Release Note for Yeastar N824

Yeastar Information Technology Co. Ltd.

FIRMWARE VERSION 42.14.0.23

DATE

July 14, 2016

CHANGES SINCE FIRMWARE RELEASE 42.13.0.27

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 support for China Mobile IMS SIP provider.
18. Added SIP advanced settings “From Field” and “To Field”.

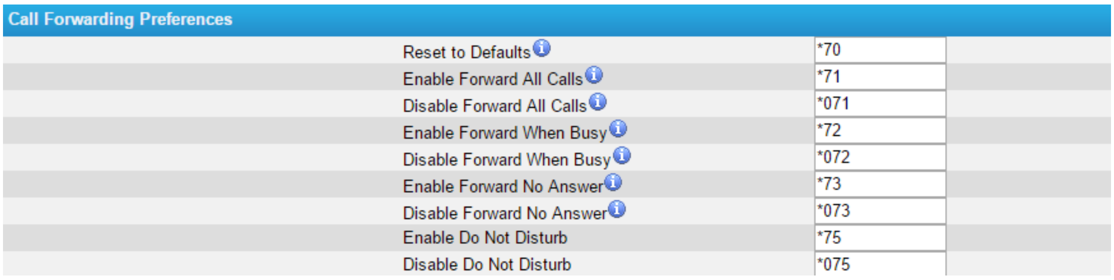
BUG FIXES

1. Fixed Network Disk issue: could not successfully mount network disk.
2. Fixed Service Provider Trunk issue: if entering an IP address in the “Hostname/IP” field and entering a domain in the “Domain” field, the trunk could not be registered.
3. Fixed the issue when you dial 9 to size a CO line and make outbound calls, it would take long time to reach the destination number.
4. Fixed the issue that the system would automatically make outbound calls through CO lines.
5. Fixed Ring Group issue: could not create a second ring group.
6. Fixed the issue that if the CO line “Answer Detection” and “Hangup Detection” were set to “Polarity”, and you call out through the CO line by using a SIP extension, you

would not be able to hear the ring back tone.

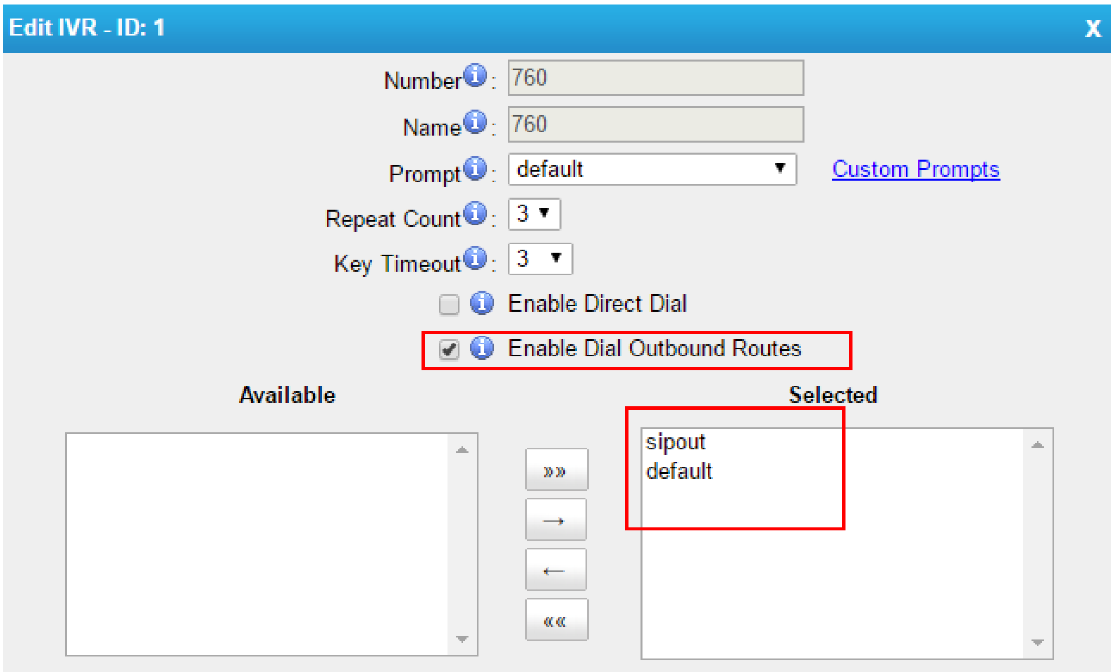
NEW FEATURES (INSTRUCTION)

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.

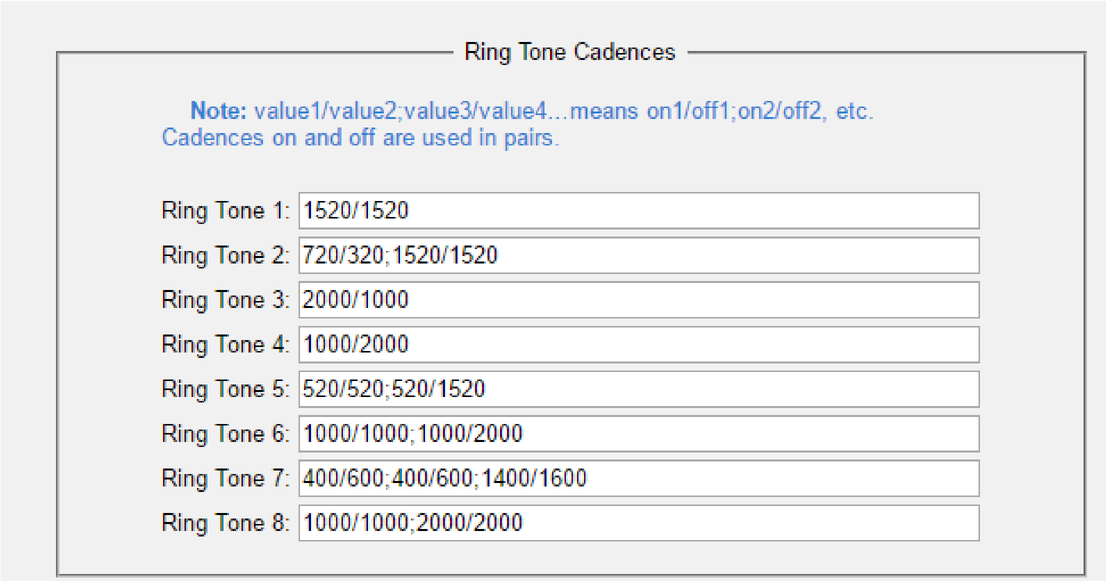


Call Forwarding Preferences	
Reset to Defaults	*70
Enable Forward All Calls	*71
Disable Forward All Calls	*071
Enable Forward When Busy	*72
Disable Forward When Busy	*072
Enable Forward No Answer	*73
Disable Forward No Answer	*073
Enable Do Not Disturb	*75
Disable Do Not Disturb	*075

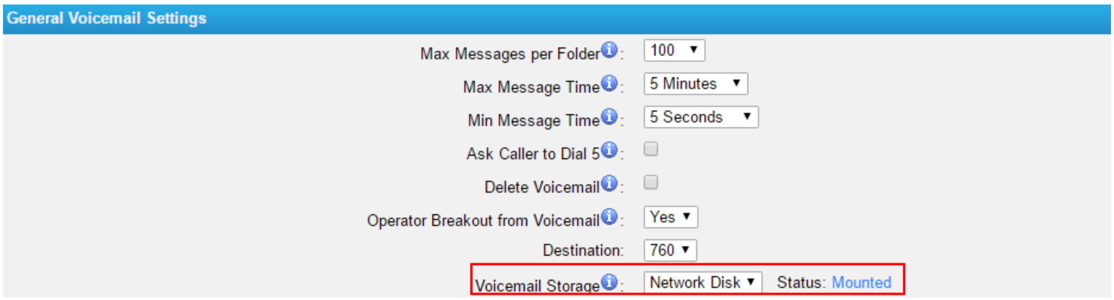
2. **Added support for dialing external numbers when in an IVR.**
   Path: PBX > Inbound Call Control > IVR
   Instruction: check the option “Enable Dial Outbound Routes” and choose the outbound route. When the caller enters the IVR, he/she can make an outbound call through the PBX.



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.**

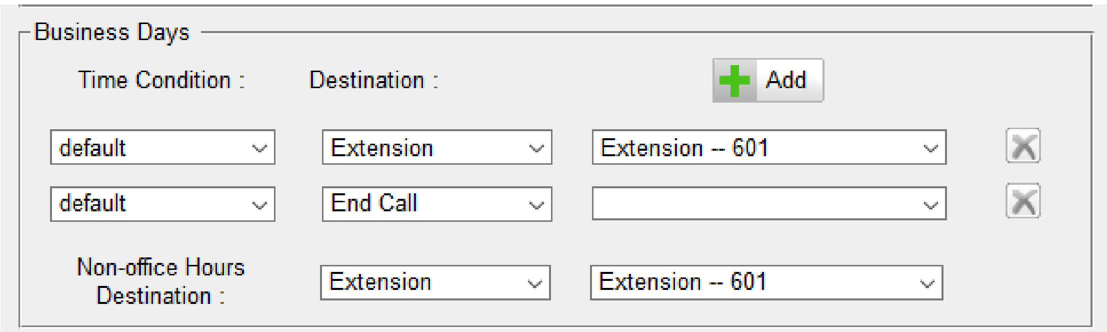
    Path: PBX > Extensions and Trunks > Trunks > CO Lines > Other Settings > Off-Hook Dial Delay

    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".

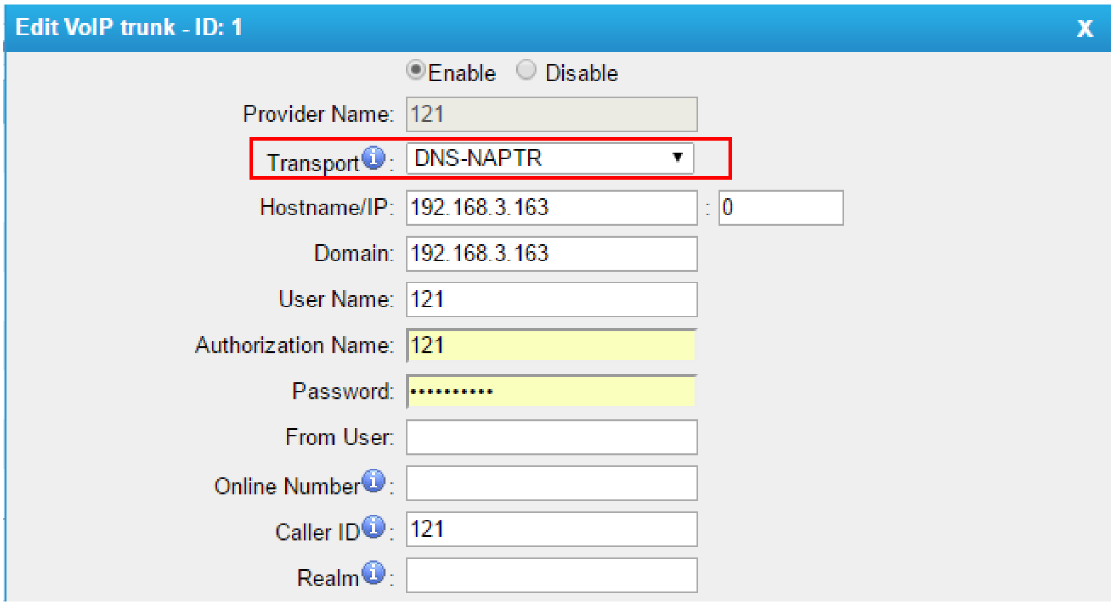
13. **Added support for multiple time conditions and destinations on Inbound Route.**

    Path: PBX > Inbound Call Control > Incoming Rules

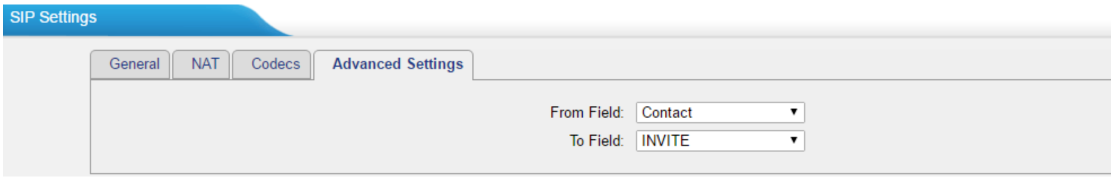
    Instruction: click "Add" to add new time condition and set the destination. The incoming calls will be routed to different destinations at different time.



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 advanced settings "From Field" and "To Field".**
    Path: PBX > Advanced Settings > SIP Settings > Advanced Settings
    Instruction: change the two settings if the N824 cannot get the correct Caller ID or DID information from the received SIP packet.
    - **From Field:** define from which SIP field N824 will get the Caller ID information.
    - **To Field:** define from which SIP field N824 will get the DID information.


SIP Settings
General
NAT
Codecs
Advanced Settings
From Field:
Contact
To Field:
INVITE