SIPURA
technology, inc.

Release Notes for 3.1.3(GW) -- Sipura Phone Adapter

SPA-3000 -- 1 Port FXS, 1 Port FXO, 1 Ethernet Interface

Bug Fixes
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    If inbound INVITE to/from header with user-id longer than 79 chars
    or uri parameters longer than 63 chars, then subsequent outbound
    requests from the SPA within the same dialog might have errors in
    the from/to headers.

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    If <Refer-To Target Contact> is no, the Refer-To header might be
    wrong if traget's Contact hase uri parameter longer than 63 chars.

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    Setting <MWI Serv> to "no" cannot turn off stutter dial tone

.
    SPA may crash if the call peer's display name field is longer than 29 chars and
    enclosed in a pair of double quotes in the SIP message.

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    SPA will ignore any 6xx response to its outbound INVITE

.
    Reboot when invoking IVR

New Features and Enhancements
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   Support escape \" and \\ for " and \ within quoted string in SIP messages

. Added <Escape Display Name> option to escape the configured
    <Dislay Name> and enclosed the

string in double quotes in outbound SIP messages. If option is enabled and display name contains " or \, it will be escaped to \" and \\ inside the double quotes.

. Removed polarity reversal in danish caller-id generation

. Allow Call Replaces during outbound ringing/calling state to support sylantro flavor of call pick up and group call pick up

. Added configurable backoff schedule on error retries for resyncs and upgrades.

. Support auto-adjustment of Daylight Saving Time, using the new config parameter <Daylight Saving Time Rule>

. Handle BYE/Also call transfer properly

. Allow non-sip URL scheme in inbound SIP messages (instead of treating them as bad messages)

. Support SIP 491 response per RFC3261 to resolve glare when both sides sending reINVITE at the same time.

. Mute audio line in when playing call-waiting tone or generating CWCID so that far end will not hear them.

. When CED/CNG tone is detected, unit will automatically go into fax/modem passthrough mode where it will,
a) change codec to G711 (and also send reINVITE or NSE event to far end as needed)
b) disable echo canceller
c) disable silence suppression
d) disable call waiting
e) increase jitter buffer size to 100ms and do not allow it to shrink
All these changes will last until the end of the current call

If the unit receives NSE event from the far end, it will also go into fax/modem passthrough mode (if <FAX Process NSE> is enabled)

. Added <FAX MODEM Line> Line 1/2 parameter. If this parameter is set to "yes", the Line will be in fax/modem passthrough mode for all incoming or outgoing calls. Default value is "no"

. Added <Modem Line Toggle Code> Regional parameter. User can dial this star code before making a call to enable the fax/modem passthrough mode for the next call.

. Change default modem pass through code from *03 to *99

. Added <Sticky 183> boolean paramter under Ext 1/2/3/4. If set to "yes", SPA will ignore further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. This avoids double ring-back tone in case the 180 response comes after 183 out-of-order

- Wait 4 seconds after hang-up before gracefully reboot so that we have a chance to tear down the call.

- When inbound reINVITE codec list mismatches current list, try to allocate a fresh codec list.

- Reply 481 to SIP OPTIONS in-dialog request if no matching dialog found

- When a PSTN caller is automatically routed to a VoIP destination due to a) hotline w/o authentication, or b)call forwarding, the SPA will not take the FXO port off-hook until the VoIP destination answers the call. If the VoIP call leg fails (busy, etc), the PSTN call will not be picked up by the SPA at all.
  The old behavior for this scenario is that the SPA will off-hook the FXO port first before calling the VoIP destination. To keep the old behavior, set the new [PSTN Line] paramter <Off Hook While Calling VoIP> to "yes" (default is "no")

- In a PSTN<->VoIP gateway call, the SPA will process out-of-band hook flash signal sent from the VoIP peer by hook-flashing (on-hook momentarily) at the FXO port. This allows the VoIP peer to initiate 3-way or conference call (and call transf subsequently). The hook-flash timing (duration of a brief on-hook) can be set in the new [PSTN Line] parameter <PSTN Hook Flash Len> in seconds. Default is 0.25. The range is limited to 0.02 to 1.6s.

- When a caller is talking with a PSTN line peer from Line 1 (FXS), he can send a hook flash signal to the PSTN line by hook flashing the phone twice within 0.5s. This feature is enabled by setting the new [PSTN Line] parameter <<Line 1 Signal Hook Flash To PSTN> to "Double Hook Flash". The default value is "Disabled" which disables this feature. If the Line 1 caller is currently not connected to a PSTN line peer, double hook flash event is ignored by the SPA. Normal single hook flash from the FXS port is processed by the SPA internally as before.