Release Notes for -- Linksys SPA-942 5.2.5

SPA942 -- IP Phone, 2 Ethernet Interface, PoE
SPA922 -- IP Phone, 2 Ethernet Interface, PoE


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      New Features
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Since version 5.1.10

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1. Support Broadsoft CMS extension mobility with optional HTTP Authentication
   on configuration provisioning. There are seven new parameters:
       EM_Enable            ("Yes" or "No", has no macro, web configurable only)
       EM_User_Domain       (macro = "$PDOM", web configurable)
       EM_Login_State       (macro = "$EMS")
       EM_Phone_User_ID     (macro = "$PUID")
       EM_Phone_Password    (macro = "$PPWD")
       EM_Mobile_User_ID    (macro = "$MUID")
       EM_Mobile_Password   (macro = "$MPWD")
   To facilitate HTTP Authentication, profile rules have two new
   tags, "uid" and "pwd". Example profile rule:
   ("$EMS" eq "mobile" and "$MUID" ne "" and "$MPWD" ne "") ?
     [--uid $MUID$PDOM --pwd $MPWD] http://domain/mobileprofile |
     ("$PUID" ne "" and "$PPWD" ne "") ? [--uid $PUID --pwd $PPWD]
     http://domain/hostprofile
   EM_Login_State has three possible values: "", "host", or "mobile"

2. Support customized logo picture to show on the phone screen during boot
process.

3. Support customized background picture. Below are the new parameters:
   [Phone]<Select Logo>: It has 4 options {"default","BMP picture",
                         "text logo","none"}
   [Phone]<BMP Picture Download URL>  (Replaces <Downloadable Background Picture
URL>)
   [Phone]<Select Background Picture> (Replaces <Current Background Picture>):
                         It has 3 options {"none", "BMP picture", "Text logo"}
   BMP picture requirements: black and white only (i.e. 1 bit per pixel),
   with maximum size: 128x64 pixels (but 96x30 is recommended for better
readability)
   Behavior: If user sets <Select Background Picture> to "BMP Picture"
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or "Text Logo", Phone will show the background picture in the middle of the window.
User ID will be drawn on top the of background picture. The information
window is on top of the background picture window and user id window.
For example, if user enables DND, Call-Forward or has missed calls, the information
window will cover the background picture, the portion of the user id will
be covered.

4. Added a <Caller ID Header> option to specify whether the phone should extract caller-ID
information from an inbound INVITE message from the FROM header,
the P-Asserted-Identity header, or the Remote-Party-ID header

5. Added <Notify 1xx On Refer> option (default "yes"). If option is set to "no",
unit will not send a NOTIFY w/ Event=refer on receiving 1xx response to
its INVITE to the transfer target when it acts as the transferee

6. Show the call progress string extracted from a 18x response's startline
on the phone GUI when making an outbound call

7. If an outbound SIP re-INVITE request to hold/resume a call was rejected
w/ 6xx response, revert the call back to the last connected/holding state
instead of ending the call as in previous versions

8. When transferring one party to another failed (due to SIP REFER request
failed for any reason), disconnect with the transfer target but reconnect
back with the transferee, instead of ending all call legs as in previous
versions. Also show the transfer problem on the phone GUI to notify user
if the REFER is rejected

9. Increased acceptable entry length for <Auth ID>, <Password>, and <Display Name>
parameters to 50 characters

10.Support reception and display of text messages sent to the phone using the SIP
MESSAGE method. A few new parameters to control this features:

[REGIONAL]<Alert Tone>: A TONE SCRIPT. This should be a brief tone which plays whenever the phone receives a new text message

[USER]<Text Message>: Enable/disable reception of text messages sent by the SIP Proxy server.Default is "yes"

[USER]<Text Message From 3rd Party>: Enable/disable reception of text messages sent by a third party (other than the SIP Proxy server). Default is "no".

[USER]<Alert Tone Off>: Enable/disable playing the Alert tone on reception of new text messages. Default is "no"

Notes:

. Message information is displayed on the phone main screen. The Following information
is shown: TimeStamp, From, Message body

. Phone will reject message with a 403 response if message reception is disabled
. Phone will overwrite the currently displayed message, if any, in favor of a newly received message.
. When the message is displayed, an "OK" softkey will appear to let user confirm the message can be dismissed. Other keys, except the line keys, the speakerphone key, the headset key, the mute key, the setup key, the voicemail key, and the volume up/down keys, will not take effect until user dismisses the text message.
. If the message is longer than the screen can display, user can use navigation key(up and down) to sroll up and down.
. Max length of the message is 255 characters.
. Only plain-text content type is supported in the SIP MESSAGE method

11.Improved Speakerphone?s full duplex audio performance

12.Added new LED blinking patterns on the power indication LED to indicate RC (remote customization) states:
. Red-orange slow blink while contacting RC server.
. Red-orange double blink if file not found on the RC server, or is corrupt

13.Configuration of Provisioning Server from IP Phone menu. New parameters:
[Provisioning]<User Configurable Resync>:Yes/No, Default is "yes"
New behavior:
. Add entry "Profile Rule" in the Setup Menu of the Phone GUI. This entry can be protected by the Admin Password (along w/ other protected settings)
. User can enter the link (can be either IP or the whole URL) and press "Resyn" softkey to resync the profile (just like entering a resync link from the web page)
. When user enters the link, he can input the whole profile rule including the scheme (tftp/http/https, case sensitive), server IP, port, path, file name. Or he can just enter the server's IP address. When user enters only an IP address, phone will use TFTP, and append the current path or default profile path (e.g., "/spa942default.cfg") as the profile rule (e.g., tftp://<Server_IP_Address>/spa942default.cfg).
. An error message window will pop up if the profile rule is invalid.
. If phone resyncs successfully, phone will reboot right away if there is parameter changes; otherwise phone will show the resync result.
. Use <User Configurable Resync> to enable/disable this feature. The Admin can hide this page from menu once it is disabled. By default, it is enabled.
. Phone will exit admin mode when user exits setup menu or logs out.

14.Support "Photo Album" feature by honoring the HTTP Refresh Timer when downloading GUI Background Picture from the server (specified in the parameter <BMP Picture Download URL>) via HTTP. New behavior:
. Support parsing the HTTP Refresh header in the HTTP Response
. Phone will automatically issue a GET for a new picture if the HTTP Refresh Timer is set in the server's response.
. If the URL is set in the Refresh header, after refresh time (seconds), phone will download the picture from the link and display in the screen.
. Otherwise phone will download the picture from the same link if no URL is defined in the Refresh header.

. Phone will only save the downloaded picture via refresh to the DRAM
  instead of flash.
. Phone will save the picture in flash when user changes the
  <BMP Picture Download URL> parameter
. The minimum refresh timer is 5 seconds.

15.When replying to inbound SIP re-INVITE w/o SDP, the phone repeats the last SDP offer
to the peer in the response instead of starting a brand new offer

16.Map incoming caller-ID with the dial plan specified in the new <Caller ID Map>
parameter before displaying the translated final number on the phone GUI. Default value
of this parameter is blank, which tells the phone to display the inbound
caller-ID as is (which is compatible will old behavior)

17.Support '+' in <Dial Plan> and <Caller ID Map>. For example in <+44:0>xxxxxx,
the + will take effect. Previous version will drop the '+' as if it is not present.

18.Added Screen Saver with User Password Protection
New Parameters
[User] <Screen Save Wait>:   Idle time in seconds, to wait before starting
                               screen saver. Default is 300
[User] <Screen Save Enable>: Boolean (y/n). Default is "no"
[User] <Screen Save Icon>:   This is a choice among {StationTime, Lock, Phone,
                               DateTime, Background Picture}. Default is
                               "Station Time"
New Behavior: When the phone is in idle state for a certain time (idle state
means no key press or hook events and no ongoing call events), phone will
be triggered to enter the screen saver mode. User can enter the screen
saver mode directly from the phone?s setup menu.
Any key press or on/off hook event will trigger the phone to return to the
normal phone mode. If the user password is set, user will be prompted to
enter the password to exit the screen save mode. A screen saver mode window
will show "Press any key to unlock your phone" info window on top scrolling
from left to right. User can select a different screen saver icon from web
config/phone menu. If the phone is locked and screen saver service is
enabled, phone will ask for user password when the phone boots up. Note that
on incoming calls during screen saver mode, the phone will ring and LED
will blink but the screen will still show the screen saver window only.

19.User can enter admin mode by entering admin password from GUI in a locked phone.
Phone will exit the admin mode if user selects logout or exits the setup menu

20.Phone will re-REGISTER when it reaches 80% of the nominal expires value instead of
the old 95%

21.When sending SIP keep alive messages, unit will re-REGISTER immediately if
a) response to keep alive message indicates change in external
IP/SIP-Port in the Via header (with the "received" and "rport" parameter), or

b) no response to the keep alive message, or
c) encounter ICMP error when sending the message.
In addition to re-REGISTER under the above conditions, unit will also send a re-INVITE to the peer on any connected calls

22. When matching a numeric value to a number of patterns in a configuration parameter that allows wildcards '*' and '?', the unit will pick the most specifically matched pattern (instead of the 1st match). For equally best matched patters, the unit will pick the first matched pattern in the list. The matching score is the number of non-wildcard characters that are matching. For example: 12345 matches 12*, 123*, 1234? and 1234? will be selected. This behavior affects parameters such as <SIT1 RSC>, <Retry Reg RSC>, <Try Backup RSC>, etc.

23. Un-REGISTER with the SIP Registrar before a graceful reboot

24.
    1. Reverse back the DNS maximum hosts to 5.
    2. Randomly chose 5 hosts if the dns answers are more than 5
    3. If all answers TTL is 1, treat it as a long TTL (>200 days)

25. During CMS login/logout process, the correct status message is displayed on the phone screen. Also, the line keys are turned off when logging out.

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      Bug Fixes
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Since version 5.1.10

1. Issue: Make a 3 way conference call (with the phone acts as the conference bridge),turn on the speakerphone, and mute the phone, the other 2 parties in the conference cannot talk to each other (Ref=CSCsi97304)

2. Issue: The phone should indicate Ethernet link is lost if Ethernet is disconnected on the Internet port, regardless whether PC Ethernet port is connected or not. Currently the phone still does not indicate Ethernet link lost even if only the PC Ethernet port is connected.

3. Issue: Call logs displayed on webpage are incorrect

4. Issue: Ringtone name display under the Directory->New Entry->Ring option in the phone GUI is wrong

5. Issue: Fixed call statistics history information wrong character

6. Issue: When adding a new entry to the personal directory form the phone GUI, the default ring type is set to 'no ring' instead of 'default'

7. Issue: Default settings of the <Second Preferred Codec> and <Third Preferred Codec> are not backward compatible to earlier versions before these 2 parameters were added

8. Issue: Entries shown on the Redial List (under the Phone's Call History Menu) have their phone number cut by one character at the end

9. Issue: Some SIP Registrars may change the client's Contact address in their response to the SPA's SIP REGISTER request (e.g, when the SPA is behind a NAT and the client uses the private address in the Contact), so that the SPA cannot find the corresponding Contact in the response and therefore may not be able to extract the proper expires value inserted by the server. This is a problem if the expires value inserted by the server in the response is smaller than the value in the original SIP REGISTER request. This issue is fixed by letting phone to use the first entry in the Contact header returned by the SIP Registrar if an exactly matching address is not found.

10.Issue: SDP version should increment by one for each new update during a call. At present the phone is a new randomized version on each SDP update.

11 Issue: "dnd" and "-dnd" softkey (to enable or disable Do Not Disturb feature) may go into a wrong state (Ref=CSCsc14307)

12.Issue: Parameters shown under the [Ext 3] and [Ext 4] tabs are not consistent with those shown under the [Ext 1] and [Ext 2] tabs on the Phone's Web page (Ref=CSCsi25423)

13.Issue: Customized bootup logo picture may not work after certain sequence of operations (Ref=CSCsj66087)

14.Issue: After signing a dictionary using dictsign (for GUI localization), the XML file cannot be fully viewed using MS Internet Explorer

15.Issue: Speakerphone howls sometimes (Ref=CSCsj89627,CSCsj89604)

16.Issue: Daylight Saving Time rule is not correctly applied if the start time is later than the end time of the year in the <Daylight Saving Time Rule> parameter

17.Issue: Enabling silence suppression mutes speakerphone input (Ref=CSCsk13542)

18.Issue: When far end hangs up, the phone should terminate any pending outbound INVITE transaction within a relatively short time instead of following the normal INVITE time out values. For example, they should terminate within 4s after far end hangs up instead of waiting for the full 32s

19.Issue: Unit should ignore in-dialog re-INVITE if it has not received ACK to the 200 response to the initial INVITE from the caller. This condition might happen if the ACK was lost but the first re-INVITE has already been sent by the peer

20.Issue: Phone should not include port number in FROM and TO headers in outbound SIP requests per RFC3261

21.Issue: Inbound SIP messages exceeding 3 Kb are truncated by the Phone. Big SIP messages usually arise when the content is a large XML document (such as dialog-info for 1 or more parties). It can happen with SIP transported over TCP or UDP

22.Issue: When user hangs up immediately after answering a call, where a SIP 200 response has been sent but the ACK has not been received yet, the phone should still send out a BYE but it does not. Note that this is a very special case where the time between answering and hanging up has to be really short (relative to network delay) in order to trigger this problem

23. Issue: Phone should reuse the last good address list for a given hostname when the current DNS query for the hostname failed

24. Reboot reason should be "User Requested" when reboot/restart from the phone GUI

25. remove "acoustic echo canceller" debug flag from web ui

26. Issue: When connected to an emergency number, the unit will set the call to idle state internally if the phone is onhook for more than 30s and without sending out any BYE message to the peer. Instead, the unit should keep the call connected indefinitely while onhook until the peer hangs up (Ref=CSCsl41231)

27. Fixed this problem: When making an outbound call from the phone, if the 18x/2xx response to the SIP INVITE contains a tel url in the Remote-Party-ID or P-Asserted-Identity header, the phone may replace the called peer number shown on the screen with a blank number

28. Fixed this problem: After attended or blind transfer a call, the phone will not release the call with the transferee until the transfer target picks up or rejects the SIP INVITE from the transferee