



Release Note for Yeastar N824

Yeastar Information Technology Co. Ltd.

FIRMWARE VERSION 42.14.0.23

DATE

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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	i	*70
Enable Forward All Calls	i	*71
Disable Forward All Calls	i	*071
Enable Forward When Busy	i	*72
Disable Forward When Busy	i	*072
Enable Forward No Answer	i	*73
Disable Forward No Answer	i	*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.

The screenshot shows the 'Edit IVR - ID: 1' configuration window. Fields include Number (760), Name (760), Prompt (default), Repeat Count (3), and Key Timeout (3). The 'Enable Direct Dial' checkbox is unchecked, while 'Enable Dial Outbound Routes' is checked and highlighted with a red box. Below this, there are two list boxes: 'Available' (empty) and 'Selected' (containing 'sipout default'), with the 'Selected' box also highlighted by a red box. Navigation buttons (»», →, ←, ««) are located between the list boxes.

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”.

Ring Tone Cadences

Note: value1/value2;value3/value4...means on1/off1;on2/off2, etc.
Cadences on and off are used in pairs.

Ring Tone 1: 1520/1520

Ring Tone 2: 720/320;1520/1520

Ring Tone 3: 2000/1000

Ring Tone 4: 1000/2000

Ring Tone 5: 520/520;520/1520

Ring Tone 6: 1000/1000;1000/2000

Ring Tone 7: 400/600;400/600;1400/1600

Ring Tone 8: 1000/1000;2000/2000

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.

General Voicemail Settings

Max Messages per Folder: 100

Max Message Time: 5 Minutes

Min Message Time: 5 Seconds

Ask Caller to Dial:

Delete Voicemail:

Operator Breakout from Voicemail: Yes

Destination: 760

Voicemail Storage: Network Disk Status: Mounted

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:

Caller ID Settings

Caller ID Signalling: DTMF

Sending Mode: Ring + Caller ID + Ring

DTMF Duration: 120 ms

DTMF Gap: 120 ms

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.

The image shows a 'Business Days' configuration window. It contains three rows of settings. The first row has 'Time Condition' set to 'default', 'Destination' set to 'Extension', and a value of 'Extension -- 601'. The second row has 'Time Condition' set to 'default', 'Destination' set to 'End Call', and an empty value field. The third row is for 'Non-office Hours Destination' with 'Extension' and 'Extension -- 601'. There is an 'Add' button with a green plus sign and two 'X' buttons to remove rows.

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.

The image shows the 'Edit VoIP trunk - ID: 1' configuration window. It has an 'Enable' radio button selected. The 'Provider Name' is '121'. The 'Transport' dropdown menu is highlighted with a red box and set to 'DNS-NAPTR'. The 'Hostname/IP' is '192.168.3.163' and the port is '0'. The 'Domain' is '192.168.3.163'. The 'User Name' is '121'. The 'Authorization Name' is '121' and the 'Password' is masked with dots. There are also fields for 'From User', 'Online Number', 'Caller ID' (set to '121'), and 'Realm'.

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