Release Notes for -- Linksys SPA-942 5.2.8

SPA942 -- IP Phone, 2 Ethernet Interface, PoE

/** NOTICE **/

==
      New Features
==

- Added "dir" softkey in the blind call transfer dial window.
  User can then set transfer target from contact list.

- Show message window only when phone window is the
  foreground window.
  When the phone is on a call, pass the DTMF to the remote end.
  When the foreground window is other windows,
  ex setup menu, dir menu, volume setting window, phone only plays
  the message tone, does not show the message window untill user
  returns back to phone window.

==
      Bug Fixes
==

- CSCsm20842: Phone was requesting DHCP option 150 parameter
  even if voice VLAN is disabled on the switch or CDP is
  uninitialized.

- CSCsm22738: During onhook dialing, cannot press '#' key to
  force the phone to start dialing, even if the dialplans allow it.
  Fix: Compare dial string to all the dial plans configured for every
  enabled Line key.
  (user can choose any Line to dial during onhook dialing)
  whenever a # key is entered, and all dial plans
  indicate the dial string is complete, the phone will start dialing
  as if "dial" softkey is pressed.

- CSCsl76422: UCS-2 localization files cannot be accepted.

- CSCsk36468: Calling pagegroup (with SPA9000) using g726 (at any bit rate)
  causes bad audio

- CSCsi63429: When used with a SPA9000, corporate directory cannot be retrieved from the SPA9000 if EXT 1 is not enabled on the phone.
  Notes - Problem is caused by the phone relies on EXT 1 to communicate with the SPA9000 for system information.
  Fix: by using the first EXT that is registered successfully with a SPA9000

- CSCsi65072: Setting <Debug Server> or <Syslog Server> to the same IP address of the phone and the <SIP Port> of one of the enabled EXTension can cause the phone to reboot every 10s.
  Notes - Problem is caused by the phone receiving a debug message at the SIP port, then tried to log it again, hence causing an infinite loop.
  Fix is by detecting such illegal messages at the SIP port and to drop them w/o any logging or processing.

- CSCsg28440: The "Setup" top title on the menu is not translated right after a new language is selected and saved in the setup menu.
  Fix: by redrawing the screen, then the title will be translated.

- CSCsk09820: phone is muted when using speaker phone and adjusting speaker volume.
  Root Cause: The discontinuity of the audio stream due to volume adjustment will harm the echo cancellation capability of the AEC, resulting in echo.
  The original design was to mute the mic, so that echo cannot be looped back. The new implementation is to freeze the adaptation of the AEC during these times.
  The mute is traded for minor residual echo.

- CSCsi68660: Phone uses <Shared User ID> instead of <User ID> for REGISTERing an EXTension when that extension is not shared
  Notes - The phone mistakenly use <Shared User ID> (if the parameter is specified and different from <User ID>) in the Contact header of the REGISTER message even if the EXTension is not a shared extension (i.e., <Share Ext> = "private"). The phone should just use the <User ID> if the Extension is not shared. This is fixed. However, Phone will use <Shared User ID> for REGISTER an EXTendsion as long as <Share Ext> = "shared" for that EXT; even if no Line Key corresponds to that EXT is "shared".
  This is correct behavior. If user does not want to the phone to use <Shared User ID> at all, the user must leave this entry blank.

- CSCsc94405: phone does not reboot automatically (for new settings to take effect) when <Dial Plan> is modified for EXT 3 or 4 on the web page
  Notes - If <Dial Plan> is modified for EXT 1 and 2, the phone does not automatically reboot for the new <Dial Plan> settings to take effect. This problem is fixed such that <Dial Plan> changes on EXT 3 and 4 will also automatically reboot the phone.

- CSCse48260: voice mail access key wrong behavior
  Notes - The current issue is that each keypress of the voicemail button starts a new
  call to voicemail; the last call connected to voicemail will be placed on
  hold. Changed the behavior to such that further keypresses of the
  voicemail button are ignored if there is already a call to voicemail. Pressing
  voicemail button will be handled only if the last call to voicemail has ended (ssin)

- CSCse32964: Phone does not switch to show calling screen when pressing voicemail key
  to call voicemail while the setup GUI menu is shown
  Notes - It is expected that the phone will switch from the setup screen to the
  call screen once user presses the voicmail key to call voicemail.

- CSCse74048: Phone does not automatically reboot when <Mini Certificate> is
  cleared (or changed) for it to take effect

- CSCsc99617: Phone will ring indefinitely withou timing out when calling from one EXT
  to another EXT on the same phone
  Notes - generally, when calling out while there is an incoming call, the phone
  plays ringback tone for the outgoing call, but never times out the incoming
  ring and rejects it (like it should normally after 30 or 60s of ringing).
  The problem is caused by the phone not playing ring tone or call waiting tone
  for the incoming call in this scenario and therefore cannot does not have the
  corresponding "tone end" event to trigger the rejection of the call.
  Fixed by generating an equivalent ring tone or call waiting
  tone ended event (even if the tone is playing out loud).

- CSCsi28470: Userinfo of To-URI truncated after 79 bytes when SUBSCRIBE to "dialog;sla"
  in a SIP-B installation
  Notes - The Request-URI of the SUBSCRIBE is formed correctly; only the TO header is
  truncated. The TO header user-id in this case should follow that in the
  Request-URI.

- CSCsk11863: wrong tone display in GUI and web UI

- CSCsm28353: Phone does not re-register immediately when there is no response to
  the last NAT Keep Alive SIP message (such as a SIP PING or SIP NOTIFY message)
  Notes - Re-Register at once on no response to a Keep Alive SIP request is a feature
  added in 5.2.x relesase, and is not working.

- CSCsf28625: When the phone is alerting for an incoming call, pressing the hold key (the hand-figure one) makes the ringtone restart from the beginning of the cadence.
  The more frequent it is pressed, the faster the tone restarts.

- CSCsf28691: Local Date can be set to 2/30(mm/dd) in the setup menu.
  Root Cause: The date validation logic is loose. Any day no larger than 31 is considered valid, without regard to which month it belongs.
  Fix: The validation is rewritten, taking into account leap year, and which month it is. Also, the new date will not be stored if it is not valid.

- CSCse49603: Wrong DTMF tone level using speaker phone.
  Root Cause: When using speakerphone, the DTMF tone is mixed with the local DTMF echo playback.
  Fix: Not mixing in the microphone input under such situation.

- CSCsl10378: Reboots every 500 or so, when calling to outside

- CSCsl93285: Reboots under stress test with SPA9000 share line

- CSCsk68778: Outgoing Call Progess message not updated according to 18x response unless <SIP Remote-Party-ID> is enabled.
  Notes: If <SIP Remote-Party-ID> is not enabled, the phone will simply show "Called Party Ringing" regardless what is in the Startline of the 18x response. For 5.2.x, the proper behavior is to show the message embedded in the 18x response. This is corrected in 5.2.x
  Note: For 5.1.x and 4.1.x, the phone is expected to always show "Call Party Ringing" during call progress and ignore the message in the 18x response.

- CSCsk69012: Phone does not re-REGISTER when response to keep alive SIP message indicates change in the external IP
  Notes: This is a 5.2.x feature where the phone should re-REGISTER immediately when the response to its last SIP keep alive message (such as PING or NOTIFY) indicate a change in the external IP or external SIP port (as indicated in the received= and rport= parameters in the VIA header of the SIP response message).

- CSCsj23404: Last character of caller name removed in call history if name has leading double quote but no ending double quote.
  Notes: When there is no ending double quote, the leading double quote should be included as part of the caller name. For example, if caller name is: "John Xyz the call history incorrectly shows: John Xy (that should be: "John Xyz).

On the other hand, if caller name is: "John Xyz"
the call history correctly shows: John Xyz

- CSCsg40578: No Ring or Ring does not time out on concurrent incoming calls
  Notes - When A and B calls the phone at the same time, it only plays the ring for the first call (does not switch to the ring for the 2nd ring). Then when A hangs up, the phone stops ringing but B is still calling and hearing the ringback tone.
  Fix: The phone swtiches to the ring corresponding to the newest incoming call.
  And when the ringing incoming call is ended by the caller, the phone should switch to ring one of the remaining incoming call (the order in this case is not deterministic).

- CSCsg06355: *98 blind-transfer causes beeping tone
  Notes - After blind transfer a call by pressing *98 followed by the target number, the phone will go to idle. However there is some short beeping tone before dial tone when offhook the phone again to make the next call. Beeping tone disappear afterwards for subsequent calls.

- CSCsf28730: one-way audio when 962 behind router and transfer target
  Notes - When 962 acts as a transfer traget in a call transfer operation, and the 962 is behind a NAT router and relies on STUN for NAT traversal, the call can be established with the transferee on completion of the transfer but the 962 cannot recieve audio from the transferee (but the transferee can from the 962). The problem is due to the phone not performing another STUN request to map RTP ports before accepting the call from the transferee. This is fixed by adding the STUN step before accepting the call from the transferee.

- CSCsm75062: Extra "" in SIP header with <Escape Display Name> enabled but <Display Name> set to blank
  Notes - Outbound SIP messages have the FROM header such as:
  FROM: "" <sip: ....>
  which is not wrong, but the "" should be eliminated.

- CSCsk78150: No version change when using 5.2.2 exe upgrade tool
  Notes - When using the upgrade .exe tool to upgrade phone firmware from version 5.2.x to other versions, the tool does not show the new version number when reporting upgrade successful; it still shows the old version number even when upgrade is successful. The problem is due to the tool not allowing enough time for the firmware to completely updated and phone successfull unRegister before rebooting, so that the phone is still running the old version code when the tool requesting the version number from the phone.
  Fixed: Tto wait extra 5s before checking the version of the upgraded phone.

- CSCsk00111: share-line issue: pick up hold call to itself will fail and cannot be picked up further by another station
  Fix:

1. Allow the call to be picked up another station if the last attempt
   by a station to pick up the shared held call has failed for any reason,
2. Do not allow a station to pick up a held call that results in a call with itself

- CSCsl69592: Phone reboot every 30s and cannot factory reset when PPPoE fails
  Notes - When phone is configured to use PPPoE instead of DHCP and PPPoE fails,
  we cannot factory reset the phone to go back to DHCP.