OM Series IPPBX Software Release Notes

Version: Rev 2.1.5.101   Date: May 27, 2016

Firmware Modules

Model / Component	OM80
Software Version	Rev 2.1.5.101
OS Version	Kernel 2.0.9

New Functions

1. Supports multilingual IVR and allows users to upload their own voice prompt packages.
2. Supports assigning an IP address for each of the two Ethernet ports.
3. Supports port redundancy providing automatic port failover when a port or connection in use becomes unavailable.
4. Supports HTTPS for Web GUI access.
5. Detects changes of the reflexive address of the access router through STUN query, and trigger re-registration to SIP server. The factory default STUN server is the New Rock STUN server.
6. Supports multi-level auto attendant. See “Configuration Guide of Multi-level IVR” for more information.
7. Supports time schedules of auto attendant, allowing users to setup multiple auto attendant schedules and rules based on business hours per week or per month. For details of configurations, see “Configuration Guide for Auto Attendant Template”.
8. Supports logs to be saved on the built-in SD card.
9. Supports simplified DISA which allows an external terminal to access the device through verifying the caller ID.
10. Supports binding caller numbers to an extension such that the calls from the callers will be directed to the bound extension without going through auto attendant.

Functions Improved

1. The hot line delay time is allowed to be configured on **Extension** >**Analog** >**Advanced** page.
2. Supports sending caller display name for intercom between IP extensions.

3. Supports configuring a caller display name in Chinese character for an extension.
4. Adds the **Obtain caller ID from** parameter on **Trunk** > **IP trunk** > **Registrar options** page to choose the caller ID either from P-Asserted-Identity header or From header.
5. Makes the look and feel of OM80 Web GUI similar to that of OM20/50.
6. Provides the description for failure cause of text-to-speech conversion.
7. Provides a voice prompt for a failed outbound call.
8. Adds feature codes *99 for feature management through voice menu, *66 and *67 for direct calling between caller and callee (bypass auto attendant), on **Advanced** > **Feature code** page.
9. Adds "Turkey**"** option on **Advanced** > **Tone > Country/Region** page.
10. Allows to configure the number of ringing cycles for no-answer forwarding on **Advanced** > **System** page.
11. Allows to import DID configuration in excel format on **Trunk** > **IP trunk** page.
12. Supports obtaining DNS server address through DHCP.
13. Allows to configure no-answer forwarding and busy forwarding separately.
14. Adds a parameter to enable registration to multiple ITSPs.
15. Numerous enhancements and bugs fixed on OM API.
16. Auto Provisioning:
    - Supports getting routing configuration from ACS.
    - The **FIRM_UPGRADE** parameter provisioned from ACS takes effect immediately rather than next time when the device visits ACS.
    - Supports access to ACS through HTTPS.
    - Supports DHCP option 66 to obtain ACS address.
17. Changes of factory default:
    - The default ringing timeout of extension (**ringing timeout before voice prompt** on **Basic** > **Auto attendant** page) is extended to 50 seconds.
    - The digit map rule x.T is changed to xxxxxxxxxx.T.
    - Allowed to response to Ping requests.
    - Display the SIP trunk account password in encrypted format instead of plain text.
18. Supports E.164 numbering plan with plus sign on **SIP trunk ID** and **Username** on **Trunk** > **IP trunk** > **Add** page.
19. Supports modification of the extension numbers in batch on **Extension** > **Analog** page.
20. Supports dial tone detection on FXO port.
21. Supports PSTN failover upon the failure of DNS lookup for SIP server address.
22. Supports for detecting caller ID before first ring and between first and second ring without preconfiguring the caller ID type.

    Protection is added to avoid picking up a false ring signal.

23. 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.
24. Supports re-register and calling again when the 403 message is received for an outgoing call made by a registered SIP trunk.
25. The next available extension in the operator hunt group will be selected to terminate the call if the current extension does not answer.
26. Allows the device to take API requests from multiple servers or terminals with a dynamic or static IP address.

Problems Fixed

1. Fixed the call forwarding failure issue with DID trunk.
2. Fixed an issue that the outgoing call on a SIP trunk would not failover to PSTN upon the SIP INVITE is time out.
3. Fixed bug that caused INVITE message carried incorrect local SIP port number.
4. A call to an unreachable IP extension within a hunt group will be directed to an alternative extension within the same group after INVITE timeout.
5. Fixed bug that caused duplicate call records in the CDR server.
6. 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 caller (extension user) cannot put the call on hold by pressing **.
7. Fixed the issue with activating call hold feature using voice menu (*99 feature code).
8. Fixed the problem with sending DTMF signal via RFC2833, especially when G.279 is used.