SIPURA technology, inc.

Support

company products partners support contact

Release Notes for 2.0.13(GWg) -- Sipura Phone Adapter

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

Bug Fixes
===

. URI parameter in Authorization header may not matching request uri.

. SPA ignores STUN responses if the message is larger than 64 bytes

. OOB DTMF via SIP INFO still transmitted when call is on hold; result is that holding peer can hear DTMF as caller dialed 3rd party in a 3-way call.

. SPA does not include all available codecs when transferred; it only includes a trimmed down set as a result of the original call establishment.

. <Dial Plan 8> does not work for VOIP to PSTN calls

. After entering invalid PSTN pin, next PSTN PIN tone does not time out until a new PIN digit is entered.

. SPA does not support compact form SIP EVENT header 'o'.

. SPA does not support changes in SDP in successive SIP 183 responses.

New Features and Enhancements
===

. Increased maximum allowed SIP URL parameter length to 574 characters in To, From, Contact, Route, and Refer-To headers.

. Increased maximum allowed SIP URL user-id length to 206 characters in To, From, Contact, Route, and Refer-To headers.

. Support RTP keep alive at the interval specified in <NAT Keep Alive Intvl> and is enabled if <NAT Keep Alive Enable> is "yes". RTP keep alive

```
  is active only when normal RTP transmission is paused due to call hold
  or silence suppression.

. Allow up to 199 chars of Call-ID and Branch values in inbound SIP messages.

. Support <No UDP Checksum> (SIP) option for outbound RTP packets

. Added <Referor Bye Delay>, , and
  parameters for Line 1/2, to control when SPA sends BYE to terminate
  stale call legs on transfer completion as the Referor, Referee, and
  Refer Target respectively.

. Added <Refer-To Target Contact> Line 1/2 parameter to control whether
  to use the Refer Target's contact or it's public address
  in the Refer-To header of the REFER request when the SPA acts as the
  referor.

. Support Denmark/Netherlands PSTN line side caller-ID detection

. Added <Stats In BYE> (SIP) parameter. If enabled, SPA will include
  P-RTP-Stat header a BYE or response to a BYE message
```