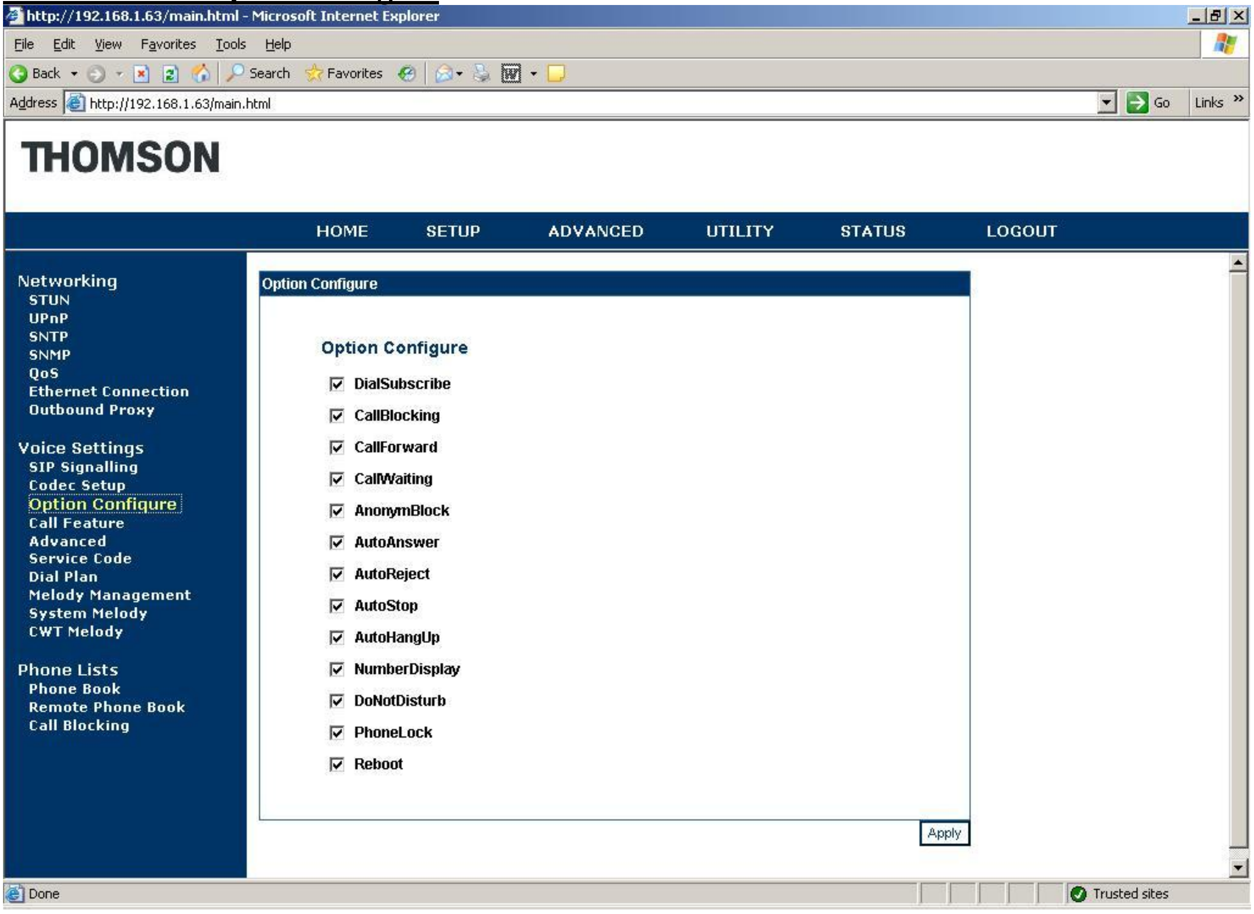
Disable:

Enable: G.711A, G.711U, G.729AB, G.723_63

Higher / Lower

Codec	Packetization	Jitter Buffer Length min	nom	max	
G.711U	10 ms 20 ms 30 ms	1 = 10 ms	2 = 20 ms	4 = 40 ms	○ Fixed ● Adaptive
G.711A	10 ms 20 ms 30 ms	1 = 10 ms	2 = 20 ms	4 = 40 ms	○ Fixed ● Adaptive
G.723_63	30 ms 60 ms	1 = 30 ms	2 = 60 ms	4 = 120 ms	○ Fixed ● Adaptive
G.729AB	10 ms 20 ms 30 ms	1 = 10 ms	2 = 20 ms	4 = 40 ms	○ Fixed ● Adaptive

ADVANCED: Option configure

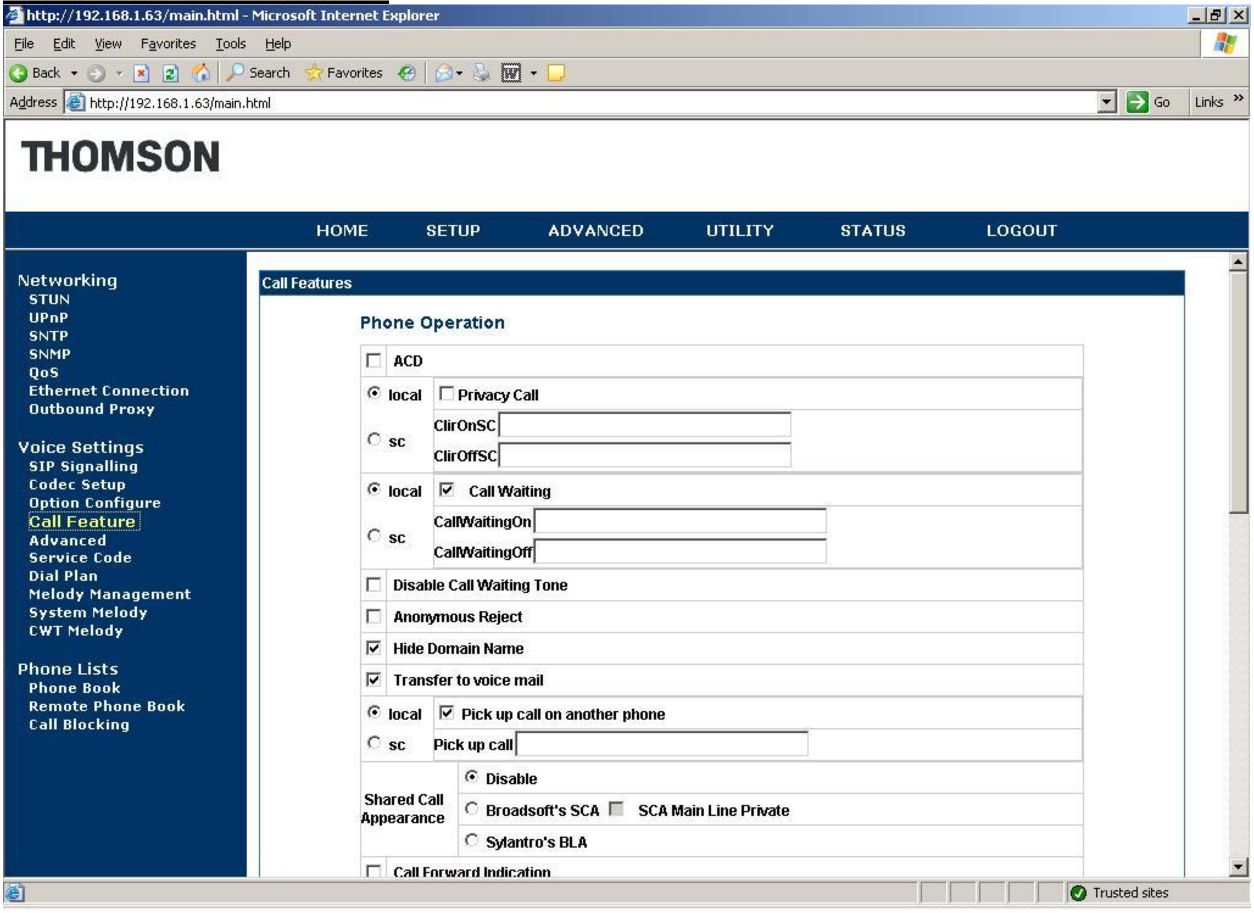


Option Configure

Option Configure

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

ADVANCED: Call Features



Call Features

Phone Operation

☐ ACD

● local ☐ Privacy Call

○ sc ClirOnSC ClirOffSC

● local ☑ Call Waiting

○ sc CallWaitingOn CallWaitingOff

☐ Disable Call Waiting Tone

☐ Anonymous Reject

☑ Hide Domain Name

☑ Transfer to voice mail

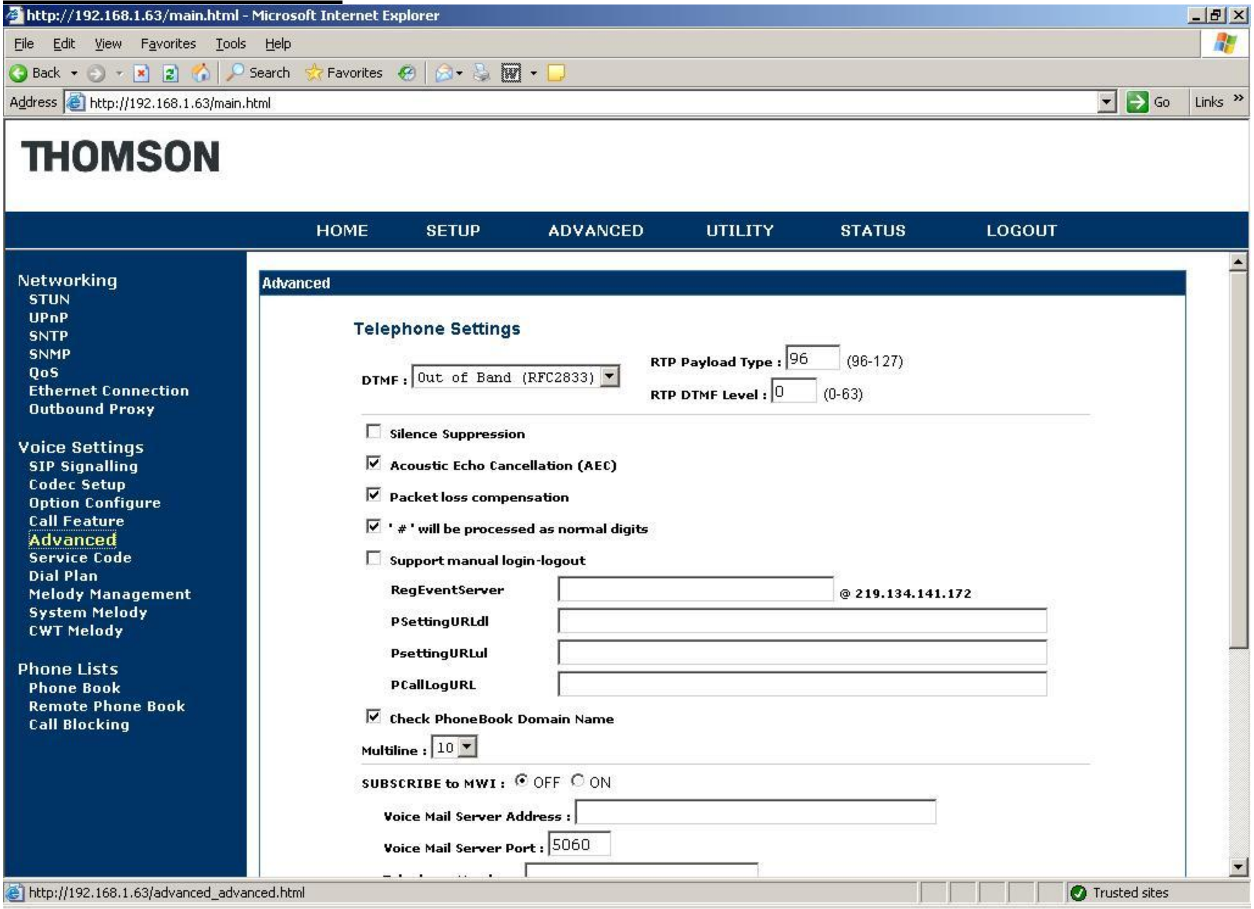
● local ☑ Pick up call on another phone

○ sc Pick up call

Shared Call Appearance
● Disable
○ Broadsoft's SCA ☐ SCA Main Line Private
○ Sylantro's BLA

☐ Call Forward Indication

ADVANCED: Advanced



Advanced

Telephone Settings

DTMF : Out of Band (RFC2833)
RTP Payload Type : 96 (96-127)
RTP DTMF Level : 0 (0-63)

☐ Silence Suppression

☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ ' # ' will be processed as normal digits

☐ Support manual login-logout

RegEventServer @ 219.134.141.172
PSettingURLdl
PsettingURLul
PCallLogURL

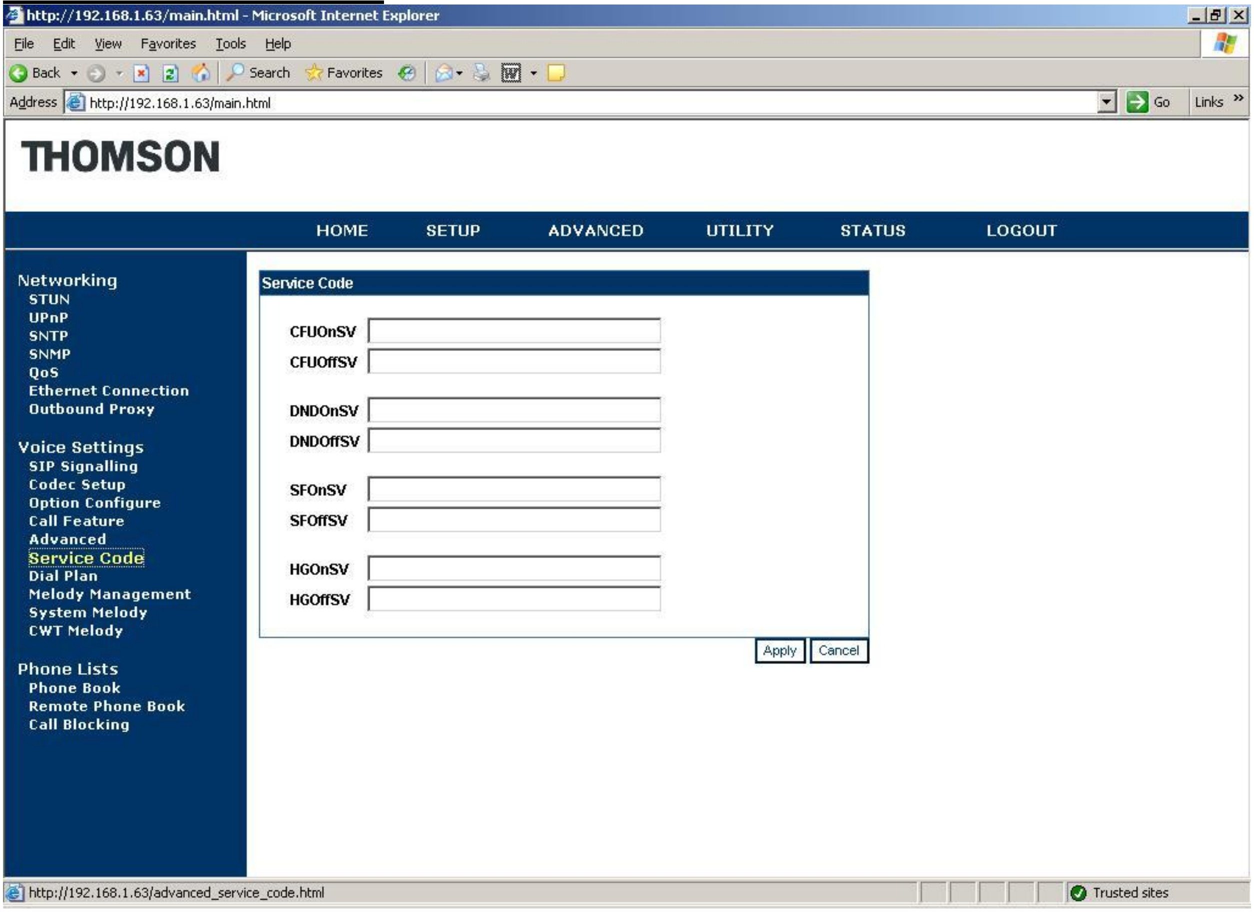
☑ Check PhoneBook Domain Name

Multiline : 10

SUBSCRIBE to MWI : ● OFF ○ ON

Voice Mail Server Address :
Voice Mail Server Port : 5060

ADVANCED: Service Code



Service Code

CFUOnSV

CFUOffSV

DNDOnSV

DNDOffSV

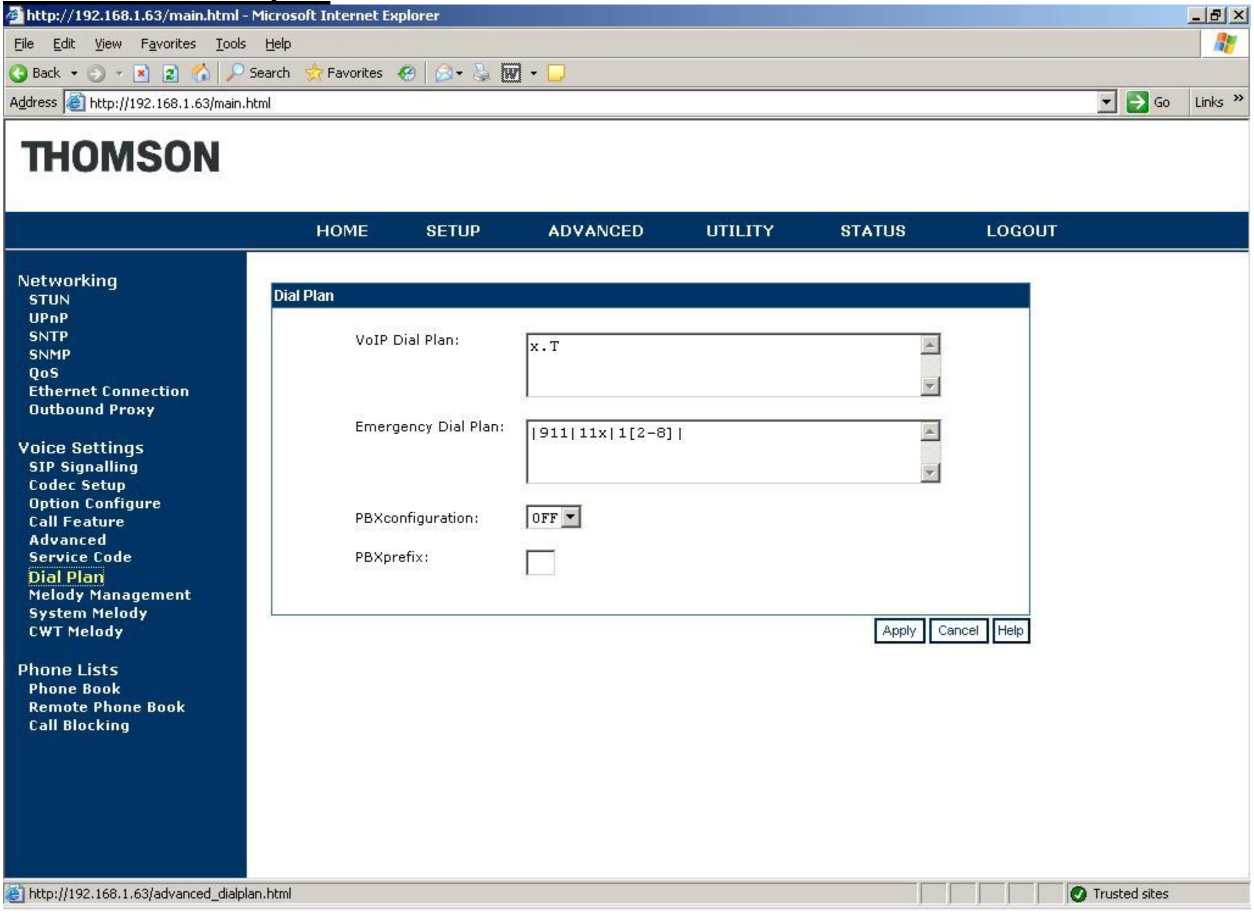
SFOnSV

SFOffSV

HGOnSV

HGOffSV

ADVANCED: Dial plan



Dial Plan

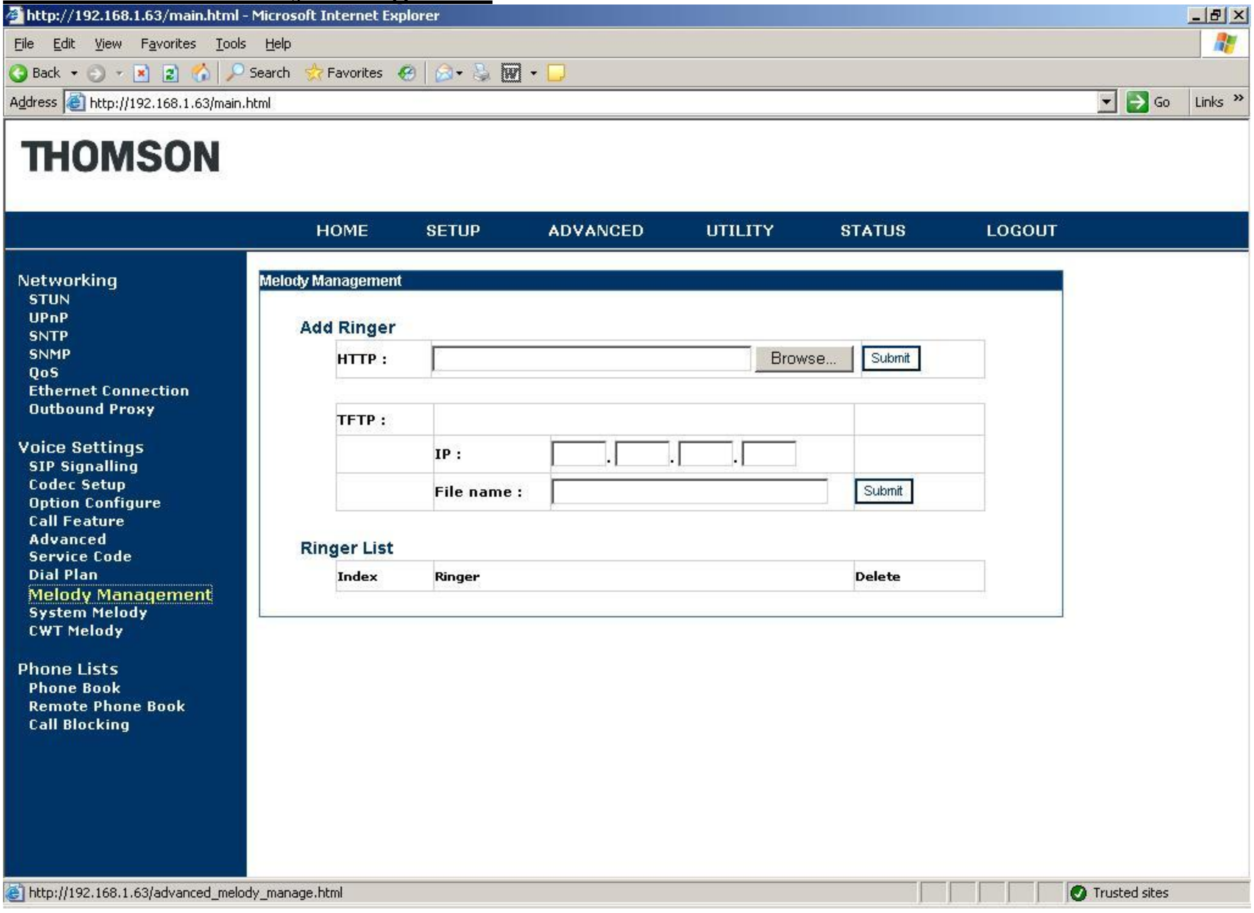
VoIP Dial Plan: x.T

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

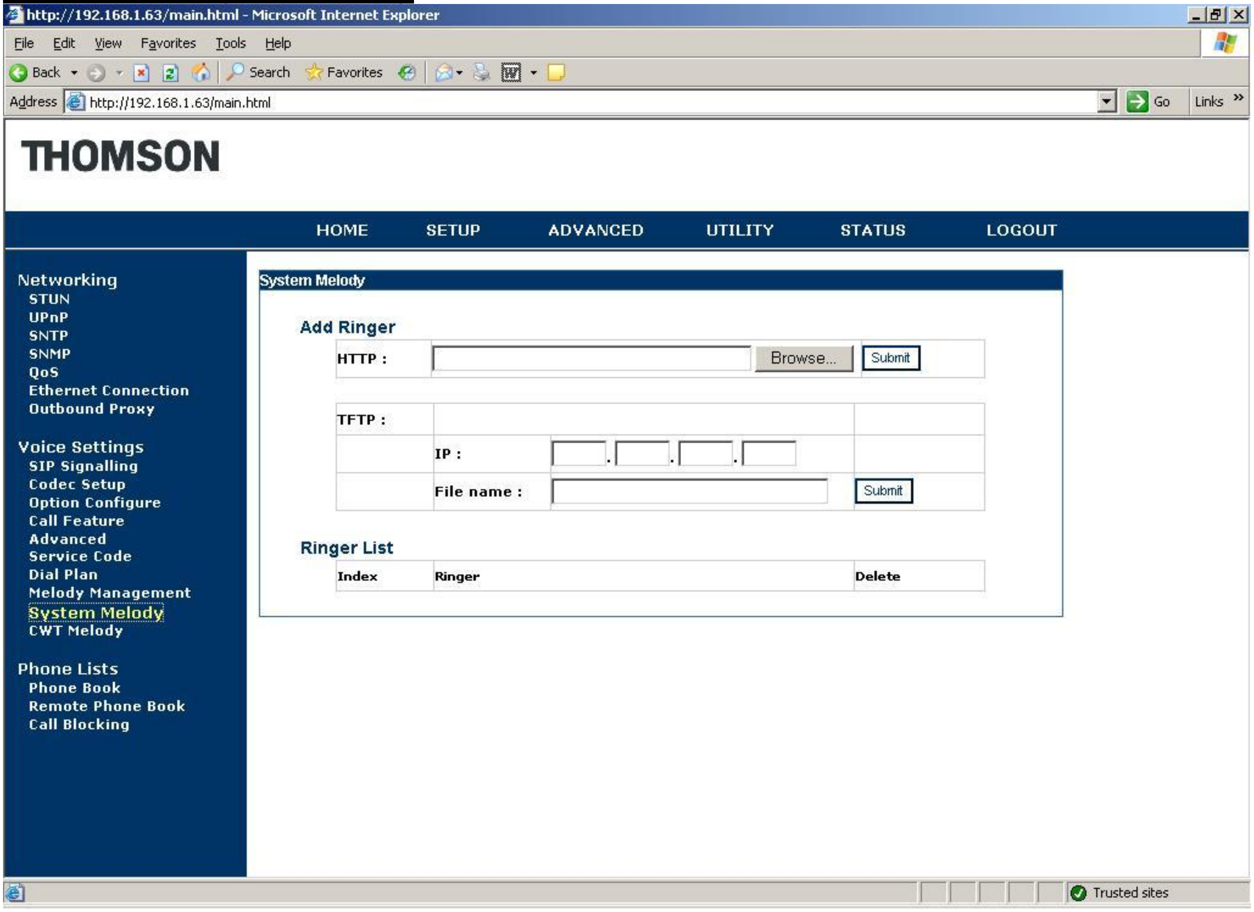
PBXconfiguration: OFF

PBXprefix:

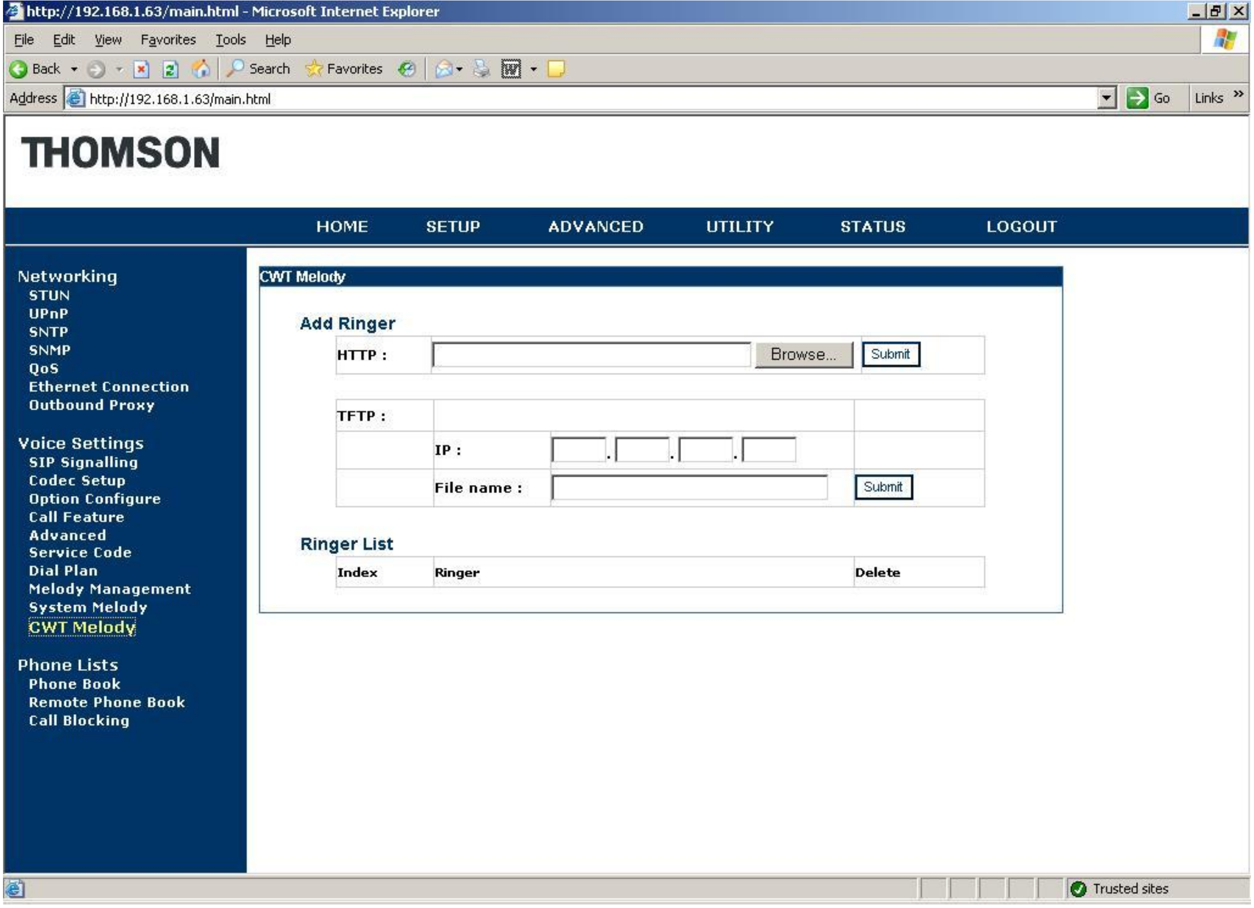
ADVANCED: Melody management



ADVANCED: System Melody



ADVANCED: CWT Melody



CWT Melody

Add Ringer

HTTP :

TFTP :

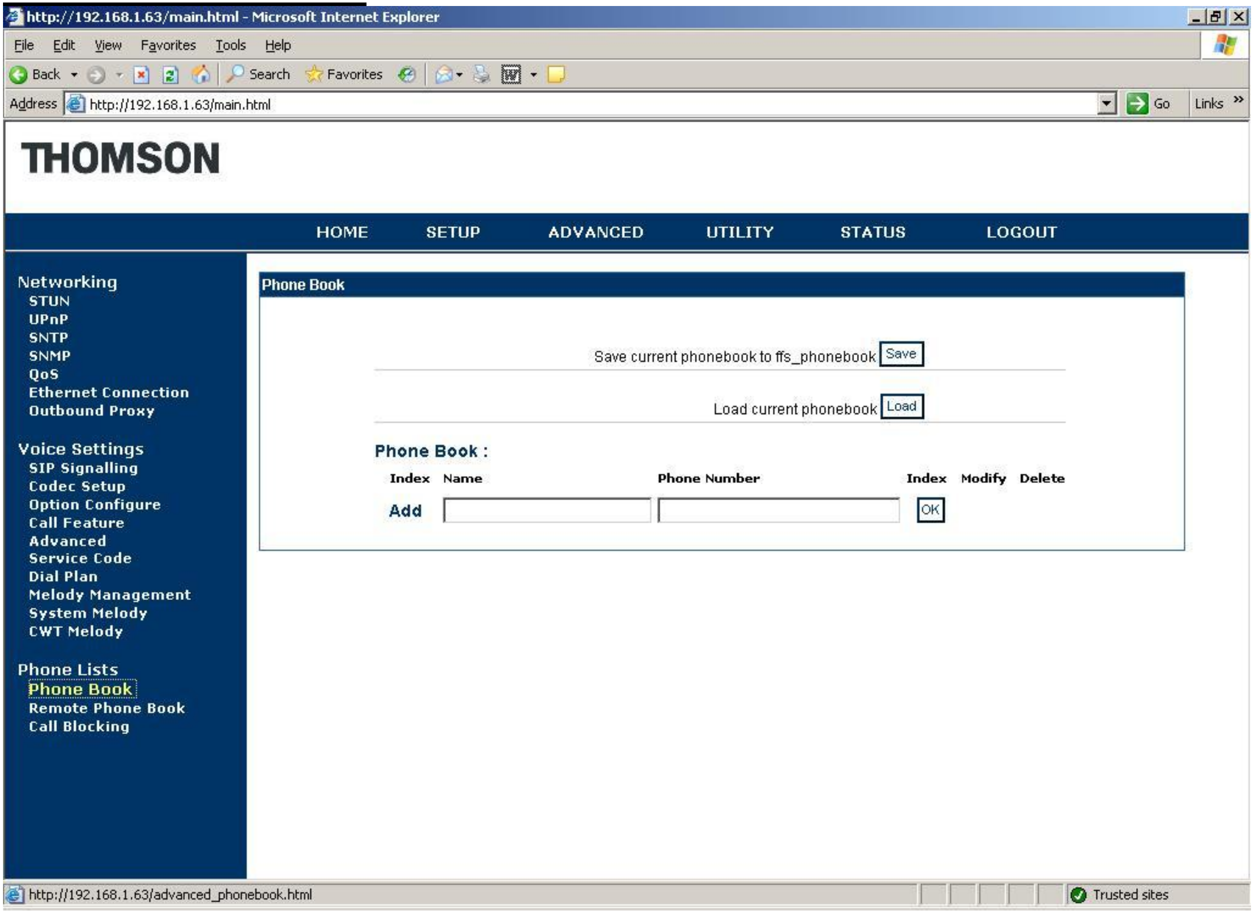
IP : . . .

File name :

Ringer List

Index	Ringer	Delete

ADVANCED: Phone book



Phone Book

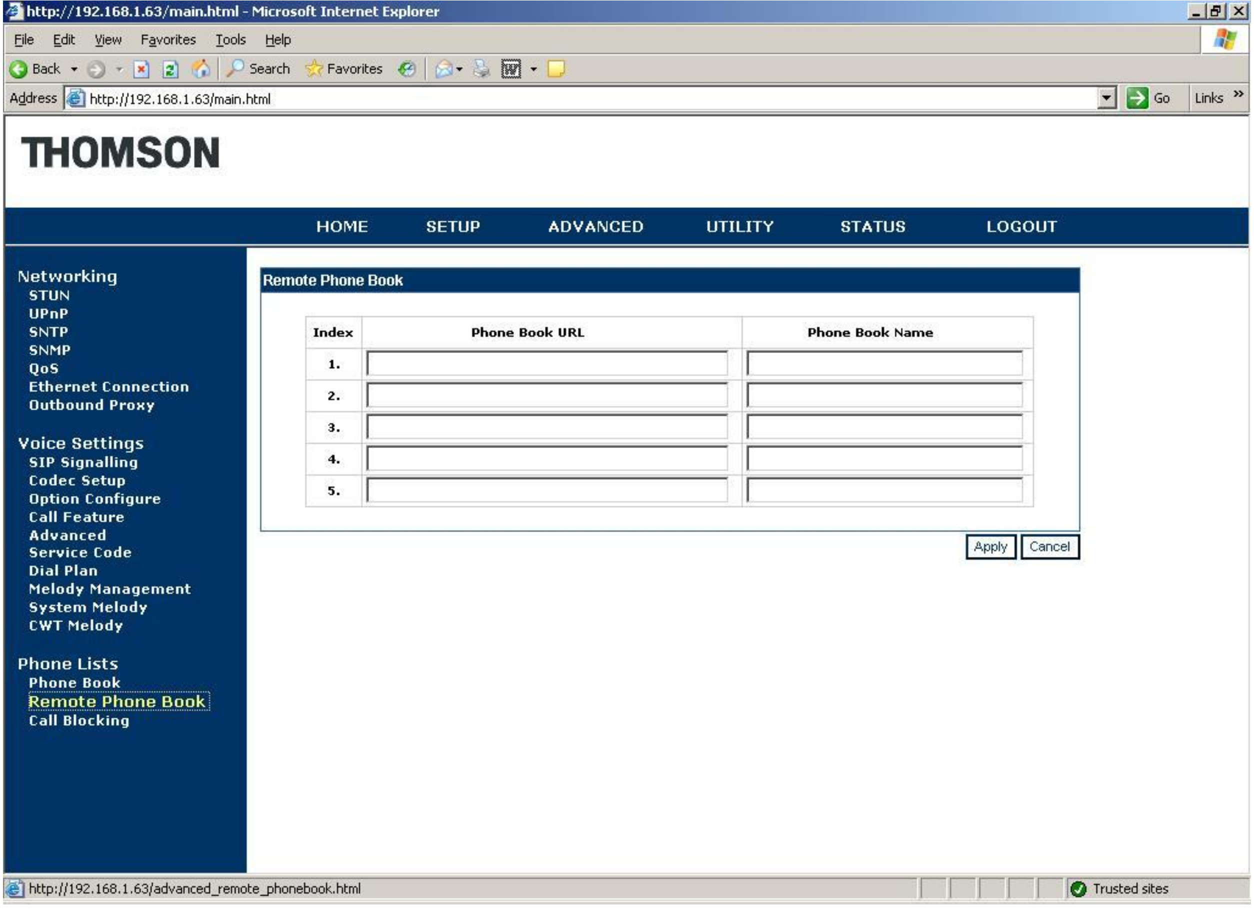
Save current phonebook to ffs_phonebook

Load current phonebook

Phone Book :

Index	Name	Phone Number	Index	Modify	Delete
Add					

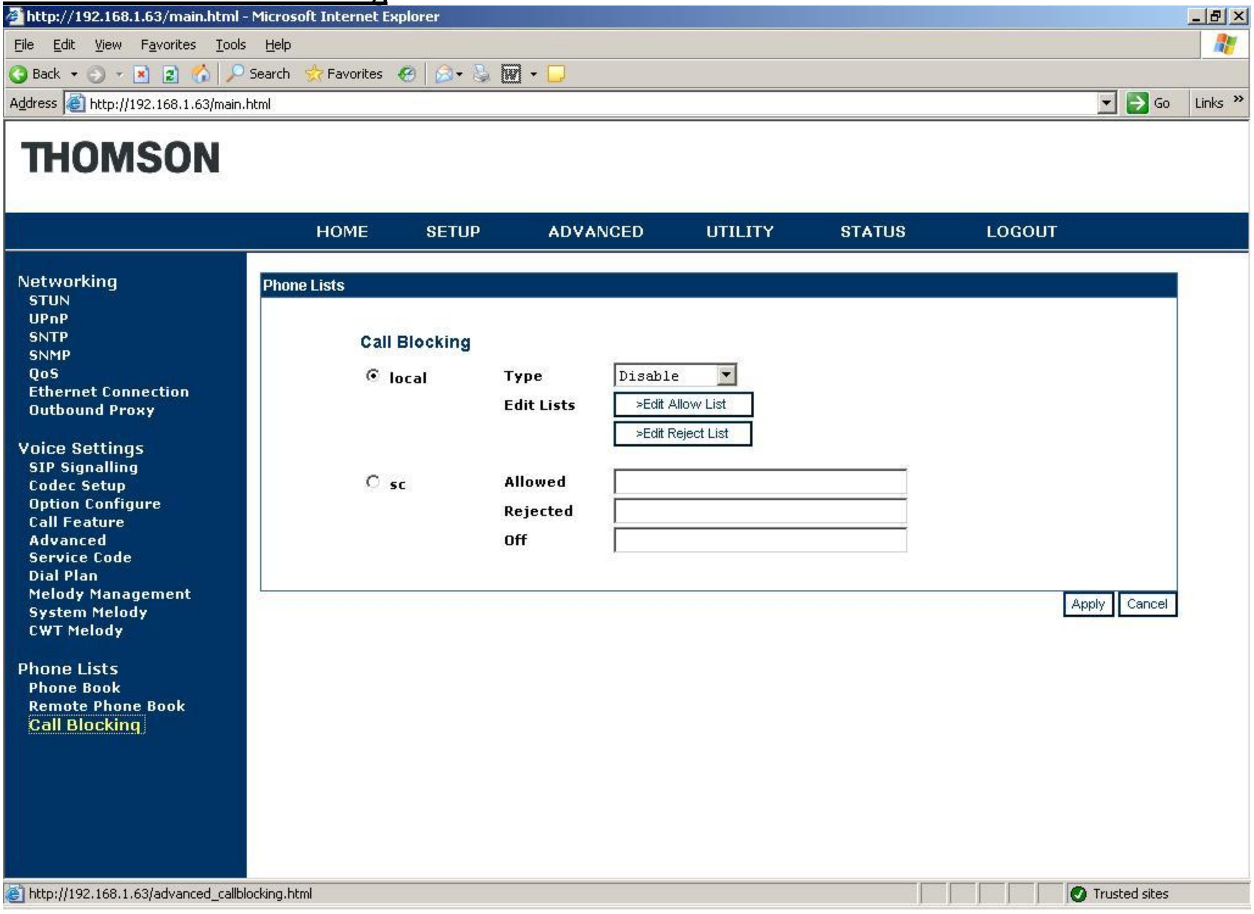
ADVANCED: Remote phone book



Remote Phone Book

Index	Phone Book URL	Phone Book Name
1.		
2.		
3.		
4.		
5.		

ADVANCED: Call blocking



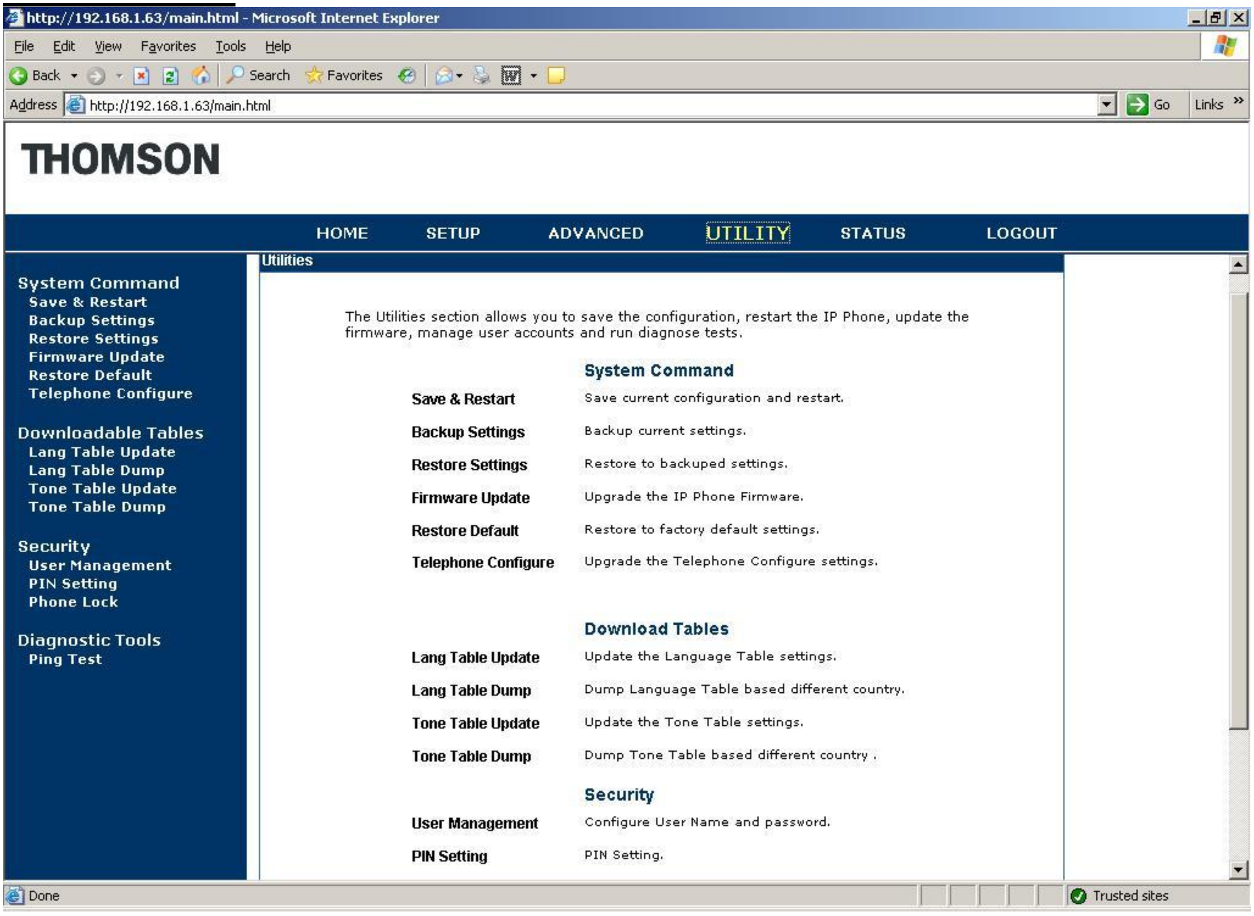
Phone Lists

Call Blocking

● local
Type: Disable
Edit Lists

○ sc
Allowed:
Rejected:
Off:

UTILITY PAGE



The Utilities section allows you to save the configuration, restart the IP Phone, update the firmware, manage user accounts and run diagnose tests.

System Command

Save & Restart	Save current configuration and restart.
Backup Settings	Backup current settings.
Restore Settings	Restore to backuped settings.
Firmware Update	Upgrade the IP Phone Firmware.
Restore Default	Restore to factory default settings.
Telephone Configure	Upgrade the Telephone Configure settings.

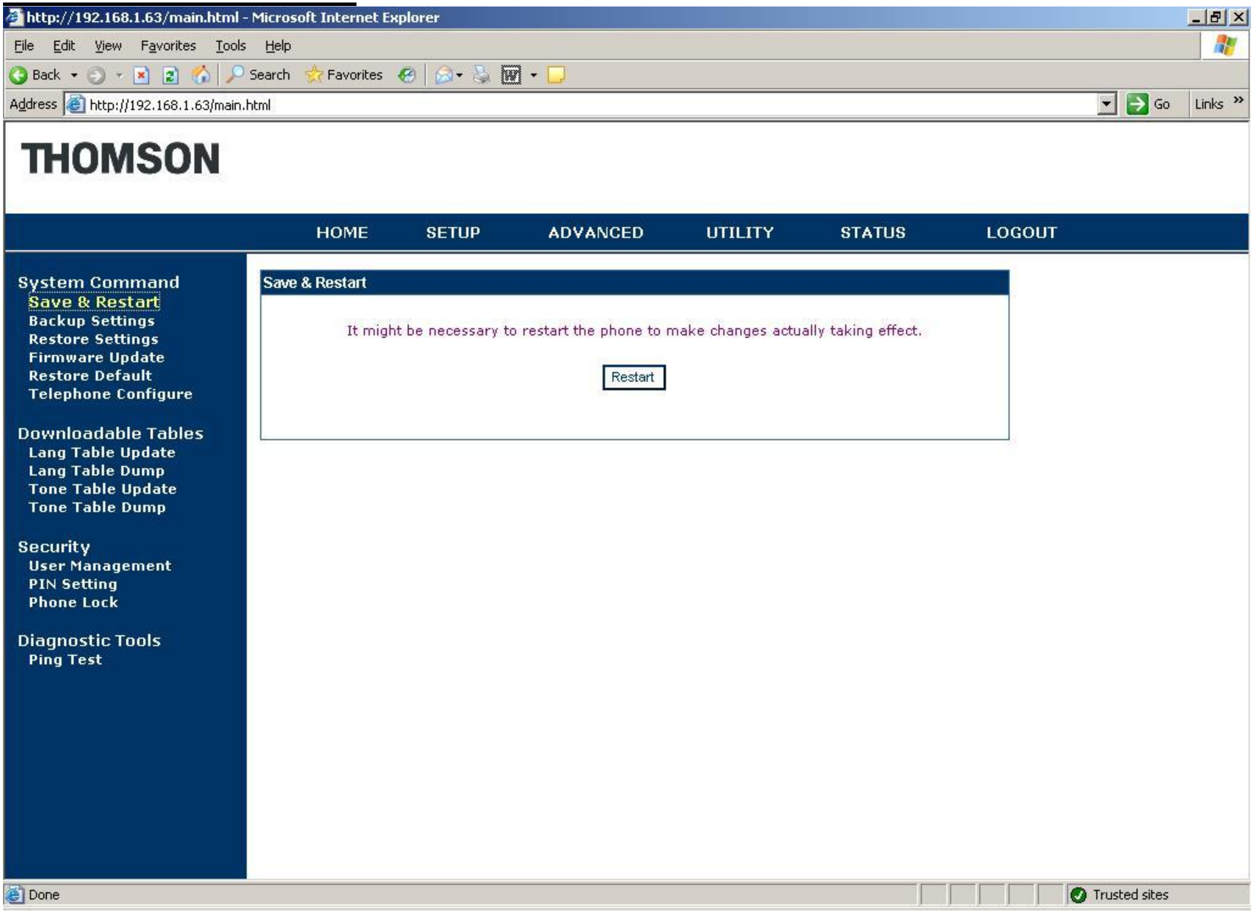
Download Tables

Lang Table Update	Update the Language Table settings.
Lang Table Dump	Dump Language Table based different country.
Tone Table Update	Update the Tone Table settings.
Tone Table Dump	Dump Tone Table based different country .

Security

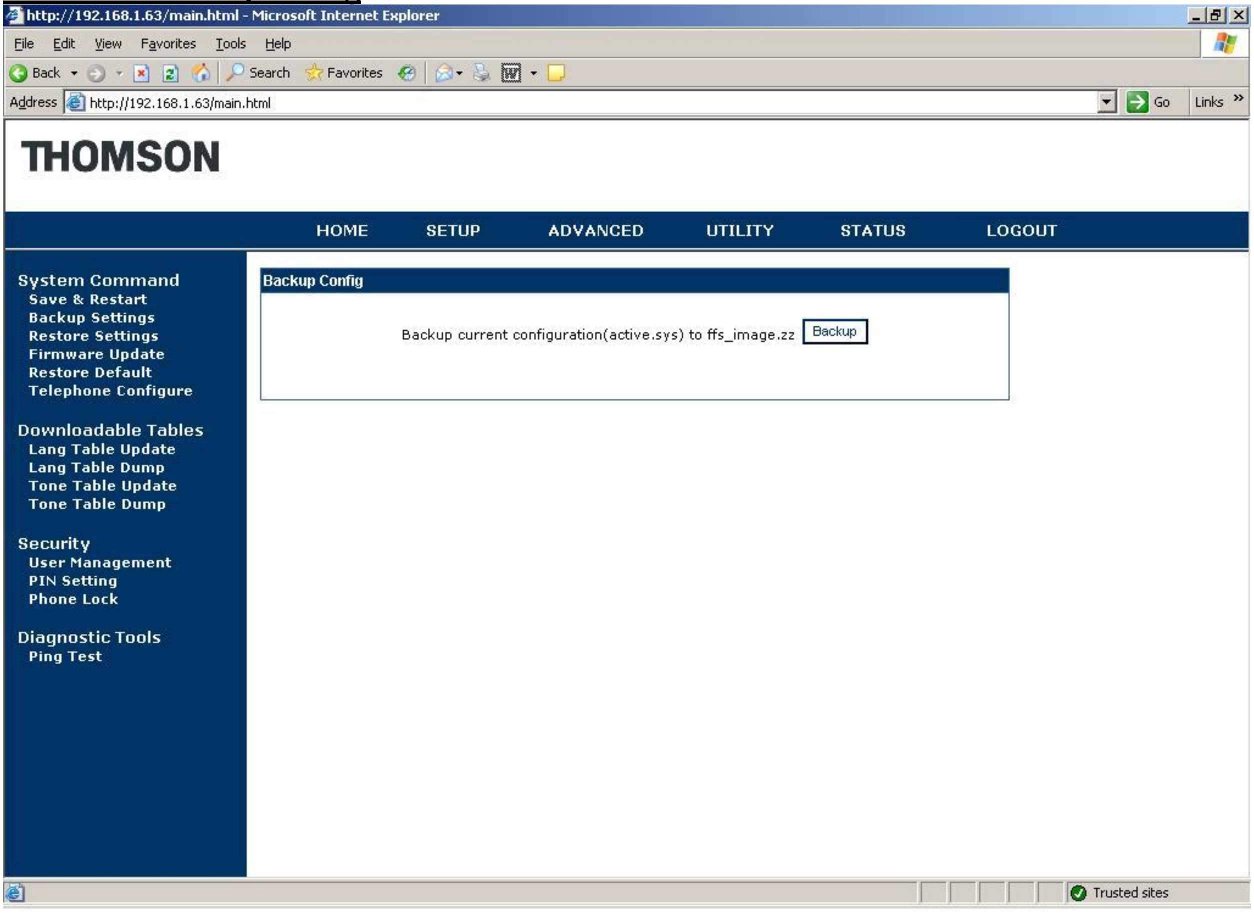
User Management	Configure User Name and password.
PIN Setting	PIN Setting.

UTILITY: Save & restart



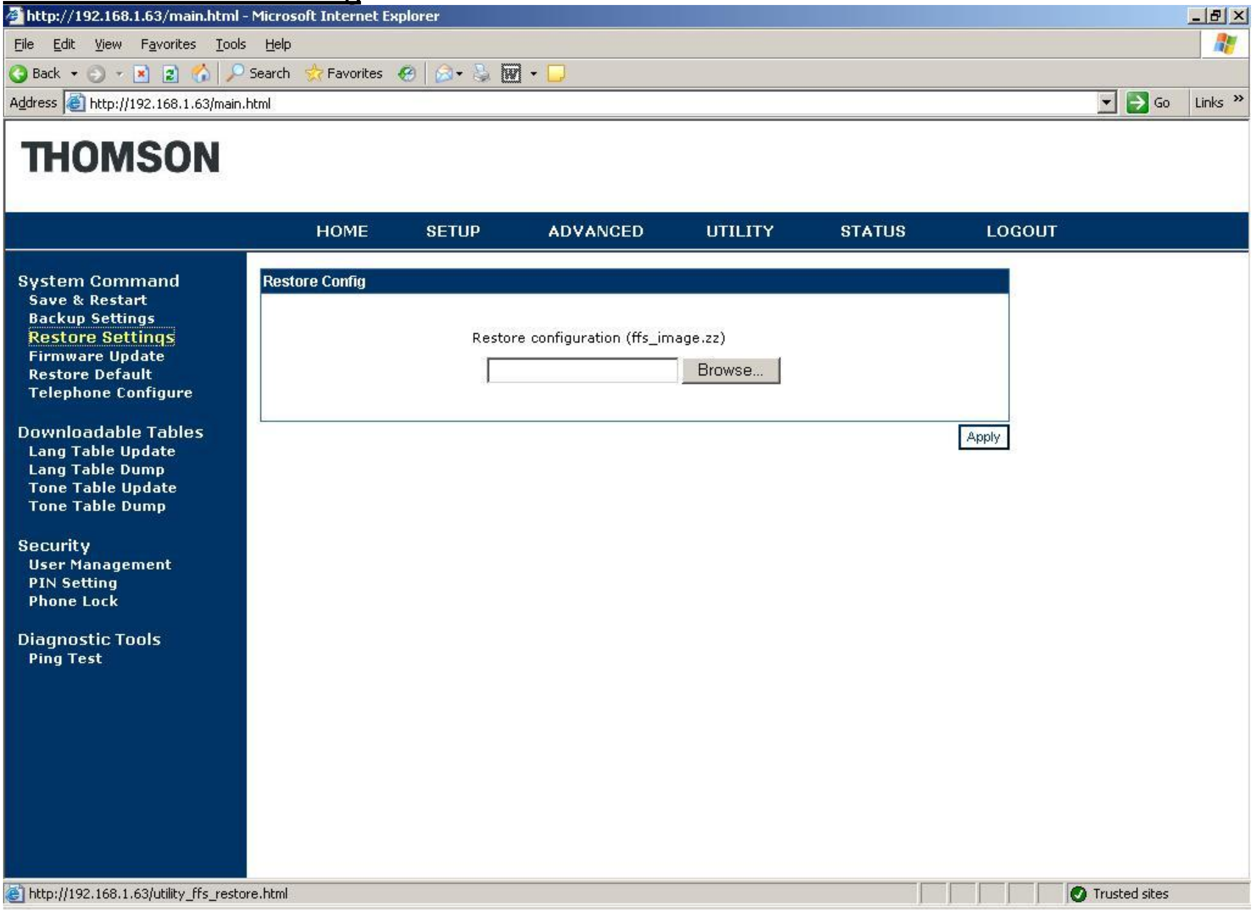
It might be necessary to restart the phone to make changes actually taking effect.

UTILITY: Backup setting



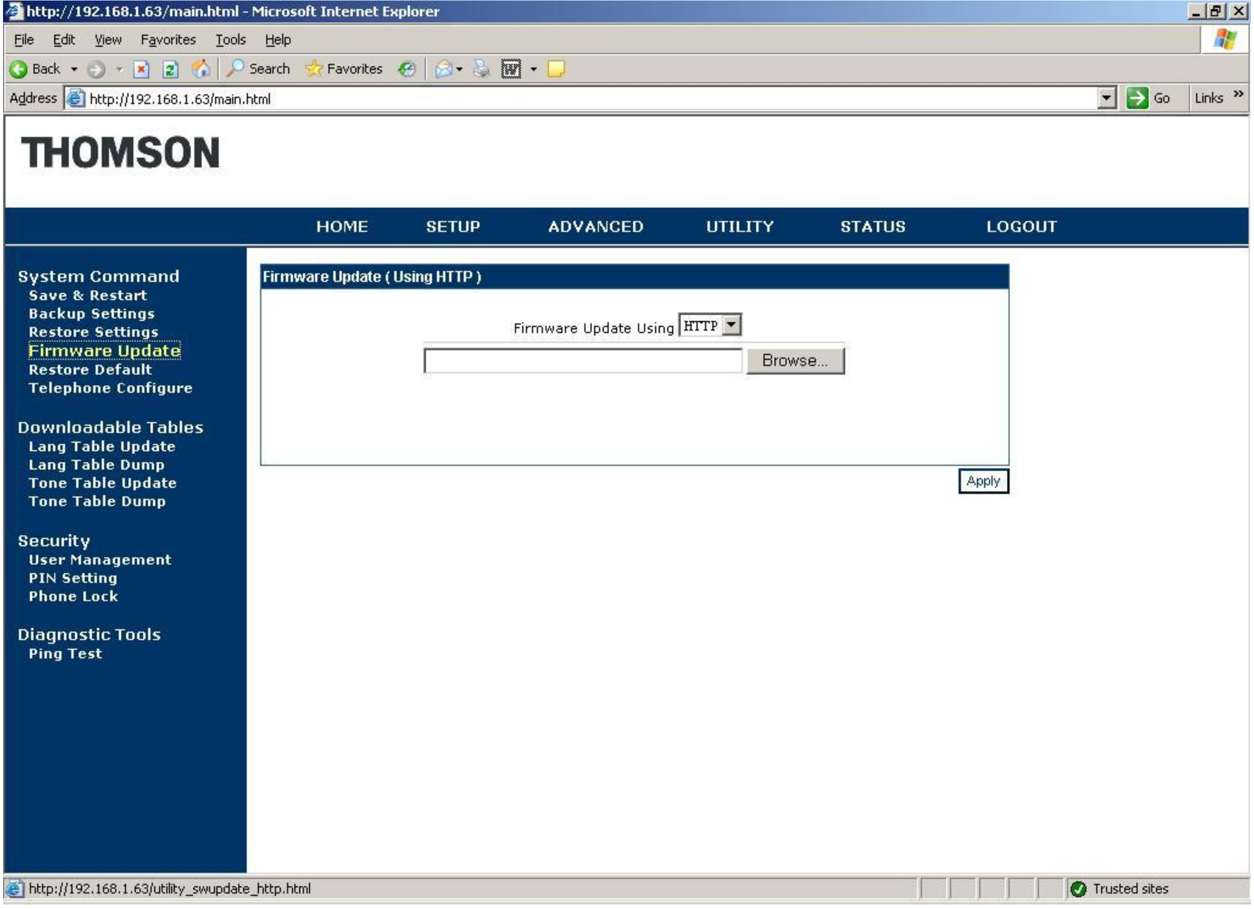
Backup current configuration(active.sys) to ffs_image.zz

UTILITY: Restore setting



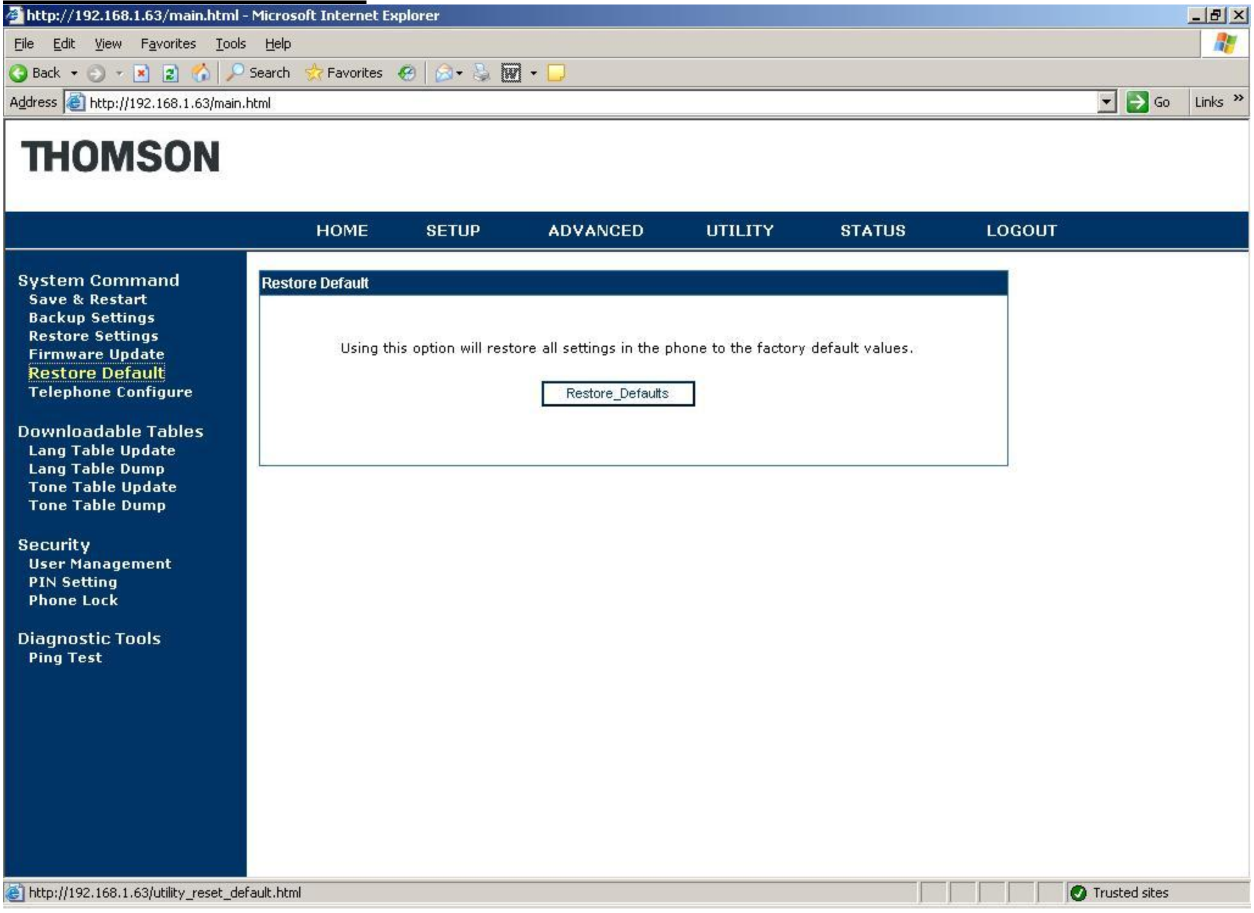
Restore configuration (ffs_image.zz)

UTILITY: Firmware Update



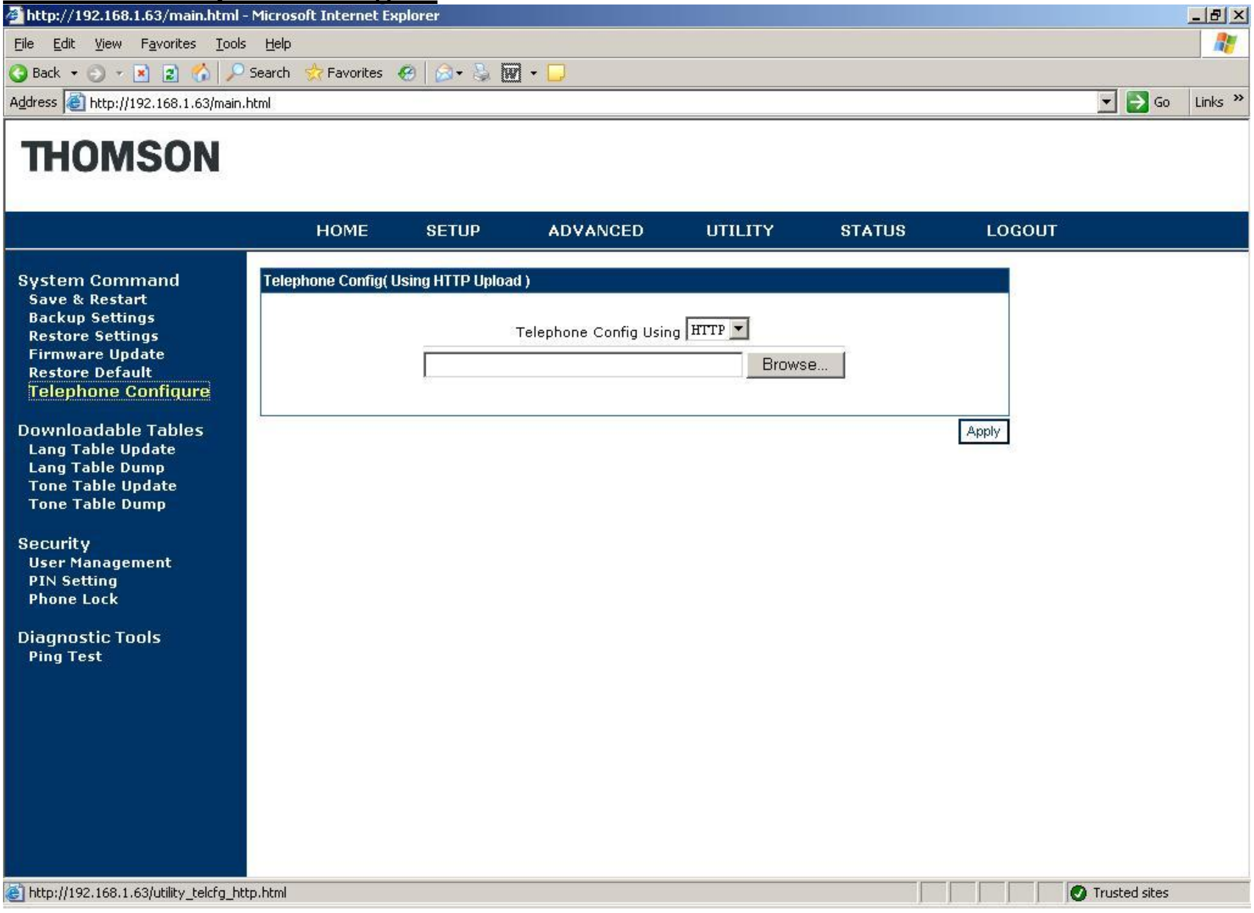
Firmware Update Using HTTP

UTILITY: Restore default



Using this option will restore all settings in the phone to the factory default values.

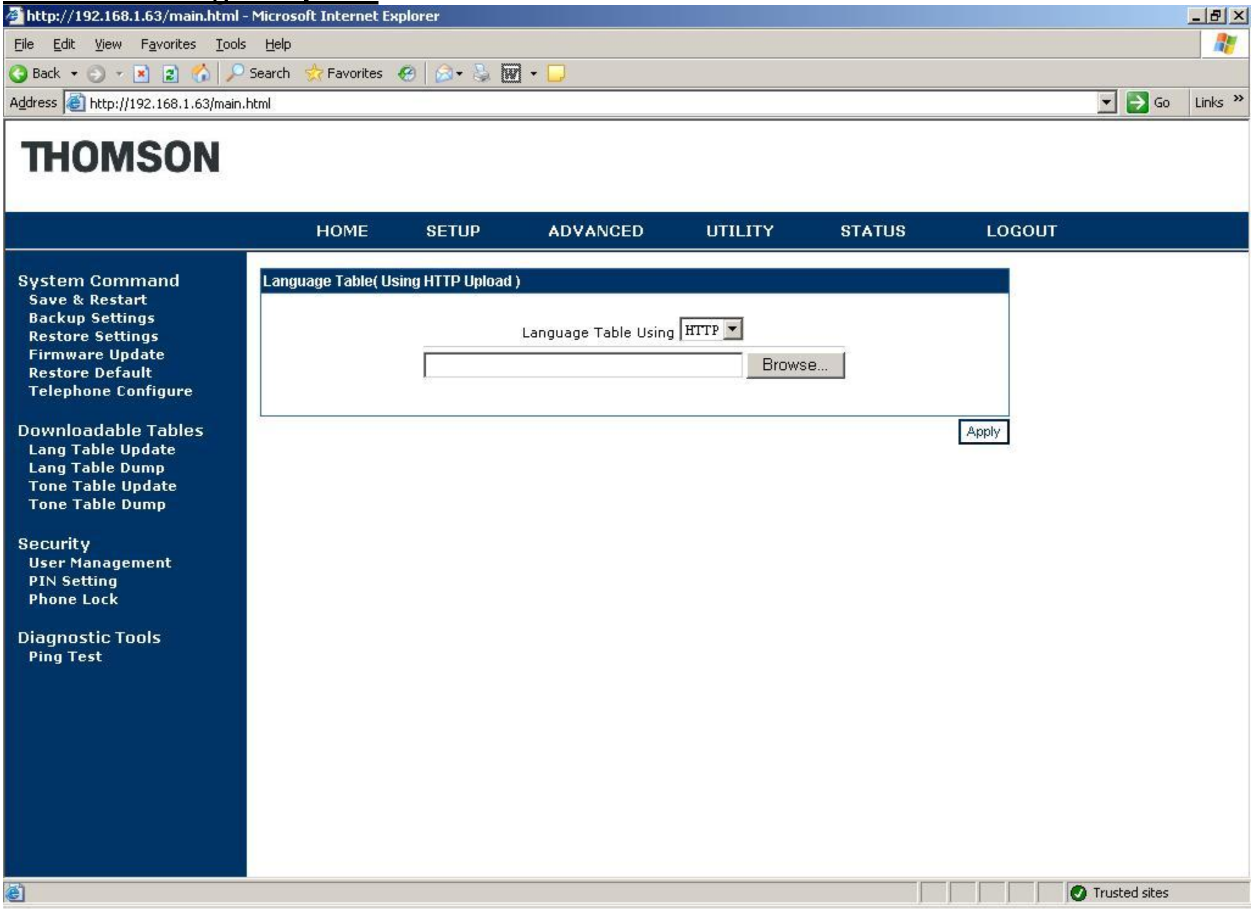
UTILITY: Telephone configure



Telephone Config(Using HTTP Upload)

Telephone Config Using: HTTP

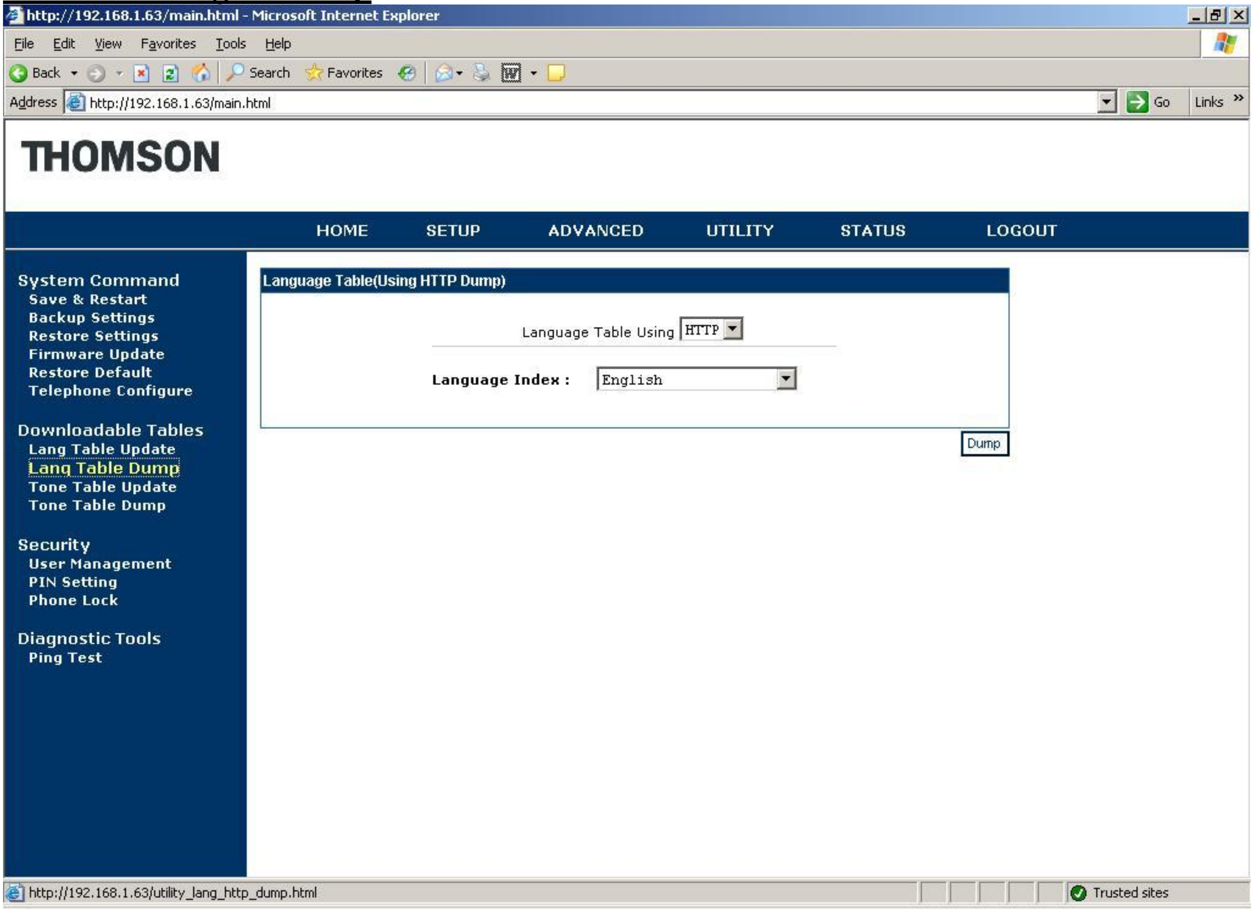
UTILITY: LangTbl update



Language Table(Using HTTP Upload)

Language Table Using: HTTP

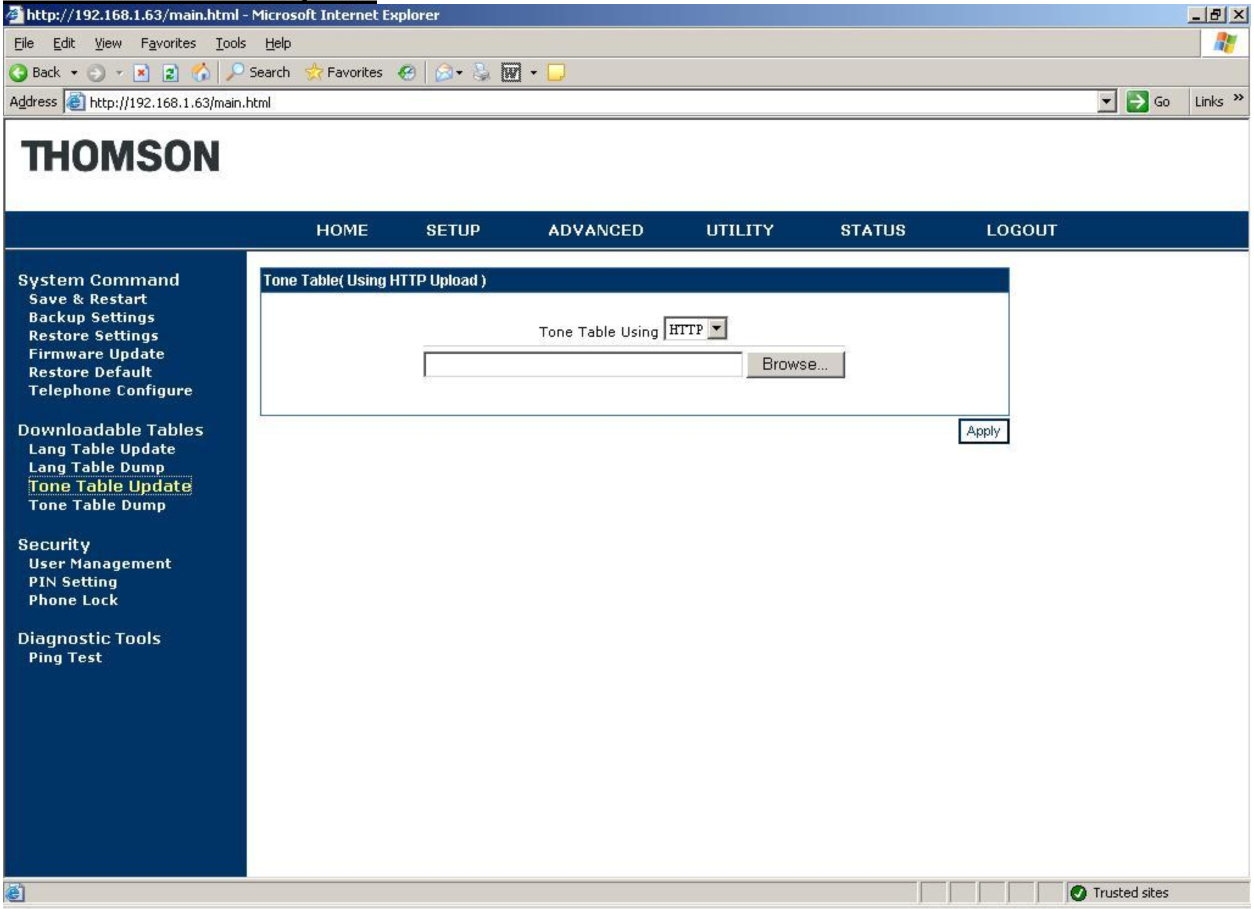
UTILITY: LangTb1 dump



Language Table Using HTTP

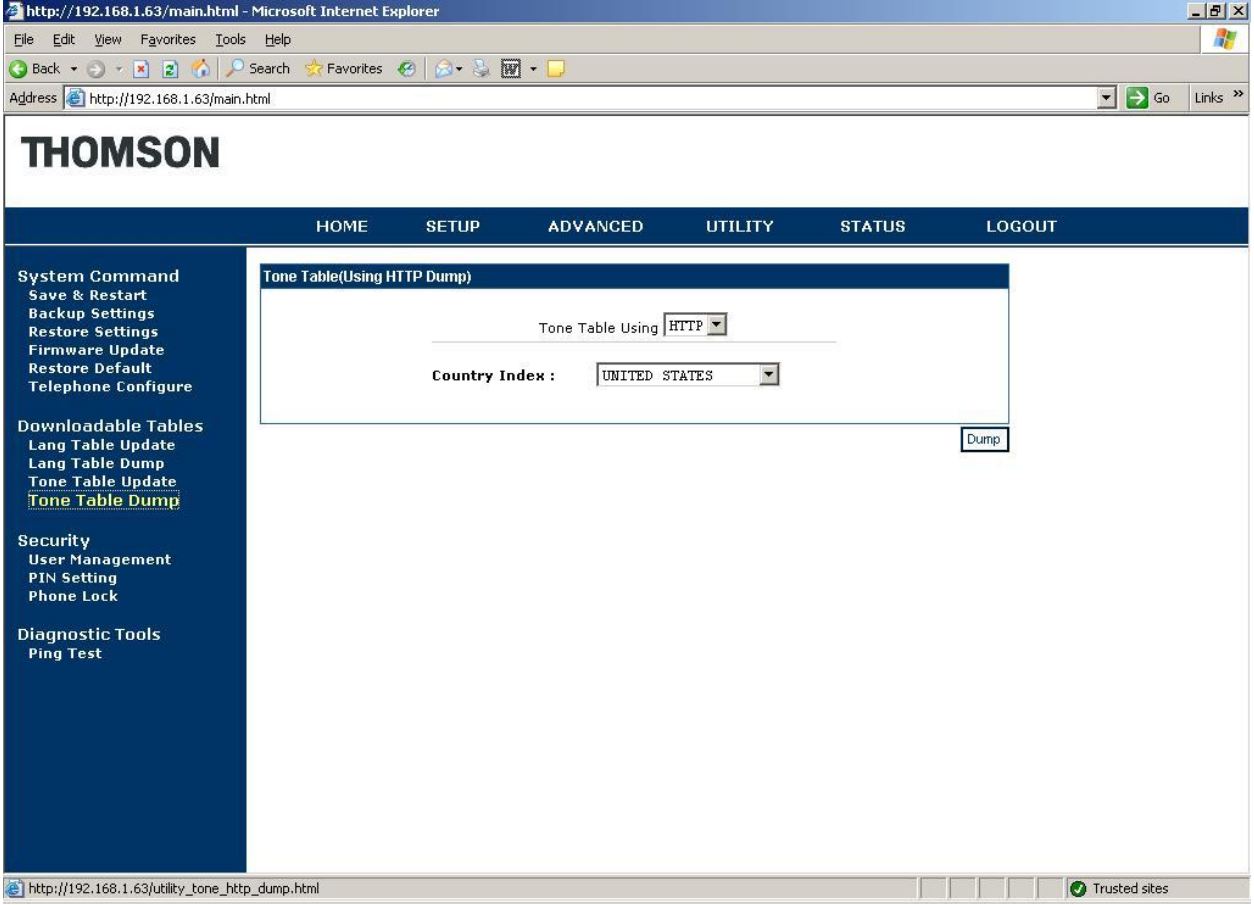
Language Index : English

UTILITY: ToneTbl update



Tone Table Using HTTP

UTILITY: ToneTbl dump

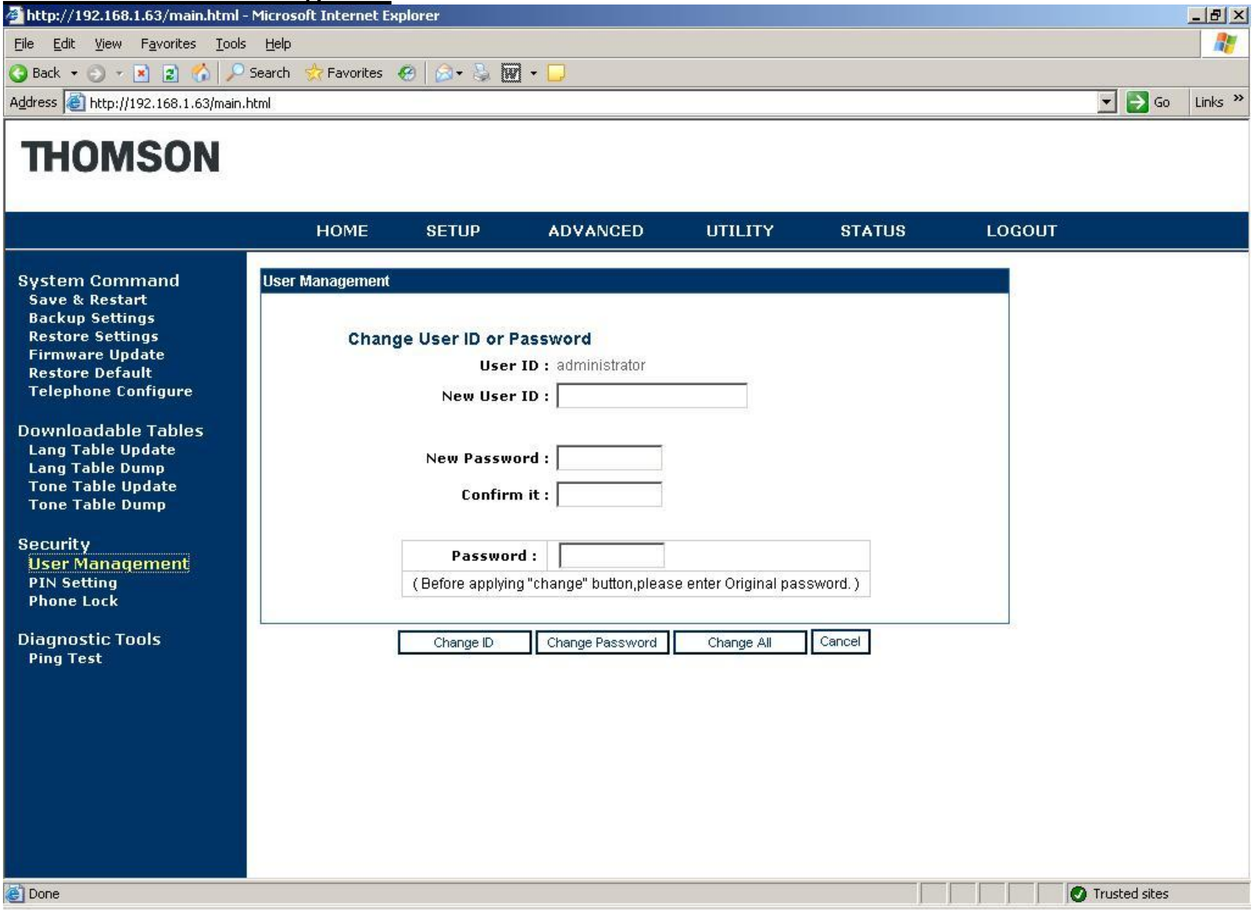


Tone Table(Using HTTP Dump)

Tone Table Using HTTP

Country Index : UNITED STATES

UTILITY: User managment



User Management

Change User ID or Password

User ID : administrator

New User ID :

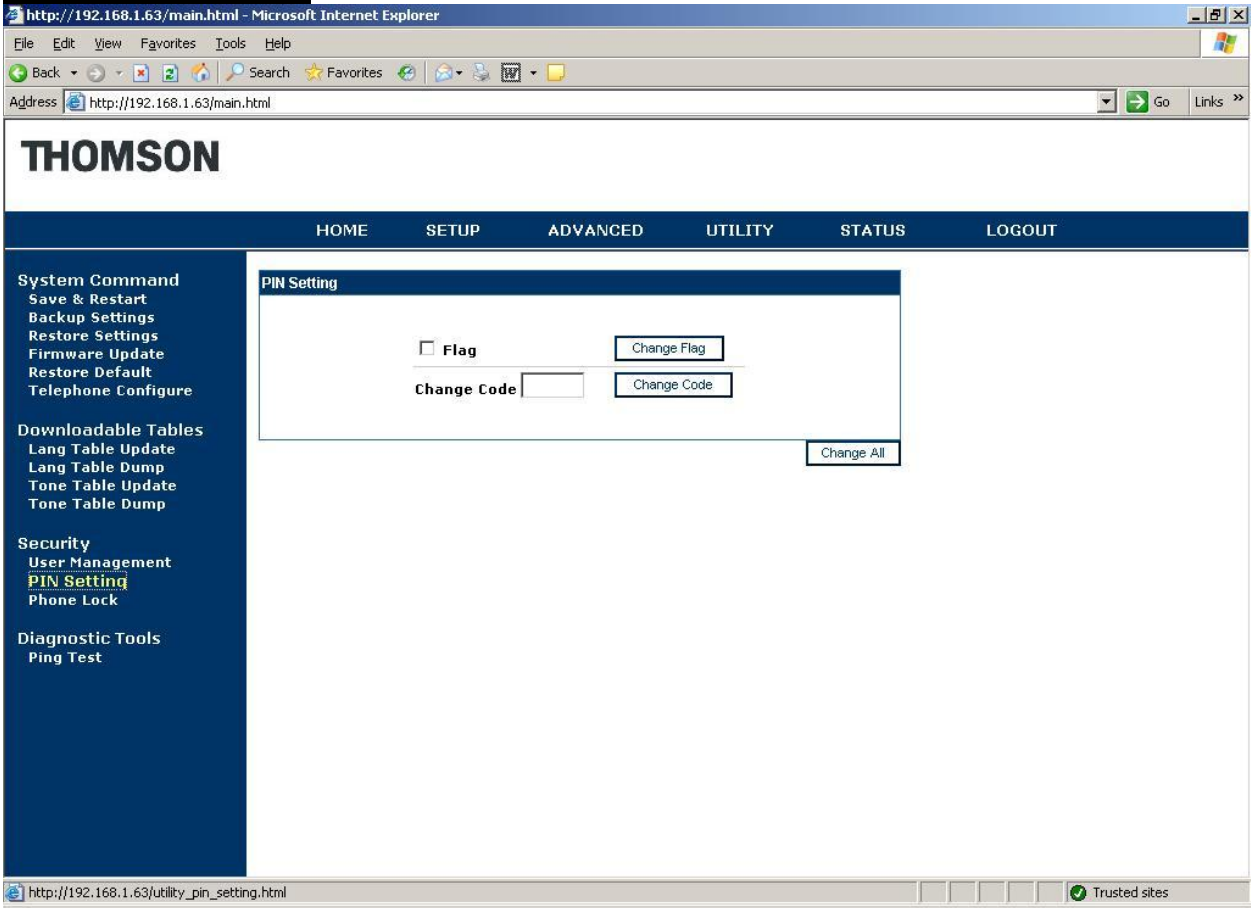
New Password :

Confirm it :

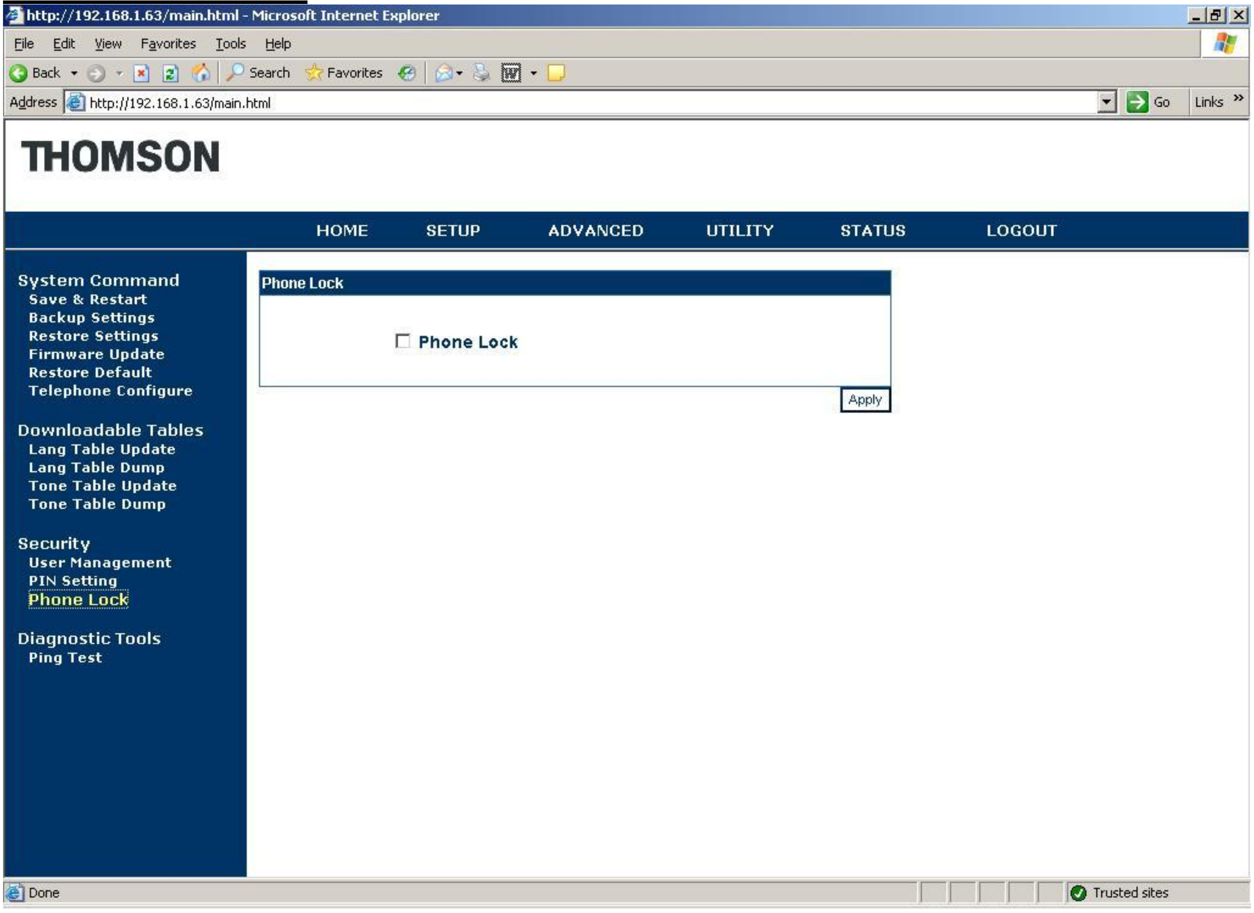
Password :

(Before applying "change" button,please enter Original password.)

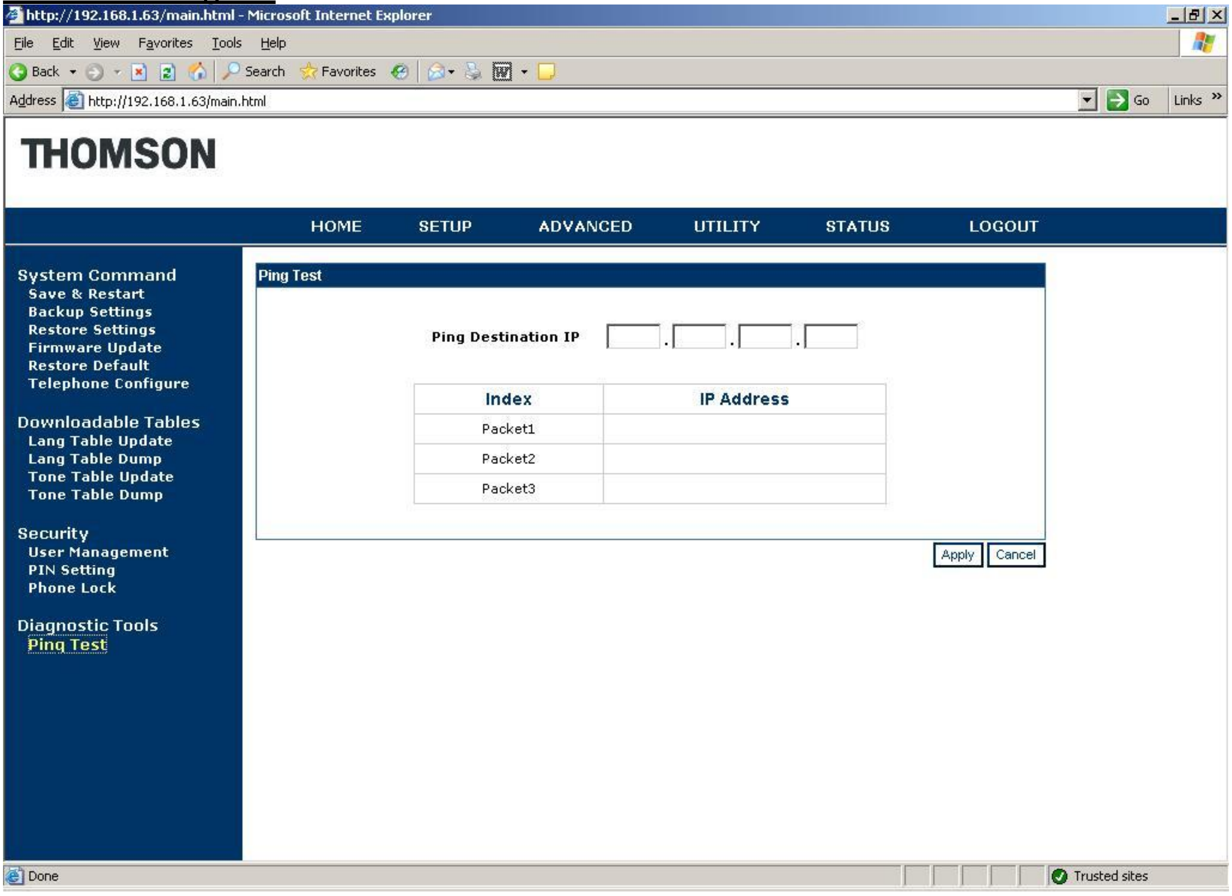
UTILITY: PIN setting



UTILITY: Phone lock



UTILITY: Ping test

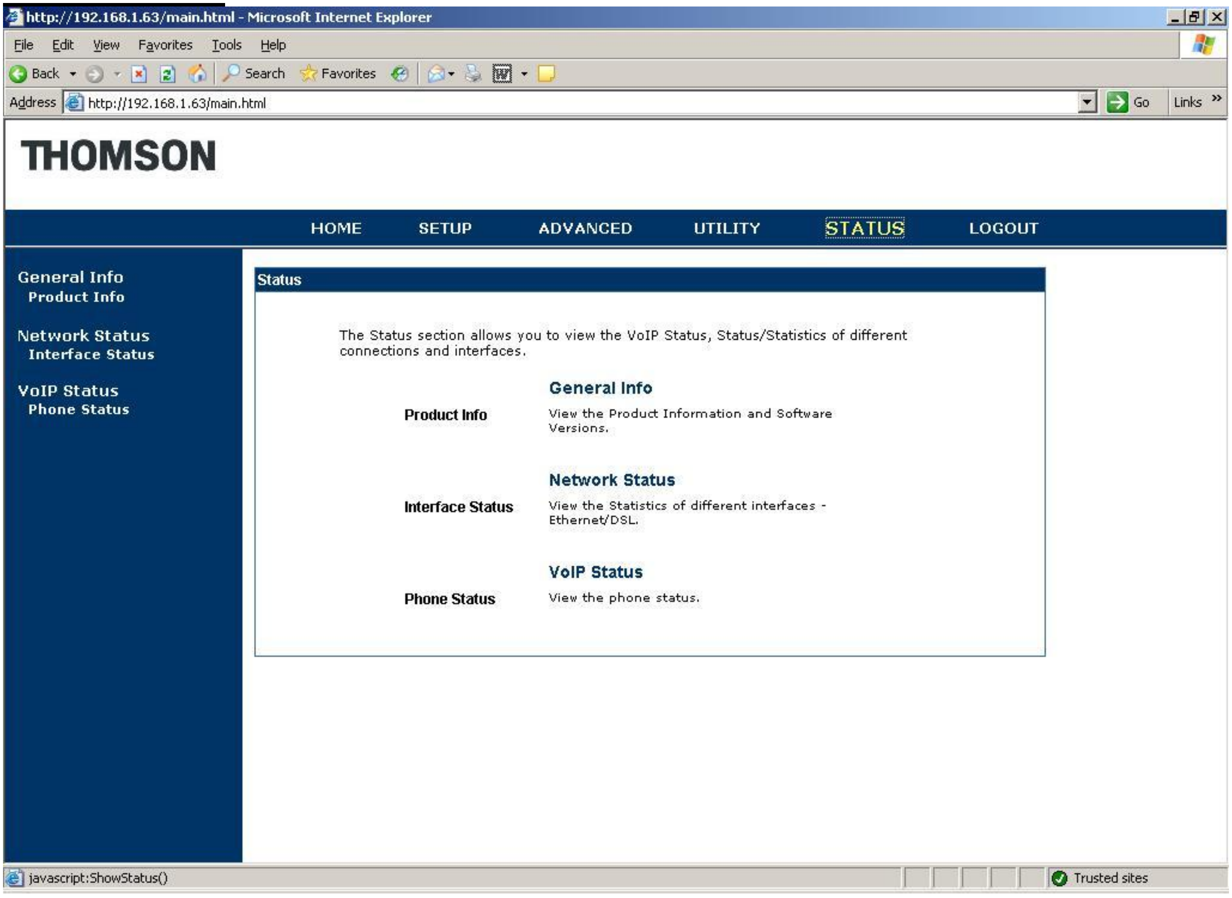


Ping Test

Ping Destination IP ___ . ___ . ___ . ___

Index	IP Address
Packet1	
Packet2	
Packet3	

STATUS PAGE

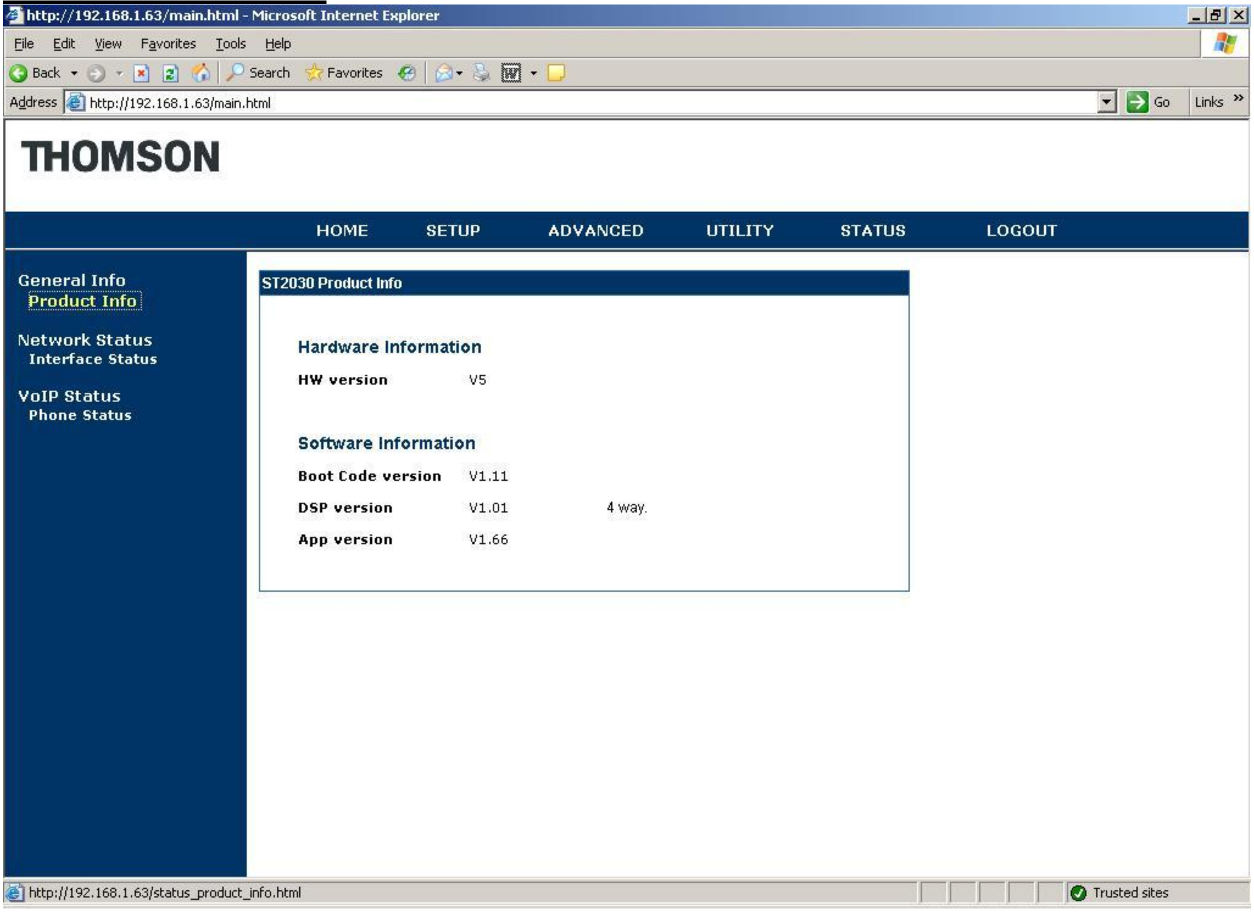


Status

The Status section allows you to view the VoIP Status, Status/Statistics of different connections and interfaces.

	General Info
Product Info	View the Product Information and Software Versions.
	Network Status
Interface Status	View the Statistics of different interfaces - Ethernet/DSL.
	VoIP Status
Phone Status	View the phone status.

STATUS : Product info



ST2030 Product Info

Hardware Information

HW version V5

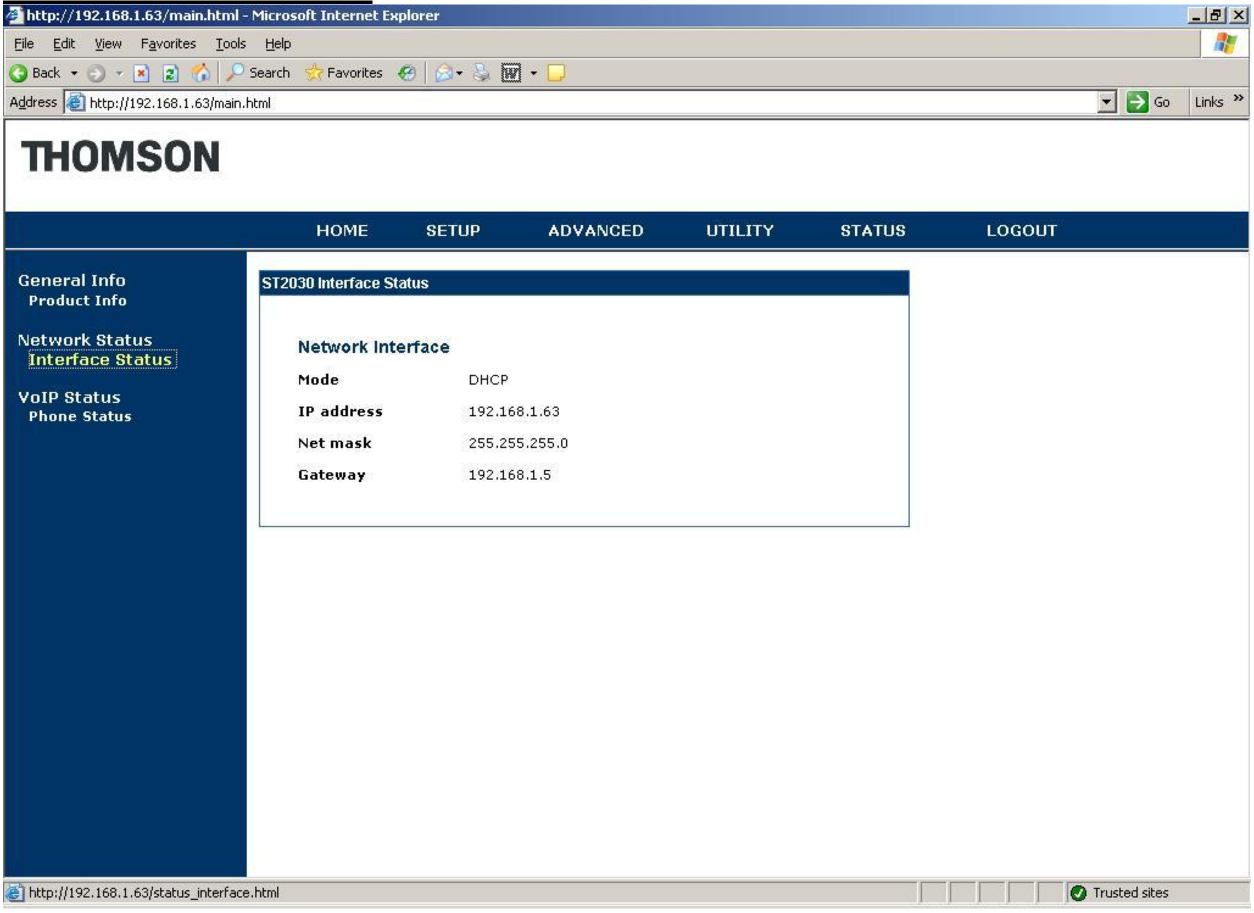
Software Information

Boot Code version V1.11

DSP version V1.01 4 way.

App version V1.66

STATUS : Interface Status



ST2030 Interface Status

Network Interface

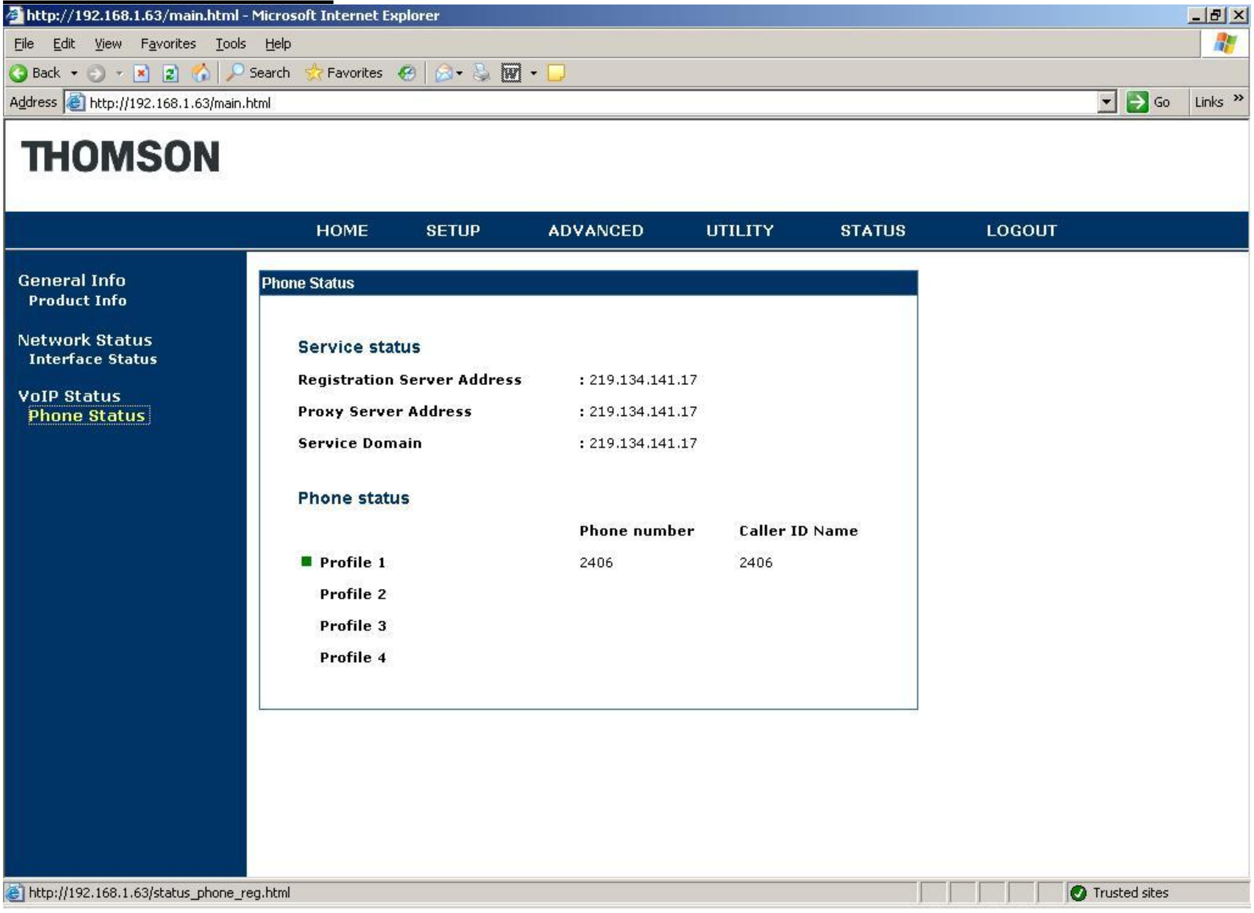
Mode DHCP

IP address 192.168.1.63

Net mask 255.255.255.0

Gateway 192.168.1.5

STATUS : Phone Status



Phone Status

Service status

Registration Server Address : 219.134.141.17

Proxy Server Address : 219.134.141.17

Service Domain : 219.134.141.17

Phone status

	Phone number	Caller ID Name
🟩 **Profile 1**	2406	2406
Profile 2		
Profile 3		
Profile 4		

Part 7 Auto-Provisioning

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

Please find below the way to provision your IP Phone.

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 set of audio parameters, which configures the phone for the audio performance.

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 (dhcpd.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
```

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
    - Reboot
    - 2nd Remote loading of file INF
    - 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
    - 2nd Attempt of remote loading of the specific file MAC if he is
    - (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.

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

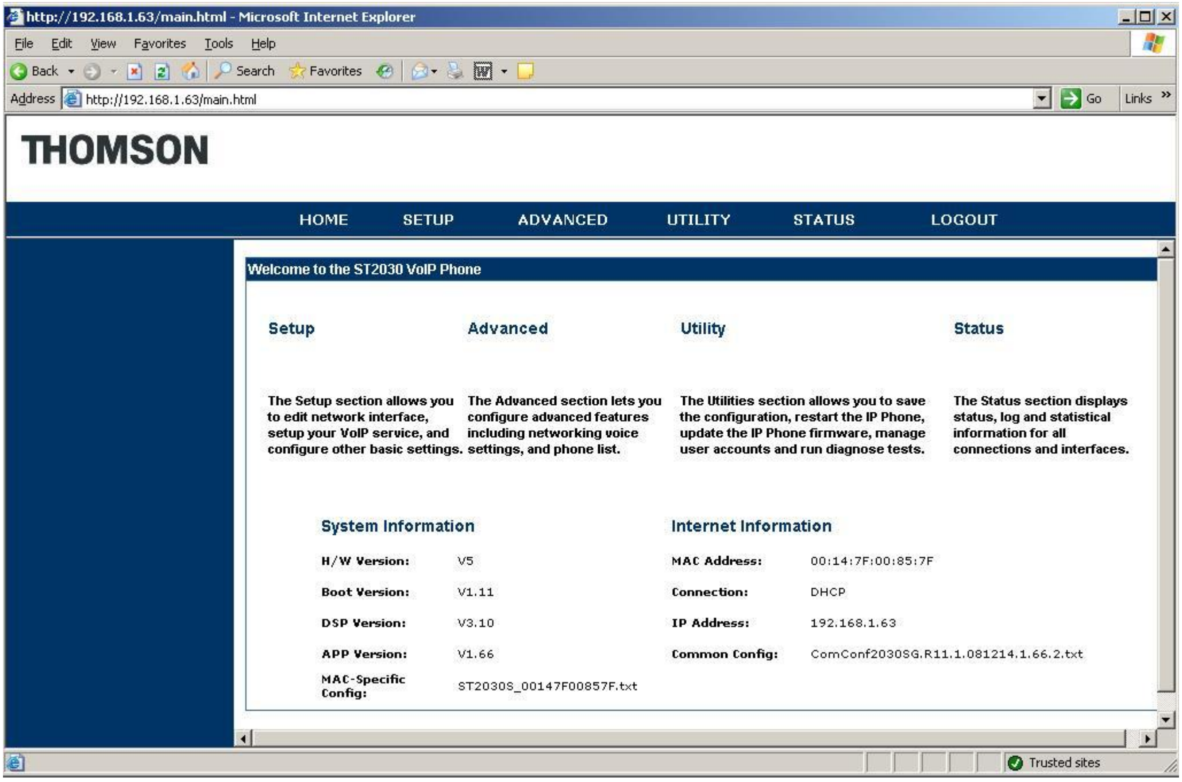
6. Open a WEB browser (Internet Explorer, Mozilla,…)

7. Enter the bar of address : IPaddressofphone/admin.html

The IPaddressofthephone being that noted in (5).The Web interface appears.

8. Login is : administrator
   Password is: 784518

9. This page is displayed:



Welcome to the ST2030 VoIP Phone

Setup	Advanced	Utility	Status
The Setup section allows you to edit network interface, setup your VoIP service, and configure other basic settings.	The Advanced section lets you configure advanced features including networking voice settings, and phone list.	The Utilities section allows you to save the configuration, restart the IP Phone, update the IP Phone firmware, manage user accounts and run diagnose tests.	The Status section displays status, log and statistical information for all connections and interfaces.

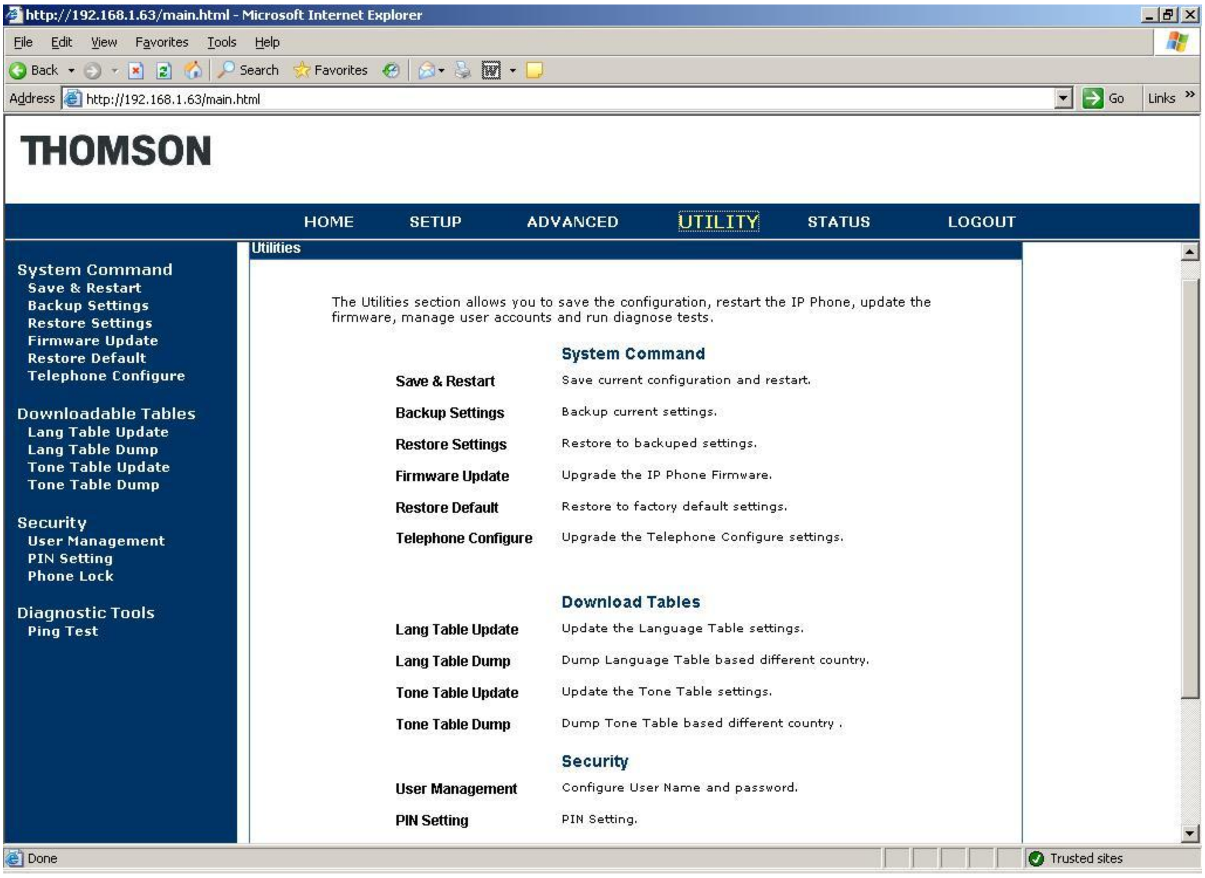
System Information

H/W Version: V5
Boot Version: V1.11
DSP Version: V3.10
APP Version: V1.66
MAC-Specific Config: ST2030S_00147F00857F.txt

Internet Information

MAC Address: 00:14:7F:00:85:7F
Connection: DHCP
IP Address: 192.168.1.63
Common Config: ComConf2030SG.R11.1.081214.1.66.2.txt

10. Select « UTILITY ».

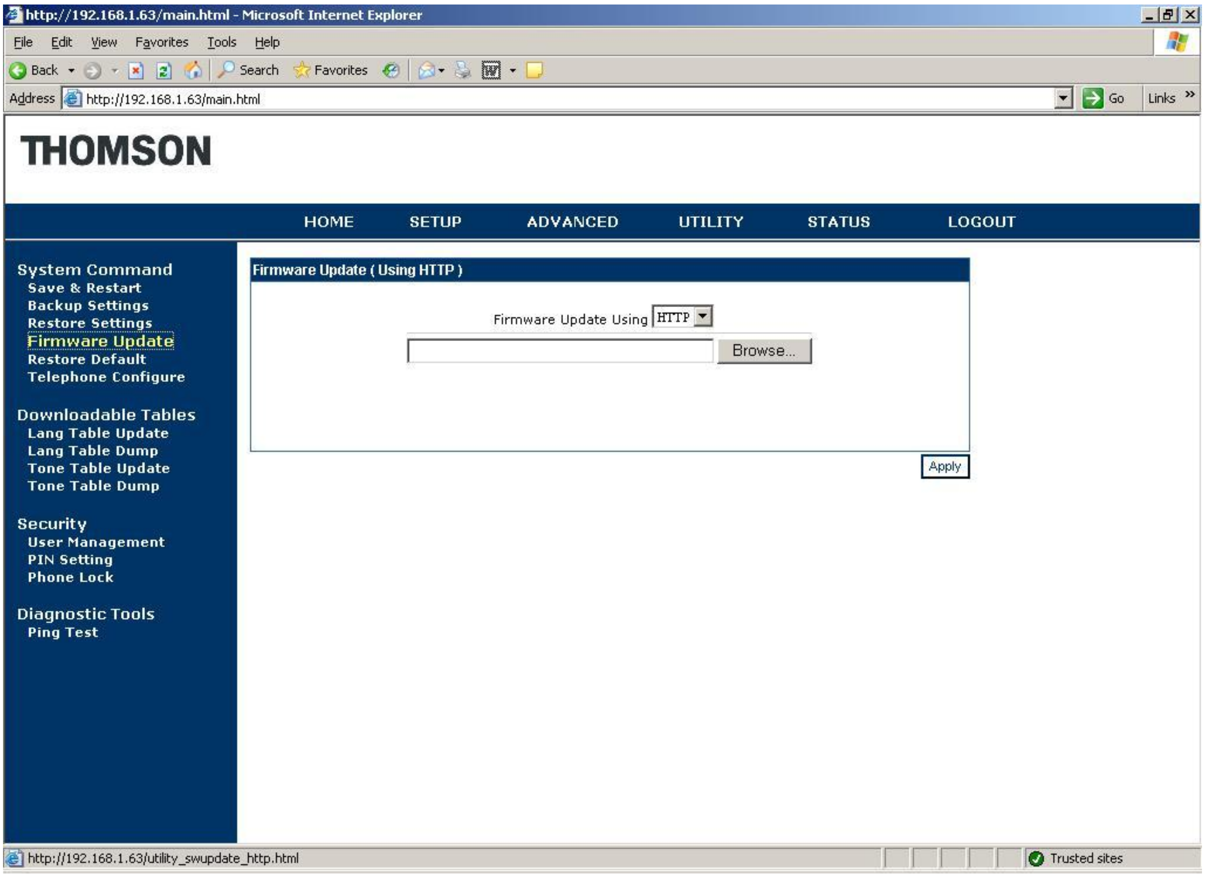


Utilities

The Utilities section allows you to save the configuration, restart the IP Phone, update the firmware, manage user accounts and run diagnose tests.

	System Command
Save & Restart	Save current configuration and restart.
Backup Settings	Backup current settings.
Restore Settings	Restore to backuped settings.
Firmware Update	Upgrade the IP Phone Firmware.
Restore Default	Restore to factory default settings.
Telephone Configure	Upgrade the Telephone Configure settings.
	Download Tables
Lang Table Update	Update the Language Table settings.
Lang Table Dump	Dump Language Table based different country.
Tone Table Update	Update the Tone Table settings.
Tone Table Dump	Dump Tone Table based different country .
	Security
User Management	Configure User Name and password.
PIN Setting	PIN Setting.

11. Select « Firmware Update »



Firmware Update (Using HTTP)

Firmware Update Using HTTP

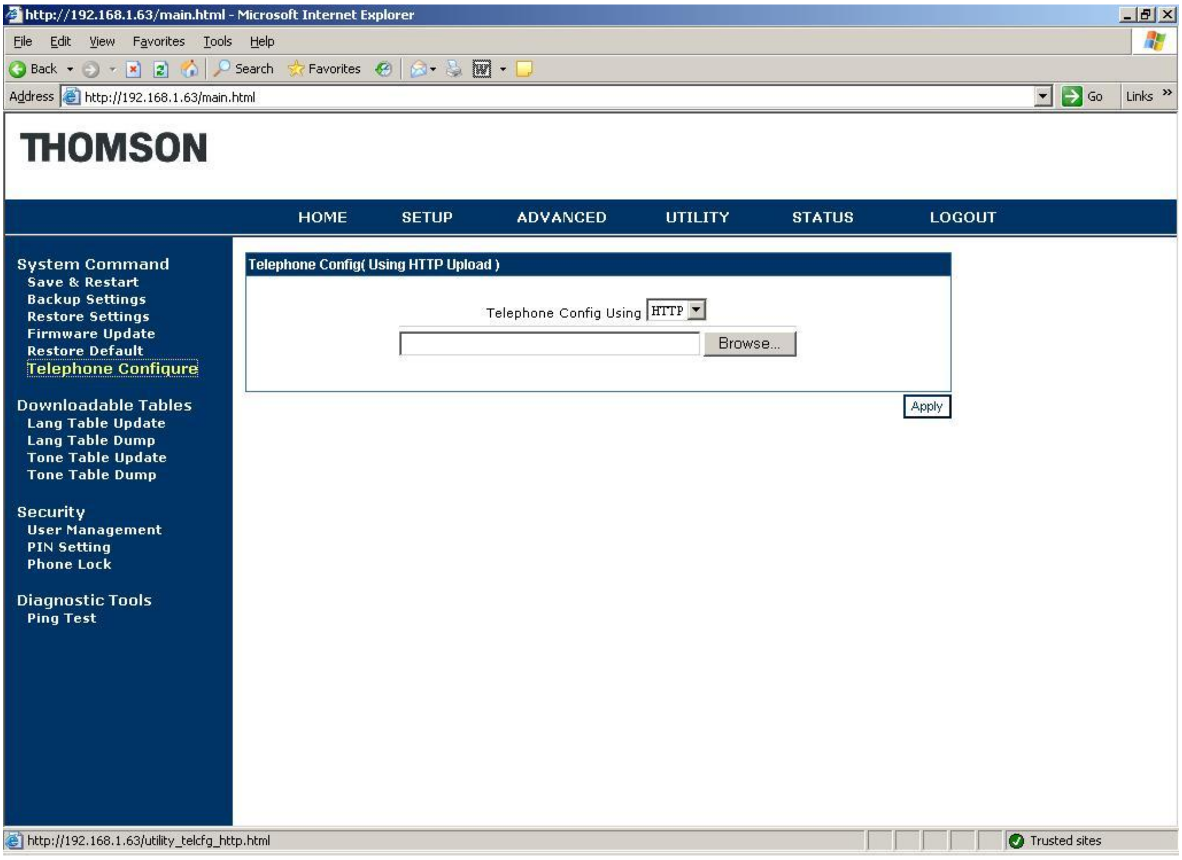
http://192.168.1.63/utility_swupdate_http.html

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. Start again from step 7) to step 11). Select « Telephone Configure »



Telephone Config(Using HTTP Upload)

Telephone Config Using HTTP

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.

Part 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:

```
Halloween:d=4,o=5,b=180:8d6,8g,8g,8d6,8g,8g,8d6,8g,8d#6,8g,8d6,8g,8g,8d6,8g,8g,8
d6,8g,8d#6,8g,8c#6,8f#,8f#,8c#6,8f#,8f#,8c#6,8f#,8d6,8f#,8c#6,8f#,8f#,8c#6,8f#,8f#,8c
#6,8f#,8d6,8f#
```

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
- www.xgsmonline.com/smartsmslogos.shtml
- 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.

1. Select a phone: Alcatel | Alcatel One-Touch 300

2. Select a ringtone: Select category | Select a ringtone...

Select a ringtone using the selection boxes above, or paste your own RTTTL ringtone in here.

3. Click this button to convert the ringtone into a key press sequence for your mobile phone

Thank you for using the Ringtone Converter Software.

Getting started is easy. First select a phone (make and model), then select a ringtone. Finally click the 'convert' button to generate instructions on how to add the ringtone to your phone.

If you want to enter your own ringtone in RTTTL or Nokia Composer format, simply paste the code into the large box above, then click the 'convert' button to convert the tone into entry instructions for your phone. This allows you to add almost any ringtone to your phone as most tunes are available on the internet in RTTTL or Nokia format.

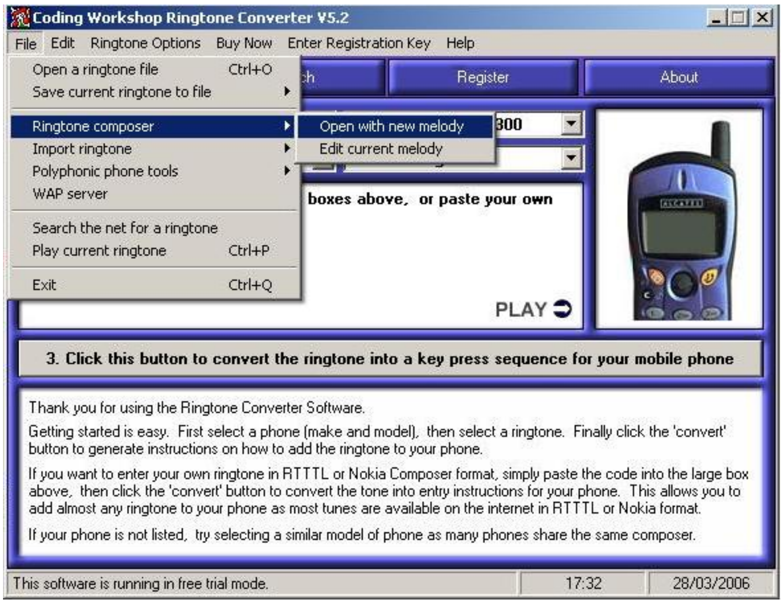
If your phone is not listed, try selecting a similar model of phone as many phones share the same composer.

This software is running in free trial mode. | 17:29 | 28/03/2006

Figure 8.1 Ringtone Converter

You can download this software here:
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"



Thank you for using the Ringtone Converter Software.

Getting started is easy. First select a phone (make and model), then select a ringtone. Finally click the 'convert' button to generate instructions on how to add the ringtone to your phone.

If you want to enter your own ringtone in RTTTL or Nokia Composer format, simply paste the code into the large box above, then click the 'convert' button to convert the tone into entry instructions for your phone. This allows you to add almost any ringtone to your phone as most tunes are available on the internet in RTTTL or Nokia format.

If your phone is not listed, try selecting a similar model of phone as many phones share the same composer.

Figure 8.2 Ringtone Converter

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

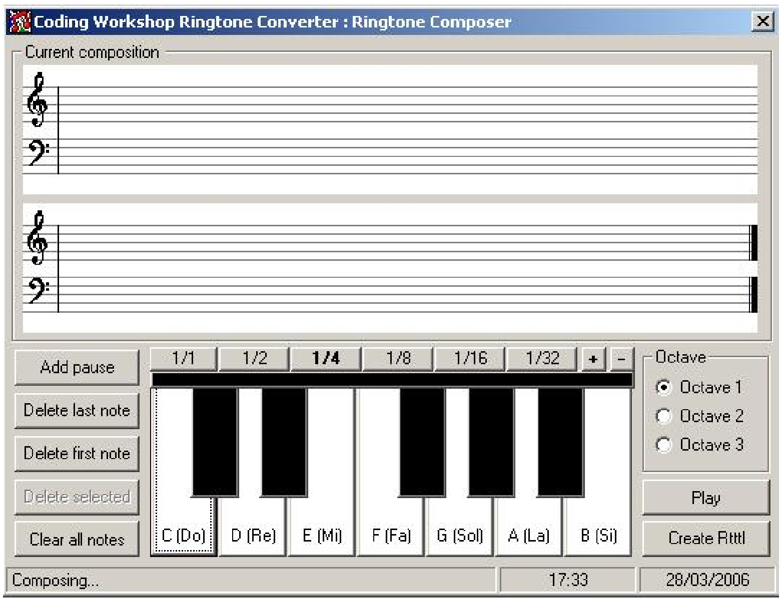
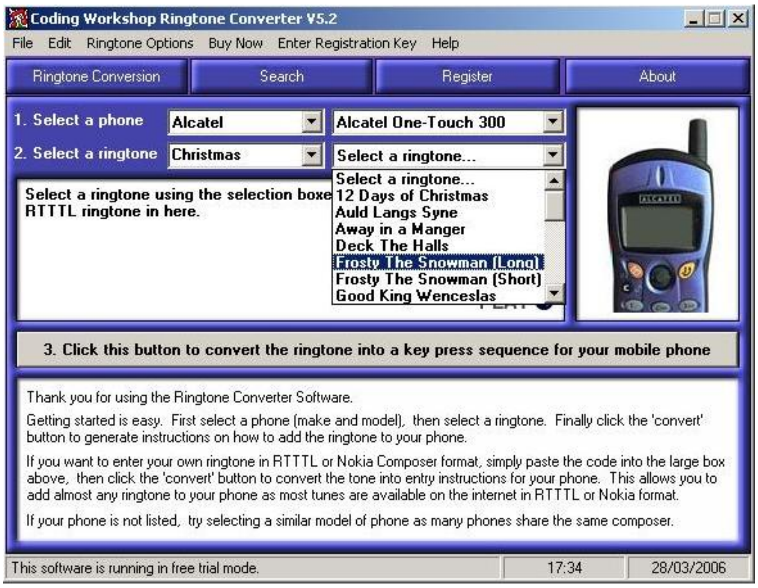


Figure 8.3 Ringtone Composer

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.



1. Select a phone: Alcatel — Alcatel One-Touch 300

2. Select a ringtone: Christmas — Select a ringtone...

Select a ringtone using the selection boxe
RTTTL ringtone in here.

Thank you for using the Ringtone Converter Software.

Getting started is easy. First select a phone (make and model), then select a ringtone. Finally click the 'convert' button to generate instructions on how to add the ringtone to your phone.

If you want to enter your own ringtone in RTTTL or Nokia Composer format, simply paste the code into the large box above, then click the 'convert' button to convert the tone into entry instructions for your phone. This allows you to add almost any ringtone to your phone as most tunes are available on the internet in RTTTL or Nokia format.

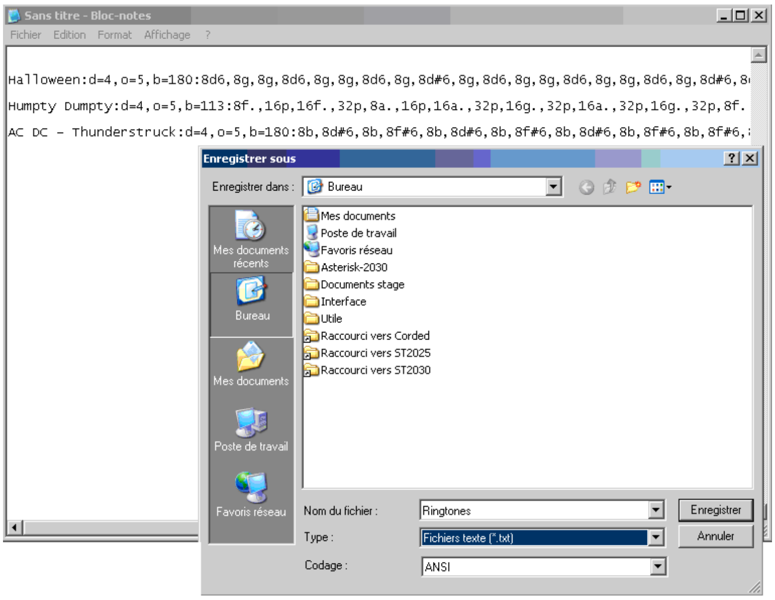
If your phone is not listed, try selecting a similar model of phone as many phones share the same composer.

This software is running in free trial mode. 17:34 28/03/2006

Figure 8.4 Ringtone Composer

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*.



```
Halloween:d=4,o=5,b=180:8d6,8g,8g,8d6,8g,8g,8d6,8g,8d#6,8g,8d6,8g,8g,8d6,8g,8g,8d6,8g,8d#6,
Humpty Dumpty:d=4,o=5,b=113:8f.,16p,16f.,32p,8a.,16p,16a.,32p,16g.,32p,16a.,32p,16g.,32p,8f.
AC DC - Thunderstruck:d=4,o=5,b=180:8b,8d#6,8b,8f#6,8b,8d#6,8b,8f#6,8b,8d#6,8b,8f#6,8b,8f#6,
```

Nom du fichier : Ringtones

Type : Fichiers texte (*.txt)

Codage : ANSI

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

If MGCP phone:

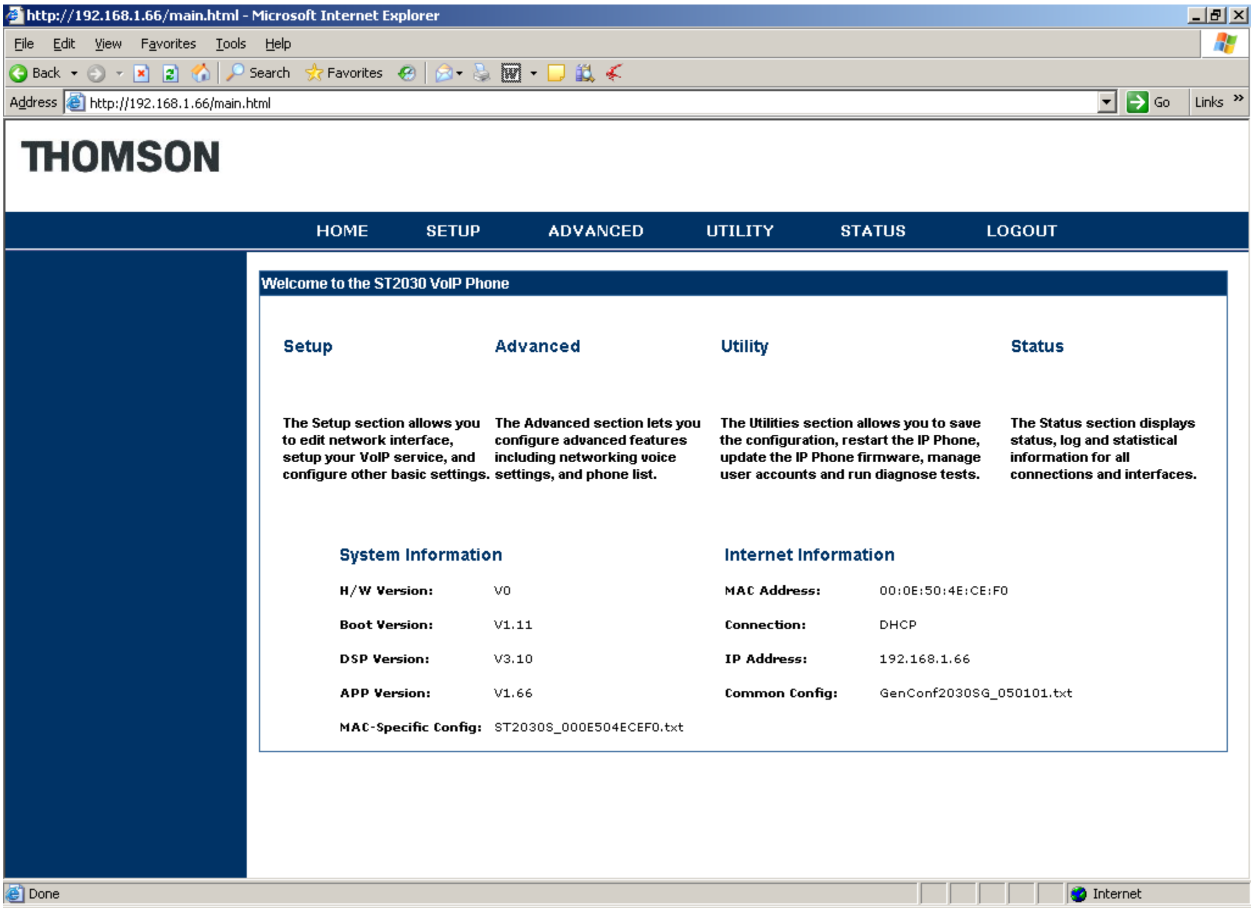
- Press on the key menu of the phone
- Press on the softkey admin
- Press on the button OK of the phone
- Select Network configuration while going down with the arrows
- Select IP address and note the address indicated

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

So this screen is displayed:



Welcome to the ST2030 VoIP Phone

Setup	Advanced	Utility	Status
The Setup section allows you to edit network interface, setup your VoIP service, and configure other basic settings.	The Advanced section lets you configure advanced features including networking voice settings, and phone list.	The Utilities section allows you to save the configuration, restart the IP Phone, update the IP Phone firmware, manage user accounts and run diagnose tests.	The Status section displays status, log and statistical information for all connections and interfaces.

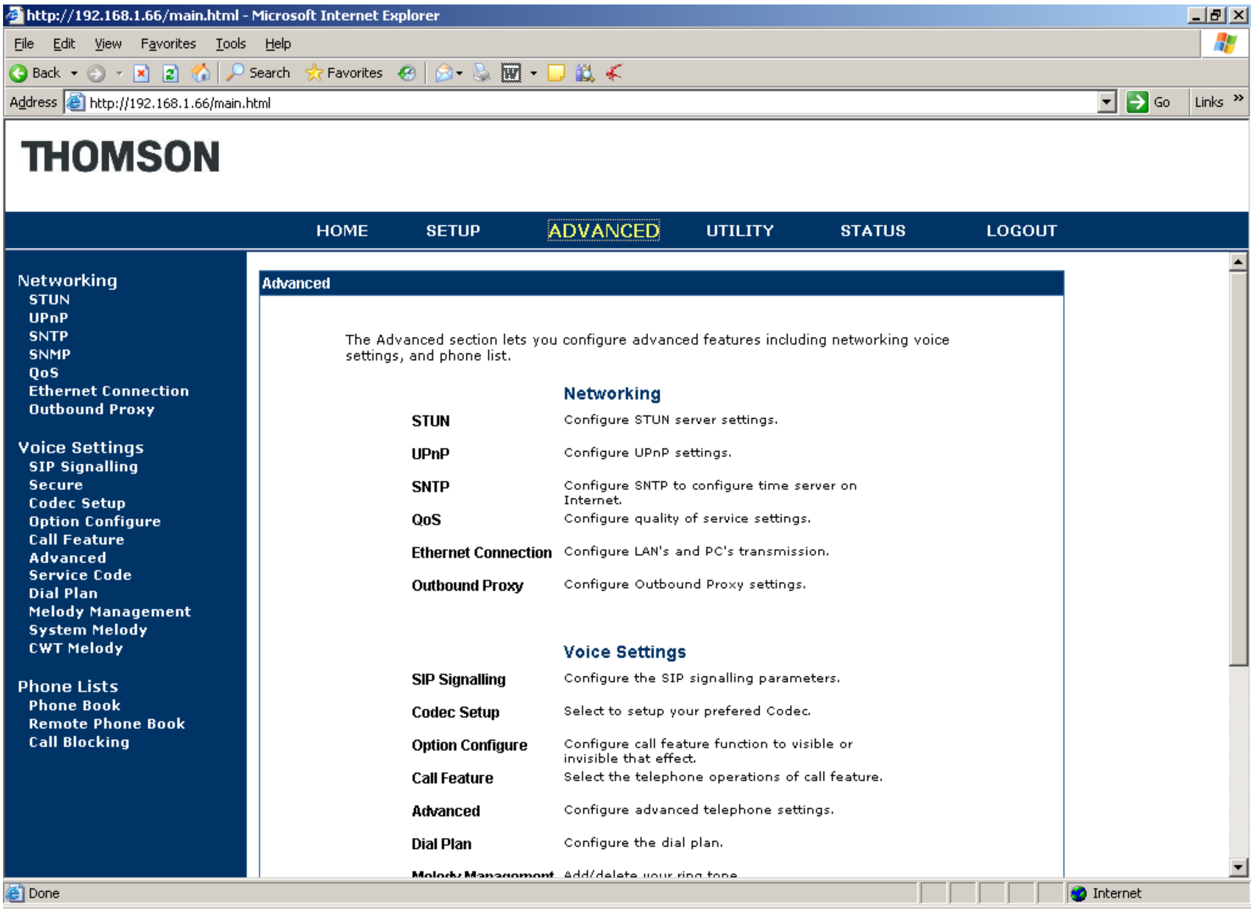
System Information

H/W Version:	V0
Boot Version:	V1.11
DSP Version:	V3.10
APP Version:	V1.66
MAC-Specific Config:	ST2030S_000E504ECEF0.txt

Internet Information

MAC Address:	00:0E:50:4E:CE:F0
Connection:	DHCP
IP Address:	192.168.1.66
Common Config:	GenConf2030SG_050101.txt

Next, we must click on « Advanced »:



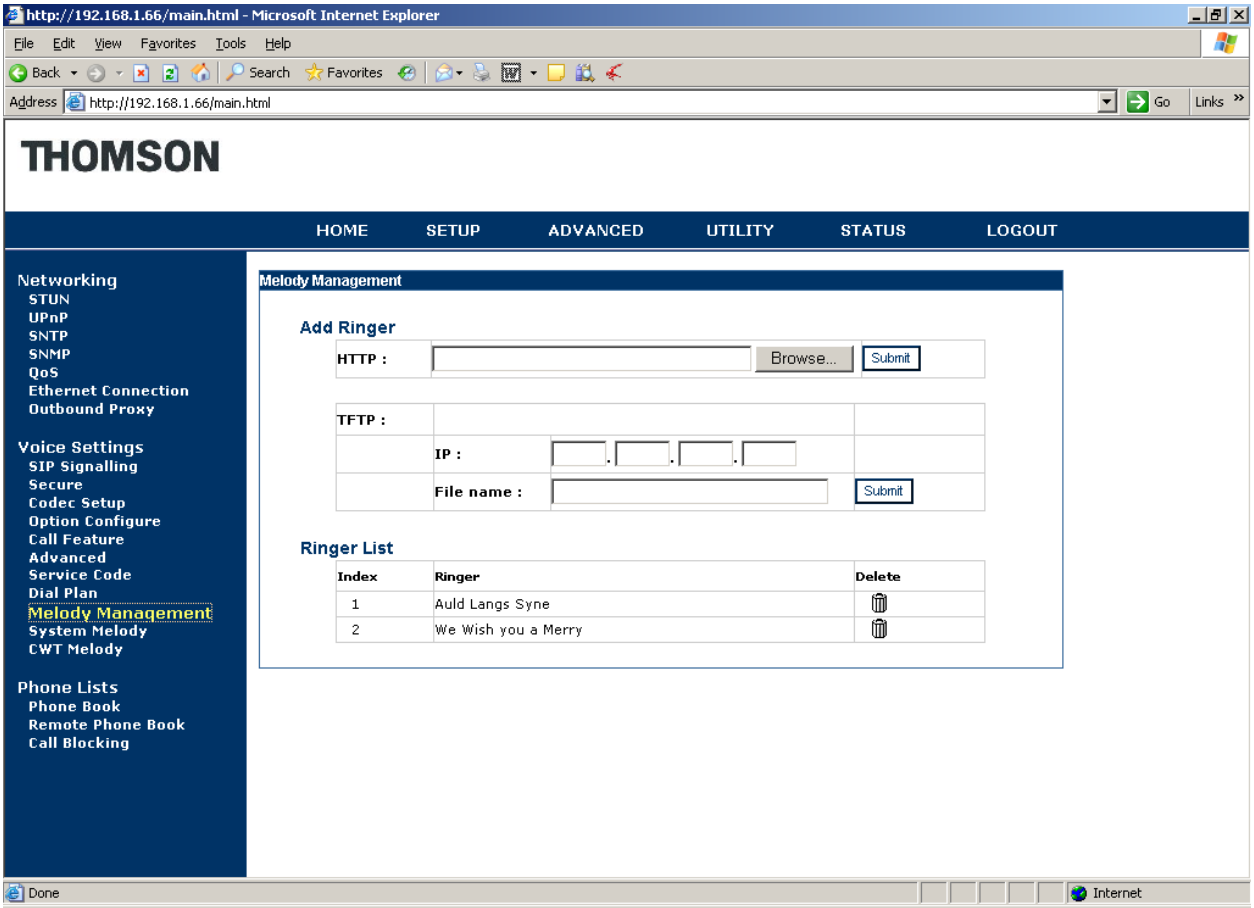
Advanced

The Advanced section lets you configure advanced features including networking voice settings, and phone list.

	Networking
STUN	Configure STUN server settings.
UPnP	Configure UPnP settings.
SNTP	Configure SNTP to configure time server on Internet.
QoS	Configure quality of service settings.
Ethernet Connection	Configure LAN's and PC's transmission.
Outbound Proxy	Configure Outbound Proxy settings.

	Voice Settings
SIP Signalling	Configure the SIP signalling parameters.
Codec Setup	Select to setup your prefered Codec.
Option Configure	Configure call feature function to visible or invisible that effect.
Call Feature	Select the telephone operations of call feature.
Advanced	Configure advanced telephone settings.
Dial Plan	Configure the dial plan.
Melody Management	Add/delete your ring tone

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.



Melody Management

Add Ringer

HTTP :

TFTP :

IP : . . .

File name :

Ringer List

Index	Ringer	Delete
1	Auld Langs Syne	🗑
2	We Wish you a Merry	🗑

Part 9 Feature Overview

This part is to show the number of new features implemented in each software new release, up to 2.68.

Note

This administrator guide is based on firmware v.2.68, you can download the latest administrator guide on:
www.thomsonbroadbandpartner.com

ST20XX SIP New Features (SG vx.68.5)

Overview

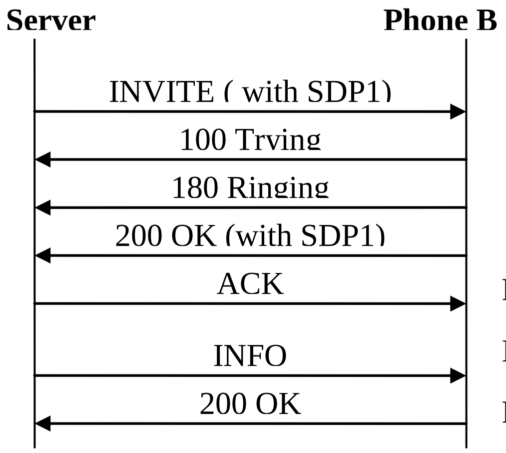
This document describes a set of features included in ST2030 and S2022 SIP V2.68.5 and 4.68.5 respectively in order to improve their usability in different environments.

Hua Wei SoftSwitch

Starting with this new release, ST20xx have implement Softswitch (Hua Wei SoftX3000 V300R006) PSTN call handling including call waiting and switching.

Flow descriptions:

Below is to list out a flow which shows PSTN 1 call to IP phone B in conversation, then PSTN 2 call to IP phone B.



PSTN1 and phone B are in conversation

INFO: tone-type, duration, gap, times

Phone will play the beep sound

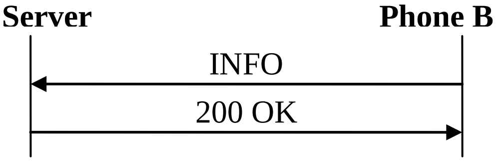
[**Answer**] and [**Reject**] softkey displays on the screen.

ST2030s (Phone B)

Date	Hour	Icons
01 PSTN Call		
Answer	**Reject**	

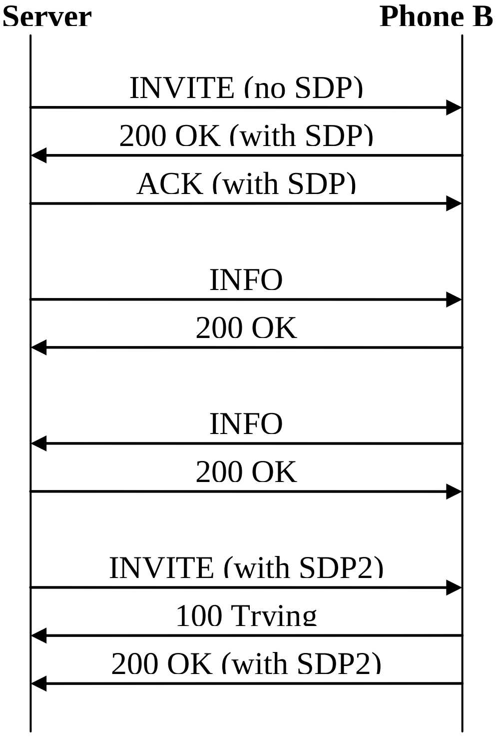
ST2022s (Phone B)

01		
Answer	**Reject**	



Press [Answer] to take the PSTN 2 call.
INFO: flashhook when press [Answer]

Remark: Go back to previous screen when [**Reject**] is pressed.



PSTN1 is on hold and B stops the call waiting tone

INFO: tone-type, duration, gap, times

Phone sends INFO to take the call, INFO: digit =2

Phone is able to talk with PSTN 2

ST2030s (Phone B)

Date	Hour	Icons
01 PSTN Call/phone number		
Taking		Duration
Hold	**Switch**	**Conf**

ST2022s (Phone B)

01 PSTN Call/phone number		
Talking		Duration
Hold	**Switch**	**Conf**

Remark: Pressing [**Hold**], [**Conf**] and [**Transfer**] will keep the same as current implementation.

Switch implementation

Requirement

- [**Switch**] softkey is required to caller parties to switch between caller during conversation

Below is to list out a flow which shows IP Phone B call to IP phone A in conversation, then PSTN 1 call to IP phone B. B and PSTN 1 are in conversation. Phone B presses [**Switch**] back to phone A and start conversation.

Flow descriptions:

Phone A ← → Phone B, operate in normal SIP protocol

PSTN 1 → phone B, call waiting tone is played, same as the call waiting flow above → call activated

Phone B presses [**Switch**] to talk back with phone A

(PSTN 1 and Phone B are in conversation)

ST2030s (Phone B)

Date	Hour	Icons
01 PSTN Call/phone number		
Taking		Duration
Hold	**Switch**	**Conf**

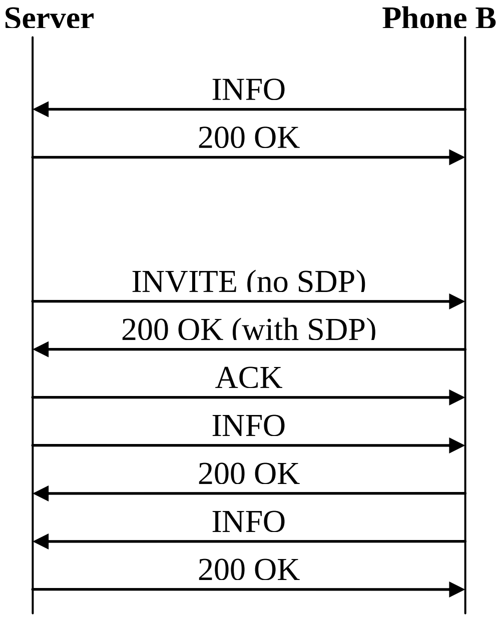
Transf >

ST2022s (Phone B)

01 PSTN Call/phone number		
Talking		Duration
Hold	**Switch**	**Conf**

Transf >

When [**Switch**] is pressed, switching is processed.

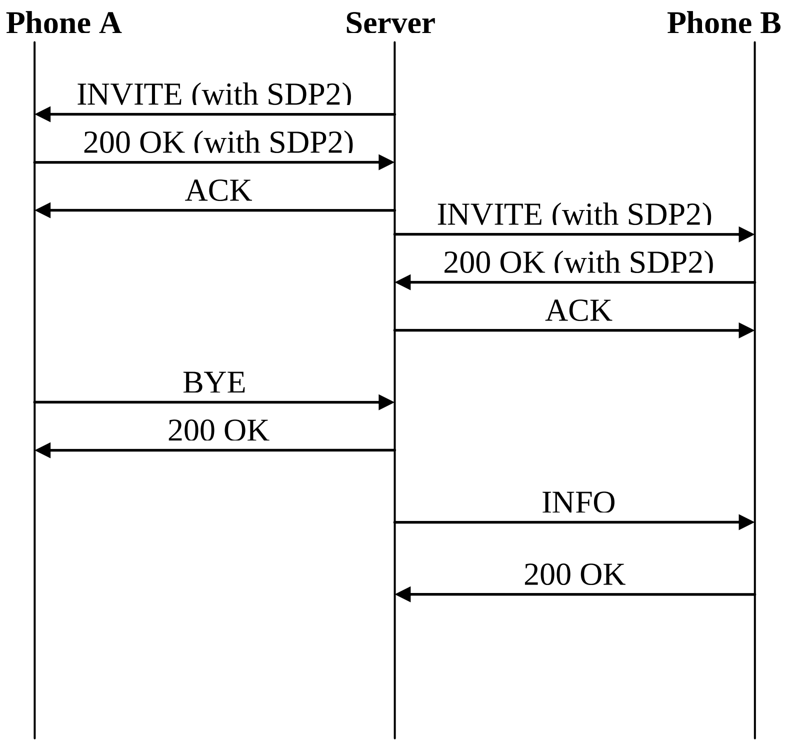


Phone B press [Switch] to start switching to other party
INFO: flashhook

PSTN1 is on hold and B stops the call waiting tone

INFO: Tone-type=specialdial, dial-timer=0

Phone send “2” to enable the take the call,
INFO: digit =2



Server invites phone A

Server invites phone B

Phone A hang up the call

INFO: tone-type=busy
Phone B know that the waiting phone is hang up and the [Switch] soft key should be changed back to [Transf]

ST2030s (Phone B)

Date	Hour	Icons
01 PSTN Call/phone number		
Taking		Duration
Hold	**Transf**	**Conf**

ST2022s (Phone B)

01 PSTN Call/phone number		
Talking		Duration
Hold	**Transf**	**Conf**

Feature Activation

A. Via APS

APS in [sys] section of both Common and Specific-MAC config files with the new parameter – **MGC_service**.

MGC_service = **0** disable (default)
MGC_service = **1** Hua Wei Softswitch PSTN handling

```
[sys]
…
MGC_service=0                              // Disable (default)
…
```

B. *Via Telnet*

To configure, open a command line console, and telnet the phone:

Example:

1. MGC_service is set to disable (**sys set mgc_service 0**)

```
[administrator]# sys set mgc_service 0
[OK] Set OK

[administrator]#
```

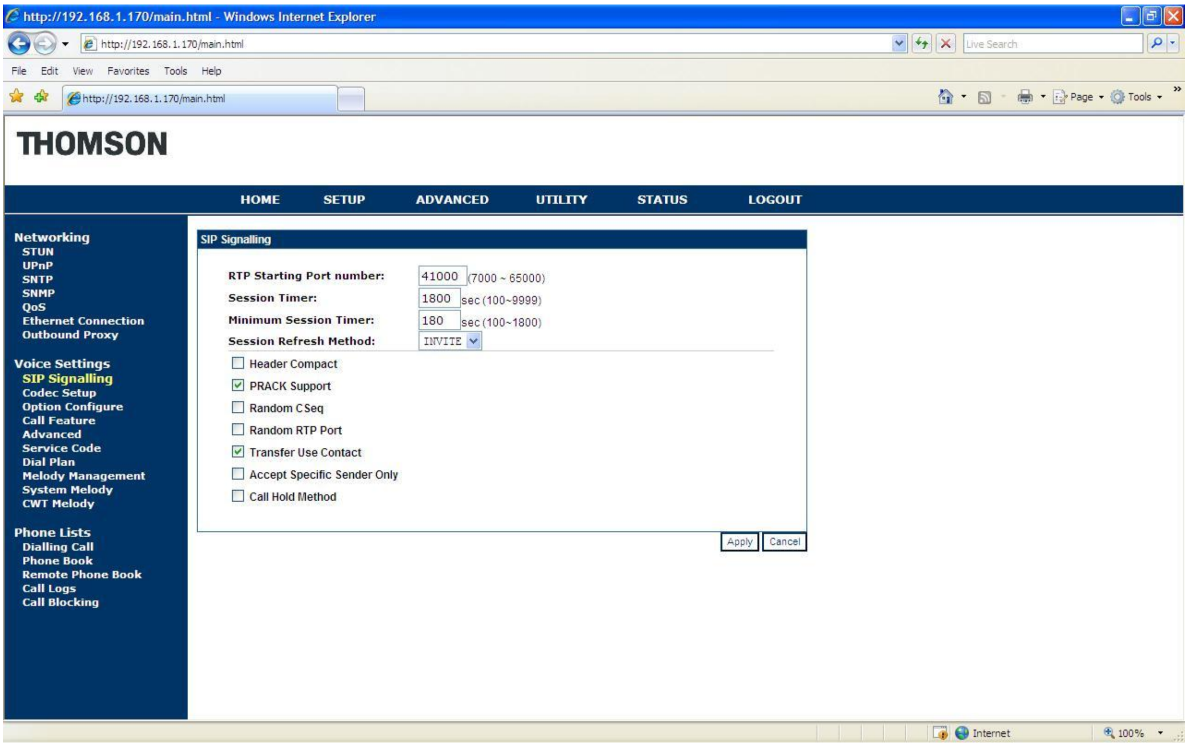
2. MGC_service is set to Hua Wei Softswitch PSTN handling (sys set mgc_service 1)

```
[administrator]# sys set mgc_service 1
[OK] Set OK

[administrator]#
```

Then type **commit** and then **activate** to apply the change.

Config without reload

Starting from this version changing config/setting in SIP Signalling, Option Configure, Call feature and Advanced Web page, ST20XX do not need to reboot.

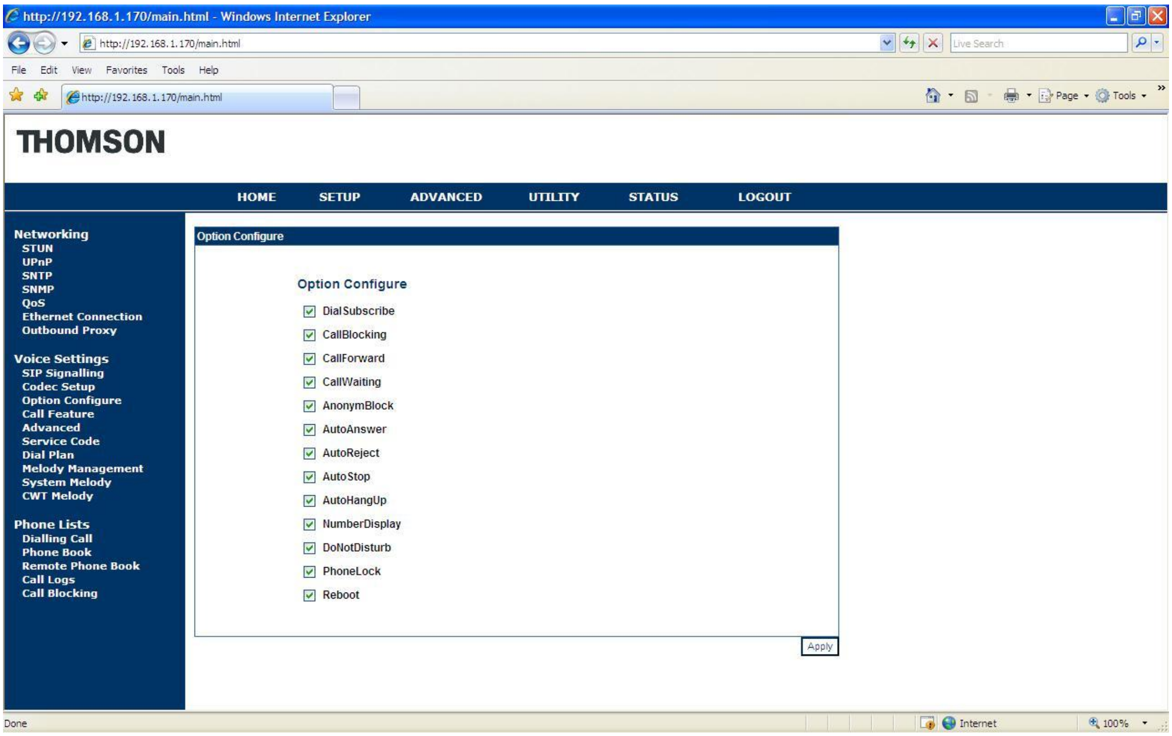


SIP Signalling

RTP Starting Port number: 41000 (7000 ~ 65000)
Session Timer: 1800 sec (100~9999)
Minimum Session Timer: 180 sec (100~1800)
Session Refresh Method: INVITE

☐ Header Compact
☑ PRACK Support
☐ Random CSeq
☐ Random RTP Port
☑ Transfer Use Contact
☐ Accept Specific Sender Only
☐ Call Hold Method

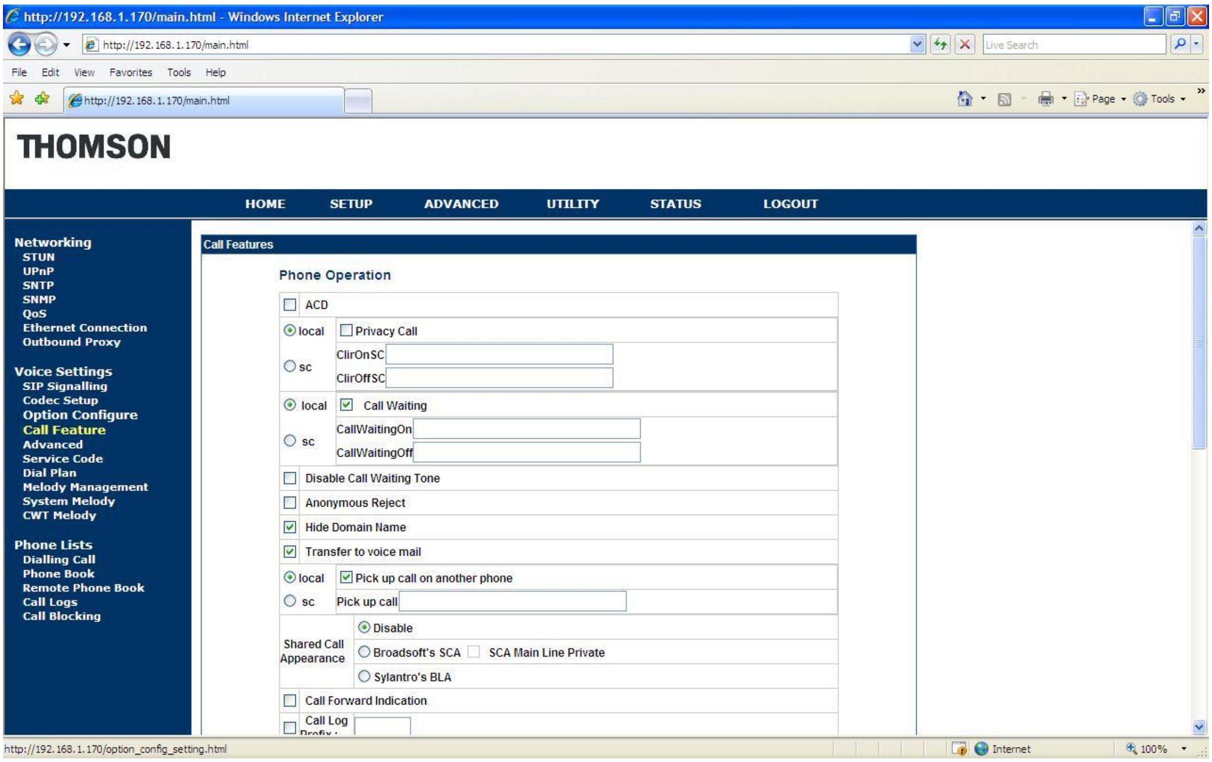
Except "RTP Starting Port Number" and "Minimum Session Timer", all parameter changed in SIP Signalling page are configured without reload.



Option Configure

☑ DialSubscribe
☑ CallBlocking
☑ CallForward
☑ CallWaiting
☑ AnonymBlock
☑ AutoAnswer
☑ AutoReject
☑ AutoStop
☑ AutoHangUp
☑ NumberDisplay
☑ DoNotDisturb
☑ PhoneLock
☑ Reboot

All parameters changed in Option Configure page are configured without reload.



Phone Operation

☐ ACD

● local ☐ Privacy Call
○ sc ClirOnSC ClirOffSC

● local ☑ Call Waiting
○ sc CallWaitingOn CallWaitingOff

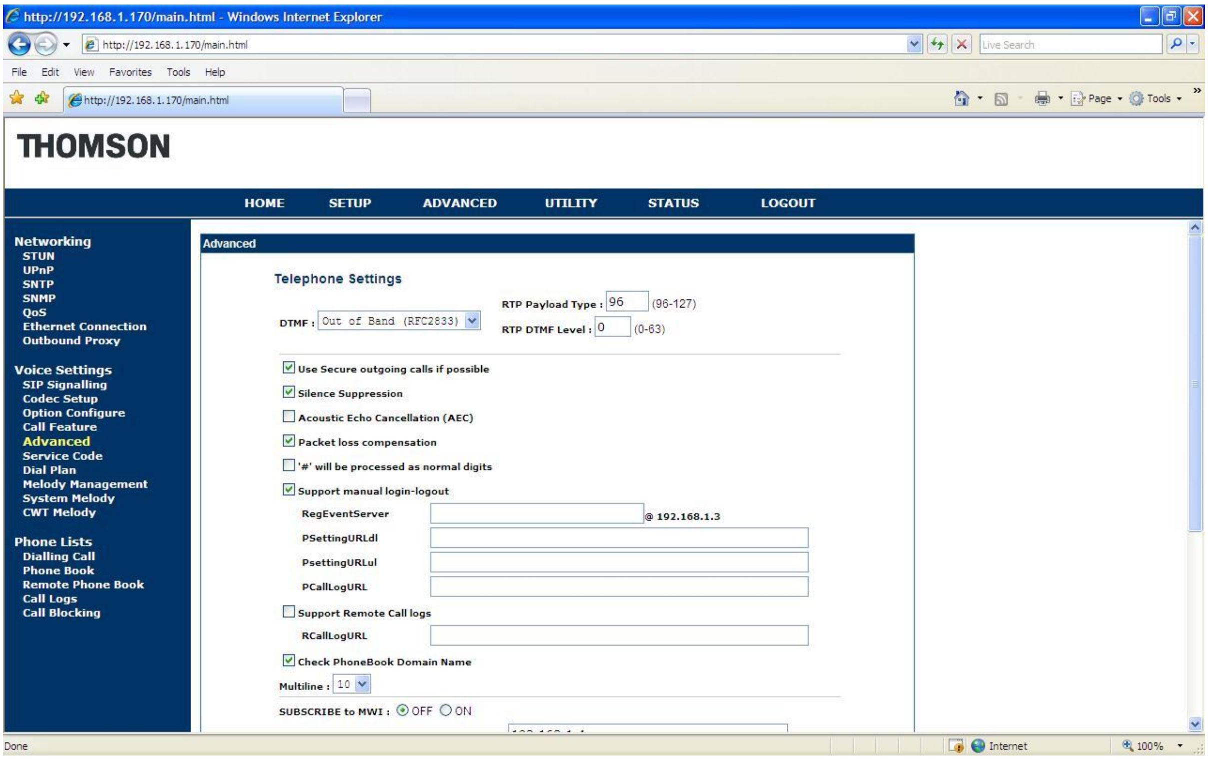
☐ Disable Call Waiting Tone
☐ Anonymous Reject
☑ Hide Domain Name
☑ Transfer to voice mail

● local ☑ Pick up call on another phone
○ sc Pick up call

Shared Call Appearance: ● Disable ○ Broadsoft's SCA ☐ SCA Main Line Private ○ Sylantro's BLA

☐ Call Forward Indication
☐ Call Log Prefix :

Except “ACD”, “Shared Call Appearance”, “Conference Mode”, “Start Spare FK” and “BLF Type”, all parameters changed in Call Features page are configured without reload.



Advanced

Telephone Settings

DTMF : Out of Band (RFC2833)

RTP Payload Type : 96 (96-127)

RTP DTMF Level : 0 (0-63)

☑ Use Secure outgoing calls if possible

☑ Silence Suppression

☐ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☐ '#' will be processed as normal digits

☑ Support manual login-logout

RegEventServer @ 192.168.1.3

PSettingURLdl

PsettingURLul

PCallLogURL

☐ Support Remote Call logs

RCallLogURL

☑ Check PhoneBook Domain Name

Multiline : 10

SUBSCRIBE to MWI : ● OFF ○ ON

Apart from “Use Secure outgoing calls if possible”, “Silence Suppression”, “Acoustic Echo Cancellation (AEC)”, “Packet Loss Compensation”, all parameters changed in Advance page are configured without reload.

CCBS Enable/Disable

A new parameter is added for CallBack feature. This parameter is configurable in common config file and telnet.

auto_cb = 0/1

// 1: Enable CallBack feature

Date	Hour	Icons
Remote Unavailable!		
XXXXXXXX		
CBack		**Back**

// 0: Disable CallBack feature

Date	Hour	Icons
Remote Unavailable!		
XXXXXXXX		
		Back

The default value is 1.

C. *Via APS*

APS in [ipp] section of both Common and Specific-MAC config files with the new parameter – **auto_cb**.

```
 [ipp]
…
auto_cb=1                                    // Enable (default)
…
```

D. *Via Telnet*

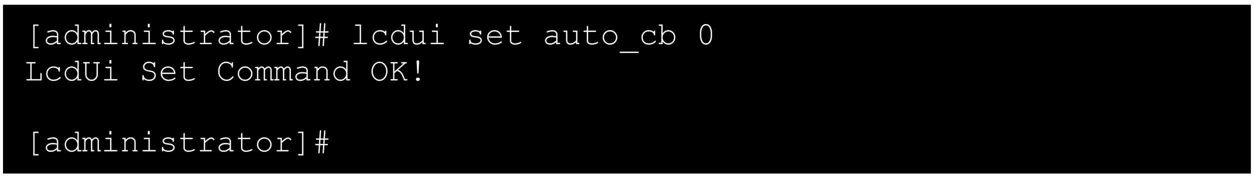
To configure, open a command line console, and telnet the phone:

Example:

1. Auto_cb is set to enable (**lcdui set auto_cb 1**)

```
[administrator]# lcdui set auto_cb 1
LcdUi Set Command OK!

[administrator]#
```

2. Auto_cb is set to disable. (**lcdui set auto_cb 0**)



```
[administrator]# lcdui set auto_cb 0
LcdUi Set Command OK!

[administrator]#
```

Then type **commit** and then **activate** to apply the change.

ST20XX SIP New Features (SG vx.67.2)

Overview

This document describes a set of features included in ST2030 and S2022 SIP V2.67.2 and 4.67.2 respectively in order to improve their usability in different environments.

Security Features

Starting with the new release, **SG x.67.2**, ST20xx have done a lot of improvements on the security implementation. In this section, you will know more about the operation of the features of SIPS/TLS/SRTP on the phone.

Overview:

Basically, SIPS/TLS/SRTP can be enabled/disabled separately by configuration.

Enabling means the feature is default option for outgoing actions only, and switching on-off mechanism is supported according to capability of remote side, no matter the feature is enabled or disabled.

SIPS

1. Switching to SIPS if receiving SIP message contained uri type (sips:)
2. Switching to SIPS if outgoing uri type is (sips:): for example: speed dial a SIPS uri etc…

TLS

3. TLS would be activated together with SIPS

4. The cipher-suites of TLS is supported by OpenSSL library(0.9.7d), and cipher-suites negotiation is supported according to capability of remote side
5. Switching transport layer to TLS while receiving SIP messages with any one of following content:
   - Transport parameter(transport=tls) in Contact and Record-Route header etc…
   - Uri type (sips:)

4. Support the Import/Export certification and private key via Web interface

SRTP

6. Switching to RTP or SRTP according to capability of remote side described on SDP.

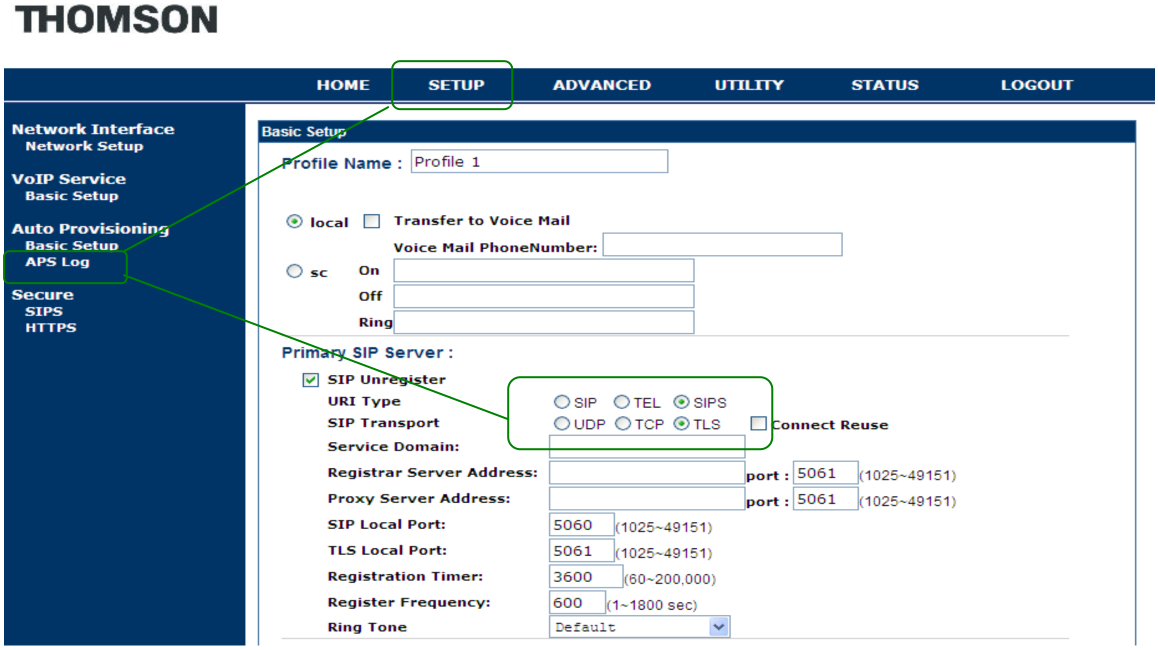
7. Supported crypto-suites:
   - AES_CM_128_HMAC_SHA1_32
   - F8_128_HMAC_SHA1_80
   - AES_CM_128_HMAC_SHA1_80

8. Crypto-suites negotiation is supported according to capability of remote side described on SDP

Feature Activation

E. <u>Via WebGui</u>

Visit the SETUP → Basic Setup → select Profile page, as shown like below:



Profile Name : Profile 1

● local ☐ Transfer to Voice Mail

Voice Mail PhoneNumber:

○ sc On

Off

Ring

Primary SIP Server :

☑ SIP Unregister

URI Type ○ SIP ○ TEL ● SIPS

SIP Transport ○ UDP ○ TCP ● TLS ☐ Connect Reuse

Service Domain:

Registrar Server Address: port : 5061 (1025~49151)

Proxy Server Address: port : 5061 (1025~49151)

SIP Local Port: 5060 (1025~49151)

TLS Local Port: 5061 (1025~49151)

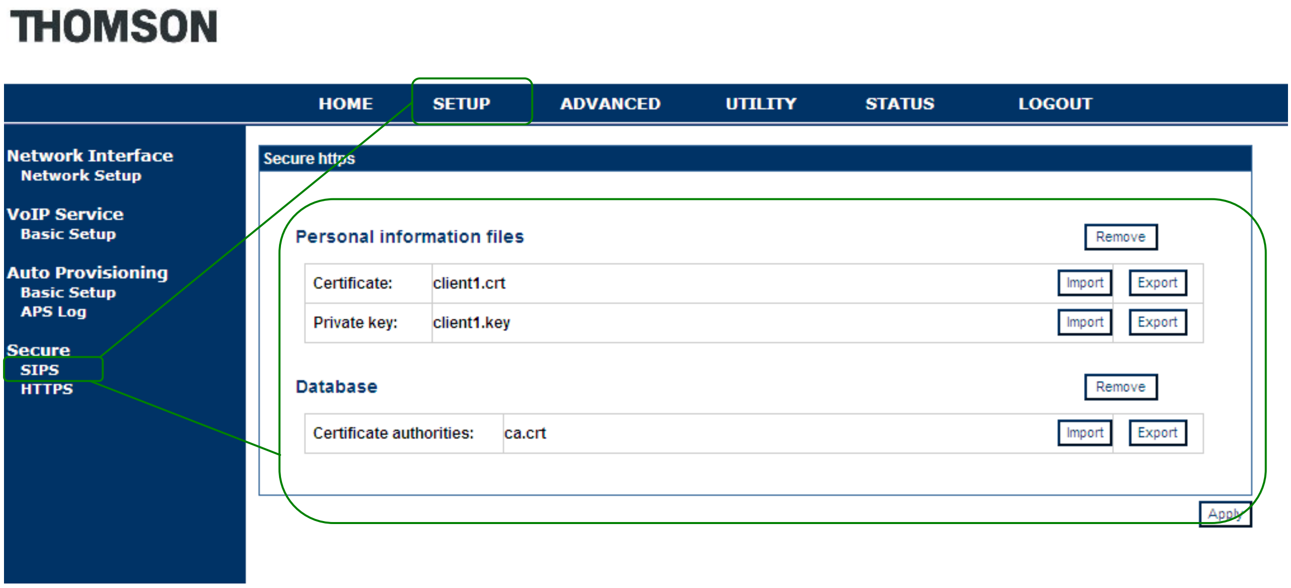
Registration Timer: 3600 (60~200,000)

Register Frequency: 600 (1~1800 sec)

Ring Tone Default

With the activation of **SIPS + TLS**, you are required to upload a Certificate and Private Key for authentication on the phone. Set port to be **5061.** The certificate from trusted authorities can be imported to create an internal Certificate Authority (CA).

Visit the SETUP → Basic Setup → SIPS page, as shown like below:



Certificate: client1.crt

Private key: client1.key

Certificate authorities: ca.crt

F. *Via APS*

APS in [sip] section of both Common and Specific-MAC config files with the new parameters:

For **SIPS**, the parameters are **URLTypeMP**[**profile_id**] = and **URLTypeBk**[**profile_id**] = ,
where, 0 – sip URL type, (default)
1 – tel URL type,
2 – sips URL type

```
[sip]
…
URLTypeMP1=0                    // sip url type for profile 1
URLTypeMP2=1                    // tel url type for profile 2
URLTypeMP3=2                    // sips url type for profile 3
URLTypeMP4=0                    // sip url type  for profile 4

URLTypeBK1=0                    // sip url type for backup profile 1
URLTypeBK2=1                    // tel url type for backup profile 2
URLTypeBK3=2                    // sips url type for backup profile 3
URLTypeBK4=0                    // sip url type  for backup profile 4
…
```

For **TLS**, the parameters are **TransportFlgMP**[**profile_id**] = and **TransportFlgBK**[**profile_id**] = ,
where, 0 – UDP (default)
1 – TCP
2 – TLS

```
[sip]
…
TransportFlgMP1=0                   // UDP for profille 1
TransportFlgMP2=1                   // TCP for profille 2
TransportFlgMP3=2                   // TLS for profille 3
TransportFlgMP4=2                   // TLS for profille 4

TransportFlgBK1=0                   // UDP type for backup profile 1
TransportFlgBK2=1                   // TCP type for backup profile 2
TransportFlgBK3=2                   // TLS type for backup profile 3
TransportFlgBK4=2                   // TLS type for backup profile 4
…
```

APS in [sys] section of both Common and Specific-MAC config files with the new parameters:

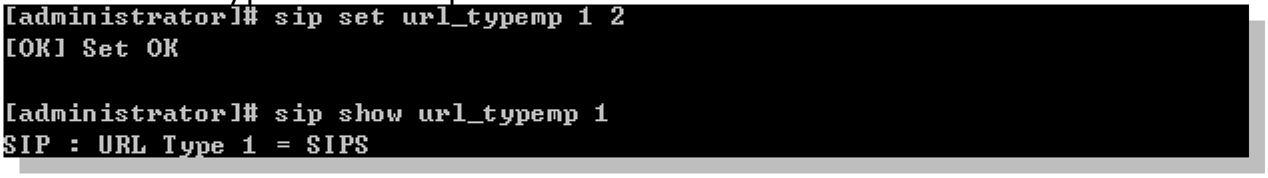
For **SRTP**, the parameter is **SRTPFlag= ,** where 0 disable (default), 1 - enable.

```
[sys]
…
SRTPFlag=1                     // Disable (default)
…
```

G. Via Telnet

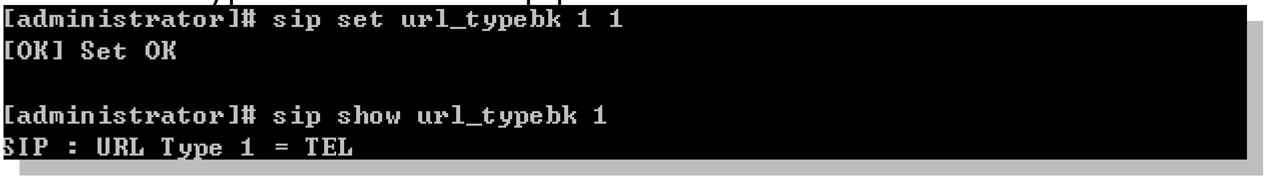
To configure, open a command line console, and telnet the phone:
Example:

3. SIPS URL type is set for profile 1



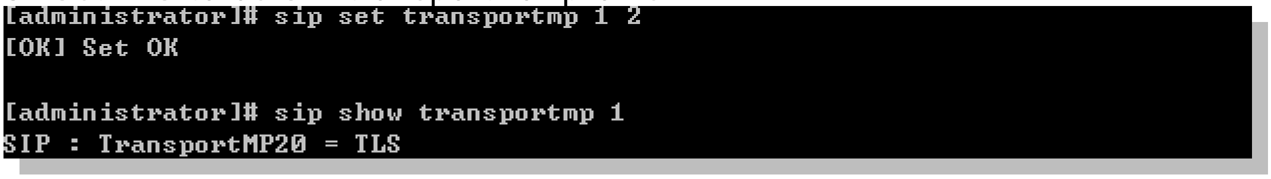
```
[administrator]# sip set url_typemp 1 2
[OK] Set OK

[administrator]# sip show url_typemp 1
SIP : URL Type 1 = SIPS
```

4. TEL URL type is set for backup profile 1



```
[administrator]# sip set url_typebk 1 1
[OK] Set OK

[administrator]# sip show url_typebk 1
SIP : URL Type 1 = TEL
```

5. Set TLS to be SIP transport for profile 1



```
[administrator]# sip set transportmp 1 2
[OK] Set OK

[administrator]# sip show transportmp 1
SIP : TransportMP20 = TLS
```

6. Set SRTP ON

```
[administrator]# sys set srtp 1
[OK] Set OK

[administrator]# sys show srtp
SRTP : ON
```

Improve HTTPS APS feature

Now, starting with the new release, SG x.67.2, St20xx HTTPS auto provisioning supports both client and server authentication with using certificates. This improvement is totally enhanced a reliable and secure provisioning based on HTTPS requests from ST20xx to server and server to ST20xx.

With using HTTPS, the file transfer between ST20xx and the provisioning server are protected. The encryption method for the data between ST20xx and server is based on symmetric key cryptography. This methodology makes use of a secret key, which is shared between ST20xx and the provisioning server over a secure channel. It ensures that the server and St20xx cannot be interfered by other devices on the network.

Moreover, server and client authentication is performed using public/private key encryption, with certificates containing the public key. Files encrypted with a public key can be decrypted only by its corresponding private key (and vice verse).

In server side, you should have an SSL server certificate. ST20xx will recognize only the certificate in the server side and try to authenticate the certificate when connecting via HTTPS, or reject any server certificate if not valid.

Likewise, in client side, St20xx should also carry a unique client certificate which is used to identify information about each individual server.

Therefore, before doing HTTPS APS, the phone needs to have the certificate and the private key imported via webGui. And the server certificate should also be ready as well.

Feature Activation

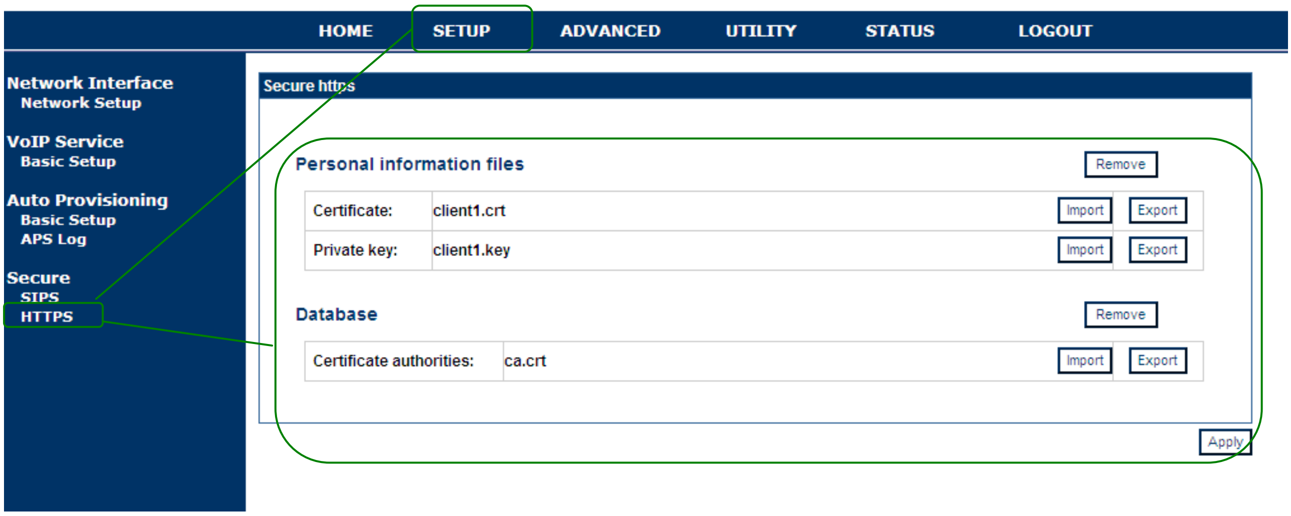
A. Via WebGui

To import the client certificate in ST20xx, you can visit the SETUP → Basic Setup → HTTPS page, as shown like below:

Click the import button to upload the certificate and private key files.
Click the import button to upload the certificate authority root certificate.

You are also able to export the certificate by clicking Export.

Finally, press “Apply” to save the change.



Secure https

Personal information files

Certificate: client1.crt

Private key: client1.key

Database

Certificate authorities: ca.crt

Support Encryption of Configuration files in APS

In current implementation, file transfer via APS TFTP and HTTP is transparent, the confidential information, e.g. accounts, password, are not protected.

Therefore, starting with the new release, x.67.2, the transfer of encrypted configuration files is supported to prevent data exposure during APS.

Encrypted files support:

- Common configuration file (Comconfig file)
- MAC configuration file (MacConfig file)

Encryption method:
Symmetric method, advance encryption standard (AES) 128bit

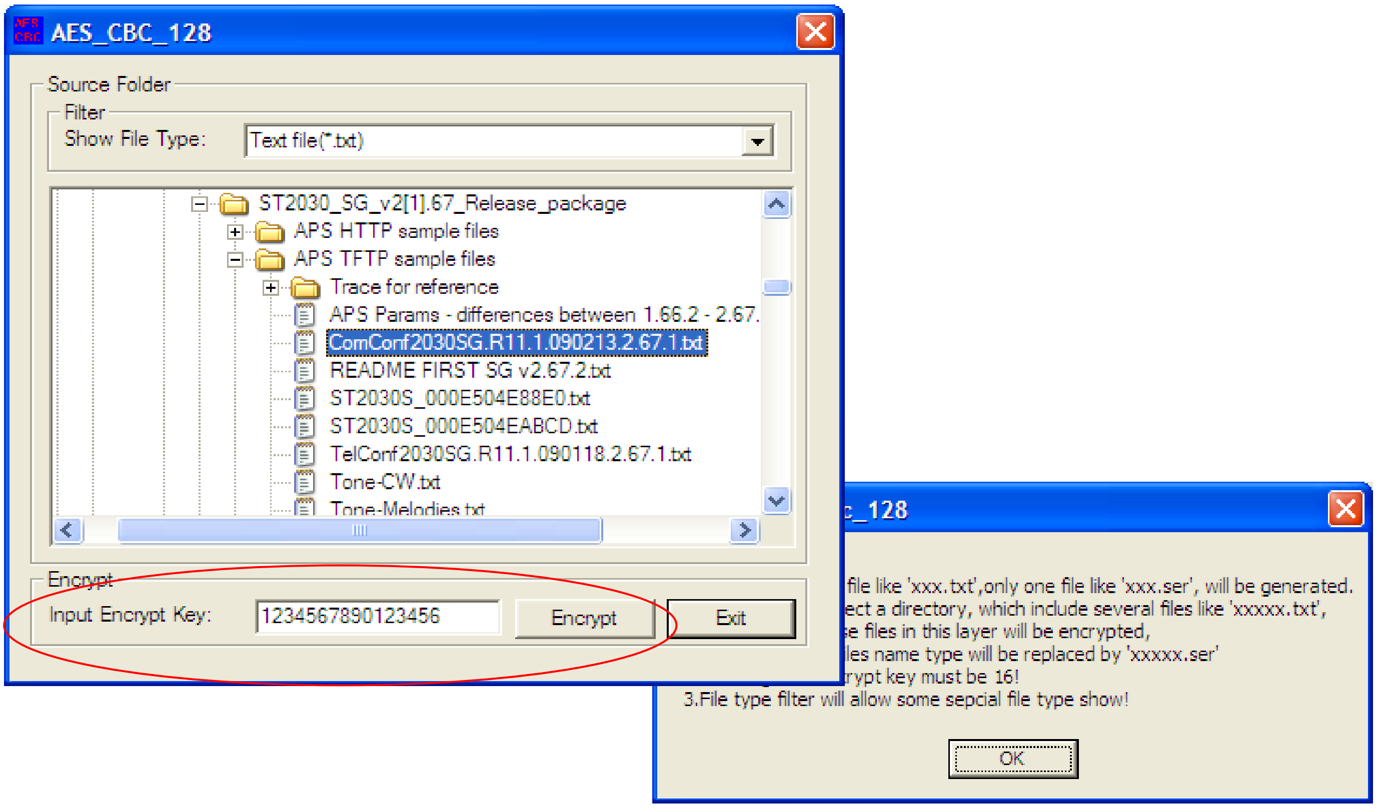
Encryption/Decryption tools:

- Name: Window_Aes_cbc_128.exe (including in release package/ Decryption tool folder)
- Press F1 for get HELP window

<u>In Window Platform</u>

File name: Window_Aes_cbc_128.exe in ../Encryption Tool folder.

Select the Configuration file and input a 16-digits key for encryption.
Press [Encrypt] to start the process. Press 'F1' to get help window for more information.



AES_CBC_128

Source Folder

Filter

Show File Type: Text file(*.txt)

```
ST2030_SG_v2[1].67_Release_package
    APS HTTP sample files
    APS TFTP sample files
        Trace for reference
        APS Params - differences between 1.66.2 - 2.67.
        ComConf2030SG.R11.1.090213.2.67.1.txt
        README FIRST SG v2.67.2.txt
        ST2030S_000E504E88E0.txt
        ST2030S_000E504EABCD.txt
        TelConf2030SG.R11.1.090118.2.67.1.txt
        Tone-CW.txt
        Tone-Melodies.txt
```

Encrypt

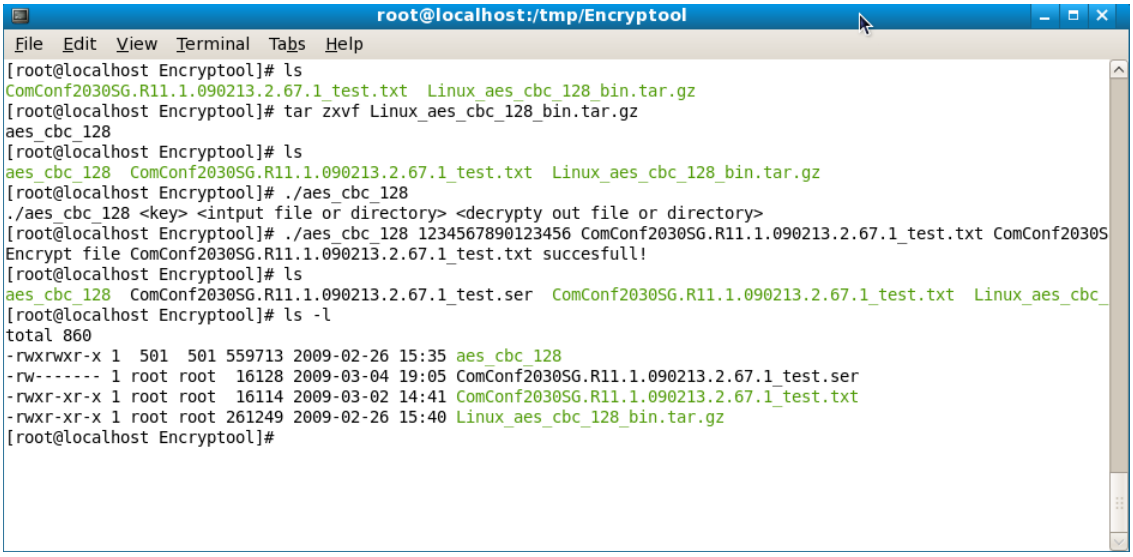
Input Encrypt Key: 1234567890123456

file like 'xxx.txt',only one file like 'xxx.ser', will be generated.
ect a directory, which include several files like 'xxxxx.txt',
e files in this layer will be encrypted,
les name type will be replaced by 'xxxxx.ser'
rypt key must be 16!
3.File type filter will allow some sepcial file type show!

In Linux Platform:

Filename: Linux_aes_cbc_128_bin.tar.gz ../Encryption Tool folder.

- ➔ tar zxvf FileName.tar.gz
- ➔ cd FileName
- ➔ ./aes_cbc_128 <key> <source file or directory> <destination file or directory>
- ➔ '*.ser' file will be generated



```
[root@localhost Encryptool]# ls
ComConf2030SG.R11.1.090213.2.67.1_test.txt  Linux_aes_cbc_128_bin.tar.gz
[root@localhost Encryptool]# tar zxvf Linux_aes_cbc_128_bin.tar.gz
aes_cbc_128
[root@localhost Encryptool]# ls
aes_cbc_128  ComConf2030SG.R11.1.090213.2.67.1_test.txt  Linux_aes_cbc_128_bin.tar.gz
[root@localhost Encryptool]# ./aes_cbc_128
./aes_cbc_128 <key> <intput file or directory> <decrypty out file or directory>
[root@localhost Encryptool]# ./aes_cbc_128 1234567890123456 ComConf2030SG.R11.1.090213.2.67.1_test.txt ComConf2030S
Encrypt file ComConf2030SG.R11.1.090213.2.67.1_test.txt succesfull!
[root@localhost Encryptool]# ls
aes_cbc_128  ComConf2030SG.R11.1.090213.2.67.1_test.ser  ComConf2030SG.R11.1.090213.2.67.1_test.txt  Linux_aes_cbc_
[root@localhost Encryptool]# ls -l
total 860
-rwxrwxr-x 1  501  501 559713 2009-02-26 15:35 aes_cbc_128
-rw------- 1 root root  16128 2009-03-04 19:05 ComConf2030SG.R11.1.090213.2.67.1_test.ser
-rwxr-xr-x 1 root root  16114 2009-03-02 14:41 ComConf2030SG.R11.1.090213.2.67.1_test.txt
-rwxr-xr-x 1 root root 261249 2009-02-26 15:40 Linux_aes_cbc_128_bin.tar.gz
[root@localhost Encryptool]#
```

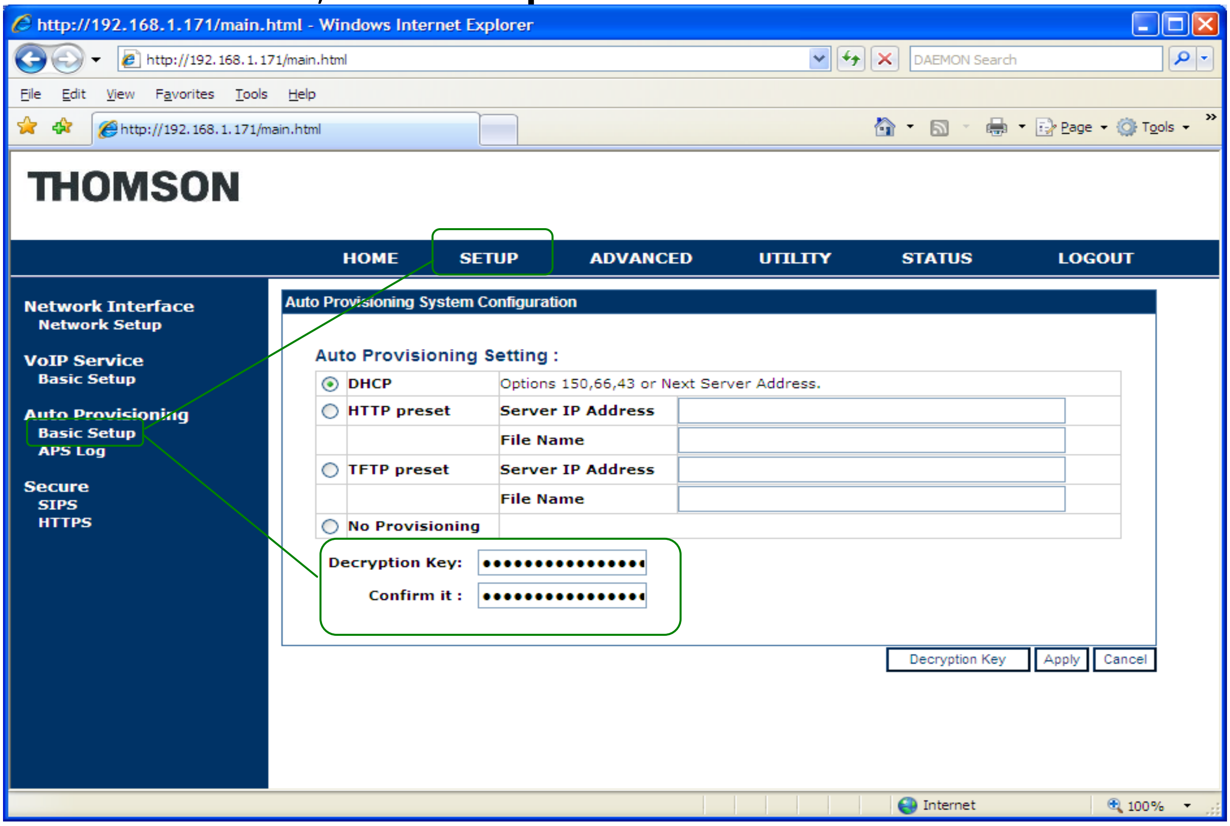
Things to know:

The ComConfig/ MacConfig file should be encrypted with symmetric key offline before doing APS. The decryption will be handled on the phone by using the same key.

Feature Activation

A. Via WebGui

Visit the **SETUP** → , **Basic Setup** will be shown like below:



Auto Provisioning System Configuration

Auto Provisioning Setting :

● DHCP	Options 150,66,43 or Next Server Address.	
○ HTTP preset	Server IP Address	
	File Name	
○ TFTP preset	Server IP Address	
	File Name	
○ No Provisioning		

Decryption Key: ••••••••••••••••

Confirm it : ••••••••••••••••

Input the decryption key (16 digits) in the field and confirm field. If the key is different with the key for confirmation, an error message will prompt you to input the key again.

Press the [**Decryption key**] to save the key settings in the phone.
→ Phone reboot.

If the key doesn't match with the key for encrypted files during APS, the files are not allowed to download to the phone.

You are recommended to check the **APS log** after doing APS to make sure its process is running correctly.

APS log (if Decryption key Match):

```
Begin Common Config download...
AES decrypt common config file successfully!
CommonConfig: ComConf2030SG.R11.1.090213.2.67.1.ser download successfully!
Begin Mac config download...
MacConfig: ST2030S_00147F00DFE4.ser download successfully!
AES decrypt mac file successfully!
Serial number is the same!
Begin upgrading config file...
Check config file syntax
Check config file syntax successfully!
Upgrading config file successfully!
```

APS log (if Decryption key not-Match):

```
Begin Common Config download...
AES decrypt common config file successfully!
CommonConfig: ComConf2030SG.R11.1.090213.2.67.1.ser download successfully!
Begin Mac config download...
MacConfig: ST2030S_00147F00DFE4.ser download successfully!
AES decrypt mac file successfully!
Error: Parse config file failed or config_sn tag not found!
Begin upgrading config file...
Check config file syntax
Error: Config file format error
```

On the **MMI** of the phone, you also be noted that the **Provision Error!** displaying on the screen when the .ser is failed to download.

B. Via APS

APS in [autoprovision] section of both Common and Specific-MAC config files with the new parameter:
Decryption_Key=Th0mson2$8s8@9z! (Default)

```
[autoprovision]
…
Decryption_Key=Th0mson2$8s8@9z!
…
```

Call Dialing from WebGui

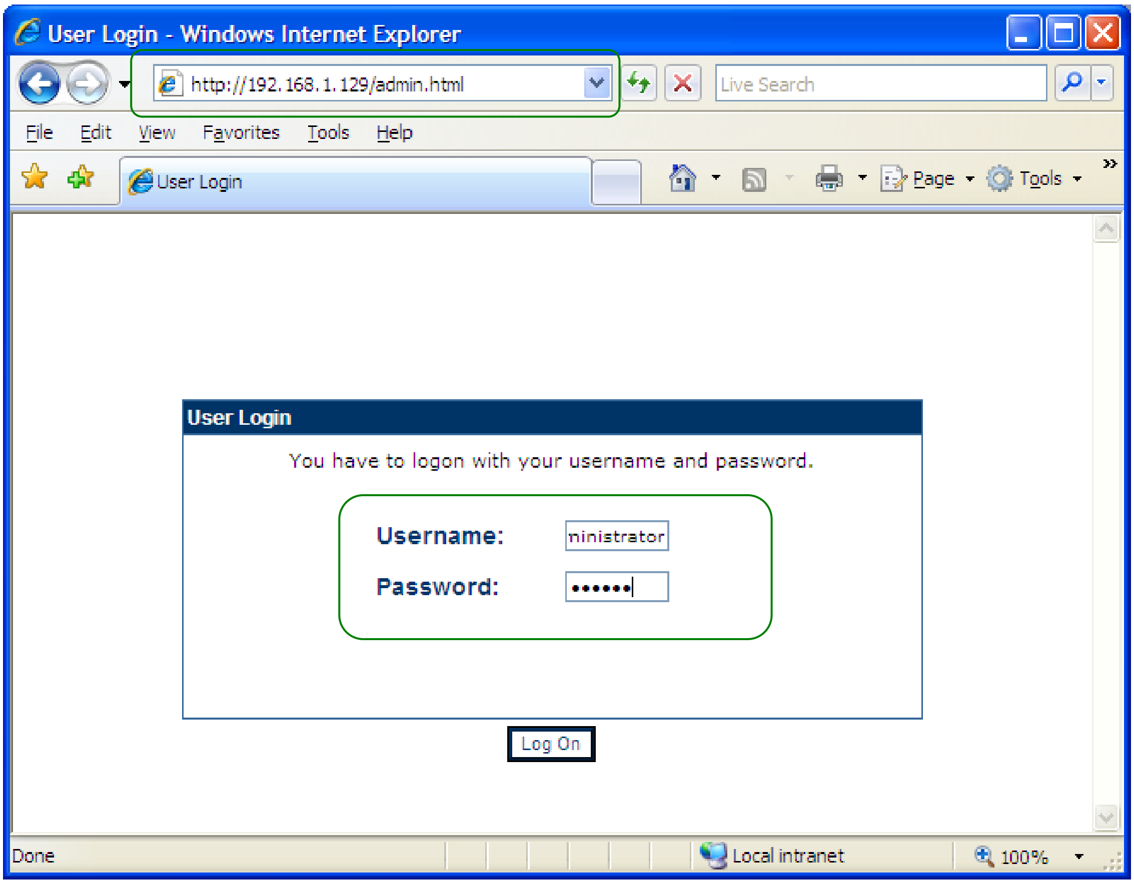
Currently, the feature of call dialing from web interface is not supported in ST20xx. For the new release today, x.67.2, the new feature is implemented. It provides the convenience for user to dial an outgoing call, reject and hang up incoming call on the web interface.

Feature Activation

Web interface

Enter the IP address of phone on the web browser
Enter the **Username** and **Password**

- Username(default): administrator
- Password(default) : 784518



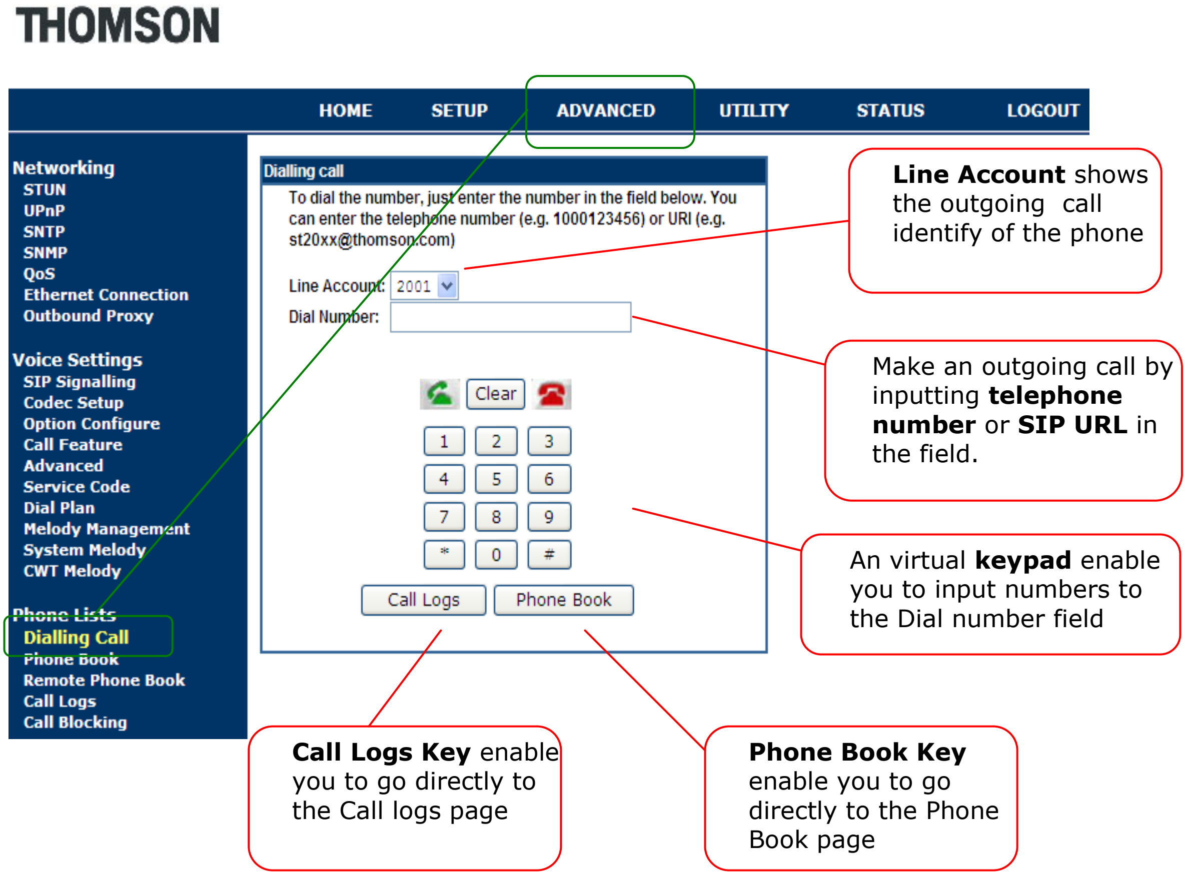
User Login

You have to logon with your username and password.

Username: ninistrator

Password: ••••••

Visit the ADVANCED → Dialling Call page, the Dialling Call interface will be shown like below:



Dialling call

To dial the number, just enter the number in the field below. You can enter the telephone number (e.g. 1000123456) or URI (e.g. st20xx@thomson.com)

Line Account: 2001

Dial Number:

Line Account shows the outgoing call identify of the phone

Make an outgoing call by inputting **telephone number** or **SIP URL** in the field.

An virtual **keypad** enable you to input numbers to the Dial number field

Call Logs Key enable you to go directly to the Call logs page

Phone Book Key enable you to go directly to the Phone Book page

To make an outgoing call, you can just choose the active outgoing identity from the pull-down menu of Line Account, and input the telephone numbers or SIP URL in the field. Click the ☎ icon to make an outgoing call. A status will be shown on the screen to indicate the call is successfully dialed out. Hang up the call by clicking the ☎ icon.

The Dial key ☎ has the following behaviors:

If the number is valid in the field,
1. In idle mode, dial out a call after pressing the key
2. Incoming call mode (Ringing), Line 2 will be used to dial out a call after pressing the key
3. Taking mode, the current call will be held and used Line 2 to dial out a call after pressing the key

No action is taken if the field is empty.

Also, the EndCall key ☎ has the following behaviors:

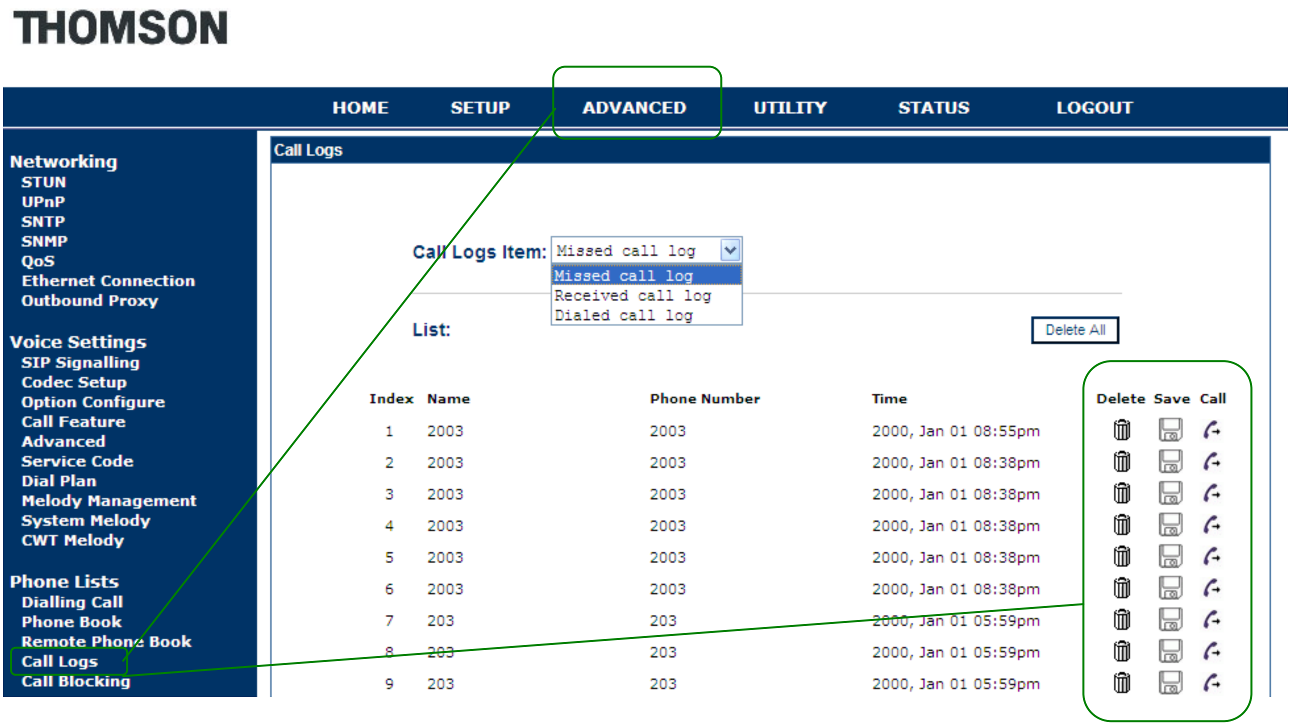
1. Dailing mode, stop the outgoing call and return to idle mode after pressing the key
2. Incoming call mode (Ringing), reject the call and return to idle mode after pressing the key
3. Taking mode, hang up the call after passing the key

Call Logs
Same as the local phone, there are 3 call logs supported on the web interface, including **Missed Call log**, **Received Call log** as well as the **Dialled Call log**.

You can press the Call logs key from Call dialling page or go to ADVANCED → Call Logs to access the page. Select the check logs by pulling-down the Call log menu.
The maxium entries in each call log are **30**.

In the call log page, you are allowed to delete entry, save entry and call back to the entry if you want.

- Click 🗑 to delete the entry
- Click 💾 to save the entry in phone book
- Click ↪ to access to the call dialling page to make an outgoing call



Call Logs Item: Missed call log

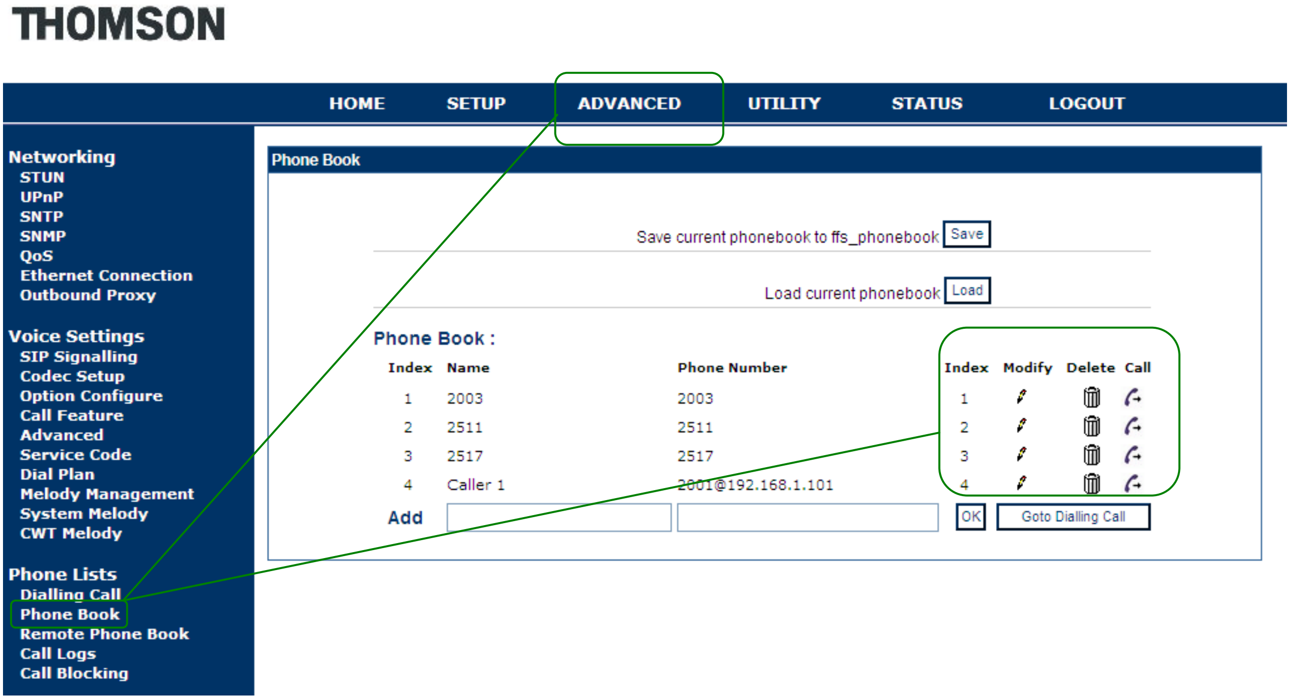
Index	Name	Phone Number	Time	Delete	Save	Call
1	2003	2003	2000, Jan 01 08:55pm	🗑	💾	☎
2	2003	2003	2000, Jan 01 08:38pm	🗑	💾	☎
3	2003	2003	2000, Jan 01 08:38pm	🗑	💾	☎
4	2003	2003	2000, Jan 01 08:38pm	🗑	💾	☎
5	2003	2003	2000, Jan 01 08:38pm	🗑	💾	☎
6	2003	2003	2000, Jan 01 08:38pm	🗑	💾	☎
7	203	203	2000, Jan 01 05:59pm	🗑	💾	☎
8	203	203	2000, Jan 01 05:59pm	🗑	💾	☎
9	203	203	2000, Jan 01 05:59pm	🗑	💾	☎

Phone Book

From the phone book page, the enhancement enable you to delete entry, modify entry and call back to the entries you want.

You can press the button from call dailling page or go to ADVANCED → Phone book to access the page.

- Click 🗑 to delete the entry
- Click ✎ to modify the entry's information
- Click ☎ to access to the call dialling page to make an outgoing call



Save current phonebook to ffs_phonebook

Load current phonebook

Phone Book :

Index	Name	Phone Number	Index	Modify	Delete	Call
1	2003	2003	1	✎	🗑	☎
2	2511	2511	2	✎	🗑	☎
3	2517	2517	3	✎	🗑	☎
4	Caller 1	2001@192.168.1.101	4	✎	🗑	☎

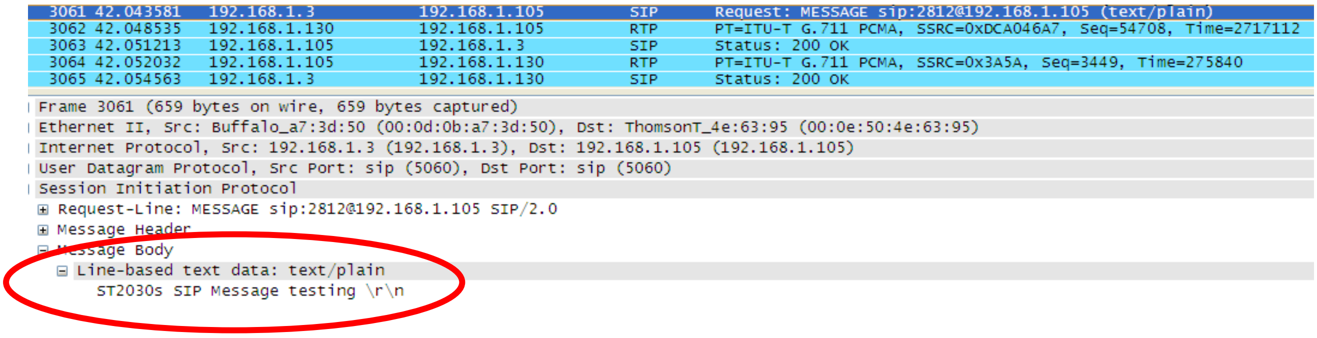
SIP Message during call (rfc 3428)

Currently, ST20xx supports SIP message method. But it is limited to in idle status. In other words, the phone can only receive short message pushed by server in idle mode to indicate its current status.

Now, starting from the new release, SG x.67.2, St20xx not only supports the receving message in idle mode, but also in conversation mode.

The existing parameter still keeps using with purpose to avoid attacks. Messages coming from other servers which were not the configured one on parameter will be rejected.

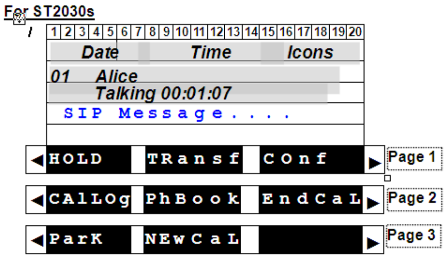
From the below network trace, it shows that the phone receives SIP message “ST2030s SIP Message testing” from server side.



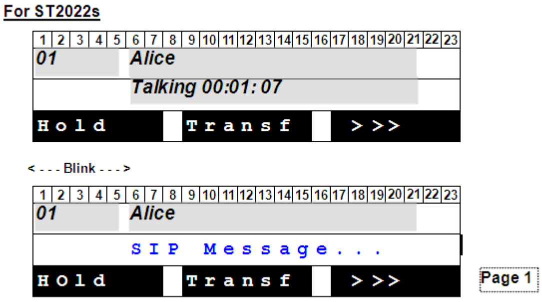
```
3061 42.043581   192.168.1.3        192.168.1.105     SIP    Request: MESSAGE sip:2812@192.168.1.105 (text/plain)
3062 42.048535   192.168.1.130      192.168.1.105     RTP    PT=ITU-T G.711 PCMA, SSRC=0xDCA046A7, Seq=54708, Time=2717112
3063 42.051213   192.168.1.105      192.168.1.3       SIP    Status: 200 OK
3064 42.052032   192.168.1.105      192.168.1.130     RTP    PT=ITU-T G.711 PCMA, SSRC=0x3A5A, Seq=3449, Time=275840
3065 42.054563   192.168.1.3        192.168.1.130     SIP    Status: 200 OK

Frame 3061 (659 bytes on wire, 659 bytes captured)
Ethernet II, Src: Buffalo_a7:3d:50 (00:0d:0b:a7:3d:50), Dst: ThomsonT_4e:63:95 (00:0e:50:4e:63:95)
Internet Protocol, Src: 192.168.1.3 (192.168.1.3), Dst: 192.168.1.105 (192.168.1.105)
User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
Session Initiation Protocol
 ⊞ Request-Line: MESSAGE sip:2812@192.168.1.105 SIP/2.0
 ⊞ Message Header
 ⊟ Message Body
   ⊟ Line-based text data: text/plain
       ST2030s SIP Message testing \r\n
```

The text contained in the SIP MESSAGE body is displayed on the 4th line of the LCD for ST2030 and on the 2nd line of the LCD on the ST2022, same as previously.

SIP MESSAGE text location on ST2030:



SIP MESSAGE text location on ST2022:



Feature Activation

Same as the current implementation. This feature does not need specific activation to be supported. With using the current parameter, it is to avoid messages attacks from non desired sources.

The parameter is AuthMessageServer. Default value is 0.0.0.0, which means phone accepts all messages received from everywhere. Otherwise, to limit from which server messages can be accepted, this parameter should contain either SIP messages server IP address or domain name.

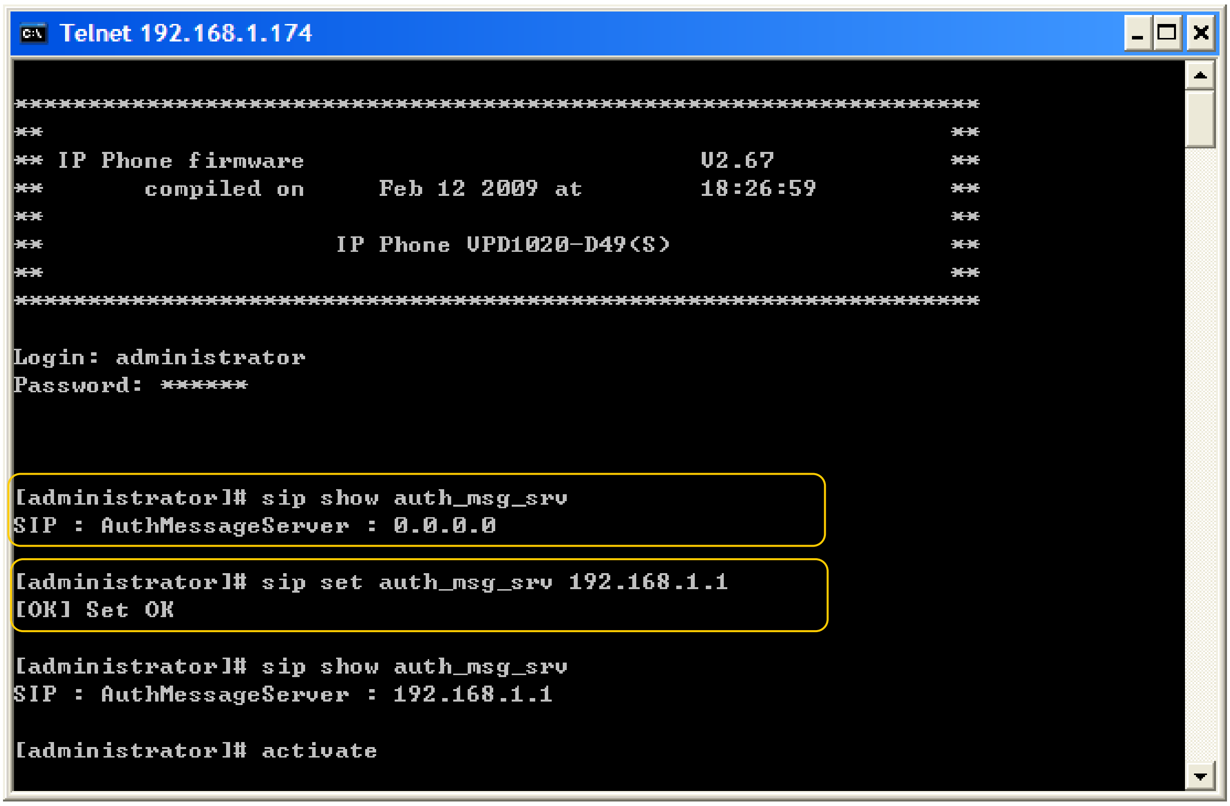
A. Via APS

This parameter only can be configured in section [sip] of common or MAC config files.

For example:

```
[sip]
...
AuthMessageServer=192.168.1.1 (or AuthMessageServer=domain.com)
...
```

B. Via Telnet



```
*****************************************************************
**                                                             **
** IP Phone firmware                          V2.67            **
**       compiled on      Feb 12 2009 at      18:26:59         **
**                                                             **
**                    IP Phone VPD1020-D49(S)                  **
**                                                             **
*****************************************************************

Login: administrator
Password: ******


[administrator]# sip show auth_msg_srv
SIP : AuthMessageServer : 0.0.0.0

[administrator]# sip set auth_msg_srv 192.168.1.1
[OK] Set OK

[administrator]# sip show auth_msg_srv
SIP : AuthMessageServer : 192.168.1.1

[administrator]# activate
```

DNS Query (Circular Method)

In the new release today, x.67.2, ST20xx have done an improvement on the mechanism of the DNS Query, which increases the efficiency on the DNS request.

Current Implementation (up to SG1.66)
For each SIP message sent from the ST20xx to the proxy server, it have to wait for the IP resolution done on the Pri DNS server. In case of the Pri DNS failed, after timout, phone will request to the Sec DNS server afterwards.

Once the Pri DNS server resume and with the timeout of Sec DNS server, phone will request the Pri DNS agina for IP resolution. The mechanism will lower the processing speed of the call and the response of the phone in case the Pri DNS down for a long time.

New Implementation (SGx.67.2)
A new parameter "**DNSFB**" is introduced and found in Common and MAC configuration files.

If the value of the parameter is **0**, the implmentation is the same as the original one, like SG x.66.2, which the phone always queries to the Pri DNS server.

While, if the value of the parameter is **1**, DNS query will be using a circular method, which the phone always starts with the DNS that has provided the answer to the previous request. If the DNS fails, then try the other and if the other answers, then use the other in subsequent requests until this one fails. and so on.

The parameter is configurable via APS and Telnet.

Feature Activation

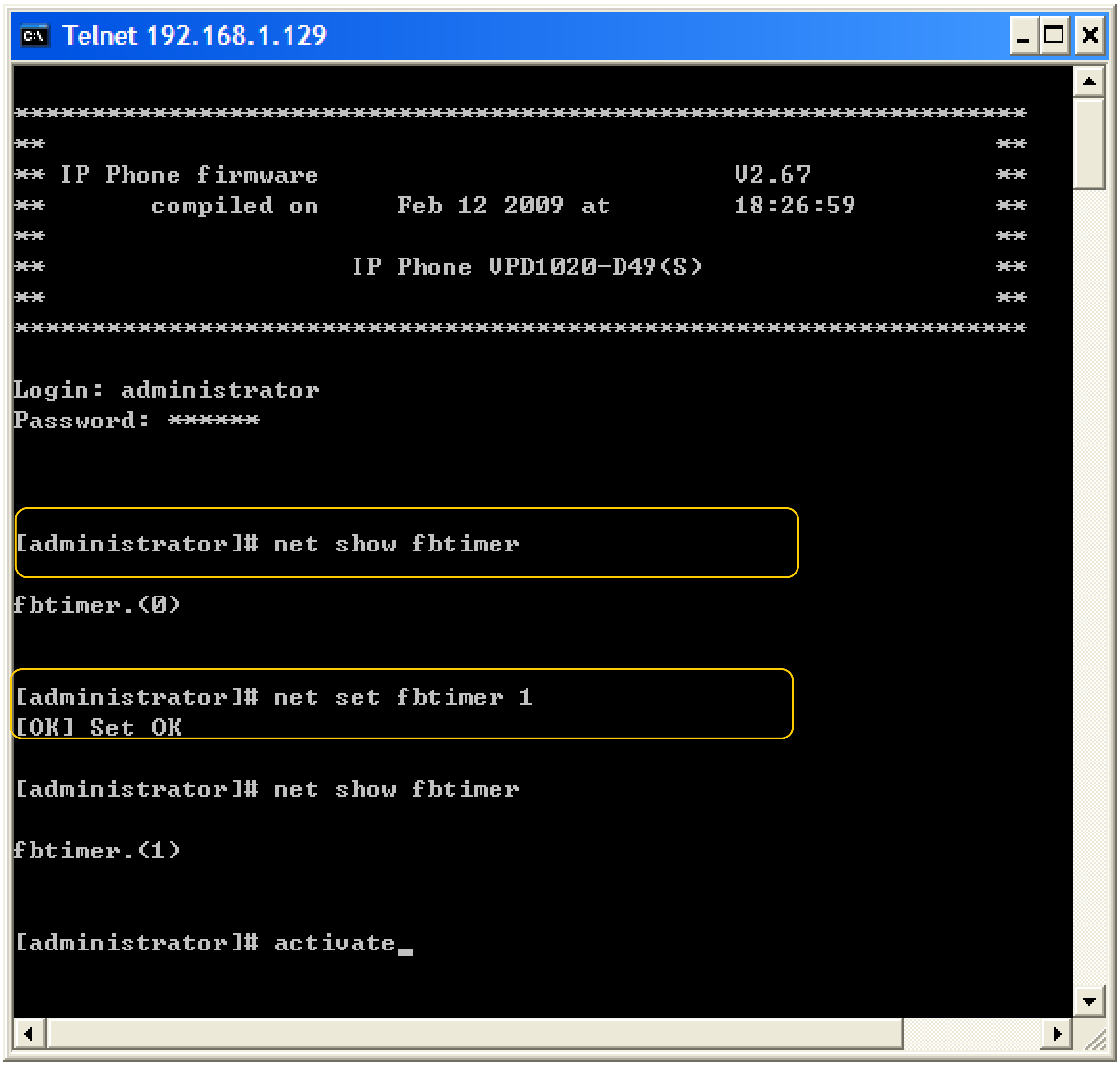
A. Via APS

APS in [net] section of both Common and Specific-MAC config files with the new parameter:

```
[net]
DNSFLG=1
DNSFB=0
DSCPflag=0
ESWITCH_LAN=1
ESWITCH_PC=1
```

B. ***Via Telnet***

To configure, open a command line console, and telnet the phone:



```
*******************************************************************
**                                                              **
** IP Phone firmware                          V2.67             **
**       compiled on      Feb 12 2009 at      18:26:59          **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
*******************************************************************

Login: administrator
Password: *******


[administrator]# net show fbtimer

fbtimer.(0)


[administrator]# net set fbtimer 1
[OK] Set OK

[administrator]# net show fbtimer

fbtimer.(1)


[administrator]# activate_
```

Support Remote call logs for local user

Remote call logs are currently supported for login/logout function in ST20xx. However, it doesn’t support for normal user.

In this section, you will know the implemetation of the remote call log for normal user in the new release SG x.67.2.

Implementation
Actually, the implementation of the remote call-log is the same as the call log for login/logout, expect for normal user this time.

The call-log file contains 3 sections: the missed calls, the received calls and the dialed calls. Each section can have up to 30 calls.

The calls shall be classified from the most recent to the oldest.

Each call is described by the phone number, phone name, date and time of the call. The call-log will use the following XML format.

```
<ThomsonRemoteCallLog>
        <MissedCalls>
                <Call>
                        <Name>John</Name>
                        <Telephone>01234</Telephone>
                        <Date>20070626</Date>
                        <Time>0130PM</Time>
                </Call>
                <Call>
                        <Name>Jack</Name>
                        <Telephone>56789</Telephone>
                        <Date>20070625</Date>
                        <Time>0315PM</Time>
                </Call>
                <Call>
                        <Name></Name>                               (name not available)
                        <Telephone>98765</Telephone>
                        <Date>20070620</Date>
                        <Time>0915AM</Time>
                </Call>
        </MissedCalls>
               <ReceivedCalls>
                <Call>
                        <Name>James</Name>
                        <Telephone>34567</Telephone>
                        <Date>20070627</Date>
                        <Time>0845AM</Time>
                </Call>
               </ReceivedCalls>
               <DialedCalls>
                <Call>
                        <Name>John</Name>
                        <Telephone>01234</Telephone>
                        <Date>20070625</Date>
                        <Time>0245PM</Time>
                </Call>
               </DialedCalls>
     </ThomsonRemoteCallLog>
```

Each time user enters in the call log menu by pressing the “**CalLog**” softkey, the phone will download the relevant file from the server using the HTTP GET method.

A new parameter “**RCallLogFlg**” is introduced to decide enabling either local call-log or remote call-log.
The default value is 0, which means the local call log is enabled,
While, the value is 1, which means the remote call log is enabled.

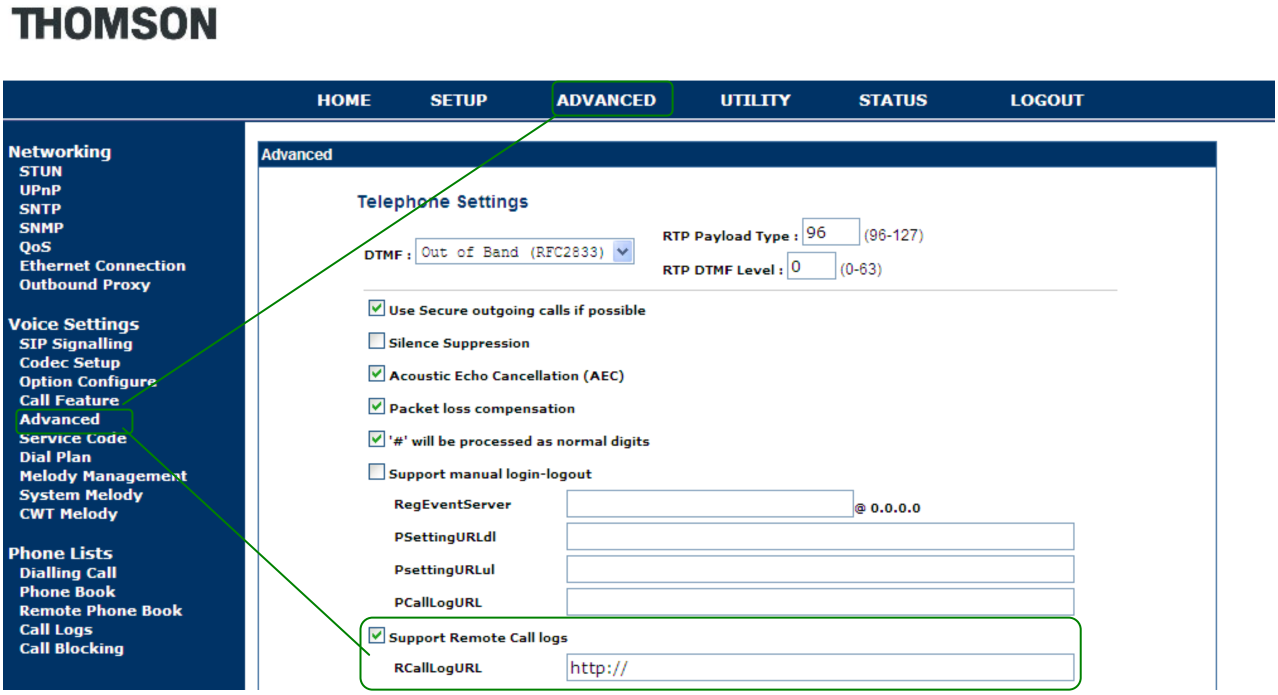
In addition, the URL for this remote call-log file is configurable using a new parameter: “**RCallLogURL”**.

The data are stored on the phone until the user quits the call log menu by pressing [Back] soft key or "C" key or the phone returns to the idle screen after a timeout.

Feature Activation

A. Via WebGui

Visit ADVANCED → advanced, click the support Remote call logs check box. Enter the remote call log URL in the field.



Telephone Settings

DTMF : Out of Band (RFC2833)

RTP Payload Type : 96 (96-127)

RTP DTMF Level : 0 (0-63)

☑ Use Secure outgoing calls if possible

☐ Silence Suppression

☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ '#' will be processed as normal digits

☐ Support manual login-logout

RegEventServer @ 0.0.0.0

PSettingURLdl

PsettingURLul

PCallLogURL

☑ Support Remote Call logs

RCallLogURL http://

B. Via APS

An **RCallLogFlg** parameter is created and added in the [sys] section of both Common and Specific-MAC config files.

RCallLogFlg = 0, (by default), the local call logs will be in use.
RCallLogFlg = 1, the remote call logs will be in use, instead.

An **RCallLogURL** parameter should be added in the [sys] section.

This parameter is set with the URL used to download the file containing the remote call log.

Examples:
RCallLogURL=http://192.168.1.1/search.php?Name=call_log&number=#LOGIN&IP_addr=#IP&mac=#MAC&pass=#PASSWORD

RCallLogURL=https://192.168.1.1/search.php?Name=call_log&number=#LOGIN&IP_addr=#IP&mac=#MAC&pass=#PASSWORD

Where,
#LOGIN is the phone number of the active profile
#PASSWORD is the SIP password of the active profile
#IP is the IP address of the phone
#MAC is the MAC address of the phone

```
[sys]
…
RCallLogFlg=1            // the remote call logs is enabled
RCallLogURL=http://
…
```

ST20XX SIP New Features (SG vx.66.2)

Overview

This document describes a set of features included in ST2030 and S2022 SIP v1.66.2 and 3.66.2 respectively in order to improve their usability in different environments.

Transfer On ringing

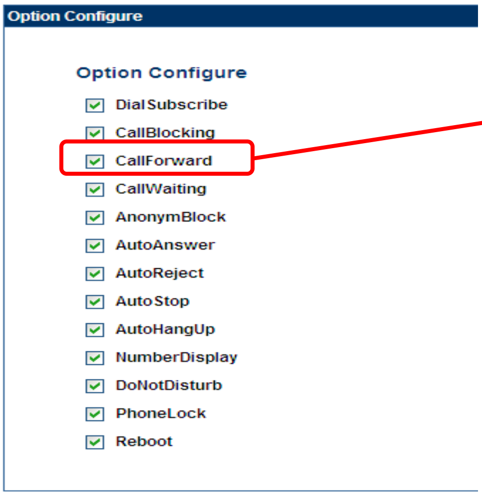
[Transf] softkey should be able to show on ringing in MMI selectively with/without respect to the CallForward bit in parameter "OptionVisible".

Current implementation

Since x.61, [Transf] softkey enable/disable on ringing in MMI with respect to the call forward bit in parameter "OptionVisible" accordingly.

In other words,
CallForward bit in parameter "OptionVisible" = 1 ➔ [Transf] softkey on ringing shows on the screen.

CallForward bit in parameter "OptionVisible" = 0 ➔ [Transf] softkey on ringing NOT shows on the screen.



Option Configure

Option Configure

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

[Transf] softkey enable/disable on ringing in MMI is directly followed the activation / deactivation of callforward bit accordingly.

New Implementation

A new parameter in Common and MAC configuration files is added to increase the flexibility to assign the [Transf] softkey on ringing shown in MMI with/without respect to the call forward bit in parameter "OptionVisible" accordingly.

TransfOnRingFlag =<0/1>,   0 is disable (by default), 1 is enable

TransfOnRingFlag = 0, [Transf] softkey is always enable without respect to the call forward bit in parameter "OptionVisible.

TransfOnRingFlag = 1, [Transf] softkey with respect to the call forward bit in parameter "OptionVisible" enable/disable. (Same as the current implementation above)

Table below describes the Enable/Disable of the [Transf] softkey on Ringing with Call forward bit in "OptionVisible".

	Call forward bit in "OptionVisible" = 0	**Call forward bit in "OptionVisible" = 1**
TransfOnRingFlag = 0	Enable	Enable
TransfOnRingFlag = 1	Disable	Enable

Feature Activation

A new parameter is needed to configure this function and with some additional definitions listed below.

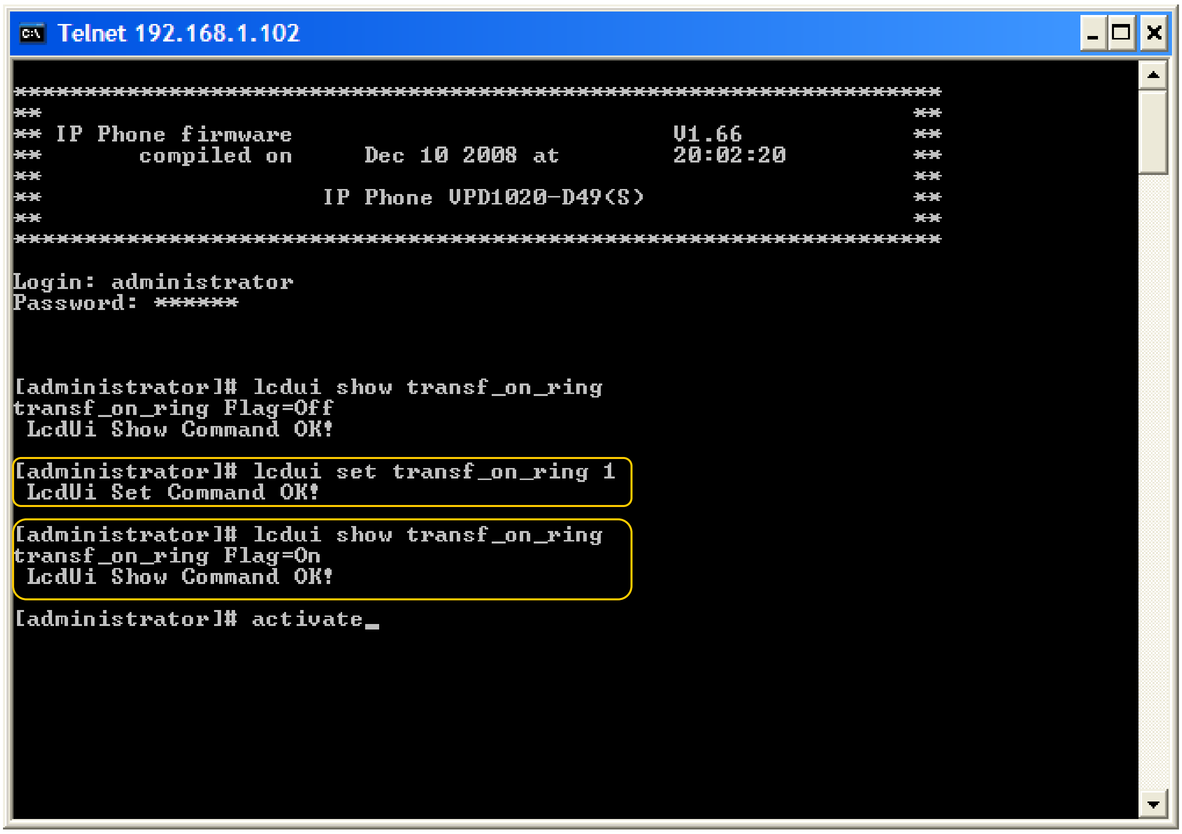
H. *Via APS:*

APS in [ipp] section of both Common and Specific-MAC config files with the new parameter:

```
[ipp]
…
OptionVisible=8191
TransfOnRingFlag =0          //by default, transfer on ringing is always enabled
……
```

I. *Via Telnet*

To configure, open a command line console, and telnet the phone:



```
*******************************************************************
**                                                              **
** IP Phone firmware                           V1.66            **
**       compiled on      Dec 10 2008 at       20:02:20         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
*******************************************************************

Login: administrator
Password: *******


[administrator]# lcdui show transf_on_ring
transf_on_ring Flag=Off
 LcdUi Show Command OK!

[administrator]# lcdui set transf_on_ring 1
 LcdUi Set Command OK!

[administrator]# lcdui show transf_on_ring
transf_on_ring Flag=On
 LcdUi Show Command OK!

[administrator]# activate_
```

Call-hold "inactive" method

To enhance the compatibility with different softswitches, a new call-hold method is introduced in this section.

Current implementation

Currently, ST20xx implements "**send-only**" method for holding a call, carried in SDP session.

```
Content-Type: application/sdp
Content-Length: 221

v=0
o=2215 1325412 1325413 IN IP4 192.168.1.102
s=-
c=IN IP4 192.168.1.102
t=0 0
m=audio 41000 RTP/AVP 18 96
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=sendonly
```

Now, an alternative new call-hold method "**inactive**" has implemented in ST20xx to enhance the compatibility working on different softswitches.

```
Content-Type: application/sdp
Content-Length: 221

v=0
o=2215 1807412 1807413 IN IP4 192.168.1.102
s=-
c=IN IP4 192.168.1.102
t=0 0
m=audio 41000 RTP/AVP 18 96
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15
a=inactive
```

Feature Activation

A new parameter is needed to configure this function and with some additional definitions listed below.

call_hold_method=<0|1>, 0 is by default
call_hold_method=0, "send-only" call-hold method is used
call_hold_method=1, "inactive" call-hold method is used

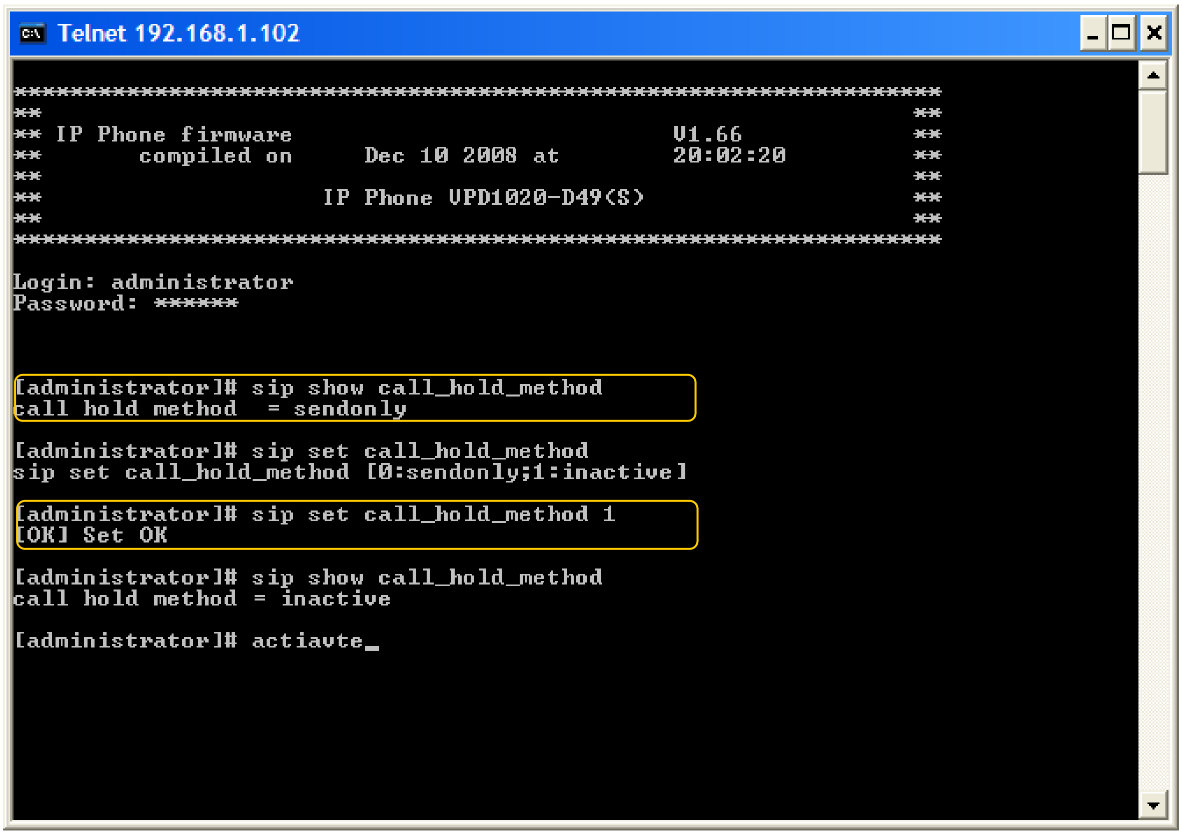
A. Via APS:

APS in [sip] section of both Common and Specific-MAC config files with the new parameter:

```
[sip]
…
call_hold_method=0            // by default, "send-only" is in use
……
```

B. Via Telnet

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.66            **
**       compiled on     Dec 10 2008 at        20:02:20         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show call_hold_method
call hold method  = sendonly

[administrator]# sip set call_hold_method
sip set call_hold_method [0:sendonly;1:inactive]

[administrator]# sip set call_hold_method 1
[OK] Set OK

[administrator]# sip show call_hold_method
call hold method = inactive

[administrator]# actiavte_
```

Comverse: Line and Service supervision (reg & ua-profile event package)

The purpose of this feature is to increase the ability of ST20xx to request and monitor the registration status, change user profile and receive notifications related to a set of services of other supervised SIP phones in Comverse environment, like:

- DND activation status
- Call Forward activation status
- Etc…

Busy Line Field (BLF)

Dialog event supervision

User-oriented BLF has already supported in ST20xx. In Comverse environment, this feature allows monitoring other line status, and speed dialing.

For each supervised phone, ST20xx generates an initial SUBSCRIBE request to the SIP server in order to be notified for the state of all dialogs for this specific agents.

Then, the network generates a NOTIFY request on any dialog update for the supervised PUI. The dialog states are « **Early** », « **Confirmed** » and « **Void** » (state not provided).

Upon reception of a NOTIFY request, the ST20xx updates the lamp associated to the supervised phone accordingly.

For User-oriented BLF service, please refer to the *ST20xx SIP New Features SG vx.52.1 P.230*

Registration event supervision with DND

One new feature is added in St20xx for working in Comverse environment. This feature allows ST20xx to monitor other line registration status with DND of the supervised phones.

For each supervised phone, ST20xx generates an initial SUBSCRIBE request to the SIP server in order to be notified for the state of all registrations for this specific phone.

Afterwards, the network generates a NOTIFY request on any registration update for the supervised PUI. The registration states are « **Registered** » and « **Unregistered** ». The service variable for DND supervision will be discussed in next section.

Upon reception of a NOTIFY request, ST20xx update the lamp associated to the supervised phone accordingly.

Implementation and configuration

Table below describes an overall ST20xx's behavior that is used for implementation, based on the **Dialog** and **Registration** and **DND supervision** of a same target:

DND state	Registration State	Dialog state	Aggregated state	Lamp state	When BLF Key is pressed
Off	Registered	Void	Free	Off	Call initiate
		Early	Ringing	Blinking fast	Call pick up
		Confirmed	Busy	On	Call initiate
	Unregistered	Any	Not available	Blinking slow	No action
On	Any	Any	Not available	Blinking slow	No action

If user presses the function Key:

- BLF Key is pressed in **OFF** state --> phone initiates a call toward the supervised SIP-URI
- BLF Key is pressed in **STEADY ON** state --> phone initiates a call toward the supervised SIP-URI
- BLF Key is pressed in **BLINKING SLOW** state --> NO action is taken
- BLF Key is pressed in **BLINKING FAST** (supervised phone is ringing) state --> phone launches a call Pickup of the supervised phone

Call pickup is launched by initiating a call toward a service code that includes the number to pickup. It can work with Soft Key or with Function key (user orientated BLF)

For call pick up service, please refer to "ST20xx SIP New Features SG vx.62 P. 145 → How to activate call Pick-up Service.

Remark:
If the status of the phone is in ringing, in communication and all others, except idle, function key should remain inactive if the user presses them. But, the supervision (LED blinking or other, including state change) should remain active during call.

Service variable supervision

Now, ST20xx has already implemented the service variable supervision, the “ua-profile” event package is supported. But working in Comverse environment, this feature has been enhanced, which not only allows supervising phone itself, but also supervised with other phones.

The service allows ST20xx to be periodically notified about the state of a set of services implemented by the network. As mentioned before, those are

- DND activation status
- Call Forward activation status
- Etc…

There are three stages for the profile delivery process, including **Enrollment**, **Content retrieval** and **Change notification**. This document will not go into detail. For more information, please refer to “St20xx SIP New Features SG vx.62 P. 145

The service variables are defined in the below table:

Name	Possible values	Description
cfu_on	Integer: { 0, 1 }	Activation state of the Unconditional Call Forward. 0 means deactivated. 1 means activated.
dnd	Integer: { 0, 1 }	Activation state of the Do Not Disturb feature. 0 means deactivated. 1 means activated.
sf_on	Integer: { 0, 1 }	Activation state of the Secretarial Filtering feature. Applicable only when a Manager line is supervised. 0 means deactivated. 1 means activated.
hg_rdy	Integer: { 0, 1 }	Indicates whether the CPE that initiated the supervision is ready on the supervised Hunt Group. Applicable only when a Hunting Group virtual line is supervised. 0 means not ready. 1 means ready.

Where,

CallFwd service is linked to “cfu_on” variable
DND service is linked to “dnd” variable
SecFilter service is linked to “sf_on” variable
HuntGroup service is linked to “hg_rdy” variable

When ST20xx receives the SIP NOTIFY message, indicating successful profile enrollment, it makes it effective immediately and displays service status to the user. The duration of the subscription is 3600 by default.

Implementation and configuration

One function key will be affected to each service. Depending on the state of the service, the corresponding function key's LED will be turned ON or OFF.

Referring to the above table, two status of service variable are well defined:

0 means Deactivation state
1 means Activation state

Table below describes lamp state with the corresponding variable state.

Variable State	Lamp state
0	Off
1	On

Service supervision is associated to a key of the phone; ST20xx provides the ability to change the variable state by pressing the key. In all cases, it exists one couple of activation/deactivation for each supervised service of the supervised phone.

Activation and deactivation consists in initiating a basic call toward the corresponding supervised phone, where the phone used depends on the current status of the service.

Table below describes the two services which are used for activation/deactivation for the supervised phone in Comverse environment.

Services	Variable State	Lamp state	When key is pressed	FCA
Secretarial Filtering	0	OFF	INVITE sip: *270x*@domain.com SIP/2.0 From: <…> To: <sip:*270x*@domain.com> (Where x is the phone number of manager)	*270x*
	1	ON	INVITE sip: *271x*@domain.com SIP/2.0 From: <…> To: <sip:*271x*@domain.com> (Where x is the phone number of manager)	*271x*
Hunting Group	0	OFF	INVITE sip: *70x#@domain.com SIP/2.0 From: <…>	*70x#

			To: <sip:*70x#@domain.com> (where x is the hunt group extension)	
	1	ON	INVITE sip: *71x#@domain.com SIP/2.0 From: <…> To: <sip:*71x#@domain.com> (where x is the hunt group extension)	*71x#

Remark:

- **Call Forward**: it is only possible to activate/deactivate for the self phone. Not support for the supervision of other phone in this case.

- **DND**: it is also only possible to activate/deactivate for the self phone. Not support for the supervision of other phone in this case.

- **SecFilter**: it is supported to activate/deactivate for the supervised phone. The typical use case allows a secretary to have information on his/her phone by using a LED of the activation status of the Secretarial Filtering on the Line of the Boss. By pressing the feature key, secretary can change the activation status of the secretarial filtering on the boss line.

- **HuntGroup**: it is supported to activate/deactivate for the supervised phone
  The typical use case allows a Hunting Group Member to have information on his/her phone by using a LED of his/her logon status regarding in a given Hunting Group. By pressing the feature key, member can change the logon status in this Hunting Group.

Feature Activation

Below is the typical example, including the supervision of self phone and other phones, with making use of the governed parameters below:

- A (2001) is the self-phone number.
- B (2003 ~ 2005) supervised phone lines
- D (2007) is the Boss phone number
- E (2022) is the Hung group number

FK1 & FK2:	Line
FK3 to FK4:	Use to supervise others phone with dialog, registration and DND services (Supervised phones: 2003 to 2004)

FK5 to FK6: Use to monitor the status of CFW and DND of the supervised phone (Supervised phones: 2005)

FK7: CFU (Self Phone) LED indicates ON/Off, Press Key for activation or Deactivation depends on the current status of the service.

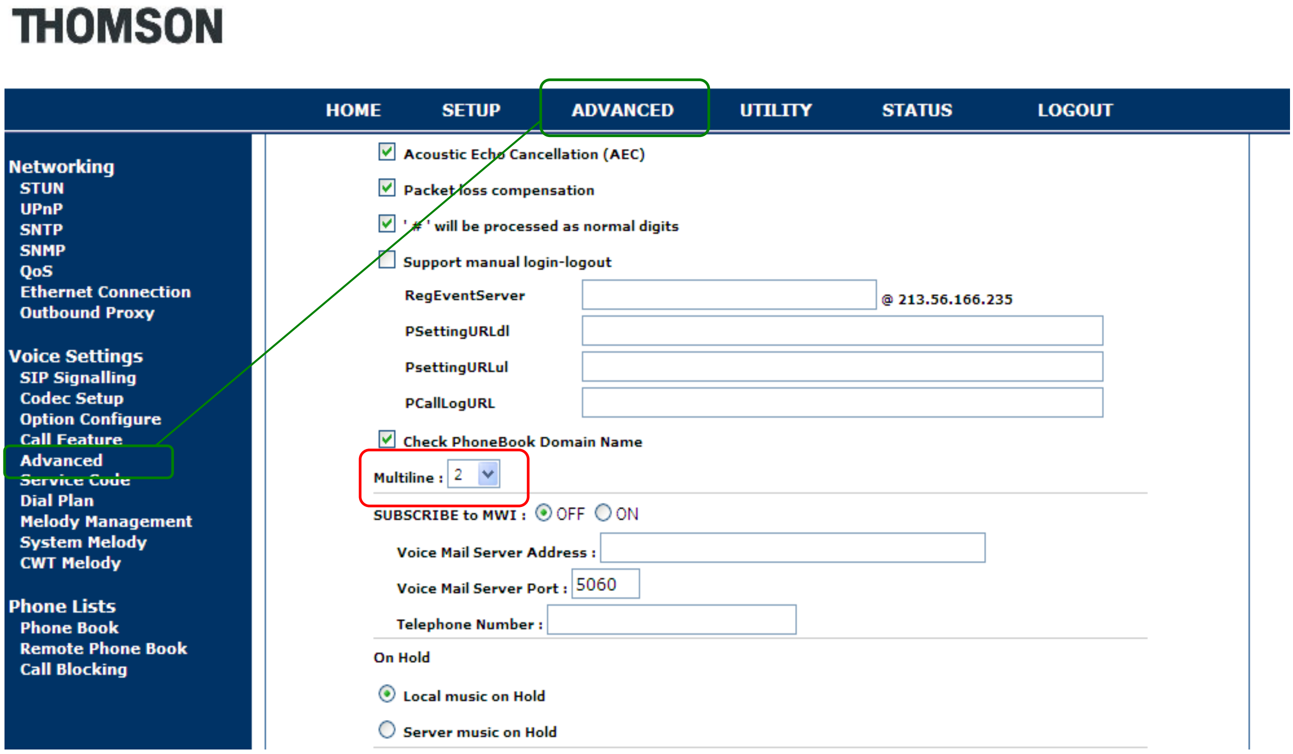
FK8: DND (Self Phone) LED indicates ON/Off, Press Key for Activation or Deactivation depends on the current status of the Service.

FK9: Use to supervise other phone with SF service. LED indicates ON/Off, Press Key for Activation or Deactivation depends on the current status of the Service.

FK10: Use to supervise other phone with HG service. LED indicates ON/Off, Press Key for Activation or Deactivation depends on the current status of the Service.

A. <u>Via WebGui:</u>

Firstly, it is required to reduce the number of Multiline to reserve the number of supervised lines in Function Key, in Advanced → Advanced. Change the Multiline to be 2 and apply.



☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ ' # ' will be processed as normal digits

☐ Support manual login-logout

RegEventServer: @ 213.56.166.235

PSettingURLdl:

PsettingURLul:

PCallLogURL:

☑ Check PhoneBook Domain Name

Multiline : 2

SUBSCRIBE to MWI : ● OFF ○ ON

Voice Mail Server Address :

Voice Mail Server Port : 5060

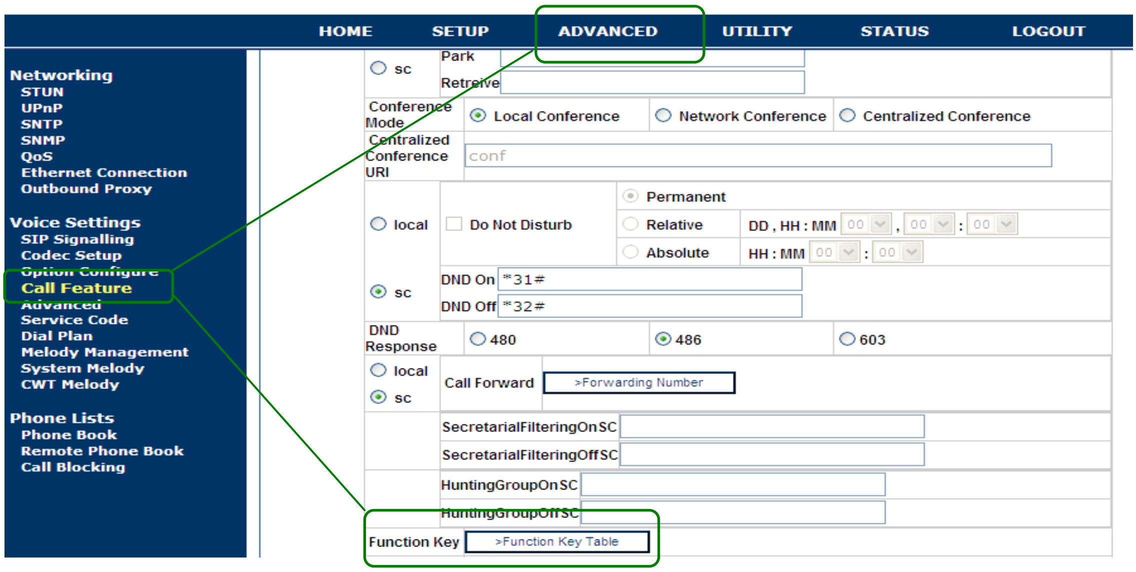
Telephone Number :

On Hold

● Local music on Hold

○ Server music on Hold

Then, you should go to Function key table through Advanced → Call feature → Function Key table.



○ sc Park Retreive

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

○ local ☐ Do Not Disturb ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00

● sc DND On *31# DND Off *32#

DND Response: ○ 480 ● 486 ○ 603

○ local ● sc Call Forward

SecretarialFilteringOnSC

SecretarialFilteringOffSC

HuntingGroupOnSC

HuntingGroupOffSC

Function Key

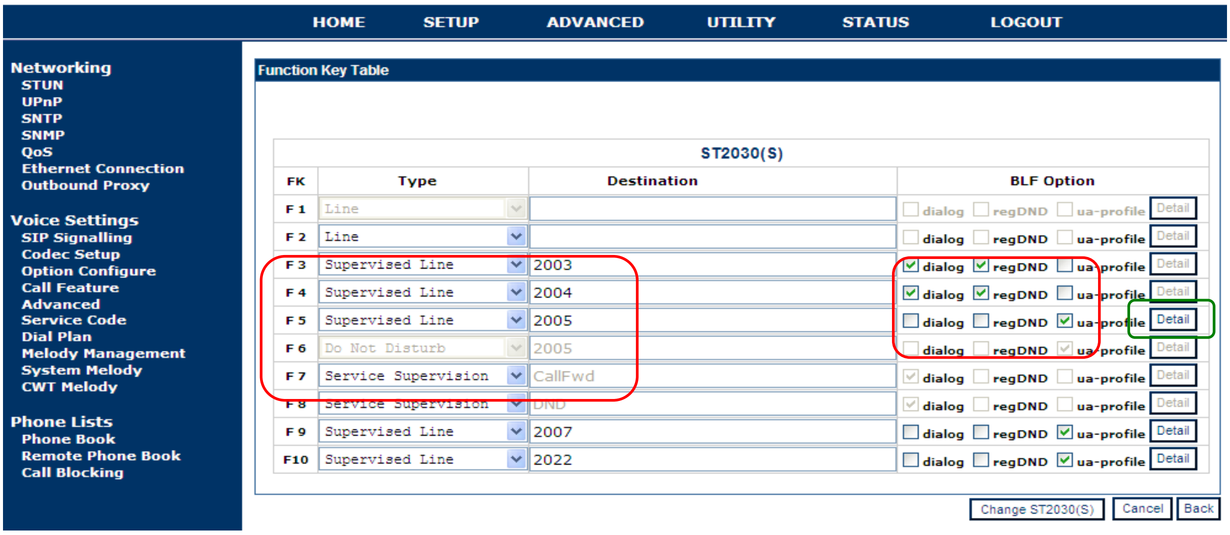
Select the item in the roll down “Type” menu: **Supervised Line** and enter the supervised phone numbers in the corresponding field of the function keys.

In this case, the function keys 3 to 4 will monitor the dialog, registration with DND service for the corresponding supervised phones. Therefore, the boxes for **dialog** and **regDND** for each supervised phones are checked.

Likewise, the function key 5 is to monitor the status of CFW and DND of the supervised phones, the **ua-porfile** option must be checked. Click the Change ST2030(S) to apply.

Then, click the **Detail** button to go into the sub-menu of the **ua-profile**

THOMSON



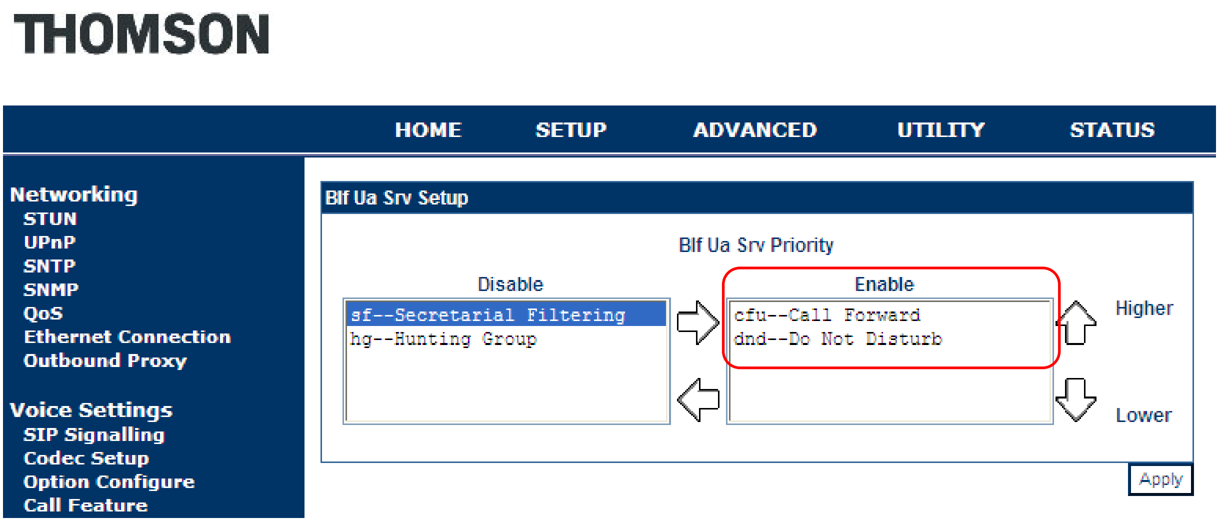
Function Key Table

ST2030(S)

FK	Type	Destination	BLF Option			
F 1	Line		☐ dialog	☐ regDND	☐ ua-profile	Detail
F 2	Line		☐ dialog	☐ regDND	☐ ua-profile	Detail
F 3	Supervised Line	2003	☑ dialog	☑ regDND	☐ ua-profile	Detail
F 4	Supervised Line	2004	☑ dialog	☑ regDND	☐ ua-profile	Detail
F 5	Supervised Line	2005	☐ dialog	☐ regDND	☑ ua-profile	Detail
F 6	Do Not Disturb	2005	☐ dialog	☐ regDND	☑ ua-profile	Detail
F 7	Service Supervision	CallFwd	☑ dialog	☐ regDND	☐ ua-profile	Detail
F 8	Service Supervision	DND	☑ dialog	☐ regDND	☐ ua-profile	Detail
F 9	Supervised Line	2007	☐ dialog	☐ regDND	☑ ua-profile	Detail
F10	Supervised Line	2022	☐ dialog	☐ regDND	☑ ua-profile	Detail

Enable the **cfu- Call Forward** and **dnd - Do Not Disturb** services by highlighting the service and click the right arrow. Then, click Apply.

For Feature Key 9 and 10, the procedures are the same.



Blf Ua Srv Setup

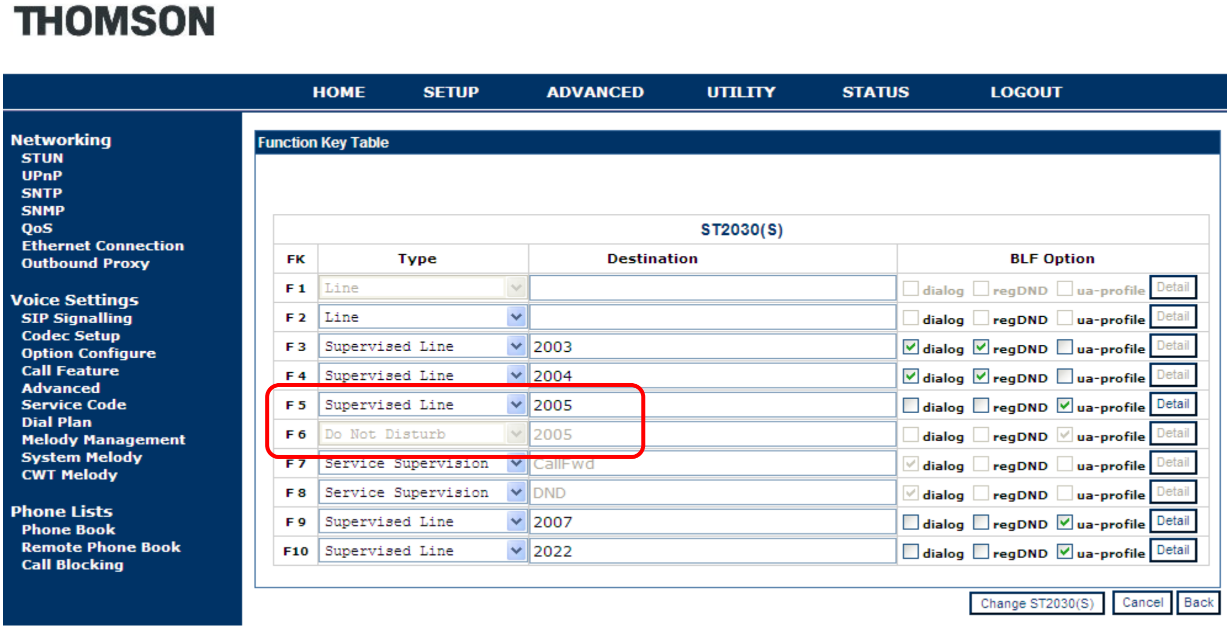
Blf Ua Srv Priority

Disable:
```
sf--Secretarial Filtering
hg--Hunting Group
```

Enable:
```
cfu--Call Forward
dnd--Do Not Disturb
```

You can also press the up and down arrow to change the priority for the enabled service. The priority is to decide which services would be used to the first assigned function key. The higher the priority, the first assigned function key will be used.

Therefore, the below feature key 5 is assigned with Call forward service and feature key 6 is assigned with DND.



Function Key Table

ST2030(S)

FK	Type	Destination	BLF Option
F 1	Line		☐ dialog ☐ regDND ☐ ua-profile Detail
F 2	Line		☐ dialog ☐ regDND ☐ ua-profile Detail
F 3	Supervised Line	2003	☑ dialog ☑ regDND ☐ ua-profile Detail
F 4	Supervised Line	2004	☑ dialog ☑ regDND ☐ ua-profile Detail
F 5	Supervised Line	2005	☐ dialog ☐ regDND ☑ ua-profile Detail
F 6	Do Not Disturb	2005	☐ dialog ☐ regDND ☑ ua-profile Detail
F 7	Service Supervision	CallFwd	☑ dialog ☐ regDND ☐ ua-profile Detail
F 8	Service Supervision	DND	☑ dialog ☐ regDND ☐ ua-profile Detail
F 9	Supervised Line	2007	☐ dialog ☐ regDND ☑ ua-profile Detail
F10	Supervised Line	2022	☐ dialog ☐ regDND ☑ ua-profile Detail

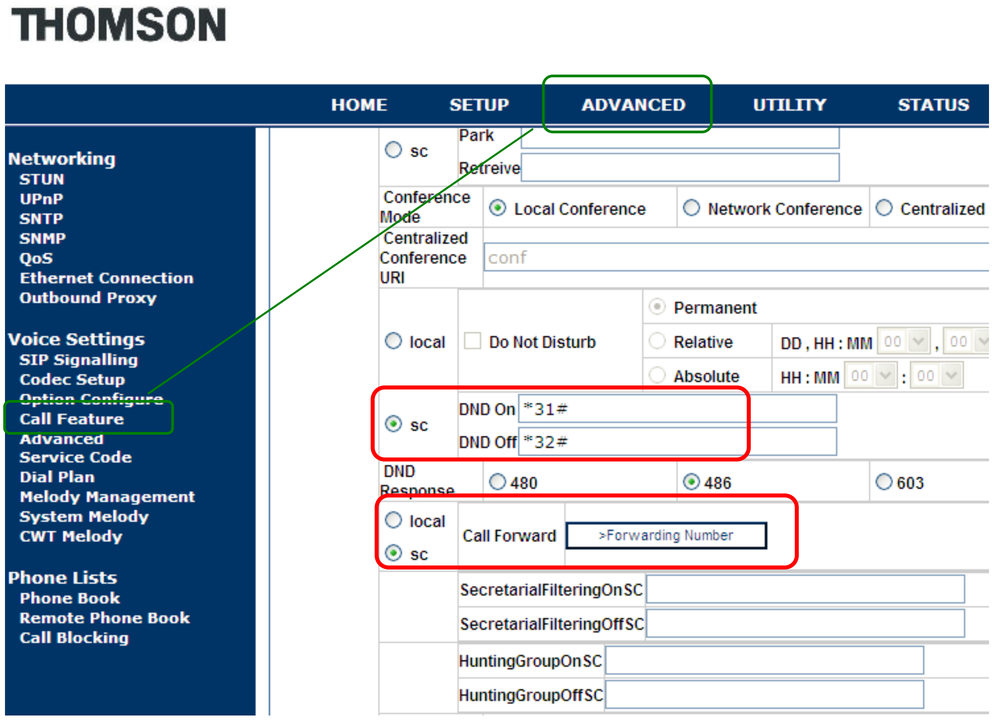
For the configuration of service supervision of self-phone, it only can be done by APS. Besides, it cannot be modifiable thought neither the MMI nor the WebGui.

One of the important topics need to be mentioned is the configuration of the starcode.

In this example, there are several starcodes involved. This document will give a brief description and make use of the startcodes working in Comverse environment. For more detail, please refer to "*ST20xx SIP New Features SG vx.62" P.145*

Remark:
Now, if the Starcodes are used for self-phone, those will be denoted by SC. While, if the Starcodes are used for supervised phones, which will be denoted by SV.

Then, go to Advanced → Call feature; enter the Starcode for **DND On/Off** and **Call Forward**.



○ sc Park / Retreive

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized

Centralized Conference URI: conf

○ local ☐ Do Not Disturb ○ Permanent ○ Relative DD , HH : MM 00 , 00 ○ Absolute HH : MM 00 : 00

● sc DND On *31# DND Off *32#

DND Response ○ 480 ● 486 ○ 603

○ local ● sc Call Forward >Forwarding Number

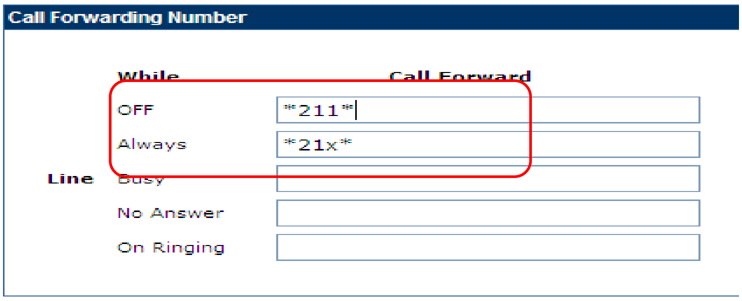
SecretarialFilteringOnSC

SecretarialFilteringOffSC

HuntingGroupOnSC

HuntingGroupOffSC

Under the **call forward** sub-menu, enter the starcode for Call-forward OFF and call-forward Always ON for the couple of activation/deactivation of the service. x is represented the phone number, which the self-phone forwarded to.

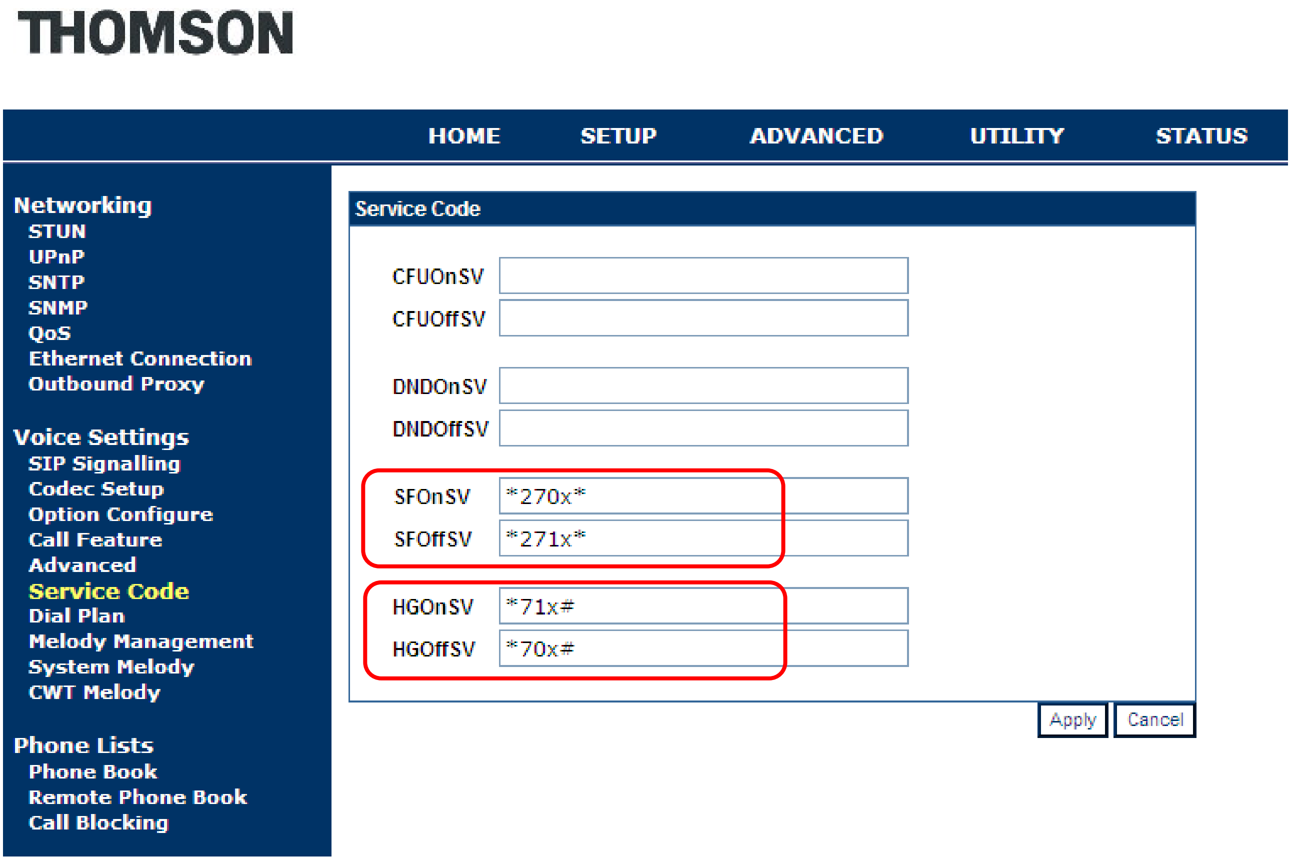


Call Forwarding Number

	While	Call Forward
Line	OFF	*211*
	Always	*21x*
	Busy	
	No Answer	
	On Ringing	

A new page is added called **Service code** in Advanced -> Service code. All the Starcode, which are used for monitoring to the supervised phones will be put there and be denoted by SV.

Enter the starcodes for **SFOnSV** , **SFOffSV**, **HGOnSV** and **HGOffSV** respectively and Apply.



CFUOnSV
CFUOffSV
DNDOnSV
DNDOffSV
SFOnSV *270x*
SFOffSV *271x*
HGOnSV *71x#
HGOffSV *70x#

B. *Via APS:*

In order not to conflict with the original user-oriented BLF, a new parameter will be added in Common and MAC configuration files to enable the feature of line supervision of registration and service supervision for each function key.

The existing parameters for user-oriented BLF will be used:

Current_Max_Multiline=10 (default)
FeatureKeyExtXX =S/<sip:xxxx>

New parameter:
FeatureKeyOptXX=Dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0)),

Where,

0 is disable, 1 is enable
XX is the number function keys used with supported extension modules, from Ext 01~ 66.

Dialog – current BLF (Dialog only)
regDND - registration status with DND
cfu - Un-conditional call forward
dnd – Do Not Disturb
sf - Secretary filter
hg – Hunted group

Default setting:
FeatureKeyOptXX=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))

```
[sys]
…
Current_Max_Multiline=2
FeatureKeyExt03 =S/<sip:2003>
FeatureKeyExt04 =S/<sip:2004>
FeatureKeyExt05 =S/<sip:2005>
FeatureKeyExt09 =S/<sip:2007>
FeatureKeyExt10 =S/<sip:2022>

FeatureKeyOpt03=dialog(1)regDND(1)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt04=dialog(1)regDND(1)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt05=dialog(0)regDND(0)ua-profile(1:cfu(1)dnd(1)sf(0)hg(0))
FeatureKeyOpt09=dialog(0)regDND(0)ua-profile(1:cfu(0)dnd(0)sf(1)hg(0))
FeatureKeyOpt10=dialog(0)regDND(0)ua-profile(1:cfu(0)dnd(0)sf(0)hg(1))
…
ServiceSupervisionStart=7
ServiceSupervisionOrder=CallFwd(1)DND(2)SecFilter(0)HuntGroup(0)
```

```
….

[sip]
….
CallFwdFlg=sc                        // StartCode for call forward (self phone)
CallFwdOffSC=*211*
CallFwdAlwaysSC=*21x*
DNDFlg=sc                            // StartCode for DND (self phone)
DNDOnSC=*31#
DNDOffSC=*32#
SFOnSV=*270x*                        // StartCode for SF (supervised phone)
SFOffSV=*271*
HGOnSV=*71x#                         // StartCode for HG (supervised phone)
HGOffSV=*70x#
…
```

Remark:

- The default sequence of ua-profile can be changed, like dnd(0)cfu(0)hg(0)sf(0)
- ua-profile value should be set to 1 in order to activate the services supervision.
- 0 for disable, default value; 1 for enable; any other value(except 0 or 1) or empty is considered as disable
- if duplicated, cfu(0)cfu(1)dns(0)sf(0)hg(0), the first one is valid.

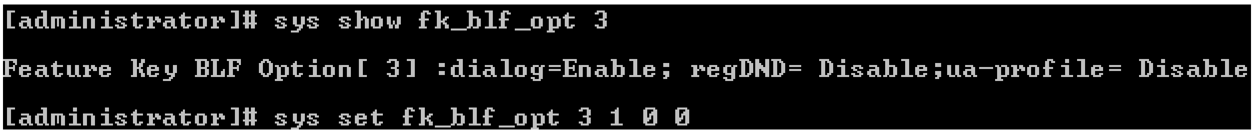
C. ***Via Telnet:***

Below shows some significant telnet commands for this case. For more information, please refer to *ST20XXS_Config File Syntax_V0034.pdf*.

1. Set the function key to be **Supervised line** with phone number

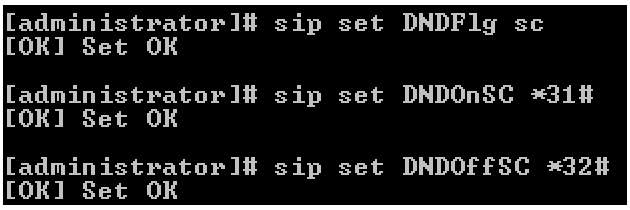
```
[administrator]# sys set fk 3 S/<sip:2003>
[OK] Set OK
```

2. Enable the line supervision and service supervision



```
[administrator]# sys show fk_blf_opt 3

Feature Key BLF Option[ 3] :dialog=Enable; regDND= Disable;ua-profile= Disable

[administrator]# sys set fk_blf_opt 3 1 0 0
```

3. Enter the starcodes flag with starcode



```
[administrator]# sip set DNDFlg sc
[OK] Set OK

[administrator]# sip set DNDOnSC *31#
[OK] Set OK

[administrator]# sip set DNDOffSC *32#
[OK] Set OK
```

Comverse: Redirecting identity and name presentation

Currently, the ST20xx do not support the "display-name" string of the Diversion header.

With working in Comverse environment, the feature of redirecting identity and name presentation allows its displays on the terminating phone. And the redirecting identity and name information is received on the initial INVITE request.

Therefore, this section is to address the implementation of the redirecting identity in ST20xx.

Let see the below example:

A (calling): 84000532 "Alice"

B (redirecting): 81000541 "Bob the Manager"

C (called): 84000531 "Assistant"

For a basic incoming call, ST20xx will receive the following SIP INVITE message:

Initial INVITE from NETWORK received by C:

```
INVITE sip:84000531@10.165.2.31:5060 SIP/2.0
Via: SIP/2.0/UDP 10.165.2.200:5060;branch=z9hG4bK4901036f-6756-5198
Call-Id: CI_375_84000531_120
From: "Alice" <sip:84000532@10.165.2.200:5060>;tag=10.165.2.200-6745-8537
To: <sip:84000531@10.165.2.31:5060>
Max-Forwards: 70
Allow: REGISTER, INVITE, BYE, ACK, CANCEL, REFER, INFO, OPTIONS, SUBSCRIBE
Session-Expires: 120
CSeq: 375 INVITE
Contact: <sip:84000532@10.165.2.200:5060>
Supported: timer
Diversion: "Bob the Manager" <sip:81000541@10.165.2.200:5060>
Content-Type: application/sdp
Content-Length: 217

SDP
```

There are three states need to be considered:

1. In Ringing (During call incoming) state

St20xx display the Caller and the Redirecting name alternatively on the screen, and the redirecting name will be displayed in reverse video. The time interval for the swap will be in 1 second.

If the diversion header has Redirecting party Name, phone should display "Bob the Manager", but if it doesn't have Redirecting party Name, phone should display Number "81000541".

2. In Talking state:

St20xx display the Caller name with number.

3. In Call log:

The call log list shows the detail of caller information.

Remark:
If the phone number of "Bob the manager" is saved in phone book, the display should show the name in the phone book.

Only top header is showed.

Feature Activation

3. Via APS:

A new parameter will be introduced to activate/deactivate the feature of redirecting identity and name presentation in [sip] section of both Common and Specific-MAC config files.

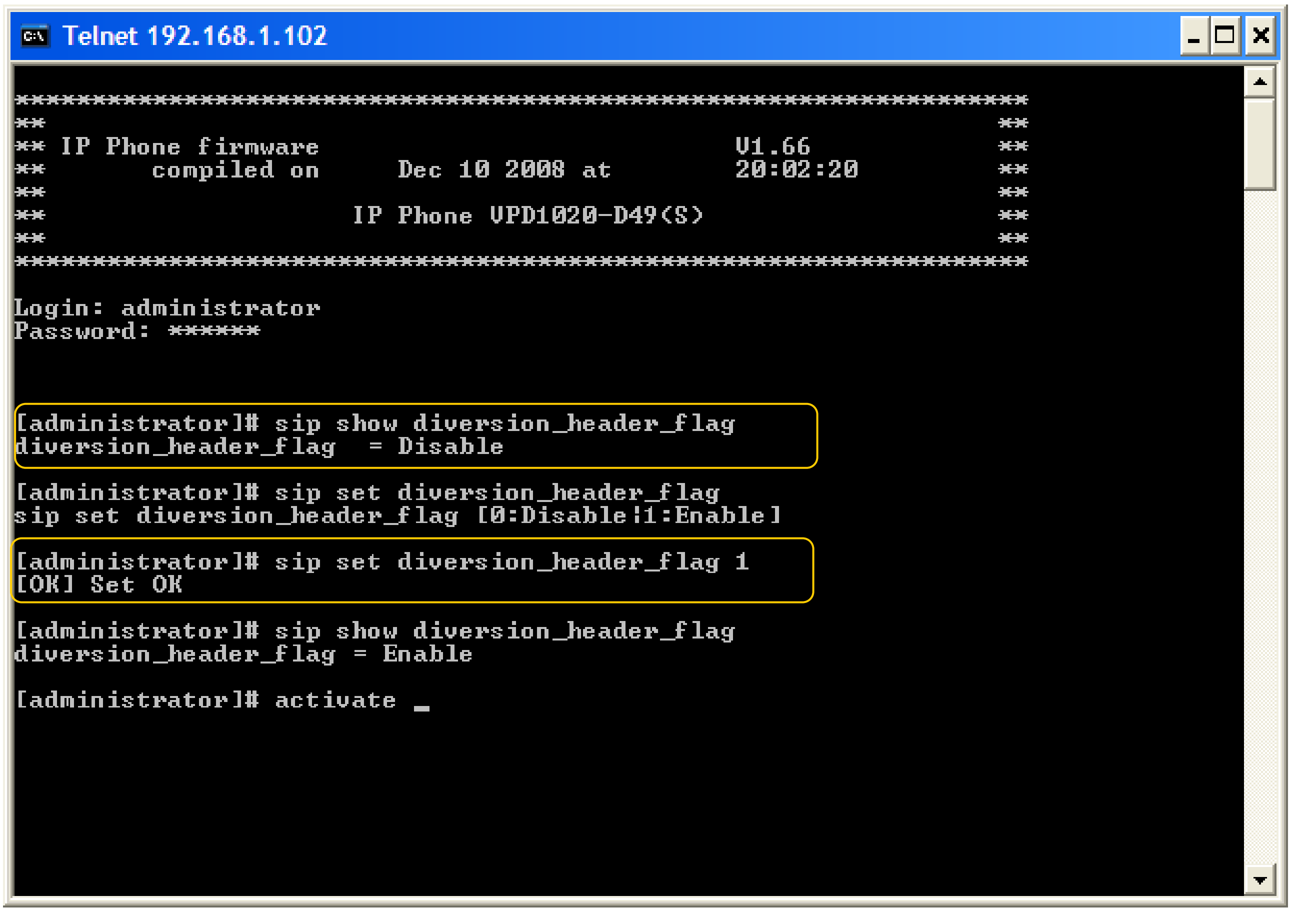
DiversionHeaderFlag= <0|1>,

Where, 0 is disable (default setting),
1 is enable

```
[sip]
…..
DiversionHeaderFlag =0                    // by default, disable
DiversionHeaderFlag =1                    // Diversion header is supported
……
```

4. Via Telnet

To configure, open a command line console, and telnet the phone:



```
*******************************************************************
**                                                              **
** IP Phone firmware                          V1.66             **
**        compiled on      Dec 10 2008 at     20:02:20          **
**                                                              **
**                     IP Phone VPD1020-D49(S)                  **
**                                                              **
*******************************************************************

Login: administrator
Password: *******


[administrator]# sip show diversion_header_flag
diversion_header_flag  = Disable

[administrator]# sip set diversion_header_flag
sip set diversion_header_flag [0:Disable|1:Enable]

[administrator]# sip set diversion_header_flag 1
[OK] Set OK

[administrator]# sip show diversion_header_flag
diversion_header_flag = Enable

[administrator]# activate _
```

ST20XX SIP New Features (SG vx.65.1)

Overview

This document describes a set of features included in ST20xx vx.65.1 in order to improve their usability in different environments.

Multi-registration on a same server

St20xx is able to support multi-registration on the same server.

Same as before, the main line will be the master.

“**Registered**” status icon will be driven by the main line registration status. Function key 1 will always be assigned to the main line. Multi registration can include Main line.

The phone behaviour will be the same for the main line and other registered lines for all features including register.

The phone supports multi-registration so that each line will register with server with different credential and extension number.

This document addresses the improvements to be done in ST20xx to enhance its support for this feature.

Feature Activation

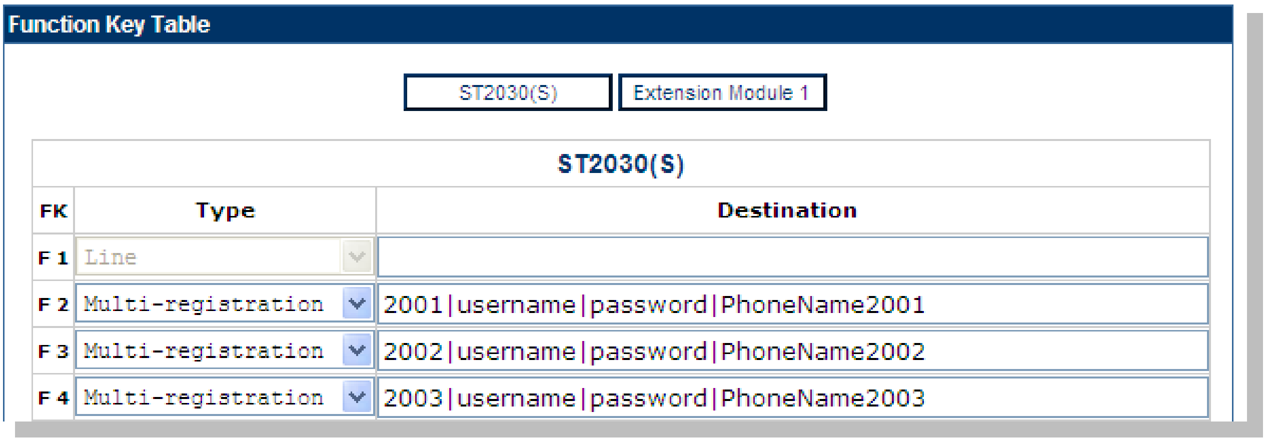
A new parameter is needed to configure this function and with some additional definitions listed below.

J. Via the web GUI:

An additional item will be created in the roll down “Type” menu: **Multi-registration.** This roll down type will always be shown.

The function key assigned to multi-registration will have the following parameters separated by “**|**” (vertical bar):
- Line number
- Authentication username
- SIP password
- Phone Name (Optional)

Example:



ST2030(S)		
FK	**Type**	**Destination**
F 1	Line	
F 2	Multi-registration	2001|username|password|PhoneName2001
F 3	Multi-registration	2002|username|password|PhoneName2002
F 4	Multi-registration	2003|username|password|PhoneName2003

K. *Via APS:*

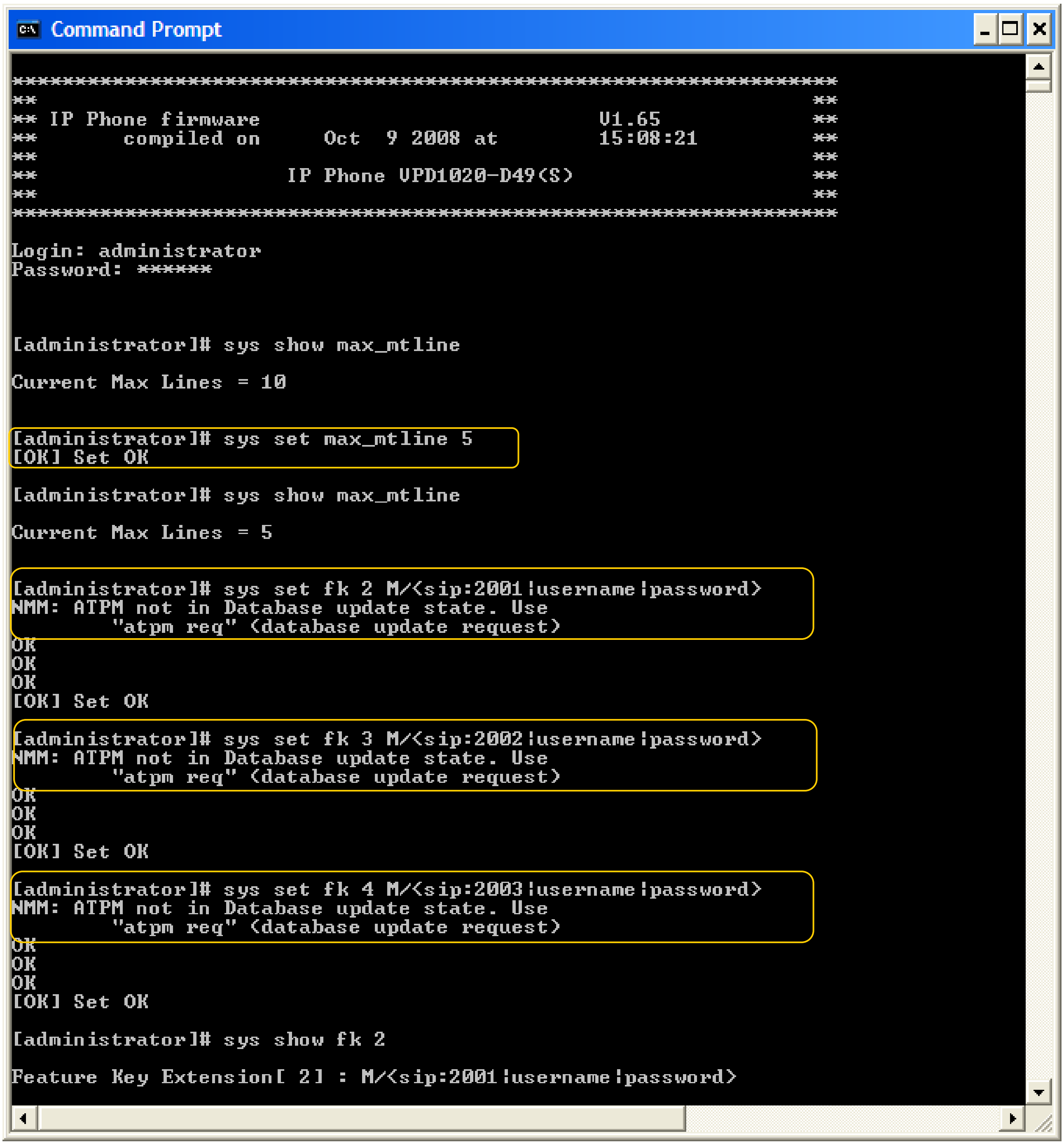
APS in [sys] section of both Common or Specific-MAC config files with the new
Parameter:

M will be used to denote the option for multi-registration, like below:

```
[sys]

…..
Current_Max_Multiline=5       (available overall number of Mulit-registration line is
Multiline-1)

FeatureKeyExt02=M/:<sip:2001|username|password|PhoneName2001>
FeatureKeyExt03=M/:<sip:2002|username|password|PhoneName2002>
FeatureKeyExt04=M/:<sip:2003|username|password|PhoneName2003>
….
```

L. *Via Telnet*

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.65            **
**       compiled on     Oct  9 2008 at        15:08:21         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sys show max_mtline

Current Max Lines = 10

[administrator]# sys set max_mtline 5
[OK] Set OK

[administrator]# sys show max_mtline

Current Max Lines = 5

[administrator]# sys set fk 2 M/<sip:2001|username|password>
NMM: ATPM not in Database update state. Use
        "atpm req" (database update request)
OK
OK
OK
[OK] Set OK

[administrator]# sys set fk 3 M/<sip:2002|username|password>
NMM: ATPM not in Database update state. Use
        "atpm req" (database update request)
OK
OK
OK
[OK] Set OK

[administrator]# sys set fk 4 M/<sip:2003|username|password>
NMM: ATPM not in Database update state. Use
        "atpm req" (database update request)
OK
OK
OK
[OK] Set OK

[administrator]# sys show fk 2

Feature Key Extension[ 2] : M/<sip:2001|username|password>
```

ST20XX SIP New Features (SG vx.64.2)

Overview

This document describes a set of features included in ST20xx vx.64.2 in order to improve their usability in different environments.

Support Private Number (ST2030 only)

Our customer's architecture includes a private identifier (called "**pn**") in the INVITE message in order to notify the end-user with a more legible number notification.

Current Implementation (up to SG v1.63)
 - "pn" parameter is currently ignored

New Implementation (SG v1.64)
For this purpose, a new parameter has been included in section [sip] of common or MAC config files in each profile.

Remark:
use_PrivateNumber=0 (default)

New policy:
If header to be used for incoming call identity display (depending on CLIPPriorDisplay and received headers) contains an uri like this:

header: <sip:<number>;pn=<privatenumber>@<hostpart>,
then

- if use_PrivateNumber is set to 0 (default),
  - show and store in Call log <number>@<hostpart> , as current behaviour

- if use_PrivateNumber is set to 1,
  - show <privatenumber> in incoming call and talking screen
  - show <privatenumber> in call log,
  - store the full <user part>@<hostpart>, in call log, for dialing purposes

Feature Activation

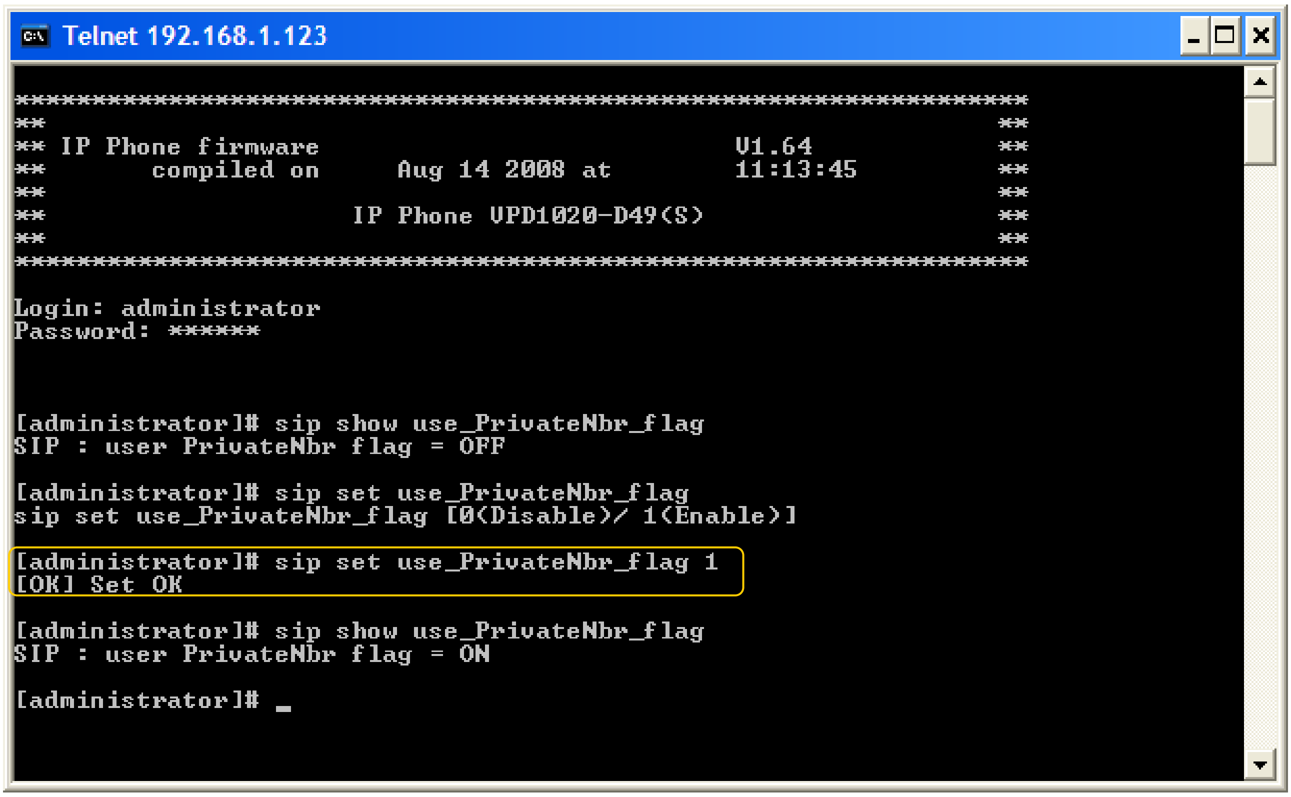
A) Through APS:

In Common or Mac Config file, user can set enable or disable the use of private number to 0 or 1 respectively. Default value is 0

```
[sip]
…
use_PrivateNumber=0 (default)
use_PrivateNumber=1 (Notify the end-user with a more legible number)
…
```

B) Through Telnet:

To configure, open a command line console, and telnet the phone:

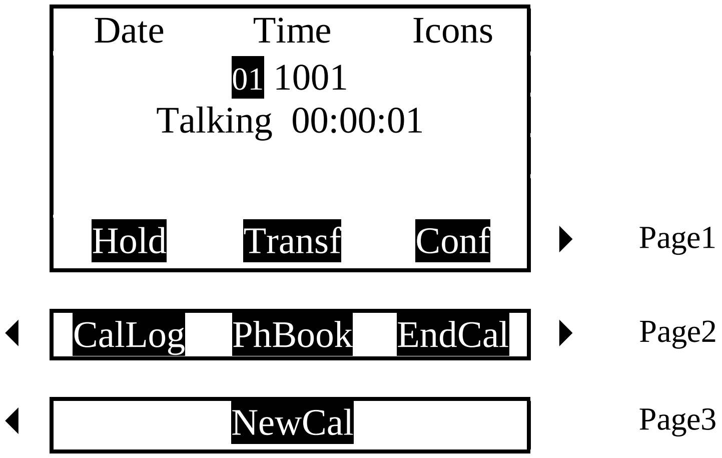


```
*******************************************************************
**                                                              **
** IP Phone firmware                          V1.64             **
**         compiled on       Aug 14 2008 at   11:13:45          **
**                                                              **
**                     IP Phone VPD1020-D49(S)                  **
**                                                              **
*******************************************************************

Login: administrator
Password: ******


[administrator]# sip show use_PrivateNbr_flag
SIP : user PrivateNbr flag = OFF

[administrator]# sip set use_PrivateNbr_flag
sip set use_PrivateNbr_flag [0(Disable)/ 1(Enable)]

[administrator]# sip set use_PrivateNbr_flag 1
[OK] Set OK

[administrator]# sip show use_PrivateNbr_flag
SIP : user PrivateNbr flag = ON

[administrator]# _
```

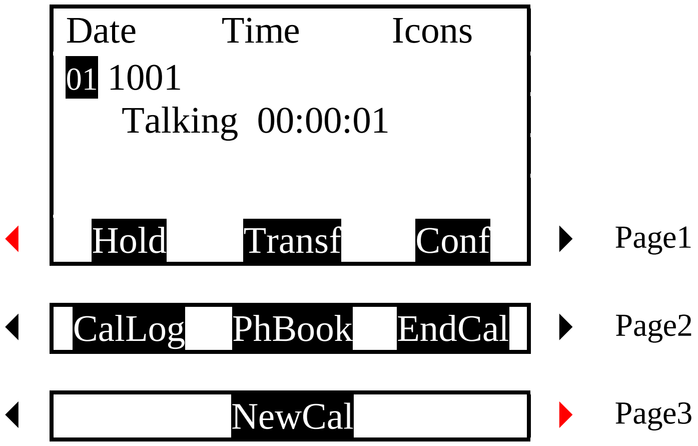
Softkey left scrolling in talking mode (ST2030 only)

In order to improve the MMI and have the same behaviour between idle mode and talking mode, the navigation thru the soft keys has to be changed

Current Implementation (up to SG v1.63)
Currently you cannot go directly from Page1 to Page3. You have to go to Page2 then Page3. When you're on Page3 you cannot go directly to Page1



New Implementation (SG v1.64)
Add left arrow on Page1 allowing direct jump to Page3
Add right arrow on Page3 allowing direct jump to Page1



Feature Activation

This is always active.

ST20XX SIP New Features (SG vx.63)

Overview

This document describes a set of features included in ST2030 and S2022 SIP v1.63 and 3.63 respectively in order to improve their usability in different environments.

Incoming Call during transfer

When users try to transfer a call and receive a call at that moment, the transfer shall not be cancelled.

The incoming call shall be placed in the queue on the first available line (Line 2, Line 3 ...) or rejected if no more free line is available, and it also shall not disturb the transfer process.

However, for blind transfer a mechanism needs to be provided to answer incoming calls once the transfer has been launched.

Thus, this section addresses the improvements on Attended transfer and blind transfer of ST2030s and provides a number of call flows for your reference.

1- Attended Transfer

When an incoming call arrives, it is automatically placed on the first available line. Depending on whether the 2nd call has been already initiated or not yet, the display shows the incoming call on the screen.

To make the implementation easier, if the user presses “C” key, the display will return to the “Transfer” screen to allow the user to resume the on going transfer.

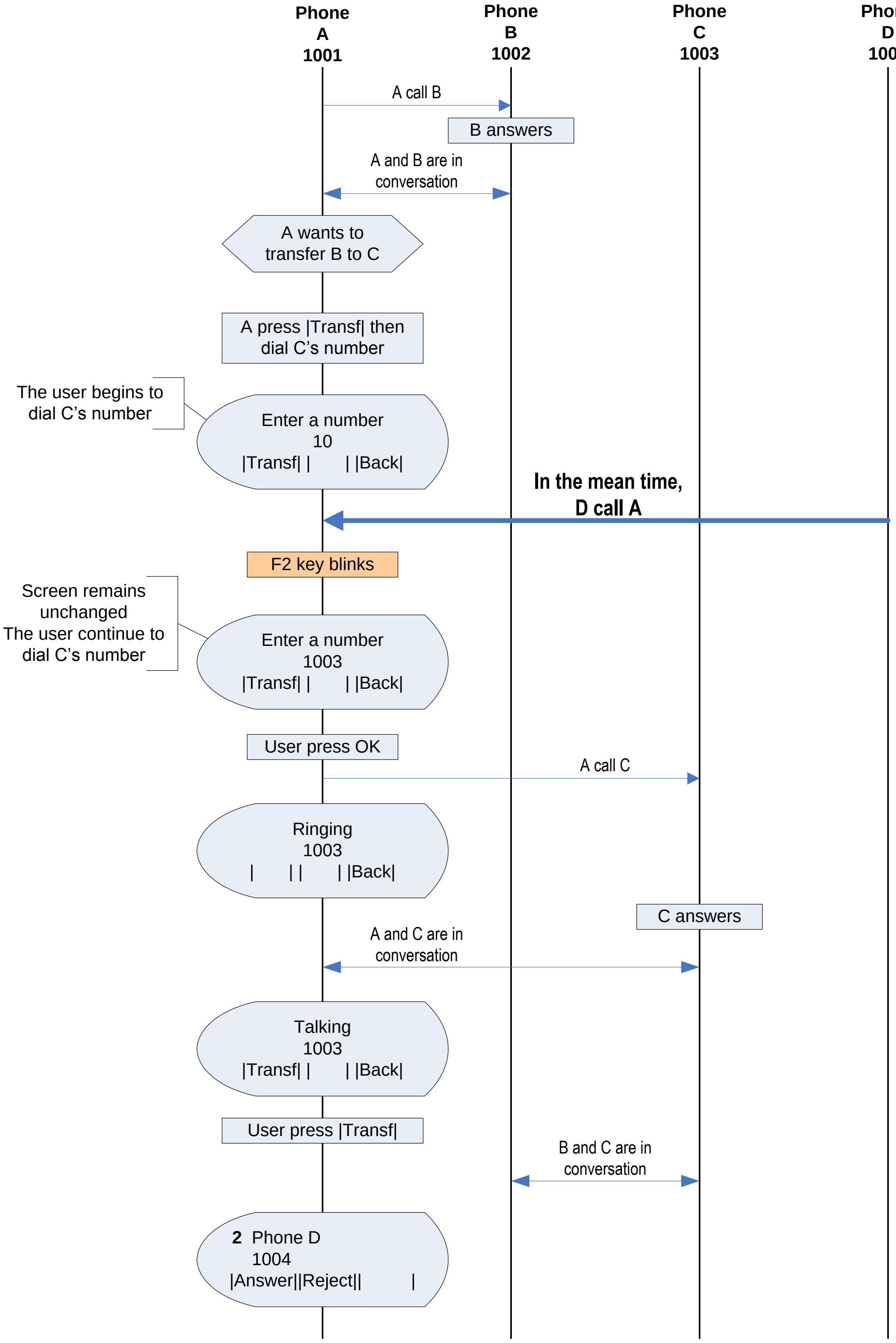
Once the transfer finished, the phone will show back to the incoming call screen. Then the user can answer to the 2nd call.

Remarks:
In the call flows, the following cases are not shown. As it is assumed the following behaviour is acceptable as it is and do not need to be modified.

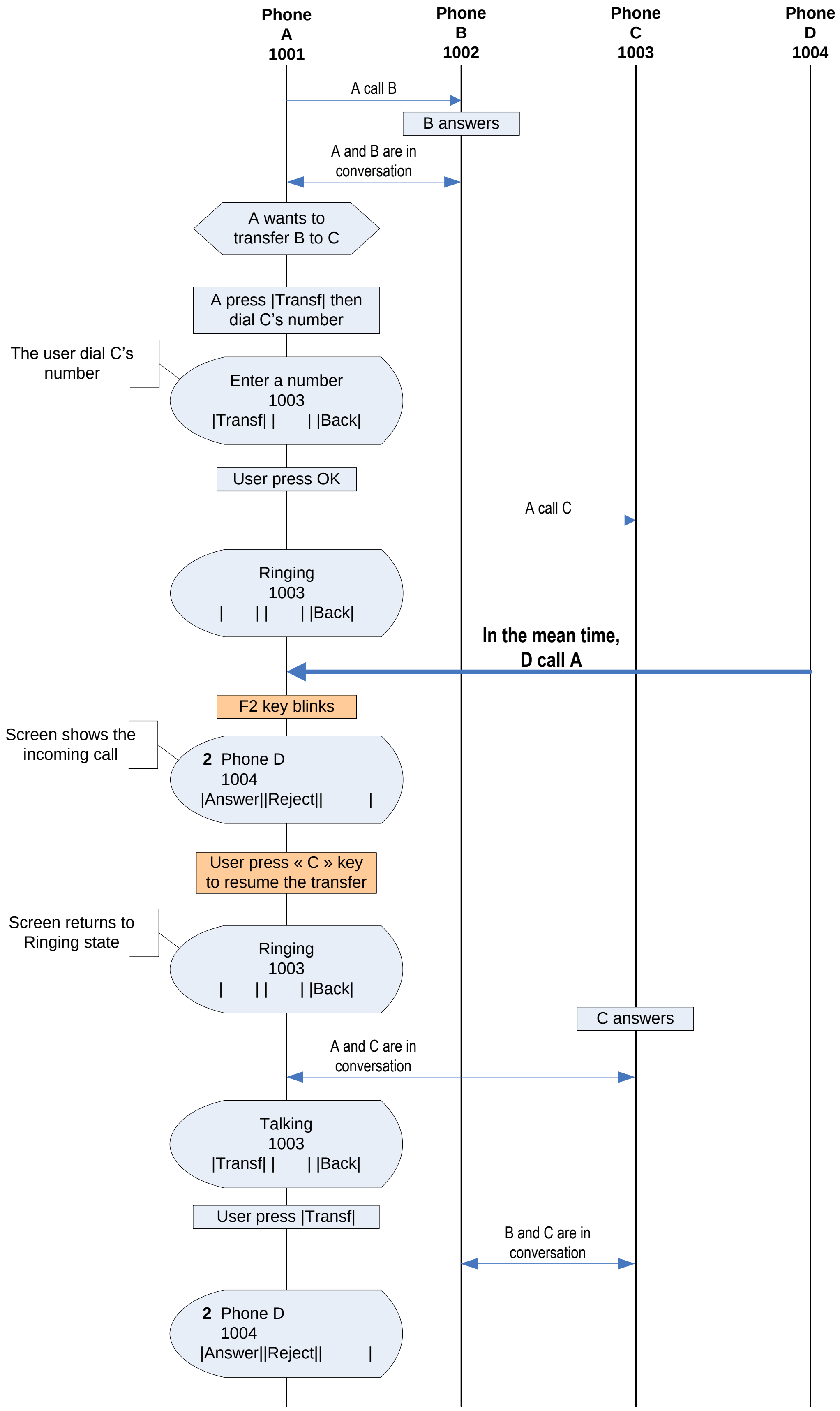
-if user A answers the call from D, in fact the transfer is cancelled; A keeps the call with B only with B held

-if user A rejects the call from D, transfer screen is resumed

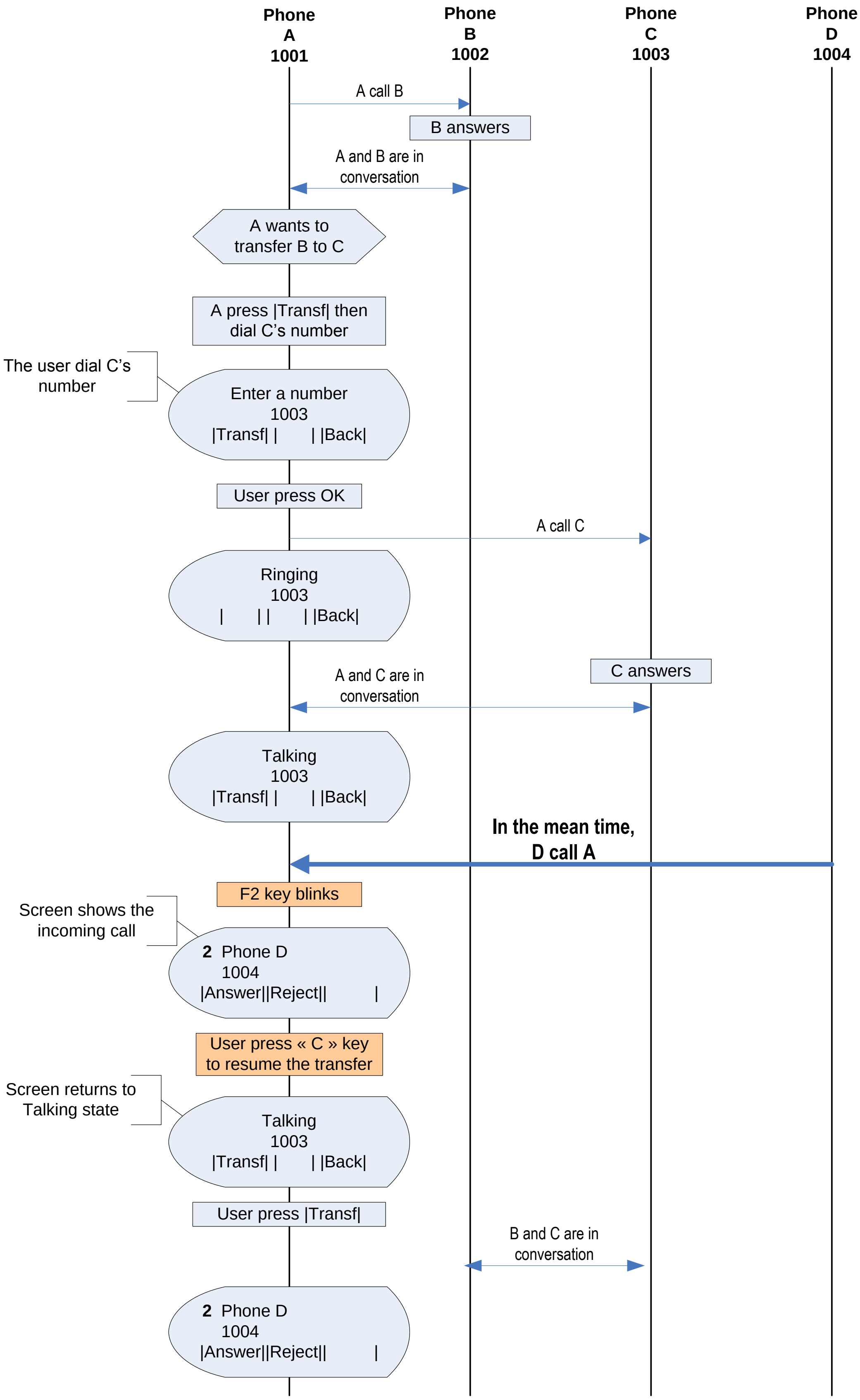
Call flow 1:



Call flow 2:


Phone A 1001
Phone B 1002
Phone C 1003
Phone D 1004
A call B
B answers
A and B are in conversation
A wants to transfer B to C
A press |Transf| then dial C's number
The user dial C's number
Enter a number
1003
|Transf| | | |Back|
User press OK
A call C
Ringing
1003
| | | | |Back|
In the mean time, D call A
F2 key blinks
Screen shows the incoming call
2 Phone D
1004
|Answer||Reject|| |
User press « C » key to resume the transfer
Screen returns to Ringing state
Ringing
1003
| | | | |Back|
C answers
A and C are in conversation
Talking
1003
|Transf| | | |Back|
User press |Transf|
B and C are in conversation
2 Phone D
1004
|Answer||Reject|| |

Call flow 3:


Phone A 1001
Phone B 1002
Phone C 1003
Phone D 1004
A call B
B answers
A and B are in conversation
A wants to transfer B to C
A press |Transf| then dial C's number
The user dial C's number
Enter a number
1003
|Transf| | | |Back|
User press OK
A call C
Ringing
1003
| | | | |Back|
C answers
A and C are in conversation
Talking
1003
|Transf| | | |Back|
In the mean time, D call A
F2 key blinks
Screen shows the incoming call
2 Phone D
1004
|Answer||Reject|| |
User press « C » key to resume the transfer
Screen returns to Talking state
Talking
1003
|Transf| | | |Back|
User press |Transf|
B and C are in conversation
2 Phone D
1004
|Answer||Reject|| |

2- Blind Transfer

When the user initiates a blind transfer, the phone launches a “recovery timer”. This timer ends if the 2nd correspondent answers and transfer is then completed or if first dialog is recovered due to failure in the transfer or timer expiration.

During this timer, in current implementation the transfer is not able to answer to any incoming call, which is not a desirable behaviour in many environments.
When an incoming call arrives, it is automatically placed on the first available line. If the user press “C” key, an early BYE shall be sent to end the dialog with the 1st correspondent.

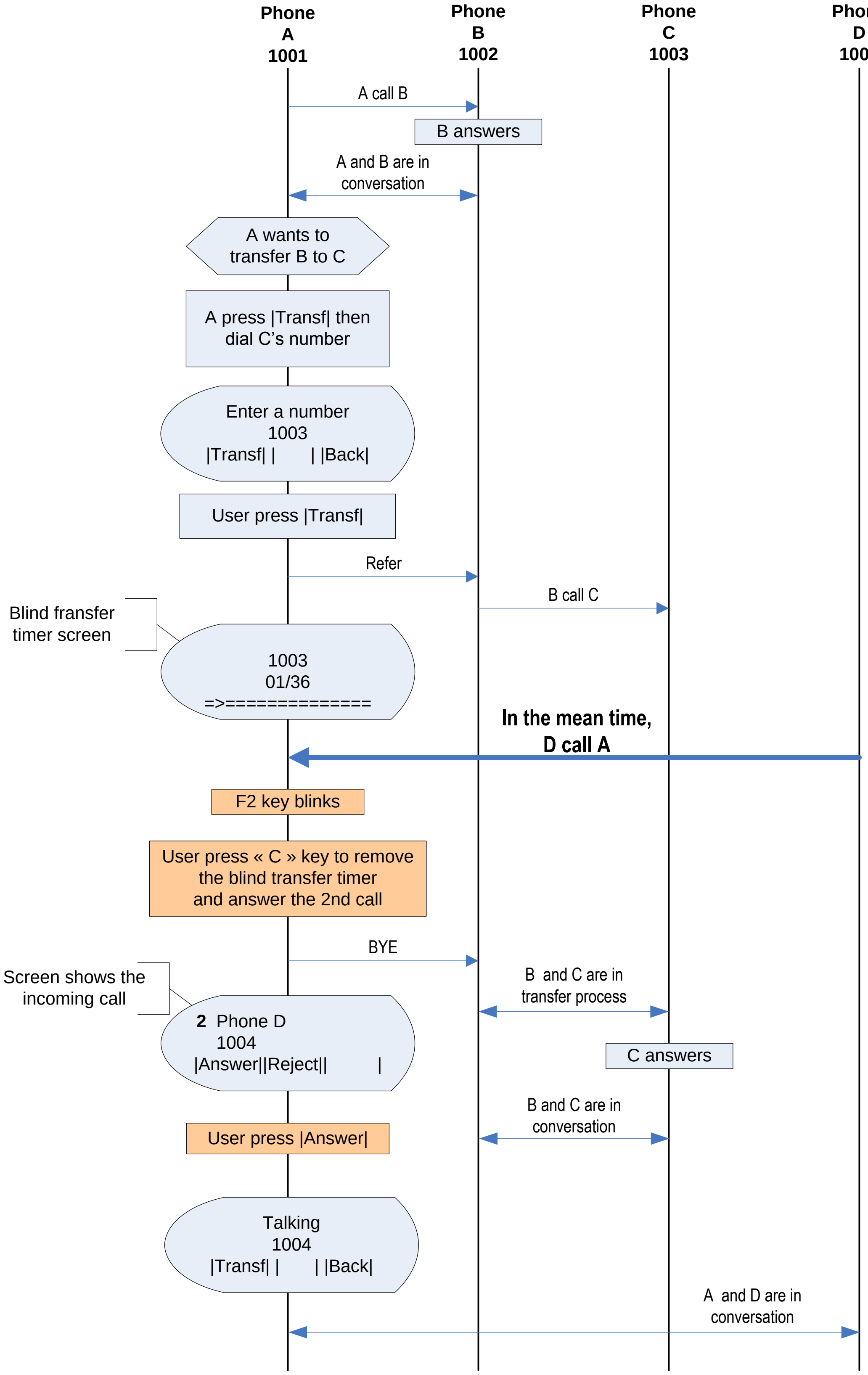
Now, the early BYE must be sent only after the transferee has accepted the REFER to avoid to cancel the on-going transfer.

Doing that, it is assumed there’s no way to recover the 1st call if, for any reason, the transfer fails.

The display will show the incoming call screen. Then the user can answer to the 2nd call

See example below:

Call flow 4:


Phone
A
1001
Phone
B
1002
Phone
C
1003
Phone
D
1004
A call B
B answers
A and B are in conversation
A wants to transfer B to C
A press |Transf| then dial C's number
Enter a number
1003
|Transf| | | |Back|
User press |Transf|
Refer
B call C
Blind fransfer timer screen
1003
01/36
=>==============
In the mean time, D call A
F2 key blinks
User press « C » key to remove the blind transfer timer and answer the 2nd call
BYE
Screen shows the incoming call
2 Phone D
1004
|Answer||Reject|| |
B and C are in transfer process
C answers
B and C are in conversation
User press |Answer|
Talking
1004
|Transf| | | |Back|
A and D are in conversation

Number display in Standby

St2030s is able to show a different number in standby that the one used for registration.

For this purpose, two new parameters have been included in section [sip] of common or MAC config files in each profile.

DisplayNumFlag value is disable by default

DisplayNum could be any char. The maximum number of digits is 20. It shall not replace any parameter of the sip profile (display name, phone number, auth name), it's only affecting the displayed info on the screen.

The default setting would be:
DisplayNumFlagX=0 , where X is profile 1,2,3,4
DisplayNumX=

Example:
DisplayNumFlag1=1
DisplayNum1=2001

On profile 1, the option of number display is enable. As a result, the 2001 will be used and replaced the default telephone number displaying on the phone screen.

Remark:
Hide_Phone_Number_Display have priority over DisplayNumFlagX.
Hide_Phone_Number_Display=1 and DisplayNumFlagX =1. Number is Not Show in strandby.

" DisplayName" is still shown on the screen and in the same position, independently of DisplayNumFlagX and DisplayNumX value.

Feature Activation

A) Through APS:

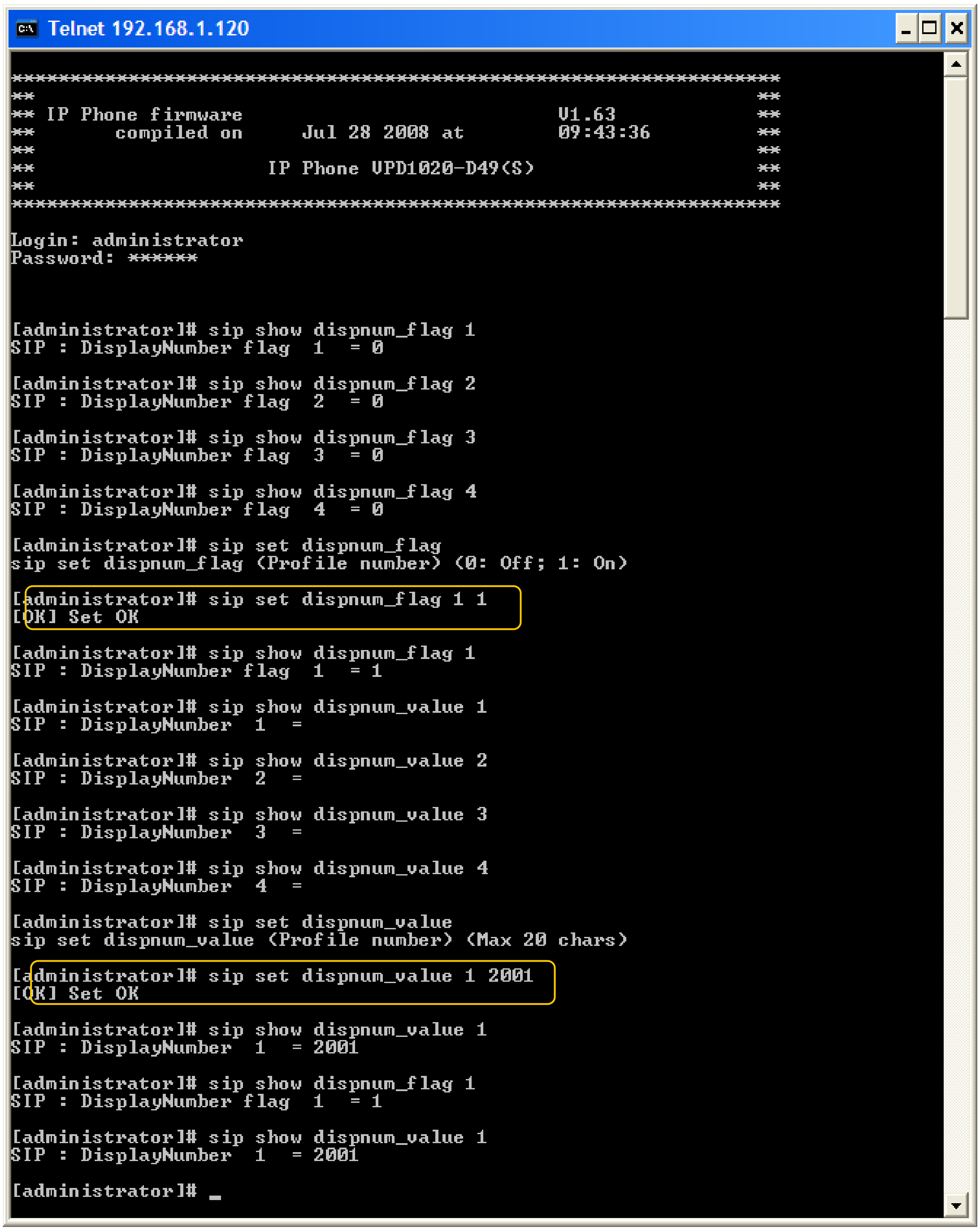
In Common or Mac Config file, user is enable or disable the number display flag to 0 or 1 respectively. Default value is 0.

```
[sip]
…
DisplayNumFlag1=0 (Default telephone number will be used and displayed on screen)
DisplayNumFlag2=0
DisplayNumFlag3=0
DisplayNumFlag4=1 (DisplayNum will be used and replace the default number displaying on
                   the screen)
DisplayNum1=
```

DisplayNum2=
DisplayNum3=
DisplayNum4=2001 (max digits is 20)
…

B) Through Telnet:

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                              V1.63         **
**       compiled on      Jul 28 2008 at          09:43:36      **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show dispnum_flag 1
SIP : DisplayNumber flag  1  = 0

[administrator]# sip show dispnum_flag 2
SIP : DisplayNumber flag  2  = 0

[administrator]# sip show dispnum_flag 3
SIP : DisplayNumber flag  3  = 0

[administrator]# sip show dispnum_flag 4
SIP : DisplayNumber flag  4  = 0

[administrator]# sip set dispnum_flag
sip set dispnum_flag (Profile number) (0: Off; 1: On)

[administrator]# sip set dispnum_flag 1 1
[OK] Set OK

[administrator]# sip show dispnum_flag 1
SIP : DisplayNumber flag  1  = 1

[administrator]# sip show dispnum_value 1
SIP : DisplayNumber  1  =

[administrator]# sip show dispnum_value 2
SIP : DisplayNumber  2  =

[administrator]# sip show dispnum_value 3
SIP : DisplayNumber  3  =

[administrator]# sip show dispnum_value 4
SIP : DisplayNumber  4  =

[administrator]# sip set dispnum_value
sip set dispnum_value (Profile number) (Max 20 chars)

[administrator]# sip set dispnum_value 1 2001
[OK] Set OK

[administrator]# sip show dispnum_value 1
SIP : DisplayNumber  1  = 2001

[administrator]# sip show dispnum_flag 1
SIP : DisplayNumber flag  1  = 1

[administrator]# sip show dispnum_value 1
SIP : DisplayNumber  1  = 2001

[administrator]# _
```

List oriented BLF V2 (ST2030 only)

List-oriented BLF is already supported by ST2030. In Broadsoft environment, this feature allowed so far monitoring other line status, and speed dialing.

Call capture was not possible on FW 1.62.3 or previous version. A number of improvements have been done on server side.

To benefit from them, some changes have been done due to improvements in Broadsoft implementation (R14).

LED state:

If [in a monitoring NOTIFY] for the same <resource uri> you get several <state> tags (different dialogs):

- LED blinking - if any of the state tags for that resource uri is "Proceeding" and direction=recipient, a global Proceeding state will be signaled

```
<?xml version="1.0" encoding="UTF-8"?><dialog-info
xmlns="urn:ietf:params:xml:ns:dialog-info" version="6" state="full"
entity="thomson2@as.iop1.broadworks.net"><dialog id="bG9jYWxIb3N0NDU5NDY5OjA="
direction="recipient"><state>proceeding</state>
<local><identity display="Broadsoft
SIP2">sip:thomson2@as.iop1.broadworks.net</identity><identity display="Broadsoft
SIP2">tel:+12408881412;ext=1412</identity></local>
<remote><identity display="BroadSoft
SIP4">sip:1414@as.iop1.broadworks.net;user=phone</identity></remote>
</dialog></dialog-info>
```

- LED steady lit - if no Proceeding (direction=recipient) tags are present, and any of the state tags for that resource uri is Proceeding with direction = originator or any of the state tags is Confirmed

```
<?xml version="1.0" encoding="UTF-8"?><dialog-info
xmlns="urn:ietf:params:xml:ns:dialog-info" version="11" state="full"
entity="thomson4@as.iop1.broadworks.net"><dialog id="bG9jYWxIb3N0NDU5NDY4OjA="
direction="initiator"><state>confirmed</state>
<local><identity display="BroadSoft
SIP4">sip:thomson4@as.iop1.broadworks.net</identity><identity display="BroadSoft
SIP4">tel:+12408881414;ext=1414</identity></local>
<remote><identity display="Broadsoft
SIP1">sip:1411@as.iop1.broadworks.net;user=phone</identity></remote>
</dialog></dialog-info>
```

- LED off - if all the state tags are Terminated

```
<?xml version="1.0" encoding="UTF-8"?><dialog-info
xmlns="urn:ietf:params:xml:ns:dialog-info" version="7" state="full"
entity="thomson2@as.iop1.broadworks.net"><dialog id="bG9jYWxIb3N0NDU5NDY5OjA="
direction="recipient"><state>terminated</state>

<local><identity display="Broadsoft
SIP2">sip:thomson2@as.iop1.broadworks.net</identity><identity display="Broadsoft
SIP2">tel:+12408881412;ext=1412</identity></local>
</dialog></dialog-info>
```

User action

There are three different statuses defined when the BLF Key is pressed:

1. LED in idle - it causes speed dial provisioned number
2. LED steady on – it causes speed dial provisioned number
3. LED blinking –it causes launch call pickup (See below 3.3)

Call pickup (directed pickup)

The method to launch pickup will depend on the local/sc configuration flag.

The number used to launch pickup when BLF key is pressed in led blinking state will be taken from the dialog-info section of the received NOTIFY.

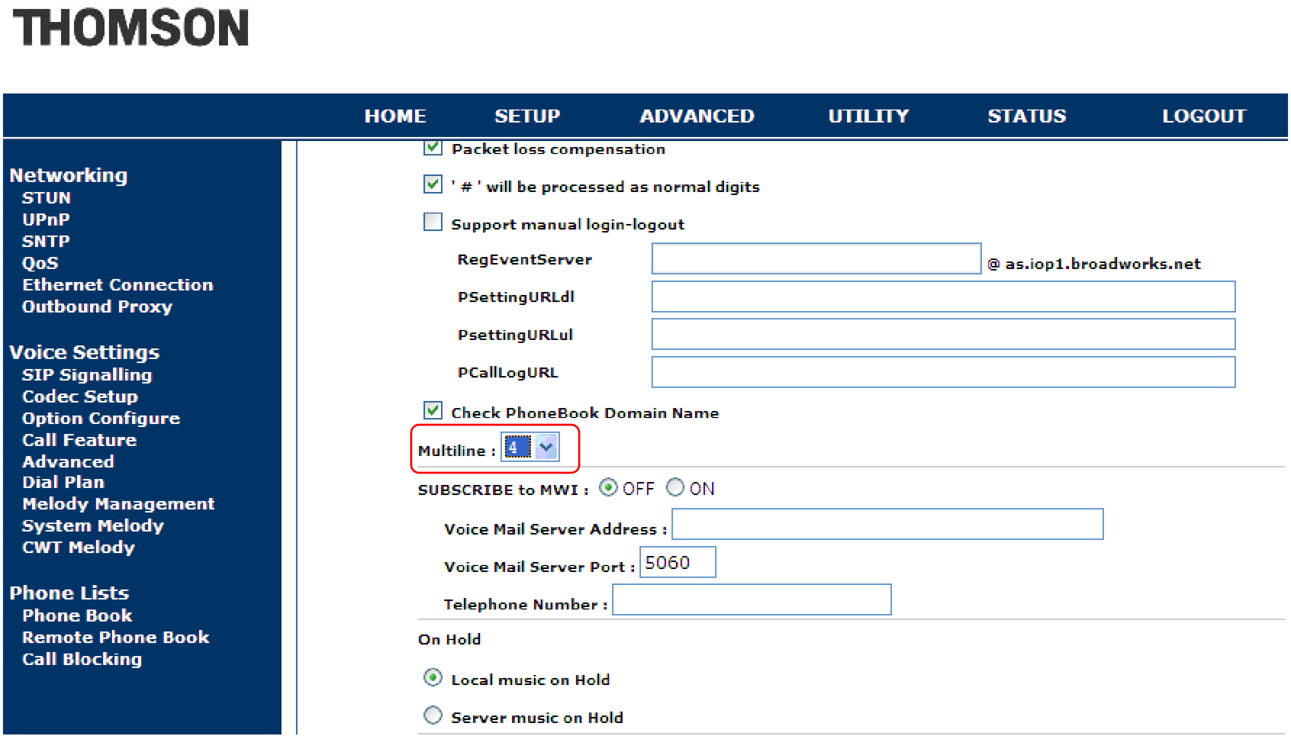
The starcode used for pickup will be in accordance with your server configuration.

Feature Activation

A) Through WebGui:

In order to have List-oriented BLF active, APS or web gui can be used. Please note default mode for BLF is User-Based.

First you have to decrease Multiline to reserve the number of supervised lines in Function Key, in Advanced → Advanced.



☑ Packet loss compensation

☑ ' # ' will be processed as normal digits

☐ Support manual login-logout

RegEventServer @ as.iop1.broadworks.net

PSettingURLdl

PsettingURLul

PCallLogURL

☑ Check PhoneBook Domain Name

Multiline : 4

SUBSCRIBE to MWI : ● OFF ○ ON

Voice Mail Server Address :

Voice Mail Server Port : 5060

Telephone Number :

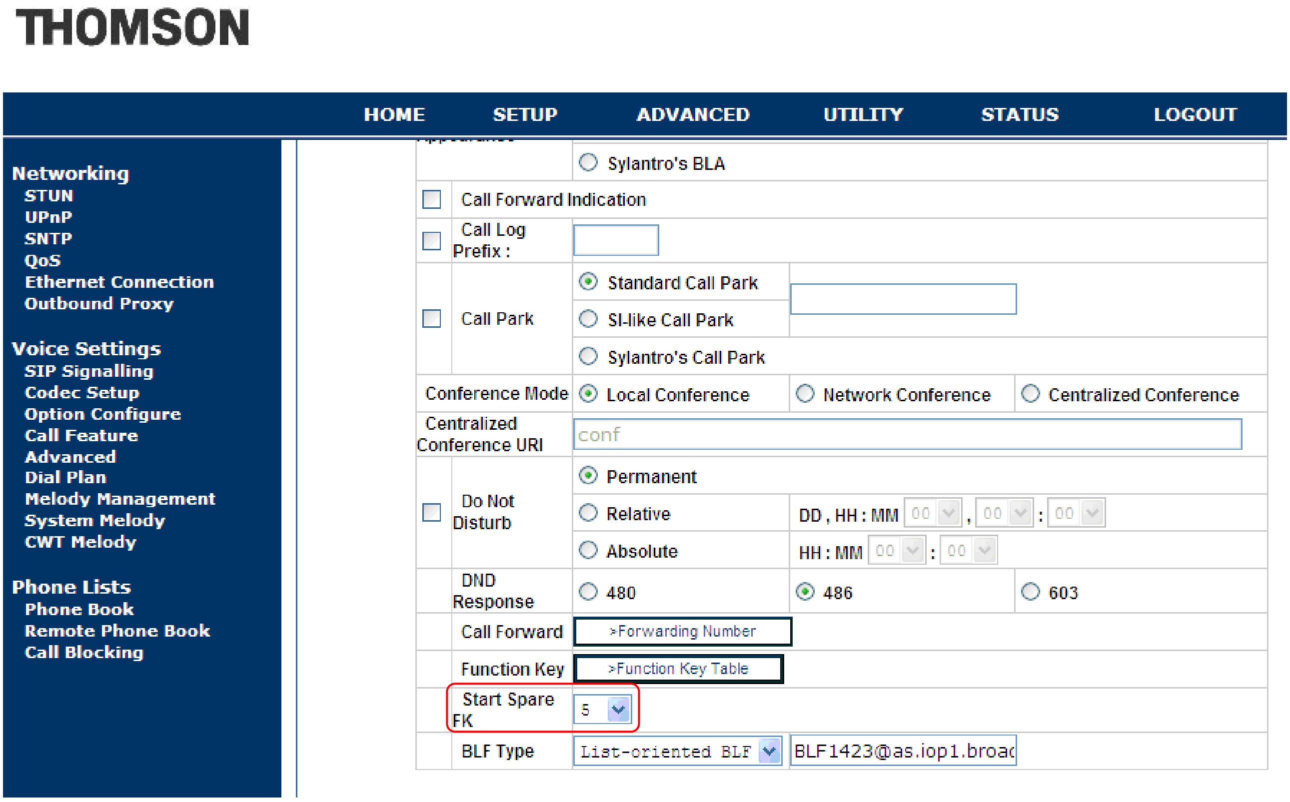
On Hold

● Local music on Hold

○ Server music on Hold

Then go to Advanced → Call feature. Parameter "Start Spare Fk" indicates the first function key which will be dynamically provisioned.

This parameter, by default, is automatically set to Multiline+1. If you want to keep some keys reserved for speeddial, please change this value



○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

☐ Call Park: ● Standard Call Park ○ SI-like Call Park ○ Sylantro's Call Park

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

☐ Do Not Disturb: ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00

DND Response: ○ 480 ● 486 ○ 603

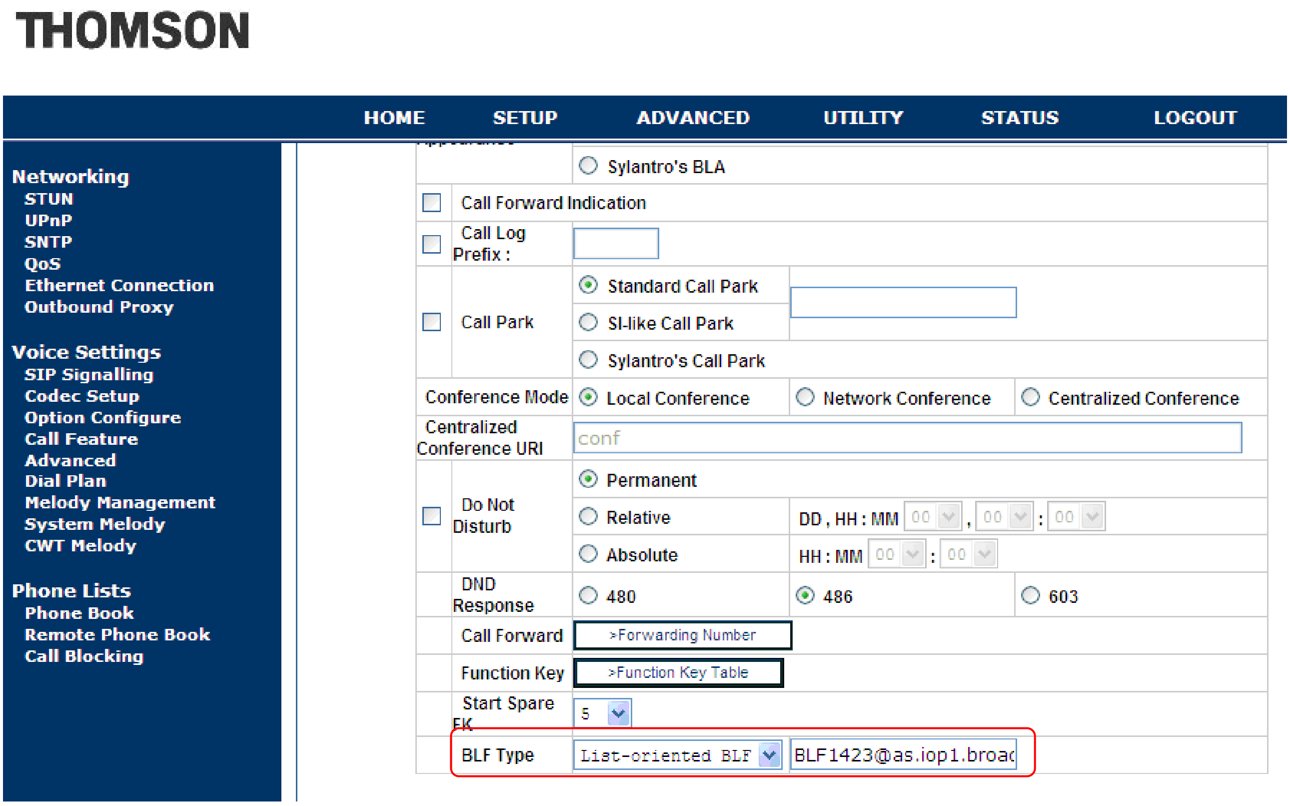
Call Forward

Function Key

Start Spare FK: 5

BLF Type: List-oriented BLF BLF1423@as.iop1.broad

Select "List-oriented BLF", and configure the List-uri to which the phone should subscribe, in accordance with your server configuration



○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

☐ Call Park: ● Standard Call Park ○ SI-like Call Park ○ Sylantro's Call Park

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

☐ Do Not Disturb: ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00

DND Response: ○ 480 ● 486 ○ 603

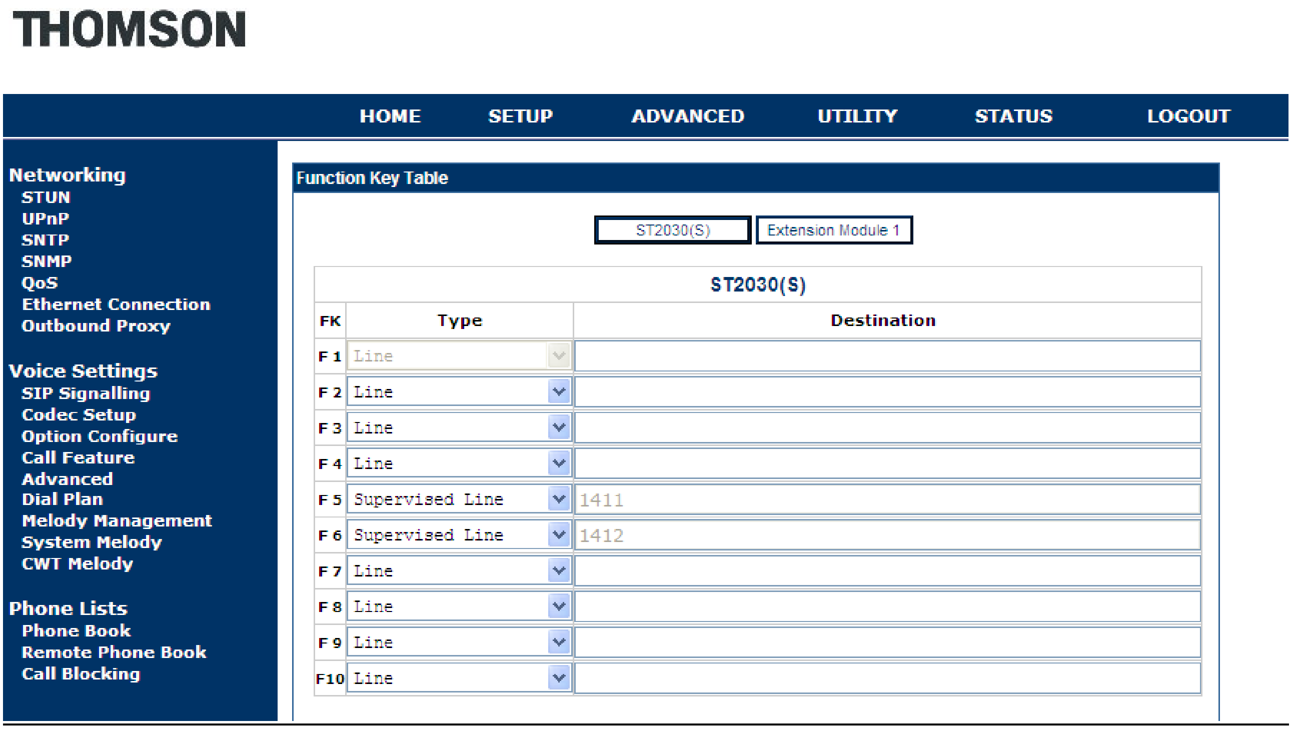
Call Forward

Function Key

Start Spare FK: 5

BLF Type: List-oriented BLF BLF1423@as.iop1.broac

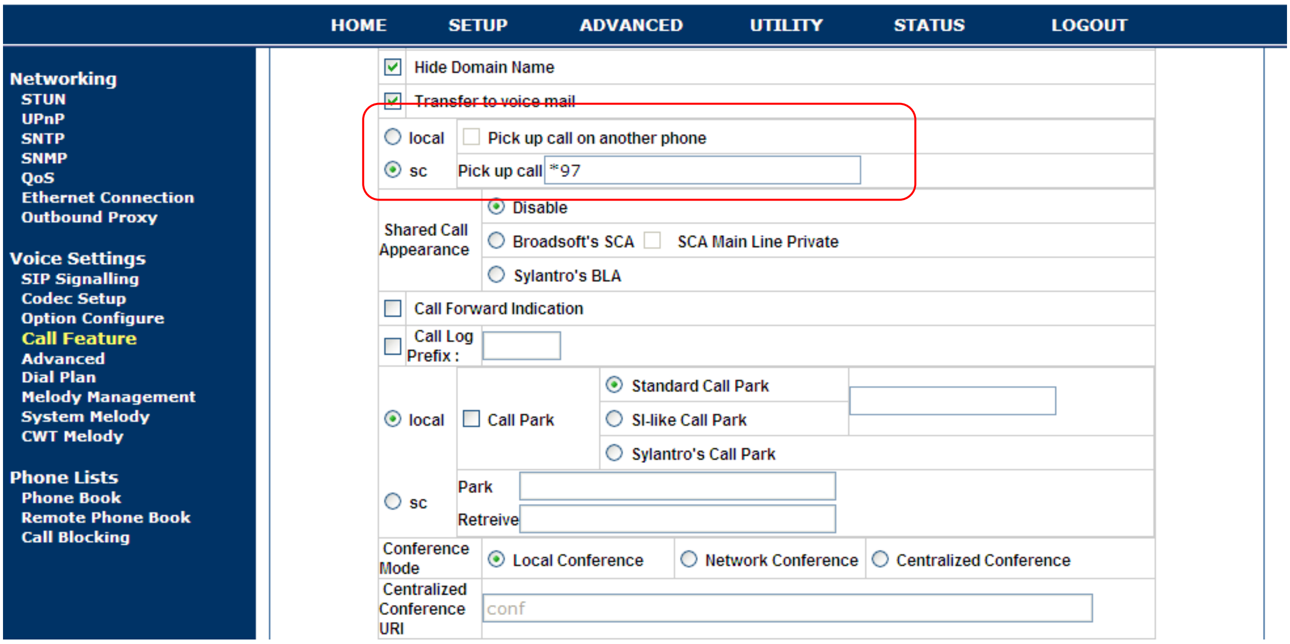
Then, you can check the configuration through Advanced → Call feature → Function Key table.



ST2030(S)

FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Line	
F 4	Line	
F 5	Supervised Line	1411
F 6	Supervised Line	1412
F 7	Line	
F 8	Line	
F 9	Line	
F10	Line	

To enable the call pickup feature from function key BLF, please enter the starcode defined in your server.



☑ Hide Domain Name

☑ Transfer to voice mail

○ local ☐ Pick up call on another phone

● sc Pick up call *97

Shared Call Appearance: ● Disable ○ Broadsoft's SCA ☐ SCA Main Line Private ○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

● local ☐ Call Park ● Standard Call Park ○ SI-like Call Park ○ Sylantro's Call Park

○ sc Park Retreive

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

Please refer to "*ST20xx SIP New Features SG vx.62*" P.145 → How to activate call Pick-Up Service.

B) <u>Through APS:</u>
The parameters governing this feature, both included in the [sys] section, are:

BLFType
Value : 0 or 1
Meaning : 0 : User-oriented BLF is active
1 : List-oriented BLF is active
Default value: 0

BLFListSipUri
Value: L/<sip:user@host>
Meaning: contains the URI of the List the phone will need to subscribe in order to monitor users. This URI has to be provided by your sip server administrator.

StartSpareFK
Value : numeric, from Current_Max_Multiline+1 to the number of function keys available
Meaning : n : Fn will be the first position to be automatically filled in via List-oriented BLF dynamic provisioning.
Default value: Current_Max_Multiline+1

Current_Max_Multiline
Value: 1 to 10
Meaning: number of simultaneous calls the phone will handle. BLF is not possible with keys assigned to multiline, that is why this parameter needs to

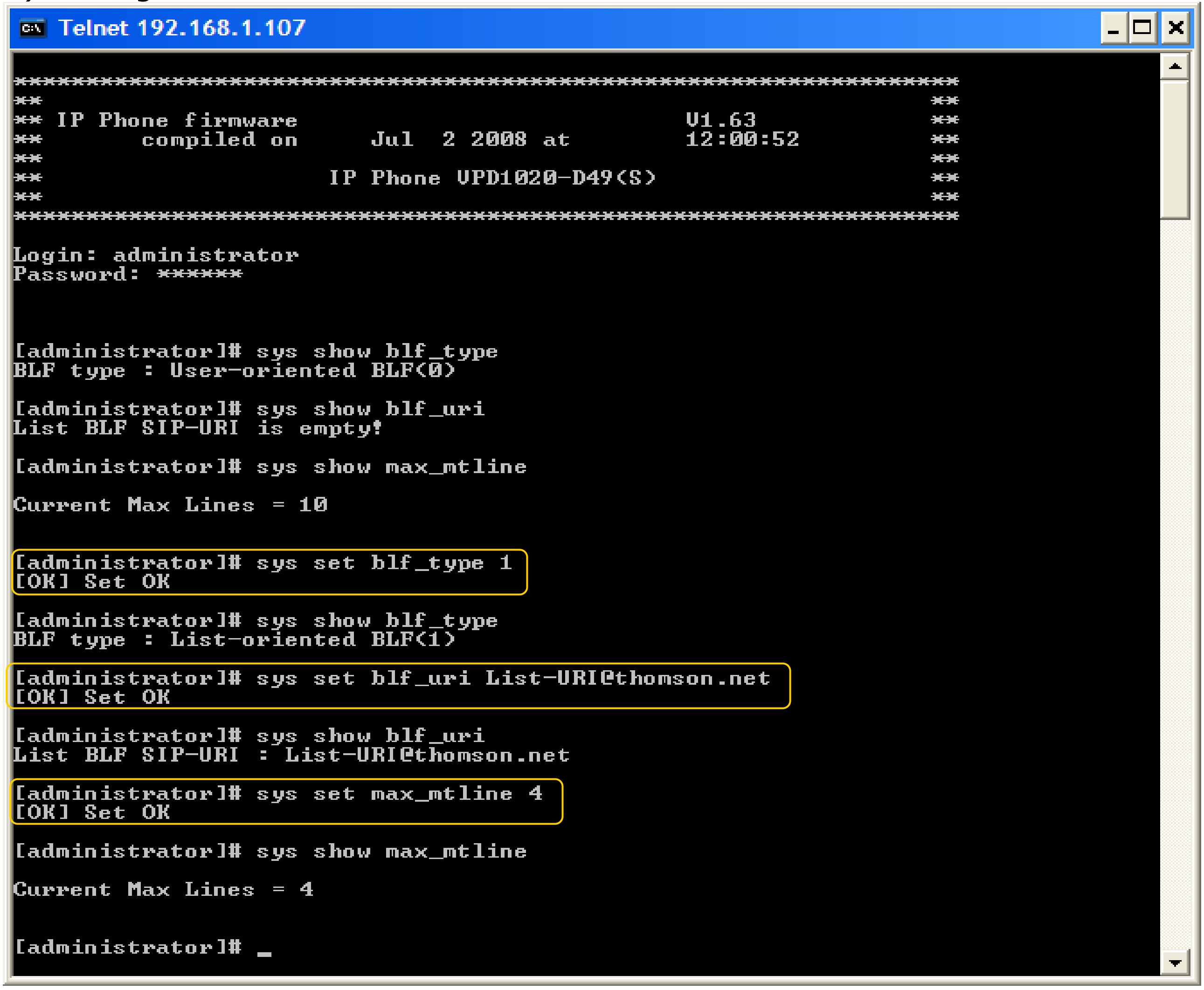
be adjusted in order to use BLF. If you intend to use an extension module for BLF, then you need not change this parameter.

```
[sys]
BLFType=1
BLFListSipUri=L/<sip:List-URI@thomson.net>
Current_Max_Multiline=4
StartSpareFK=6
….
```

With this parameters, List-oriented BLF is enabled, the List Uri is List-URI@thomson.net and the first key to be provisioned is F6

For the call pick up service, please refer to "ST20xx SIP New Features SG vx.62" P.145 → How to activate call Pick-Up Service.

C) Through Telnet:



```
*********************************************************************
**                                                                 **
** IP Phone firmware                            V1.63              **
**       compiled on      Jul  2 2008 at        12:00:52           **
**                                                                 **
**                     IP Phone VPD1020-D49(S)                     **
**                                                                 **
*********************************************************************

Login: administrator
Password: ******


[administrator]# sys show blf_type
BLF type : User-oriented BLF(0)

[administrator]# sys show blf_uri
List BLF SIP-URI is empty!

[administrator]# sys show max_mtline

Current Max Lines = 10

[administrator]# sys set blf_type 1
[OK] Set OK

[administrator]# sys show blf_type
BLF type : List-oriented BLF(1)

[administrator]# sys set blf_uri List-URI@thomson.net
[OK] Set OK

[administrator]# sys show blf_uri
List BLF SIP-URI : List-URI@thomson.net

[administrator]# sys set max_mtline 4
[OK] Set OK

[administrator]# sys show max_mtline

Current Max Lines = 4

[administrator]# _
```

Improved Shared Call Appearance (SCA)

Thomson ST20xx currently supports SCA according to Broadsoft R14 specification. However, due to its single line architecture, which only allows 1 SIP account active at a time.

This section addresses the improvements to be done in ST20xx to enhance its support for this feature and its application developed for Broadworks R14 only.

Improvements

1. The phone supports "multi-lines" so that each line will register with the Broadsoft server with different credential and extension number. Registrar will see several registrations. Main line will be the master, and will determine most of the available services.

2. Multiple phones in the shared appearance group will be provisioned with the same "multi-lines" configured on each, and the only difference between the phones is that each phone will have a different extension number configured as its primary line (line 1 for example).

   The primary extension number of any phone will be configured as a "secondary" extension in all other phones. Phone provisioning system allows doing so, but admin has to decide how many lines and how many appearances per line to provide, and generate the appropriate configuration data. Function Key 1 will always correspond to the main line. Available overall number of SCA’s is Multiline-1.

3. Each phone should be able to display the status of all other lines as in standard BLF. It is important to make clear we are talking about SCA in this document, and not about BLF. BLF in Broadsoft is only supported in R13 and beyond, and its implementation and behaviour is different from Asterisk BLF.

   However, SCA framework does include subscriber notification status for each shared call appearance via Call-Info: idle, seized, progressing, alerting, active, held, held=private, bridge-active, bridge-held in R14.

4. Incoming call alerting in idle state:
   When a call comes in to the primary line of a phone, Broadworks sends INVITE to ALL SHARED CALL APPEAREANCES. Hence, all phones sharing the same line will ring when a call is received in idle state. Phones will indicate the incoming call also in the Function Key’s LED which corresponds to that appearance.

5. Behavior when an incoming call is received and another one is active:

   In the event that additional call arrives when the user is already engaged in a call, the phone should provide call waiting indicator. Calls to other lines (than primary line) should be indicated with on-screen display and Function Keys LEDs, with the same approach as above: function key led which corresponds to that appearance is the one to be used.

6. Main Line can be share or private:
   - Phone can have on main private line with Second line: Private Main line is not Share with any others phones.
   - Phone can have on main Share line with Second line: Share Main line is share with others phones.

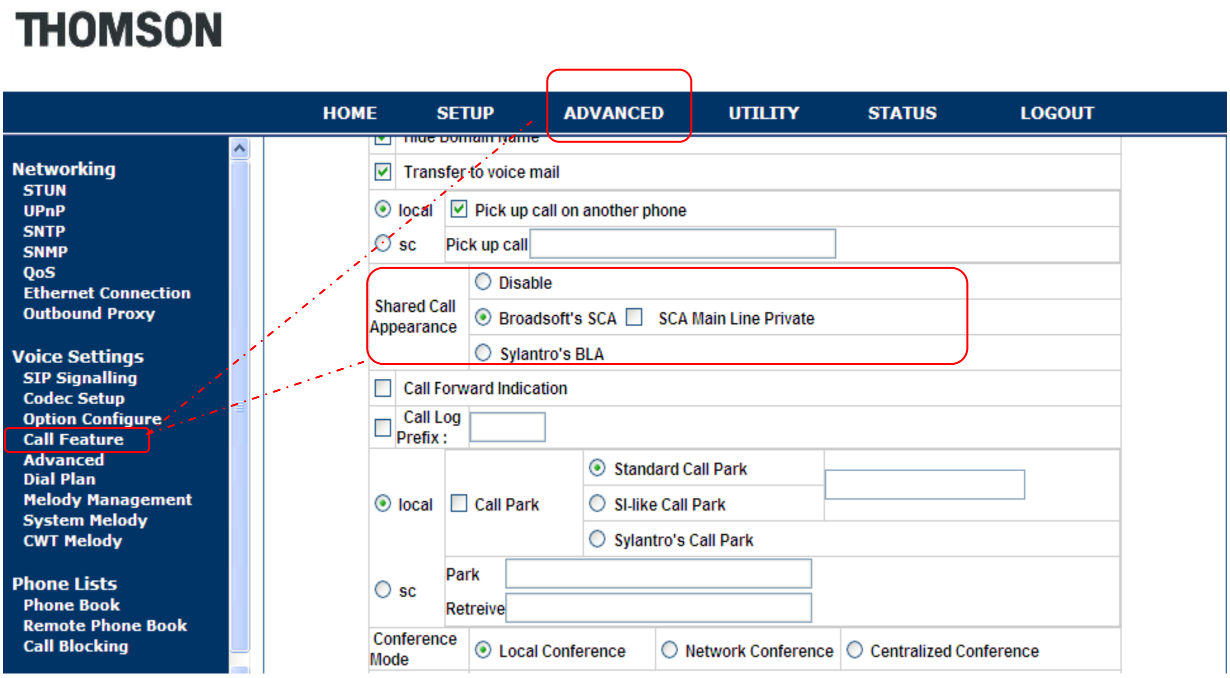
Feature Activation

To activate shared call Appearance with Main line private, a new parameter has created, described as follow:

New Parameter : SCAMainLinePrivate
Existing Parameter : SharedCall Appearance

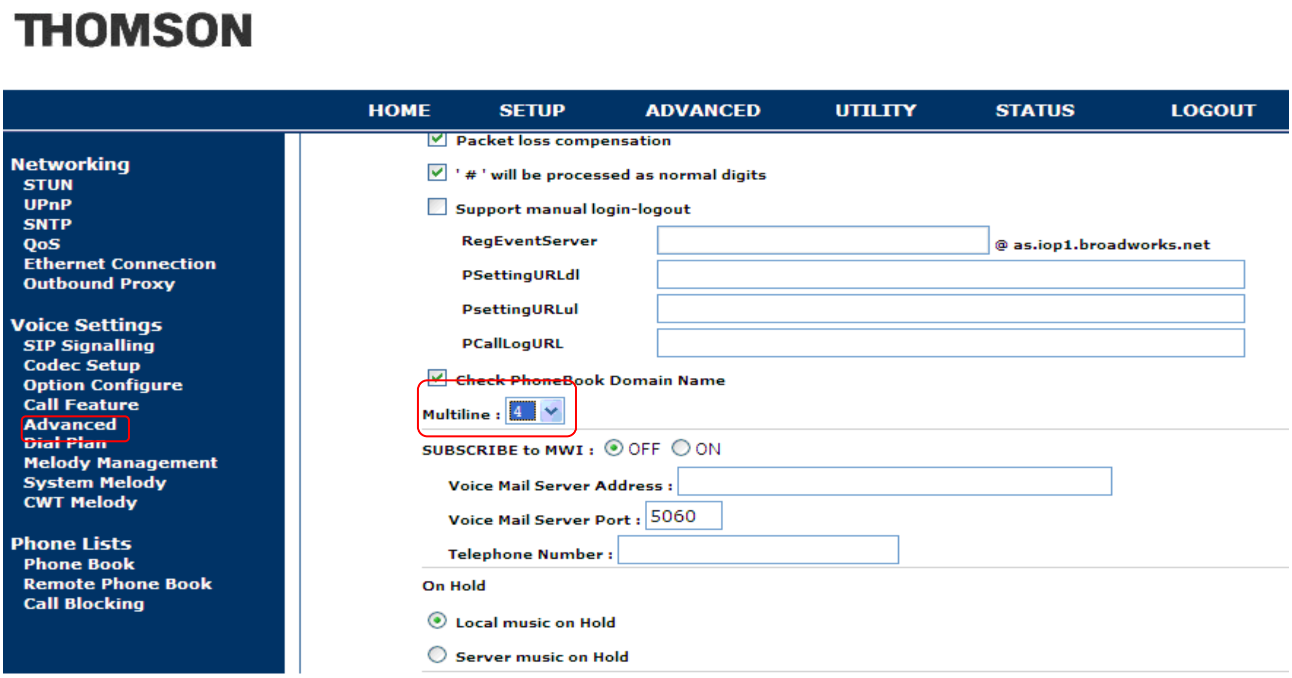
A) Through WebGui:

The activation of this feature is accessible from the web GUI in the Advanced → Call Feature. Then, enable Broadsoft's SCA or Broadsoft's SCA with SCA Main Line Private. (Shared Call Appearance is disable by default)



☑ Transfer to voice mail
● local ☑ Pick up call on another phone
○ sc Pick up call
Shared Call Appearance: ○ Disable ● Broadsoft's SCA ☐ SCA Main Line Private ○ Sylantro's BLA
☐ Call Forward Indication
☐ Call Log Prefix :
● local ☐ Call Park: ● Standard Call Park ○ SI-like Call Park ○ Sylantro's Call Park
○ sc Park Retreive
Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

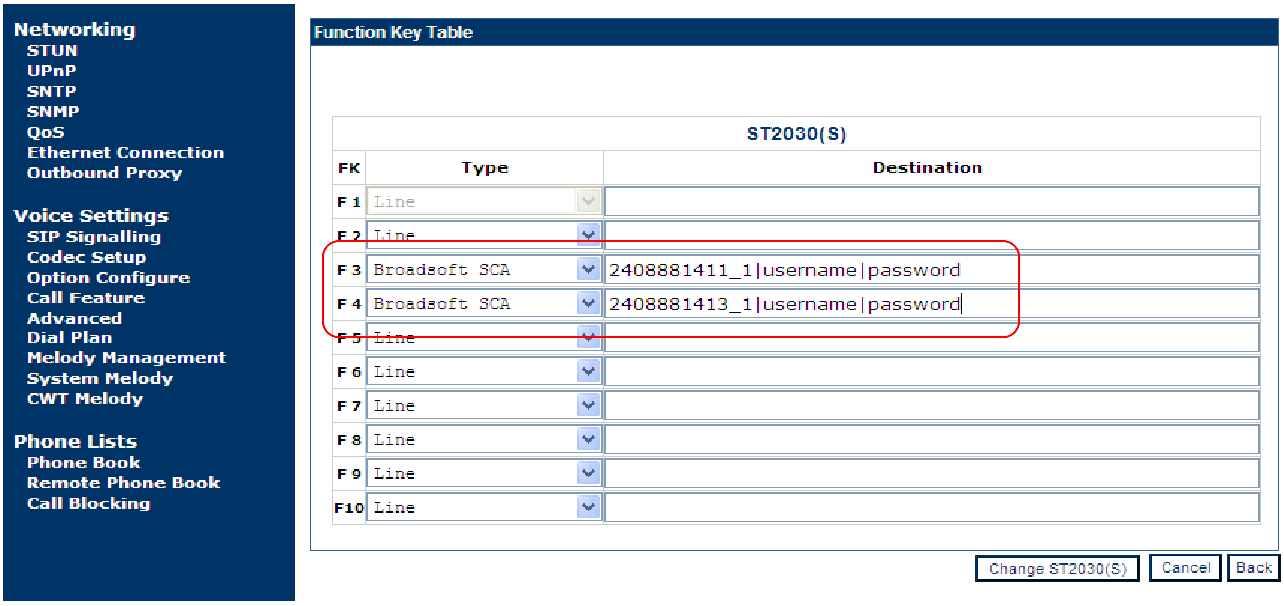
Then, you have to set the Multiline (available overall number of SCA's is Multiline-1), in Advanced → Advanced.
Function Key 1 will always correspond to the main line.



☑ Packet loss compensation
☑ ' # ' will be processed as normal digits
☐ Support manual login-logout
RegEventServer: @ as.iop1.broadworks.net
PSettingURLdl:
PsettingURLul:
PCallLogURL:
☑ Check PhoneBook Domain Name
Multiline : 4
SUBSCRIBE to MWI : ● OFF ○ ON
Voice Mail Server Address :
Voice Mail Server Port : 5060
Telephone Number :
On Hold
● Local music on Hold
○ Server music on Hold

Then go to Advanced → Call feature → Function Key table. An additional item will be created in the roll down "Type" menu: Broadsoft SCA. This roll down type will be shown when existing SharedCallAppearance=1 (Broadsoft SCA)

The function key assigned to an SCA will have the following parameters separated by "|" (vertical bar):
- Line number
- Authentication username
- SIP password



Function Key Table

ST2030(S)

FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Broadsoft SCA	2408881411_1|username|password
F 4	Broadsoft SCA	2408881413_1|username|password
F 5	Line	
F 6	Line	
F 7	Line	
F 8	Line	
F 9	Line	
F10	Line	

B) Through APS:

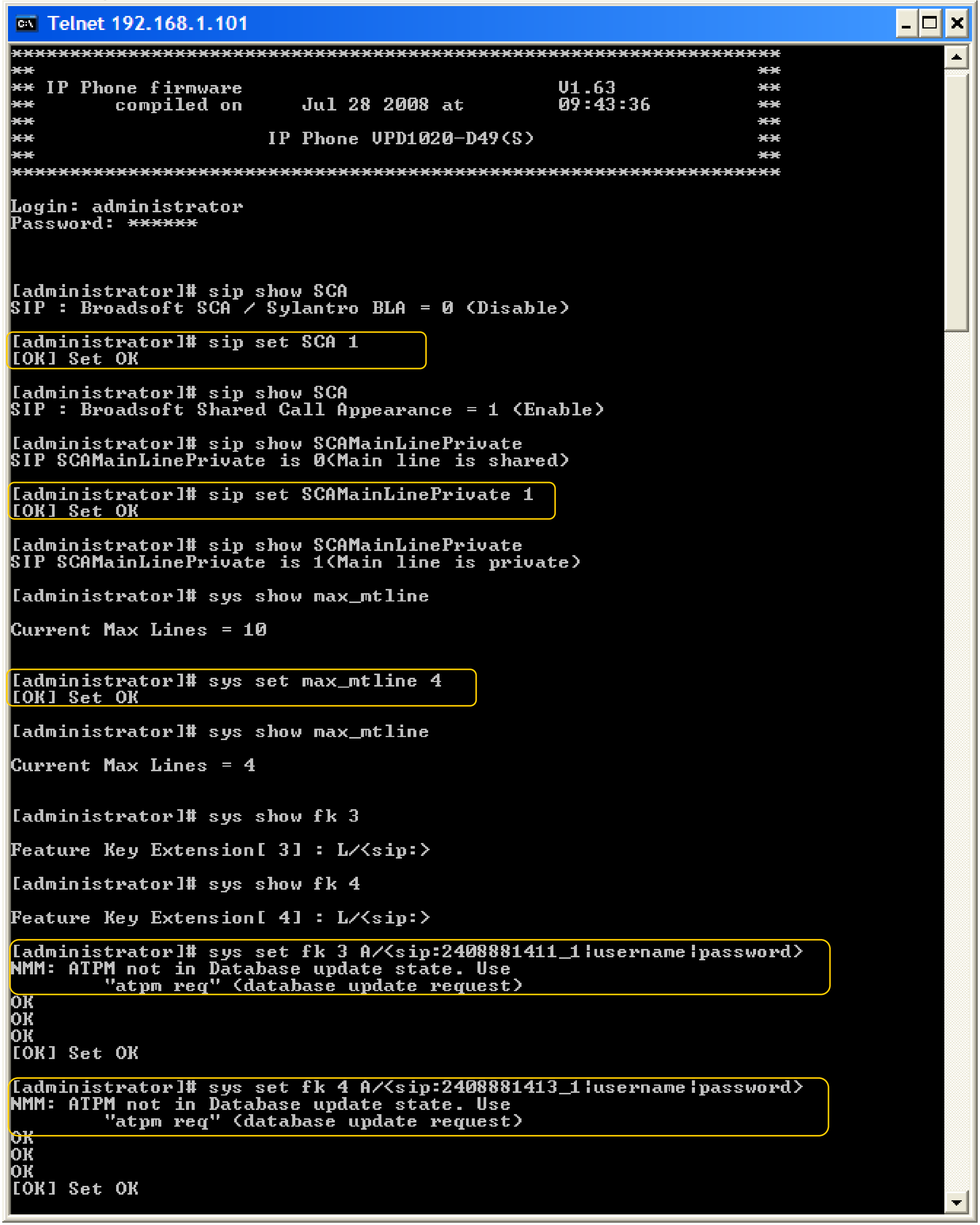
APS in [sip] section of both Common or Specific-MAC config files with the new
Parameter:

- SharedCallAppearance= 0 (0 by default, inactive; 1 to active)
- SCAMainLinePrivate= 0 (0:default, 1: Main line is private, this flag only take effect with SharedCall Appearance=1)

```
[sip]
...
SharedCallAppearance= <0|1>     (0 by default, inactive; 1 to active)
SCAMainLinePrivate=<0|1>        (0: default, main line is also shared; 1: Main line is
                                private
….                              and will not perform any of the Broadsoft SCA
                                procedures)
[sys]
Current_Max_Multiline=4         (available overall number of SCA's is Multiline-1)
FeatureKeyExt03=A/:<sip:2408881411_1|username|password>
FeatureKeyExt04=A/:<sip:2408881413_1|username|password>
…
```

C) Through Telnet:

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.63            **
**       compiled on     Jul 28 2008 at        09:43:36         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: *******


[administrator]# sip show SCA
SIP : Broadsoft SCA / Sylantro BLA = 0 (Disable)

[administrator]# sip set SCA 1
[OK] Set OK

[administrator]# sip show SCA
SIP : Broadsoft Shared Call Appearance = 1 (Enable)

[administrator]# sip show SCAMainLinePrivate
SIP SCAMainLinePrivate is 0(Main line is shared)

[administrator]# sip set SCAMainLinePrivate 1
[OK] Set OK

[administrator]# sip show SCAMainLinePrivate
SIP SCAMainLinePrivate is 1(Main line is private)

[administrator]# sys show max_mtline

Current Max Lines = 10


[administrator]# sys set max_mtline 4
[OK] Set OK

[administrator]# sys show max_mtline

Current Max Lines = 4


[administrator]# sys show fk 3

Feature Key Extension[ 3] : L/<sip:>

[administrator]# sys show fk 4

Feature Key Extension[ 4] : L/<sip:>

[administrator]# sys set fk 3 A/<sip:2408881411_1|username|password>
NMM: ATPM not in Database update state. Use
        "atpm req" (database update request)
OK
OK
OK
[OK] Set OK

[administrator]# sys set fk 4 A/<sip:2408881413_1|username|password>
NMM: ATPM not in Database update state. Use
        "atpm req" (database update request)
OK
OK
OK
[OK] Set OK
```

ST20XX SIP New Features (SG vx.62)

Overview

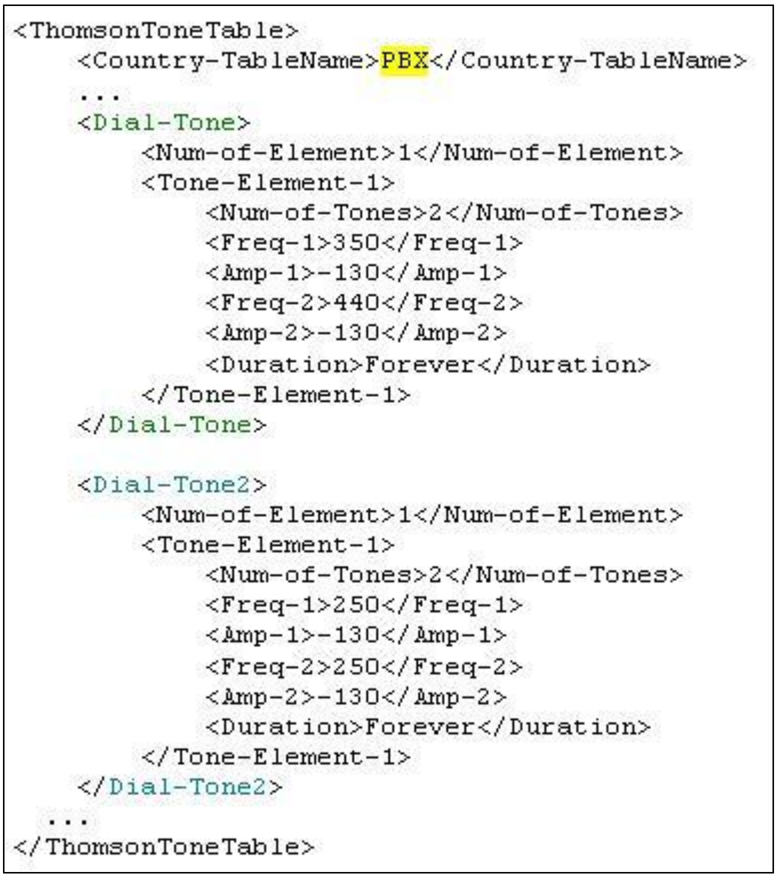
This document describes a set of features included in ST2030 and S2022 SIP v1.62 and 3.62 respectively in order to improve their usability in different environments.

Second dial-tone for PBX-like environments.

This new feature specifies how to simulate the behavior of an analog phone or a “proprietary phone” which is connected behind a PBX.
ST20xx can now generate another dial tone when the user dials the specific PBX prefix.

Refer to “ST20xx SIP New Features SG vx.59” P.200 → How-to Download and Update tone and language tables, to get a dumped tone file from the phone and configure the tones you want for the public network dial-tone and the PBX dial-tone.

As you can see in the dumped “ToneTbl.zz” file, the phone can support 3 kinds of dial tones : <Dial-Tone>, <Dial-Tone2> and <Dial-Tone3>.
<Dial-Tone> and <Dial-Tone2> from the internal ThomsonToneTable will be used to configure the tones we want to play.



```
<ThomsonToneTable>
    <Country-TableName>PBX</Country-TableName>
    ...
    <Dial-Tone>
        <Num-of-Element>1</Num-of-Element>
        <Tone-Element-1>
            <Num-of-Tones>2</Num-of-Tones>
            <Freq-1>350</Freq-1>
            <Amp-1>-130</Amp-1>
            <Freq-2>440</Freq-2>
            <Amp-2>-130</Amp-2>
            <Duration>Forever</Duration>
        </Tone-Element-1>
    </Dial-Tone>

    <Dial-Tone2>
        <Num-of-Element>1</Num-of-Element>
        <Tone-Element-1>
            <Num-of-Tones>2</Num-of-Tones>
            <Freq-1>250</Freq-1>
            <Amp-1>-130</Amp-1>
            <Freq-2>250</Freq-2>
            <Amp-2>-130</Amp-2>
            <Duration>Forever</Duration>
        </Tone-Element-1>
    </Dial-Tone2>
  ...
</ThomsonToneTable>
```

Two new parameters have been created to support this new feature:

1. **PBXconfiguration** , to indicate if the phone is behind a PBX or not.

<Dial-Tone> will be played first when user goes off hook (as it was already the case until this version), no matter the value of this parameter. So, if this parameter is set to 1 (behind PBX), you should modify the <Dial-Tone> table and configure it with your preferred PBX dial-tone.

2. **PBXprefix,** to indicate which is the PBX-prefix to access to the public network (can be 0 to 9, * or #). If the phone is not behind a PBX (previous parameter set to 0), this second parameter will not be taken into account.

If the phone is behind a PBX, the behavior of the phone is as follows:

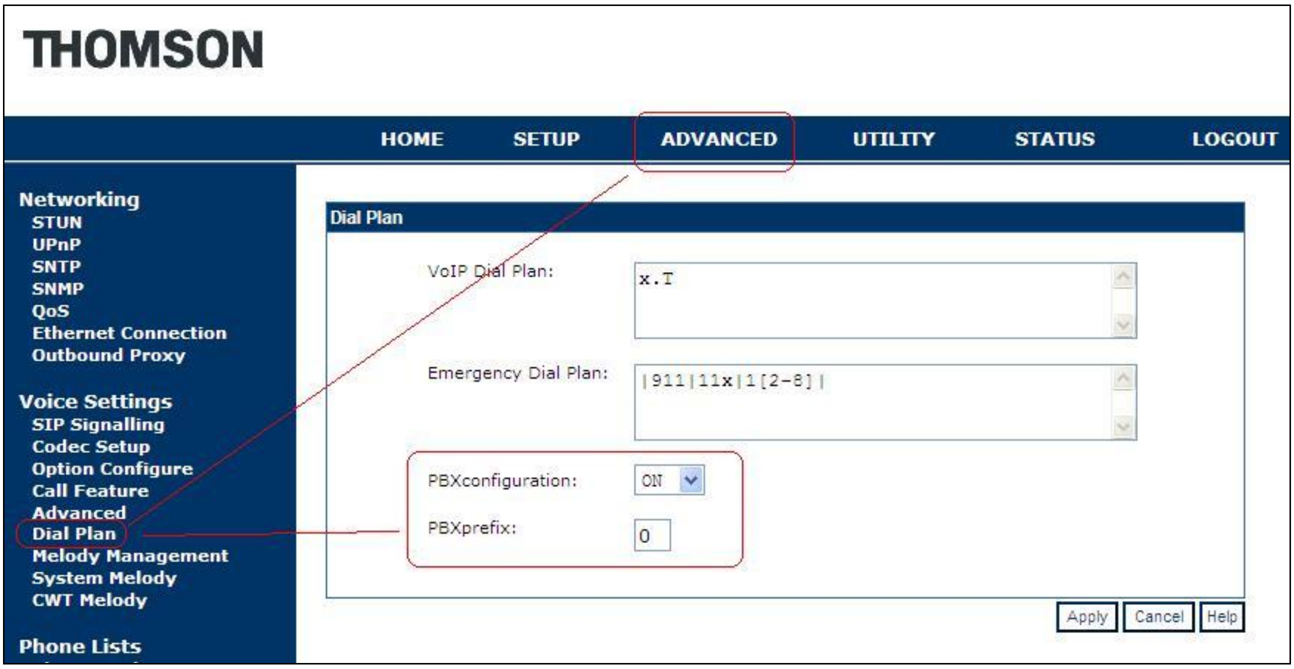
- When the user takes the line, <Dial-Tone> is generated.

- When the user enters the 1st digit:
    - If this (digit = PBXprefix) then <Dial-Tone2> is played until another key is pressed
    - In any other case, <Dial-Tone> is stopped, as current behavior.

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Dial Plan section.



VoIP Dial Plan: x.T

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

PBXconfiguration: ON

PBXprefix: 0

B) Through APS:

For this purpose, two new parameters have been included in the [sys] section of the Common/MAC config files, to be able to activate the configuration of the second dial tone behind a PBX.

```
[sys]
...
PBXconfiguration=1
PBXprefix=0
```

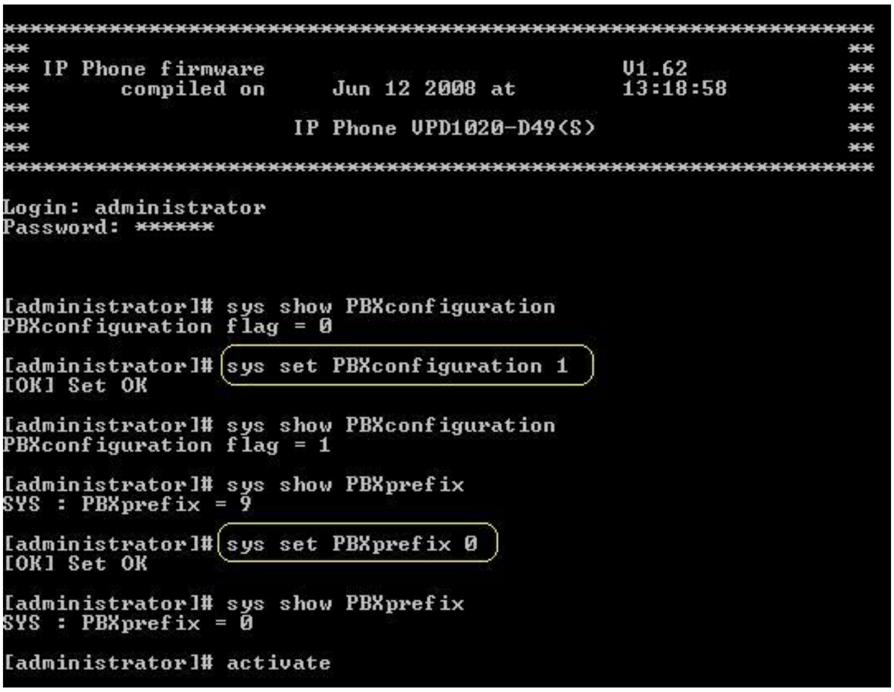
The parameter "**PBXconfiguration**" could be either 0: normal behaviour, 1: phone behind a PBX. The default value is "0".
The parameter "**PBXprefix**" could be either 0 to 9, * or #.

C) Through Telnet:

For this purpose, two new parameters have been created, described as follow:

```
sys set PBXconfiguration (0: default setting   1: phone behind a PBX)
sys set PBXprefix        (0 to 9, * or #)
```

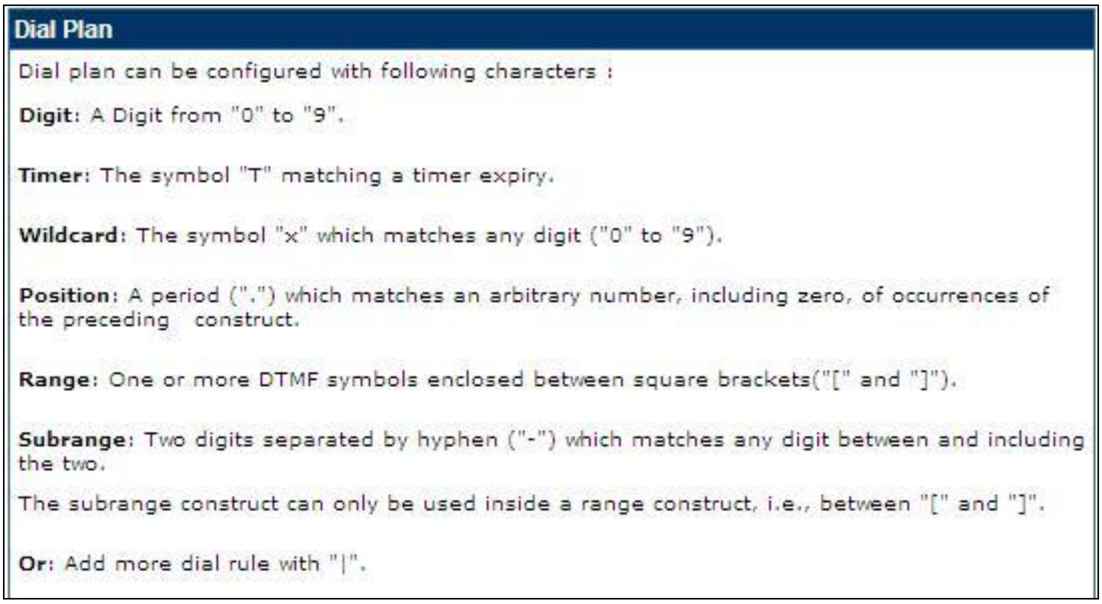
To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.62            **
**       compiled on     Jun 12 2008 at        13:18:58         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sys show PBXconfiguration
PBXconfiguration flag = 0

[administrator]# sys set PBXconfiguration 1
[OK] Set OK

[administrator]# sys show PBXconfiguration
PBXconfiguration flag = 1

[administrator]# sys show PBXprefix
SYS : PBXprefix = 9

[administrator]# sys set PBXprefix 0
[OK] Set OK

[administrator]# sys show PBXprefix
SYS : PBXprefix = 0

[administrator]# activate
```

New specific keys in the Dial Plan.

The following DigitMap syntax definition was used by ST20xx to define when the phone has to stop collecting digits and initiate a call in off hook mode:



Dial Plan

Dial plan can be configured with following characters :

Digit: A Digit from "0" to "9".

Timer: The symbol "T" matching a timer expiry.

Wildcard: The symbol "x" which matches any digit ("0" to "9").

Position: A period (".") which matches an arbitrary number, including zero, of occurrences of the preceding construct.

Range: One or more DTMF symbols enclosed between square brackets("[" and "]").

Subrange: Two digits separated by hyphen ("-") which matches any digit between and including the two.

The subrange construct can only be used inside a range construct, i.e., between "[" and "]".

Or: Add more dial rule with "|".

However it does not cover the specific functions on server side (controlled by n-digits star codes) like enable or disable access voice mail server, enable or disable voice recording, etc which often include "*" and "#".

Thus, ST2030 now implements specific keys "*" and "#" in the existing dial plan in order to enable/disable these specific functions.

Note that wildcard "x" does not include "*" or "#".

For example:

```
(0T|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
```

More examples:

"****": If dialed number exactly matches 4 "star" digits, then we send it to server.

"*xxx" : If dialed number exactly matches 4 digits and the first digit is "*", then we send it to server.

"*xxxT" : If dialed number exactly matches 4 digits and the first digit is "*", then we send it to server till timeout.

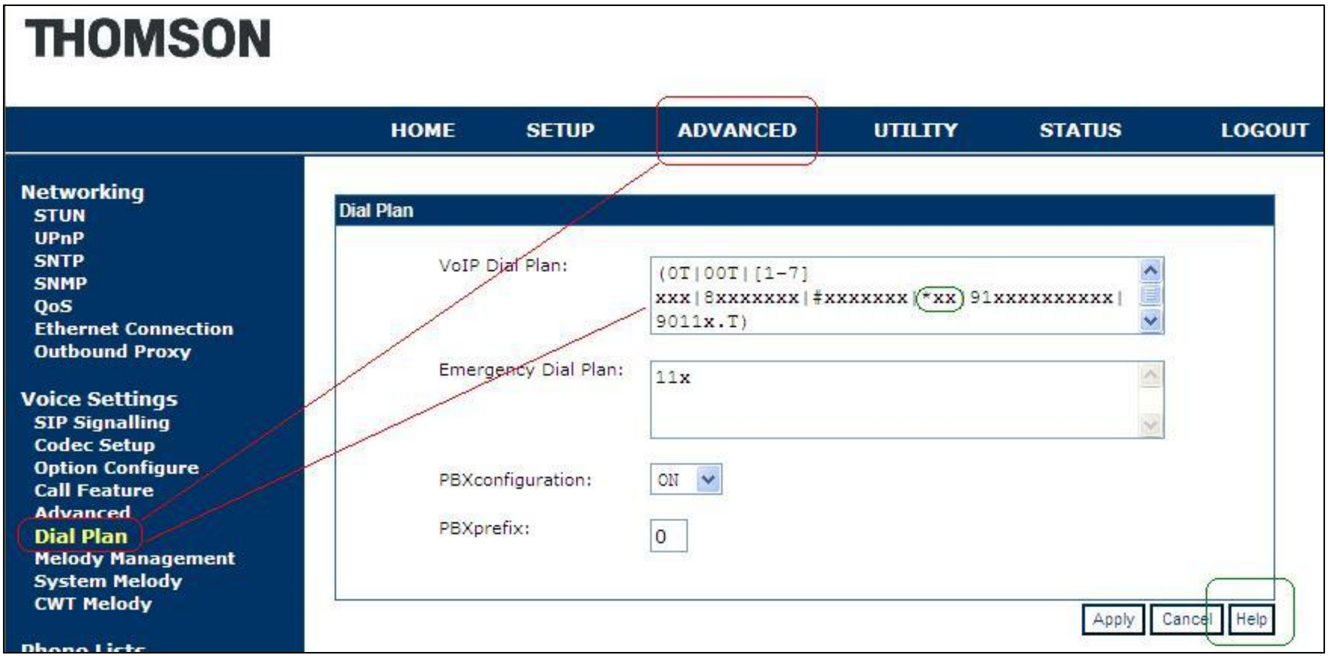
"####": If dialed number exactly matches 4 "hash" digits, then we send it to server.

"#xxx": If dialed number exactly matches 4 digits and the first digit is "#", then we send it to server.

"*xx#": If dialed number exactly matches 4 digits and the first digit is "*" and last digit is "#", then we send it to server.

Feature Activation

A) Through the WebGui:



VoIP Dial Plan: (0T|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)

Emergency Dial Plan: 11x

PBXconfiguration: ON

PBXprefix: 0

B) Through APS:

For this purpose, you have to use the following APS parameters:

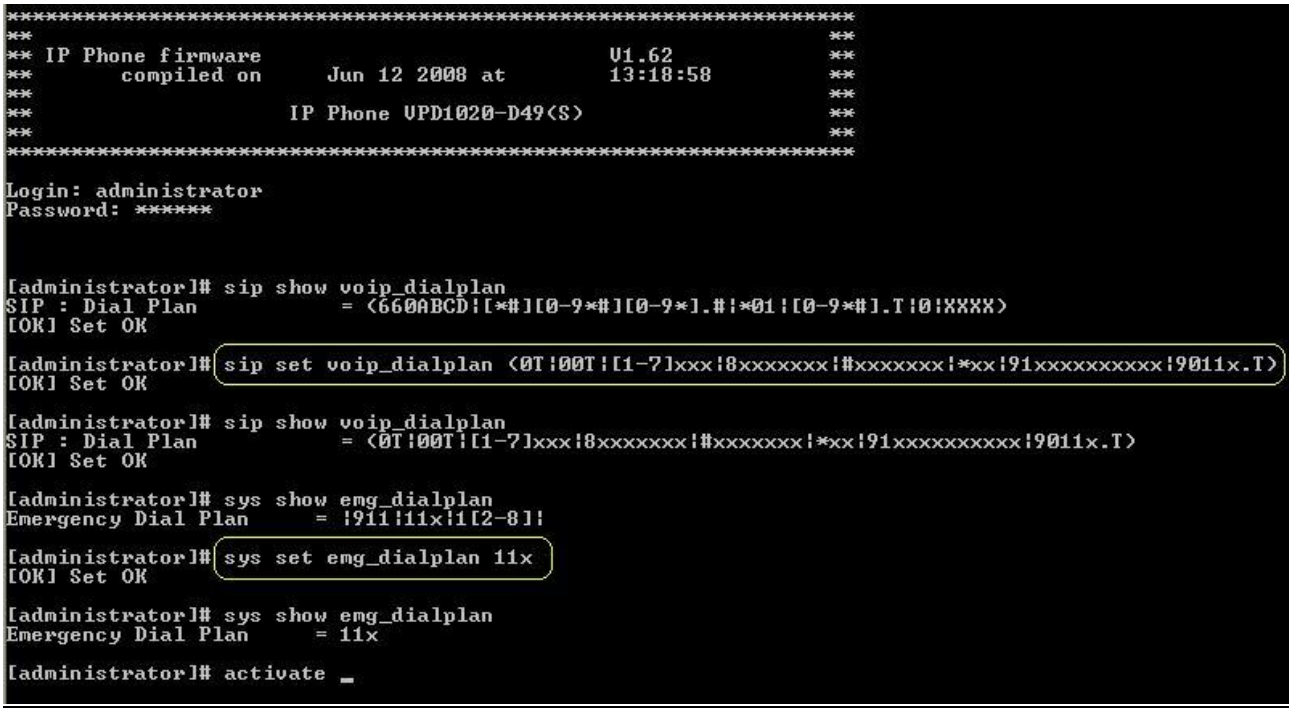
```
[sys]
…
EmergencyDialPlan=|911|11x|1[2-8]|
VOIPDialPlan=(0T|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
```

C) Through Telnet:

For this purpose, you have to use the following telnet parameters (already existing in previous versions):

```
sip set voip_dialplan (dialplan_string(max:120chars))

sys set emg_dialplan (dialplan_string(max:120chars))
```

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.62            **
**       compiled on      Jun 12 2008 at       13:18:58         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show voip_dialplan
SIP : Dial Plan           = (660ABCD|[*#][0-9*#][0-9*].#|*01|[0-9*#].T|0|XXXX)
[OK] Set OK

[administrator]# sip set voip_dialplan (0T|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
[OK] Set OK

[administrator]# sip show voip_dialplan
SIP : Dial Plan           = (0T|00T|[1-7]xxx|8xxxxxxx|#xxxxxxx|*xx|91xxxxxxxxxx|9011x.T)
[OK] Set OK

[administrator]# sys show emg_dialplan
Emergency Dial Plan     = |911|11x|1[2-8]|

[administrator]# sys set emg_dialplan 11x
[OK] Set OK

[administrator]# sys show emg_dialplan
Emergency Dial Plan     = 11x

[administrator]# activate _
```

New error message display for 403 response.

When the user dials a number that is not authorized on a network, it receives a 403 Forbidden message.

On current implementation, when the phone receives a 403 Forbidden message, it displays "Dialing Failed !".

This message could be misinterpreted by the end user who will try several times and finally call the administrator.

To enhance the MMI and reduce the need of support from customers, when the phone receives a "403 Forbidden" message, it shall display "Forbidden" which is a more appropriate message.

Additional strings in the language table:

Error Message	English	French	Spanish	German	Italian
Forbidden	Forbidden	Interdit	Prohibido	Verboten	Proibito

Error Message	Norway	Russian	Portuguese	Netherlands	
Forbidden	Forbudt	Запрещено	Proibido	Verboden	

Feature Activation

This feature is always active.

Services Supervision Feature.

This feature describes how ST20xx can request for a user profile and receive notifications related to a set of services implemented by the network like:

- DND activation status
- Call Forward activation status
- etc….

A few comments first. SIP User Agents require configuration data to function properly. A configuration data set, specific to an entity, is termed a profile. Process to get this profile could be automatic with no user intervention. In our scope, Service variables supervision is performed using the User Profile.

So, to support this new feature, ST20xx has implemented the "ua-profile" Event Package as described in "draft-ietf-sipping-config-framework-xx.txt". This framework provides a standard means of providing dynamic configuration and also addresses change notifications when profiles change. The content or format of the profile will be defined below.

There are 3 stages for the profile delivery process: enrollment, content retrieval and change notification. The behaviour of the phone in each state is as follows:

A) Enrollment: ST20xx requests and receives its profile data.

After a successful registration process, ST20xx will generate an initial SUBSCRIBE request for the event package "ua-profile" to the SIP server in order to be notified for the state of subscribed services for this account.
The initial SUBSCRIBE Request-URI, To and From headers are set to the account, and includes an Event and Accept headers as follows:

```
SUBSCRIBE sip:2205@10.0.0.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.55:5060;branch=z9hG4bK4696915359658109209-910168
From: <sip:2205@10.0.0.5:5060>;tag=c0a80101-de357
To: <sip:2205@10.0.0.5:5060>
Call-ID: e32b4-c0a80101-d-3@10.0.0.55
CSeq: 1 SUBSCRIBE
Max-Forwards: 70
Event: ua-profile;profile-type=user
Accept: application/service-user-profile+xml
Expires: 3600
Contact: <sip:2205@10.0.0.55:5060;user=phone>
Allow-Events: refer,dialog,message-summary,check-sync,talk,hold
User-Agent: THOMSON ST2030 hw0 fw1.62 00-0E-50-4E-88-E0
Content-Length: 0
```

Then, SIP Server receives successfully this Subscribe and identifies the requested profile data. It prepares a SIP NOTIFY message to ST20xx with the ua profile data inserted inside the body, and sends it immediately to complete the enrollment.

B) Content Retrieval

A successful profile enrollment leads to an initial SIP notification, and may result in subsequent change notifications. Each of these notifications can contain a profile data with the following restrictions:
The NOTIFY content is set according to the MIME type "application/service-user-profile+xml". All service variables are described in a XML document.
Inside the XML <service> node, each service variable is configured with a dedicated <variable> node. The <variable> node provides service variable name and current value.

For example:

```
NOTIFY sip:2205@10.0.0.55:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-b7d60ea12c0761993d006bb99033562e
from: <sip:2205@10.0.0.5:5060>;tag=e9c388b8d46
to: <sip:2205@10.0.0.5:5060>;tag=c0a80101-2834
Call-ID: 4fb8-c0a80101-d-2@10.0.0.55
CSeq: 2 NOTIFY
Contact: <sip:script@10.0.0.5:5060;transport=UDP>
Max-Forwards: 70
Subscription-state: active
Content-Length: 274
Event: ua-profile;profile-type=user
Content-type: application/service-user-profile+xml

<?xml version="1.0" encoding="UTF-8"?>
<service>
<variable name="cfu_on"   value="1" />
<variable name="dnd"      value="0" />
</service>
```

Following services are defined for ST20xx at this point:

Name	Possible values	Description
cfu_on	Integer: { 0, 1 }	Activation state of the Unconditional Call Forward. 0 means deactivated. 1 means activated.
sf_on	Integer: { 0, 1 }	Activation state of the Secretarial Filtering feature. Applicable only when a Manager line is supervised. 0 means deactivated. 1 means activated.
dnd	Integer: { 0, 1 }	Activation state of the Do Not Disturb feature. 0 means deactivated. 1 means activated.
hg_rdy	Integer: { 0, 1 }	Indicates whether the CPE that initiated the supervision is ready on the supervised Hunt Group. Applicable only when a Hunting Group virtual line is supervised. 0 means not ready. 1 means ready.

When ST20xx receives the SIP NOTIFY message, indicating successful profile enrollment, it makes it effective immediately and displays service status to the user.
The duration of the subscription is 3600 by default.

C) Change Notification

Each time service configuration changes, the SIP server sends a NOTIFY message to communicate the updated service variables to the ST20xx.
The NOTIFY body directly contains the new profile data.

The NOTIFY request content is set as described above and will only provide the updated service variables.
For all other service variables, the previous received values still apply.

ST20xx will apply the updated service variables and display them to the user according with MMI NOTIFICATION section below:

MMI Notification

One function key will be affected to each service. Depending on the state of the service, the corresponding function key's LED will be turned ON or OFF.

The Function Keys affected to a Service Supervision will remain inactive if the user presses them.

Things to know :

- Service Supervision will coexist with User-oriented BLF, but NOT with List-oriented BLF.

- To simplify the implementation, function keys assigned to Service supervisin will follow each other. So there will be a block of max 4 function keys.

- The position of the first function key and the order of the keys will be defined with two new different parameters.

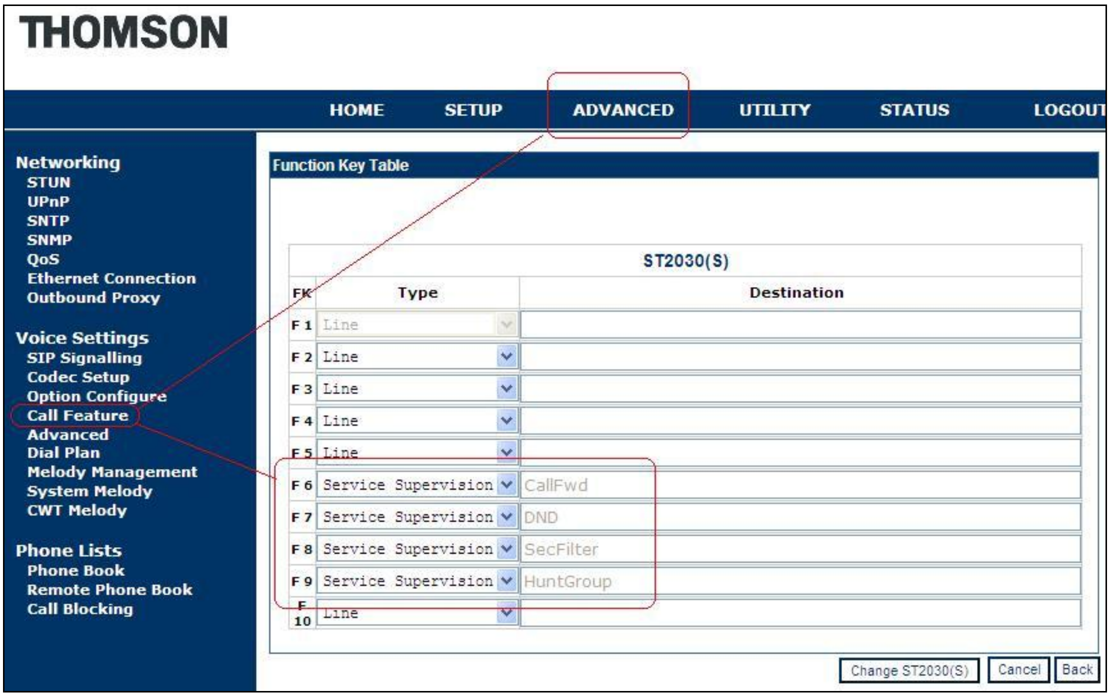
- If within the range allowed to Service Supervision there's already a function key assigned to an extension number or a supervised line, this key could be skipped or simply erased.

- Once a function key is assigned to Service Supervision, it cannot be modifiable through the MMI nor the web GUI.

Feature Activation

A) Through the WebGui:

This feature can not be configured through the Web (as mention above). However you can check the configuration through Advanced – Call Feature – Function Key Table :



Function Key Table

ST2030(S)

FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Line	
F 4	Line	
F 5	Line	
F 6	Service Supervision	CallFwd
F 7	Service Supervision	DND
F 8	Service Supervision	SecFilter
F 9	Service Supervision	HuntGroup
F 10	Line	

B) Through APS:

For this purpose, two new parameters have been included in the [sys] section of the Common/MAC config files, to be able to activate the feature.

```
[sys]
...
ServiceSupervisionFK=0
ServiceSupOrder=callfwd(0)dnd(0)secFilter(0)huntgroup(0)
...
```

The parameter “**ServiceSupervisionFK**” could be a value from Multiline+1 (default) to max number of function keys available (10, 38 or 66 if using extension modules).

The parameter “**ServiceSupOrder**=callfwd(a)dnd(b)secFilter(c)huntgroup(d)” could have the following values, where a, b, c, d values can be 0, 1, 2, 3 or 4:

- 0 means the function is not supervised.

- the other values are the order of the function key affected to this service.

CallFwd is linked to "cfu_on" valiable
DND is linked to "dnd" valiable
SecFilter is linked to "sf_on" valiable
HuntGroup is linked to "hg_rdy" valiable

For example:

```
[sys]
Current_Max_Multiline=5
ServiceSupervisionFK=7
ServiceSupOrder=callfwd(2)dnd(1)secFilter(0)huntgroup(0)
```

Here,

- Call Forward service will be supervised on function key 8.
- Do Not Disturb service will be supervised on function key 7.
- Secretarial Filtering will not be supervised at all.
- Hunting Group will not be supervised at all.

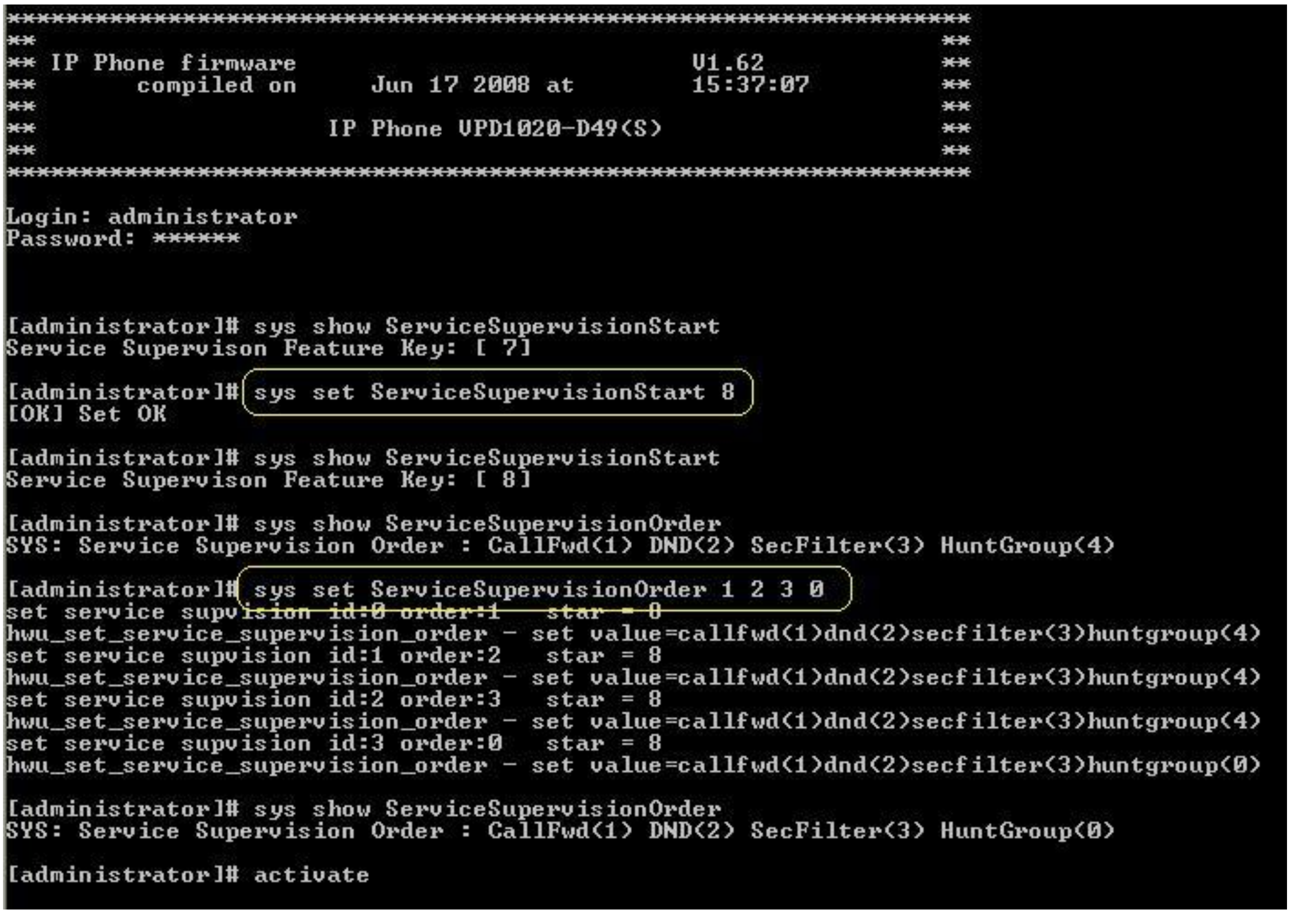
So, function keys 6, 9 and 10 could be assigned to anything else.

C) Through Telnet:

For this purpose, two new parameters have been created, described as follow:

```
sys set ServiceSupervisionStart [id]

sys set ServiceSupervisionOrder <CallFwd> <DND> <SecFilter> <HuntGroup>
                                (Value: 0~4,)
```

To configure, open a command line console, and telnet the phone:



```
*******************************************************************
**                                                              **
** IP Phone firmware                          V1.62             **
**       compiled on      Jun 17 2008 at      15:37:07          **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
*******************************************************************

Login: administrator
Password: ******


[administrator]# sys show ServiceSupervisionStart
Service Supervison Feature Key: [ 7]

[administrator]# sys set ServiceSupervisionStart 8
[OK] Set OK

[administrator]# sys show ServiceSupervisionStart
Service Supervison Feature Key: [ 8]

[administrator]# sys show ServiceSupervisionOrder
SYS: Service Supervision Order : CallFwd(1) DND(2) SecFilter(3) HuntGroup(4)

[administrator]# sys set ServiceSupervisionOrder 1 2 3 0
set service supvision id:0 order:1   star = 8
hwu_set_service_supervision_order - set value=callfwd(1)dnd(2)secfilter(3)huntgroup(4)
set service supvision id:1 order:2   star = 8
hwu_set_service_supervision_order - set value=callfwd(1)dnd(2)secfilter(3)huntgroup(4)
set service supvision id:2 order:3   star = 8
hwu_set_service_supervision_order - set value=callfwd(1)dnd(2)secfilter(3)huntgroup(4)
set service supvision id:3 order:0   star = 8
hwu_set_service_supervision_order - set value=callfwd(1)dnd(2)secfilter(3)huntgroup(0)

[administrator]# sys show ServiceSupervisionOrder
SYS: Service Supervision Order : CallFwd(1) DND(2) SecFilter(3) HuntGroup(0)

[administrator]# activate
```

Star Codes.

Star Codes (SC) or Feature Access Codes (FAC) are special char or digity patterns that are dialed from a phone to invoke particular server features.

Typically, FAC are invoked using a short sequence of digits that are dialed using the keypad on an analog phone, while IP phones users select soft-keys to invoke the same features. Generally speaking service will be invoked by sending an INVITE to a specific uri which contains the FAC.

A description of how to change the ST20xx local soft-keys configuration to activate/deactivate these features instead of the local functions, and instructions on how to use each of them is provided in the following chapters.

A special case takes place when a FAC has to be sent as DTMF during an existing SIP call. For this purpose, a specific type of function key is defined and described in chapter 6.9

Also, for other potential services invoked by FAC's and not supported by softkeys, a "Service" type of function key is defined and described in chapter 6.10.

Things to know :
1.- For each local function, a flag is used to state if the function is managed locally (as current behavior) or using the star codes. For those items which have a "local/sc" flag, whenever the flag is set to "local", the behavior will be the same as it is today. So chapters below refer only to the flag="sc" value when applicable.

2.- When function flags are set to "sc" for features launching some kind of icon or message on the display (CF, DND...), then this indication will not be present, in order to avoid un-synchronization with server state. Better not to provide any state than a wrong one!

3.- A "Star Code" can include keypad digits (0-9, * and #), as well as any chars allowed in uri's. For DTMF function key type only keypad digits are allowed.

4.- For functions related to call forward, if sc is enabled, we need to use a different call to send the sc. It is assumed the server is in charge of canceling the initial call being forwarded.

5.- When a star code is sent and there is an active call, active call is put on hold

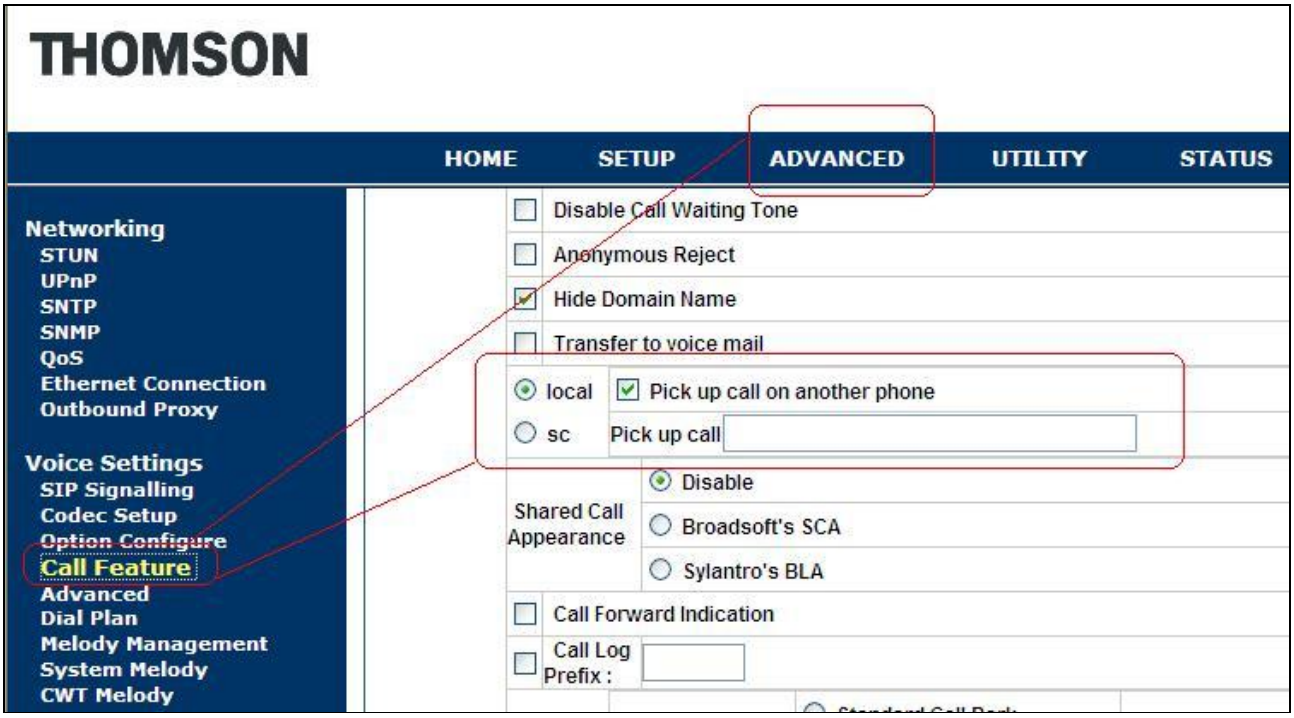
Glossary: SC= star code; FK= function key

Call Pick-Up Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



☐ Disable Call Waiting Tone

☐ Anonymous Reject

☑ Hide Domain Name

☐ Transfer to voice mail

● local ☑ Pick up call on another phone

○ sc Pick up call

Shared Call Appearance: ● Disable ○ Broadsoft's SCA ○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

B) Through APS:

For this purpose, two new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
CallPkupFlg=sc
CallPkupSC=*53
```

C) Through Telnet:

For this purpose, two new parameters have been created:

```
sip set CallPkupFlg [local|sc]
sip set CallPkupSC (call pickup star code)
```

Behavior

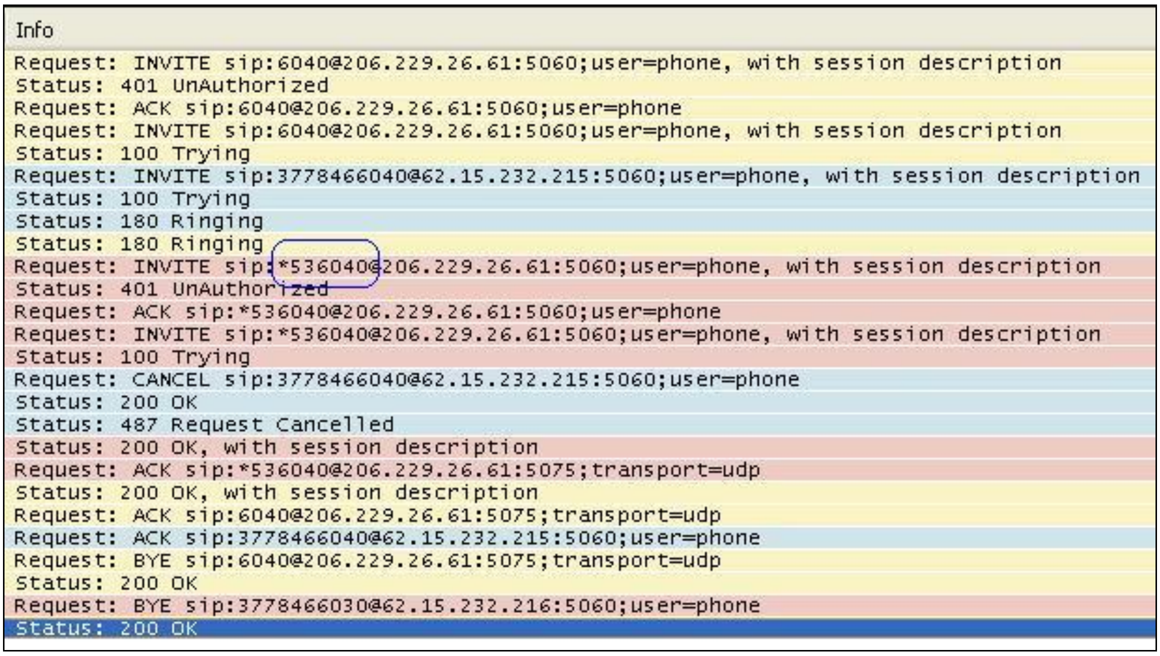
Call pick-up with Soft Key

1. Press Pick-Up soft key.
2. The phone will display "Enter a number".
3. Then, user will enter the extension number to be picked up.
4. The phone will send the INVITE with TO header set to "SC + phoneNr"

Call pick-up with Function Key (User oriented BLF)

1. Press FK when blinking (BLF = early)
2. The phone will send the INVITE with TO header = SC + phoneNr
3. Is the same behavior as above, except that phone number = supervised line (will no prompt to user).

Call Flow:



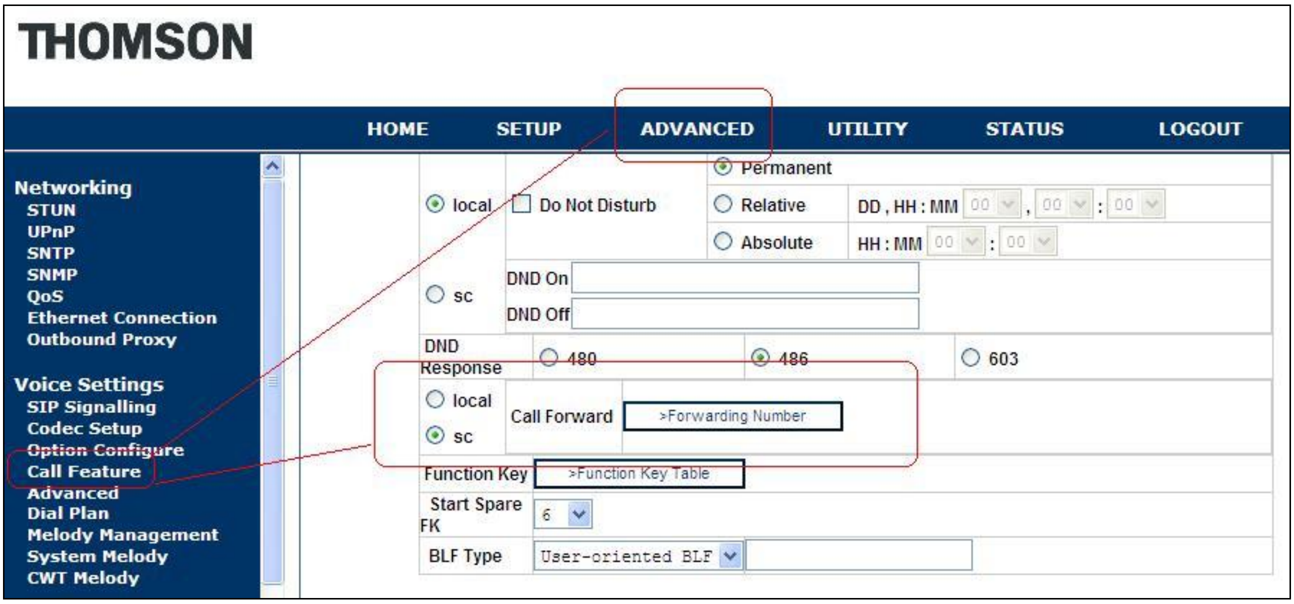
```
Info
Request: INVITE sip:6040@206.229.26.61:5060;user=phone, with session description
Status: 401 UnAuthorized
Request: ACK sip:6040@206.229.26.61:5060;user=phone
Request: INVITE sip:6040@206.229.26.61:5060;user=phone, with session description
Status: 100 Trying
Request: INVITE sip:3778466040@62.15.232.215:5060;user=phone, with session description
Status: 100 Trying
Status: 180 Ringing
Status: 180 Ringing
Request: INVITE sip:*536040@206.229.26.61:5060;user=phone, with session description
Status: 401 UnAuthorized
Request: ACK sip:*536040@206.229.26.61:5060;user=phone
Request: INVITE sip:*536040@206.229.26.61:5060;user=phone, with session description
Status: 100 Trying
Request: CANCEL sip:3778466040@62.15.232.215:5060;user=phone
Status: 200 OK
Status: 487 Request Cancelled
Status: 200 OK, with session description
Request: ACK sip:*536040@206.229.26.61:5075;transport=udp
Status: 200 OK, with session description
Request: ACK sip:6040@206.229.26.61:5075;transport=udp
Request: ACK sip:3778466040@62.15.232.215:5060;user=phone
Request: BYE sip:6040@206.229.26.61:5075;transport=udp
Status: 200 OK
Request: BYE sip:3778466030@62.15.232.216:5060;user=phone
Status: 200 OK
```

Call Forward Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



● local ☐ Do Not Disturb ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00

○ sc DND On DND Off

DND Response ○ 480 ● 486 ○ 603

○ local ● sc Call Forward >Forwarding Number

Function Key >Function Key Table

Start Spare FK 6

BLF Type User-oriented BLF


Call Forwarding Number

	While	Call Forward
Line	OFF	*73
	Always	*72
	Busy	*90
	No Answer	*92
	On Ringing	*94

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
CallFwdFlg=sc
CallFwdOffSC=*73
CallFwdAlwaysSC=*72
CallFwdBusySC=*90
CallFwdNoAnswerSC=*92
TransferOnRingSC=*94
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set CallFwdFlg [local|sc]
sip set CallFwdOffSC ( Call FwdOff star code)
sip set CallFwdAlwaysSC (Call FwdAlways star code)
sip set CallFwdBusySC (Call FwdBusy star code)
sip set CallFwdNoAnswerSC (Call FwdNoAnswe star code)
sip set TransferOnRingSC (Transfer OnRing star code)
```

Behavior

Call Forward OFF through Keypad

1. Press Menu - Option - CallForward - Select
2. Press Change
3. Enter the CF phone number
4. Select CF type = OFF
5. Press Save
6. The phone will send the INVITE with TO header = "SC"

Call Forward Always/Busy/No Answer through Keypad

1. Press Menu - Option - CallForward - Select
2. Press Change
3. Enter the CF phone number
4. Select CF type = Always/Busy/No Answer
5. Press Save
6. The phone will send the INVITE with TO header = "SC + phoneNr"

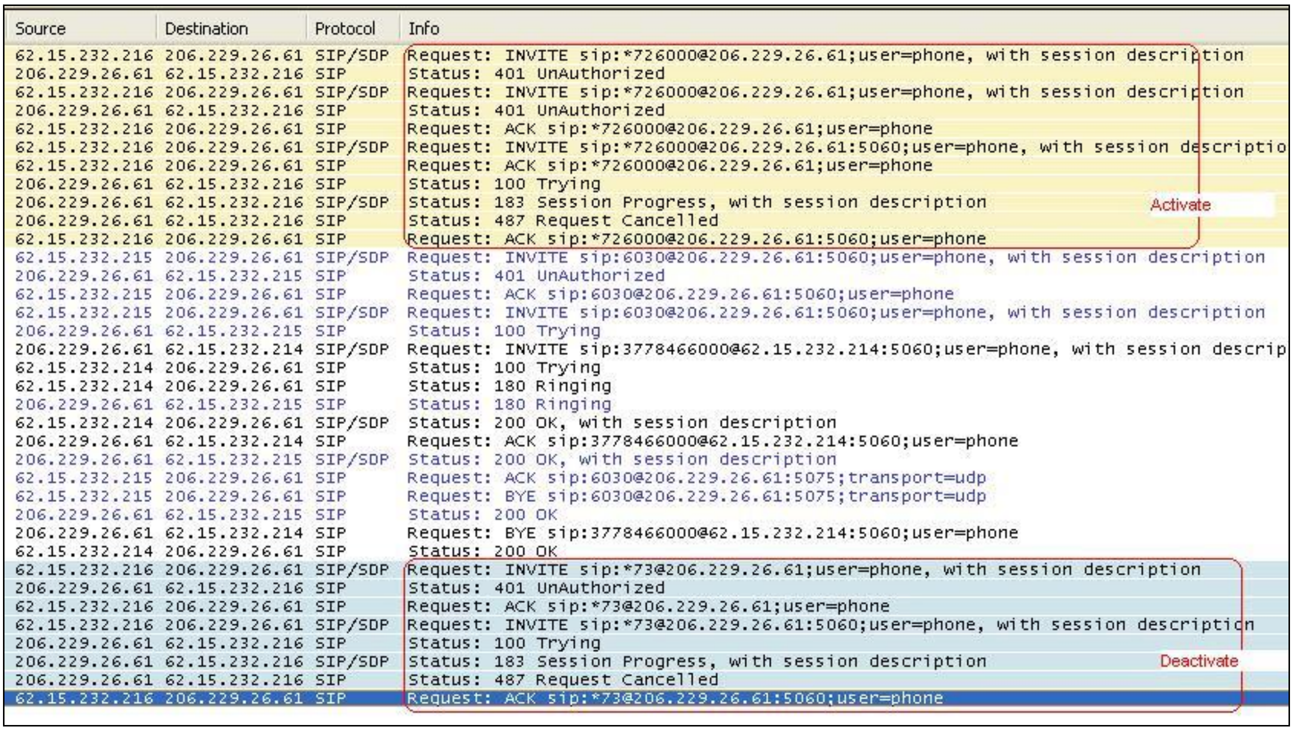
As indicated in General remarks, phone's mmi must not show any sign of this feature to be activated (icon, message on screen), to avoid synchrony issues with server

Call Forward On ringing through Soft Key

1. Press Transfer Soft Key when the phone is ringing
2. The phone displays "Enter a number"
3. User enters the destination phone number to transfer to, press OK.
4. The phone will send an INVITE with TO header = "SC + phoneNr"

The server is expected to cancel the initial call

Call Flow:



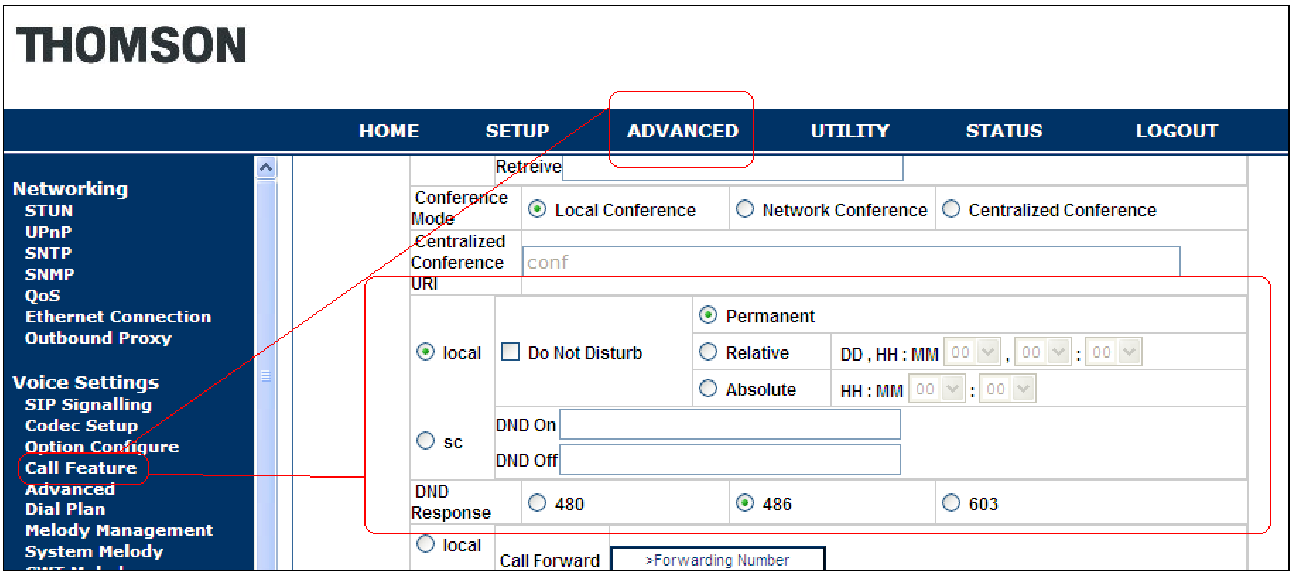
```
Source        Destination   Protocol Info
62.15.232.216 206.229.26.61 SIP/SDP  Request: INVITE sip:*726000@206.229.26.61;user=phone, with session description
206.229.26.61 62.15.232.216 SIP      Status: 401 UnAuthorized
62.15.232.216 206.229.26.61 SIP/SDP  Request: INVITE sip:*726000@206.229.26.61;user=phone, with session description
206.229.26.61 62.15.232.216 SIP      Status: 401 UnAuthorized
62.15.232.216 206.229.26.61 SIP      Request: ACK sip:*726000@206.229.26.61;user=phone
62.15.232.216 206.229.26.61 SIP/SDP  Request: INVITE sip:*726000@206.229.26.61:5060;user=phone, with session descriptio
62.15.232.216 206.229.26.61 SIP      Request: ACK sip:*726000@206.229.26.61;user=phone
206.229.26.61 62.15.232.216 SIP      Status: 100 Trying
206.229.26.61 62.15.232.216 SIP/SDP  Status: 183 Session Progress, with session description
206.229.26.61 62.15.232.216 SIP      Status: 487 Request Cancelled
62.15.232.216 206.229.26.61 SIP      Request: ACK sip:*726000@206.229.26.61:5060;user=phone
62.15.232.215 206.229.26.61 SIP/SDP  Request: INVITE sip:6030@206.229.26.61:5060;user=phone, with session description
206.229.26.61 62.15.232.215 SIP      Status: 401 UnAuthorized
62.15.232.215 206.229.26.61 SIP      Request: ACK sip:6030@206.229.26.61:5060;user=phone
62.15.232.215 206.229.26.61 SIP/SDP  Request: INVITE sip:6030@206.229.26.61:5060;user=phone, with session description
206.229.26.61 62.15.232.215 SIP      Status: 100 Trying
206.229.26.61 62.15.232.214 SIP/SDP  Request: INVITE sip:3778466000@62.15.232.214:5060;user=phone, with session descrip
62.15.232.214 206.229.26.61 SIP      Status: 100 Trying
62.15.232.214 206.229.26.61 SIP      Status: 180 Ringing
206.229.26.61 62.15.232.215 SIP      Status: 180 Ringing
62.15.232.214 206.229.26.61 SIP/SDP  Status: 200 OK, with session description
206.229.26.61 62.15.232.214 SIP      Request: ACK sip:3778466000@62.15.232.214:5060;user=phone
206.229.26.61 62.15.232.215 SIP/SDP  Status: 200 OK, with session description
62.15.232.215 206.229.26.61 SIP      Request: ACK sip:6030@206.229.26.61:5075;transport=udp
62.15.232.215 206.229.26.61 SIP      Request: BYE sip:6030@206.229.26.61:5075;transport=udp
206.229.26.61 62.15.232.215 SIP      Status: 200 OK
206.229.26.61 62.15.232.214 SIP      Request: BYE sip:3778466000@62.15.232.214:5060;user=phone
62.15.232.214 206.229.26.61 SIP      Status: 200 OK
62.15.232.216 206.229.26.61 SIP/SDP  Request: INVITE sip:*73@206.229.26.61;user=phone, with session description
206.229.26.61 62.15.232.216 SIP      Status: 401 UnAuthorized
62.15.232.216 206.229.26.61 SIP      Request: ACK sip:*73@206.229.26.61;user=phone
62.15.232.216 206.229.26.61 SIP/SDP  Request: INVITE sip:*73@206.229.26.61:5060;user=phone, with session description
206.229.26.61 62.15.232.216 SIP      Status: 100 Trying
206.229.26.61 62.15.232.216 SIP/SDP  Status: 183 Session Progress, with session description
206.229.26.61 62.15.232.216 SIP      Status: 487 Request Cancelled
62.15.232.216 206.229.26.61 SIP      Request: ACK sip:*73@206.229.26.61:5060;user=phone
```

Do Not Disturb Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



Retreive

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

● local ☐ Do Not Disturb ● Permanent

○ Relative DD , HH : MM 00 , 00 : 00

○ Absolute HH : MM 00 : 00

○ sc DND On

DND Off

DND Response ○ 480 ● 486 ○ 603

○ local Call Forward >Forwarding Number

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
DNDFlg=sc
DNDOnSC=*67
DNDOffSC=*68
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set DNDFlg [local|sc]
sip set DNDOnSC (DND on star code)
sip set DNDOffSC (DND off star code)
```

Behavior

Do Not Disturb ON through Soft Key

1. Using the navigation keys, Press DND Soft key.
2. Press Edit - select ON - press OK
3. The phone will send the INVITE with TO header = SC

Same remark on mmi as above

Do Not Disturb OFF through Soft Key

1. Using the navigation keys, Press DND Soft key.
2. Press Edit - select OFF - press OK
3. The phone will send the INVITE with TO header = SC

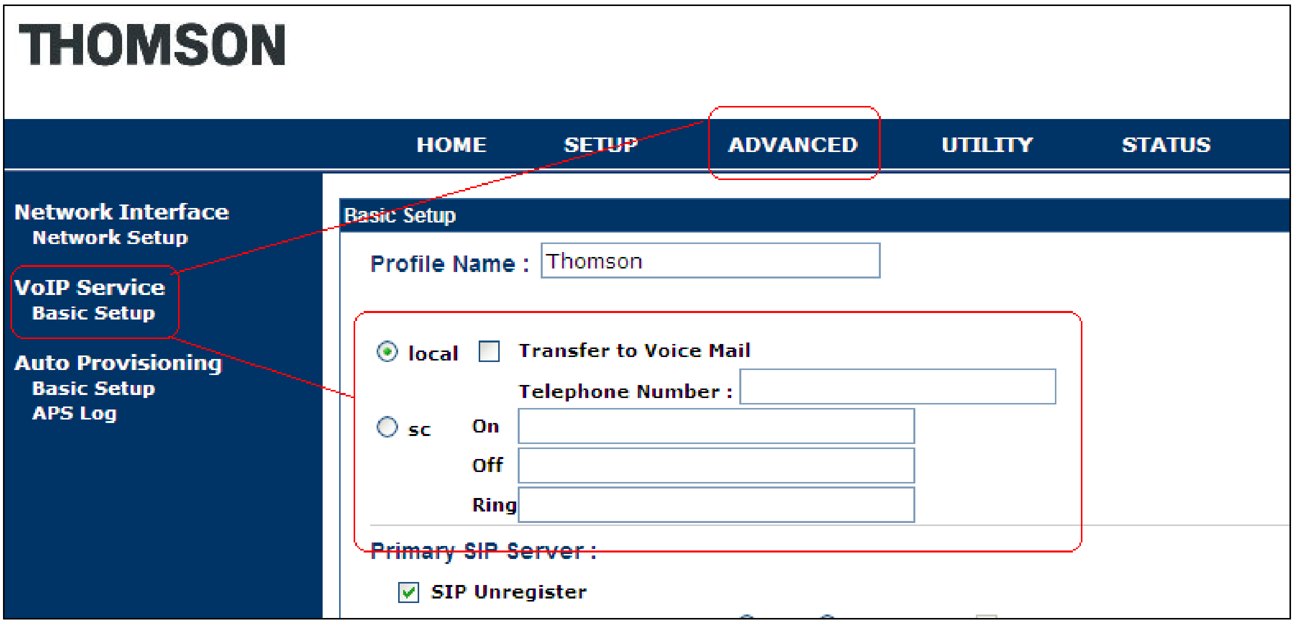
Same remark on mmi as above

Transfer to Voicemail Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Setup / Basic Setup / ProfileN section.



Profile Name : Thomson

● local ☐ Transfer to Voice Mail

Telephone Number :

○ sc On

Off

Ring

Primary SIP Server :

☑ SIP Unregister

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
TrVoiceMailFlg1=local
TrVoiceMailFlg2=local
TrVoiceMailFlg3=local
TrVoiceMailFlg4=local
TrVoiceMailOnSC1=
TrVoiceMailOnSC2=
TrVoiceMailOnSC3=
TrVoiceMailOnSC4=
TrVoiceMailOffSC1=
TrVoiceMailOffSC2=
TrVoiceMailOffSC3=
TrVoiceMailOffSC4=
TrVoiceMailRingSC1=
TrVoiceMailRingSC2=
TrVoiceMailRingSC3=
TrVoiceMailRingSC4=
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set TrVoiceMailFlg (profile number) (local|sc)
sip set TrVoiceMailOnSC  (profile number) (star code)
sip set TrVoiceMailOffSC (profile number) (star code)
sip set TrVoiceMailRingSC (profile number) (star code)
```

Behavior

Transfer to voicemail ON

1. Using the navigation keys, press TrMail, select ON
2. The phone sends the INVITE with TO header = SC

Transfer to voicemail OFF

1. Using the navigation keys, press TrMail, select OFF
2. The phone sends the INVITE with TO header = SC

Transfer to voicemail on Ringing

1. Press TrMail when the phone is ringing
2. The phone sends an INVITE with TO header = SC

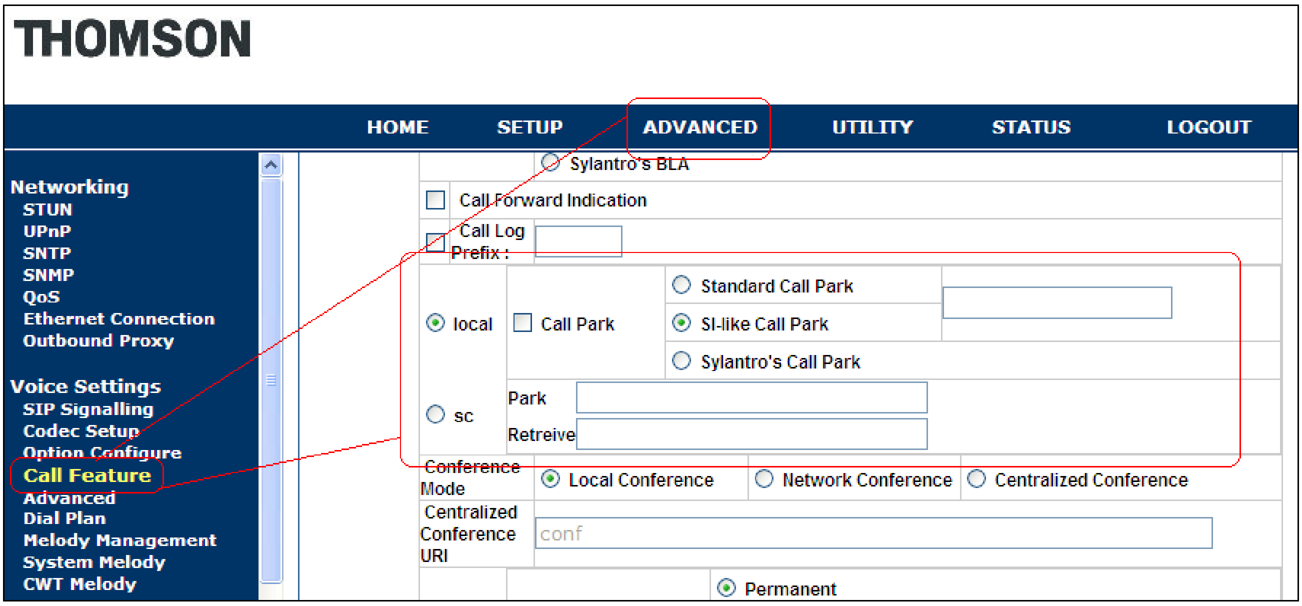
The server is expected to cancel the initial call

Call Park Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

● local ☐ Call Park ○ Standard Call Park ● SI-like Call Park ○ Sylantro's Call Park

○ sc Park Retreive

Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference

Centralized Conference URI: conf

● Permanent

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
CallParkFlg=local
CallParkSC=
CallRetreiveSC=
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set CallParkFlg [local|sc]
sip set CallParkSC ( Call Park star code)
sip set CallRetreiveSC (Call Retreive star code)
```

Behavior

Call Park soft key

1. During a conversation, press Call Park Soft key
2. The phone will dial the SC as DTMF.

Call Retrieve soft key

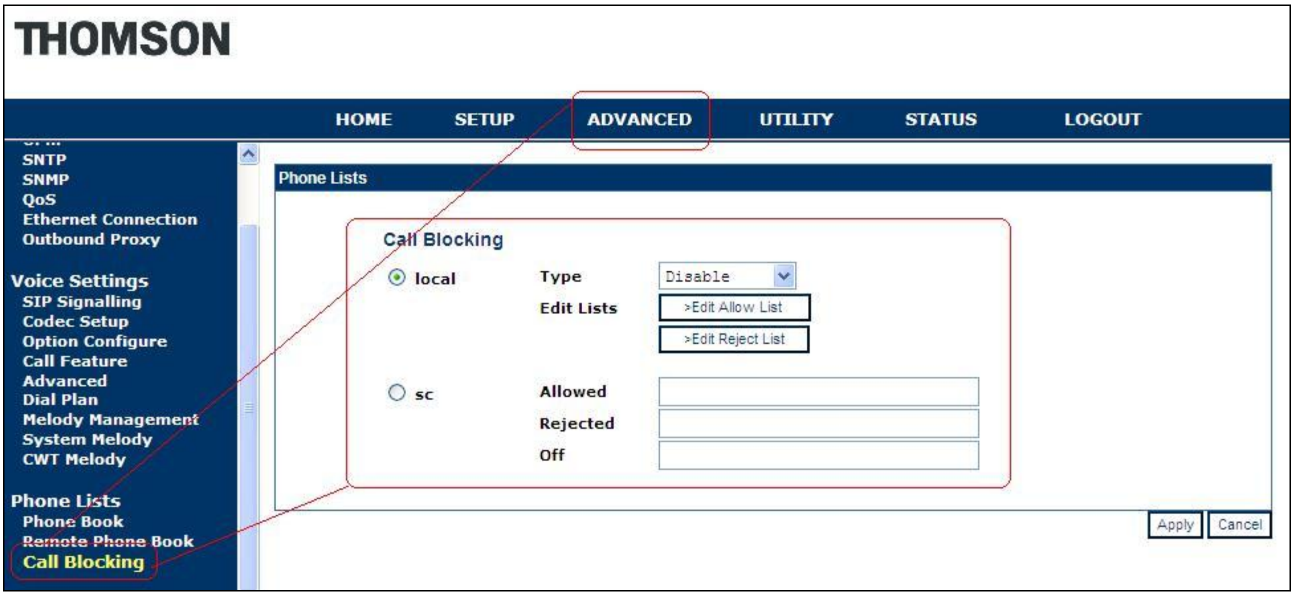
1. Press RtPark soft key
2. The phone will display "Enter a number"
3. The user will enter the orbit number
4. The phone will send the INVITE with TO header= " SC + orbit Nr"

Call Blocking Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Blocking section.



● local
Type: Disable

○ sc

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
CallBlockFlg=local
CallBlockAllowSC=
CallBlockRejectSC=
CallBlockOFFSC=
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set CallBlockFlg [local|sc]
sip set CallBlockAllowSC (Call Block Allow star code)
sip set CallBlockRejectSC (Call Block Reject star code)
sip set CallBlockOFFSC (Call Block OFF star code)
```

Behavior

Call Blocking Allowed

1. Press Menu - Option - CallBlocking - Select
2. Press Edit - Select Allowed - press Save
3. The phone will send the INVITE with TO header = SC

Call Blocking Rejected

1. Press Menu - Option - CallBlocking - Select
2. Press Edit - Select Rejected - press Save
3. The phone will send the INVITE with TO header = SC

Call Blocking OFF

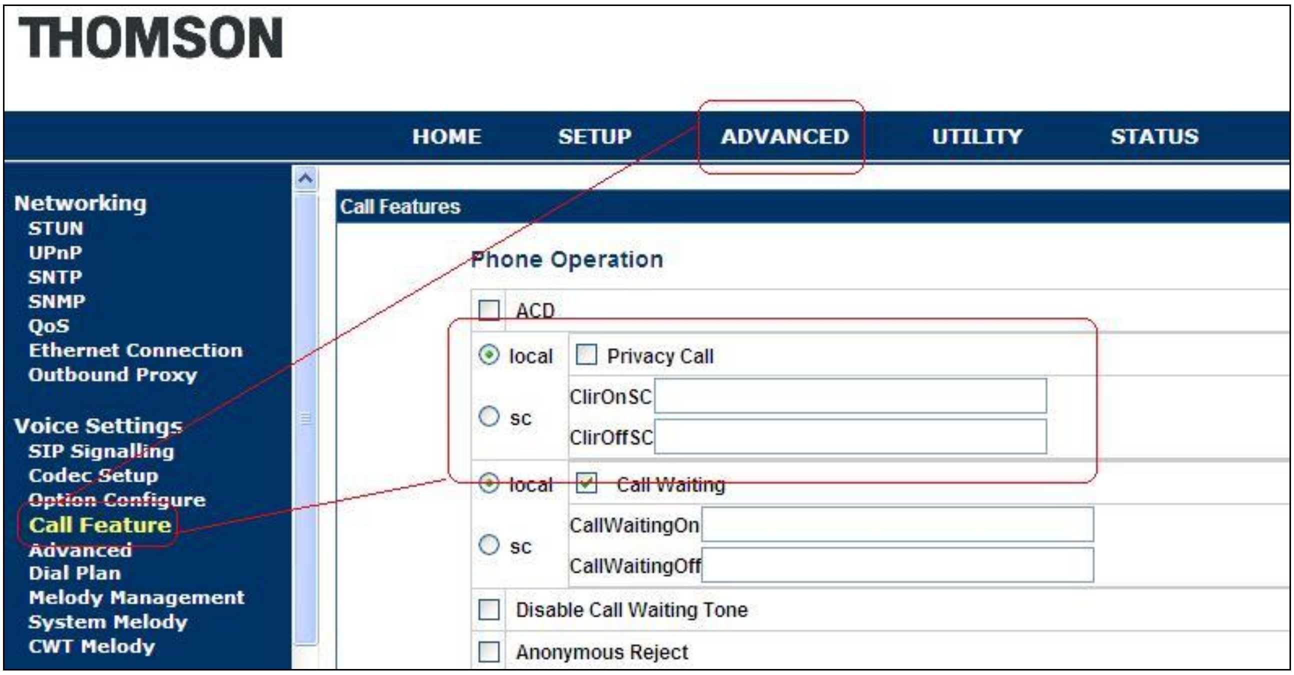
1. Press Menu - Option - CallBlocking - Select
2. Press Edit - Select Disable - press Save
3. The phone will send the INVITE with TO header = SC

Privacy Call Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
[sip]
ClirFlg=local
ClirOnSC=
ClirOffSC=
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set ClirFlg [local|sc]
sip set ClirOnSC ( Clir On star code)
sip set ClirOffSC (Clir Off star code)
```

Behavior

CLIR ON (Privacy code)

1. Press Menu - Option - NumberDisplay - Change
2. Select ON
3. The phone sends the INVITE with TO header= SC

Same remark on mmi as above

CLIR OFF (Privacy code)

1. Press Menu - Option - NumberDisplay - Change
2. Select OFF
3. The phone sends the INVITE with TO header= SC

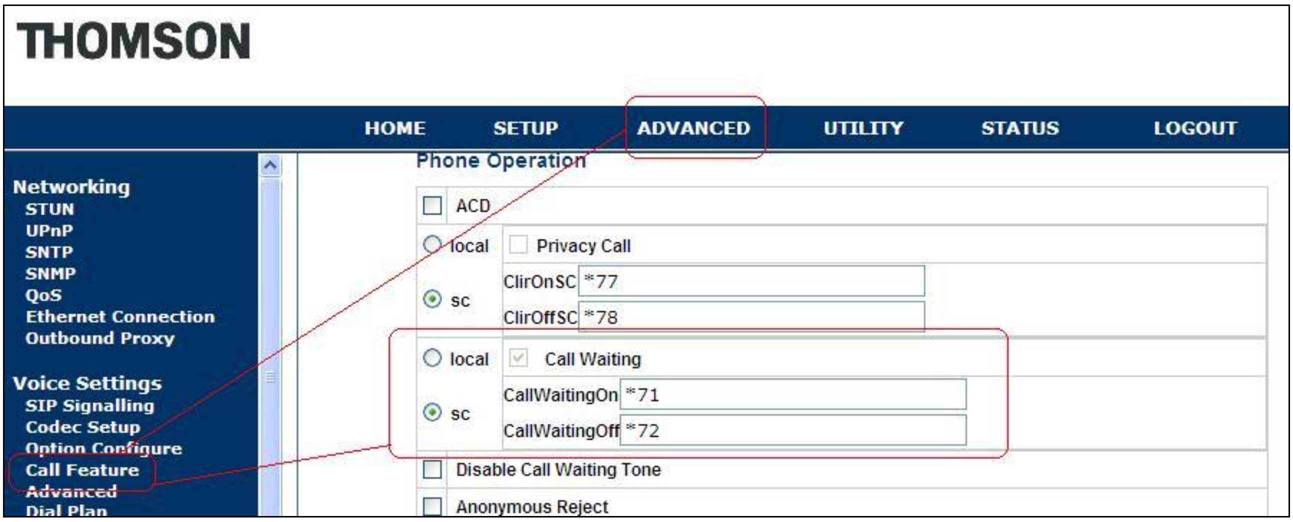
Same remark on mmi as above

Call Waiting Service

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



Phone Operation

☐ ACD

○ local ☐ Privacy Call

● sc ClirOnSC *77

ClirOffSC *78

○ local ☑ Call Waiting

● sc CallWaitingOn *71

CallWaitingOff *72

☐ Disable Call Waiting Tone

☐ Anonymous Reject

B) Through APS:

For this purpose, new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the feature.

```
 [sip]
CallWaitingFlg=local
CallWaitingOnSC=
CallWaitingOffSC=
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

```
sip set CallWaitingFlg [local|sc]
sip set CallWaitingOnSC ( Call Waiting On star code)
sip set CallWaitingOffSC (Call Waiting  Off star code)
```

Behavior

Call Waiting ON

1. Press Menu - Option - CallWaiting - Change
2. Select ON
3. The phone will send the INVITE with TO header= SC

Call Waiting OFF

1. Press Menu - Option - CallWaiting - Change
2. Select OFF
3. The phone will send the INVITE with TO header= SC

Special Services activation through DTMF

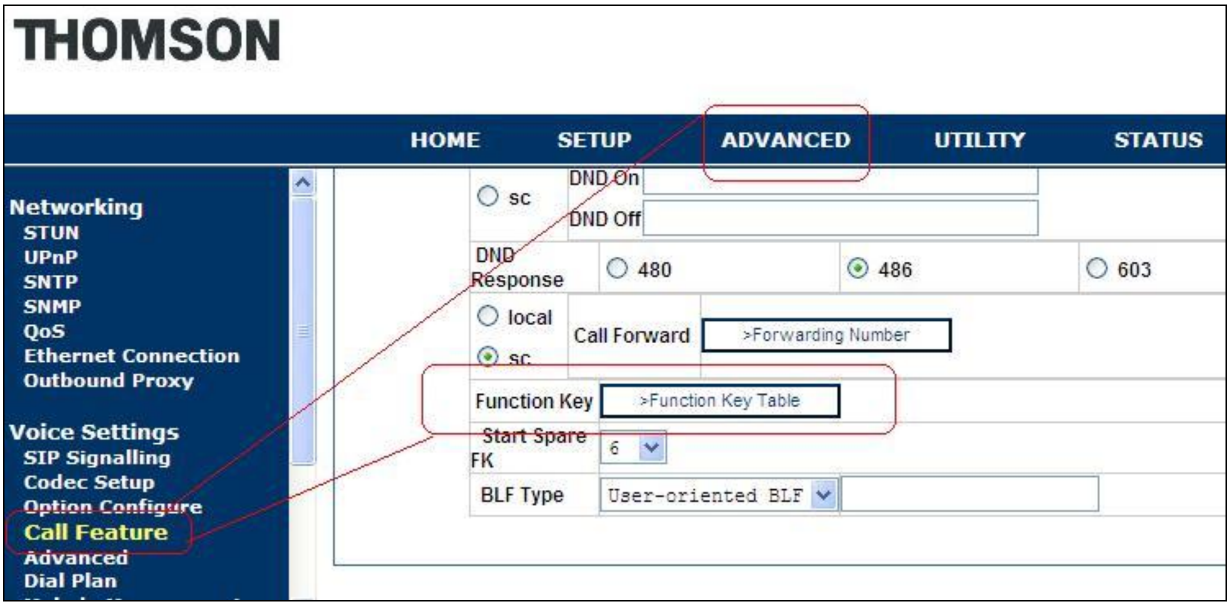
Notes:

- Allowed starting from multiline+1
- Add a new type "DTMF" (on top of existing Line and Supervised Line)

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from :



○ sc DND On DND Off

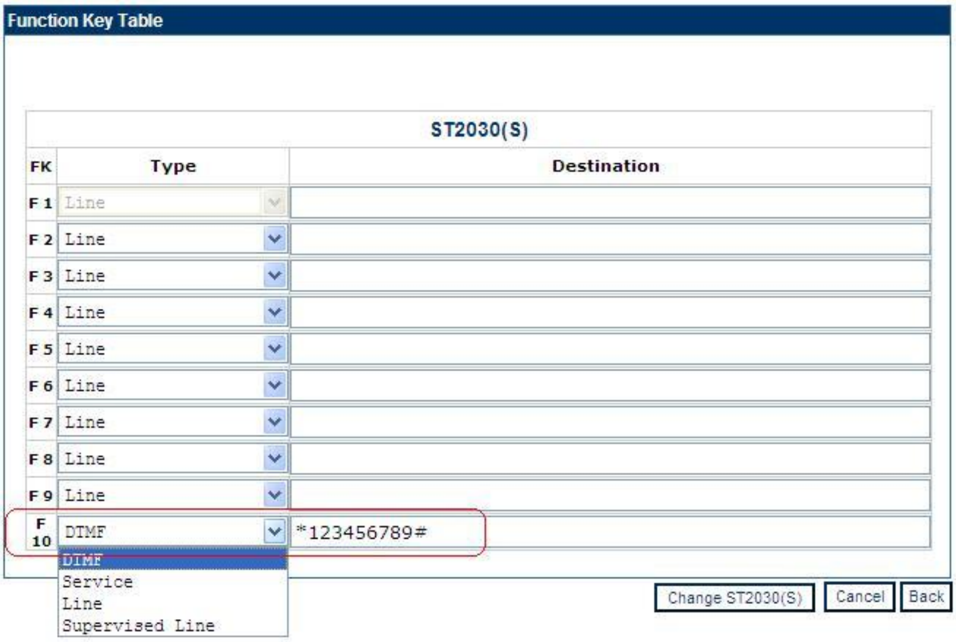
DND Response ○ 480 ● 486 ○ 603

Call Forward ○ local ● sc

Function Key

Start Spare FK: 6

BLF Type: User-oriented BLF



ST2030(S)

FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Line	
F 4	Line	
F 5	Line	
F 6	Line	
F 7	Line	
F 8	Line	
F 9	Line	
F 10	DTMF	*123456789#

B) Through APS:

For this purpose, new parameters have been included in the [sys] section of the Common/MAC config files, to be able to configure the feature.

```
[sys]
Current_Max_Multiline=5

FeatureKeyExt06=L/<sip:2206>
FeatureKeyExt07=S/<sip:2207>
FeatureKeyExt08=S/<sip:2208>
FeatureKeyExt09=S/<sip:2209>
FeatureKeyExt10=D/<sip:*123456789#>
```

C) Through Telnet:

For this purpose, new parameters have been created, described as follow:

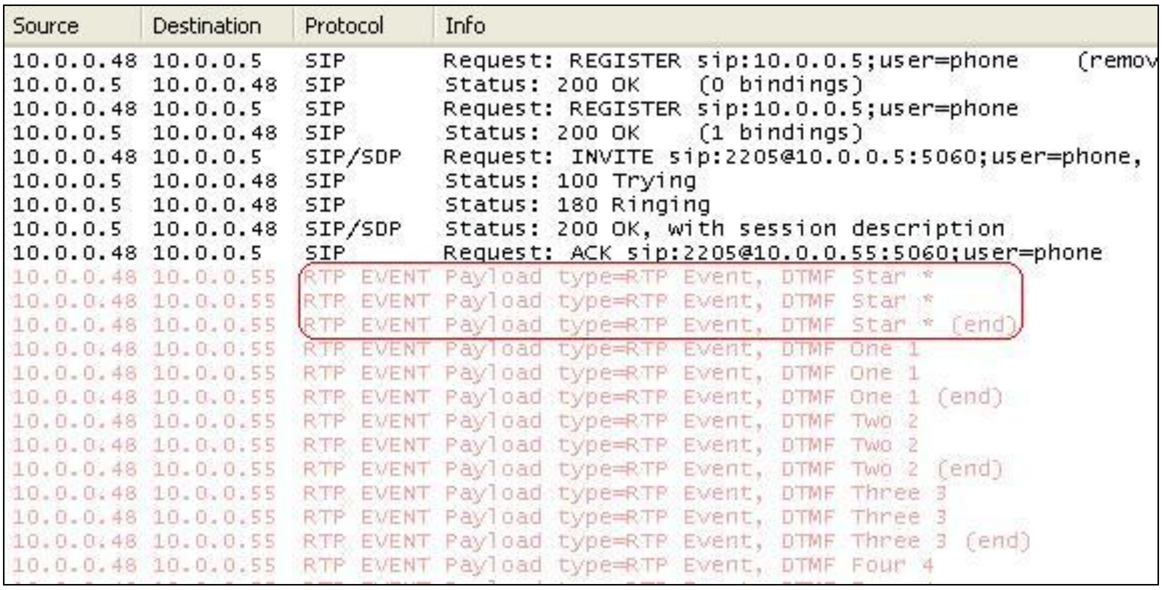
```
sys set fkuri id(1~66) uri_string
sys set fktype id(1~66) 0/1/2/3/4/5/6/7 (L/S/E/R/G/B/D/V)
```

Behavior

1. During a conversation, press the Function key
2. The phone will dial the FK number as DTMF, with configured or negotiated DTMF method.

If you press the FK in idle, there will be no action

Call Flow:



```
Source     Destination Protocol  Info
10.0.0.48  10.0.0.5    SIP       Request: REGISTER sip:10.0.0.5;user=phone    (remov
10.0.0.5   10.0.0.48   SIP       Status: 200 OK    (0 bindings)
10.0.0.48  10.0.0.5    SIP       Request: REGISTER sip:10.0.0.5;user=phone
10.0.0.5   10.0.0.48   SIP       Status: 200 OK    (1 bindings)
10.0.0.48  10.0.0.5    SIP/SDP   Request: INVITE sip:2205@10.0.0.5:5060;user=phone,
10.0.0.5   10.0.0.48   SIP       Status: 100 Trying
10.0.0.5   10.0.0.48   SIP       Status: 180 Ringing
10.0.0.5   10.0.0.48   SIP/SDP   Status: 200 OK, with session description
10.0.0.48  10.0.0.5    SIP       Request: ACK sip:2205@10.0.0.55:5060;user=phone
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Star *
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Star *
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Star * (end)
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF One 1
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF One 1
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF One 1 (end)
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Two 2
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Two 2
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Two 2 (end)
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Three 3
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Three 3
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Three 3 (end)
10.0.0.48  10.0.0.55   RTP EVENT Payload type=RTP Event, DTMF Four 4
```

Other Special Services activation

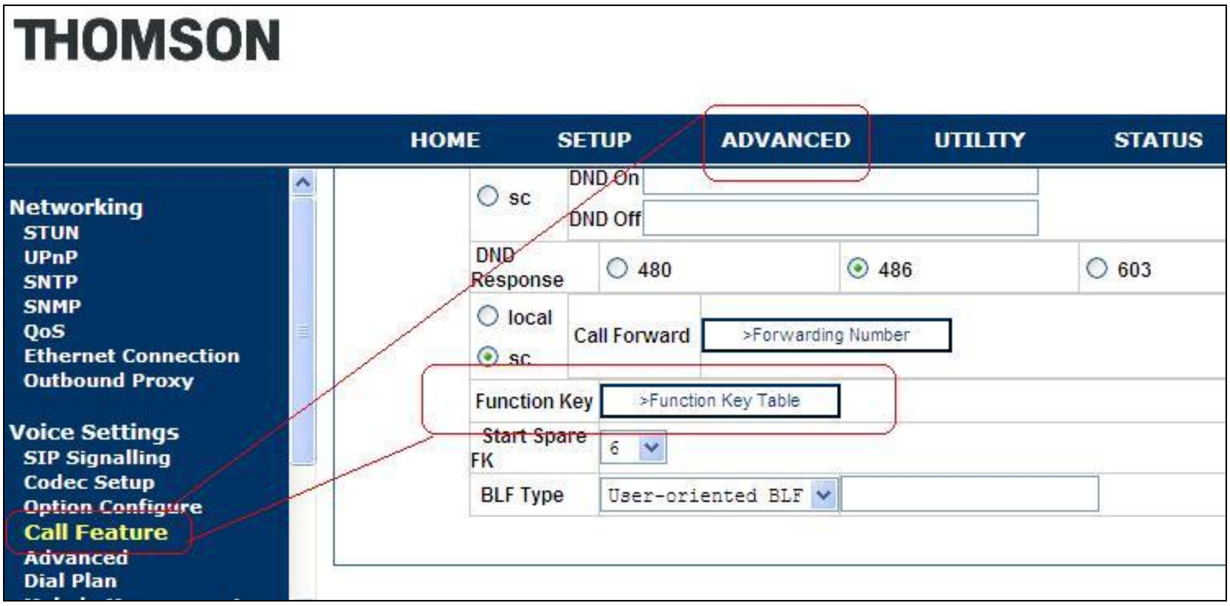
Note:

- Allowed starting from multiline+1
- Add a new type in the config file syntax "V" for serVice.
- use a separator to distinguish starcode from the phone number and from starcode trailer
  ie: FeatureKeyExt06=V/<sip:***8|1103|#**@domain.com>

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from:



○ sc | DND On | DND Off

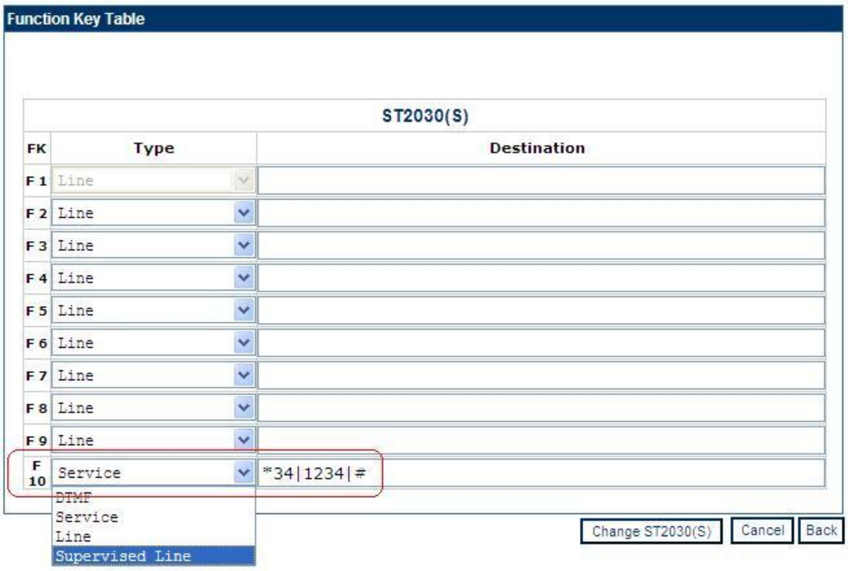
DND Response: ○ 480 ● 486 ○ 603

○ local ● sc | Call Forward

Function Key

Start Spare FK: 6

BLF Type: User-oriented BLF



ST2030(S)		
FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Line	
F 4	Line	
F 5	Line	
F 6	Line	
F 7	Line	
F 8	Line	
F 9	Line	
F 10	Service	*34|1234|#

B) Through APS:

For this purpose, new parameters have been included in the [sys] section of the Common/MAC config files, to be able to configure the feature.

```
[sys]
Current_Max_Multiline=5

FeatureKeyExt09=S/<sip:2209>
FeatureKeyExt10=V/<sip:*8|1103|#>
```

C) Through Telnet:

For this purpose, existing parameters are used, described as follows:

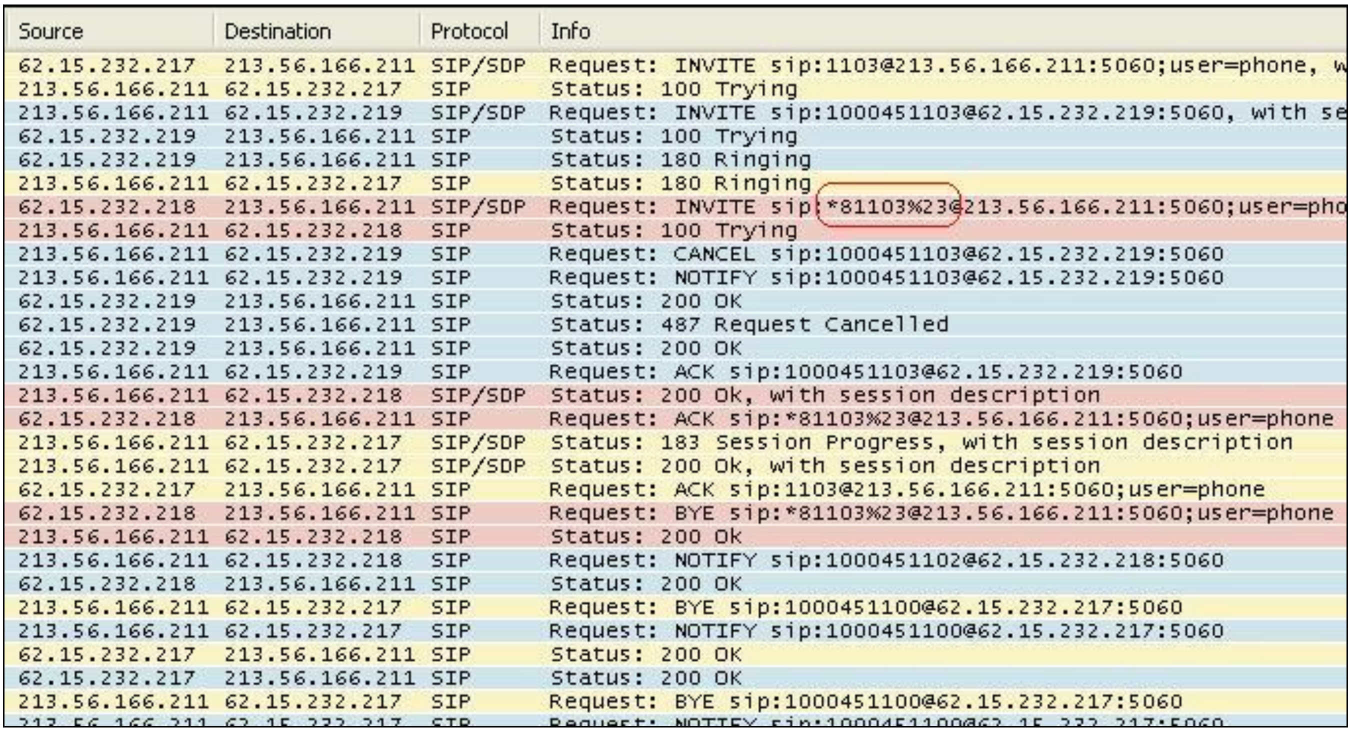
```
sys set fkuri id(1~66) uri_string
sys set fktype id(1~66) 0/1/2/3/4/5/6/7 (L/S/E/R/G/B/D/V)
```

Behavior

1. Press the Function key
2. The phone will display "Enter a number" displaying also the programmed number.
3. User will modify or not that number, and then press OK to dial out.
4. The phone will send an INVITE with TO header = SC + phoneNr + SC

If you press the FK during a call, it will be treated as another call (so the phone will put the current active call on hold, perform the action, and will wait for user to go back to the active call).

Call Flow:



Source	Destination	Protocol	Info
62.15.232.217	213.56.166.211	SIP/SDP	Request: INVITE sip:1103@213.56.166.211:5060;user=phone, w
213.56.166.211	62.15.232.217	SIP	Status: 100 Trying
213.56.166.211	62.15.232.219	SIP/SDP	Request: INVITE sip:1000451103@62.15.232.219:5060, with se
62.15.232.219	213.56.166.211	SIP	Status: 100 Trying
62.15.232.219	213.56.166.211	SIP	Status: 180 Ringing
213.56.166.211	62.15.232.217	SIP	Status: 180 Ringing
62.15.232.218	213.56.166.211	SIP/SDP	Request: INVITE sip:*81103%23@213.56.166.211:5060;user=pho
213.56.166.211	62.15.232.218	SIP	Status: 100 Trying
213.56.166.211	62.15.232.219	SIP	Request: CANCEL sip:1000451103@62.15.232.219:5060
213.56.166.211	62.15.232.219	SIP	Request: NOTIFY sip:1000451103@62.15.232.219:5060
62.15.232.219	213.56.166.211	SIP	Status: 200 OK
62.15.232.219	213.56.166.211	SIP	Status: 487 Request Cancelled
62.15.232.219	213.56.166.211	SIP	Status: 200 OK
213.56.166.211	62.15.232.219	SIP	Request: ACK sip:1000451103@62.15.232.219:5060
213.56.166.211	62.15.232.218	SIP/SDP	Status: 200 Ok, with session description
62.15.232.218	213.56.166.211	SIP	Request: ACK sip:*81103%23@213.56.166.211:5060;user=phone
213.56.166.211	62.15.232.217	SIP/SDP	Status: 183 Session Progress, with session description
213.56.166.211	62.15.232.217	SIP/SDP	Status: 200 Ok, with session description
62.15.232.217	213.56.166.211	SIP	Request: ACK sip:1103@213.56.166.211:5060;user=phone
62.15.232.218	213.56.166.211	SIP	Request: BYE sip:*81103%23@213.56.166.211:5060;user=phone
213.56.166.211	62.15.232.218	SIP	Status: 200 Ok
213.56.166.211	62.15.232.218	SIP	Request: NOTIFY sip:1000451102@62.15.232.218:5060
62.15.232.218	213.56.166.211	SIP	Status: 200 OK
213.56.166.211	62.15.232.217	SIP	Request: BYE sip:1000451100@62.15.232.217:5060
213.56.166.211	62.15.232.217	SIP	Request: NOTIFY sip:1000451100@62.15.232.217:5060
62.15.232.217	213.56.166.211	SIP	Status: 200 OK
62.15.232.217	213.56.166.211	SIP	Status: 200 OK
213.56.166.211	62.15.232.217	SIP	Request: BYE sip:1000451100@62.15.232.217:5060
213.56.166.211	62.15.232.217	SIP	Request: NOTIFY sip:1000451100@62.15.232.217:5060

ST20XX SIP New Features (SG vx.61)

Overview

This document describes a set of features included in ST2030 and S2022 SIP v1.61 and 3.61 respectively in order to improve their usability in different environments.

Ad-Hoc Conf (RFC 4579)

The ST20xx is now a "conference-aware" user agent which supports SIP conferencing call control conventions defined in the RFC-4579 (Call Control - Conferencing for User Agents) as a conference participant.

ST20xx as a conference-aware UA, is able to process SIP redirections such as described in Section 8.1.3.4 of RFC 3261. Can recognize the 'isfocus' feature parameter, can support REFER and SIP events. But as many of the servers which already support this feature as "focus UA" and "Conference factory" (like Netcentrex/Comverse, Broadsoft or Sylantro servers) don't support so far the subscriptions to conference package (RFC 4575 "Event Package for Conference State"), this subscribe mechanism is not implemented in the ST20xx.

MMI description on initiator side:

```
1. A and B are in a call.

2. A press "Conf" soft key (-> B is put on hold), then enters C phone's number.

3. C replies, A and C are in a 2nd call.

4. A press "Conf" again,  to switch in conference mode (-> C is put on hold).

5. A creates a conference by sending an INVITE to the Conference_Factory_URI (as described in the
section "5.4. INVITE: Creating a Conference Using Ad-Hoc SIP Methods" of the rfc-4579).
```

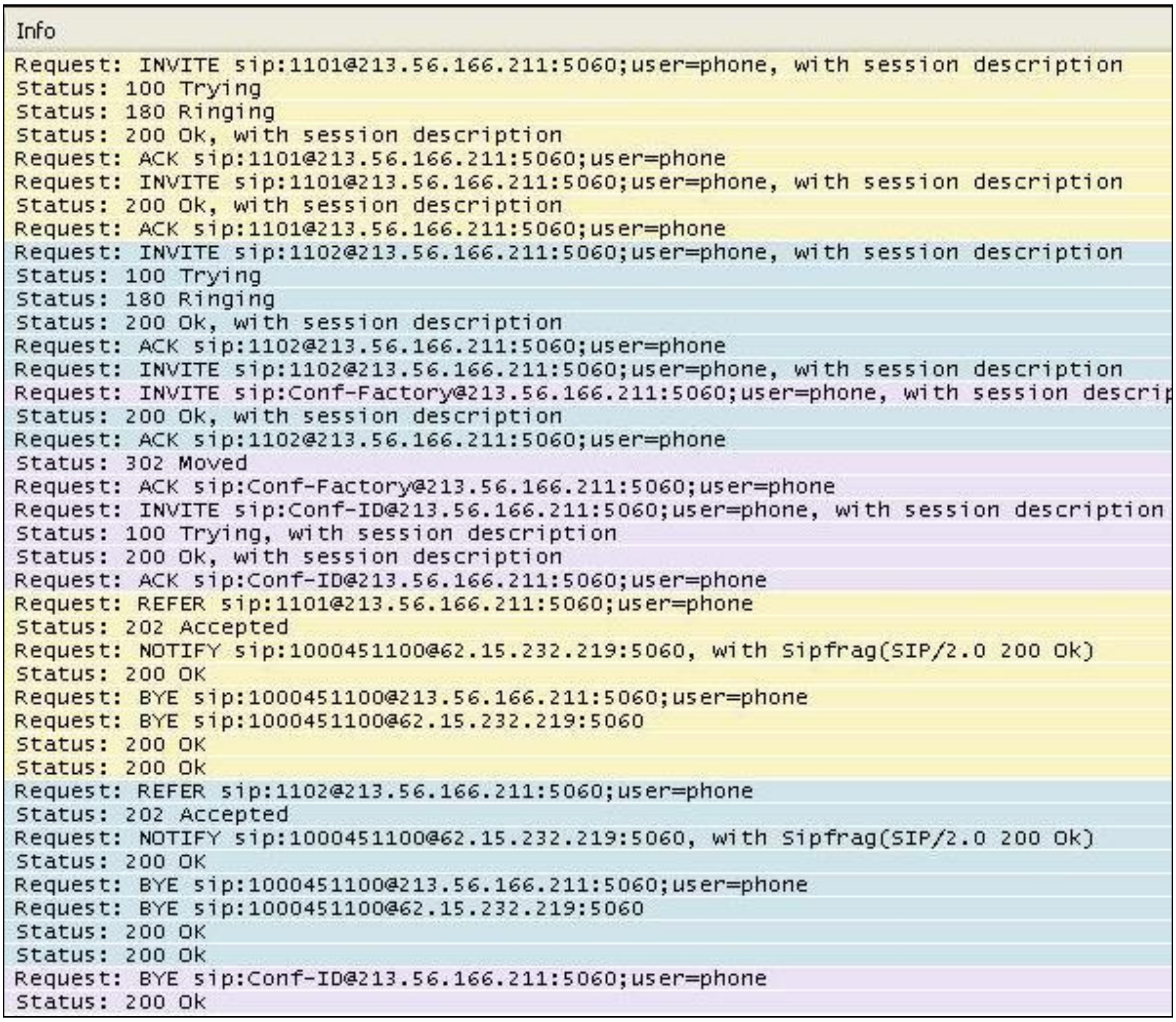
Then it will :

```
6. Retrieve the "conference ID" either in the 200 OK or in the 302 Moved response (in case the
phone receives a 302 Moved response, it will send another INVITE to the new URI in order to
really initiate the conference).

7. send REFER to B to move to conference ID (as described in section 5.6 of the rfc-4579)

8. send REFER to C to move to conference ID (as described in section 5.6 of the rfc-4579)
```

The expected behavior here is the Server takes care of capturing the Refers, then re-INVITE participants to the new call, and deletes old dialogs with initiator.

ST20xx as Initiator is ready to receive early Byes to its Refers, with or without received Notify (200)·

Here you can find an example Call-Flow of Centralized Conference:



```
Info
Request: INVITE sip:1101@213.56.166.211:5060;user=phone, with session description
Status: 100 Trying
Status: 180 Ringing
Status: 200 Ok, with session description
Request: ACK sip:1101@213.56.166.211:5060;user=phone
Request: INVITE sip:1101@213.56.166.211:5060;user=phone, with session description
Status: 200 Ok, with session description
Request: ACK sip:1101@213.56.166.211:5060;user=phone
Request: INVITE sip:1102@213.56.166.211:5060;user=phone, with session description
Status: 100 Trying
Status: 180 Ringing
Status: 200 Ok, with session description
Request: ACK sip:1102@213.56.166.211:5060;user=phone
Request: INVITE sip:1102@213.56.166.211:5060;user=phone, with session description
Request: INVITE sip:Conf-Factory@213.56.166.211:5060;user=phone, with session descri
Status: 200 Ok, with session description
Request: ACK sip:1102@213.56.166.211:5060;user=phone
Status: 302 Moved
Request: ACK sip:Conf-Factory@213.56.166.211:5060;user=phone
Request: INVITE sip:Conf-ID@213.56.166.211:5060;user=phone, with session description
Status: 100 Trying, with session description
Status: 200 Ok, with session description
Request: ACK sip:Conf-ID@213.56.166.211:5060;user=phone
Request: REFER sip:1101@213.56.166.211:5060;user=phone
Status: 202 Accepted
Request: NOTIFY sip:1000451100@62.15.232.219:5060, with Sipfrag(SIP/2.0 200 Ok)
Status: 200 OK
Request: BYE sip:1000451100@213.56.166.211:5060;user=phone
Request: BYE sip:1000451100@62.15.232.219:5060
Status: 200 OK
Status: 200 Ok
Request: REFER sip:1102@213.56.166.211:5060;user=phone
Status: 202 Accepted
Request: NOTIFY sip:1000451100@62.15.232.219:5060, with Sipfrag(SIP/2.0 200 Ok)
Status: 200 OK
Request: BYE sip:1000451100@213.56.166.211:5060;user=phone
Request: BYE sip:1000451100@62.15.232.219:5060
Status: 200 OK
Status: 200 Ok
Request: BYE sip:Conf-ID@213.56.166.211:5060;user=phone
Status: 200 Ok
```

Once A is in a call with the "focus UA", it could add another participant (D) in the same way:

```
9. A press “Conf” soft key then enters D phone’s number.

10. D replies, A and D are in a 2nd call.

11. A press “Conf” again.

10. A sends REFER to D to move to conference ID (as described in section 5.6 of the rfc-4579).
```

Another variant:

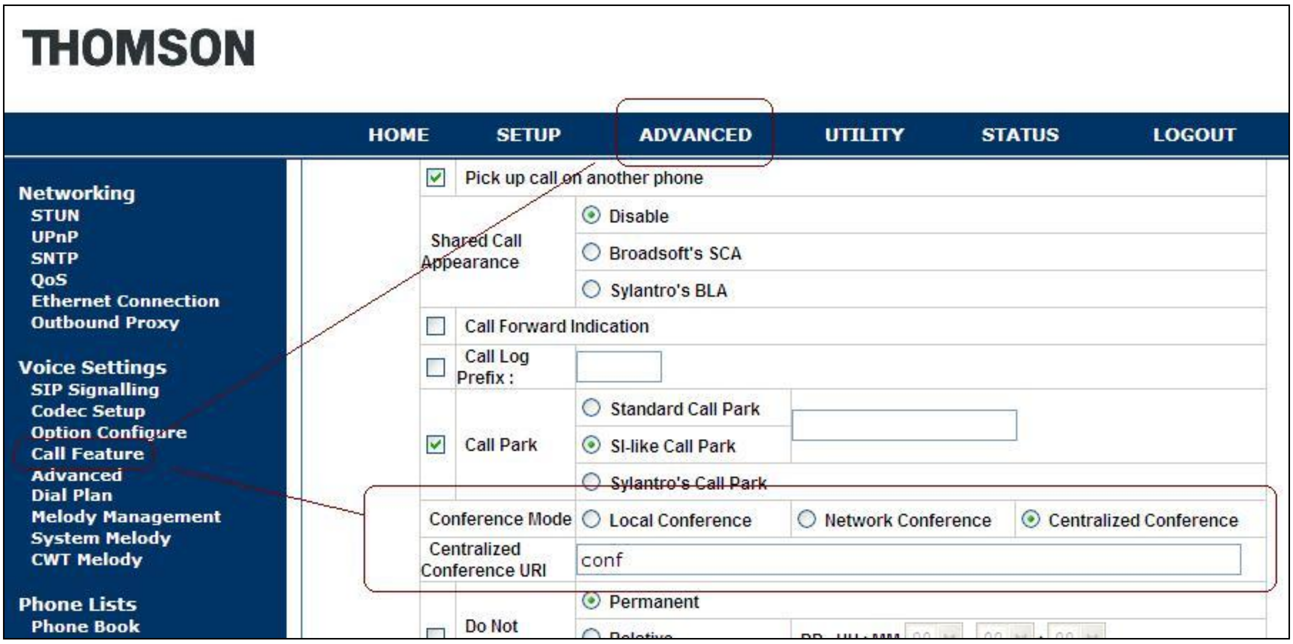
```
1. A and B are in a call.

2. C calls A

3. A replies, A and C are in a 2nd call, B is on hold.
```

```
4. A press "Join" to switch in conference mode.

5. Phone A shall send an INVITE to the Conference_Factory_URI.

6. It will retrieve the "conference ID" either in the 200 OK or the 302 Moved response.

7. send REFER to B to move to conference ID (as described in section 5.6 of the rfc-4579)

8. send REFER to C to move to conference ID (as described in section 5.6 of the rfc-4579)
```

Feature Activation

A) Through the WebGui:

The activation of this feature is accessible from the WebGui in the Advanced → Call Feature section.



☑ Pick up call on another phone

Shared Call Appearance: ● Disable ○ Broadsoft's SCA ○ Sylantro's BLA

☐ Call Forward Indication

☐ Call Log Prefix :

☑ Call Park: ○ Standard Call Park ● SI-like Call Park ○ Sylantro's Call Park

Conference Mode: ○ Local Conference ○ Network Conference ● Centralized Conference

Centralized Conference URI: conf

● Permanent

☐ Do Not

B) Through Telnet:

For this purpose, two new parameters have been created, described as follow:

```
sip set conference_mode [ 0(Local Conference)/
                          1(Network Conference)/
                          2(Centralized Conference (RFC4579))]

sip set conf_uri (Max 191 chars)
```

To configure, open a command line console, and telnet the phone:

```
*******************************************************************
**                                                              **
** IP Phone firmware                              V1.61         **
**        compiled on      Apr 28 2008 at         15:33:57      **
**                                                              **
**                      IP Phone VPD1020-D49(S)                 **
**                                                              **
*******************************************************************

Login: administrator
Password: ******


[administrator]# sip set conference_mode 2
[OK] Set OK

[administrator]# sip set conf_uri "conf"
[OK] Set OK

[administrator]# activate

[administrator]# commit

[administrator]# _
```

C) Through APS:

For this purpose, two new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the conference mode and the conference factory URI.

```
[sip]
...
ConferenceMode=2
ConferenceFactoryURI=conf
...
```

The parameter "ConferenceFactoryURI" could be either a SIP URI or a phone number. The default value is "conf".

Examples:
ConferenceFactoryURI=5000
ConferenceFactoryURI=conference@10.0.0.1
ConferenceFactoryURI=conf

Sylantro SIP-B

Correction of the known limitations and bug fixing of the Sylantro SIP-B provided in the release SG1.58.
Please refer to the document "ST20XX SIP Sylantro SIP-B features.pdf" for more Info.

Identity Header Precedence

This new feature specifies how the "source" for the caller/called party identity display, will be determined.

There are a number of SIP headers that can include calling (INVITE) or called (in 200 OK) party information.
A little history from "**draft-ietf-sip-privacy-xx**" to "**RFC3325**":

1. The calling party number was stored in the SIP From header in the initial SIP RFC 2543.

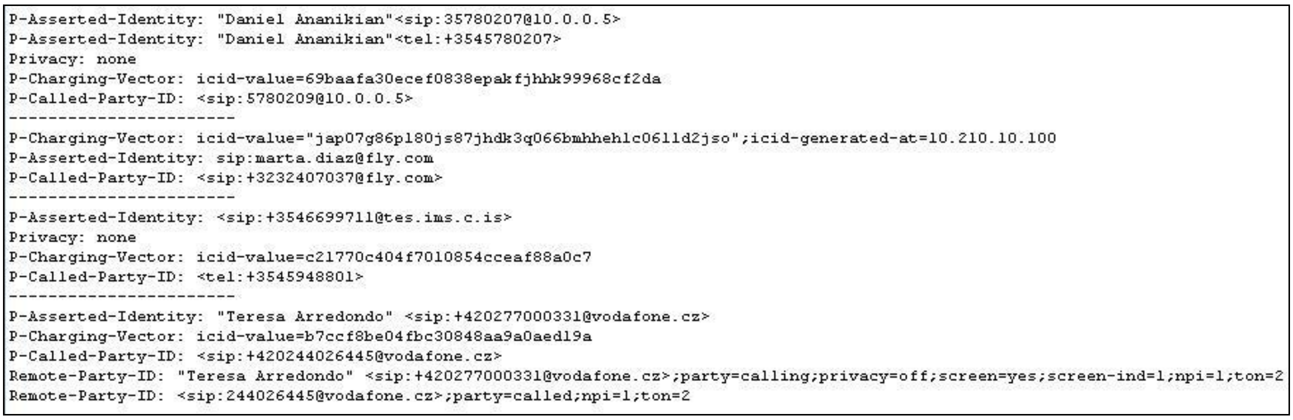
2. Later, Remote-Party-ID header was added in "draft-ietf-sip-privacy-xx" file,

3. Then Remote-Party-ID was not standard anymore and has been replaced with P-Asserted-Identity header (PAI) in RFC3325.

4. Finally, as User Agent can have multiple Identities, P-Preferred-Identity header was added to carry the identity the user wishes to be used.

This feature allows use of any of the possible headers where the calling number can be stored.

Some examples:



```
P-Asserted-Identity: "Daniel Ananikian"<sip:35780207@10.0.0.5>
P-Asserted-Identity: "Daniel Ananikian"<tel:+3545780207>
Privacy: none
P-Charging-Vector: icid-value=69baafa30ecef0838epakfjhhk99968cf2da
P-Called-Party-ID: <sip:5780209@10.0.0.5>
------------------------
P-Charging-Vector: icid-value="jap07g86pl80js87jhdk3q066bmhhehlc0611d2jso";icid-generated-at=10.210.10.100
P-Asserted-Identity: sip:marta.diaz@fly.com
P-Called-Party-ID: <sip:+3232407037@fly.com>
------------------------
P-Asserted-Identity: <sip:+3546699711@tes.ims.c.is>
Privacy: none
P-Charging-Vector: icid-value=c21770c404f7010854cceaf88a0c7
P-Called-Party-ID: <tel:+3545948801>
------------------------
P-Asserted-Identity: "Teresa Arredondo" <sip:+420277000331@vodafone.cz>
P-Charging-Vector: icid-value=b7ccf8be04fbc30848aa9a0aed19a
P-Called-Party-ID: <sip:+420244026445@vodafone.cz>
Remote-Party-ID: "Teresa Arredondo" <sip:+420277000331@vodafone.cz>;party=calling;privacy=off;screen=yes;screen-ind=1;npi=1;ton=2
Remote-Party-ID: <sip:244026445@vodafone.cz>;party=called;npi=1;ton=2
```

Different softswitch vendors use one or more of the different headers to provide information, and there is not a clear common criterion to all of them on the priority to be applied.

Moreover, ST20xx phones have a local phonebook.
Phonebook shall be always the 1st priority, but the header to look for in the phonebook will depend on the priority of the headers.

So two new variables are defined for “header priority”, default value being as follows:

```
CLIPDisplayPrior=ppreferred(1)passerted(2)remoteparty(3)from(4)

CalledDisplayPior=dialed(1)passerted(2)remoteparty(3)
```

Precedence mechanism to be applied is:

When a call is received, phone will check which is the header with the highest priority present in the INVITE ("presentation candidate") comparing with **`CLIPDisplayPrior`**. Once found, will check whether this address exists in the local phonebook or not. If it does, it will present phonebook info, otherwise, it will present the "candidate".

For outgoing calls, it is the same: check highest priority (dialed digits or header in 200 OK), to determine “presentation candidate” comparing with **`CalledDisplayPior`**. Will use that to check the local phonebook. If existing in the phonebook, it will present phonebook info. Otherwise, it will present the "candidate".

Feature Activation

A) Through the WebGui:

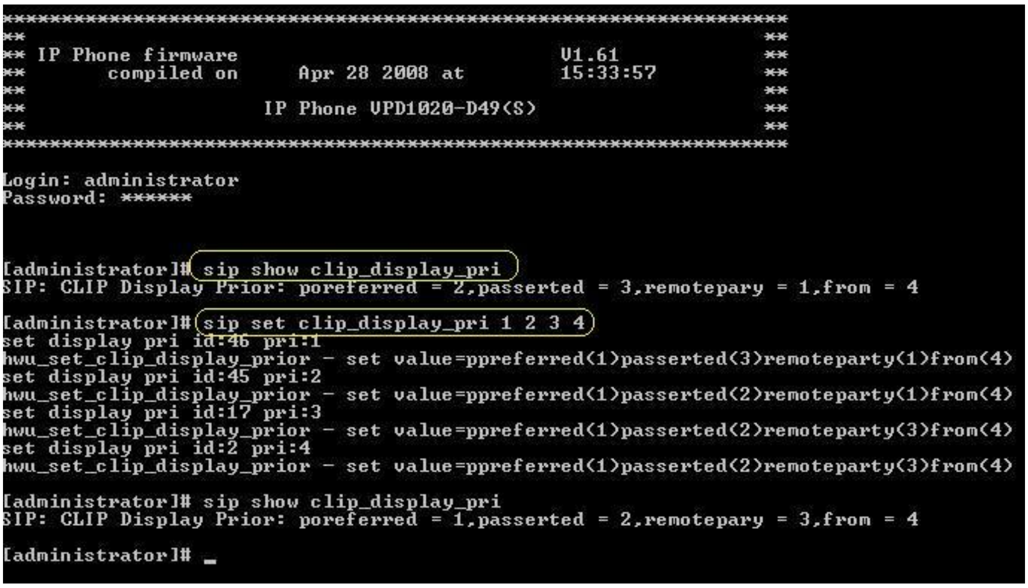
This feature can not be configured through the Web.

B) Through Telnet:

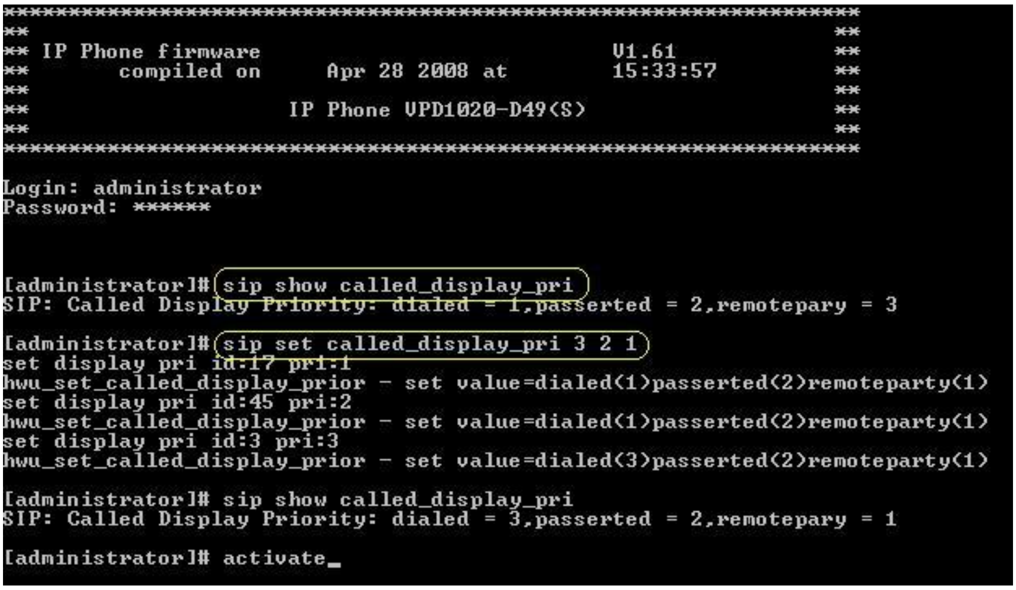
For this purpose, two new parameters have been created, described as follow:

```
sip set clip_display_pri <ppreferred> <passerted> <remoteparty> <from>
(Priority: 1~4, 1:Highest, 4:Lowest)

sip set called_display_pri <dialed> <passerted> <remoteparty>  (Priority: 1~3,
1:Highest, 3:Lowest)
```

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                            V1.61           **
**       compiled on      Apr 28 2008 at        15:33:57        **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show clip_display_pri
SIP: CLIP Display Prior: poreferred = 2,passerted = 3,remotepary = 1,from = 4

[administrator]# sip set clip_display_pri 1 2 3 4
set display pri id:46 pri:1
hwu_set_clip_display_prior - set value=ppreferred(1)passerted(3)remoteparty(1)from(4)
set display pri id:45 pri:2
hwu_set_clip_display_prior - set value=ppreferred(1)passerted(2)remoteparty(1)from(4)
set display pri id:17 pri:3
hwu_set_clip_display_prior - set value=ppreferred(1)passerted(2)remoteparty(3)from(4)
set display pri id:2 pri:4
hwu_set_clip_display_prior - set value=ppreferred(1)passerted(2)remoteparty(3)from(4)

[administrator]# sip show clip_display_pri
SIP: CLIP Display Prior: poreferred = 1,passerted = 2,remotepary = 3,from = 4

[administrator]# _
```



```
******************************************************************
**                                                              **
** IP Phone firmware                            V1.61           **
**       compiled on      Apr 28 2008 at        15:33:57        **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show called_display_pri
SIP: Called Display Priority: dialed = 1,passerted = 2,remotepary = 3

[administrator]# sip set called_display_pri 3 2 1
set display pri id:17 pri:1
hwu_set_called_display_prior - set value=dialed(1)passerted(2)remoteparty(1)
set display pri id:45 pri:2
hwu_set_called_display_prior - set value=dialed(1)passerted(2)remoteparty(1)
set display pri id:3 pri:3
hwu_set_called_display_prior - set value=dialed(3)passerted(2)remoteparty(1)

[administrator]# sip show called_display_pri
SIP: Called Display Priority: dialed = 3,passerted = 2,remotepary = 1

[administrator]# activate_
```

C) Through APS:

For this purpose, two new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the priority.

```
[sip]
...
CLIPDisplayPrior=ppreferred(1)passerted(2)remoteparty(3)from(4)
CalledDisplayPrior=dialed(1)passerted(2)remoteparty(3)
...
```

The number in parenthesis means priority, where 1 means the highest one.

NAT Keep Alive

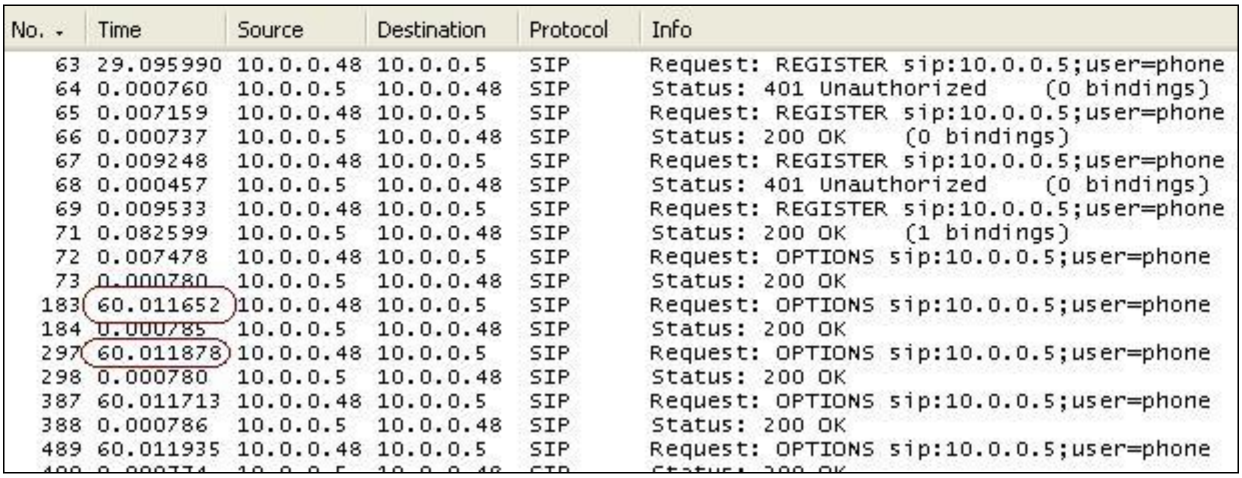
When the phone is installed behind a gateway or in another LAN than the Proxy, it is likely that we will face NAT traversal problems.
One solution is the automatic allocation of ports by the border router or gateway. But this allocation is not maintained forever.

So once registered with the SIP Registrar, the phone must maintain the channel open by sending keep-alive packets to the SIP server before the binding expires in the NAT device.

SIP OPTIONS message (cf. RFC3261 page 67) will be used to keep the NAT open. An example of the Option packet is as follow:

```
OPTIONS sip:10.0.0.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.48:5060;branch=z9hG4bK5708696430465310310-6772
From: <sip:2207@10.0.0.5:5060>;tag=c0a80101-1a74
To: <sip:10.0.0.5>
Call-ID: 34ac-c0a80101-2-2@10.0.0.48
CSeq: 1 OPTIONS
Max-Forwards: 70
Contact: <sip:2207@10.0.0.48:5060;user=phone>
Accept: application/sdp
User-Agent: THOMSON ST2030 hw5 fw1.61 00-14-7F-E1-81-F9
Content-Length: 0
```

Call flow example:



No.	Time	Source	Destination	Protocol	Info
63	29.095990	10.0.0.48	10.0.0.5	SIP	Request: REGISTER sip:10.0.0.5;user=phone
64	0.000760	10.0.0.5	10.0.0.48	SIP	Status: 401 Unauthorized (0 bindings)
65	0.007159	10.0.0.48	10.0.0.5	SIP	Request: REGISTER sip:10.0.0.5;user=phone
66	0.000737	10.0.0.5	10.0.0.48	SIP	Status: 200 OK (0 bindings)
67	0.009248	10.0.0.48	10.0.0.5	SIP	Request: REGISTER sip:10.0.0.5;user=phone
68	0.000457	10.0.0.5	10.0.0.48	SIP	Status: 401 Unauthorized (0 bindings)
69	0.009533	10.0.0.48	10.0.0.5	SIP	Request: REGISTER sip:10.0.0.5;user=phone
71	0.082599	10.0.0.5	10.0.0.48	SIP	Status: 200 OK (1 bindings)
72	0.007478	10.0.0.48	10.0.0.5	SIP	Request: OPTIONS sip:10.0.0.5;user=phone
73	0.000780	10.0.0.5	10.0.0.48	SIP	Status: 200 OK
183	60.011652	10.0.0.48	10.0.0.5	SIP	Request: OPTIONS sip:10.0.0.5;user=phone
184	0.000785	10.0.0.5	10.0.0.48	SIP	Status: 200 OK
297	60.011878	10.0.0.48	10.0.0.5	SIP	Request: OPTIONS sip:10.0.0.5;user=phone
298	0.000780	10.0.0.5	10.0.0.48	SIP	Status: 200 OK
387	60.011713	10.0.0.48	10.0.0.5	SIP	Request: OPTIONS sip:10.0.0.5;user=phone
388	0.000786	10.0.0.5	10.0.0.48	SIP	Status: 200 OK
489	60.011935	10.0.0.48	10.0.0.5	SIP	Request: OPTIONS sip:10.0.0.5;user=phone

Two parameters have been created:

1. The destination address for the SIP OPTIONS request. It could be either an SIP URI (or only the host part), an ip address or a domain name.

Examples:

`KeppAliveDest`=server@domain.com

`KeppAliveDest`=122.22.22.22

`KeppAliveDest`=domain.com

2. The time interval between 2 SIP OPTIONS requests.
The value is in seconds and between 0 and 600.
With 0 meaning disable sending Keep-Alive packets (default value).

Examples:

`KeepAliveTimer`=0

`KeepAliveTimer`=120

Feature Activation

A) Through the WebGui:

This feature can not be configured through the Web.

B) Through APS:

For this purpose, two new parameters have been included in the [sip] section of the Common/MAC config files, to be able to configure the nat.

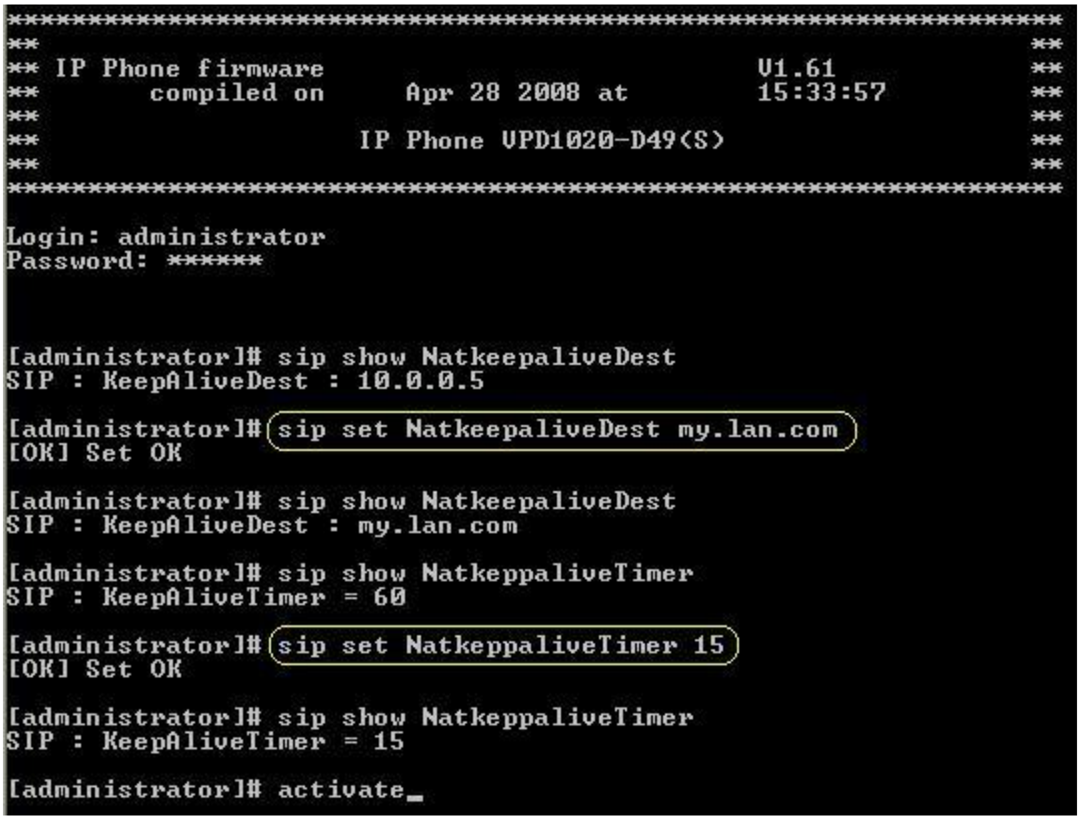
```
[sip]
...
KeppAliveDest=0.0.0.0
KeepAliveTimer=0
...
```

C) Through Telnet:

For this purpose, two new parameters have been created, described as follow:

```
sip set NatkeepaliveDest (Max 127 chars)

sip set NatkeppaliveTimer (range 0 - 600)
```

To configure, open a command line console, and telnet the phone:



```
*******************************************************************
**                                                               **
** IP Phone firmware                          V1.61              **
**       compiled on      Apr 28 2008 at      15:33:57           **
**                                                               **
**                    IP Phone VPD1020-D49(S)                    **
**                                                               **
*******************************************************************

Login: administrator
Password: ******


[administrator]# sip show NatkeepaliveDest
SIP : KeepAliveDest : 10.0.0.5

[administrator]# sip set NatkeepaliveDest my.lan.com
[OK] Set OK

[administrator]# sip show NatkeepaliveDest
SIP : KeepAliveDest : my.lan.com

[administrator]# sip show NatkeppaliveTimer
SIP : KeepAliveTimer = 60

[administrator]# sip set NatkeppaliveTimer 15
[OK] Set OK

[administrator]# sip show NatkeppaliveTimer
SIP : KeepAliveTimer = 15

[administrator]# activate_
```

Trusted IP for Notify “check-sync”

Up to now it was possible to send a SIP NOTIFY message to the phone, to launch remotely the automatic provisioning.
Extract of the APS document "ST2030-ST2022 AutoProvisioning-V0030.doc" :

3.4 SIP NOTIFY (SIP only)

The phone launches provisioning upon reception of a SIP NOTIFY :

```
NOTIFY sip: CLI@IP_ADDRESS_SIP_SERVER:5060;transport=UDP SIP/2.0
Call-ID: 03945c5ecd70248d6e76b8f0e6c64a13@10.107.111.189
CSeq: 460015 NOTIFY
From: <sip:webadmin@operator.com>;tag=8542
To: <sip: CLI@operator.com:5060>
Via: SIP/2.0/UDP
82.91.55.123:5061;branch=z9hg4bk917de6b406ebc8d056971e03991fdde6
Max-Forwards: 70
Contact: <sip:82.91.55.123:5061;transport=UDP>
Event: check-sync
Content-Length: 0
```

The phone may reboot or not, whether an upgrade is actually performed or not (i.e. a new FW or configuration is available).

There is a security issue: a cracker may broadcast such packets to the network frequently and cause the phones to reboot constantly. Therefore the usage of this feature is configurable via a “Use SIP NOTIFY to launch provisioning” parameter. Default value is 'No'.

To avoid attacks once the feature is enabled (as mentioned above), an additional parameter has been created to contain the authorized server from which the Notify is going to be allowed.
NOTIFY messages with event=check-sync, coming from other servers are ignored.
To minimize the impact of such attacks, the phone will not send any response.

The default value will be 0.0.0.0, which means the phone will accept all NOTIFY (check-sync) messages coming from everywhere.
This parameter shall be either an IP address or a domain name.

Remember that the proper APS pre-configuration should be:

```
[sip]
sw_notify_autoprovision=1
AuthNotifyCheckSync=10.0.0.1

[autoprovision]
AutoprovisionFlag=1
AutoprovisionTFTPServer=10.0.0.1
AutoprovisionConfigname=ST2030S.inf
```

An example Call Flow:

```
4.936427    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000817    10.0.0.5  10.0.0.55  SIP  Status: 401 Unauthorized    (0 bindings)
0.007231    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.003256    10.0.0.5  10.0.0.55  SIP  Status: 200 OK    (0 bindings)
0.005964    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.000456    10.0.0.5  10.0.0.55  SIP  Status: 401 Unauthorized    (0 bindings)
0.009705    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.003246    10.0.0.5  10.0.0.55  SIP  Status: 200 OK    (1 bindings)
1141.633840 10.0.0.1  10.0.0.55  SIP  Request: NOTIFY sip:2205@10.0.0.55
0.006190    10.0.0.55 10.0.0.5   SIP  Status: 200 OK
0.021143    10.0.0.55 10.0.0.1   TFTP Read Request, File: ST2030S_000E504E88E0.inf, Transfer type: octet
0.003462    10.0.0.1  10.0.0.55  TFTP Error Code, Code: File not found, Message: TFTP Error: File does not exist
10.007936   10.0.0.55 10.0.0.1   TFTP Read Request, File: st2030s.inf, Transfer type: octet
0.080049    10.0.0.55 10.0.0.1   TFTP Read Request, File: ST2030S_000E504E88E0.txt, Transfer type: octet
4.934673    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.000779    10.0.0.5  10.0.0.55  SIP  Status: 401 Unauthorized    (0 bindings)
0.008198    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.003850    10.0.0.5  10.0.0.55  SIP  Status: 200 OK    (1 bindings)
13.961019   10.0.0.1  10.0.0.55  SIP  Request: NOTIFY sip:2205@10.0.0.55
0.006157    10.0.0.55 10.0.0.5   SIP  Status: 200 OK
0.021461    10.0.0.55 10.0.0.1   TFTP Read Request, File: ST2030S_000E504E88E0.inf, Transfer type: octet
0.009948    10.0.0.1  10.0.0.55  TFTP Error Code, Code: File not found, Message: TFTP Error: File does not exist
10.006132   10.0.0.55 10.0.0.1   TFTP Read Request, File: st2030s.inf, Transfer type: octet
0.079233    10.0.0.55 10.0.0.1   TFTP Read Request, File: ST2030S_000E504E88E0.txt, Transfer type: octet
4.939727    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.000611    10.0.0.5  10.0.0.55  SIP  Status: 401 Unauthorized    (0 bindings)
0.005699    10.0.0.55 10.0.0.5   SIP  Request: REGISTER sip:10.0.0.5;user=phone
0.003553    10.0.0.5  10.0.0.55  SIP  Status: 200 OK    (1 bindings)
```

Feature Activation

A) Through the WebGui:

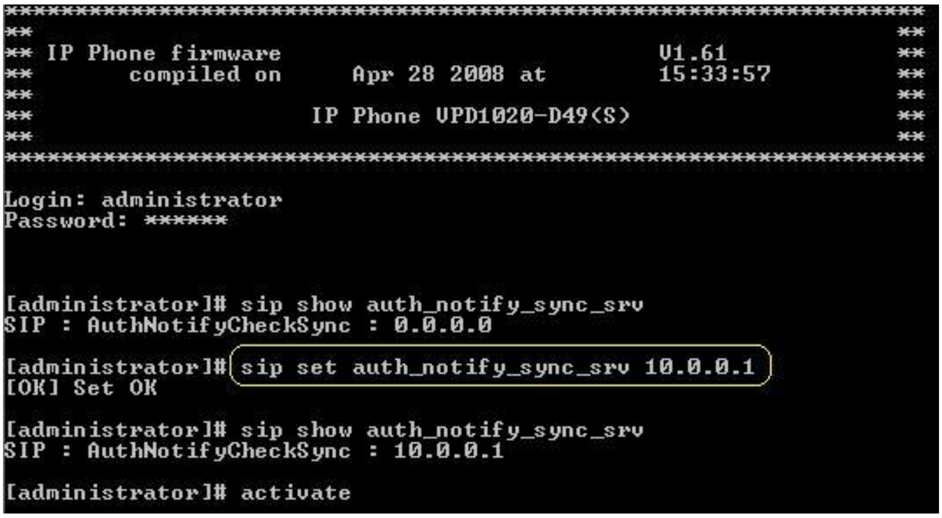
This feature can not be configured through the Web.

B) Through Telnet:

For this purpose, a new parameter has been created, described as follow:

```
sip set auth_notify_sync_srv (Max 127 chars)
```

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                            V1.61           **
**        compiled on      Apr 28 2008 at       15:33:57        **
**                                                              **
**                      IP Phone VPD1020-D49(S)                 **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show auth_notify_sync_srv
SIP : AuthNotifyCheckSync : 0.0.0.0

[administrator]# sip set auth_notify_sync_srv 10.0.0.1
[OK] Set OK

[administrator]# sip show auth_notify_sync_srv
SIP : AuthNotifyCheckSync : 10.0.0.1

[administrator]# activate
```

C) Through APS:

For this purpose, a new parameter has been included in the [sip] section of the Common/MAC config files, to be able to configure the server.

```
[sip]
...
AuthNotifyCheckSync=10.0.0.1
...
```

Reboot on Notify “check-sync; reboot=true”

In the implementation existing prior to this new version, when the phone received NTFY check-sync, it did not force reboot, but start APS procedure. Therefore, it was necessary to have TFTP/HTTP server and ".inf" filename pre-configured. Otherwise, APS could not not start.

But in scenarios where the APS must be done by DHCP, NTFY check-sync is not enough, because pre-configuration has a higher priority than the options obtained by DHCP.

So a new parameter will now be used to recognize when a NTFY check-sync needs to force the reboot of the phone.

This parameter will be located in the Event header of the NOTIFY like:

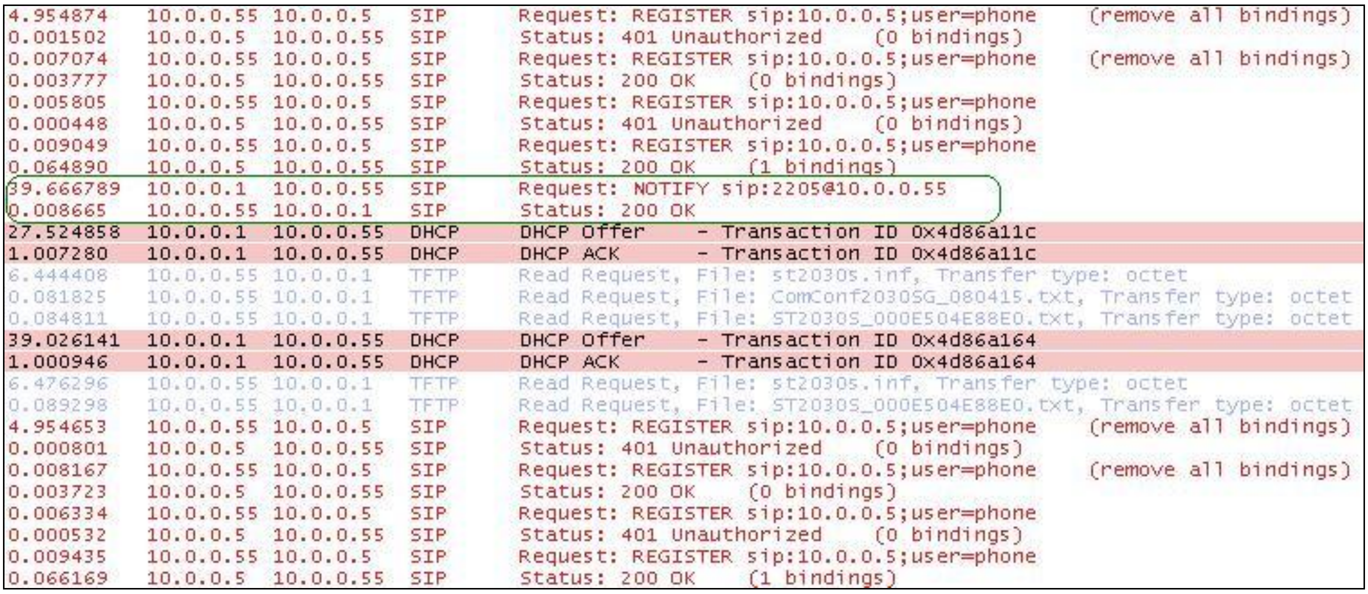
```
Event: check-sync                --> default, no reboot, only APS, preconfig needed
Event: check-sync;reboot=false --> default, no reboot, only APS, preconfig needed

Event: check-sync;reboot=true  --> force reboot, no need for preconfig (can still be used)
```

This parameter is optional.

Remember now, that the proper minimum configuration would be:

```
[sip]
sw_notify_autoprovision=1
AuthNotifyCheckSync=10.0.0.1

[autoprovision]
AutoprovisionFlag=1
```

An example Call Flow where the phone takes the APS configuration from the DHCP options would be:



```
4.954874   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.001502   10.0.0.5  10.0.0.55  SIP    Status: 401 Unauthorized    (0 bindings)
0.007074   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.003777   10.0.0.5  10.0.0.55  SIP    Status: 200 OK    (0 bindings)
0.005805   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone
0.000448   10.0.0.5  10.0.0.55  SIP    Status: 401 Unauthorized    (0 bindings)
0.009049   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone
0.064890   10.0.0.5  10.0.0.55  SIP    Status: 200 OK    (1 bindings)
39.666789  10.0.0.1  10.0.0.55  SIP    Request: NOTIFY sip:2205@10.0.0.55
0.008665   10.0.0.55 10.0.0.1   SIP    Status: 200 OK
27.524858  10.0.0.1  10.0.0.55  DHCP   DHCP Offer    - Transaction ID 0x4d86a11c
1.007280   10.0.0.1  10.0.0.55  DHCP   DHCP ACK      - Transaction ID 0x4d86a11c
6.444408   10.0.0.55 10.0.0.1   TFTP   Read Request, File: st2030s.inf, Transfer type: octet
0.081825   10.0.0.55 10.0.0.1   TFTP   Read Request, File: ComConf2030SG_080415.txt, Transfer type: octet
0.084811   10.0.0.55 10.0.0.1   TFTP   Read Request, File: ST2030S_000E504E88E0.txt, Transfer type: octet
39.026141  10.0.0.1  10.0.0.55  DHCP   DHCP Offer    - Transaction ID 0x4d86a164
1.000946   10.0.0.1  10.0.0.55  DHCP   DHCP ACK      - Transaction ID 0x4d86a164
6.476296   10.0.0.55 10.0.0.1   TFTP   Read Request, File: st2030s.inf, Transfer type: octet
0.089298   10.0.0.55 10.0.0.1   TFTP   Read Request, File: ST2030S_000E504E88E0.txt, Transfer type: octet
4.954653   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000801   10.0.0.5  10.0.0.55  SIP    Status: 401 Unauthorized    (0 bindings)
0.008167   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.003723   10.0.0.5  10.0.0.55  SIP    Status: 200 OK    (0 bindings)
0.006334   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone
0.000532   10.0.0.5  10.0.0.55  SIP    Status: 401 Unauthorized    (0 bindings)
0.009435   10.0.0.55 10.0.0.5   SIP    Request: REGISTER sip:10.0.0.5;user=phone
0.066169   10.0.0.5  10.0.0.55  SIP    Status: 200 OK    (1 bindings)
```

Downloadable/Uploadable Local Phonebook

As requested by many customers, we have included the possibility to save, edit and load the local phonebook of the IP phone.

This feature allows you (the user) to copy the local Phonebook directory from your ST2030 phone to your PC. So for example, when a phone is out of order or must be replaced or be reset to default, the user can now make a backup and then recover all the entries stored in its phonebook.

A directory entry will have:

- Name
- Phone number

The personal phonebook will use the following XML format:

```
<ThomsonPhoneBook>
    <DirectoryEntry>
        <Name>Alice Abbot</Name>
        <Telephone>2200</Telephone>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Bernard Bishop</Name>
        <Telephone>2201</Telephone>
    </DirectoryEntry>
    <DirectoryEntry>
        <Name>Carol Camps</Name>
        <Telephone>2202</Telephone>
    </DirectoryEntry>
</ThomsonPhoneBook>
```

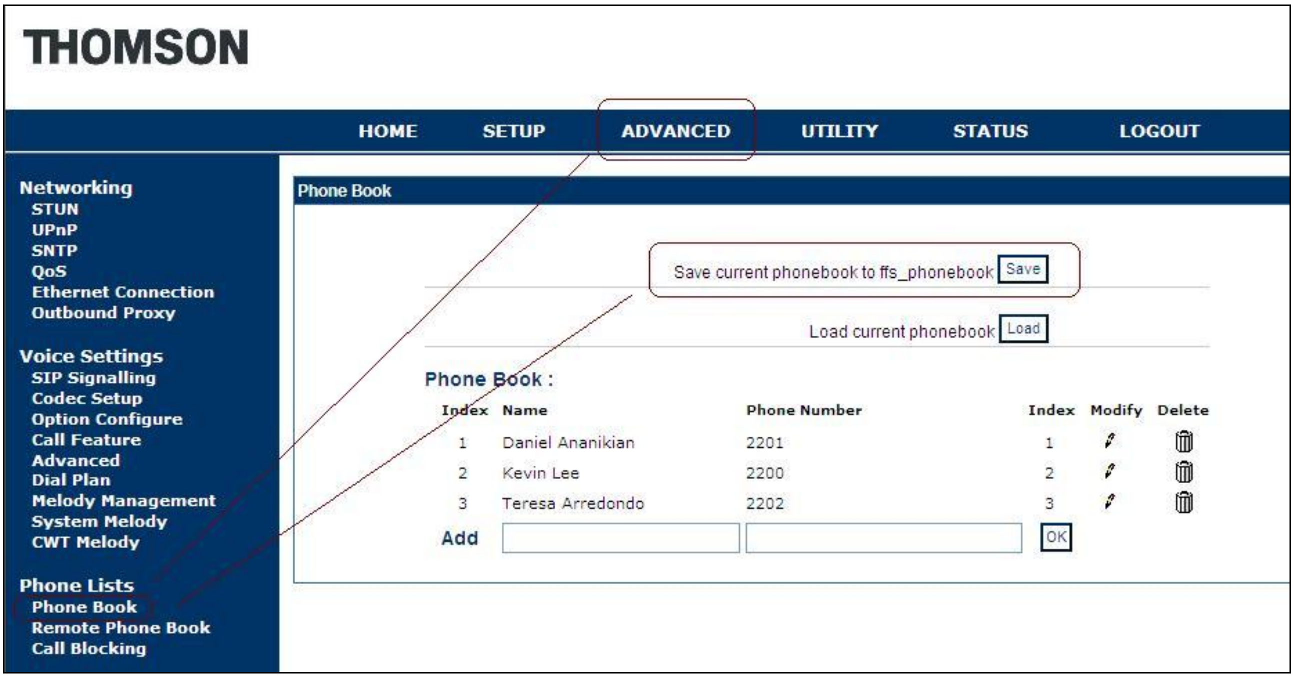
Up to 100 entries can be stored in the ST20xx phones.

Note that if you are going to upload a file through the WebGui, all current entries will be deleted, and all the entries which are in the file will be added.

Feature Activation

A) Through the WebGui:

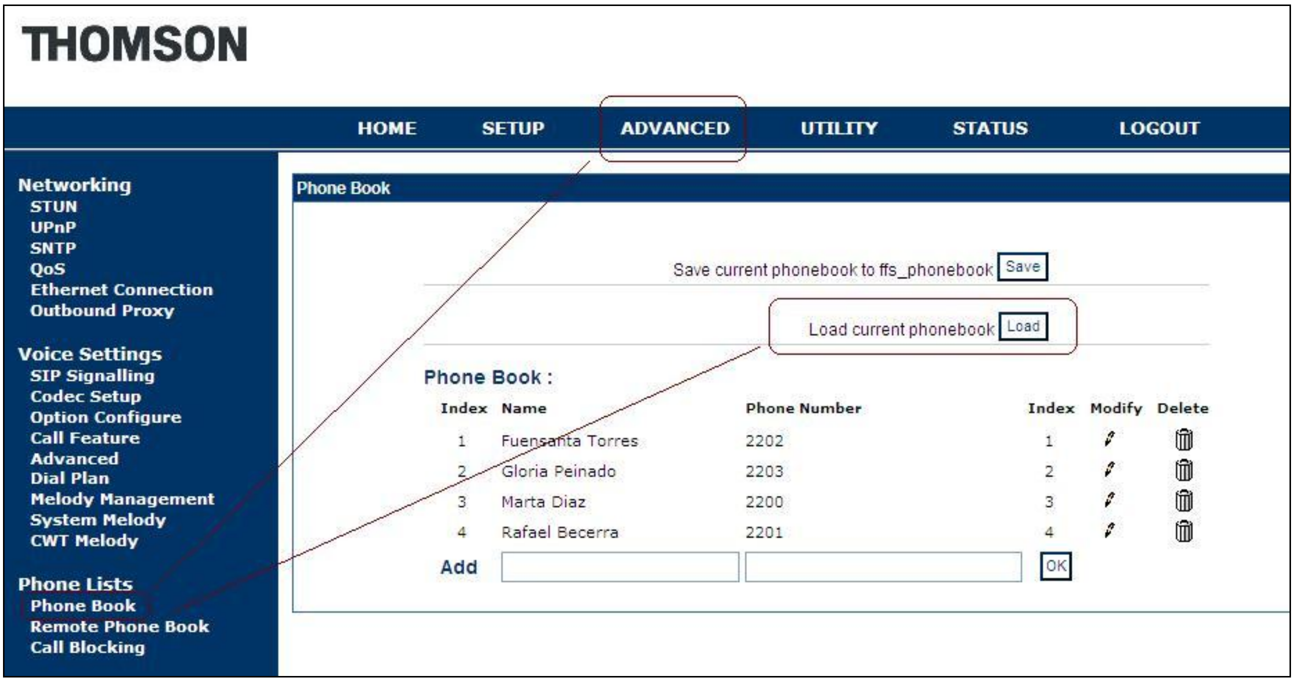
You can load/download through the “Advanced → Phone Book” Web page:



Phone Book :

Index	Name	Phone Number	Index	Modify	Delete
1	Daniel Ananikian	2201	1		
2	Kevin Lee	2200	2		
3	Teresa Arredondo	2202	3		

And :

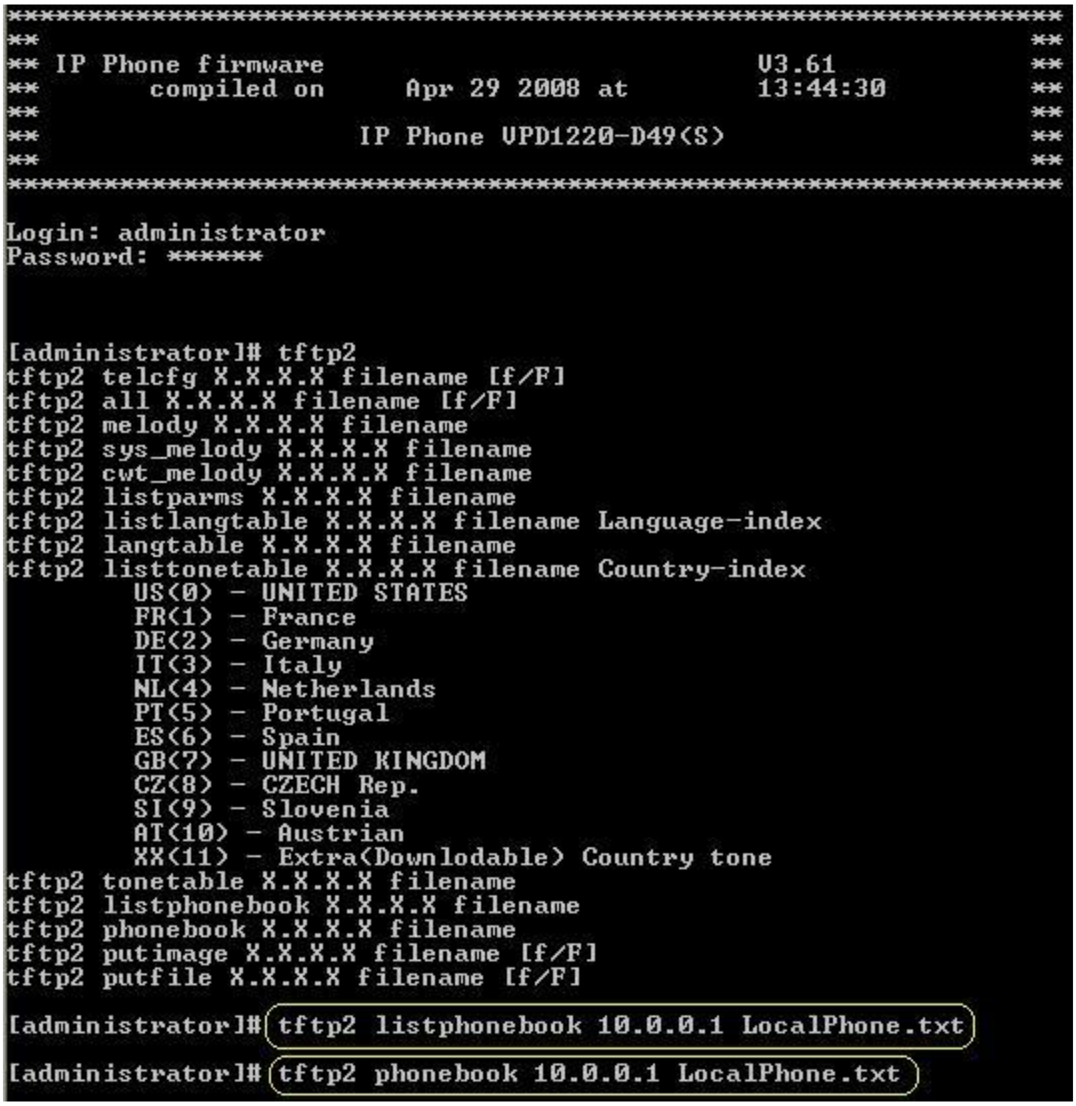


Phone Book :

Index	Name	Phone Number	Index	Modify	Delete
1	Fuensanta Torres	2202	1		
2	Gloria Peinado	2203	2		
3	Marta Diaz	2200	3		
4	Rafael Becerra	2201	4		

B) Through Telnet:

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                          V3.61             **
**       compiled on      Apr 29 2008 at      13:44:30          **
**                                                              **
**                    IP Phone VPD1220-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# tftp2
tftp2 telcfg X.X.X.X filename [f/F]
tftp2 all X.X.X.X filename [f/F]
tftp2 melody X.X.X.X filename
tftp2 sys_melody X.X.X.X filename
tftp2 cwt_melody X.X.X.X filename
tftp2 listparms X.X.X.X filename
tftp2 listlangtable X.X.X.X filename Language-index
tftp2 langtable X.X.X.X filename
tftp2 listtonetable X.X.X.X filename Country-index
        US(0) - UNITED STATES
        FR(1) - France
        DE(2) - Germany
        IT(3) - Italy
        NL(4) - Netherlands
        PT(5) - Portugal
        ES(6) - Spain
        GB(7) - UNITED KINGDOM
        CZ(8) - CZECH Rep.
        SI(9) - Slovenia
        AT(10) - Austrian
        XX(11) - Extra(Downlodable) Country tone
tftp2 tonetable X.X.X.X filename
tftp2 listphonebook X.X.X.X filename
tftp2 phonebook X.X.X.X filename
tftp2 putimage X.X.X.X filename [f/F]
tftp2 putfile X.X.X.X filename [f/F]

[administrator]# tftp2 listphonebook 10.0.0.1 LocalPhone.txt

[administrator]# tftp2 phonebook 10.0.0.1 LocalPhone.txt
```

Mixed paging Remote Phonebook

In the implementation prior to this sw version, ST20xx could accept 2 kinds of syntax in the XML reply from the server:

1- Less than 32 results to the query with directory entries:

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

2- Or More than 32 results to the query with menu items:

```
<ThomsonPhoneMenu>
<MenuItem>
      <Name>Arrh to Foulard</Name>
      <URL>http://www.server.com/get32results_1.php</URL>
</MenuItem>
...
<MenuItem>
      <Name>Roger to Ziad </Name>
      <URL>http://www.server.com/get32results_7.php</URL>
</MenuItem>
</ThomsonPhoneMenu>
```

If the phone could support a mix of these 2 syntaxes, the server could send a list of phone numbers followed by a link to the next part of the reply.
This could make the research more intuitive for the user.

So we have included the following feature: The phone can support a mixed syntax in the XML reply like:

```
<PhoneBook>
<MenuItem>
      <Name>Previous-Page...</Name>
      <URL>http://www.server.com/Paging.php?NAME=M&amp;PAGE=0</URL>
</MenuItem>
<DirectoryEntry>
      <Name>Maldonado, Juan</Name>
      <Telephone>10022382</Telephone>
</DirectoryEntry>
<DirectoryEntry>
      <Name>Manilla, Antonio</Name>
      <Telephone>10022383</Telephone>
</DirectoryEntry>
```

```
<DirectoryEntry>
      <Name>Marco, Tomás</Name>
      <Telephone>10022384</Telephone>
</DirectoryEntry>
<MenuItem>
      <Name>Next-Page...</Name>
      <URL>http://www.server.com/Paging.php?NAME=M&amp;PAGE=2</URL>
</MenuItem>
</PhoneBook>
```

In previous example, for "MenuItem" elements, the content of the tag "Name" would be literally shown on the screen, and when selected would launch the url contained in "URL" tag.

As an alternative, an additional tag has been defined within "MenuItem", in order to support keywords. The new tag is "Item", and this would be a usage example:

```
<PhoneBook>
<MenuItem>
      <Item>previous-page</Item>
      <URL>http://www.server.com/Paging.php?NAME=M&amp;PAGE=0</URL>
</MenuItem>
<DirectoryEntry>
      <Name>Maldonado, Juan</Name>
      <Telephone>10022382</Telephone>
</DirectoryEntry>
<DirectoryEntry>
      <Name>Manilla, Antonio</Name>
      <Telephone>10022383</Telephone>
</DirectoryEntry>
<DirectoryEntry>
      <Name>Marco, Tomás</Name>
      <Telephone>10022384</Telephone>
</DirectoryEntry>
<MenuItem>
      <Item>next-page</Item>
      <URL>http://www.server.com/Paging.php?NAME=M&amp;PAGE=2</URL>
</MenuItem>
</PhoneBook>
```

In that case, the text between <Item> start-tag and end-tag will not be displayed as it is. Rather the phone will display an associated string in the language table and thus linked to the selected language on the phone.
2 keywords are supported: "next-page" and "previous-page".

Additional strings generated for the language table:

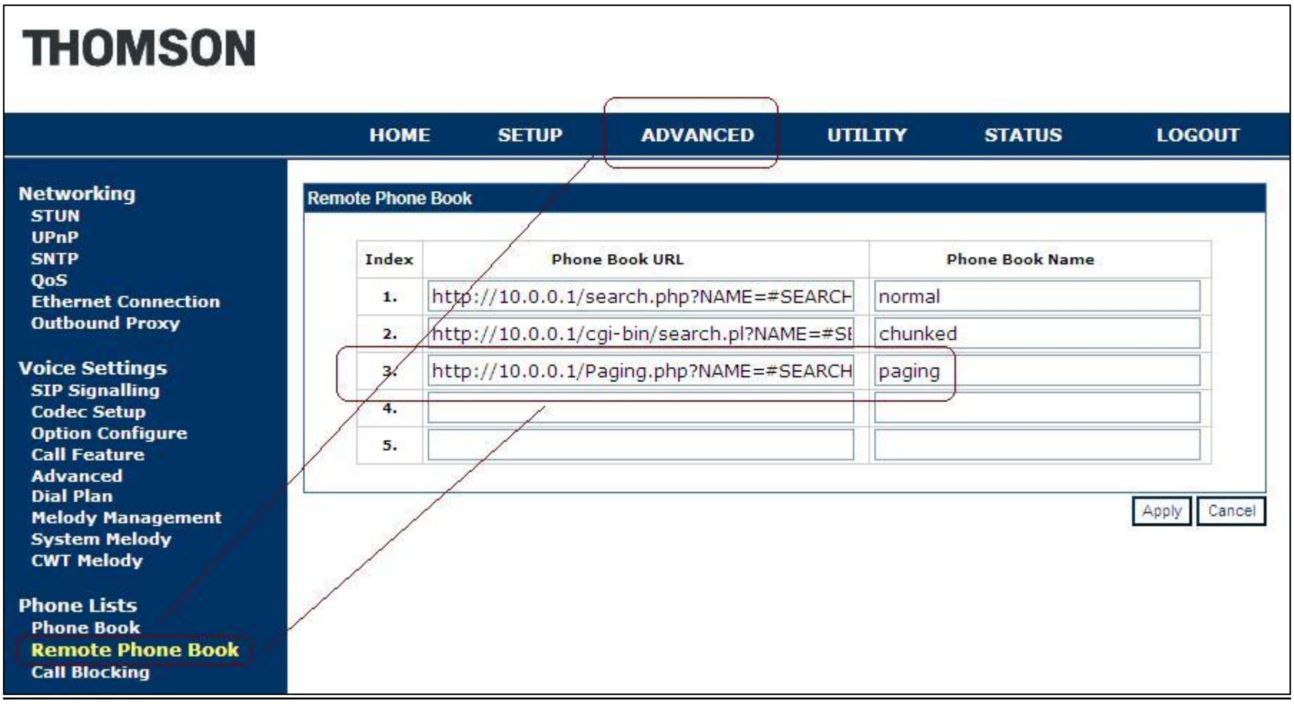
Keywords	**English**	**French**	**Spanish**	**German**	**Italian**
next-page	Next	Suivant	Siguiente	Nächste	Seguente
previous-page	Previous	Précédent	Anterior	Vorig	Precedente

Keywords	Norway	Russian	Portuguese	Netherlands
next-page	Neste	Следующий	Seguinte	Volgende
previous-page	Tidligere	Предыдущий	Precedente	Vorig

Feature Activation

Feature is activated by default, all you need is a remote phonebook using this structure.

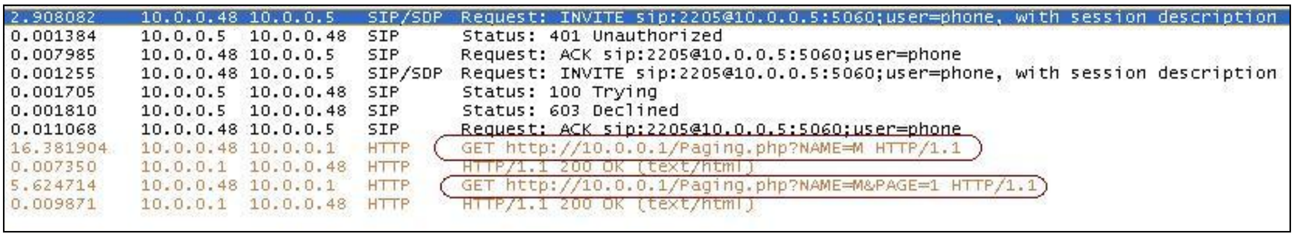
You can check this new feature by configuring the remote phonebook like this, and uploading to your http server the example provided with this release:



Remote Phone Book

Index	Phone Book URL	Phone Book Name
1.	http://10.0.0.1/search.php?NAME=#SEARCH	normal
2.	http://10.0.0.1/cgi-bin/search.pl?NAME=#SE	chunked
3.	http://10.0.0.1/Paging.php?NAME=#SEARCH	paging
4.		
5.		

An example of a normal Call Flow:



```
2.908082     10.0.0.48 10.0.0.5    SIP/SDP  Request: INVITE sip:2205@10.0.0.5:5060;user=phone, with session description
0.001384     10.0.0.5  10.0.0.48   SIP      Status: 401 Unauthorized
0.007985     10.0.0.48 10.0.0.5    SIP      Request: ACK sip:2205@10.0.0.5:5060;user=phone
0.001255     10.0.0.48 10.0.0.5    SIP/SDP  Request: INVITE sip:2205@10.0.0.5:5060;user=phone, with session description
0.001705     10.0.0.5  10.0.0.48   SIP      Status: 100 Trying
0.001810     10.0.0.5  10.0.0.48   SIP      Status: 603 Declined
0.011068     10.0.0.48 10.0.0.5    SIP      Request: ACK sip:2205@10.0.0.5:5060;user=phone
16.381904    10.0.0.48 10.0.0.1    HTTP     GET http://10.0.0.1/Paging.php?NAME=M HTTP/1.1
0.007350     10.0.0.1  10.0.0.48   HTTP     HTTP/1.1 200 OK (text/html)
5.624714     10.0.0.48 10.0.0.1    HTTP     GET http://10.0.0.1/Paging.php?NAME=M&PAGE=1 HTTP/1.1
0.009871     10.0.0.1  10.0.0.48   HTTP     HTTP/1.1 200 OK (text/html)
```

Login/logout – disable Subscribe to dialog

In the context of the Login/Logout feature, the phone subscribes to the Dialog Event before trying to register with a dynamic personal profile or static personal profile, to ensure any phone potentially registered with this profile is not in conversation or ringing).

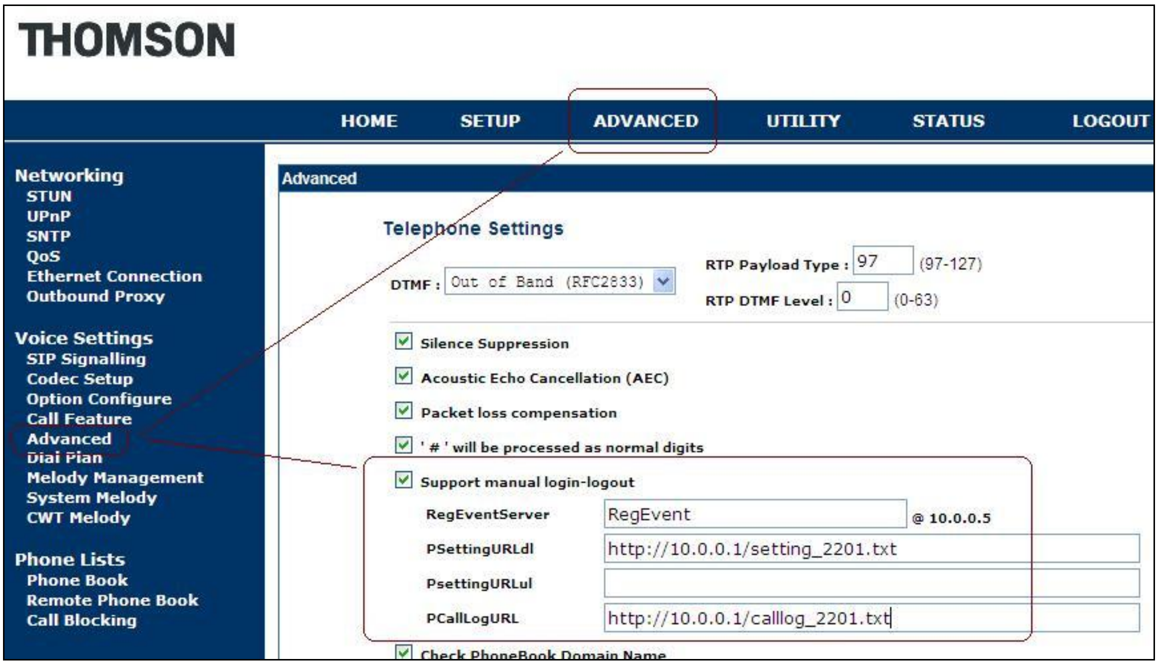
Some servers don't allow such subscriptions, or the real phone number is different from the Authentication ID (login parameter).
So the Login procedure is aborted due to non response from server side, or to an error message other than expected 404 Not Found.

To be able to set up a simple Login/Logout procedure without sending such Subscribe to Dialog event, a new parameter will be created:

1. If value=0 => the phone will skip the Subscription to the dialog event and directly Register with the personal dynamic profile.

2. If value=1 => Enable Subscribe to Dialog event before Register. If the SUBSCRIBE gets a 404 Not Found response, then PhoneA will consider the account is not in use and will proceed with the login process. If receives a Notify with the status as "terminated" then the login can go on. If it's not the case, the phone must abort the login procedure.

Default value will be 1 which means the phone will Subscribe to dialog event before registering (behavior prior to this new sw release).



Telephone Settings

DTMF : Out of Band (RFC2833)

RTP Payload Type : 97 (97-127)

RTP DTMF Level : 0 (0-63)

☑ Silence Suppression

☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ ' # ' will be processed as normal digits

☑ Support manual login-logout

RegEventServer: RegEvent @ 10.0.0.5

PSettingURLdl: http://10.0.0.1/setting_2201.txt

PsettingURLul:

PCallLogURL: http://10.0.0.1/calllog_2201.txt

☑ Check PhoneBook Domain Name

Feature Activation

A) Through the WebGui:

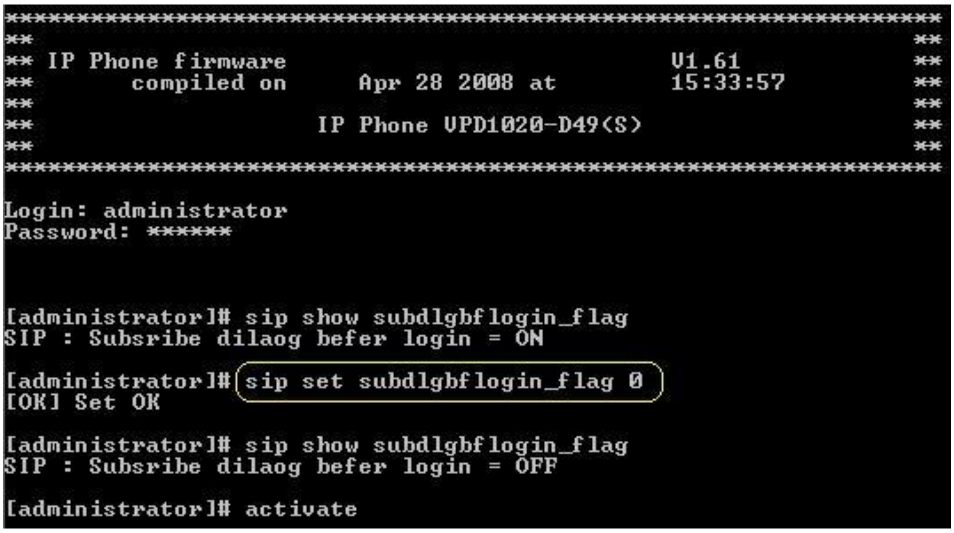
This feature can not be configured through the Web.

B) Through Telnet:

For this purpose, a new parameter has been created, described as follow:

```
sip set subdlgbflogin_flag [0(Disable)/ 1(Enable)]
```

To configure, open a command line console, and telnet the phone:



```
******************************************************************
**                                                              **
** IP Phone firmware                          V1.61             **
**       compiled on      Apr 28 2008 at      15:33:57          **
**                                                              **
**                     IP Phone VPD1020-D49(S)                  **
**                                                              **
******************************************************************

Login: administrator
Password: ******


[administrator]# sip show subdlgbflogin_flag
SIP : Subsribe dilaog befer login = ON

[administrator]# sip set subdlgbflogin_flag 0
[OK] Set OK

[administrator]# sip show subdlgbflogin_flag
SIP : Subsribe dilaog befer login = OFF

[administrator]# activate
```

C) Through APS:

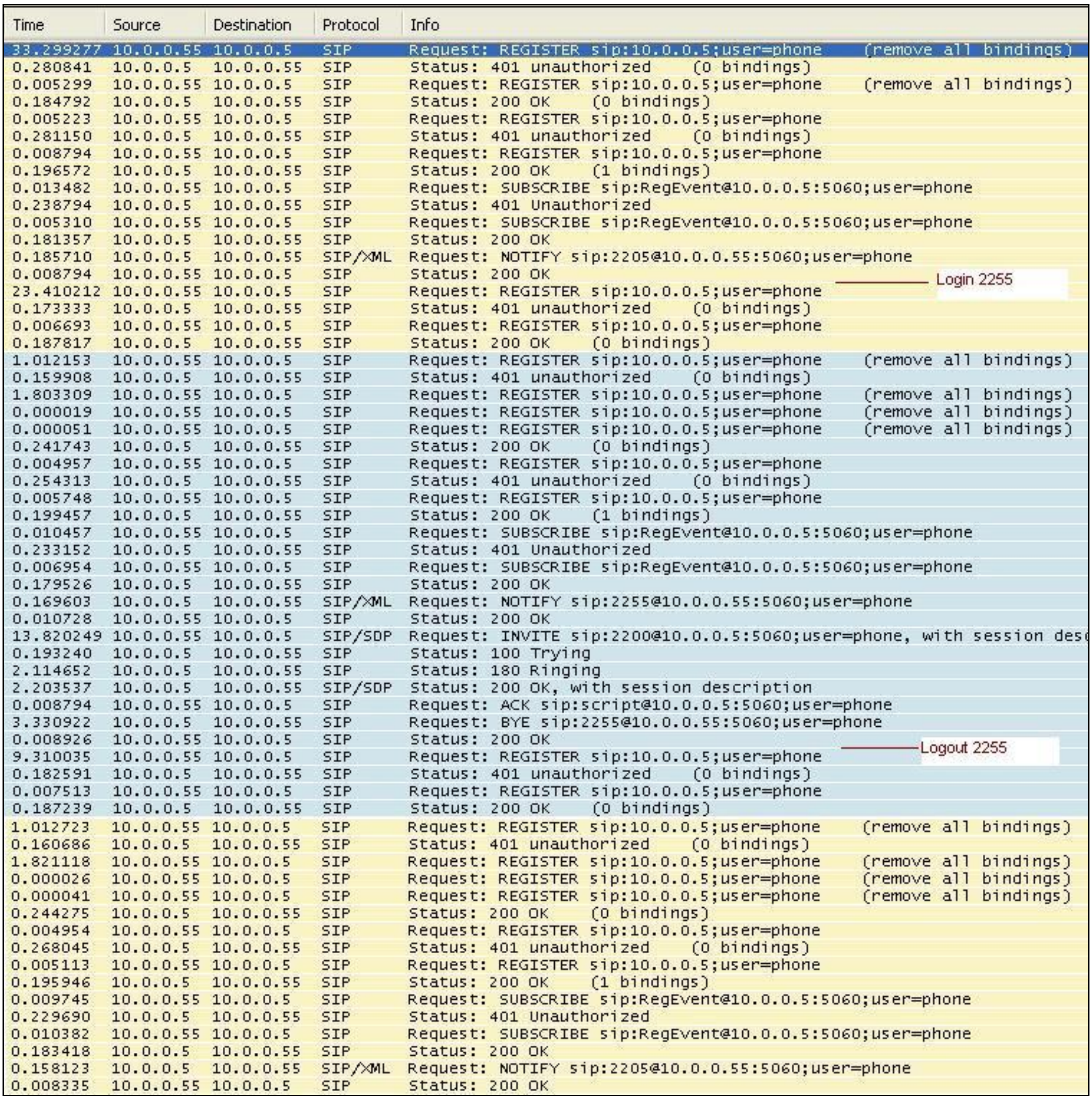
For this purpose, a new parameter has been included in the [sip] section of the Common/MAC config files, to be able to configure the feature. Remember the proper APS configuration to activate the Login/Logout feature at the highest feature level:

```
[ipp]
ManualLog=1

[sip]
SubscribeDilaogBeforeLogin=0
RegEventServer=RegEvent

[sys]
PSettingURLul=http://www.server.com/upload.php?Login_ID=#LOGIN;Passwrd=#PASSWORD
PSettingURLdl=http://www.server.com/download.php?Login_ID=#LOGIN;Passwrd=#PASSWORD
PCallLogURL=http://www.server.com/call-log.php?Login_ID=#LOGIN;Passwrd=#PASSWORD
```

And the minimum configuration for a basic Login/logout without Subscribe to dialog would be:

```
[ipp]
ManualLog=1

[sip]
SubscribeDilaogBeforeLogin=0
```

A call flow for reference :



```
Time      Source     Destination Protocol Info
33.299277 10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.280841  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.005299  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.184792  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (0 bindings)
0.005223  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.281150  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.008794  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.196572  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (1 bindings)
0.013482  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.238794  10.0.0.5   10.0.0.55   SIP      Status: 401 Unauthorized
0.005310  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.181357  10.0.0.5   10.0.0.55   SIP      Status: 200 OK
0.185710  10.0.0.5   10.0.0.55   SIP/XML  Request: NOTIFY sip:2205@10.0.0.55:5060;user=phone
0.008794  10.0.0.55  10.0.0.5    SIP      Status: 200 OK
23.410212 10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.173333  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.006693  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.187817  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (0 bindings)
1.012153  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.159908  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
1.803309  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000019  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000051  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.241743  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (0 bindings)
0.004957  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.254313  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.005748  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.199457  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (1 bindings)
0.010457  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.233152  10.0.0.5   10.0.0.55   SIP      Status: 401 Unauthorized
0.006954  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.179526  10.0.0.5   10.0.0.55   SIP      Status: 200 OK
0.169603  10.0.0.5   10.0.0.55   SIP/XML  Request: NOTIFY sip:2255@10.0.0.55:5060;user=phone
0.010728  10.0.0.55  10.0.0.5    SIP      Status: 200 OK
13.820249 10.0.0.55  10.0.0.5    SIP/SDP  Request: INVITE sip:2200@10.0.0.5:5060;user=phone, with session desc
0.193240  10.0.0.5   10.0.0.55   SIP      Status: 100 Trying
2.114652  10.0.0.5   10.0.0.55   SIP      Status: 180 Ringing
2.203537  10.0.0.5   10.0.0.55   SIP/SDP  Status: 200 OK, with session description
0.008794  10.0.0.55  10.0.0.5    SIP      Request: ACK sip:script@10.0.0.5:5060;user=phone
3.330922  10.0.0.5   10.0.0.55   SIP      Request: BYE sip:2255@10.0.0.55:5060;user=phone
0.008926  10.0.0.55  10.0.0.5    SIP      Status: 200 OK
9.310035  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.182591  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.007513  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.187239  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (0 bindings)
1.012723  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.160686  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
1.821118  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000026  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.000041  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone    (remove all bindings)
0.244275  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (0 bindings)
0.004954  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.268045  10.0.0.5   10.0.0.55   SIP      Status: 401 unauthorized    (0 bindings)
0.005113  10.0.0.55  10.0.0.5    SIP      Request: REGISTER sip:10.0.0.5;user=phone
0.195946  10.0.0.5   10.0.0.55   SIP      Status: 200 OK    (1 bindings)
0.009745  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.229690  10.0.0.5   10.0.0.55   SIP      Status: 401 Unauthorized
0.010382  10.0.0.55  10.0.0.5    SIP      Request: SUBSCRIBE sip:RegEvent@10.0.0.5:5060;user=phone
0.183418  10.0.0.5   10.0.0.55   SIP      Status: 200 OK
0.158123  10.0.0.5   10.0.0.55   SIP/XML  Request: NOTIFY sip:2205@10.0.0.55:5060;user=phone
0.008335  10.0.0.55  10.0.0.5    SIP      Status: 200 OK
```

ST20XX SIP New Features (SG vx.59)

Overview

This document describes a set of features included in ST2030 and S2022 SIP v1.59 and 3.59 respectively in order to improve their usability in different environments.

Tracing tool

Tracing tool is a new feature that allows administrators to get SIP level traces remotely. Through telnet phone interface, sent and received SIP messages flow can be obtained. SIP data provided by this tool is shown split in headers and packet body as follows:

```
Recv from udp: 10.0.0.5:5060 00:00:00:01:066 (937 bytes)

INVITE sip:2205@10.0.0.55:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK2b55f27f;rport
From: "Thomson-2204" <sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>
Contact: <sip:2204@10.0.0.5>
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 INVITE
User-Agent: Asterisk 1.4.11
Max-Forwards: 70
Remote-Party-ID: "Thomson-2204" <sip:2204@10.0.0.5>;privacy=off;screen=no
Date: Mon, 14 Jan 2008 14:17:14 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Alert-Info: <http://notused.com>;info=Clocks
P-Asserted-Identity: <sip:Asterisk@10.0.0.5>
Content-Type: application/sdp
Content-Length: 254

v=0
o=root 2036 2036 IN IP4 10.0.0.5
s=session
c=IN IP4 10.0.0.5
t=0 0
m=audio 17836 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

At first glance it can be noted there are two parts in the information for each packet.

The first one includes:

- packet sent or received
- destination address and port (for a sent packet) or source (for received packet).
- a time stamp, to locate the packets in the real time
- packet size

Second part is packet content itself.

Feature Activation

To use Tracing tool, a telnet connection has to be open. Then, feature will be enabled or disabled through a command: `sip tracer on / sip tracer off`.

Usual steps and output example:

```
telnet <ip>
Login: administrator
Password: 789234

[administrator]# sip tracer on

[administrator]#

[administrator]#  Recv from udp: 10.0.0.5:5060 00:00:00:01:066 (937 bytes)

INVITE sip:2205@10.0.0.55:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK2b55f27f;rport
From: "Thomson-2204" <sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>
Contact: <sip:2204@10.0.0.5>
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 INVITE
User-Agent: Asterisk 1.4.11
Max-Forwards: 70
Remote-Party-ID: "Thomson-2204" <sip:2204@10.0.0.5>;privacy=off;screen=no
Date: Mon, 14 Jan 2008 14:17:14 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Alert-Info: <http://notused.com>;info=Clocks
P-Asserted-Identity: <sip:Asterisk@10.0.0.5>
Content-Type: application/sdp
Content-Length: 254

v=0
o=root 2036 2036 IN IP4 10.0.0.5
s=session
c=IN IP4 10.0.0.5
t=0 0
m=audio 17836 RTP/AVP 0 8 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv

Sent to udp: 10.0.0.5:5060 00:00:00:01:096 (259 bytes) SIP/2.0 100 Trying

Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK2b55f27f;rport
From: "Thomson-2204"<sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 INVITE
Content-Length: 0

Sent to udp: 10.0.0.5:5060 00:00:00:01:126 (472 bytes)  SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK2b55f27f;rport
From: "Thomson-2204"<sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>;tag=c0a80101-79d86
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 INVITE
Allow: INVITE,ACK,BYE,CANCEL,OPTIONS,PRACK,SUBSCRIBE,NOTIFY,UPDATE,REFER,REGISTER,INFO
Contact: <sip:2205@10.0.0.55:5060>
Allow-Events: reg,refer,dialog,message-summary,check-sync,talk,hold
Content-Length: 0
```

```
Sent to udp: 10.0.0.5:5060 00:00:00:02:896 (691 bytes)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK2b55f27f;rport
From: "Thomson-2204"<sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>;tag=c0a80101-79d86
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 INVITE
Allow: INVITE,ACK,BYE,CANCEL,OPTIONS,PRACK,SUBSCRIBE,NOTIFY,UPDATE,REFER,REGISTER,INFO
Contact: <sip:2205@10.0.0.55:5060>
Allow-Events: reg,refer,dialog,message-summary,check-sync,talk,hold
Content-Type: application/sdp
Content-Length: 191

v=0
o=2205 499958 499958 IN IP4 10.0.0.55
s=-
c=IN IP4 10.0.0.55
t=0 0
m=audio 41000 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

Recv from udp: 10.0.0.5:5060 00:00:00:02:908 (445 bytes)

ACK sip:2205@10.0.0.55:5060 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK6c7d8777;rport
From: "Thomson-2204" <sip:2204@10.0.0.5>;tag=as1b93c676
To: <sip:2205@10.0.0.55:5060>;tag=c0a80101-79d86
Contact: <sip:2204@10.0.0.5>
Call-ID: 7b068be424cee3565279b5cb075e248b@10.0.0.5
CSeq: 102 ACK
User-Agent: Asterisk 1.4.11
Max-Forwards: 70
Remote-Party-ID: "Thomson-2204" <sip:2204@10.0.0.5>;privacy=off;screen=no
Content-Length: 0
```

Download and update tone and language tables

Downloading and updating Tone table or Language table capability has been added to improve and enhance admin possibilities in terms of managing languages and tones of his/her phone/s.

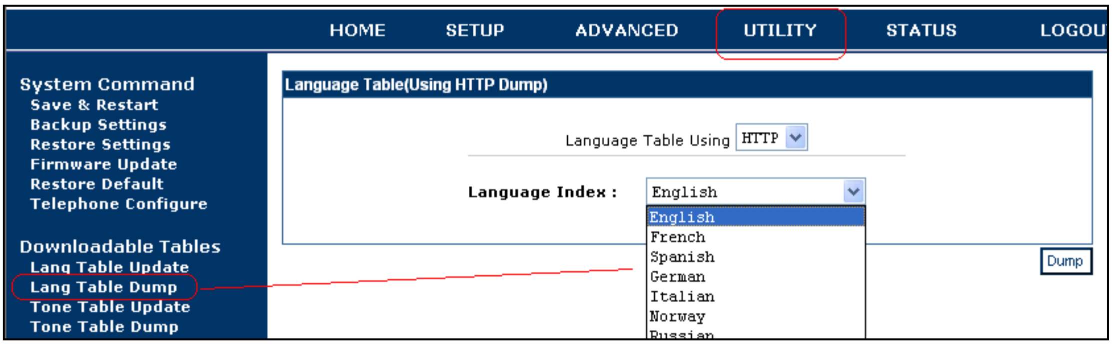
Phone provides 9 language tables (English, French, Spanish, German, Italian, Norway, Russian, Portuguese, Deutsch), and now, 1 more extra language that can be uploaded by the admin.

Likewise, phone provides 11 tone tables (United States, France, Germany, Italy, Netherlands, Portugal, Spain, United Kingdom, Czech Rep, Slovenia, Austrian), and now, 1 more extra tone table that can be uploaded by admin.

Currently, ST20xx only support one additional table for Language and Tone, and user **can not** modify the original Tone or Language tables.

How to get the Templates:

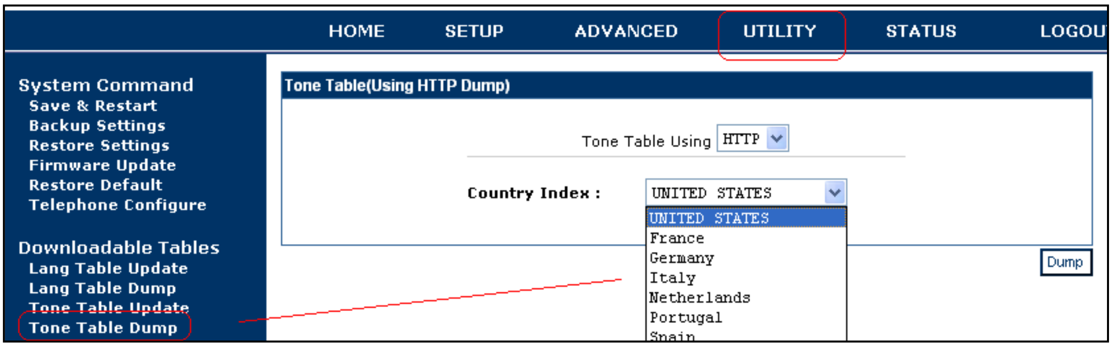
To create the new extra Tone or Language table, first you must dump one file via Web GUI, then modify and upload it, based on the dumped file.



Language Table(Using HTTP Dump)

Language Table Using HTTP

Language Index : English



Tone Table(Using HTTP Dump)

Tone Table Using HTTP

Country Index : UNITED STATES

You can also do it through Telnet:

```
[administrator]# tftp2
tftp2 telcfg X.X.X.X filename [f/F]
tftp2 all X.X.X.X filename [f/F]
tftp2 melody X.X.X.X filename
tftp2 sys_melody X.X.X.X filename
tftp2 cwt_melody X.X.X.X filename
tftp2 listparms X.X.X.X filename
tftp2 listlangtable X.X.X.X filename Language-index
tftp2 langtable X.X.X.X filename
tftp2 listtonetable X.X.X.X filename Country-index
        US(0) - UNITED STATES
        FR(1) - France
        DE(2) - Germany
        IT(3) - Italy
        NL(4) - Netherlands
        PT(5) - Portugal
        ES(6) - Spain
        GB(7) - UNITED KINGDOM
        CZ(8) - CZECH Rep.
        SI(9) - Slovenia
        AT(10) - Austrian
        XX(11) - Extra(Downlodable) Country tone
tftp2 tonetable X.X.X.X filename
tftp2 putimage X.X.X.X filename [f/F]
tftp2 putfile X.X.X.X filename [f/F]
```

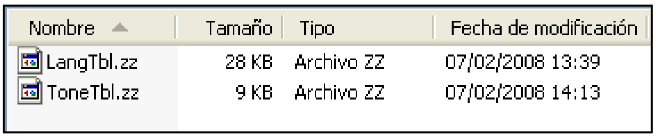
For example, to get US :

```
[administrator]# tftp2 listtonetable 10.0.0.1 ToneTbl.zz 0
```

or, to get English :

```
[administrator]# tftp2 listlangtable 10.0.0.1 LangTable.zz 1
```

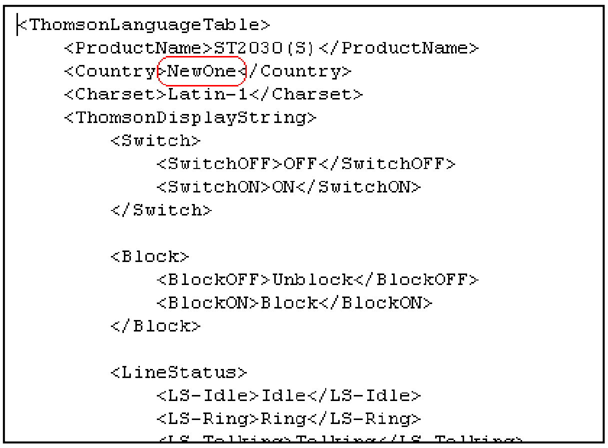
Finally we will have the Templates to create the new tables:



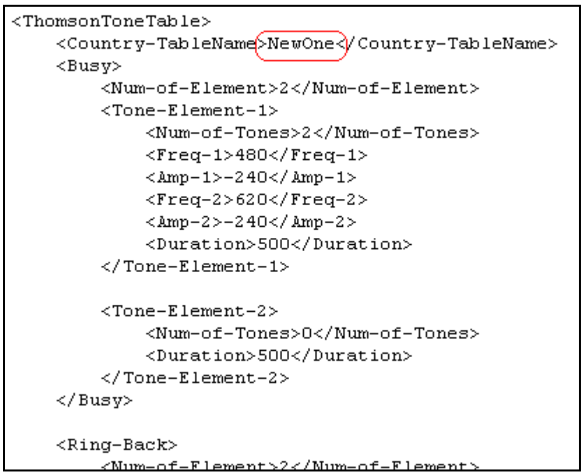
Nombre	Tamaño	Tipo	Fecha de modificación
LangTbl.zz	28 KB	Archivo ZZ	07/02/2008 13:39
ToneTbl.zz	9 KB	Archivo ZZ	07/02/2008 14:13

How to modify the Templates:

Tone/Language template files are “txt” files and follow XML format:



```
<ThomsonLanguageTable>
    <ProductName>ST2030(S)</ProductName>
    <Country>NewOne</Country>
    <Charset>Latin-1</Charset>
    <ThomsonDisplayString>
        <Switch>
            <SwitchOFF>OFF</SwitchOFF>
            <SwitchON>ON</SwitchON>
        </Switch>

        <Block>
            <BlockOFF>Unblock</BlockOFF>
            <BlockON>Block</BlockON>
        </Block>

        <LineStatus>
            <LS-Idle>Idle</LS-Idle>
            <LS-Ring>Ring</LS-Ring>
            <LS-Talking>Talking</LS-Talking>
```



```
<ThomsonToneTable>
    <Country-TableName>NewOne</Country-TableName>
    <Busy>
        <Num-of-Element>2</Num-of-Element>
        <Tone-Element-1>
            <Num-of-Tones>2</Num-of-Tones>
            <Freq-1>480</Freq-1>
            <Amp-1>-240</Amp-1>
            <Freq-2>620</Freq-2>
            <Amp-2>-240</Amp-2>
            <Duration>500</Duration>
        </Tone-Element-1>

        <Tone-Element-2>
            <Num-of-Tones>0</Num-of-Tones>
            <Duration>500</Duration>
        </Tone-Element-2>
    </Busy>

    <Ring-Back>
        <Num-of-Element>2</Num-of-Element>
```

If you modify:

```
<ThomsonToneTable>
    <Country-TableName>Spain 2</Country-TableName>
    <Busy>
        <Num-of-Element>2</Num-of-Element>
        ...
    </Busy>
...
</ThomsonToneTable>
```

this means the name of the extra tone table will be "Spain 2".

And:

```
<ThomsonLanguageTable>
    <ProductName>ST2030(S)</ProductName>
    <Country>Czech</Country>
    <Charset>Latin-2</Charset>
    <ThomsonDisplayString>
    ...
...
</ThomsonLanguageTable>
```

means the name of the extra language table will be "Czech".

The new language table should be based on Latin-1, Latin-2, Cyrillic and Hebrew character set. The character table name must be indicated in the language file in the tag:

```
<ThomsonLanguageTable>
    <ProductName>ST2030(S)</ProductName>
    <Country>Czech</Country>
    <Charset>Latin-2</Charset>
    <ThomsonDisplayString>
        <Switch>
            <SwitchOFF>OFF</SwitchOFF>
            <SwitchON>ON</SwitchON>
        </Switch>

        <Block>
            <BlockOFF>Odblokovat</BlockOFF>
            <BlockON>Blokovat</BlockON>
        </Block>

        <LineStatus>
            <LS-Idle>Neèinný</LS-Idle>
            ...
...
</ThomsonLanguageTable>
```

Words the admin can modify are the ones between xml tags.

Limitations:

1. Tone table limits - >

- Country-TableName is up to 64 characters,
- Num-of-Element is up to 6,

- Num-of-Tones is up to 4,
- Freq (frequency) is up to 3000Hz,
- Amp is between -800 ~ +30 (*0.1 db),
- Duration is up to 60000 seconds.

2. Language table limits - >

Length limit - > refer to the limit of MMI display

- LCD Central Line – 20 characters (ST2030),
  24 characters (ST2022);
- right upper corner of LCD – 6 characters;
- soft keys – 6 characters (ST2030),
  7 characters (ST2022).

Reserved characters should be used as follows within Language and Tone tables:

& → &

< → <

> → >

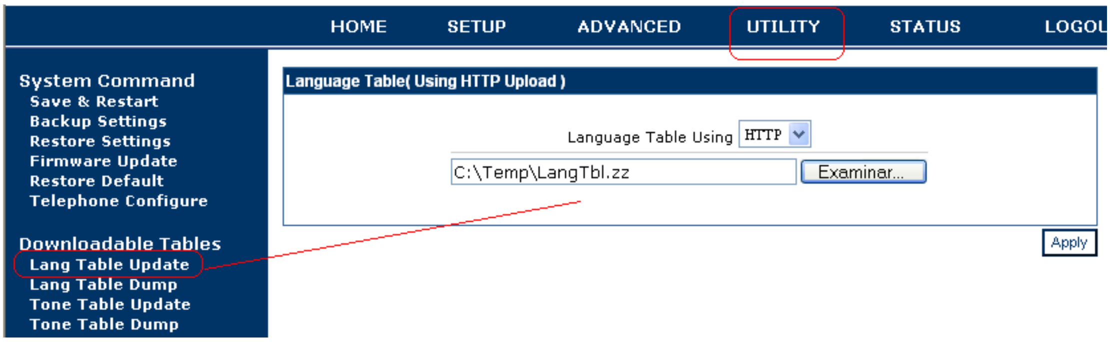
' → '

" → "

How to upload the Templates:

Language and tone tables can be uploaded using APS, through telnet application or via Web GUI.

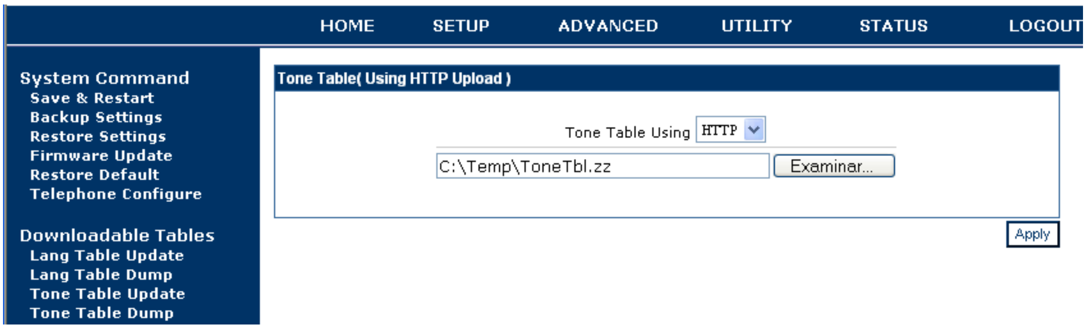
A) Through the Web Gui:



Language Table(Using HTTP Upload)

Language Table Using HTTP

C:\Temp\LangTbl.zz



Tone Table(Using HTTP Upload)

Tone Table Using HTTP

C:\Temp\ToneTbl.zz

B) Through Telnet

```
[administrator]# tftp2 langtable 10.0.0.1 LangTable.zz
```

```
[administrator]# tftp2 tonetable 10.0.0.1 ToneTbl.zz
```

C) Through APS:

- File Syntax for TFTP APS:

```
[application]
fw_filename=v2030SG.080227.1.59.3.zz

[config]
telcfg=TelConf2030SG_v1.59.3.txt
common_config=ComConf2030SG_v1.59.3.txt
melodies=Melodies.txt
system_melodies=Sys_Ringtones.txt
call_waiting_tone=Bellcore_CW.txt
tone_table=ToneTable.txt
language_table=LangTable.txt
```

- File Syntax for HTTP APS:

```
[application]
fw_url=http://10.8.1.217/v2030SG.080227.1.59.3.zz

[config]
common_config=http://10.8.1.217/ComConf2030SG_v1.59.3.txt
telcfg=http://10.8.1.217/TelConf2030SG_v1.59.3.txt
melodies=http://10.8.1.217/Melodies.txt
system_melodies=http://10.8.1.217/Sys_Ringtones.txt
call_waiting_tone=http://10.8.1.217/ Bellcore_CW.txt
tone_table=http://10.8.1.217/tonetable.txt
language_table=http://10.8.1.217/langtable.txt
config=http://10.8.1.217/
```

The files are temporarily uploaded, so they will be deleted after reset to default.

Feature Activation

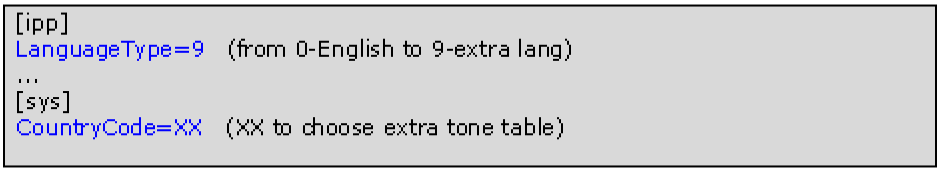
Once the files have been uploaded, user can activate them by the following means:

A) Through Keypad : Only Language table can be changed. Menu → Config → Personalize → Lang option → Edit → choose your own table and save.

B) Through Telnet : Only Tone table can be changed:

```
[administrator]# sys set country

        US - UNITED STATES
        FR - France
        DE - Germany
        IT - Italy
        NL - Netherlands
        PT - Portugal
        ES - Spain
        GB - UNITED KINGDOM
        CZ - CZECH Rep.
        SI - Slovenia
        AT - Austrian
        XX - Spain 2
[Error] Syntax error

[administrator]# sys set country XX
[OK] Set OK
```

C) Through APS :



```
[ipp]
LanguageType=9   (from 0-English to 9-extra lang)
...
[sys]
CountryCode=XX   (XX to choose extra tone table)
```

ST20XX SIP New Features (SG vx.58.6)

Overview

This document describes a set of features included in ST2030 and S2022 SIP v1.58.6 and 3.58.6 respectively in order to improve their usability in different environments.

SIP MESSAGE support (rfc 3428) for Status display applications

More and more, services supported locally by the phone are also supported centrally by the IP PBX or softswitch (call forwarding, call rejection, call block …).
Unlike the local services, there is no indication on the screen when the user activates these services on the server.
With the SIP MESSAGE method, the server could push short messages to the phone indicating its current status. These messages will be displayed on the phone's screen.
A parameter has been created with purpose to avoid attacks. Messages coming from other servers which were not the configured one on parameter will be rejected.
The text contained in the SIP MESSAGE body is displayed on the 4th line of the LCD for ST2030 and on the 2nd line of the LCD on the ST2022.

SIP MESSAGE text location on ST2030:

	Date	Hour	Icons	
	Phone Name			
	Phone Number			
	SIP MESSAGE text location			
<	Softkey1	Softkey2	Softkey3	>

SIP MESSAGE text location on ST2022:

	Date	Hour	Icons	
	Phone Number			
<	Softkey1	Softkey2	Softkey3	>

<-Blink->

	Date	Hour	Icons	
	SIP MESSAGE text			
<	Softkey1	Softkey2	Softkey3	>

Feature Activation

This feature does not need specific activation to be supported. But a parameter has been created to avoid messages attacks from non desired sources. Parameter is AuthMessageServer. Default value is 0.0.0.0, which means phone accepts all messages received from everywhere. Otherwise, to limit from which server messages can be accepted, this parameter should contain either SIP messages server IP address or domain name.

This parameter only can be configured in section [sip] of common or MAC config files.

For example:

```
[sip]
...
AuthMessageServer=192.168.1.1   (or AuthMessageServer=domain.com)
...
```

Disable Call Waiting Tone

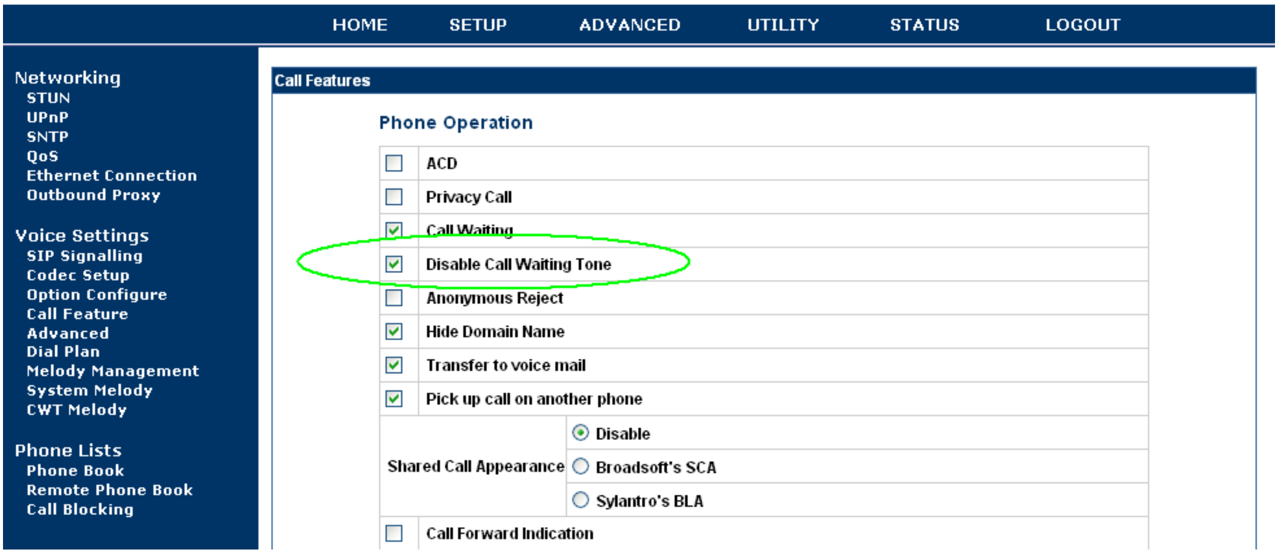
This new feature, we add the possibility to enable or disable the call waiting tone, in such a way that disable call waiting tone produces all the call waiting tones are muted.

Feature Activation

For this purpose, a new parameter has been included in section [sip] of common or MAC config files. So, you can enable or disable call waiting tone setting to 0 or 1, respectively. Default value is 0.

```
[sip]
...
DisableCWtone=0 (the call waiting tone is played)
DisableCWtone=1 (the call waiting tone is not played)
...
```

The activation of this feature is also accessible from the Web Gui in the Advanced | Call Features section.



Call Features

Phone Operation

☐ ACD
☐ Privacy Call
☑ Call Waiting
☑ Disable Call Waiting Tone
☐ Anonymous Reject
☑ Hide Domain Name
☑ Transfer to voice mail
☑ Pick up call on another phone
Shared Call Appearance: ● Disable ○ Broadsoft's SCA ○ Sylantro's BLA
☐ Call Forward Indication

Soft keys reordering

Up to now, the soft keys position was fixed. If you removed a soft key (by web GUI or APS), its position remained empty. So you could have 2 soft keys on the first page, another one on the second…
In order to avoid this situation, soft key reordering is possible since this version.
Also it could be interesting to put on the first page the most useful soft keys.

Feature Activation

The reordering of the soft keys only can be configured using APS. The soft keys order is indicated in section [sys] of common or MAC config files. Each function or service is associated to a soft key. It is the administrator responsibility to ensure a function is not used twice or is missing.
The function names syntax is independent from the language table. In case the function name is modified in any language, the above syntax must remain unchanged.
Following the country language selected, the corresponding wording will be displayed.

The following list has to be added to common or MAC file to determine the order:

```
[sys]
...
softkey01=TrVoiceMail
softkey02=CallLog
```

```
softkey03=PickUp
softkey04=LockPhone
softkey05=VoiceMail
softkey06=RetrievePark
softkey07=DNDstate
softkey08=ShortCut1
softkey09=ShortCut2
softkey10=ACDCheckIn
softkey11=ACDAvailable
softkey12=Login
...
```

Defined soft keys location is displayed on MMI screen as follow:

	Date	Hour	Icons	
		Phone Name		
		Phone Number		
<	Softkey1	Softkey2	Softkey3	>
<	Softkey4	Softkey5	Softkey6	>
<	Softkey7	Softkey8	Softkey9	>
<	Softkey10	Softkey11	Softkey12	>

Early Media Type parameter

Regarding Early media and RTP, policy so far was: 18x with sdp triggers RTP to be played, whereas 180 ringing without sdp triggers local ringing signal generation.
Some systems however have been found with call flows incompatible with this policy.

For this reason, a parameter has been created in order to decide whether a 180Ringing will generate local ringing or will continue playing RTP previously negotiated in the early session.

New parameter is Earlymediatype. See below in order to know how to configure it.

Feature Activation

This parameter can be configured using APS through common and MAC config files, where you can find Earlymediatype in section [sip]. It can be also modified via telnet using command sip set early_media_type.

Default value is 0. That means phone will switch to local ring tone if it receives a 180 Ringing (no sdp) response, regardless if an RTP stream corresponding to the early media session is present.
Setting Earlymediatype parameter to 1, phone will continue playing RTP stream corresponding to the early media session even if a 180 Ringing (no sdp) response is received.

```
[sip]
...
Earltmediatype=0 (switch to local ringing generation
                  if 180 (no sdp) is received)
...
```

Or

```
[sip]
...
Earlymediatype=1 (do not switch to local ringing if 180 (no sdp) is
                  received, and still playing the incomming RTP pkg)
...
```

ST2030 SIP New Features (SG v1.56)

Overview

This document describes a set of features included in ST2030 SIP v1.56 in order to improve its usability in different environments.

Login/Logout

Login/Logout feature allows the user to register easily with his own parameters (username and password) on any ST2030S or ST2022S in a location.

A Registration Event server as described by RFC3680 (A Session Initiation Protocol (SIP) Event Package for Registrations) is needed for this feature to work. Interested parties can ask for a complete specificacion of this service to their technical customer support.

There are two different applications for login/logout: substitution and free sitting.
In Substitution scenario, user will be able to login on any phone on the same network in order to get, for a while, the rights allowed to your telephone line (ie: external calls, international calls, your remote phone book …).
Free-sitting is a typical application in call-centers. When the employee arrives at work, he takes place in front of any free phone; he presses login key on the ST2030S then enters his login and password.

When this feature is enabled, behaviour will depend on active SIP profile as follows.

In Substitution scenario, phones have one active profile which they will normally use (can be profiles 1, 2 or 3), plus a backup profile for emergency which is profile 4.
In Free-sitting scenario, phones have as active profile the profile 4. This profile will be populated with emergency account information, which they will use while no dynamic personal profile is active.

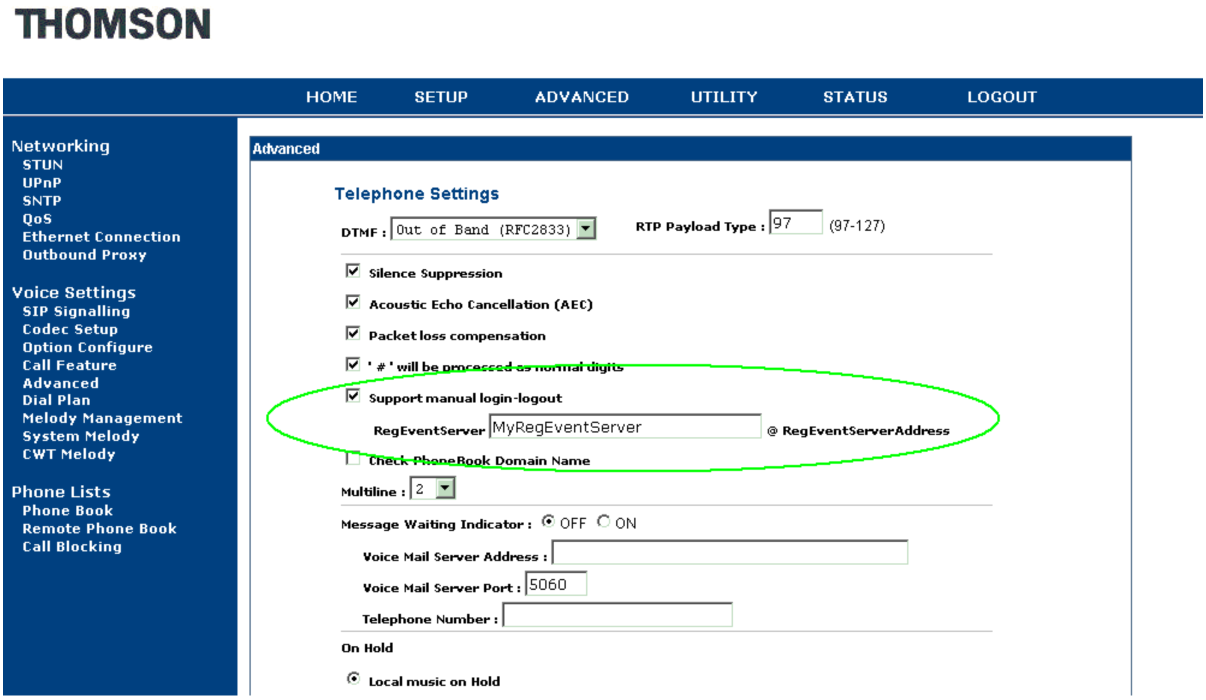
The login function allows the phone to register (and/or authenticate) on a SIP server with two parameters: username and password. The parameter username is common to the fields “Phone Number”, “Phone Name” and “Authentication ID”. The parameter password is the same as the “Password” field. Each user must have his own parameters.
The logout function allows the phone to return to its initial profile, recovering the user his static personal profile in case of phone is in substitution scenario or the backup profile if phone is in free-sitting scenario.

Feature activation

This feature can be cofigured via APS setting to 1 ManualLog parameter and adding the user part of the registration events server uri in RegEventServer parameter in common or MAC file:

```
[ipp]
...
ManualLog=1
...

[sip]
...
RegEventServer=MyRegEventServer
...
```

Or via Web GUI in section “Advanced | Voice Settings | Advanced” where it is necessary to enable Support manual login-logout to fill RegEventServer field. This field contains the user part of the registration events server uri, and the domain is automatically added when you enter Domain Name Server in the active profile.



Telephone Settings

DTMF : Out of Band (RFC2833) RTP Payload Type : 97 (97-127)

☑ Silence Suppression

☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ ' # ' will be processed as normal digits

☑ Support manual login-logout

RegEventServer MyRegEventServer @ RegEventServerAddress

☐ Check PhoneBook Domain Name

Multiline : 2

Message Waiting Indicator : ● OFF ○ ON

Voice Mail Server Address :

Voice Mail Server Port : 5060

Telephone Number :

On Hold

● Local music on Hold

Broadsoft Shared Call Appearance

This new feature makes the phone compatible with Broadsoft Shared Call Appearance facility.

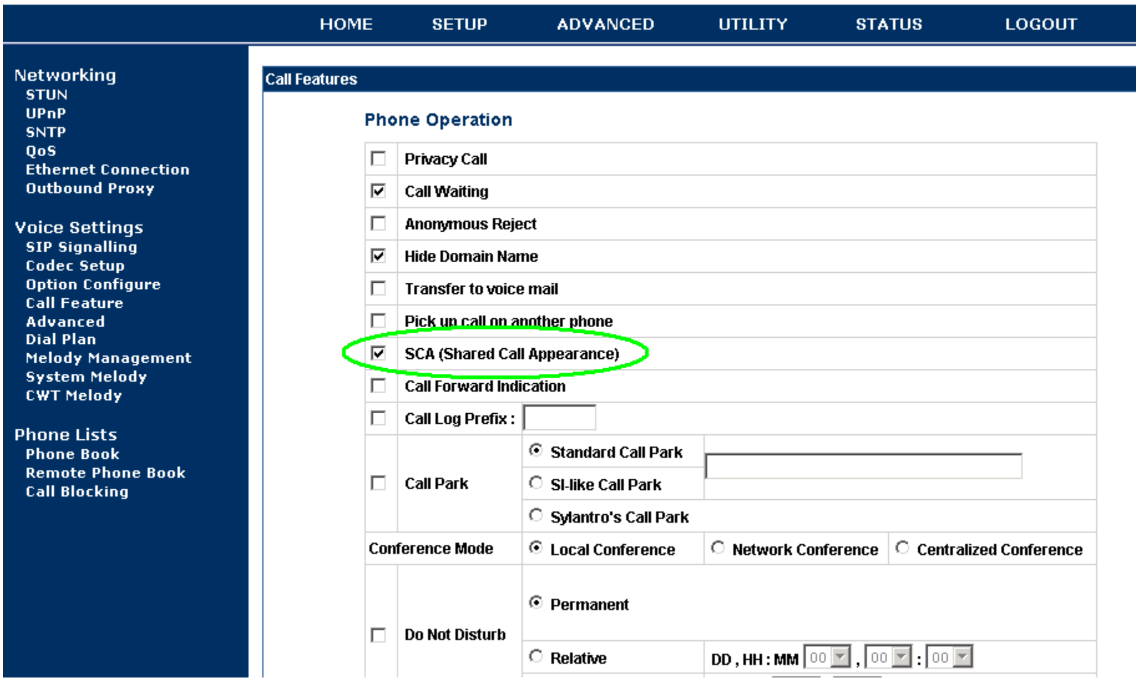
This function allows several users to share a phone number (the one in the active SIP profile), so that they can receive the same calls, make a call from the same number, unhold a previously held call, or even conference.
For incoming calls, all the phones with this service active (SCAs from now on) will ring simultaneously and the first picking up will catch the call.
For outgoing, the SCA will take the line, if available, by sending a subscription and the caller number is the same independent of which SCA is doing the call.
In order to support this, the phone establishes 2 new kind of subscriptions: event=line-seize and event=call-info. The first is for taking the (shared) line and the second to be informed about the status of this (shared) line: idle, seized, progressing, active, held, held-private, bridged-active and bridge-held.

Enriched Display

A new screen and some softkeys are available specifically for this feature.
The screen shows the status of the shared line: as soon as there is an active call in any of the phones using the same Shared Call Appearance (SCAs), it will be launched in the rest of them and refreshed when any change in the state of this call occurs.
The softkeys are:
"Retriv": retrieve selected call appearance which is in Held state.
"BargIn": barge in selected call appearance which is in Active/Bridge-Active/Brideg-Held state. After that, we’ll have a 3-way call between the initial shared call appearance, the phone which did the Barg-In and the remote party.
"NewCal": make a new call
"Back": cancel the display of call appearance(S) state and back to previous state.

Feature activation

To activate Shared Call Appearance user just have to click on the SCA flag in Advanced->Call feature menu in the Web GUI:



Call Features

Phone Operation

☐ Privacy Call
☑ Call Waiting
☐ Anonymous Reject
☑ Hide Domain Name
☐ Transfer to voice mail
☐ Pick up call on another phone
☑ SCA (Shared Call Appearance)
☐ Call Forward Indication
☐ Call Log Prefix :
☐ Call Park: ● Standard Call Park ○ SI-like Call Park ○ Sylantro's Call Park
Conference Mode: ● Local Conference ○ Network Conference ○ Centralized Conference
☐ Do Not Disturb: ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00

Or by APS in [sip] section of both Common or Specific-MAC config files with the new parameter SharedCallAppearance (0 by default, inactive; 1 to active):

```
[sip]
...
SharedCallAppearance=1
...
```

The rest of the configuration will be done by your BroadSoft accounts administrator.

ST20XX SIP New Features (SG vx.54.2)

Overview

This document describes a set of features included in both ST2030 SIP v1.54.2 and ST2022 SIP v3.54.2 in order to improve its usability in different environments.

Automatic Hang Up

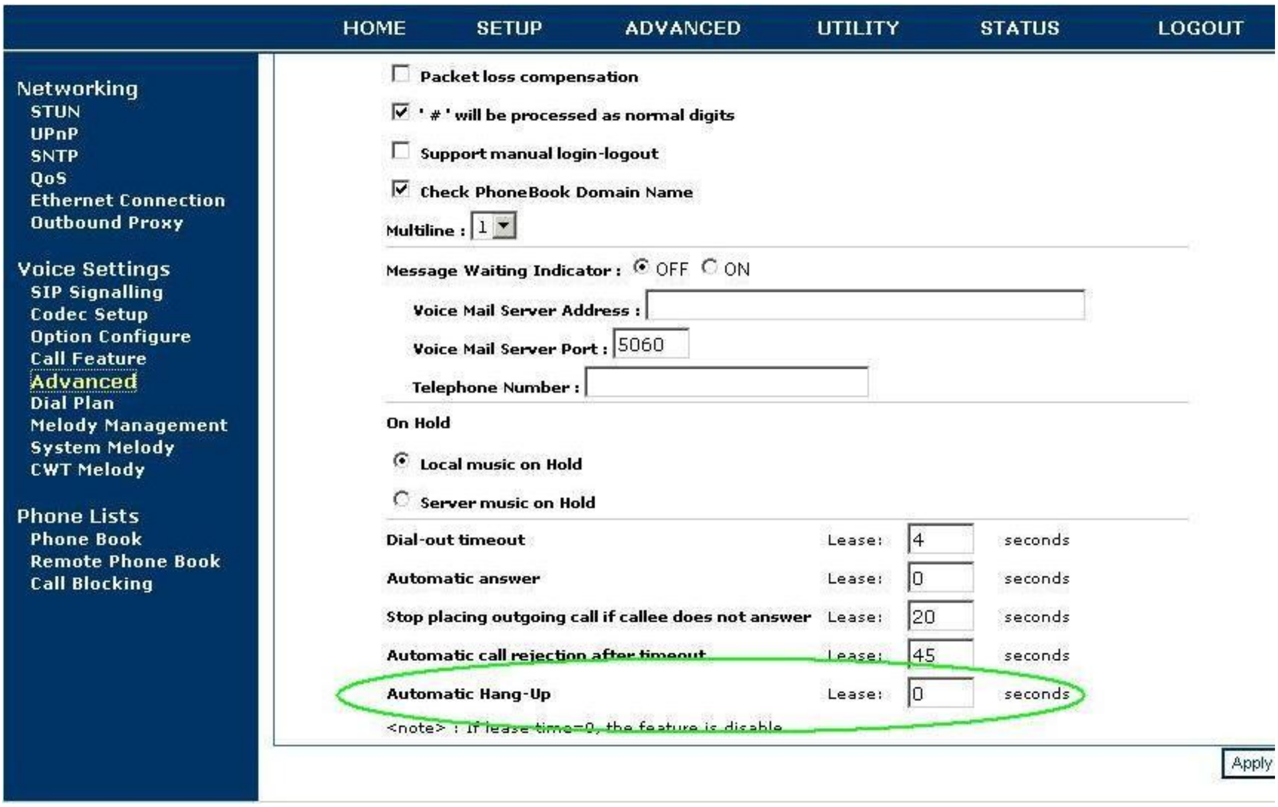
This new feature allows the phone to hang up automatically, stopping the audio output (regardless it's from speaker, headset or handset) and turning the phone into idle state. It can be configured to wait x seconds before the hang up is done. If the value of x is set to 0, phone understands that this feature is deactivated.

Obviously this doesn't enter in conflict with the signalling since the count of seconds stars after a BYE is received.

If the phone hangs-up automatically and the handset is not physically on-hook, on the next incoming call the user must on-hook first then off hook to answer the call (or press the Answer softkey).

Feature activation

It can be configured via Web GUI with a new parameter sitted on Advanced | Avanced menu:



☐ Packet loss compensation

☑ ' # ' will be processed as normal digits

☐ Support manual login-logout

☑ Check PhoneBook Domain Name

Multiline : 1

Message Waiting Indicator : ● OFF ○ ON

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
Stop placing outgoing call if callee does not answer	Lease:	20	seconds
Automatic call rejection after timeout	Lease:	45	seconds
Automatic Hang-Up	Lease:	0	seconds

<note> : If lease time=0, the feature is disable

Or by the Autoprovisioning with the parameter AutoHangUp in the [ipp] section of APS config files (both Common or specifi-MAC):

```
[ipp]
…
AutoHangUpTimer = 3
…
```

In both cases, via Web and via Autoprovisioning, the behaviour is the same. User introduces the number of seconds he wants the phone to wait (after the BYE) for hanging up; 0 to deactivate.

Call Park Type

This is not exactly a new feature but a new way of supporting traditional Call Park and Park Retrieve. From now on there are 2 modes for implementing them: The first (set by default) is the one recommended by draft-sipping-service-examples and the other one is quite similar to a blind transfer. The latter was already supported as unique Call Park/Retrieve method in ST2030SI versions.

Call Park/Retrieve Park implementation modes

Both modes requires a proper enviroment that supports Call Park.

Mode 0:
Call park is achieved by st20xx sending a REFER message to the park extension with

- Request-URI: the park extension
- Refer-to: the party which is going to be parked
- Replaces (param inside Refer-to header): dialog between parked and the one who parks.

Retrieve Park is achieved by st20xx sending a SUBSCRIBE message to the park extension and then, when park answers a NOTIFY with the dialog identifiers, it has to send an INVITE (with Replaces header) again to the previously parked party.

Mode 1:
Call park is achieved by st20xx sending a REFER message to the other party equal to the one sent if a blind transfer to the Park extension was done. Park Retrieve is achieved by st20xx sending an INVITE message to the park extension. This one will manage the rest of signalling to finalize the retrieve.

Also in both cases, park extension is configured as it used to be, by the proper option in Advanced | Call Features menu in Web GUI.

Feature activation

This feature can only be set by Autoprovisioning with the parameter Call_park_type in [sip] section of the config files (both common and specific-MAC):

```
[sip]
…
CallParkType=0 (default value. Mode 1 is activated with value 1)
…
```

APS upgrade between SIP and MGCP

This new feature allows a particular st20xx be used for both SIP and MGCP enviroments toggling application via a simple APS upgrade.

Process and Requirements

As this compatibility between SIP and MGCP st20xx phones is new, admin has to make sure that his/her phone is using approppriate versions of boot, dsp and application firmware. Also bear in mind that target version must also support this feature.

Namely:

Bootcode v1.11
DSP code 1.01
Application:
- SG1.54.2 or newer
- MC1.53 or newer
- MX1.52 or newer

These are the steps to be followed to make sure the process will work properly. You can skip those steps for which requisite is already met by your phone.

This is a generic procedure for both conversions, from SIP to MGCP and from MGCP to SIP:

1. Upgrade the boot code to v1.11
2. Upgrade the dsp code to v1.01
3. Upgrade to the SAME protocol type version supporting Protocol swap:
    - SG older version must upgrade to SG v1.54.2
    - MC older version upgrades to MC v1.53
    - MX older version upgrades to MX v1.52
4. Then, SIP ⇔ MGCP firmware change will be possible via APS
    - SG v1.54.2 or newer ⇔MC v1.53 or newer
    - SG v1.54.2 or newer ⇔ MX v1.52 or newer

Ringer in the headset. (st2030 only)

Just available in st2030. The ringer shall always be audible in the headset whatever the ringer level selected on the loudspeaker.

The volume of the ringer in the headset follows the headset volume setting.

Feature activation

This is always active; it’s a new behaviour rather than a feature. Just pressing the volume keys when headset is active will adjust the volume for the rest of the current call and for the nexts, also the ringer volume, until next change.

Ringer Off

Now it’s possible to turn off the ringer on the loudspeaker.
There are 9 steps currently and the lowest step turns off the ringer.

Feature activation

This is always active; it’s a new behaviour rather than a feature. Just pressing the volume keys when st20xx is ringing will adjust the volume for the rest of the current ringin and for the nexts, until next change.

Ignoring Firmware update

In the INF file, if the name of the Firmware starts with the letter X (lower or upper case) or is empty, the IP Phone is ignoring this parameter and keep using its old Firmware. Moreover, the IP Phone doen’t display an error message because of this.

However, if the Telconf or config files are not compatible with this old firmware, an error message will be displayed.

ST20XX SIP New Features (SGx.53)

Overview

This document describes a set of features included in both ST2030 SIP v1.53 and ST2022 SIP v3.53 in order to improve its usability in different environments.

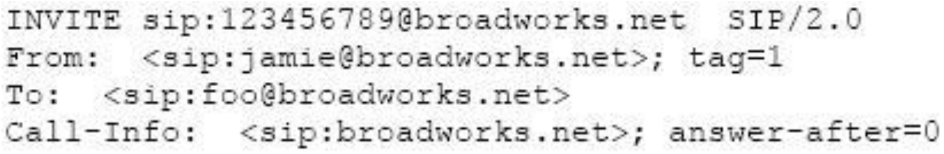
Call-info header with Answer-after parameter

This new feature allows the phone to answer a call automatically and in handsfree mode if the phone is on-hook which is comfortable for the user since he doesn't have to take any action to be able to speak.

But this already existed in previous versions as a local feature of the phone. The main difference of this new supported parameter is that the automatic answer is indicated by the server for a particular call, not by the user and for all calls .This new behaviour is used by some servers (e.g: BroadSoft) to implement Click-to-call facility where the party that appears in the call as "From" URI shouldn't ring as the "To" does.

Feature activation

It can't be activated or deactivated. This new supported feature doesn't enter in conflict with any other. It's implicitly activated when receiving and INVITE of this kind:



```
INVITE sip:123456789@broadworks.net  SIP/2.0
From:  <sip:jamie@broadworks.net>; tag=1
To:  <sip:foo@broadworks.net>
Call-Info:  <sip:broadworks.net>; answer-after=0
```

The Call-Info header field provides additional information about the caller or callee, depending on whether it is found in a request or response. To support intercom and click-to-call scenarios, we introduce a parameter called "answer-after". When present in the Call-Info header of an incoming INVITE request, it indicates how many seconds should be waited by the UAS before the call is automatically answered.

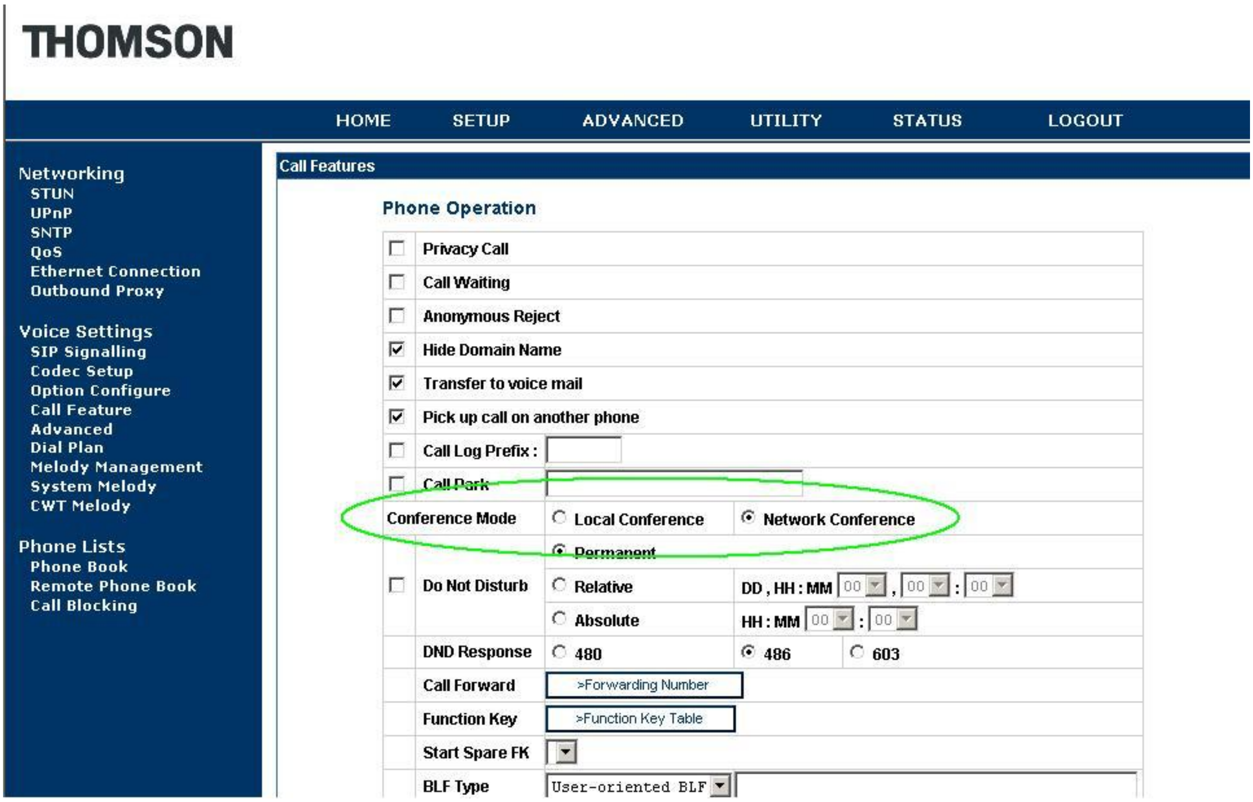
If the "answer-after" value is 0, then the call should be automatically answered without applying any alert tones. If it's 1,2,3… then phone will ring 1,2,3… seconds before the call is answered.

Network Conference

This is not exactly a new feature but it’s a new way of supporting traditional Conference (called Local Conference from now on).
Network conference allows the phone to be interoperable with servers that are able (an prefer) to carry out the conference by theirselves.

Feature activation

This feature can be configured via web with a new flag:



☐ Privacy Call
☐ Call Waiting
☐ Anonymous Reject
☑ Hide Domain Name
☑ Transfer to voice mail
☑ Pick up call on another phone
☐ Call Log Prefix :
☐ Call Park
Conference Mode ○ Local Conference ● Network Conference
☐ Do Not Disturb ● Permanent
○ Relative DD , HH : MM 00 , 00 : 00
○ Absolute HH : MM 00 : 00
DND Response ○ 480 ● 486 ○ 603
Call Forward >Forwarding Number
Function Key >Function Key Table
Start Spare FK
BLF Type User-oriented BLF

or by the APS files with a new parameter in Config Files (either common.txt or mac.txt):
ConferenceMode=0 (Local is the default mode) or 1 (for Network Conference)
Which is placed in [sip] section and set to 0 by default.

Managing Network Conference

With Network Conference activated phone is registered a bit different: REGISTER sent has to include in Allow-Events a new one called Conference:

```
…
Allow-Events:refer,dialog,message-summary,check-
sync,talk,hold,conference
```

…

After that, when initiating a Conference (by Conf or Join softkeys, same as traditional Local Conference) the phone just have to send a NOTIFY (event: conference) containing in its Refer-to header, location info of the 3rd, 4th, 5th… party that it wants to add to the conversation. Server will send the proper INVITES and BYES (or any other message) to complete the Conference.

Of course when Local Conference is the one activated, phone behaviour is the same as in previous versions.

Configurable Refer-To header population

The purpose of this feature is being able to populate Refer-To header with two different (but similar in concept) information. This is thought for Attended Transfer where refer-to header included in REFER message has to contain the URI of the target.

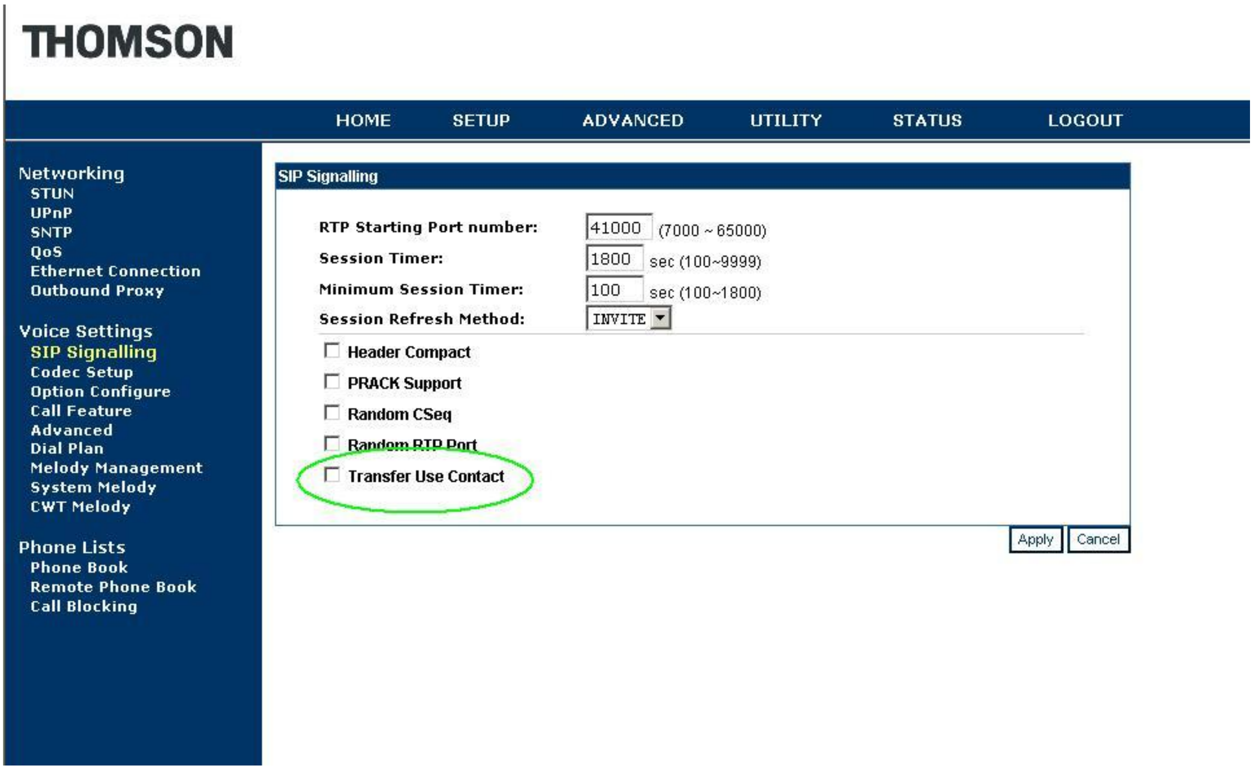
This will have to be configured in the phone which transfers (transferor) and allows it to be compatible with more servers in what is related to Attended Transference.

The two options are:

1. Populating it with the Request –URI of the messages sent to this target (e.g: INVITE sent by transferor to the target in what we call Consultative call)
2. Populating it with the Contact info of the target. This is obtained by the transferor analyzing Responses of the requests sent to the target (e.g: 200 OK asnwered in already mentioned Consultative call)

Feature activation

For option 2 (recommended by RFCs) the Flag has to be activated and for option 1 deactivated.
In this example, option 1 has been configured via web:



SIP Signalling

RTP Starting Port number: 41000 (7000 ~ 65000)
Session Timer: 1800 sec (100~9999)
Minimum Session Timer: 100 sec (100~1800)
Session Refresh Method: INVITE

☐ Header Compact
☐ PRACK Support
☐ Random CSeq
☐ Random RTP Port
☐ Transfer Use Contact

In APS config files (common and mac) this issue will be configured using the parameter TransferUseContact=1 (flag ON) or 0 (flag OFF)
Which is placed in [sip] section and set to 1 by default

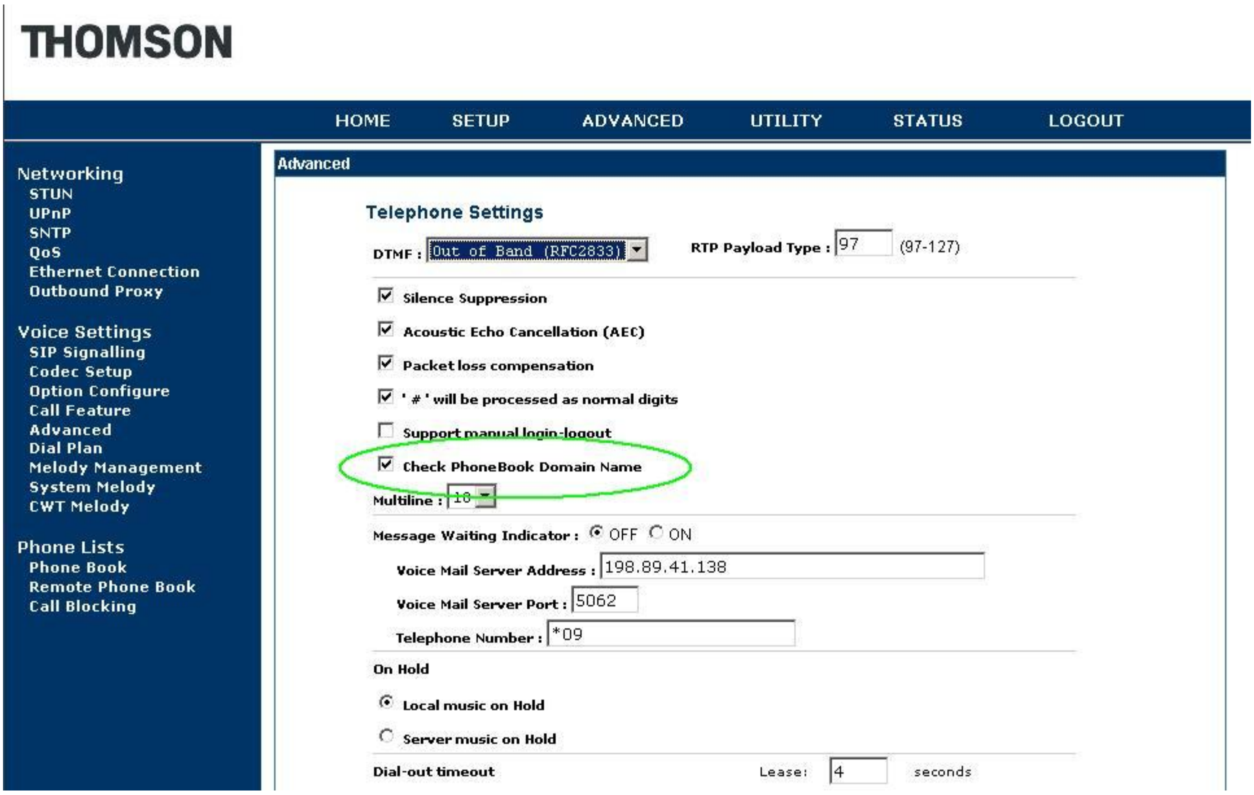
Check phonebook Domain name

For users who use to registered the phone in different accounts, with different servers or at least domain names, this can be use to get more freedom of action when storing numbers in phonebook.

There is a flag which can be activated or deactivated and basically decides if the phone will check the domain name or just the user part to recognise phonebook entries (both on incoming and outgoing calls)

Feature activation

It can be set via web:



Advanced

Telephone Settings

DTMF : Out of Band (RFC2833)
RTP Payload Type : 97 (97-127)

☑ Silence Suppression
☑ Acoustic Echo Cancellation (AEC)
☑ Packet loss compensation
☑ ' # ' will be processed as normal digits
☐ Support manual login-logout
☑ Check PhoneBook Domain Name

Multiline : 18

Message Waiting Indicator : ● OFF ○ ON
Voice Mail Server Address : 198.89.41.138
Voice Mail Server Port : 5062
Telephone Number : *09

On Hold
● Local music on Hold
○ Server music on Hold

Dial-out timeout Lease: 4 seconds

Or by APS, with a new parameter Check_phonebook_domain_name=1 (or 0 to deactivate) included in config files (common or MAC).
Exactly in [ipp] section and having 1 as default value

Phonebook entries recognition

Behaviour of the phone related to this option can be resumed in this points (points 3 and 4 are interesting but independent of this new flag):

1.- Call from/to 1234(@my_domain) will be recognised by phone as phonebook entry 1234@any_domain, showing its assigned name<=>Flag inactive

2.-Call from/to 1234(@my_domain) will JUST be recognised by phone as phonebook entry 1234@my_domain, showing its assigned name<=>Flag active

3.- Phonebook entry storaged as 1234 (no domain included) will always be recognised, showing assigned name, both in incoming calls and outgoing for any state of the flag.

4.- Also for any flag state, INVITEs to phonebook entries type name@domain will be launched with that domain and type name (no domain included) will go with current registered domain

APS improvement

TelConf file will only be downloaded if its name has changed from the previous downloaded one.

Changes in APS are deeply explained in several documents in APS&FW upgrade of this Release package.

Talk and hold event packages for click-to-answer and other 3PCC scenarios

This extension provides the ability for an Application Server to send an asynchronous NOTIFY event to our phone, using an existing INVITE dialog. This will allow an user responds hold and unhold a call from an application in a PC (e.g: Broadsoft Call Manager) without touching any key in the phone.

When a UAC sends an INVITE to a UAS, it adds an Allow-Events header to the request, indicating all of the event packages it supports. When a UAS responds to the INVITE with an 18x provisional response or a 200 OK response, it adds an Allow-Events header indicating all of the event packages it supports.

Talk event

The Allow-Events header in the 180 Ringing indicates to the Application Server that our phone supports remote call control primitives. When the user (e.g: handling a PC) selects the incoming call and clicks on “talk” – an event is sent from the call client to the Application Server, indicating that the user is requesting that the incoming call be answered. The Application Server reacts by sending a NOTIFY to the SIP phone.

The IP phone indicates that it honors the request by responding to the NOTIFY with a 200 OK. The phone then automatically answers the incoming call by forcing off-hook and activating the speaker.

Hold event

The call is set up as usual. Now the user at the call client decides to remotely hold the call and sends a “hold” request to the Application Server. The Application Server discovered that the endpoint supported the hold event package through an Allow-Events header in the 180 Ringing provisional response.

The Application Server reacts by sending a NOTIFY with a hold event. Note that this NOTIFY is sent using the same dialog as the session it is acting on.

Persistent VLAN

This is an improvement in VLAN configuration via DHCP. This is deeply explained in VLAN provisioning via DHCP.pdf placed in the same folder as this.

ST2030 SIP New Features (SG1.52.1)

Overview

This document describes a set of features included in ST2030 SIP v1.52.1 in order to improve its usability in different environments.

List-oriented BLF

The purpose of this feature is to supervise other phones as is the case of standard BLF (from now on, referred to as User-Based BLF).
This feature is taylored to Broadworks environments.

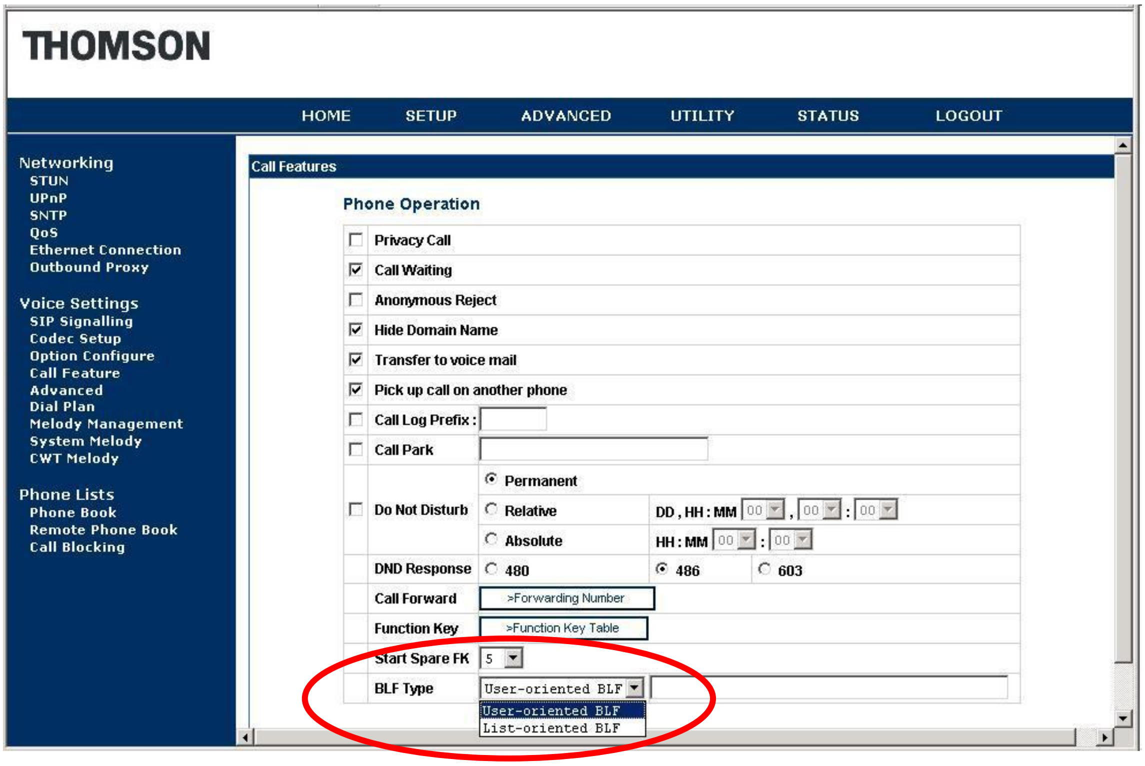
The main difference from administrator point of view is the dynamic provisioning via NOTIFY of all the supervised parties contained in an URI-List-versus static provisioning for User-Based BLF

Feature activation

In order to have List-oriented BLF active, APS or web gui can be used.
Please note default mode for BLF is User-Based.

A. <u>Via Web GUI:</u>

First you have to decrease Multiline, (or use an extension module) in Advanced→Advanced
Then go to Advanced→Call feature

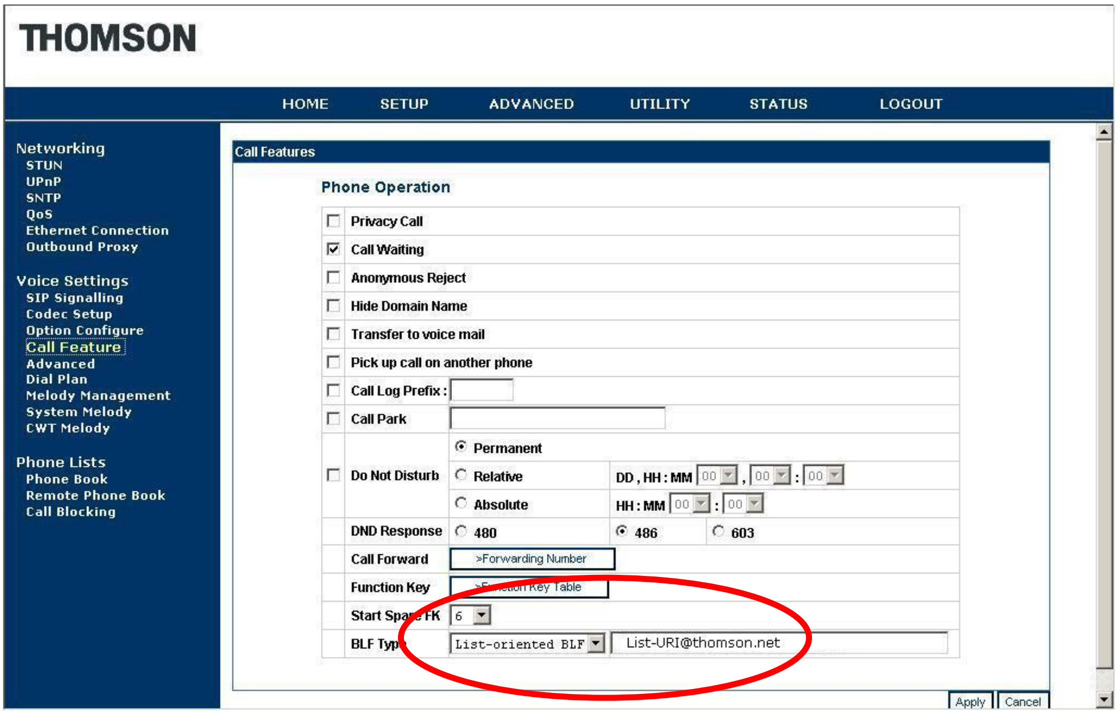


Phone Operation

☐ Privacy Call
☑ Call Waiting
☐ Anonymous Reject
☑ Hide Domain Name
☑ Transfer to voice mail
☑ Pick up call on another phone
☐ Call Log Prefix :
☐ Call Park
☐ Do Not Disturb: ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00
DND Response: ○ 480 ● 486 ○ 603
Call Forward: >Forwarding Number
Function Key: >Function Key Table
Start Spare FK: 5
BLF Type: User-oriented BLF
User-oriented BLF
List-oriented BLF

Select "List-oriented BLF", and configure the List-uri to which the phone should subscribe, in accordance with your server configuration.

Parameter "Start Spare Fk" indicates the first function key which will be dynamically provisioned. This parameter, by default, is automatically set to Multiline+1. If you want to keep some keys reserved for speeddial, please change this value.

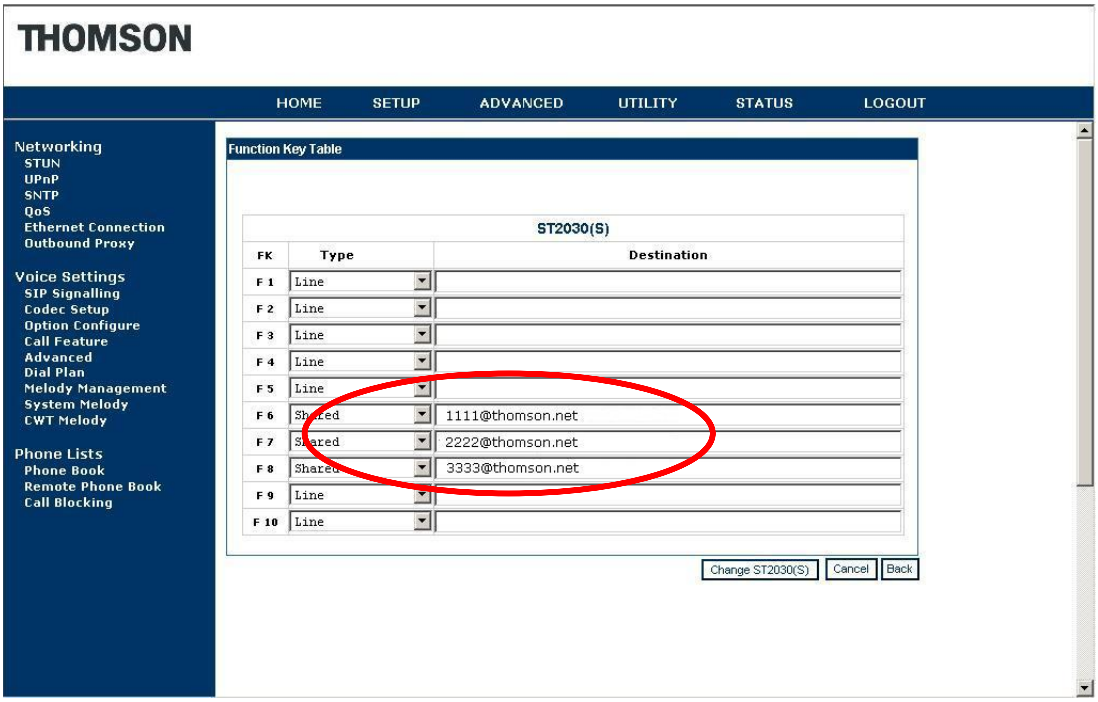


Call Features

Phone Operation

☐ Privacy Call
☑ Call Waiting
☐ Anonymous Reject
☐ Hide Domain Name
☐ Transfer to voice mail
☐ Pick up call on another phone
☐ Call Log Prefix :
☐ Call Park
☐ Do Not Disturb: ● Permanent; ○ Relative DD , HH : MM 00 , 00 : 00; ○ Absolute HH : MM 00 : 00
DND Response: ○ 480 ● 486 ○ 603
Call Forward: >Forwarding Number
Function Key: >Function Key Table
Start Spare FK: 6
BLF Type: List-oriented BLF List-URI@thomson.net

When you press Apply and reboot the phone, it will subscribe to the list and fill in automatically the function key table.



Function Key Table

ST2030(S)

FK	Type	Destination
F 1	Line	
F 2	Line	
F 3	Line	
F 4	Line	
F 5	Line	
F 6	Shared	1111@thomson.net
F 7	Shared	2222@thomson.net
F 8	Shared	3333@thomson.net
F 9	Line	
F 10	Line	

B. ***For APS***, parameters are defined in the next paragraphs

Keyword definition and usage

The parameters governing this feature, both included in the [sys] section, are:

BLFType

Value : 0 or 1
Meaning :
  0 : User-oriented BLF is active
  1 : List-oriented BLF is active
Default value: 0

BLFListSipUri
Value: L/<sip:user@host>
Meaning: contains the URI of the List the phone will need to subscribe in order to monitor users. This URI has to be provided by your sip server administrator.

StartSpareFK
Value : numeric, from Current_Max_Multiline+1 to the number of function keys available
Meaning :
  n : Fn will be the first position to be automatically filled in via List-oriented BLF dynamic provisioning.
Default value: Current_Max_Multiline+1

Current_Max_Multiline
Value: 1 to 10
Meaning: number of simultaneous calls the phone will handle. BLF is not possible with keys assigned to multiline, that is why this parameter needs to be adjusted in order to use BLF. If you intend to use an extension module for BLF, then you need not change this parameter.

Example:

```
[sys]
BLFType=1
BLFListSipUri=L/<sip:List-URI@thomson.net>
Current_Max_Multiline=4
StartSpareFK=6
```

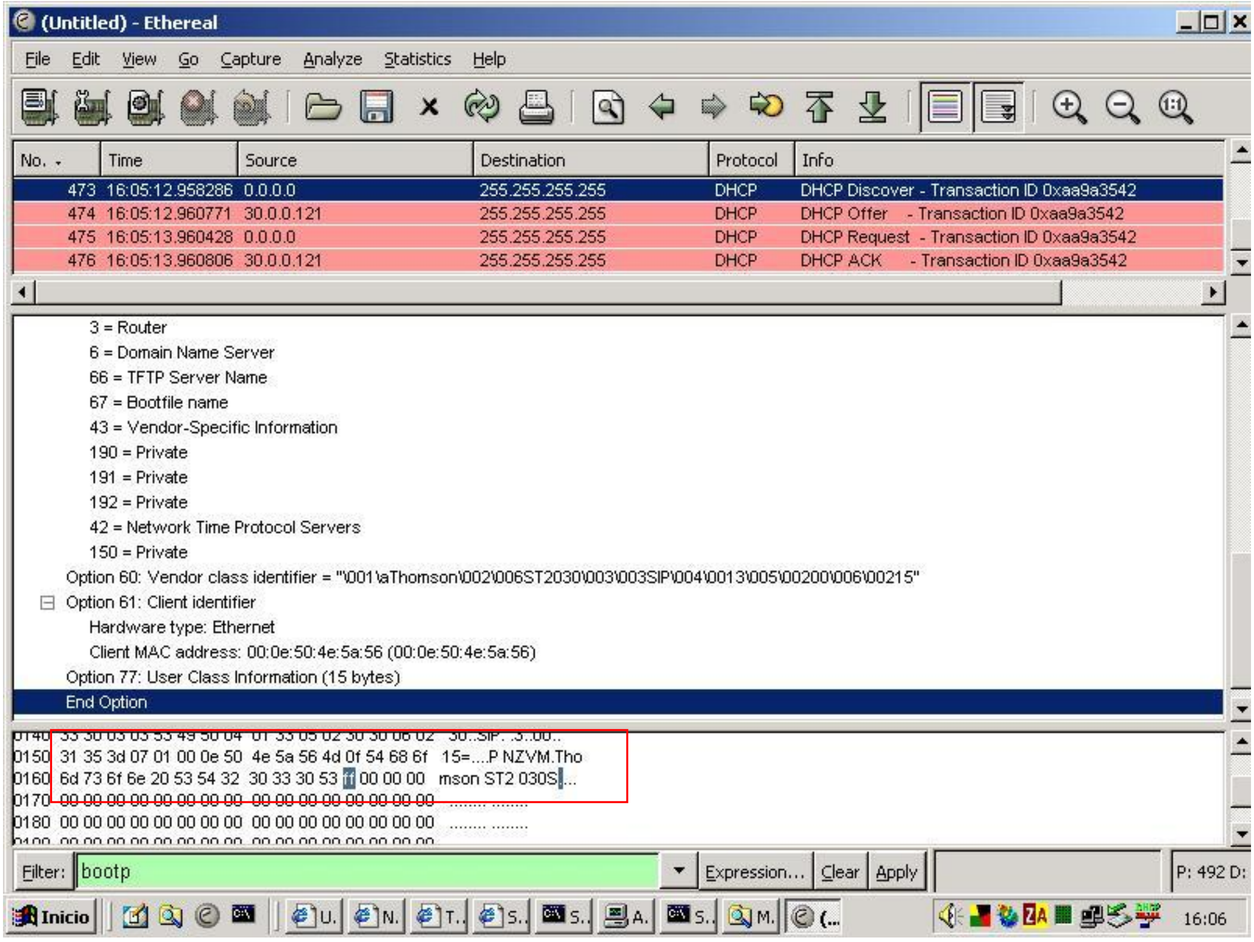
With this parameters, List-oriented BLF is enabled, the List Uri is List-URI@thomson.net and the first key to be provisioned is F6

User Class Identifier

User Class identifier is a DHCP option (Opt 77) which allows DHCP servers to serve different devices differently.
In particular, this option is useful for Provisioning scenarios with more than one model using a similar strategy for provisioning parameter transport using options 66/67/150.

ST2030 will send option 77 in all DHCP DISCOVER and DHCP REQUEST messages.

The content delivered by ST2030 within this option depends on the protocol loaded.
Hence it will be "Thomson ST2030S" or "Thomson ST2030M"
As an example, this is what would be seen with a packet sniffer with an SG version



No.	Time	Source	Destination	Protocol	Info
473	16:05:12.958286	0.0.0.0	255.255.255.255	DHCP	DHCP Discover - Transaction ID 0xaa9a3542
474	16:05:12.960771	30.0.0.121	255.255.255.255	DHCP	DHCP Offer - Transaction ID 0xaa9a3542
475	16:05:13.960428	0.0.0.0	255.255.255.255	DHCP	DHCP Request - Transaction ID 0xaa9a3542
476	16:05:13.960806	30.0.0.121	255.255.255.255	DHCP	DHCP ACK - Transaction ID 0xaa9a3542

```
        3 = Router
        6 = Domain Name Server
        66 = TFTP Server Name
        67 = Bootfile name
        43 = Vendor-Specific Information
        190 = Private
        191 = Private
        192 = Private
        42 = Network Time Protocol Servers
        150 = Private
    Option 60: Vendor class identifier = "\001\aThomson\002\006ST2030\003\003SIP\004\0013\005\00200\006\00215"
⊟   Option 61: Client identifier
        Hardware type: Ethernet
        Client MAC address: 00:0e:50:4e:5a:56 (00:0e:50:4e:5a:56)
    Option 77: User Class Information (15 bytes)
    End Option
```

```
0150  31 35 3d 07 01 00 0e 50  4e 5a 56 4d 0f 54 68 6f   15=....P NZVM.Tho
0160  6d 73 6f 6e 20 53 54 32  30 33 30 53 ff 00 00 00   mson ST2 030S....
0170  00 00 00 00 00 00 00 00  00 00 00 00 00 00 00 00   ........ ........
0180  00 00 00 00 00 00 00 00  00 00 00 00 00 00 00 00   ........ ........
```

Filter: bootp

Feature activation

This feature is always active and does not need any configuration.

Additional Softkey Control

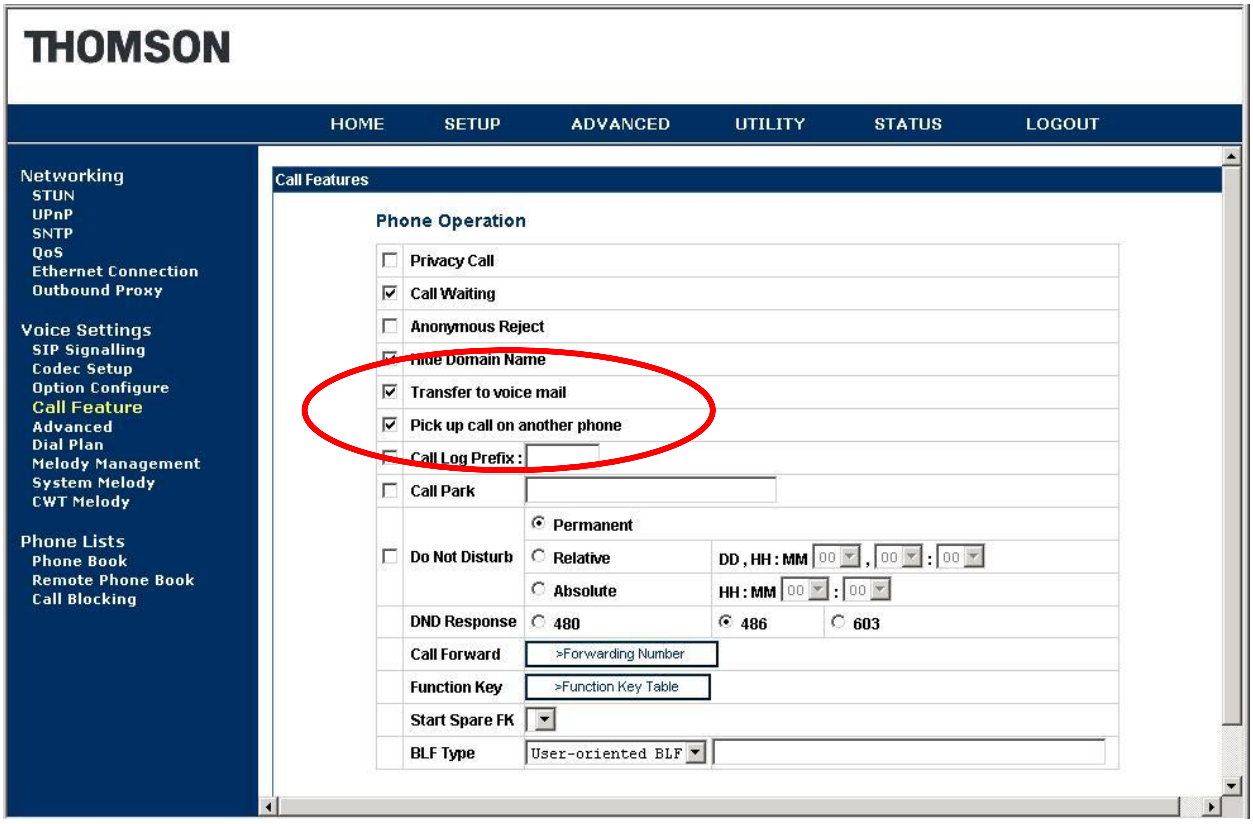
The purpose of this feature is being able to deactivate/activate "Transfer to voice mail" and "Pick up" soft keys, to accomodate to environments in which these features are not supported.

Feature activation

Web GUI and APS can be used to control these features.

In the admin Web GUI, go to Advanced→ Call Feature page, 2 new entries can be found:

"Transfer to voice mail"	corresponding to "TrMail" soft key
"Pick up call on another phone"	corresponding to "PickUp" soft key



☐ Privacy Call
☑ Call Waiting
☐ Anonymous Reject
☑ Hide Domain Name
☑ Transfer to voice mail
☑ Pick up call on another phone
☑ Call Log Prefix :
☐ Call Park
☐ Do Not Disturb: ● Permanent ○ Relative DD , HH : MM 00 , 00 : 00 ○ Absolute HH : MM 00 : 00
DND Response: ○ 480 ● 486 ○ 603
Call Forward
Function Key
Start Spare FK
BLF Type: User-oriented BLF

You can tick/untick the options to enable/disable the features. They are enabled by default

As for APS, related parameters are described in next paragraph.

Keyword definition and usage

The parameters governing this feature, included in the [ipp] section, are:

Transfer_to_voice_mail

Values : 0 or 1
Meaning :
0 : Transfer to Voicemail softkey will not appear on phone screen
1 : Transfer to Voicemail softkey will appear on phone screen
Default: default value is “1”

Pick_up_call

Values : 0 or 1
Meaning :
0 : Call Pickup function is disabled. Softkey will not appear on phone screen
1 : Call Pickup function is enabled. Softkey will appear on phone screen
Default: default value is “1”

Example:

```
[ipp]
Transfer_to_voice_mail=1
Pick_up_call=0
```

Call Progress Indication Control

For environments in which early media is provided, e.g. to convey network progress tones, the status information shown on the display of the phone may in some cases result confusing.

For example, when you receive a busy tone via early media, and the phone shows "Ringing", according to its call progress status.

To overcome this situation, a control flag is provided to administrators. So, they may decide whether to show call progress indication in this early state.

Feature activation

APS is currently used to control this setting.
Involved APSarameter is described in next paragraph.

Keyword definition and usage

The parameter governing this feature, included in the [ipp] section, is:

Disable_call_progress
Values : 0 or 1
Meaning :
    0 : Progress indication in early state is shown
    1 : Progress indication in earlyh state is not shown
Default : 0 (Progress is indicated)

Example :

```
[ipp]
Disable_call_progress=1
```

Phone Number Display in Idle Mode Control

ST2030 idle screen shows both display name and user part of the sip uri in the active account (phone number).

In some systems, however, the phone number has some additional characters like MAC address, for example, which makes the screen look unfriendly to users.

An option has been implemented to be able to hide phone number in idle screen.

Feature activation

APS is currently used to control this setting.
Involved APSarameter is described in next paragraph.

Keyword definition and usage

The parameter governing this feature, included in the [ipp] section, is:

Hide_Phone_Number_Display

Values : 0, 1
Meaning :
  0 : the phone displays its phone number on the idle screen
  1 :: the phone does not display its phone number on the idle screen
Default : 0 (number is displayed)

Example :

```
[ipp]
Hide_Phone_Number_Display=1
```

ST2030 SIP New Features (SEG1.50t3)

Overview

This document describes a set of features included in ST2030 SIP in order to improve its usability in different environments.

Automatic call when offhook

The purpose of this feature is to allow the user to call a number that is pre-registered in the phone. The call will be placed as it is unhooked (when you use handset, headset or handsfree). This behaviour will remain the same as long as it is not disabled by APS.

This functionality can be used to emit emergency calls. In this case, the user is directly directed towards an urgent number when he pickups the phone.

For example, this can be used in an elevator which is blocked.

Feature activation

APS is currently used to activate/deactivate this feature, via the parameters described in next paragraph.

Keyword definition and usage

The parameters governing this feature, both included in the [ipp] section, are:

Autocall
Values : 0 or 1
Meaning :
- 0 : Autocall feature is disabled
- 1 : Autocall feature is enabled

AutocallNumber
Value : character string
Meaning : the string represents the phone number or URI to be called

Example:

```
 [ipp]
Autocall=1
AutocallNumber=0805
```

Hide Missed Calls flag

The purpose of this feature is being able to deactivate/activate Missed Calls message display on the phone screeen.

This will have mainly interest within the framework of call centres in which calls are sent to all Agents and only picked up by one of them, whereas it is not desirable to have all other phones of the group announcing a missed call.

Feature activation

APS is currently used to activate/deactivate this feature, via the parameter described in next paragraph.

Keyword definition and usage

The parameter governing this feature, included in the [ipp] section, is:

HideMissedCall
Values : 0 or 1
Meaning :
0 : Missed Calls message will be displayed
1 : Missed Calls message will not be displayed

Example :

```
[ipp]
HideMissedCall=1
```

Autoanswer device routing

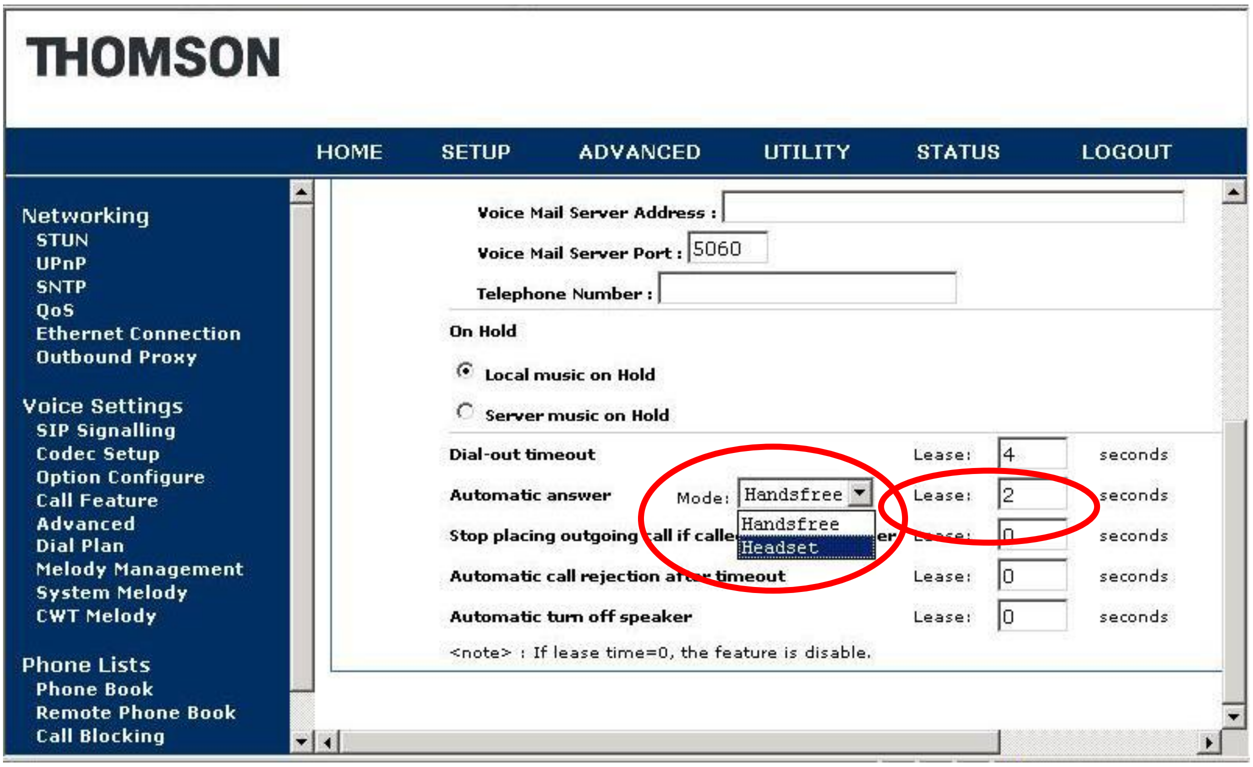
The purpose of this feature is being able to select which audio devices will be activated when Autoanswer feature is enabled.

In previous versions, default device was handsfree speaker/mic. In Call Centre environments it makes more sense to use headset for this purpose.

Feature activation

Web GUI or APS are currently used to control this feature. Please note the Autoanswer timer must be different from 0 for Autoanswer to be activated.

In the web gui, go to Advanced→Advanced section:



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 Mode: Handsfree Lease: 2 seconds

Stop placing outgoing call if calle er Lease: 0 seconds

Automatic call rejection af r timeout Lease: 0 seconds

Automatic turn off speaker Lease: 0 seconds

<note> : If lease time=0, the feature is disable.

Involved APS parameters are described in next paragraph.

Keyword definition and usage

The parameter governing this feature, included in the [ipp] section, is:

AutoAnsMode
Values : 0 or 1
Meaning :
0 : Autoanswered call routed to handsfree
1 : Autoanswered call routed to headset

Example:

```
[ipp]
AutoAnsMode=1
AutoAnsTimer=2
```

Daylight saving refinements

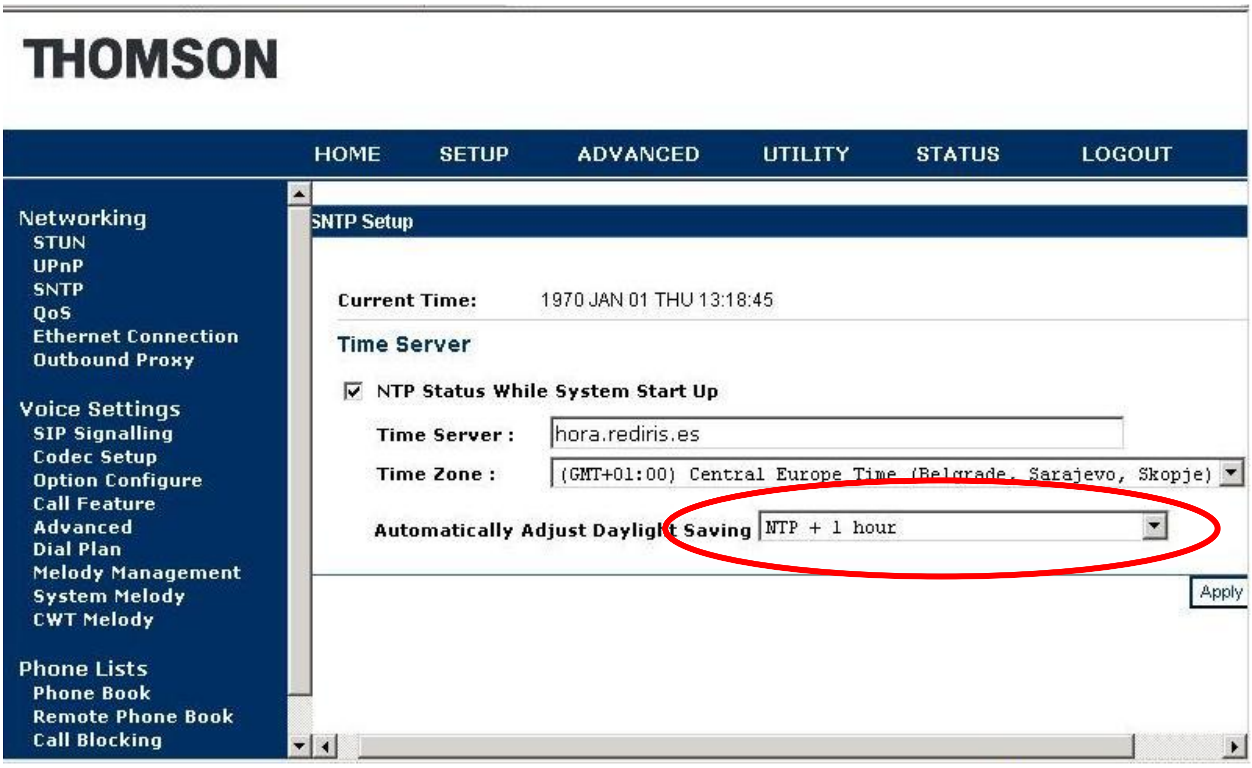
In previous ST2030 versions, parameter NtpDaylight exists in order to allow the telephone to automatically switch to summer-time.

But some states in some countries have a particular and varying switching time which makes it difficult to handle internally on a global basis.

In order to overcome this problem, a new value for the existing parameter is created. When this setting is in place, phone will add one hour to NTP received time.

Feature activation

Web GUI or APS are currently used to control this setting.
In the Web GUI, go to Advanced→SNTP section:



Current Time: 1970 JAN 01 THU 13:18:45

☑ NTP Status While System Start Up

Time Server : hora.rediris.es

Time Zone : (GMT+01:00) Central Europe Time (Belgrade, Sarajevo, Skopje)

Automatically Adjust Daylight Saving NTP + 1 hour

Involved APSarameters are described in next paragraph.

Keyword definition and usage

The parameter governing this feature, included in the [ntp] section, is:

NtpDaylight
Values : 0, 1 or 2
Meaning :
- 0 : NTP time unchanged
- 1 : Automatic Daylight saving applied to NTP time
- 2 : NTP +1

Example :

```
[ntp]
NTPFlag=1
NtpIP=192.43.244.18
NtpDaylight=2
NtpZoneNum=50
NtpSyncTime=1
```

NTP address source configuration

In previous ST2030 versions, NTP server address was either taken from the DHCP server, or from stored values if DHCP server was not serving this parameter.

This feature allows the administrator to decide which will be the source of NTP address.

Feature activation

APS is currently used to control this setting.

Keyword definition and usage

The parameter governing this feature, included in the [ntp] section, is:

NtpMode
Values : 0, 1
Meaning :
- 0 : NTP address retrieved from stored value
- 1 : NTP address retrieved preferably from DHCP server

Example :

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

Information shortcut-“Menu” long press

The purpose of this feature is to facilitate Support tasks, by granting fast access to basic phone information: HW and FW version, MAC and IP address, etc

This information was already provided by pressing User→Information
Now, access to this submenu is also available by long pressing “Menu” key

ST2030 SIP Monitoring Extension Feature (BLF) v1.47

1 SUBSCRIBE/NOTIFY support for monitoring extension states- Overview

The objective of this document is to present the way ST2030 SIP uses backlit function keys to monitor the state of other extensions. The backlit function keys will indicate if an extension is idle, in use, or ringing. This feature is known as BLF or Monitoring Extension function.

ST2030 extension module can be used to have more programmable keys on board. There are basically two scenarios:

- **Server-to-phone scenarios**: there is a SIP proxy which controls the state of each extension and sends the appropriate messages to the supervisor phone. Asterisk is an example and the reference followed for implementation.

- **Phone-to-phone scenarios**: each phone is responsible for notifying its own state to supervisor phones

2 Functionality

Mechanism used for BLF is SUBSCRIBE/NOTIFY messages described in RFC-3265.

The device will subscribe to the state of the extension of interest and receive status notifications from this extension or from the proxy in order to drive the LEDs.

2.1 High level technical description

When the phone is initialized, it will send out a SUBSCRIBE message to the proxy for each extension it would like to monitor. This is to subscribe to the state of the other extension. The proxy (and in some cases the other extension) needs to support these subscriptions. At this point 2 things can happen depending on the Proxy, which correspond to the scenarios listed above:

• **Phone-to-phone scenarios** : the proxy will forward the SUBSCRIBE to the far phone. Then the phone needs to log all extensions which subscribed, and send out NOTIFY messages when changes happens in its state.

• **Server-to-phone scenarios** : the proxy captures the SUBSCRIBE, does not forward it and sends state changes as the body of a NOTIFY on behalf of the phone. Asterisk Server does this.

The "<state>early|confirmed|void</state>" in the received NOTIFY msgs determines the LEDs activity.

- If the line on the extension rings, the programmable key LED on the phone blinks. Related state is early.

- If the line on the extension is busy, the programmable key LED on the phone is steadily lit. Related state is confirmed.

- If the programmable key LED on the phone is off, then the programmable key can act as a speed dial key for the same supervised destination number . Related state is void (state not provided), or terminated.

Other tags could be supported in future releases.

2.2 User interaction

2.2.1 In idle mode

Pressing a key whose LED is **off** (phone in **idle** mode, state void) performs a call to the corresponding phone.

Pressing a key whose LED is **on** (phone **busy**, state confirmed) performs a call to the corresponding phone, a normal call.

Pressing a key whose LED is **blinking** (**phone ringing**, state early) performs a call pickup of the corresponding phone.

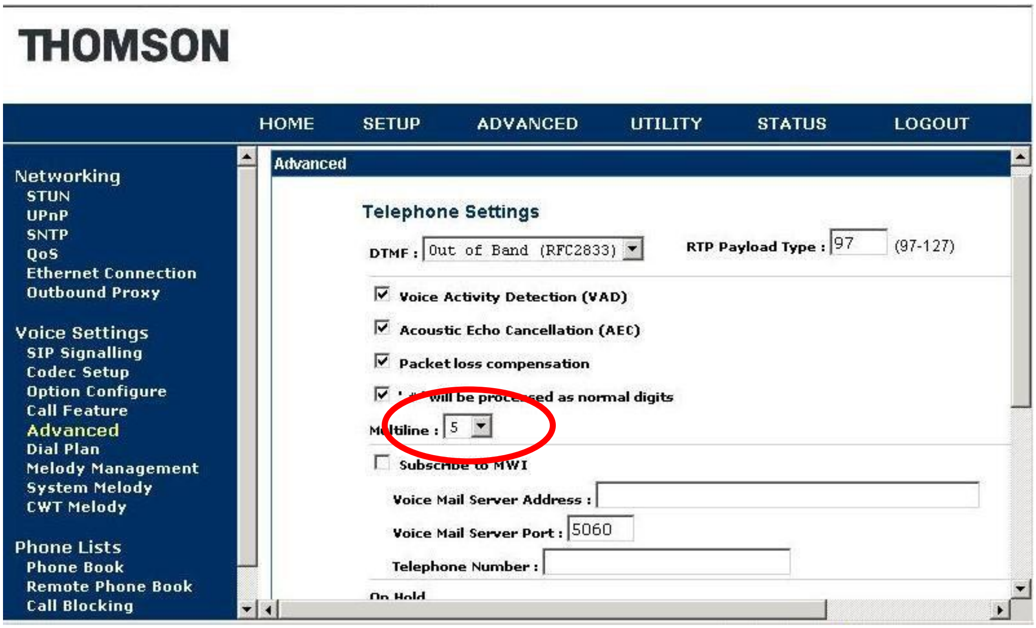
2.2.2 During a call or conference

Same behaviour as in idle mode except it is in a second call (or third…). This would be equivalent to the user pressing the softkey "New Call" and placed a call to that extension.

3. BLF Configuration

3.1 Web GUI

The setup of the function keys can be done using the ST2030 web interface. First, if you are going to use any of the function keys in the phone instead of an extension module, you need to reconfigure the max number of lines in Advanced→ Advanced. In this example it has been set to 5, which leaves room for 5 supervision lines:



Telephone Settings

DTMF : Out of Band (RFC2833) RTP Payload Type : 97 (97-127)

☑ Voice Activity Detection (VAD)

☑ Acoustic Echo Cancellation (AEC)

☑ Packet loss compensation

☑ will be processed as normal digits

Multiline : 5

☐ Subscribe to MWI

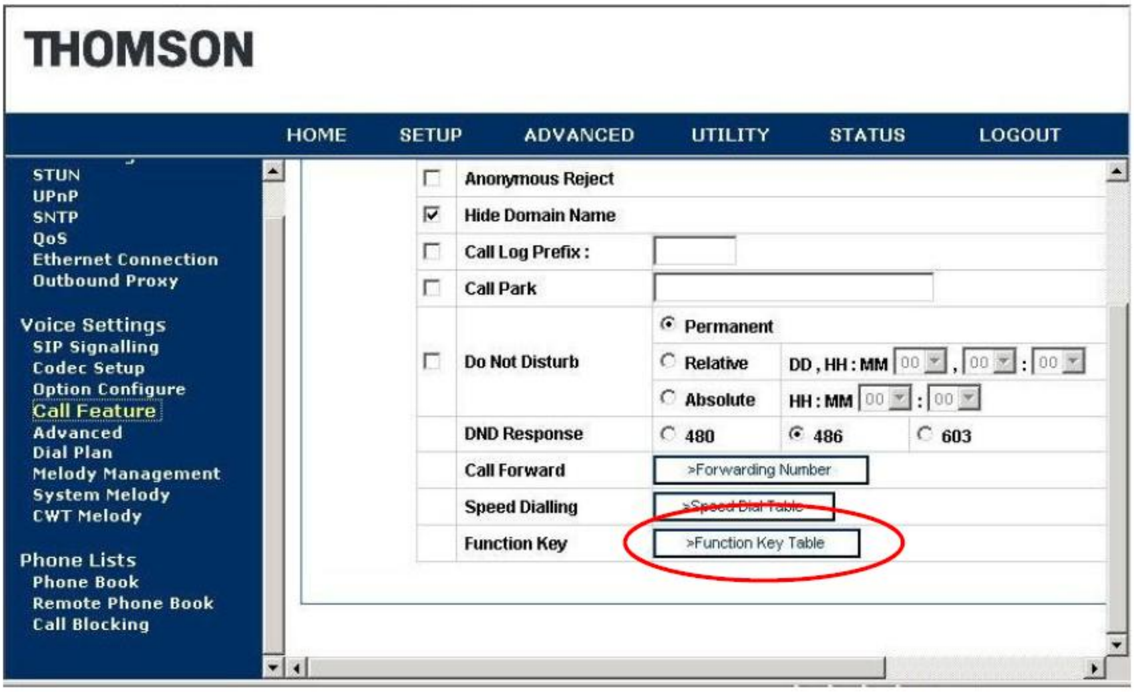
Voice Mail Server Address :

Voice Mail Server Port : 5060

Telephone Number :

On Hold

To configure the function keys, visit Advanced → Call features and click on Function keys:



☐ 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

DND Response ○ 480 ● 486 ○ 603

Call Forward

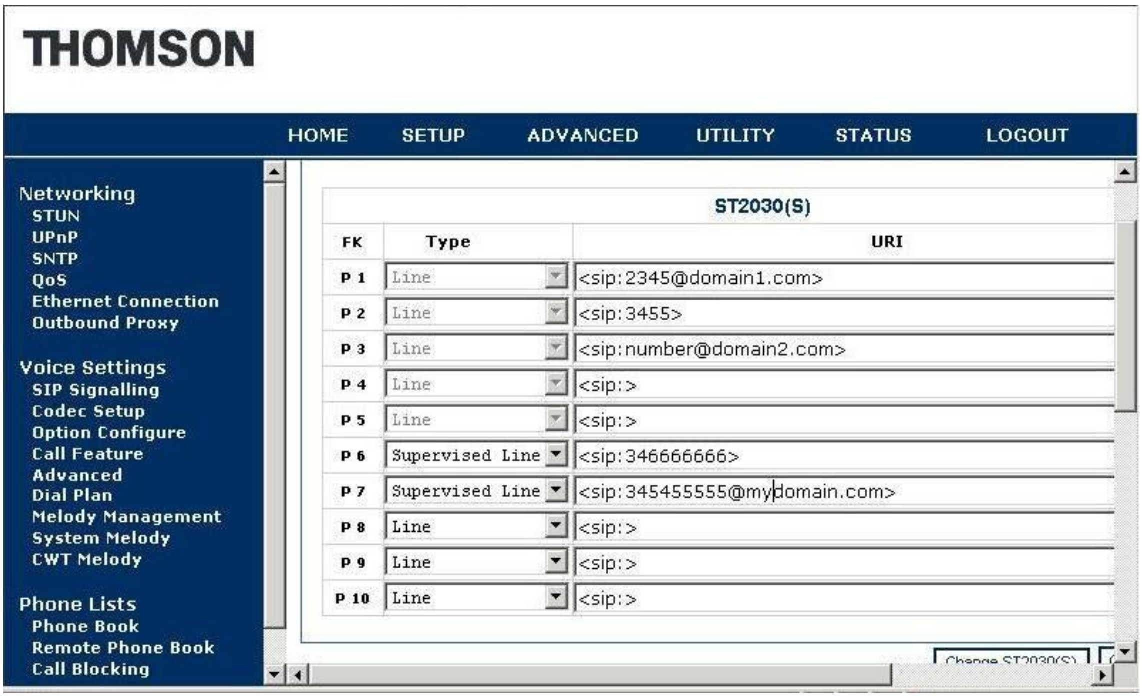
Speed Dialling

Function Key

The feature keys are split in 3 pages:
F1 to F10
F11 to F38 (1st extension module, will only appear if the module is detected)
F39 to F66 (2nd extension module, will only appear if the module is detected)

On each page, the feature keys are detailed in a list.
For each line the administrator is able to setup:

- The line type:
  Line (normal speedial)
  or
  Supervised Line (Supervised behavior described above)
- The corresponding extension



ST2030(S)

FK	Type	URI
P 1	Line	<sip:2345@domain1.com>
P 2	Line	<sip:3455>
P 3	Line	<sip:number@domain2.com>
P 4	Line	<sip:>
P 5	Line	<sip:>
P 6	Supervised Line	<sip:346666666>
P 7	Supervised Line	<sip:345455555@mydomain.com>
P 8	Line	<sip:>
P 9	Line	<sip:>
P 10	Line	<sip:>

3.2 Provisioning files (APS)

Feature keys are configurable using configuration files, either common or MAC specific, and syntax is the same for both. These parameters, as can be checked in ST2030S_Config file syntax document and sample files,
will be included in [sys] section.

The parameters will follow this pattern:
FeatureKeyExtXX=Y/<sip:2006@10.0.0.5>
With: 01 <= XX <= 66
With: Y = [L:S]
L= Line (no supervision)
S= Supervised Line
Ex:

```
FeatureKeyExt01=L/<sip:2345@domain1.com>
FeatureKeyExt02=L/<sip:3455>
FeatureKeyExt03=L/<sip:number@domain2.com>
FeatureKeyExt04=L/<sip:>
FeatureKeyExt05=L/<sip:>
FeatureKeyExt06=S/<sip:346666666>
FeatureKeyExt07=S/<sip:345455555@mydomain.com>
```

4 Examples with Asterisk SIP server as notifier

```
------------------------------------------------------------------------------
Sent to udp:10.0.0.5:5060 at 19/10/2005 12:26:20:770 (425 bytes):
#With this message 2200@lan subscribes to the state of 2205. The message
will be forwarded to 2205 or not depending on Proxy capabilities
SUBSCRIBE sip:2205@lan.net;user=phone SIP/2.0
Via: SIP/2.0/UDP 10.0.0.1:2051;branch=z9hG4bK-wkwhbwu19wgc;rport
From: <sip:2200@lan.net>tag=gumvap0ha0
To: <sip:2205@lan.net;user=phone>
Call-ID: 3c2675eaad57-2fjnztudzq25@ST2030
CSeq: 1 SUBSCRIBE
Max-Forwards: 70
Contact: <sip:2200@10.0.0.1:2051;line=xjqldyhz>
Event: dialog
Accept: application/dialog-info+xml
Expires: 3600
Content-Length: 0
------------------------------------------------------------------------------
The UA 2200@lan should receive the 200 OK to the SUBSCRIBE above ....
-------------------------------------------------------------------------------


------------------------------------------------------------------------------
Received from udp:10.0.0.5:5060 at 19/10/2005 12:35:57:280 (580 bytes):
#This turns the LED corresponding to 2205 OFF at 2200@lan
NOTIFY sip:2200@10.0.0.1:2051;line=xjqldyhz SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-
44e587400f6d8ef6a8d98ddeecf6edca
From: <sip:2205@lan.net;user=phone>tag=6sam28oefu
To: <sip:2200@lan.net>tag=gumvap0ha0
Call-ID: 3c2675eaad57-2fjnztudzq25@ST2030
CSeq: 18 NOTIFY
Max-Forwards: 70
Event: dialog
Subscription-State: active
Content-Type: application/dialog-info+xml
Content-Length: 150
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="18"
state="full"
```

```
entity="sip:2205@lan.net"></dialog-info>
------------------------------------------------------------------------------
.... The UA 2200@lan should send the 200 OK to the NOTIFY above ....
------------------------------------------------------------------------------

------------------------------------------------------------------------------
Received from udp:10.0.0.5:5060 at 19/10/2005 12:35:57:430 (930 bytes):
#This make the led FLASH
NOTIFY sip:2200@10.0.0.1:2051;line=xjqldyhz SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-
a566a0acd02fdaf578d455bc48614953
Page 6 de 9
From: <sip:2205@lan.net;user=phone>tag=6sam28oefu
To: <sip:2200@lan.net>tag=gumvap0ha0
Call-ID: 3c2675eaad57-2fjnztudzq25@ST2030
CSeq: 19 NOTIFY
Max-Forwards: 70
Event: dialog
Subscription-State: active
Content-Type: application/dialog-info+xml
Content-Length: 500
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="19"
state="full" + entity="sip:2205@lan.net"><dialog id="dummy" call-
id="feb21493- d84a9bd6@192.168.0.113" +
local-tag="6yln5cag4d" remote-tag="fe9c856f35980aao0" +
direction="recipient"><state>early</state> +
<local><identity>sip:2205@lan.net</identity><target +
uri="sip:2205@lan.net"/></local><remote> +
<identity>sip:22052@lan.net</identity><target +
uri="sip:22052@lan.net"/></remote></dialog></dialog-info>
------------------------------------------------------------------------------
.... The UA 2200@lan should send the 200 OK to the NOTIFY above ....
-------------------------------------------------------------------------------

------------------------------------------------------------------------------
Received from udp:10.0.0.5:5060 at 19/10/2005 12:36:00:220 (934 bytes):
#This makes the led STEADY ON
NOTIFY sip:2200@10.0.0.1:2051;line=xjqldyhz SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-
a718cece01b3c69c1d666b3a7a03c366
From: <sip:2205@lan.net;user=phone>tag=6sam28oefu
To: <sip:2200@lan.net>tag=gumvap0ha0
Call-ID: 3c2675eaad57-2fjnztudzq25@ST2030
CSeq: 20 NOTIFY
Max-Forwards: 70
Event: dialog
Subscription-State: active
Content-Type: application/dialog-info+xml
```

```
Content-Length: 504
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="20"
state="full" +
entity="sip:2205@lan.net"><dialog id="dummy" call-id="feb21493-
d84a9bd6@192.168.0.113" +
local-tag="6yln5cag4d" remote-tag="fe9c856f35980aao0" +
direction="recipient"><state>confirmed</state> +
<local><identity>sip:2205@lan.net</identity><target
uri="sip:2205@lan.net"/> +
</local><remote><identity>sip:22052@lan.net</identity><target +
uri="sip:22052@lan.net"/></remote></dialog></dialog-info> +
-------------------------------------------------------------------------------
.... The UA 2200@lan should send the 200 OK to the NOTIFY above ....
-------------------------------------------------------------------------------


-------------------------------------------------------------------------------
Received from udp:10.0.0.5:5060 at 19/10/2005 12:36:04:050 (580 bytes):
Page 7 de 9
#This turns the led BACK OFF
NOTIFY sip:2200@10.0.0.1:2051;line=xjqldyhz SIP/2.0
Via: SIP/2.0/UDP 10.0.0.5:5060;branch=z9hG4bK-
05f4e06451c0349ccf9fb79c55280932
From: <sip:2205@lan.net;user=phone>tag=6sam28oefu
To: <sip:2200@lan.net>tag=gumvap0ha0
Call-ID: 3c2675eaad57-2fjnztudzq25@ST2030
CSeq: 21 NOTIFY
Max-Forwards: 70
Event: dialog
Subscription-State: active
Content-Type: application/dialog-info+xml
Content-Length: 150
<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="21"
state="full" entity="sip:2205@lan.net"></dialog-info>
-------------------------------------------------------------------------------
.... The UA 2200@lan should send the 200 OK to the NOTIFY above ....
-------------------------------------------------------------------------------
```

ST2030 SIP Distinctive ringing and CWT using <Alert-Info> header

1. Overview

This document describes how ST2030 uses <Alert-Info> SIP header in INVITE requests for distinctive ringing and distinctive CWT (Call waiting tone) features.

2. <Alert-Info> Header and Ringing management

This chapter specifies how the ST2030 SIP handles the <Alert-Info> header and parameters in SIP when the phone is in idle state, i.e., no other call is currently active.

2.1 System ringers

When ST2030 is asked to play a ringer it should look for it within previously downloaded system ringers.

Example:
Alert-Info: MyMelody8 will trigger "MyMelody8" (system)

2.2 Void or not understood Alert-Info

When ST2030 receives a void or not understood <Alert-Info> header, it plays a default Distinctive ringing melody which is not configurable.
Example of a void Alert-Info:

Alert-Info: will trigger Default Distinctive ringing melody

Example of not understood (unable to play):

Alert-Info: <http://music.com/boom.mp3>

Alert-Info: Toto if there is no "Toto" ringer downloaded.

2.3 Alert-Info with local URL

When ST2030 receives a Alert-Info with a local URL 127.0.0.1 as argument, it plays the ringers named as the character string following "http://127.0.0.1/"

Example of Alert-Info with local URL:

Alert-Info: http://127.0.0.1/MyMelody5 will trigger "MyMelody5"

2.4 Alert-Info with URN

When ST2030 receives a Alert-Info targeting a URN designating a local melody, it plays the ringers named as the character string received.

Example of Alert-Info with URN:
Alert-Info: MyMelody8
will trigger "MyMelody8"

2.5 Alert-Info with info parameter

When ST2030 receives a Alert-Info with an info parameter specified, it plays the ringers named as the character string received in the info parameter.

Example of Alert-Info with info parameter:
Alert-Info: <http://www.notused.com>;info=MyMelody9
will trigger "MyMelody9"

3. <Alert-Info> Header and Call Waiting Tones management

This chapter specifies how the ST2030 SIP handles the <Alert-Info> header and parameters in SIP when the phone has already an active call.
In this case, the header will determine which Call Waiting tone will be played.

3.1 Description

The same behaviour as in chapter 1 will be applied. The difference is the phone will look for the tones within previously downloaded Call Waiting Tones instead.

In case the header is not understood or no match is found with downloaded tones, default tone will be applied.

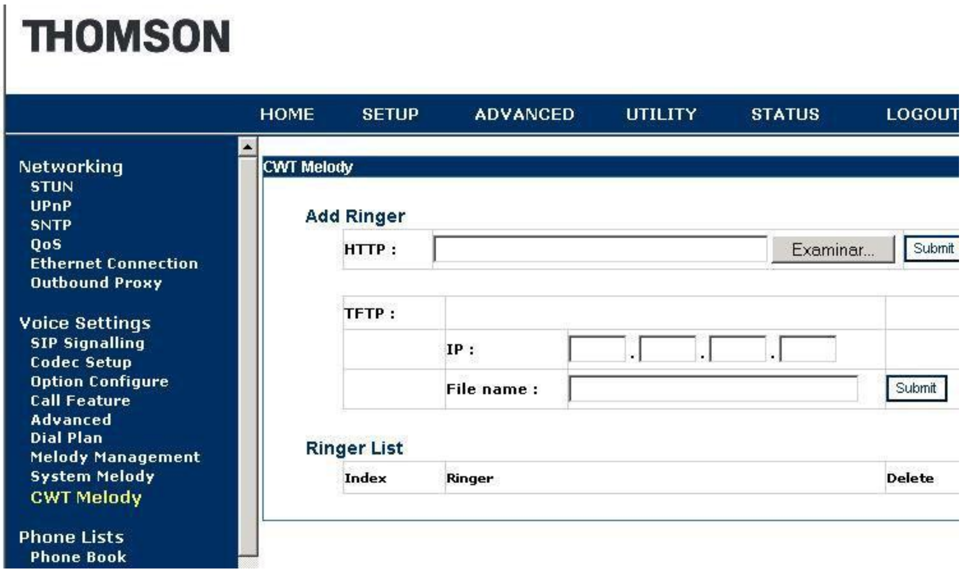
4. System melodies and Call Waiting Tones download

Melodies are described using RTTTL files as documented in the Admin Guide.
There are two ways to download files including System Melodies and Call Waiting tones to ST2030:

Web GUI (admin): section Advanced → System Melody or Advanced → CWT Melody. The management in this sections is exactly the same as in Melody Management, described in the Admin Guide.

Voice Settings

SIP Signalling	Configure the SIP signalling parameters.
Codec Setup	Select to setup your prefered Codec.
Option Configure	Configure call feature function to visible or invisible that effect.
Call Feature	Select the telephone operations of call feature.
Advanced	Configure advanced telephone settings.
Dial Plan	Configure the dial plan.
Melody Management	Add/delete your ring tone.
System Melody	Update system melody.
CWT Melody	Update CWT(call waiting tone) melody.



APS: file names can be included in the information file (*.inf) within the parameters:

system_melodies=
call_waiting_tone=

in [config] section.

Please check ST2030-AutoProvisioning-V0026.pdf for details.

Some examples of melodies and call waiting tones, and their usage within an .inf file can be found in this release package, "APS sample files" folder.

Part 10 - Remote Phonebook Specification

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 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.

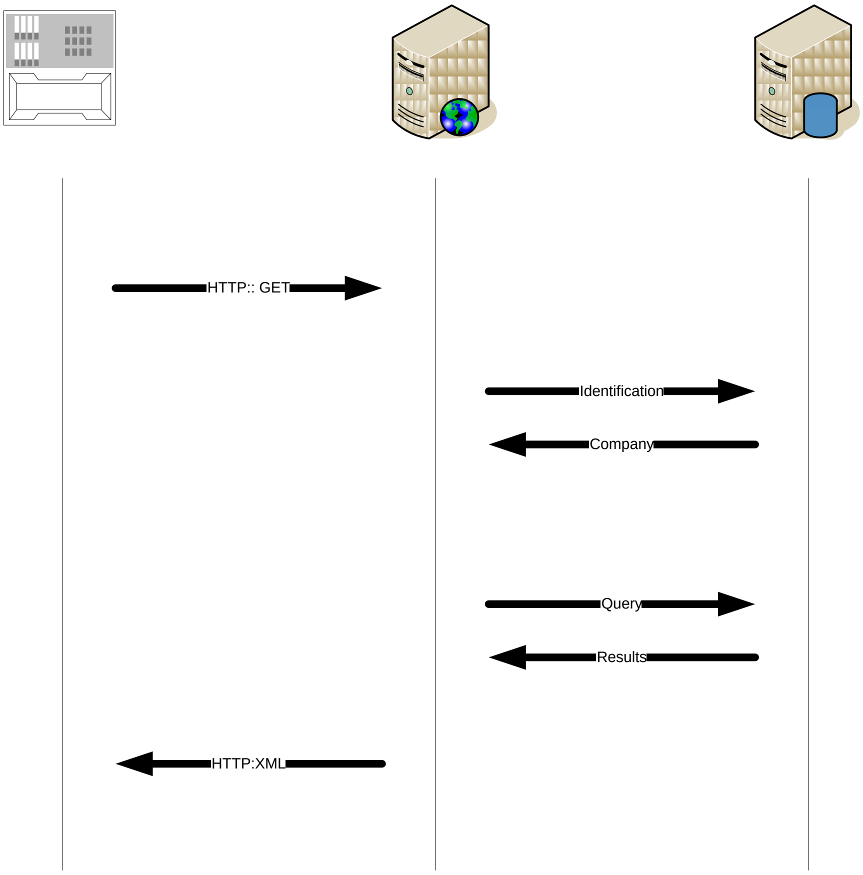
Example:

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

2. Server specifications

2.1 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.



2.2 Phone query

The ST2030 is able to send several parameters to the server in the *GET* request:

Parameter	Description
#IP	IP address of the phone
#MAC	MAC address of the phone
#SEARCH	The string to search (most likely the first letters of the last name of the person we want to lookup the telephone number)

Some of those parameters will be sent by the telephone in the 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

2.3 XML Tags used by ST2030

<ThomsonPhoneBook>
</ ThomsonPhoneBook>

<DirectoryEntry>
</DirectoryEntry>

```
<Name>
</Name>

<Telephone>
</Telephone>
```

See following paragraphs for some examples.

2.4 Answer to the request from the server

2.4.1 No answers

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>
```

2.4.2 Less than 32 results to the query

If the server finds less than 32 answers to the query, it should send to the telephone this kind of 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>
```

2.4.3 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 once. 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.

3. ST2030 Phone Specifications

3.1 Presentation

The Thomson ST2030 will be able to send the queries to the HTTP server, and display the XML pages sent back from the server, and enable the user to navigate in simple menus, in order to select an entry, view the number, and call the contact.

3.2 Configuration

The URL of the query CGI that has to called in order to perform a search in the phonebooks must be specified in the configuration file of the Thomson ST2030. They are written in the following way:

http://www.server.com/services/phonebookl/search.php?IP_ADDR=#IP&MAC_ADDR=#MAC&NAME=#SEARCH

In those URL if some pattern starting with the "#" character are found and they match some predefined strings, they are replaced by the corresponding values. Here are the conversions:

#IP → replaced by the IP address in decimal. Ex: 192.68.0.1
#MAC → replaced by the MAC address in hexa. Ex: 000E504EA77B
#SEARCH → replaced by the entry of the search request. EX: dup

An example of config file:
[st2030s_common.txt]

```
        …
        Phonebook1_url =
        http://www.server.com/services/phonebookl/search.php?IP_ADDR=#IP
        &NAME=#SEARCH
        Phonebook1_name = Thomson Telecom
        Phonebook2_url =
        http://www.server1.com/pb/search.php?NAME=#SEARCH
        Phonebook2_name = Inventel
        Phonebook3_url =
        http://www.server2.com/services/phonel/s.php?MAC_ADDR#MAC&Na
        me=#SEARCH
        Phonebook3_name = Yellow pages
        …
```

3.3 Cancel Key

At anytime, if the Cancel hard key is pressed, the phone returns to its IDLE mode.

3.4 Accessing the Phonebooks

3.4.1 MGCP

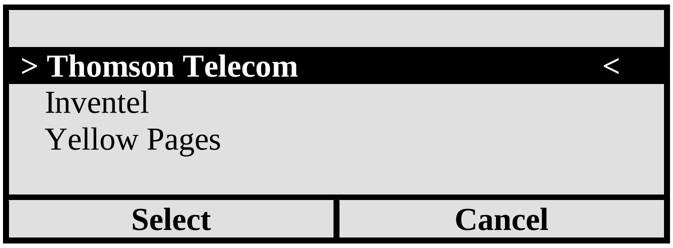
One phonebook

If only one phonebook is available (i.e. Phonebook2_url is empty), ST2030 goes directly to the Query Menu.

Several phonebooks

If more than one phonebook is available (i.e. Phonebook2_url is set), ST2030 goes to the Phonebook menu.

If the ST2030 finds more than 1 phonebook in the config file, it displays the phone book names. The user is able to browse in the list of phonebooks using the *up* and *down* keys, and select one of them using a softkey or the *OK* button. He is of course also able to cancel his search.



> Thomson Telecom <
Inventel
Yellow Pages
Select **Cancel**

3.4.2 SIP

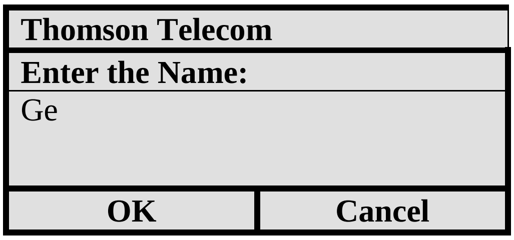
The PBX (or network) based phonebook(s) are displayed along with the Personal Phonebook that is inside the ST2030.

The user will be able to browse in the list of phonebooks using the *up* and *down* keys, and select one of them using a softkey or the *OK* button. He will of course also be able to cancel his search.

> Personal Phonebook <
Thomson Telecom
Inventel
Yellow Pages
Select **Cancel**

3.5 Query Menu

When the user enters in the phonebook service, he is asked to enter the search query. He can enter the search string by using the DTMF keys, and can send or cancel the search with 2 softkeys.



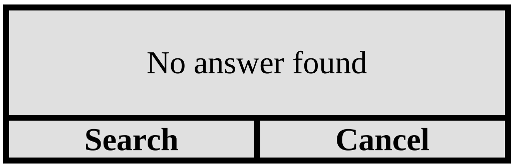
Thomson Telecom
Enter the Name:
Ge
OK **Cancel**

3.6 Incoming call during search

If the phone receives an incoming call anytime during the search, the search is cancelled and the phone shows up the call, in the same way as if it was in IDLE mode before the call arrives.

3.7 Displaying empty phonebook

If no results are found, the phone will receive an empty *ThomsonPhoneBook* XML page. The user will have the choice to search another string, or to abort.


No answer found

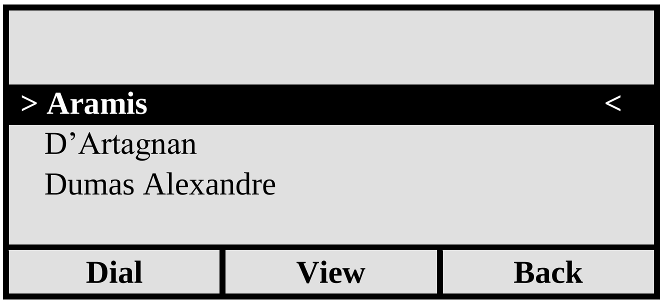
If the user selects "*Search*", he will return to the previous menu.
If the user selects "*Cancel*", the phone goes back to IDLE mode.

3.8 Displaying less than 32 results phonebook

If less than 32 answers are found, the server will send back a ThomonPhoneBook XML page, containing all the contacts' information. The phone will display the names of the contact in a list.

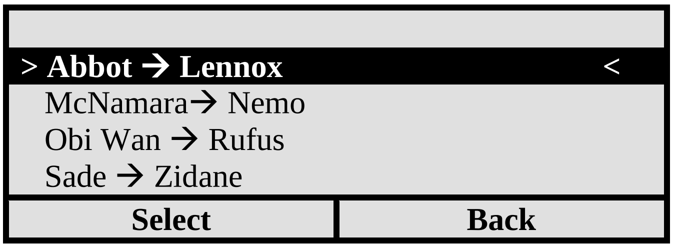
The user will be able to browse the list of answer using the up and down keys. He will have 3 softkeys which will enable the user to directly call the contact, display his number, or go back to previous screen.

The phone displays in the list the content of the XML *Name* tag of each entry (XML *Directory* XML) of the list. If the *Dial* softkey or OK button is selected, the call is initiated. If the *Display* softkey is selected, the telephone number of the contact is displayed. If the *Cancel* softkey is selected, the user returns to the phonebook query page.



3.9 Displaying more than 32 results phonebook

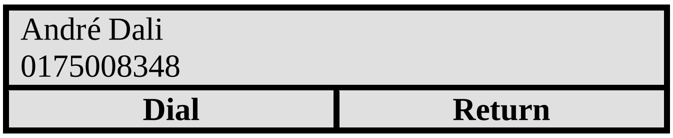
If the server finds more than 32 entries to the query, it will not return all the answers, but will instead send a *ThomsonMenu* XML page, giving a list of URL. These URL will be used to fetch the results by blocks of 32 entries. The phone will display the menu sent by the server. The user will be able to browse in the list of URLs using the *up* and *down* keys, and select one of them using a softkey or the *OK* button.



> Abbot → Lennox **<**
McNamara→ Nemo
Obi Wan → Rufus
Sade → Zidane

3.10 Displaying an entry in the phonebook

When the users choose to display the telephone number of a user, the phone will display on the screen the telephone number of this contact, and propose to call this contact. He will also propose to go back to the previous listing of the results of his query.



André Dali
0175008348

If the user selects the softkey *Dial*, the phone will initiate a call to this contact. If the return softkey is selected, the phone will display the previous page.

3.11 Calling an entry in the phonebook

When the users orders the phone to call a contact, the phone directly dial the number as if it was typed and validated on the keypad.

3.12Timeout

If any request of the ST2030 takes more than 5 seconds to be answered, ST2030 will display an alert display “Service unavailable” during 2 seconds and then go to IDLE mode.

Note

To know more the installation of remote phone book, you can download the installation and Setup guide on:
www.thomsonbroadbandpartner.com

Part 11 Telnet Activation

Introduction

This part reviews and summarizes ST20xx Telnet commands useful for remote management of the phone.

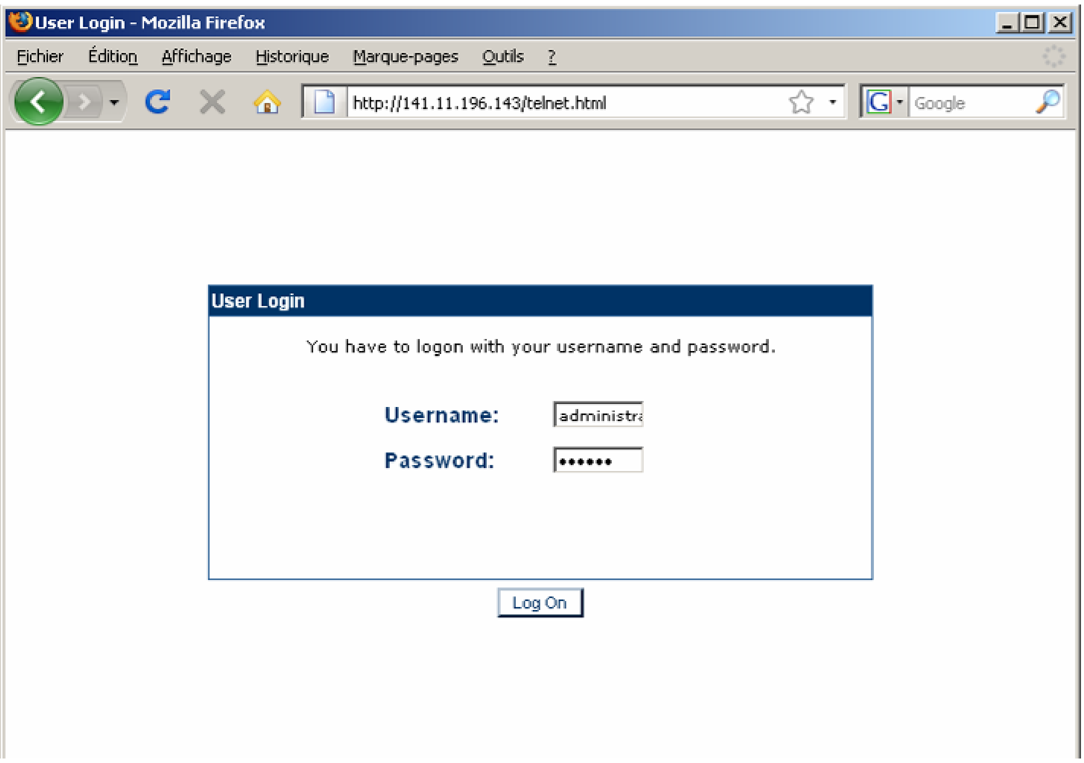
Other commands (like configuration of the phone's parameters) are described in another Technical document: "Syntax of ST20xx Configuration file". It is issued in the firmware release package.

Enable Telnet service

By default, the Telnet service is disabled for security reasons.
So the Telnet service must be enabled before trying to open a telnet session on the phone.

Enter the following URL on you web browser:
http://ip_address_of_the _phone/telnet.html

Then enter the username and password (same as the web interface):
Username (default): administrator
Password (default): 784518



http://141.11.196.143/telnet.html

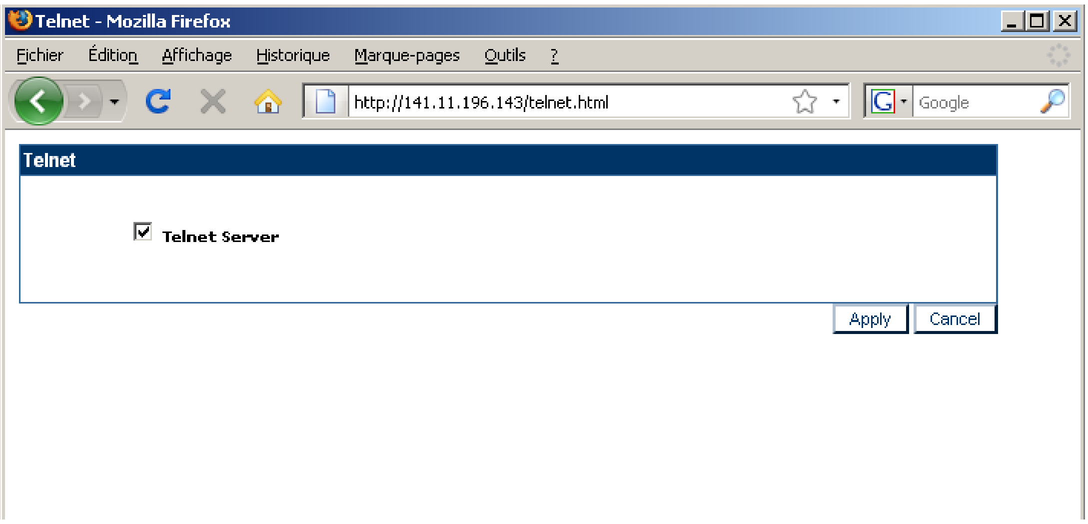
User Login

You have to logon with your username and password.

Username: administra

Password: ••••••

Then check off the Telnet Server to activate it.

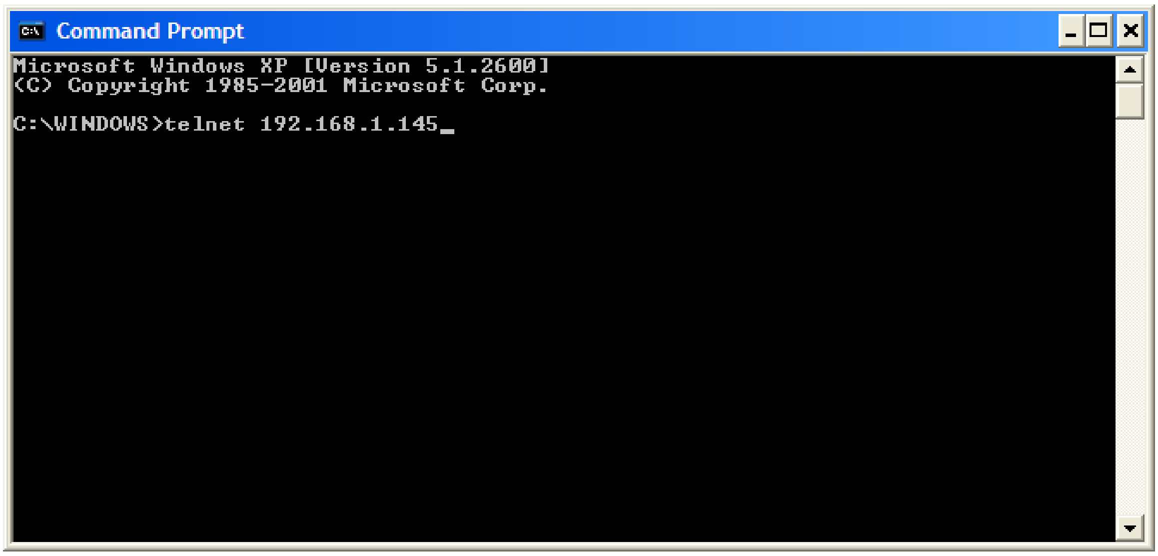


Telnet

☑ Telnet Server

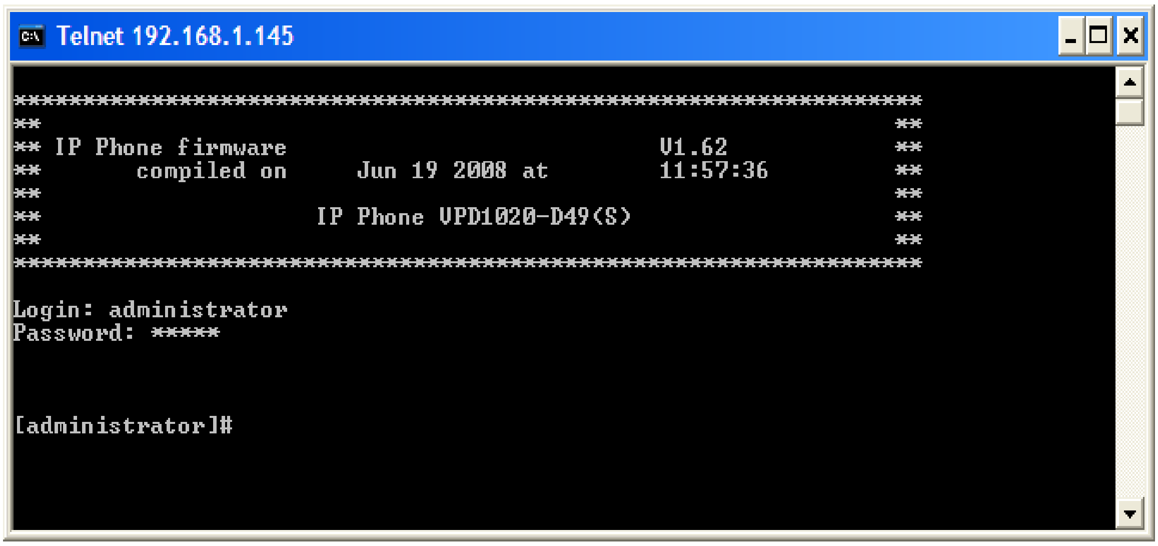
Open a Telnet session

In Windows OS: Click on “Start” , select “Execute”, type “cmd” then press “Enter”
In the CMD window, type “telnet <ip address>”



```
Microsoft Windows XP [Version 5.1.2600]
(C) Copyright 1985-2001 Microsoft Corp.

C:\WINDOWS>telnet 192.168.1.145_
```

Enter the usename and password:
Login (default): administrator
Password (default): 789234



```
******************************************************************
**                                                              **
** IP Phone firmware                           V1.62            **
**       compiled on     Jun 19 2008 at        11:57:36         **
**                                                              **
**                    IP Phone VPD1020-D49(S)                   **
**                                                              **
******************************************************************

Login: administrator
Password: *****



[administrator]#
```

Telnet commands

The table below, lists the telnet commands mainly focused on the remote management.

Command	Descriptions	Examples
Tftp	Upgrade Firmware tftp X.X.X.X filename [f/F] [TFTP server should be ready and select the corresponding directory]	[administrator]# tftp 192.168.1.3 v2030SG.080619.1.62.3.zz
Tftp2	Upgrade a number of defined files (See below) tftp2 telcfg X.X.X.X filename [f/F] tftp2 melody X.X.X.X filename tftp2 sys_melody X.X.X.X filename tftp2 cwt_melody X.X.X.X filename tftp2 listparms X.X.X.X filename tftp2 listlangtable X.X.X.X filename Language-index tftp2 langtable X.X.X.X filename tftp2 listtonetable X.X.X.X filename Country-index tftp2 tonetable X.X.X.X filename tftp2 listphonebook X.X.X.X filename tftp2 phonebook X.X.X.X filename tftp2 putimage X.X.X.X filename [f/F]	[administrator]# tftp2 sys_medlody 192.168.1.3 tone-RG.txt

	tftp2 putfile X.X.X.X filename [f/F] [TFTP server should be ready and select the corresponding directory]	
Reboot	Trigger phone reboot	[administrator]# Reboot now
Activate	Enable services	[administrator]# Activate
Commit	Save settings	[administrator]# commit
Version	Show boot, dsp and FW version	[administrator]# Version
Info	Show phone info	[administrator]# info
sip show reg	Show registrar info	[administrator]# sip show reg
Ping	Ping the specified Host ping <X.X.X.X> [-t|-n <Count>]	[administrator]# ping 192.168.1.3

Reset to default

Use the following procedure to reset the phone to factory default settings

Commands	Description	Example
sys set rel 0 ffs format ffs commit ffs commit flash clean nmm reboot immediate	Reset to default	[administrator]# sys set rel 0 [administrator]# ffs format [administrator]# ffs commit [administrator]# ffs commit [administrator]# flash clean nmm [administrator]# reboot immediate

Part 12 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.

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

Table 9.1 Troubleshooting

Part 13 Appendix

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	Deactivated	x	x				Activated : SIPS + TLS

Table 9.1 Configurable Parameters and Commands

Examples of files to configure auto provisioning via TFTP

Example of file INF SIP:

```
[provision_mode]
provision_mode=auto_provisioning

[application]
fw_filename=st2030SEG_fw.zz
dsp_filename=v2030_dsp_ver_.zz
boot_filename =v2030_boot_ver.zz

[config]
telcfg=TelConf2030SEG_telcfg.txt
common_config=st2030s_common.txt

melodies=Melodies.txt
system_melodies=S_Melodies.txt
call_waiting_tone=ringtone.txt
tone_table=ToneTb.txt
language_table=langtable.txt
```

Example of file INF MGCP:

```
[provision_mode]
provision_mode=auto_provisioning

[application]
fw_filename=v2030MEC_fw.zz
dsp_filename=v2030_dsp_ver_.zz
boot_filename =v2030_boot_ver.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_telcfg.txt
```

```
common_config=st2030m_common.txt
melodies=Melodies.txt
system_melodies=S_Melodies.txt
call_waiting_tone=ringtone.txt
beep_tone=beeptone.txt
```

Example of file Common_config SIP:

```
[ipp]
AutoAnsMode=0
AutoAnsTimer=0
Autocall=0
AutocallNumber=
AutoRejectTimer=0
AutoStopTimer=0
AutoHangUpTimer=3
CaLogPrefix=
Check_phonebook_domain_name=1
Disable_call_progress=0
Hide_Phone_Number_Display=0
HideMissedCall=0
LanguageType=0
LocalMusicOnHold=1
ManualLog=0
OptionVisible=8191
PasswordString=0000
PhoneLock=0
Pick_up_call=1
Ringer_default_vol=4
Ringer_lock=0
RingToneMP1=0
RingToneMP2=0
RingToneMP3=0
RingToneMP4=0
Suppress-DomainName-Flag=1
Transfer_to_voice_mail=1
TransfOnRingFlag=0


[net]
DNSSrv1=0.0.0.0
DNSSrv2=0.0.0.0
DNSFLG=1
DSCPflag=0
ESWITCH_LAN=1
ESWITCH_PC=1
Gateway=0.0.0.0
ipwan=10.1.24.88
IpDFflag=0
MaskWan=0.0.0.0
PersistentVLANfromDHCP=0
TelnetTime=240
```

```
TelnetSrv=1
VLAN=0
WanModFlag=1



[pstn]

[sip]
ACD=0
AuthMessageServer=0.0.0.0
KeppAliveDest=0.0.0.0
KeepAliveTimer=0
AuthNotifyCheckSync=0.0.0.0
CallBlockType=0
CF_Address1=
CFNA_Time1=30
CF_domain=
ConnectReuseFlgMP1=0
ConnectReuseFlgMP2=0
ConnectReuseFlgMP3=0
ConnectReuseFlgMP4=0
ConnectReuseFlgBK1=0
ConnectReuseFlgBK2=0
ConnectReuseFlgBK3=0
ConnectReuseFlgBK4=0
ConferenceMode=0
ConferenceFactoryURI=conf
CallParkType=0
CLIPDisplayPrior=ppreferred(1)passerted(2)remoteparty(3)from(4)
CalledDisplayPrior=dialed(1)passerted(2)remoteparty(3)
CallPkupFlg=local
CallPkupSC=
CallFwdFlg=local
CallFwdOffSC=
CallFwdAlwaysSC=
CallFwdBusySC=
CallFwdNoAnswerSC=
CallParkFlg=local
CallParkSC=
CallRetreiveSC=
CallBlockFlg=local
CallBlockAllowSC=
CallBlockRejectSC=
CallBlockOFFSC=
ClirFlg=local
ClirOnSC=
ClirOffSC=
CallWaitingFlg=local
CallWaitingOnSC=
CallWaitingOffSC=
call_hold_method=0
```

```
CFUOnSV=
CFUOffSV=
DisplayNumFlag1=0
DisplayNumFlag2=0
DisplayNumFlag3=0
DisplayNumFlag4=0
DisplayNum1=
DisplayNum2=
DisplayNum3=
DisplayNum4=
DNDFlg=local
DNDOnSC=
DNDOffSC=
DNDOnSV=
DNDOffSV=
DNDResp=1
DisplayName1=
DisplayName2=
DisplayName3=
DisplayName4=
DisplayNameFlag1=0
DisplayNameFlag2=0
DisplayNameFlag3=0
DisplayNameFlag4=0
DiversionHeaderFlag=0
ExpireTimeMP1=3600
ExpireTimeMP2=3600
ExpireTimeMP3=3600
ExpireTimeMP4=3600
ExpireTimeBK1=3600
ExpireTimeBK2=3600
ExpireTimeBK3=3600
ExpireTimeBK4=3600
Earlymediatype=0
HGOnSC=
HGOffSC=
HGOnSV=
HGOffSV=
MissedCallSummary=0
OutBoundProxy0=
OutBoundPort0=5060
OutBoundType=0
ProfileName1=Profile 1
ProfileName2=Profile 2
ProfileName3=Profile 3
ProfileName4=Profile 4
PrivacyFlag1=4
ProxyServerMP1=
ProxyServerMP2=
ProxyServerMP3=
ProxyServerMP4=
ProxyServerBK1=
```

```
ProxyServerBK2=
ProxyServerBK3=
ProxyServerBK4=
P-AssertedIDforCallee=0
regid1=
regid2=
regid3=
regid4=
regpwd1=
regpwd2=
regpwd3=
regpwd4=
RegisterServerMP1=
RegisterServerMP2=
RegisterServerMP3=
RegisterServerMP4=
RegisterServerBK1=
RegisterServerBK2=
RegisterServerBK3=
RegisterServerBK4=
RtpPort=41000
ReTransTimer1=500
ReTransTimer2=4
RegisterFrequencyMP1=600
RegisterFrequencyMP2=600
RegisterFrequencyMP3=600
RegisterFrequencyMP4=600
RegisterFrequencyBK1=600
RegisterFrequencyBK2=600
RegisterFrequencyBK3=600
RegisterFrequencyBK4=600
REQ_RUI_with_port=1
RFC2833=1
RFC2833_rtp_pltype=96
RFC3262=0
RandomCSeqFlag=0
SubscribeDilaogBeforeLogin=1
RandomRTPPortFlag=0
RegEventServer=
ServiceDomainMP1=
ServiceDomainMP2=
ServiceDomainMP3=
ServiceDomainMP4=
ServiceDomainBK1=
ServiceDomainBK2=
ServiceDomainBK3=
ServiceDomainBK4=
SessionExpire=1800
SessionExpireMin=100
SessionMethod=0
SessionRefresher=0
sip_portMP1=5060
```

```
sip_portMP2=5060
sip_portMP3=5060
sip_portMP4=5060
sip_portBK1=5060
sip_portBK2=5060
sip_portBK3=5060
sip_portBK4=5060
sip_reg_srv_portMP1=5060
sip_reg_srv_portMP2=5060
sip_reg_srv_portMP3=5060
sip_reg_srv_portMP4=5060
sip_reg_srv_portBK1=5060
sip_reg_srv_portBK2=5060
sip_reg_srv_portBK3=5060
sip_reg_srv_portBK4=5060
sip_pxy_srv_portMP1=5060
sip_pxy_srv_portMP2=5060
sip_pxy_srv_portMP3=5060
sip_pxy_srv_portMP4=5060
sip_pxy_srv_portBK1=5060
sip_pxy_srv_portBK2=5060
sip_pxy_srv_portBK3=5060
sip_pxy_srv_portBK4=5060
SubscriptionExpire=3600
SubscriptionExpireCB=60
SharedCallAppearance=0
SCAMainLinePrivate=0
Specific_IP_flag=0
HeaderCompact=0
subscribe_event=0
sw_anon_reject=0
sw_CF=0
sw_CFA=0
sw_CFNA=0
sw_CFB=0
sw_not_disturb=0
sw_call_park=0
sw_notify_autoprovision=0
sw_tr_vmail_address1=
sw_tr_vmail_address2=
sw_tr_vmail_address3=
sw_tr_vmail_address4=
sw_tr_vmail_flag=0
sw_park_srv_addr1=
sip_instance_id=0
SFOnSC=
SFOffSC=
SFOnSV=
SFOffSV=
TransferOnRingSC=
TrVoiceMailFlg0=local
TrVoiceMailFlg1=local
```

```
TrVoiceMailFlg2=local
TrVoiceMailFlg3=local
TrVoiceMailFlg4=local
TrVoiceMailOnSC0=
TrVoiceMailOnSC1=
TrVoiceMailOnSC2=
TrVoiceMailOnSC3=
TrVoiceMailOnSC4=
TrVoiceMailOffSC0=
TrVoiceMailOffSC1=
TrVoiceMailOffSC2=
TrVoiceMailOffSC3=
TrVoiceMailOffSC4=
TrVoiceMailRingSC0=
TrVoiceMailRingSC1=
TrVoiceMailRingSC2=
TrVoiceMailRingSC3=
TrVoiceMailRingSC4=
TEL1Number=
TEL2Number=
TEL3Number=
TEL4Number=
TransportFlgMP1=0
TransportFlgMP2=0
TransportFlgMP3=0
TransportFlgMP4=0
TransportFlgBK1=0
TransportFlgBK2=0
TransportFlgBK3=0
TransportFlgBK4=0
TransferUseContact=1
URLTypeMP1=0
URLTypeMP2=0
URLTypeMP3=0
URLTypeMP4=0
URLTypeBK1=0
URLTypeBK2=0
URLTypeBK3=0
URLTypeBK4=0
USRPhoneFlg=1
UnRegister_priorFlgMP1=1
UnRegister_priorFlgMP2=1
UnRegister_priorFlgMP3=1
UnRegister_priorFlgMP4=1
UnRegister_priorFlgBK1=1
UnRegister_priorFlgBK2=1
UnRegister_priorFlgBK3=1
UnRegister_priorFlgBK4=1
use_PrivateNumber=0
VoiceMailAddr=
VoiceMailTelNum=
VoiceMailPort=5060
```

```
[snmp]
commRW1=0
commRW2=0
commRW3=0
flag=1
filter=0
snmpManager1=0.0.0.0
snmpManager2=0.0.0.0
snmpManager3=0.0.0.0
snmpManager4=0.0.0.0
snmpManager5=0.0.0.0
sysTrapSrv=0.0.0.0
sysCommName1=public
sysCommName2=
sysCommName3=
sysContact=Thomson
sysDescr=Thomson IP Phone
sysLocation=France
sysName=ST2030 SIP
TrapLevel=0


[sys]
2833_volume=0
AEC=1
BLFType=0
BLFListSipUri=L/<sip:>
CountryCode=US
config_sn=200402190001
Current_Max_Multiline=10
CodecJitterBufMult=g711a(1/2/4)g711mu(1/2/4)g729(1/2/4)g723(1/2/4)
CodecPktime=g711a(10)g711mu(10)g729(10)g723(30)
CodecAdaptivePlayout=g711a(1)g711mu(1)g729(1)g723(1)
DisableCWtone=0
Dialednum_timeout=4
dtmf_mode_flag=1
EmergencyDialPlan=|911|11x|1[2-8]|
FeatureKeyExt01=L/<sip:>
FeatureKeyExt02=L/<sip:>
FeatureKeyExt03=L/<sip:>
FeatureKeyExt04=L/<sip:>
FeatureKeyExt05=L/<sip:>
FeatureKeyExt06=L/<sip:>
FeatureKeyExt07=L/<sip:>
FeatureKeyExt08=L/<sip:>
FeatureKeyExt09=L/<sip:>
FeatureKeyExt10=L/<sip:>
FeatureKeyExt11=L/<sip:>
FeatureKeyExt12=L/<sip:>
FeatureKeyExt13=L/<sip:>
```

```
FeatureKeyExt14=L/<sip:>
FeatureKeyExt15=L/<sip:>
FeatureKeyExt16=L/<sip:>
FeatureKeyExt17=L/<sip:>
FeatureKeyExt18=L/<sip:>
FeatureKeyExt19=L/<sip:>
FeatureKeyExt20=L/<sip:>
FeatureKeyExt21=L/<sip:>
FeatureKeyExt22=L/<sip:>
FeatureKeyExt23=L/<sip:>
FeatureKeyExt24=L/<sip:>
FeatureKeyExt25=L/<sip:>
FeatureKeyExt26=L/<sip:>
FeatureKeyExt27=L/<sip:>
FeatureKeyExt28=L/<sip:>
FeatureKeyExt29=L/<sip:>
FeatureKeyExt30=L/<sip:>
FeatureKeyExt31=L/<sip:>
FeatureKeyExt32=L/<sip:>
FeatureKeyExt33=L/<sip:>
FeatureKeyExt34=L/<sip:>
FeatureKeyExt35=L/<sip:>
FeatureKeyExt36=L/<sip:>
FeatureKeyExt37=L/<sip:>
FeatureKeyExt38=L/<sip:>
FeatureKeyExt39=L/<sip:>
FeatureKeyExt40=L/<sip:>
FeatureKeyExt41=L/<sip:>
FeatureKeyExt42=L/<sip:>
FeatureKeyExt43=L/<sip:>
FeatureKeyExt44=L/<sip:>
FeatureKeyExt45=L/<sip:>
FeatureKeyExt46=L/<sip:>
FeatureKeyExt47=L/<sip:>
FeatureKeyExt48=L/<sip:>
FeatureKeyExt49=L/<sip:>
FeatureKeyExt50=L/<sip:>
FeatureKeyExt51=L/<sip:>
FeatureKeyExt52=L/<sip:>
FeatureKeyExt53=L/<sip:>
FeatureKeyExt54=L/<sip:>
FeatureKeyExt55=L/<sip:>
FeatureKeyExt56=L/<sip:>
FeatureKeyExt57=L/<sip:>
FeatureKeyExt58=L/<sip:>
FeatureKeyExt59=L/<sip:>
FeatureKeyExt60=L/<sip:>
FeatureKeyExt61=L/<sip:>
FeatureKeyExt62=L/<sip:>
FeatureKeyExt63=L/<sip:>
FeatureKeyExt64=L/<sip:>
FeatureKeyExt65=L/<sip:>
```

```
FeatureKeyExt66=L/<sip:>
FeatureKeyOpt01=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt02=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt03=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt04=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt05=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt06=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt07=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt08=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt09=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt10=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt11=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt12=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt13=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt14=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt15=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt16=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt17=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt18=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt19=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt20=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt21=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt22=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt23=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt24=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt25=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt26=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt27=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt28=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt29=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt30=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt31=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt32=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt33=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt34=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt35=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt36=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt37=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt38=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt39=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt40=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt41=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt42=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt43=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt44=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt45=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt46=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt47=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt48=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt49=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt50=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt51=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
```

```
FeatureKeyOpt52=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt53=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt54=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt55=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt56=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt57=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt58=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt59=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt60=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt61=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt62=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt63=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt64=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt65=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
FeatureKeyOpt66=dialog(1)regDND(0)ua-profile(0:cfu(0)dnd(0)sf(0)hg(0))
NormalDigitFlag=1
PCallLogURL=
PSettingURLdl=
PSettingURLul=
Phonebook1_url=
Phonebook2_url=
Phonebook3_url=
Phonebook4_url=
Phonebook5_url=
Phonebook6_url=
Phonebook7_url=
Phonebook8_url=
Phonebook1_name=
Phonebook2_name=
Phonebook3_name=
Phonebook4_name=
Phonebook5_name=
Phonebook6_name=
Phonebook7_name=
Phonebook8_name=
PBXconfiguration=0
PBXprefix=
softkey01=TrVoiceMail
softkey02=CallLog
softkey03=PickUp
softkey04=LockPhone
softkey05=VoiceMail
softkey06=RetrievePark
softkey07=DNDstate
softkey08=ShortCut1
softkey09=ShortCut2
softkey10=ACDCheckIn
softkey11=ACDAvailable
softkey12=Login
StartSpareFK=0
ServiceSupervisionFK=11
ServiceSupOrder=callfwd(0)dnd(0)secfilter(0)huntgroup(0)
```

```
SuppressRTCP=0
TelnetID=administrator
TelnetPWD=789234
UserID=user
UserPWD=
VOIPDialPlan=x.T
WebPWD=784518

[autoprovision]
AutoprovisionFlag=1
AutoprovisionHTTPServer=
AutoprovisionTFTPServer=
AutoprovisionTimeDays=0
Autoprovisionstarttime=00:00
AutoprovisionTimeSpan=0
AutoprovisionRetryPeriod=30

[qos]
DHCPVLANid1=1
DHCPVLANid2=1
DSCP1=46
DSCP2=40
DSCPdata=40
TOS=160
VLANid1=1
VLANid2=1
VLANTag1=6
VLANTagdata=6

[ftp]
FTPID=
FTPPWD=

[ntp]
NtpDaylight=1
NTPFlag=0
NtpIP=
NtpMode=0
NtpSyncTime=1
NtpZoneNum=23


[pppoe]
PPPoEID=
PPPoELCPMod=2
PPPoELCPTimer=60
PPPoEPWD=
PPPoEMTU=1454

[stun]
STUNSrv=
STUNPort=0
```

```
STUNFlag=0
[upnp]
UPnPFlag=1

[customer]
sw_call_wait=1
```

Example of file Common_config MGCP:

```
[ipp]
LanguageType=0


[mgcp]
ActivatedCAProfile=0
caip1=ca@[217.66.118.137]:2427
caip2=ca@[217.66.118.137]:2427
caip3=ca@[217.66.118.137]:2427
caip4=ca@[217.66.118.137]:2427
nat_keep=1
nat_keep_timeout=90
rgw=218.97.191.86
rgw_use_ip_addr=0


[net]
DNSSrv1=202.96.209.5
DNSSrv2=168.95.1.1
DSCPflag=0
ESWITCH_LAN=1
Gateway=218.97.191.81
ipwan=218.97.191.86
MaskWan=255.255.255.240
TelnetSrv=0
VLAN=0
WanModFlag=0


[sys]
CountryCode=FR
Phonebook1_url=
Phonebook1_name=
TelnetID=administrator
TelnetPWD=789234
UserID=user
UserPWD=
WebPWD=784518



[ntp]
NtpDaylight=0
```

```
NTPFlag=1
NtpIP=192.43.244.18
NtpZoneNum=0
```

Example of file TelConf: (Don't touch)

Not recommended to change.

Specific file MAC:

The content is the same as common_config file, except the parameter "config_sn=xxxxxxxxxxxx", which need to be changed for each APS update.

Files Deck: (only MGCP)

Until version 1.61.4 (11 files):

MC.080620.abrege.thd
MC.080620.call.thd
MC.080620.config.thd
MC.080620.default.thd
MC.080620.filtrage.thd
MC.080620.log.thd
MC.080620.main.thd
MC.080620.phonebook.thd
MC.080620.renvoi.thd
MC.080620.register.thd
MC.080620.supervision.thd

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