OM20/50 IPPBX Software Release Notes

Version: Rev 2.1.5.100 Date: May 12, 2016

Firmware Modules

Model / Component	OM20	OM50
Software Version	Rev 2.1.5.100	Rev 2.1.5.100
OS Version	2.0.14	2.0.14mt

New Functions

None

Functions Improved

1. Supports for detecting caller ID before first ring and between first and second ring without preconfiguring the type.
   Protection is added to avoid picking up a false ring signal.
2. For authorization code dialing and DISA as well, allows a caller to enter the destination number without adding "#" sign at the end of the authorization code. A waiting period of 2 seconds after dialing the authorization code is required before dialing the destination number.
3. Supports re-register and calling again when the 403 message is received for an outgoing call made by a registered SIP trunk.
4. The next available extension in the operator hunt group will be selected to terminate the call if the current extension does not answer.
5. Allows the device to take API requests from multiple servers or terminals with a dynamic or static IP address.

Bugs Fixed

1. Fixed the bug that could cause the App to restart when sending an API report to an unreachable server.
2. Fixed the issue that the IP extension can make outbound calls without adding a prefix, even though the dialing method was "dialing with prefix".
3. Fixed the following issues related to transfer an internal call to the third party via SIP trunk:

a) The party who does the transfer was not released thus a three-way calling was formed.

b) After the call being transferred to the third party, the extension user cannot put the call on hold by pressing **.

4. Fixed the issue that causes failure when activating call hold feature using voice menu (*99 feature code).

5. Fixed the bug which could prohibit downloading the audio file synthesized using text-to-speech feature on the Web GUI.

6. Fixed the issue that allows a user instead of an administrator to modify SIP, HTTP, and HTTPS ports and disable Telnet on **Status** > **Alarm** page on the web GUI.

7. Fixed the issue for displaying incomplete port numbers on **Extension** > **Analog** page on Web GUI using Google Chrome.

8. Fixed the bug that could cause issue when sending DTMF signal through SIP with RFC2833, especially when G.279 is used.

9. Fixed the issue that a DNS query failure could cause abnormal ring cadence.

10. Fixed the issue that the limitation of 50 voice mail messages for each analog extension did not take effect.