ST20xx SIP
New Features
SG vx.78.1
Release Notes

Contents

1 Overview

This document describes a set of new features included in ST20xx SIP vx.76 in order to improve their usability in different environments.

2 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.

3 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 <u>REJECT</u>** 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.

4 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.

5 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.

6 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 choises:

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 choise :

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

7 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

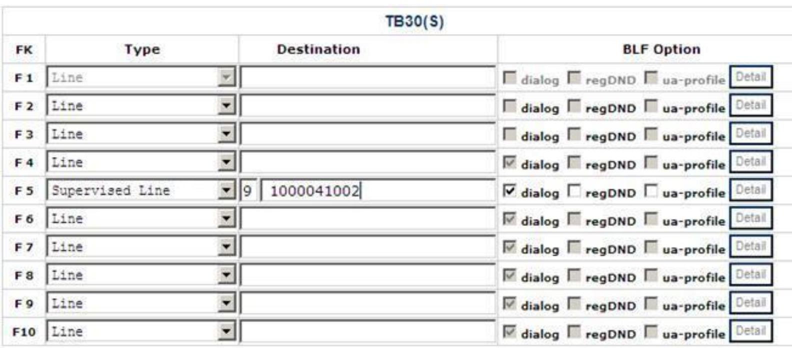
VPNesp2=x

...

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(S)

FK	Type	Destination	BLF Option
F1	Line		dialog regDND ua-profile Detail
F2	Line		dialog regDND ua-profile Detail
F3	Line		dialog regDND ua-profile Detail
F4	Line		dialog regDND ua-profile Detail
F5	Supervised Line	9 1000041002	dialog regDND ua-profile Detail
F6	Line		dialog regDND ua-profile Detail
F7	Line		dialog regDND ua-profile Detail
F8	Line		dialog regDND ua-profile Detail
F9	Line		dialog regDND ua-profile Detail
F10	Line		dialog regDND ua-profile Detail

8 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.

9 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.

10 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.

11 Carries All Codec When Transfer is processed by server 4957

1. Introduction:

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.

2. 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)

12 DHCP Hostname (4819)

1. Introduction:

One of our customer has some other IP phones and they use this system to assign the IPs via DHCP (hostname IP).

2. Requirement:

Implementation DHCP hostname.

13 Supervisor list 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 = 0/1/2 (0: Off Default)

// **0: off (Original)**

// **1: on (SupervisorListening enabled)**

// **2: on (SupervisorListening enabled + voice from handset)**

Remark : When the flag is on, original headset group listening function will be disabled.

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

// **2: 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 also be send out to remote part.**

3. When the flag is on, original headset group listening function will be disabled.

14 Unregistered 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.

15 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.

16 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 handeled 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>&gt123<\digit>

<uppercase>&gtABC<\uppercase>

<lowercase>&gtabc<\lowercase>

<Hebrew>&gtXXX<\Hebrew>

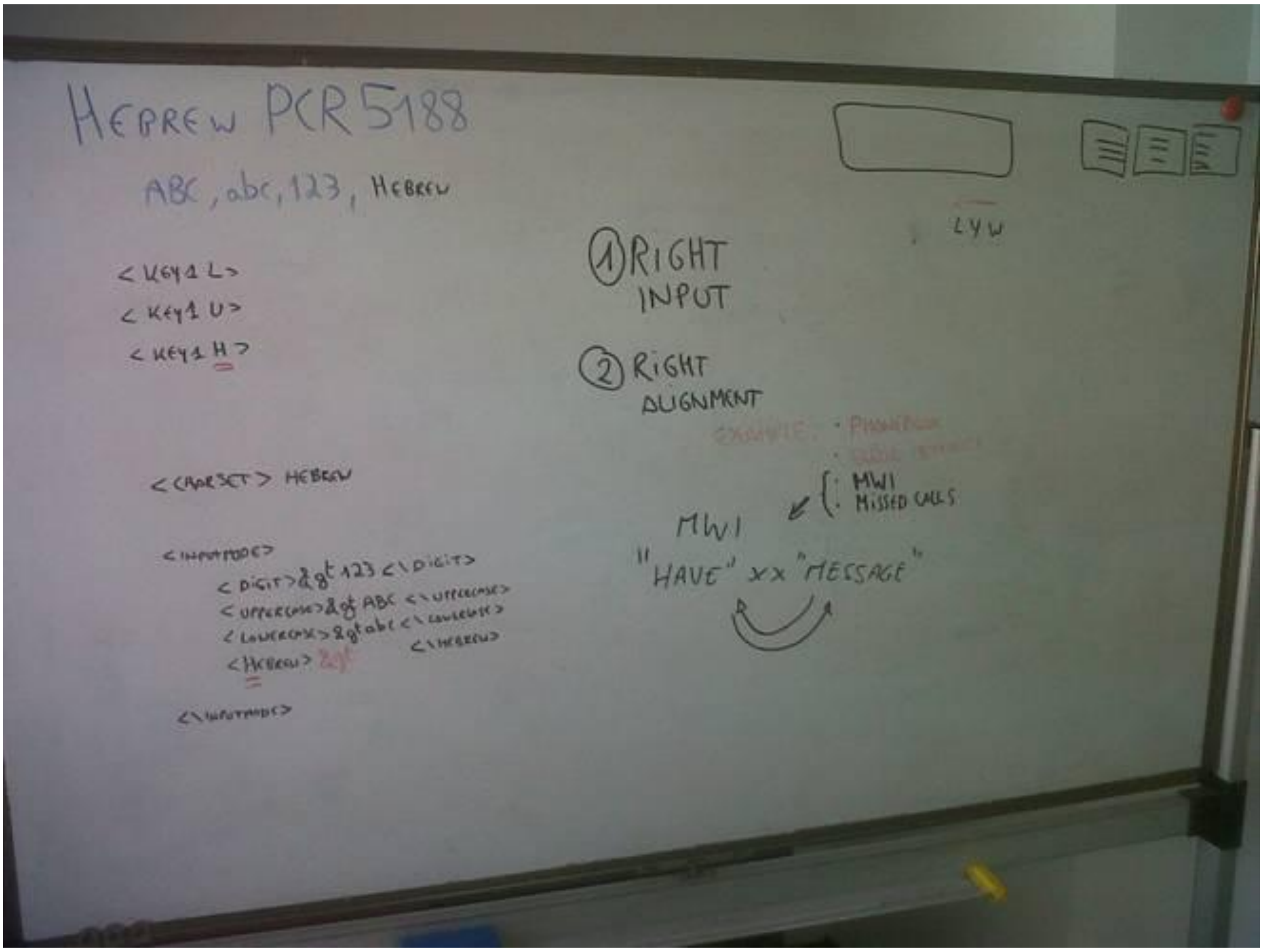
<Input Mode>

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



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