# OM Series IPPBX Software Release Notes

Version: Rev 2.1.5.95 Date: Nov 30, 2015

## Firmware Modules

| Model / Component | OM20 | OM50 |
|---|---|---|
| Software Version | Rev 2.1.5.95 | Rev 2.1.5.95 |
| OS Version | Kernel 2.0.9 | Kernel 2.0.9mt |

## New Functions

1. OM20/50 supports multilingual IVR and allows users to upload their own voice prompt packages in addition to the built-in English and Chinese packages.
2. Supports managing the device using SSH or HTTPS.
3. 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. It is disabled by default.
4. Supports multi-level IVR to create auto attendant with more menu options. For details of configurations, see “Configuration Guide of Multi-level IVR”.
5. Supports time-based 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”.
6. OM20/50 supports G.729 recording.
7. OM20/50 supports logs to be saved on the built-in USB device.
8. OM20/50 supports simplified DISA which allows an external terminal to access the device through verifying the caller ID.
9. 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. OM20/50: When there is no enough free space on the USB device, the latest recording files can overwrite the existing recording files.
2. The hot line delay time is allowed to be configured on **Extension** >**Analog** >**Advanced** page.

3. When ringing with no answer on an operator extension, the incoming call will be redirected to the next available extension in the operator group.
4. Supports caller name display for intercom when both parties are IP extension.
5. Supports caller name display in Chinese characters.
6. Add 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.
7. Optimizes text on **Advanced**>**Security**>**Ping** page.
8. Shows text-to-speech conversion failure cause in plain text rather than error code.
9. Optimizes system voice prompt.
10. 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.
11. Adds "Turkey**"** option on **Advanced** > **Tone > Country/Region** page.
12. Allows to configure the number of ringing cycles for no-answer call before automatically forwarding it to an alternative extension on **Advanced** > **System** page.
13. Allows to import DID configuration in excel format on **Trunk** > **IP trunk** page.
14. In DID mode, the outbound calls from the extension are allowed to select only its bind trunk.
15. Supports no-answer forwarding and busy forwarding as two configurable features.
16. Adds a parameter to enable registration to multiple ITSPs.
17. Numerous enhancements and bugs fixed on OMAPI.
18. Auto Provisioning:
    - Support getting routing configuration from ACS.
    - The factory default of HTTP User-Agent header is allowed to be configured in the format "Company Device Software-Version MAC-Address/SIP-Version"
    - The **FIRM_UPGRADE** parameter provisioned from ACS takes effect immediately rather than next time when the device visits ACS.
    - Support access to ACS through HTTPS.
    - ACS address can be obtained via DHCP option 66.
    - OM20/OM50 supports Kernel upgrading by obtaining .img file from ACS through auto-provisioning.
19. Changes of factory default:
    - The default ringing timeout of extension is extended to 50 seconds.
    - Allowed to response to Ping requests.
    - For OM20/50, voice mail is disabled as default
    - Display the SIP trunk account password in encrypted format instead of plain text.

## Problems Fixed

1. Corrects incorrect display of SIP signaling and RTP ports on **Basic** > **Remote access** page.
2. Fixed the issue that could add noise to the audio when the volume from analog ports is high.
3. Fixed the issue that connecting IP phones causes the device fails to boot up.
4. Fixed no dial tone issue.
5. Fixed the call forwarding failure issue with DID trunk.
6. Fixed an issue that the outgoing call through a SIP trunk would not failover to PSTN upon the SIP INVITE is time out.
7. Fixed the bug that could cause the device to crash when the IP address of the device changes.
8. Fixed the bug of SIP re-registration delay when the Ethernet port is back to work.