

technicolor



**TB30 SIP
New Features
SG v2.78
Release Notes**

Contents

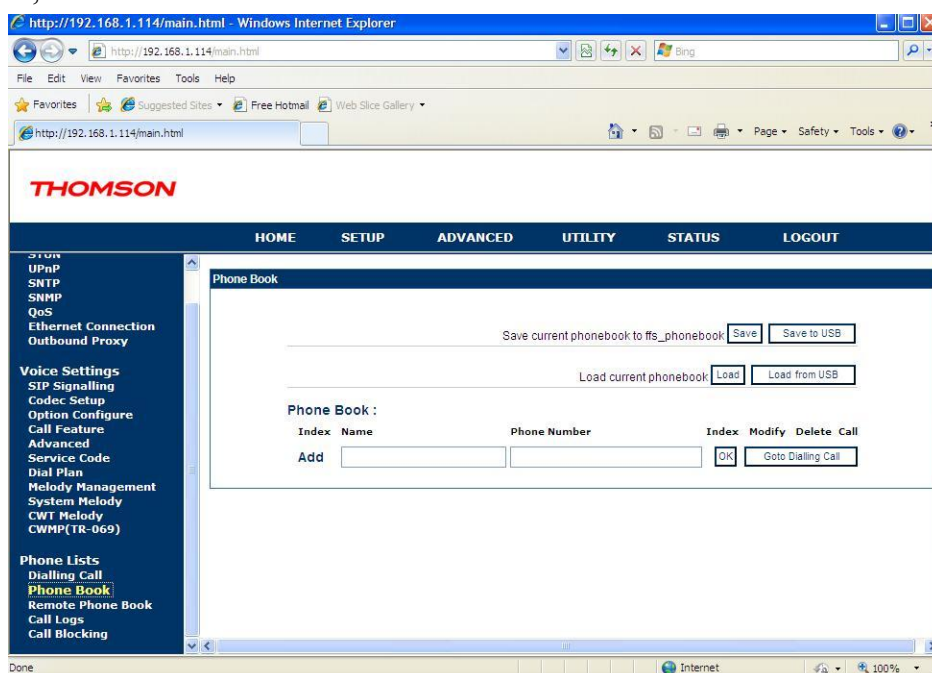
1	Overview	1
2	Save Phone Book , Config File on USB key (3996).....	1
3	Direct Corporate Phone Book Access (3997)	3
4	SIP unregister does not behave in the correct manner (4073)	4
5	LCD Backlight for Event (4172)	5
6	Never Hangup (4319).....	6
7	Pause in Speed Dial (4724).....	7
8	Display customer icon when boot (4332)	8
9	Block USB APS (4326)	9
10	Event Header(5067).....	9
11	Downloadable Ring Tone (4328).....	10
12	Conference Not Allow NW (4862).....	11
13	Group Pickup Key (4794).....	12
14	VPN Escape Code For Per Line For SF and HG(4907)	13
15	Call Join (4458).....	14
16	DNS Query (4803).....	14
17	SIP Scenario of Alternate Call with Notify Event (4311)	16
18	Default Language Table (4964)	17
19	Carries All Codec When Transfer is processed by server (4957).....	17
20	LCD Backlight for Event Indication(4172)	19
21	Improve Key Detection (5064)	20
22	Remove DHCP Option 60 (5063).....	20
23	Logo Icon Display (5085).....	21
24	Default value for Common Config File (xxxx)	21
25	Full Duplex Handfree(?)	22
26	USB Headset Support (4331).....	22
27	Call transfer is not available in double (4896).....	23
28	Direct DND (5086)	24
29	Supervisor listening on handset, while operator (5184)	25
30	The client transmits the End packets only once whereas it should be transmitted three times (5159)	26
31	Unregister phone ability to call another unregistered phone directly (5195)	27
32	BLF improvement: display the caller ID (5205).....	28
33	Universal pickup behavior-Update from PCR 4794 (5347)	29
34	Support writing from right to left for Hebrew (5188).....	32
35	Support package as-feature-event (5187)	35
36	Support Hebrew Right sorting & searching of the phonebook(5188)	37
37	HTTPS type V1 (update PCR5341).....	39
38	USB Stick Debug Log (5369).....	40
39	Extend WebGui Session Timeout (5368)	41
40	The call is not automatically released if the phone doesn't receive any rtp flow (5365)	41
41	Support Contact Header information included in the BYE message (5376).....	42
42	Language filename display on info.xml page (5377)	42
43	Support "HTTPS Web Page" (5382)	43

1 Overview

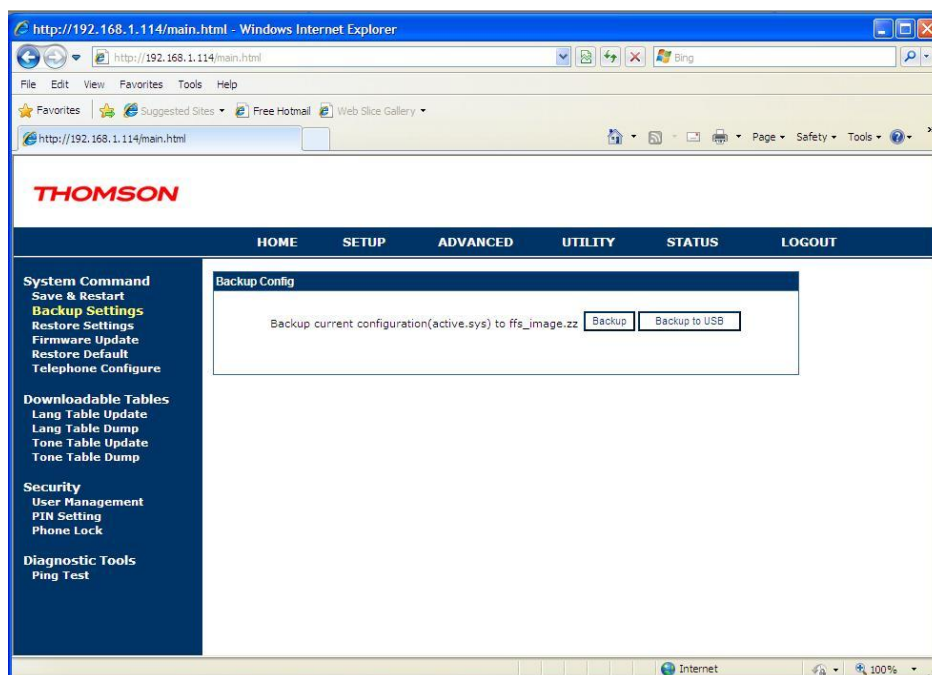
This document describes a set of new features included in TB30 SIP v1.73 in order to improve their usability in different environments.

2 Save Phone Book , Config File on USB key (3996)

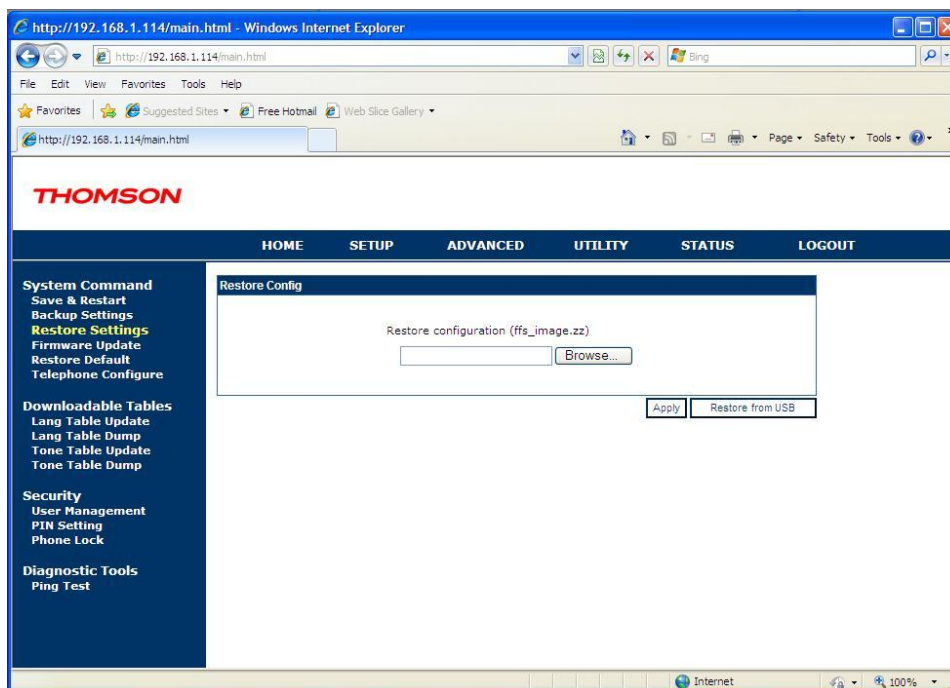
For Phone Book, add “Save to USB” and “Load from USB” button in the Web GUI.



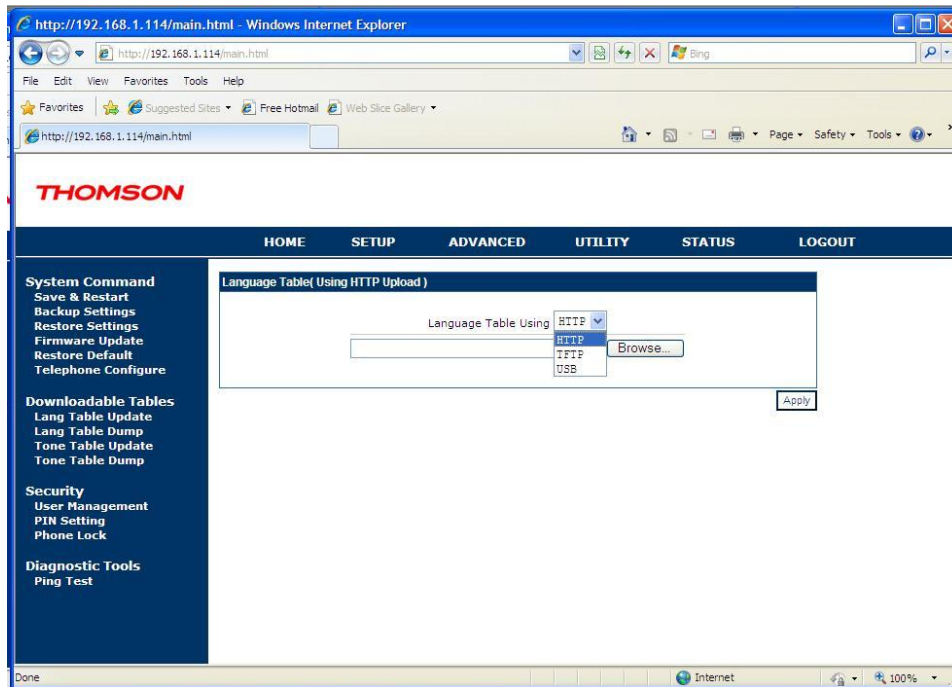
For Backup Setting, add “Backup to USB” button in the Web GUI.



For Restore Setting, add “Restore from USB” button in the Web GUI.



For Downloadable Tables (Lang Table Update, Lang Table Dump, Tone Table Update and Tone Table Dump) add USB option with the existing HTTP and TFTP.



The file must be in the root directory of the USB key. The following are their filenames.

Phonebook – ffs phonebook.zz

Setting – ffs_image.zz

Tone Table – ToneTbl.zz

Language Table – LangTbl.zz

3 Direct Corporate Phone Book Access (3997)

A new parameter is added for Direct Corporate Phone Book Access. This parameter is configurable in common config file and telnet.

LocalPB = 0/1 (1: Default)

```
// 0: disable (Disable Local Phone book)
```

```
// 1: enable (enable ...as original)
```

Telnet CLI:

lcdui set LocalPB 0/1

Remark for local phonebook disabled:

- if there is only one Corporate phone book, it will access it directly without choosing the phonebook



4 SIP unregister does not behave in the correct manner (4073)

With the SIP Unregister check-box selected a de-Register is sent with an Expiry Timer of 0 Seconds which is correct but with a Contact of * (star) rather than the correct contact of the phone

We would also expect the device to de-register during the following scenarios with its own contact rather than Contact with "*"

01) Soft Reboot

02) Change in credentials (Need to de-register old credentials and register with new credentials)

03) Change in voice profile (Need to de-register de-selected profile and register the newly selected profile)

"It's like that since the beginning. It follows the rfc 3261:

10.2.2 Removing Bindings

Registrations are soft state and expire unless refreshed, but can also be explicitly removed. A client can attempt to influence the expiration interval selected by the registrar as described in Section

10.2.1. A UA requests the immediate removal of a binding by specifying an expiration interval of ""0"" for that contact address in a REGISTER request. UAs SHOULD support this mechanism so that bindings can be removed before their expiration interval has passed.

The REGISTER-specific Contact header field value of ""*"" applies to all registrations, but it MUST NOT be used unless the Expires header field is present with a value of ""0"".

Use of the ""*"" Contact header field value allows a registering UA to remove all bindings associated with an address-of-record without knowing their precise values."

The customer has an option for REBOOT in GUI and hence we would like to see this addressed in TB30.

The contact header of Unregister message should be the correct contact of the phone not "*".

- If change profile, phone unregister with the old account.
- If change account, phone unregister with the old account.
- If system reboot from web GUI or from the Menu Options, phone unregister before reboot.
- IP of phone is change, phone unregister before reboot.



5 LCD Backlight for Event (4172)

A new parameter is added for LCD Backlight for Event Indication. This parameter is configurable in common config file and telnet.

BLBlinkMode = 0 (empty : Default, 1: MWI, 2: Incoming call, 4: Miss Call, 7: all)

// MWI: (LCD Backlight flashing when VMWI in idle mode)

// Incoming: (LCD Backlight flashing when incoming call)

// Miss Call (LCD Backlight flashing when miss call)

Telnet CLI:

lcdui set BLBlinkMode 0 to 7

Remark:

1. MWI: when receive new message, it will blink as MWI HK. If C hard key is pressed, it will be back to the idle mode and not blink any more.
2. Incoming call: LCD will blink as function key. If the call is pick up or rejected. The back light will not blinking.
3. The blinking frequency is the same as indication LED for MWI and Incoming call. For the miss call, BL is on when "Have XX missed calls" is displayed. BL is off when the message is not displayed.



6 Never Hangup (4319)

One of our customers requests the following features:

1. Only the administrator of the phones has the right to activate or dis-activate the function
2. When Activated, the function :
 1. does not allow any Call Center operator to hang up and finish the call with the calling party (as a consequence, the Call Center Operator is obliged to stay on line with the calling party until the time that the calling party has hanged up to terminate the call).
 2. has to work on the various possible ways that are normally used by Call Center operators to hang-up and terminate the call (from the head set, from the handset or from the key pad)
 3. If there is an incoming call, **the user should NOT have the possibility to REJECT** a call. The REJECT functionality should be disabled for users.

The user should be able only to answer to the phone and transfer the call to a specific Supervisor. A new parameter is added for never hangup for incoming call. This parameter is configurable in common config file and telnet.

NeverHangup = 0/1 (0: Off Default)

// 0: off (Original)

// 1: on (Never Hangup On)

Telnet CLI:

lcdui set NeverHangup 0/1

Remark:

- It is only for incoming call.
- The only way to end the call is by the calling party.
- When it is handset mode, onhook will go to handsfree mode.
- When it is headset mode, press the headset button will have no response.
- When it is handsfree mode, press the handsfree button will have no response.
- EndCal soft key: no response if it is a incoming call
- Function key: no response to press function key to end the call.
- The Telnet control should be the same response as key press.
- The way to hangup is by the SIP BYE message.
- When NeverHangup is on, the REJECT functionality should be disabled.



7 Pause in Speed Dial (4724)

One of our customer request to add a pause for a speed dial keys. Purpose is to have one key to dial out a number and then send after a pause a DTMF number.

“|” symbol is added as a special char for the speed dial to add a pause.

So the behavior is:

1. We suppose phone config FK as 1234|5678 for speed dial.
 2. Phone dial out 1234
 3. Until remote party(1234) answer the call, phone A will send 5678 in DTMF mode after TTime.
- A new parameter is added for this. This parameter is configurable in common config file and telnet.

TTime = 0 – 10 s (1s: Default)

Telnet CLI:

lcdui set TTime 0-10

Remark:

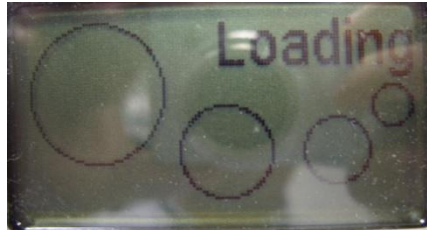
- If user dial digit key himself before this pause time, it will send the DTMF immediately.
- | symbol can be input by digit 1 in ABC/abc mode.
- TTime can only be changed by config file or telnet command by administrator.
- Speed dial can be config by phone or Web GUI.



8 Display customer icon when boot (4332)

We have a customer that is requesting a new feature on TB30: Display on his Logo icon when boot.

Current software displays Black screen. Replace by bubble logo icon.



A new parameter is added for APS.

boot_logo_bmp=xxxx.bmp (B/W bitmap file only and bitmap will be inverted)

Icon Size = 240 x 120

To clean the icon:

boot_logo_bmp= <empty>

Remark: After reset to default, it will not clear the icon.

The upgrade behavior is same as language table. It will include in XXX.inf file. Format as below:

XXX.inf with tftp

Tone_table=xxxx

Language_table=xxxx

boot_logo_bmp=xxxx.bmp

xxx.inf with http

tone_table=http://path/filename

language_table=http://path/filename

boot_logo_bmp= http://path/xxxx.bmp

9 Block USB APS (4326)

Disable USB APS. Only Telnet, or other APS need to work fine (no Web GUI need).

A new parameter is added for this feature:

NoUSBUpgrade= 0/1 (default: 0)

// 0: USB upgrade is enabled even the APS is disabled.

// 1: No upgrade possible by USB.

Telnet CLI:

autoprovision set NoUSBUpgrade 0/1

10 Event Header(5067)

One of our customer request to a change for the Subscription and Notifications.

It needs to be disabled as it causes inter-op issues with customer requirement

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

B_suppress_evt = 0/1 (0: default)

// 0: current behavior

// 1: Event header value different shall answer based on the following table

Event Header Value	V1.73	V1.73.1	B_suppress_evt = 0	B_suppress_evt = 1
dialog	200	200	200	200
message-summary	200	404	404	404
gateway-upgrade	200	200	200	200
check-sync	200	200	200	200
talk	200	200	200	200
hold	200	200	200	200
conference	200	200	200	200
Reg	200	200	200	200
call-info	200	200	200	200
line-seize	200	200	200	200
missed-call-summary	200	200	200	200
presence	200	200	200	489
ua-profile	200	200	200	200
Winfo	400	489	489	489

All other event which not known by phone will response 489.



11 Downloadable Ring Tone (4328)

One of our customer request to implement Melody plays with 711.

Requirement

A new parameter is added for APS.

melody =xxxx.bin (711 binary file)

File Size < 64k

To clean the icon:

melody= <empty>

Remark: After reset to default, it will not clear the melody.

A new parameter is added for melody on/off. This parameter is configurable in common config file and telnet.

PolyMelody = 0/1 (0: Off Default)

// 0: off (Original)

// 1: on (using downloaded melody)

Remark: if there is no melody downloaded, even it is on; it will be the same as off.



12 Conference Not Allow NW (4862)

One of our customer request to implement Conference Not Allowed NW

Prerequisites:

Tested CPE is RFC-4579 compliant.

Through GUI create a line with line template having Call Conference feature deactivated.

Configuration:

On SAU, in cpe specific ini file set the parameter [sip].holdTrfRight=1

1. A (Tested CPE) calls B. B answers the Call.
2. Press Hold Button on phone A to put B on hold.
3. Dial C from A
4. Pick up C.
5. On phone A press the Conference Button.

Expected result:

1. Check that A and B are connected and hear each other.
2. Check that A hears dial tone and B is on hold (hears hold music).
3. Phone C must ring and A hears ring back tone.
4. A and C are in active call. B is still on hold.
- 5a. Check that the REFER request sent by CPE A, is rejected with a 403 (Forbidden) response by the network.
- 5b. Ensure that as the REFER transaction fails (the final response is not 2xx), the CPE A returned back to the previous state (A is in active call with C, B is still on hold).

The right behavior is:

- The INVITE request could be rejected with a 403 (Forbidden) response by the network if the user is not allowed to perform Call Conferencing. In all cases when the INVITE transaction fails (the final response is not 2xx), the CPE MUST return back to the previous state.
- The REFER request could be rejected with a 603 (Decline) response by the network if the user reaches the maximum number of participants in a conference. In all cases when the REFER transaction fails (the final response is not 2xx), the CPE MUST return back to the previous state.



13 Group Pickup Key (4794)

One of our customer request to add a function Group Pickup Key

They want to have a soft-key like for direct pickup and have a way to configure the service code (*8#) like for direct pickup.

A new service code is added for auto pickup

AutoPkupNum =

(the length is max to 6)

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

Local_Pickup_type: 0/1/2

// 0: Manual pickup (default) it is current pick up. User need to enter the number they want to pick up.

// 1: Auto pickup: config before calling. And no need to enter the number when pick up the call.

// 2: both pick up exist

If **Local_Pickup_type=2**, then pressing pick up soft key, it will enter another page and there'll be 2 choices:

If AutoPkupNum is empty means original pick up function.

1. M_pick =>original pick up function
2. A_pick =>auto pick up function

User can choose the one they like to do pick up feature.

In WEB GUI. We will have 3 choose :

0: Manual pickup; no configuration for AutoPkupNum

1: auto pickup with configuration for AutoPkupNum

2: both pick up exist with configuration for AutoPkupNum

Another new parameter is added for pickup. This parameter is configurable in common config file and telnet.

Auto_pickup_Method=0: SUBSCRIBE will be used

//0: SUBSCRIBE will be used.

//1: the number will be sendout with INVITE method



14 VPN Escape Code For Per Line For SF and HG(4907)

One of our customer need to have the possibility to configure the service code per line basis for at least Secretarial Filtering and Hunting Group.

The reason to have a per line service code configuration is the following:

On our PF we can configure a VPN escape code “y” for a line “xxx”. In this case the subscription must be done with VPN escape code “yxxx” but the activation/deactivation of SF or HG, so service code, must be done without VPN escape code “xxx”.

For example with Secretarial Filtering supervision, if the manager has a VPN escape code “9” and line number “222”, on assistant the subscription must be done with “9222” and service codes must be *270222* for activation and *271222* for deactivation.

They want to Propose :

1. To keep the global service code configuration
2. According to service supervision selected, add the possibility to configure the service code per line key and this service code will surcharge the global service code configuration

A new parameter is added for Escape code.

This parameter is configurable in common config file and telnet.

VPNEscCode = 0/1 (0: default)

// 0: Disable

// 1: Enable VPN Escape Code

In Config files, the following items are added for Escape code.

VPNesp1=x

VPNesp2=x

...

VPNesp66=x

The length of the Escape Code is 1.

A Text Box is added in the Supervised Line for input for the escape code when VPNEscCode = 1.

TB30(\$)				
FK	Type	Destination	BLF Option	
F 1	Line		☐ dialog ☐ regDND ☐ ua-profile	Detail
F 2	Line		☐ dialog ☐ regDND ☐ ua-profile	Detail
F 3	Line		☐ dialog ☐ regDND ☐ ua-profile	Detail
F 4	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail
F 5	Supervised Line	9 1000041002	☒ dialog ☐ regDND ☐ ua-profile	Detail
F 6	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail
F 7	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail
F 8	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail
F 9	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail
F10	Line		☒ dialog ☐ regDND ☐ ua-profile	Detail

15 Call Join (4458)

One of our customer need to have one function Call Join .

A new parameter is added for call join. This parameter is configurable in common config file and telnet.

CallJoin = 0/1 (0: Default)

// **0:** (without call join – Original)

A create a conference with B and C, when A hang up, it will send BYE to B, C, then A, B, C return to idle.

// **1:** (with call join)

A create a conference with B and C, when A hang up, A transfer B to C, then after A return to idle, B and C still keep talking.

16 DNS Query (4803)

One of our customer requests the following feature

They uses some high availability system . It consists of one Call Server (CS) and one Passive Call Server (PCS). The PCS takes the control if the CS is down. To do this, they configure the DNS so when somebody asks for the CS IP address, the DNS server responds with 2 IP addresses: one of the CS and one of the PCS.

With both TB30 and ST2030, the scenario is falling and the phones get blocked sending REGISTER to the PCS although it is responding with 503.

The test result is as follows:

When the phone is started, it does a DNS query and it stores both IP addresses. It is registered to the CS. Then the CS is stopped so the PCS starts working. While it starts there is no service available at all. Then the phones sends the REGISTERs and get registered to the PCS when this is ready. Then they do an out-coming call. The phone sends several INVITE to the CS (it shouldn't as it is now registered to the PCS), but as this is KO, it finally sends it to to the PCS and the call is OK.

Then they have an incoming call, and it is OK as it is registered correctly.

Then, the CS is up again but the phone insists on going to the PCS (it answers with a 503, because PCS is now down) but it doesn't get registered to the CS so the phone gets out of service and it is needed a manual reboot

Requirement:

When the server has no response or get 503, phone will move to the second one.





17 SIP Scenario of Alternate Call with Notify Event (4311)

One of our customers requires the following feature:

SIP Scenario of Alternate Call with Notify Event=talk

A new parameter is added for alternative call. This parameter is configurable in common config file and telnet.

TIAlternativeCall = 0/1 (0: Off Default)

// 0: off (Original)

// 1: on (TI Alternative Call handling)

Remark for TIAlternativeCall = 1:

- Do not send the INVITE with sendonly. If it has been already put in inactive means phone already hold that line.
- INVITE with sendonly only in the event in which an INVITE with inactive does not have precedence received on that same one leg.
- Attached here is trace for reference.



18 Default Language Table (4964)

One of our customer complains some country/customer there is 2 to 3 official language. There is a need for those customers to be able have 3 languages.

Requirement

Two more downloadable language tables are added.

default_lang_table is replaced by three new parameters for APS.

dft_lang_table1=

dft_lang_table2=

dft_lang_table3=

Telnet Command is changed as:

```
> tftp2 dftlangtable index[1~3] X.X.X.X filename
```

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

RestoreLanguageType = x|x|x (x: 0~8, default: 0)

Telnet Command:

```
> lcdui show restore_lang
```

```
> lcdui set restore_lang content [0|0|0 ~ 8|8|8]
```

Remark:

If the language table is not exist (not download, it will have no effect).

Any setting of this feature is effective after rebooting

Default language table will be cleaned after reset default

19 Carries All Codec When Transfer is processed by server (4957)

One of our customers want to improve the following problem:

When unhold a call, it provides an offer with all codec it supports not only the one codec with which phone negotiated successfully before.

Requirement:

A new parameter is added for unhold problem. This parameter is configurable in common config file and telnet.

UnholdCodec = 0/1 (0: negotiated codec only)

```
// 0: one codec (Original - negotiated codec only)
```

```
// 1: all codec (all support codec)
```





20 LCD Backlight for Event Indication(4172)

LCD Backlight for Event Indication.

A new parameter is added for LCD Backlight for Event Indication. This parameter is configurable in common config file and telnet.

BLBlinkMode = 0 (empty : Default, 1: MWI, 2: Incoming call, 4: Miss Call, 8: Not Register, 15: all)

//MWI: (LCD Backlight flashing when VMWI in idle mode)

//Incoming: (LCD Backlight flashing when incoming call)

//Miss Call (LCD Backlight flashing when miss call)

//Not Register (LCD Backlight and F1 to F10 flashing when not register and Out of Service will display with phone number alternatively in idle)

Another new parameter is added for LCD Backlight for Event Indication. This parameter is configurable in common config file and telnet.

BLBlinkTime = 0 – 3600 seconds (0: default)

If Option ON, Blinking stop after Timer for MWI and Miss Call. If it is 0, it will not stop.

Remark:

1. MWI: when receive new message, it will blink as MWI HK. If C hard key is pressed, it will be back to the idle mode and not blink any more.
2. Not Register: If C hard key is pressed, LCD and F1 to F10 will not blink any more.
3. Incoming call: LCD will blink as function key. If the call is pick up or rejected. The back light will not blinking.
4. The blinking frequency is the same as indication LED for MWI and Incoming call. For the miss call, BL is on when “Have XX missed calls” is displayed. BL is off when the message is not displayed.



21 Improve Key Detection (5064)

One of our customers complaining about 2nd dialed digit is missed intermittently when dialing quickly on the keypad (519 returns 59, 656 returns 66 on display, etc)

Current behavior: (1.72.0) Display 1 when 2 is released.

Request Behavior:

Display 1 & 2 (1 after xx ms press, then 2 after 1 is release & 2 is still hold for xx ms)

Sequential Double key Press: (A is detect before B is press)

- If key A is press then Key B is press: Display show A & B

Sequential key Press: (A is detect before B is press, B is detect before C is press, C is detect before D is press)

- Display show ABCD...

If key A is pressed, then key B is pressed. And key B release before key A (no matter the time key B pressed), in this case, key B will be ignore. Display only show A.

22 Remove DHCP Option 60 (5063)

One of our customers request us to provide a way to remove the option 60 for DHCP DISCOVER OPTION.

A new parameter is added for remove option 60. This parameter is configurable in common config file and telnet.

DHCP_Opt60 = 0/1 (1: default)

// 0: off (remove option 60 in DHCP and ignore the option 43 in DHCP offer)

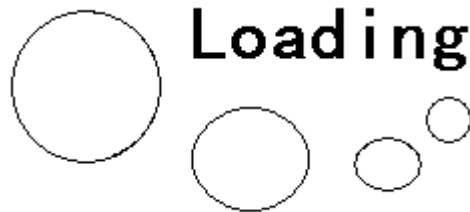
// 1: on (current behavior)



23 Logo Icon Display (5085)

One of our customer requires the following feature about booting icon:

Default booting icon should not show any TB30, Thomson or Technicolor wording or Icon. Need to remove it.



24 Default value for Common Config File (xxxx)

One of our customers requires the following feature about Default value for common config file

For the current implementation of APS mechanism, if the value is empty of some parameter, we don't change anything, for example, if you set "CWMPACSUser=" in ComConfTB30S_100204.txt, it will follow the setting you set before, so if you want to those five parameters for blank in this version, you must reset to default to reach them.

So it is no way to set the parameter to empty.

As a result two keywords <default> and <empty> are defined.

CWMPACSUser=<default> will set to default value after APS.

CWMPACSUser=<empty> will set to empty value after APS.

Remark: <empty> is only apply for string parameter.



25 Full Duplex Handfree(?)

2.74 have improved performance for hand-free audio thanks to the new DSP R12 for TI.

The release include 2 Telconf file One for TIA requirement, One for ETSI.

26 USB Headset Support (4331)

Jabra & Plantronics USB headsets are supported:

Model	Picture	Wired/Wireless
Jabra BIZ 620		Wired
Jabra GN2000		Wired
Plantronics C610		Wired
Jabra GN9350e		Wireless. Can be connected to USB or RJ45 of TB30.
Plantronics HW121N		Wired

2.74 can be use with Jabra GO660 (USB Bluetooth Wireless) for demo.

There is a small incompatibility after pressing the Jabra headset button to release the line, release time is more that few second. Occure 20% of the time.

Attention:

1. Phone can detect USB headset in standby only.
2. When USB headset is detected, TB30 will ignore the original headset.
3. For the USB Headset volume cannot be regulated by the volume key on the key on TB30.
4. If USB headset is in use, there`s no group listening mode
5. If the USB Headset is unplugged when on talk mode, TB30 will go on hook automatically.



27 Call transfer is not available in double (4896)

1.Introduction:

Once two calls have been established, it has the possibility to make a transfer from the first call to the second.

2. Requirement :

2.2 When in conference, a new soft key [Quit] is added.

When pressing this soft-key, it will exit from the conference and transfer the first call to the second.

Date	Hour	Icons
01 Tel.nr		
Conf	Time	
02 Tel.nr		
Conf	Time	
Hold CallLog	Quit PhBook NewCall	Remove

If the user put the handset on hook, the conference will be closed and both lines will be released.

2.3 When one call is on hold and a second is established, a new soft key [CalTran] is added.

2.3.1 Talking line is selected. Same as original behavior

Date	Hour	Icons
01 Tel.nr		
Talking	Time	
02 Tel.nr		
Hold	Time	
Hold CallLog Park	Transf PhBook NewCall	Conf EndCall

2.3.2 Hold line is selected, the soft key “CalTran” will appear in the place of original “Transf”

Date	Hour	Icons
01 Tel.nr		
Talking	Time	
02 Tel.nr		
Hold	Time	
UnHold CallLog Park	CalTran PhBook NewCall	Conf EndCall Join



When pressing this soft-key, it will transfer the first call to the second.

Another soft key [Quit] is added. When pressing this soft key the conference is closed and both calls are terminated.

This feature only exists for two lines.

28 Direct DND (5086)

1. Introduction:

This is an improvement in the MMI to shorten the access to the DND feature

2. Requirement

Only one key press enables the DND

When the same key is pressed again DND off.

Current Situation

Press Option

Navigate down to “DoNotDisturb – Off” (completely down in the list)

Press Select

Press Edit

Navigate up / down (to change on / off)

Press Ok

Press Back

Press Back

Situation requested with product change

Press Option

Navigate to “ DoNotDisturb – Off”

(Please put “ DoNotDisturb – Off” at the first place in the list, and move DialSubscribe down to the end of the list)

Press Change (to change on/off)

29 Supervisor listening on handset, while operator (5184)

1. Introduction

Telecom Italia wants to connect the headset in the headset plug and use the handset in order to do supervising. The operator will use the headset and the supervisor want to come and listen the call with the handset.

Scenario:

1. Operator is talking with headset on dedicated headset port
2. Supervisor is beside the agent and picks up the handset of the phone to listen to the call with the handset

2. Requirement

A new configuration parameter should added for this. This parameter is configurable in common config file and telnet.

SupervisorListening = 0/1 (0: Off Default)

// 0: off (Original)

// 1: on (HeadsetSupervisorMode enabled)

Scenario if the parameter is enabled:

// 1: on

1. Talking with headset
2. Pick up the handset, user will hear the voice from remote part but cannot hear the voice from headset. **The voice from handset will NOT be send out.**
3. When the flag is on, original headset group listening function will be disabled

To switch microphone used (on the headset or the handset), the supervisor can switch with the menu and cancel keys between using microphone on the headset or microphone on the handset.

Supervision listening OFF

Handset	Handsfree	Headset	Headset-GL	Headset-GL
---------	-----------	---------	------------	------------



Handset		1. Press Speaker key 2. Replace handset	Press Headset key	Press Speaker key	N/A
Handsfree	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	

Supervision listening ON

	Handset	Handsfree	Headset	Handset -GL	Headset-GL	Supervisor-GL
Handset		1.Press Speaker key 2.Replace handset	Press Headset key	Press Speaker key	N/A	N/A
Handsfree	Pick up handset		Press Headset key	N/A	N/A	N/A
Headset	N/A	1.Press Speaker key 2.Press headset key		N/A	Press Speaker key	Pick up handset
Handset -GL	Press Speaker key	Replace handset	N/A		Press headset key	N/A
Headset-GL	N/A	Press headset key	Press Speaker key	Pick up Handset		N/A
Supervisor-GL	Press headset key	N/A	Hang up Handset Menu key: voice from the handset Cancel key: voice from the headset	Press speaker key when voice from handset	Press speaker key when voice from headset	

30 The client transmits the End packets only once whereas it should be transmitted three times (5159)

1. Introduction

The terminal transmits the End packets only once whereas it should to be transmitted three times.

This issue is visible with Thomson ST2030 v2.67 as well as Thomson TB30 v1.73.

The RFC suggests that the final RTP event packet for each event (which will have the "E" bit set) should be transmitted three times to help prevent implementations which rely on the "E" bit to stop tone playout from extending the tone in the event of packet loss.

2. Requirement

Add a configure option for timing with 10ms as the unit.

The parameter name is "EventEndRetransmit"

The value could be set to something between 0 and 100 ($x \times 10\text{ms}$)

By default the parameter's value is 0.

EventEndRetransmit=0 : for original behavior

EventEndRetransmit=x : $x \times 10\text{ms}$

If x is different from 0, RTP Event (DTMF) end packet will be retransmit 3 times with the period of the value.

For instance

$x=1 \Rightarrow$ RTP Event (DTMF) end packet will be retransmit 3 times with the period of 10ms

$x=2 \Rightarrow$ RTP Event (DTMF) end packet will be retransmit 3 times with the period of 20ms

$x=3 \Rightarrow$ RTP Event (DTMF) end packet will be retransmit 3 times with the period of 30ms

etc...

31 Unregister phone ability to call another unregistered phone directly (5195)

1. Introduction

This is a new feature request, phone ability to make a call even when not registered on SIP server. (Peer to Peer ,without server)

2. Requirement

Use case :

(inbuilding) emergency calls to a control room,

call functionality in case of SIP server down,

For a TB30 who does not have a SIP registered, the ability to make a call.

Example:



Pressing function key which has a "phonenumber@domainname" programmed behind it.

32 BLF improvement: display the caller ID (5205)

1. Introduction

Currently, for BLF feature, only the function key is blinking when the supervised line have an incoming call. The customer wants to see the name or the number of the caller on the screen before launching the call pick-up process.

2. Requirement

In the context of a Broadsoft server, the Caller ID information are carried in the NOTIFY message's xml body. It should also work with the peer to peer BLF.

When the supervised line is called, the corresponding Function Key on the supervisor phone starts blinking. It can happen that several lines are blinking at the same time.

If the user presses a blinking function key, the phone should display the corresponding caller number and/or name as if it was an incoming call.

Date	Hour	Icons
01 Name		
Number		
Pick-up		Back

- If the user press again the function key, or press "Pick-up" the phone should start the call pick-up process. The pick-up process will be either local or star code based depending on the pick-up setting.
- If the user presses "Back" or C key, the phone will return to the previous screen. Either the idle screen or conversation or else.
- If the user doesn't presses any button, after a 5 seconds timeout the phone will return to the previous screen.

A new parameter need to be added:

BLFcheckID=0/1



0 = don't check the caller ID. Same as current behavior. It will be the default setting

1= caller ID presentation on 1st press and call pick-up on the 2nd as described above.

33 Universal pickup behavior-Update from PCR 4794 (5347)

1. Introduction

One of our customers request to change the Call Pickup behavior, so we will add the feature for universal pickup. The purpose is satisfied to most all the request for pickup service for all customers

2. Requirement

- Note:
1. Configure is mainly used for server side.
 2. The PCR will be as an update base on PCR_4794

Add an config to enable/disable this feature

U_pickup: 0: default, current implementation
 1: enable pickup feature for below implementation.

Add Another 4 parameters to config Pickup type by user themselves. Default as below

Pickup1=

Pickup2=

Pickup3=

Pickup4=

The format to setting for this parameter is **Pickup_Name|Pickup_Method|Pickup_Num.** (length<63) The definition as below:

Pickup_Name:

It is empty for default. User can define a specific name for the Pick up service themselves. Such as Direct_Pickup, Group_Pickup or any name they like. The length limitation for TB30: length<26;. The overflow part for the name doesn't display on LCD.

Pickup_Method: 0: INVITE by default
 1: SUBSCRIBE

The value means the method that pickup feature used. Currently, we have 2 methods for pickup service INVITE&SUBSCRIBE. INVITE is the default one.



Pickup_Num: x by default

The value should be the number user want to do pick up. The format for the value is combined with 0-9,*,#, x. phone number in **Pickup_Num** represented by x. e.g. **Pickup_Num** = *#x#, where x is the phone number that user input. E.g. *#20001#

Example:

1. Example for for Comverse:

Comverse wants both local pick up and group pick up exist. So we can config here:

Pickup1=Local_Pickup|1|x

Pickup2=Group_Pickup|0|*8#

Pickup1 means, the pickup name is Local_Pickup, it use SUBSCRIBE method for pickup service. It needs user to enter the pickup number.

Pickup2 means, the pickup name is Group_Pickup, it use INVITE method for pickup service. It will send out INVITE with *8# without any input by user.

Because there`re 2 type of Pickup, when you press Pickup SK, the display on LCD will as below

a

Date	Hour	Icons
name		
Phone number		
	CallLog	PickUp

b

Date	Hour	Icons
Local_Pickup		
Group_Pickup		
Select		Back

2. Other Examples:

- 1) only Pickup2 is configed: **Pickup2=pickup1 |0|x**

a

Date	Hour	Icons
name		
Phone number		
	CallLog	PickUp

Enter PickUp SK to do pickup directly



b

Date	Hour	Icons
Enter a number		
PhBook	CallLog	Back

2) **Pickup1=pickup1|0|x**

Pickup2=pickup2|0|x

Pickup4=pickup4|0|x

a

Date	Hour	Icons
name		
Phone number		
	CallLog	PickUp

After press PickUp SK, it will enter page b to list all pickup type

b

Date	Hour	Icons
pickup1		
Pickup2		
Pickup4		
Select		Back

3) And if there is empty in Pickup_Name it will not display anything on LCD and this type of pick up is not activated.

a

Date	Hour	Icons
name		
Phone number		
	CallLog	



34 Support writing from right to left for Hebrew (5188)

1. Introduction

In current implementation only the left to right display and entry modes are supported.

In Hebrew the writing is from right to left, combined with right alignment.

2. Requirement

There is no need for a new parameter, as it will be handled with the charset:

<Charset>Hebrew</Charset>

This will trigger right alignment and right-to-left input mode when selected.

Text Display

When right to left mode is selected, all the existing sentences and words (either embedded in the firmware or uploaded from a language file) that were displayed on the screen justified left, will be justified right.

When Charset is not Hebrew

Date	Hour	Icons
DialSubscribe-OFF		
CallBlocking-OFF		
CallForward-OFF		
CallWaiting-ON		
Select		Back

When Charset is Hebrew

Date	Hour	Icons
DialSubscribe-OFF		
CallBlocking-OFF		
CallForward-OFF		
CallWaiting-ON		
Select		Back

For the longer strings that require **scrolling**, the scrolling direction must be reversed as well.

Text Entry

In text entry mode, for instance adding an entry in the local phonebook, the cursor must be placed on the right.

Date	Hour	Name
<Null>		
<Null>		
Default		
Edit	Save	Cancel



Press **Edit**

Date	Hour	Name
		<Null>
		Default
Edit	Save	Cancel

And the characters will be inserted from left to right.

Text Entry for numbers

For numbers the order will change again to normal left to right. This mean when doing a sequence of Hebrew characters and numbers, the order of input can change on the fly.

This can be seen on the SNOM 300 phone example provided.

This can be seen on the Hebrew website example provided. www.hebrewbooks.org

Text Deletion for sequence of Hebrew characters and numbers

When doing “backspace” or deletion of characters in a sequence where both number and Hebrew characters are present, the order of deletion needs to respect the order of input. This also implies a change of deletion order on the fly, when the cursor goes over from deleting characters to deleting numbers and vice versa.

Notes

Add another input method indication (ABC,abc,123,**Hebrew**)

Picture from board

Language File :

<charset>Hebrew

<key1L>

<key1R>

<key1H>

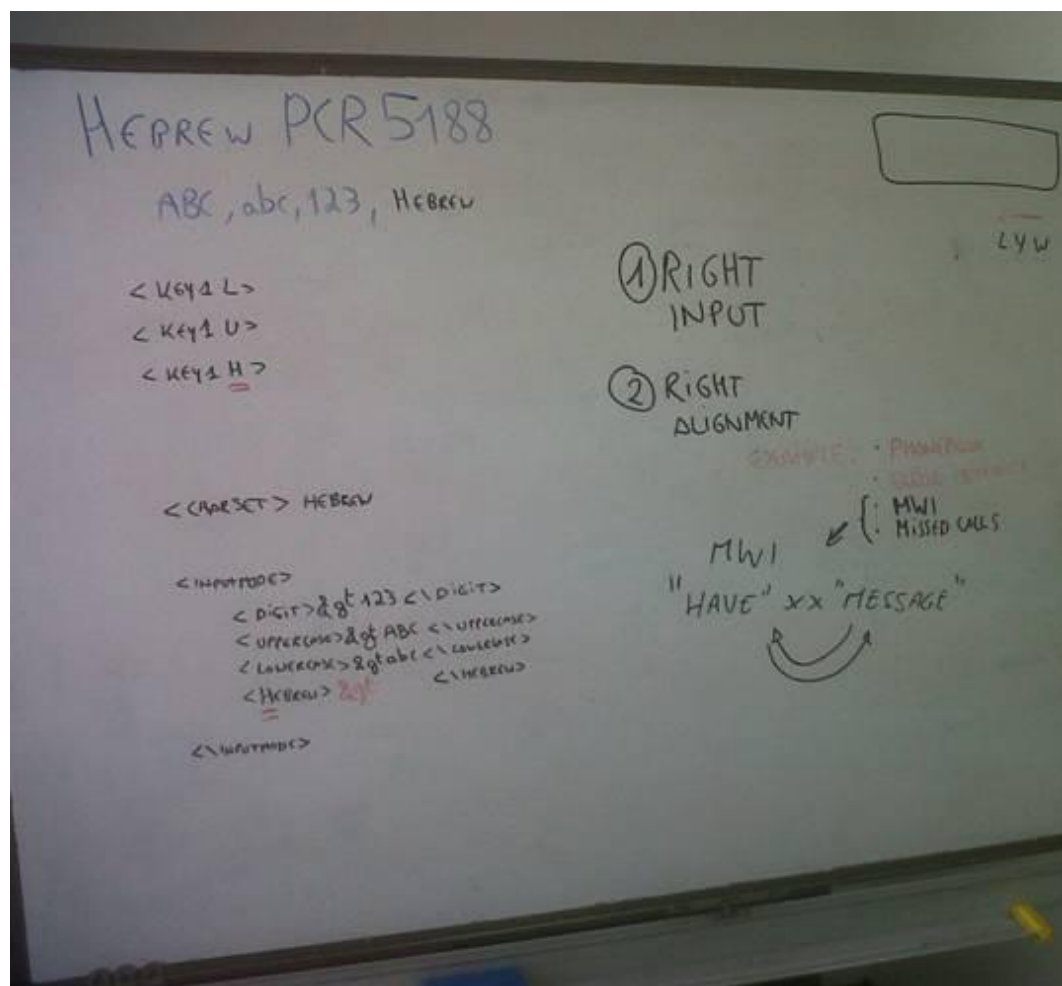


```
<Input Mode>
<digit>&gt;123<\digit>
<uppercase>&gt;ABC<\uppercase>
<lowercase>&gt;abc<\lowercase>
<Hebrew>&gt;XXX<\Hebrew>
<Input Mode>
```

Attention

The dynamic language table is used for Hebrew language, Hebrew is not implemented as standard language.
The PCR involves 2 items :

1. Right input
2. Right Alignment for User interface :
 - a. Phonebook
 - b. Error Message
 - c. MWI message
 - d. Missed Calls
 - e. Menu options
 - f. Softkey



35 Support package as-feature-event (5187)

1. Introduction

One of our customers request that TB30 supports the package « as-feature-event »

2.Requirement

There`re 5 services on spec: **DND CFA, CFB, CFNA & ACD.**

Need to use a global enable/disable flag to change the behavior of the phone between current Star Codes / local and Broadsoft like.

Add a config to enable/disable this feature



as-feature-event:

0: disable , default, current implementation

1: enable , package as-feature-event feature for below implementation.

1) DND:

Set Do Not Disturb

The supported fields are:

device – This is a mandatory field in the XML schema, but it is not used by BroadWorks. The phone can set this to any value.

doNotDisturbOn – To activate Do Not Disturb, set to “true”. To deactivate Do Not Disturb, set to “false”.

2) Call Forward

Set Forwarding

The supported fields are:

device – This is a mandatory field in the XML schema, but it is not used by BroadWorks. The phone can set this to any value.

forwardingType – BroadWorks only supports the following forward types

forwardImmediate – To set the Call Forwarding Always data on BroadWorks

forwardBusy – To set the Call Forwarding Busy data on BroadWorks

forwardNoAns – To set the Call Forwarding No Answer data on BroadWorks

Note: In order to control ringCount, there’s no need to add a flag. The CFNA time is setting with UI.

3) ACD

SetAgentState

The supported fields are:

device – This is a mandatory field in the XML schema, but it is not used by BroadWorks. The phone can set this to any value.

requestedAgentState – Use one of the following values:

loggedOn: Agent in the *Sign-In ACD* state

loggedOff: Agent in the *Sign-Out ACD* state

notReady: Agent in the *Unavailable ACD* state

ready: Agent in the *Available ACD* state

workingAfterCall: Agent in the *Wrap-Up ACD* state

Note: *agented/password/group/extensions are not mandatory request*

Original ACD, call forwarding and DND setting in web GUI will be ignored if the flag enabled.

36 Support Hebrew Right sorting & searching of the phonebook(5188)

1. Introduction

In current implementation only the left to right display and entry modes are supported.

In Hebrew the writing is from right to left, combined with right alignment.

2.Requirement

There is no need for a new parameter, as it will be handled with the charset:

<Charset>Hebrew</Charset>

This will trigger right alignment and right-to-left input mode when selected.

Text Display

When right to left mode is selected, all the existing sentences and words (either embedded in the firmware or uploaded from a language file) that were displayed on the screen justified left, will be justified right.

When Charset is not Hebrew

Date	Hour	Icons
DialSubscribe-OFF		
CallBlocking-OFF		
CallForward-OFF		
CallWaiting-ON		
Select		Back

When Charset is Hebrew

Date	Hour	Icons
DialSubscribe-OFF		
CallBlocking-OFF		
CallForward-OFF		
CallWaiting-ON		
Select		Back

For the longer strings that require **scrolling**, the scrolling direction must be reversed as well.

Text Entry

In text entry mode, for instance adding an entry in the local phonebook, the cursor must be placed on the right.

Date	Hour	Name
<Null>		
<Null>		
Default		
Edit	Save	Cancel

Press **Edit**



Date	Hour	Name
		<Null>
		Default
Edit	Save	Cancel

And the characters will be inserted from left to right.

Text Entry for numbers

For numbers the order will change again to normal left to right. This mean when doing a sequence of Hebrew characters and numbers, the order of input can change on the fly.

This can be seen on the SNOM 300 phone example provided.

This can be seen on the Hebrew website example provided. www.hebrewbooks.org

Text Deletion for sequence of Hebrew characters and numbers

When doing “backspace” or deletion of characters in a sequence where both number and Hebrew characters are present, the order of deletion needs to respect the order of input. This also implies a change of deletion order on the fly, when the cursor goes over from deleting characters to deleting numbers and vice versa.

Notes

Add another input method indication (ABC,abc,123,**Hebrew**)

Picture from board

Language File :

<charset>Hebrew

<key1L>

<key1R>

<key1H>

<Input Mode>

<digit>>123<\digit>

<uppercase>>ABC<\uppercase>

<lowercase>>abc<\lowercase>

<Hebrew>>XXX<\Hebrew>

<Input Mode>

Attention

The dynamic language table is used for Hebrew language, Hebrew is not implemented as standard language.

The PCR involves 2 items :

1. Right input
2. Right Alignment for User interface :
 - a. Phonebook
 - b. Error Message
 - c. MWI message
 - d. Missed Calls
 - e. Menu options
 - f. Softkey

3. Right sorting of the phonebook :

If the character is a Hebrew character, then the sorting follows the Hebrew alphabetical order

if the character is not a Hebrew character, but standard alphabet, then it is sorting *after the last alphabetical character in Hebrew*.

Please verify with UTF-8 for Hebrew. <http://www.alanwood.net/unicode/hebrew.html>

4. Hebrew search of the phonebook :

The search function of the phonebook needs to support searching in Hebrew.

37 HTTPS type V1 (update PCR5341)

1. Introduction

One of our customers request to make an update of about https support for our devices, we need supported Certificate Authority (CA) in double-way authentication , and it must verify if the server is trusted .

2.Requirement

Starting from the new release, both HTTPS implementations should be put in a version and a parameter “**HttpsType**” is introduced for users to opt for. This parameter is configurable in common config file and telnet.

[net]

HttpsType = 0/1

// 0: **“without authentication”** -> phone will not authenticate the server, by default

// 1: **“with authentication”** -> phone will authenticate the server ,

Remark:

without authentication -> changed the description from previous “One way authentication” --- phone will not authenticate the server .The behavior has already been implemented in x.66.2,so let’s keep as it is.

with authentication -> changed the description from previous “Double authentication”

The current phone just look if there is a CA or not on the server but not matter if it is “trusted” or not. So for the case, we need to make the phone be able to support authenticate server and consider the server must as “trusted”, this is change request.

38 USB Stick Debug Log (5369)

1. Introduction

In general , we capture some debug log or trace via serial port, our customers request to add a new feature about USB stick debug log .

2.Requirement

Telnet command: sys set USBlog on/off

(the command no use when USB stick doesn’t plug in.)

Start USB log:

Plug in USB stick first and then enable the config.

Stop USB log:

1. Plug out the USB stick
2. Disable USBlog flag
3. Cold or warm restart



39 Extend WebGui Session Timeout (5368)

1. Introduction

One of our customers want to extend the WebGui Session Timeout.

This WebGui Session Timeout is at the moment fixed for approximately 120 seconds.

Goal is to extend this Webgui Session Timeout upto 3600 seconds (1 hour).

2.Requirement

A new configuration parameter should added for this. This parameter is configurable in common config file and telnet.

webtimeout = 120 - 3600 [seconds] (120: Default)

40 The call is not automatically released if the phone doesn't receive any rtp flow (5365)

1. Introduction

One of our customers raised a bug about “The call is not automatically released if the phone does not receive any RTP flow. The config file does not contain a RTP timeout parameter. See OGDR. 30 seconds should be a good default value”.

2.Requirement

A new configuration parameter should be added for this. This parameter is configurable in common config file and telnet.

rtptimeout= 0 - 120 [seconds] (0: Default)

The default value “ 0 “ is stand for disable any other value means the timeout for auto release.

Note (20110831):

If the rtptimeout is setting , the local phone will not send RTP packet and will release the call in the following two cases:

- 1) The remote party select Silence Suppression option (in ADVANCED page) and stop speaking for a period of time ;
- 2) The remote party muted is enabled ;

Moreover, if rtptimeout is setting. Never Hang Up feature (PCR4319) will become to disabled even if Never Hang Up is enabled.

41 Support Contact Header information included in the BYE message (5376)

1. Introduction

One of our customers request to add “Contact Header” information in BYE message, according to RFC 3261 (page 162) , it isn’t a requirement in RFC, but some soft-clients and other devices seem to put the ” Contact Header” in the BYE message, so we will consider if it can be implemented or not.

2.Requirement

The new feature “contact header information included into BYE message” need to starting from the new release. This parameter is configurable in common config file and telnet.

[sip]

ContactinBye= 0/1 (default : 0)

// 0: set as disabled -> It will keep with current phone behavior by default

// 1: set as enabled ->” Contact header “information will be shown in BYE message

42 Language filename display on info.xml page (5377)

1. Introduction

In current version, the TB30 phone have a feature which user can load their own language via webpage or telnet . But it doesn't display the loaded language file on web page, we suggest to add "language filename" display on the info.xml .

2.Requirement

For APS , we can upload 4 language tables . 3 default languages and 1 dynamic language.
we need to add "language filename" displays on the info.xml page of phone, the format can likes as below

```
<Language_table> yyyyyyyyyy.zz </ Language_table >
<dft_Language_table1>xxxxxxx.zz </ dft_Language_table1 >
<dft_Language_table2>aaaaaaa.zz </ dft_Language_table2>
<dft_Language_table3>bbbbbbb.zz </ dft_Language_table3 >
```

Note : The name will be updated if the language table is changed via telnet, APS, or webpage

43 Support "HTTPS Web Page" (5382)

1. Introduction

One of our customers request to add a secure way for providing them support downloading the configuration, changing it and uploading it again later.

They want the connection from their technical support stations to the phones to be secured so none hacked PC in the network can sniff network traffic and get the passwords and then change the device configuration.

This requirement is mandatory for them if there is one way to access the phone in a secure way or at least having the passwords encrypted .

2.Requirement

The new feature "HTTPS web page" shall be introduced into our VoIP phone need to starting from the new release.



Three HTTPS page need to be added:

- 1) HTTPS://<IP of Phone>/
- 2) HTTPS://<IP of Phone>/admin.html/
- 3) HTTPS://<IP of Phone >/telnet.html

A config will be used because both HTTP and HTTPS can't be existed at the same time.

Syntax :

websecurityType =0 (default :http , 1 : https without authentication , 2: https with authentication)

When upgrade via APS ,the parameter as below :

tftp:

[config]

```
cert_httpsserver_filename=
pkey_httpsserver_filename=
calist_httpsserver_filename=
```

http:

[config]

```
cert_httpsserver_url=
pkey_httpsserver_url=
calist_httpsserver_url=
```

The following two methods please consider as possible :

- 1) The administrator can choose APS implement (these syntax lines can be added into INF file) to load the https certificates.
- 2) The administrator can upload the certificates via telnet;

Note :

- 1) HTTP web page mode will be used as default;
- 2) If we set to HTTPS mode for web access but without certificates installed, in this case, the web page connection will be failed;
- 3) All HTTPS certificates install will be removed after factory default setting ;

End of document