# Release Note for N412

Yeastar Information Technology Co. Ltd.

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# FIRMWARE VERSION 45.15.0.3

**DATE: July 25, 2017**

## CHANGES SINCE FIRMWARE RELEASE 45.15.0.1

### BUG FIXES

1. Fixed the issue that when using extensions to record custom prompts, the call would be hung up automatically and custom prompts record would fail.
2. Fixed the issue that when using analog phones to record custom prompts, the system would hang up.
3. Fixed the issue that analog phones would not play the distinctive ring tones when the call came from the BRI trunks and was delivered to the ring group.
4. Fixed the issue that system would not skip the default “Unavailable Prompt” after uploading the custom “Unavailable Prompt” to FTP voicemail folder.

# FIRMWARE VERSION 45.15.0.1

**DATE: April 26, 2017**

## CHANGES SINCE FIRMWARE RELEASE 45.14.0.23

### NEW FEATURES

1. Added support for monitoring FXS port and GSM channel on Web (Status > Reports > Packet Tool).
2. Added support for German Web GUI and Italian Web GUI.

### OPTIMIZATION

5. Optimized DOD setting: support DOD numbers starting with digit 0.
6. Optimized Distinctive Ringtone feature: distinctive ring tone will also work for FXS extensions in a ring group.
7. Optimized seizing a line to call out: the administrator can set which extensions have the permission to seize a line.
8. Optimized IVR setting: users could set Invalid Key event and timeout event to a custom prompt.
9. Optimized Queue setting: “Ring in Use” is disabled by default.
10. Optimized # key feature: users can choose whether to press # to send a call or

not (PBX > Basic Settings > General Preferences).

## BUG FIXES

1. Fixed Service Provider Trunk issue: if Qualify was disabled, the trunk could not connect to the service provider.
2. Fixed Call Log issue: call logs for conference calls were incorrect.
3. Fixed SIP trunk issue: N412 didn’t support SIP INVITE packet with the field “transport=udp”.
4. Fixed Ring Group issue: the ring strategy “Ring One Each Time” could not work properly.
5. Fixed Queue issue: the ring strategy “leastrecent” and “fewestcalls” could not work properly.
6. Fixed Call Transfer issue: the “Attended Transfer Caller ID” setting (Basic Settings > General Preferences) could not take effect.
7. Fixed the issue that when using BLF key to monitor call park extension, the BLF status was incorrect.
8. Fixed Voicemail Prompt issue: change the voicemail prompt, the system would still remain the system default voicemail prompt.
9. Fixed FXS extension Call Transfer issue: using the flash key to transfer could not work.
10. Fixed the Call Log issue: users could not download filtered call logs from N412 Web interface.
11. Fixed NTP server issue: IP phones could not obtain time from N412 NTP server.
12. Fixed compatibility issue with some analog phones: if a trunk had no Caller ID service, when users call in the trunk, and the call reached an analog phone, the phone might crash.
13. Fixed the issue that there was a 502 Bad Gateway error after logging in the N412 Web interface.
14. Fixed the Record Log issue: record logs would show incorrect callee number if users seized a line to call out.
15. Fixed the call transfer issue: using flash key to transfer a call that could not work.
16. Fixed the call quality issue of BRI trunk.
17. Fixed the SIP trunk issue: the SIP trunk registration might be failed after a period of time.
18. Fixed the network issue: if WAN and LAN interfaces were both enabled, WAN DNS could not work.
19. Fixed the NAT issue: the NAT domain name could not be resolved.
20. Fxied the issue on ring group and queue: agents would get missed call information on their phones even if the incoming call was answered by another agent.

# FIRMWARE VERSION 45.14.0.23

**DATE: October 8, 2016**

## CHANGES SINCE FIRMWARE RELEASE 45.14.0.22

### NEW FEATURES

1. Added support for switching Follow Me status using feature codes.
2. Added support for dialing external numbers when in an IVR.
3. Added support for customization of ring tones.
4. Added support for storage of Voicemail messages in the external drive.
5. Added support for going back to IVR menu if the called extension does not answer the call.
6. Added option "FXS Port Interdigit Timeout".
7. Added support for "Turkey" FXO Mode.
8. Added option "CO Line DTMF Duration" to set the duration of a DTMF tone on a CO line.
9. Added option "CO Line DTMF Gap" to set the interval between each DTMF tone.
10. Added "DTMF" Caller ID Signaling and "ETSI V23" FSK Caller ID type; added "Caller ID + Ring" and "Polarity + Caller ID + Ring" sending modes.
11. Added new ring strategy "Ring One Each Time" for Ring Group.
12. Added support for making three-way call using Flash key on an analog phone.
13. Added option "Off-Hook Dial Delay" to set the delay on CO line between off-hook and dialing digits.
14. Added support for multiple time conditions and destinations on the Inbound Route.
15. Added support for canceling attended transfer when the called person does not answer.
16. Added option "DNS-NAPTR" to search SIP trunk transport, port and server.
17. Added SIP related settings.

### OPTIMIZATION

1. Enhanced DSP stability.
2. Changed the recording file format from wav49(GSM) to wav_ulaw(ulaw).

### BUG FIXES

1. Fixed the issue that the BRI related information would remain on web even the B2 module was removed.

2. Fixed iLBC codec issue: no sound during the call if using the iLBC codec.
3. Fixed noise issue on BRI trunk.
4. Fixed the issue that N412 users would see an IP address on their phones when an unknown call through PSTN trunk reached N412.
5. Fixed the recording issue: if you used transfer key on the IP phone to do transfer, the call could not be recorded.
6. Fixed the issue that the “match” value in pjsip.conf configuration file was incorrect.
7. Fixed the Recording log issue: the “Caller” in the recording log was incorrect if you using *3 to do transfer.
8. Fixed the issue that users could not play custom music file using a specific extension.
9. Fixed the issue that the VoIP trunks would remain in outbound/inbound routes even you had deleted the trunks in bulk.
10. Fixed outbound route issue: if an unreachable SIP trunk was selected in outbound route A, other outbound routes that were behind the outbound route A could not be used.

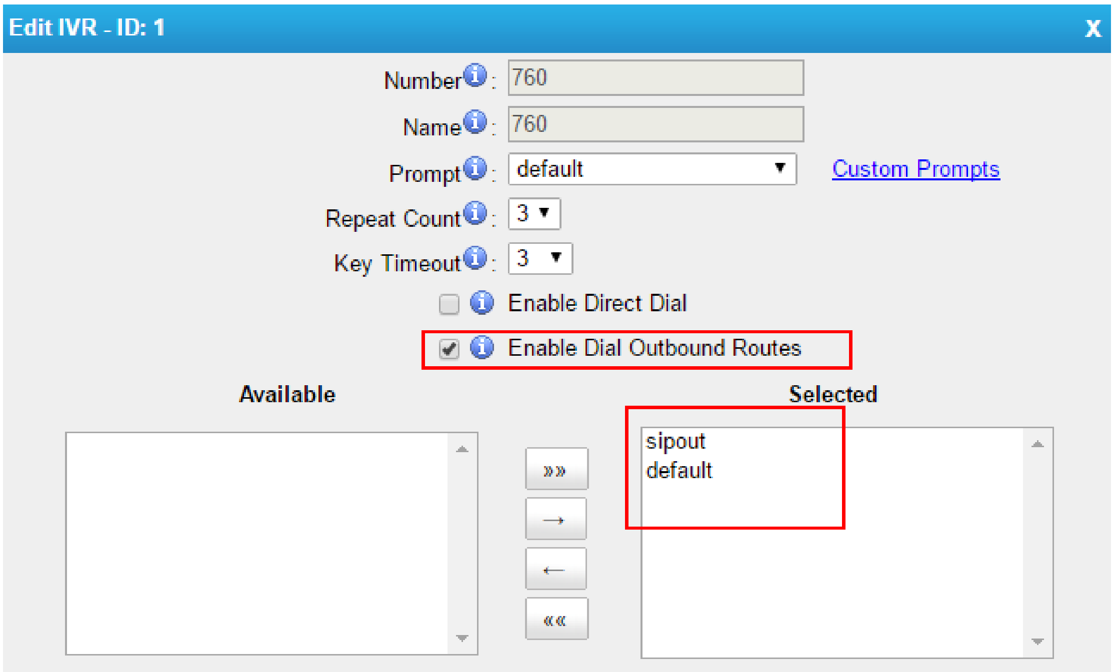
## NEW FEATURES (INSTRUCTION)

1. **Added support for switching Follow Me status using feature codes.**
   **Path:** PBX > Basic Settings > Feature Codes
   **Instruction:** dial the feature codes on your phone directly to switch the follow me status.
   **Examples:**
   - Dial *71 to forward all calls to voicemail; dial *71500 to forward all calls to number 500; dial *071 to disable forwarding of all calls.
   - Dial *72 to forward calls to voicemail when busy; dial *71500 to forward calls to number 500 when busy; dial *072 to disable when busy call forwarding.
   - Dial *73 to forward calls to voicemail when no answer; dial *73500 to forward calls to number 500 when no answer; dial *073 to disable no answer call forwarding.

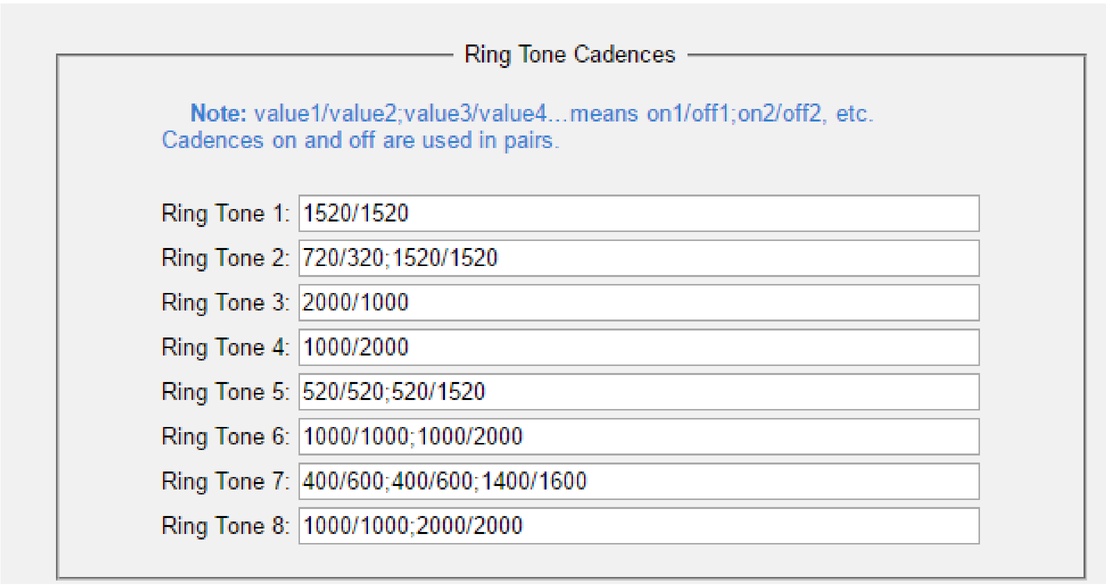
| Call Forwarding Preferences | |
|---|---|
| Reset to Defaults | *70 |
| Enable Forward All Calls | *71 |
| Disable Forward All Calls | *071 |
| Enable Forward When Busy | *72 |
| Disable Forward When Busy | *072 |
| Enable Forward No Answer | *73 |
| Disable Forward No Answer | *073 |
| Enable Do Not Disturb | *75 |
| Disable Do Not Disturb | *075 |

2. **Added support for dialing external numbers when in an IVR.**
   **Path:** PBX > Inbound Call Control > IVR
   **Instruction:** check the option “Enable Dial Outbound Routes” and choose the outbound route. When the caller enters the IVR, he/she can make an outbound

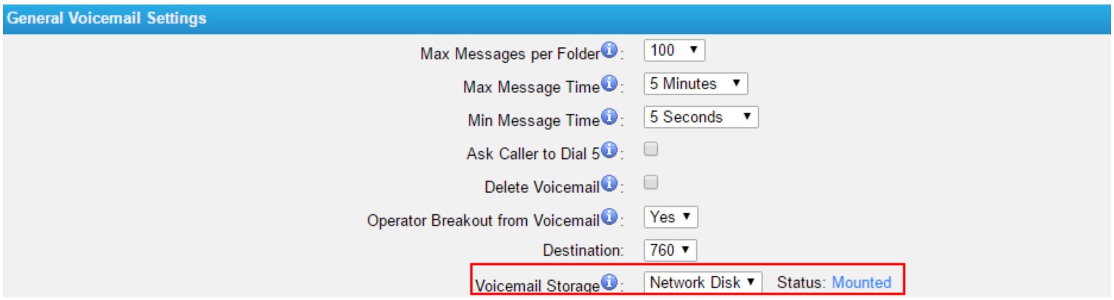
call through the PBX.




3. **Added support for customization of ring tones.**
   **Path:** PBX > Advanced Settings > Ring Tone
   **Instruction:** Customize the ring tone cadences then click “Save” and “Apply”.




4. **Added support for storage of Voicemail messages in the external drive.**
   **Path:** PBX > Basic Settings > Voicemail Settings
   **Instruction:** you can choose to store the voicemail messages to local flash, SD card or network disk.




5. **Added option "FXS Port Interdigit Timeout".**
   **Path:** PBX > Basic Settings > General Preferences
   **Instruction:** in order to allow sufficient time for a user to dial a telephone number, N824 relies on a timer referred to as the interdigit timeout. This parameter indicates the duration N824 waits after each digit is entered before it assumes the user has finished entering digits.

6. **Added support for "Turkey" FXO Mode.**
   **Path:** PBX > Basic Settings > General Preferences
   **Instruction:** you can choose the Turkey FXO Mode from the drop-down menu.

7. **Added option "CO Line DTMF Duration" to set the duration of a DTMF tone on a CO line.**
   **Path:** PBX > Basic Settings > General Preferences
   **Instruction:** this sets the duration of a DTMF tone on the CO line. The default value
   is "120".

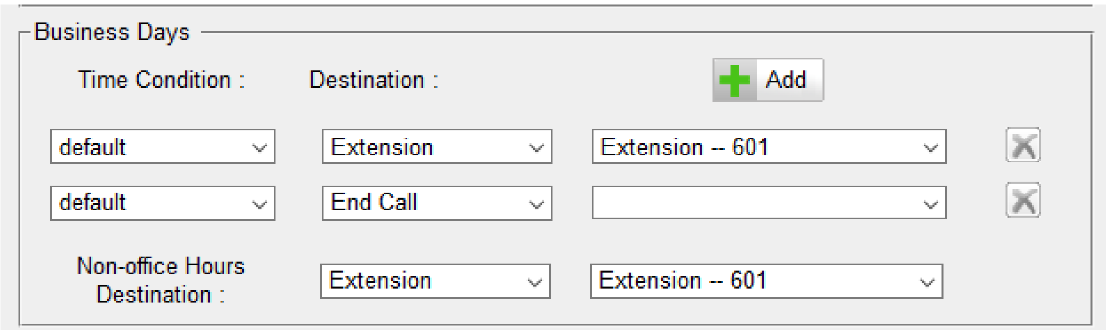
8. **Added option "CO Line DTMF Gap" to set the interval between each DTMF tone.**
   **Path:** PBX > Basic Settings > General Preferences
   **Instruction:** This sets the interval between each DTMF tone on the CO line. The default value is "120".
9. **Added "DTMF" Caller ID Signaling and "ETSI V23" FSK Caller ID type; added "Caller ID + Ring" and "Polarity + Caller ID + Ring" sending modes.**
   **Path:** PBX > Extensions and Trunks > Extensions > FXS Extensions > Advanced Settings
   **Instruction:**


10. **Added new ring strategy "Ring One Each Time" for Ring Group.**
    **Path:** PBX > Inbound Call Control > Ring Groups
    **Instruction:** "Ring One Each Time" means to ring only one available extension in the group each time sequentially.

11. **Added support for making three-way call by using Flash key on an analog phone.**
    **Path:** PBX > Basic Settings > General Preferences > Enable Three-way Calling
    **Instruction:** in the former firmware version, when you pressed hook key during an active call, you would start to use three-way calling, but could not transfer a call using the hook key. In the new version, when you press the hook key during a call, the system will treat it as transferring by default. If you want to make a three-way call, you can enable the option “enable three-way calling”.

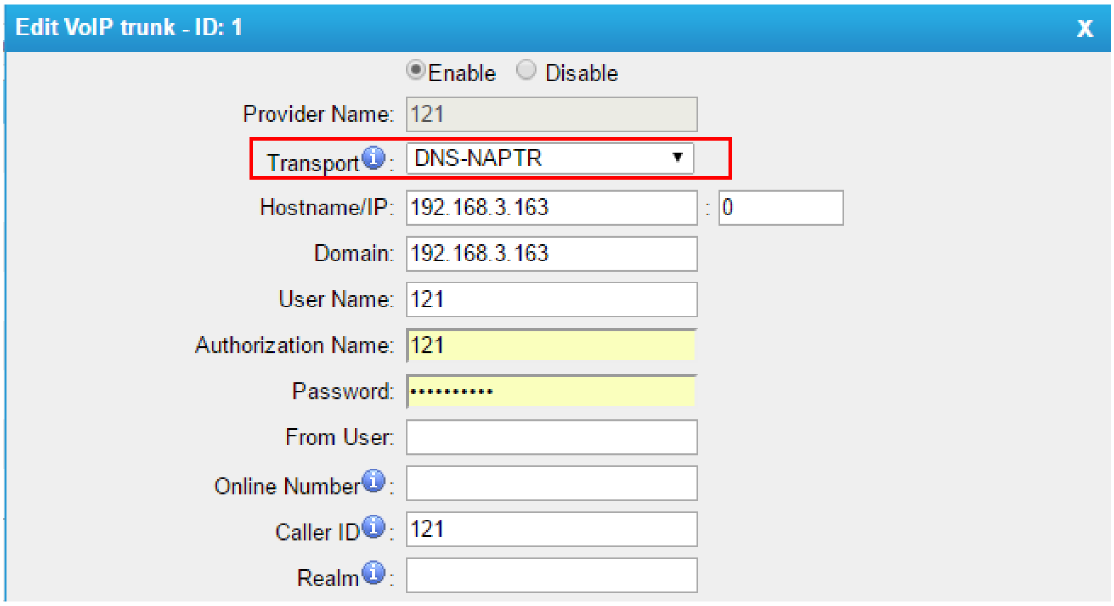
12. **Added option “Off-Hook Dial Delay” to set the delay on CO line between off-hook and dialing digits.**
    **Path:** PBX > Extensions and Trunks > Trunks > CO Lines > Other Settings > Off-Hook Dial Delay
    **Instruction:** this sets the delay on CO line between off-hook and dialing digits. The allowed range is from 0 ms to 5000 ms. Setting it to "0" means no delay. The default is "0".

13. **Added support for multiple time conditions and destinations on Inbound Route.**
    **Path:** PBX > Inbound Call Control > Incoming Rules
    **Instruction:** click “Add” to add new time condition and set the destination. The incoming calls will be routed to different destinations at different time.




14. **Added support for canceling attended transfer when the called person does not answer.**
    **Instruction:** if you make an attended transfer but the called person does not answer the call, you can dial the attended transfer feature code (*3) again to retrieve the call.
15. **Added option “DNS-NAPTR” to search SIP trunk transport, port and server.**
    **Path:** PBX > Extensions and Trunks > Trunks > VoIP Trunk
    **Instruction:**
    - If “Hostname/IP Address” is the PBX’s Hostname and the port is 0 or blank, NAPTR and SRV lookup will be executed to search for transport, port and server.
    - If “Hostname/IP Address” is a valid IP address or a designated port then UDP

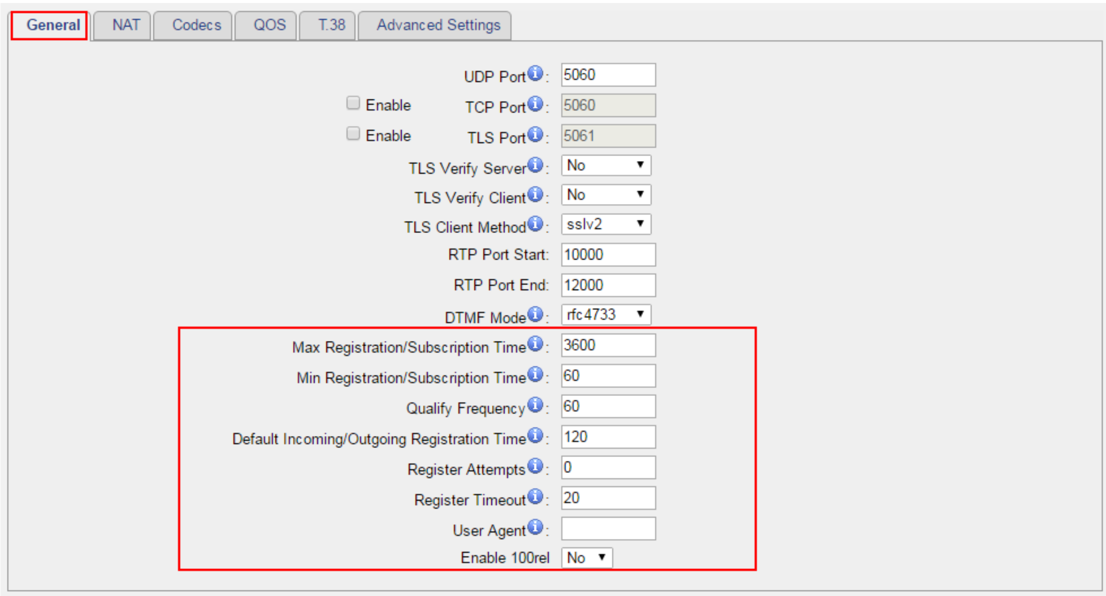
will be used.

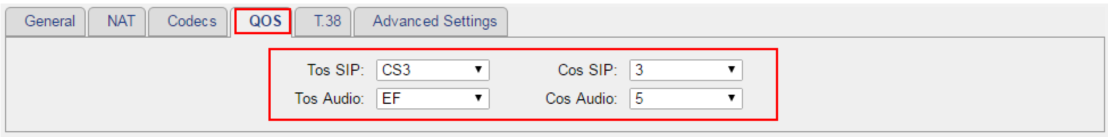


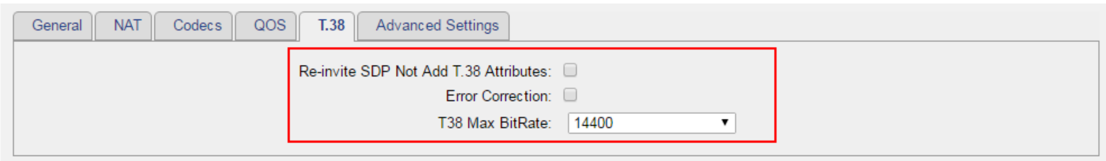

**16. Added SIP related settings.**

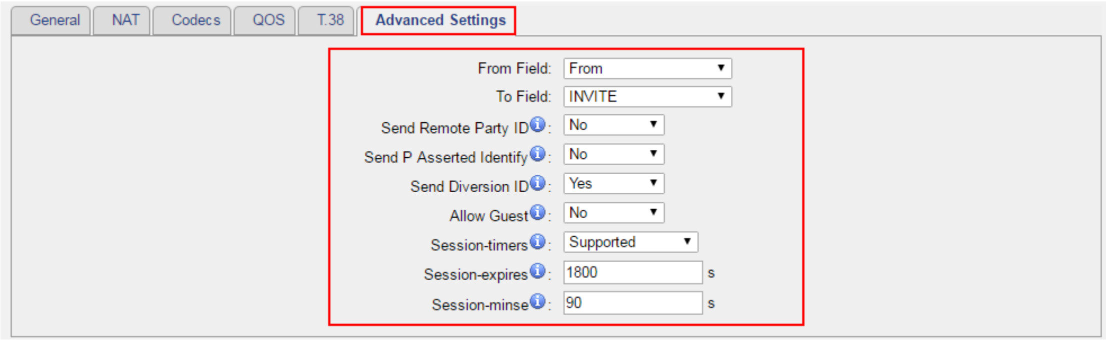
**Path:** PBX > Advanced Settings > SIP Settings

**Instruction:**




General
NAT
Codecs
QOS
T.38
Advanced Settings
From Field:
From
To Field:
INVITE
Send Remote Party ID:
No
Send P Asserted Identify:
No
Send Diversion ID:
Yes
Allow Guest:
No
Session-timers:
Supported
Session-expires:
1800
s
Session-minse:
90
s

# FIRMWARE VERSION 45.14.0.22

## DATE

May 18, 2016

## NEW FEATURES

1. Added enable/disable options for **three-way calling**.
   **Path:** PBX> Basic Settings > General Preferences
   **Instruction:** in the former firmware version, when you pressed hook key during an active call, you would start to use three-way calling, but could not transfer a call using the hook key. In the new version, when you press the hook key during a call, the system will treats it as transferring by default, if you want to make a three-way call, you can enable the option “enable three-way calling”.

## OPTIMIZATION

1. Added the detection of data accuracy and integrity on DSP; improved the stability of DSP.

## BUG FIXES

1. Fixed the issue that users could not check new voicemail message once the number of received voicemail messages reach 100.
2. Fixed the firmware upgrading failure issue when voicemail files were over a certain amount (when the flash space could not hold the firmware file).