Release Note for S-Series VoIP PBX

Yeastar Information Technology Co. Ltd.

Contents

FIRMWARE VERSION 30.6.0.16

DATE: January 11, 2018

CHANGES SINCE FIRMWARE RELEASE 30.5.0.30

NEW FEATURES

1. Added Multisite Interconnect feature.
2. Added a new SIP trunk type “Account Trunk”.
3. Added templates of SIP trunk.
4. Added support for answering an intercom/paging call.
5. Added support for importing configurations of SIP register trunks.
6. Added support for importing configurations of outbound routes and inbound routes.
7. Added support for configuring IAX jitter buffer on Web page.
8. Added a default rule of outbound restriction: an extension user can make a maximum of 5 calls in 1 minute.
9. Added FTP access permission to the voicemail files and one-touch recording files which are stored in PBX local flash.

OPTIMIZATION

1. Updated the following system prompt: English prompt, Korean prompt, and Hebrew prompt.
2. Optimized the Web user interface: the PBX’s GUI language will be synchronized to your browser language.
3. Optimized call recording: the administrator can choose whether to record internal calls.
4. Optimized blacklist feature: the administrator can restrict specific extension users from calling the blocked numbers.
5. Optimized SLA feature: the administrator can enable or disable the SLA prompt.
6. Optimized backup feature: the administrator can choose to backup system configurations, custom prompts or call logs.
7. Added support for adding network drive for Windows domains.
8. Optimized call forwarding: if an extension is unregistered, incoming calls to the extension will be forwarded to the “No answer” destination.
9. Optimized CDR Auto Cleanup: when the CDR number reaches 90% of the Max Number of CDR, the system will send notification.
10. Optimized Auto Recording Cleanup: the system will send notification when the usage of the device is about to reach the pre-configured maximum value.
11. Optimized LDAP feature: when importing LDAP, the system can automatically add new phonebook nodes.
12. Optimized LDAP feature: you can order the contacts by contact names.
13. Improved the security of shared files on cHar: the administrator can add permitted IP addresses to access the shared files.
14. Optimized billing feature: the administrator can configure whether to disconnect a user’s call if the user’s balance is insufficient.

15. Optimized Distinctive Ringtone setting: you can enter uppercase letters, lowercase letters, numbers, "-", "_" and "+".
16. Optimized Notification Contact settings: if you select a notification method without setting the email address or phone numbers, the system will not save the settings.
17. Optimized OpenVPN settings: you cannot upload a certification file with incorrect format.
18. Optimized Call log searching feature.
19. Optimized VLAN Priority setting: the default VLAN priority is 0.
20. Optimized SLA feature: Users: users can dial numbers start with "*" through an SLA trunk.
21. Optimized voicemail prompts.

BUG FIXES

1. Fixed the issue that Mobility Extension user could not change the status of time condition.
2. Fixed call log issue: when an extension and the mobility extension were ringing simultaneously, you answered the call on the extension or on the mobility extension; the system would generate 3 call logs.
3. Fixed the issue that if you switched the network type of a 4G module, the system would count the cellular data incorrectly.
4. Fixed the issue of GSM/3G/4G module: a GSM/3G/4G module could not handle calls and SMS simultaneously.
5. Fixed the issue of GSM/3G/4G module: after rebooting the PBX, the GSM/3G/4G modules could not work.
6. Fixe "Email to SMS" issue: The system could not send long emails to multiple destination numbers.
7. Fixed system email issue: the default TLS port for Sina emails could not work.
8. Fixed DOD issue on mobility extensions: if you use a mobility extension to make calls, the system would not send the DOD number of the extension.
9. Fixed the issue that after receiving a packet of cancelling Register from Panasonic, the Yeastar system would break down.
10. Fixed the prompt of Dial by Name.
11. Fixed voicemail prompt: if you set the inbound route to voicemail, you could not hear the prompt of "Ask Caller to Dial 5".
12. Fixed All Busy Mode for SIP Forking: the feature only worked for the called extension, not worked for calling extension.
13. Fixed ring group issue: if a member of the ring group canceled "Always Forward" by dialing *071; the member still could not receive calls.
14. Fixed voicemail feature: if "Ask Caller to Dial 5" was enabled, users could not dial 0 to exit voicemail.
15. Fixed BRI trunk issue: when making a call through a BRI trunk, the user could not get a prompt immediately if the called number is busy.
16. Fixed BRI trunk issue: importing DOD numbers could not work.
17. Fixed the compatibility issue with China Mobile IMS trunk: when making a call through the IMS trunk, the calling user could not get a prompt if the called person cancelled the call.
18. Fixed the compatibility issue with Loxone SIP Intercom system.

FIRMWARE VERSION 30.5.0.30

Note:

1. **This version has greatly enhanced the security mechanism; we strongly suggest all users upgrade to this new version.**
2. **After upgrading to version 30.5.0.30, to make sure Linkus can be properly used, please upgrade Linkus App to 1.4.7 in App Center.**
3. **If you are using Hot Standby, please disable it first, and upgrade the two PBXs separately bef ore enabling Hot Standby again. Otherwise, the PBX's extensions will fail to register after the upgrade.**

DATE: September 22, 2017

CHANGES SINCE FIRMWARE RELEASE 30.5.0.8

NEW FEATURES

1. Added API feature: Third-party applications can get IPPBX information via the API. This feature is supported in S50, S100, S300.

OPTIMIZATION

1. Reduced background noise when playing system prompts and global recording in Linkus Mobile Clients.
2. Optimized Web interface login security: after five unsuccessful login attempts in the Web interface, instead of locking the account for 10 minutes, the IP address will be added into the firewall blacklist.
3. Optimized extension registration security: when failed to register extension using the same IP address for five times in five minutes, the IP address will be added into the firewall blacklist.

BUG FIXES

1. Fixed Attended Transfer issue on Linkus Mobile Client: when Linkus Mobile Client user A (extension A) made an outbound call through the GSM trunk, and transferred it to the extension B during the call using the “Attended Transfer” feature, then user A pressed the “Transfer” button before extension B could press the “Answer”, the call would be hung up..
2. Fixed the issue that DTMF on Polycom Phones did not take effect.
3. Fixed the issue that when administrator sent emails containing a QR code for users to login their Linkus Mobile Client account, some emails would display garbage characters.

FIRMWARE VERSION 30.5.0.8

DATE: July 12, 2017

CHANGES SINCE FIRMWARE RELEASE 30.4.0.25

NEW FEATURES

1. Added FTP access for voicemail, One-Touch recording, Auto Recording and CDR when those are stored in external storage like TF card, USB drive or internal hard disk.
2. Added SIP Message by using MESSAGE method communicated between SIP phones.
3. Added Caller ID name modification for SIP trunk DOD settings. Note that Global DOD has been removed, and “Caller ID Number” and “Caller ID Name” have been added on trunk edit page.
4. Added All Busy Mode for SIP Forking. When one SIP terminal registers one extension is called and is already in call, the other SIP terminals registered to the same extension will be marked as busy and can’t be called. Outgoing call will not be restricted.
5. Added support for different extensions to use the same email address. Note that when email address has been used by different extensions, the user Web interface will only be logged in with extension number.
6. Added support for DSS Key type – Call Park of Yealink phones. Call can be parked directly by pressing the Call Park key as well as monitoring and pressing to retrieve parked call.
7. Added support for counting the duration from “Queue receives call” to “Agent answers call”.
8. Added 2 VLAN subinterfaces for LAN and WAN.
9. Added DTMF pass-through for FXS when connected with door phone.
10. Added warning to create backup when trying to upgrade.
11. Added CDR query condition for Caller ID name. There will have a separate column of Caller ID name in the CDR CSV file that you download.
12. Added support for Auto Recording file, One-Touch Recording file, and CDR backup in real time to stand-by device in Hot Standby configuration.
13. Added silence detection for FXO, BRI and E1.
14. Added new Italian and German Web language.
15. Added support for saving Web configuration by pressing Enter.
16. Added Local Loop trunk for IP address 127.0.0.1.
17. Added BLF monitoring support for Time Condition default feature code like *800.

OPTIMIZATION

1. Optimized SMTP authentication. Account authentication is added.
2. Optimized China PRI default configuration to match most of China PRI lines.
3. Optimized Web text on Adapt Caller ID page. “Dial Patterns” is replaced by “Adaptation Patterns”.
4. Optimized Adapt Caller ID feature. Now it is also available for Ring Group and Queue.
5. Optimized Web text in Event Center. “Storage Space Full” is replaced by “Storage Full”.
6. Added feature conflict prompt that when Char and Hotel App is enabled at the same time, the Mini Bar feature of Hotel will affected. Some feature codes will not work well.
7. Optimized the name of extension type when Char App is enabled. “Hotel Extension” is replaced by

"Normal Extension", and "Room Extension" is replaced by "cHar Extension".

8. Optimized the SIP qualify feature for Peer-Type SIP trunk. Some providers doesn't support OPTION to be qualified which makes the trunk status "unknown" and calls can't be made. Currently when Quality option is unchecked, it would show "unmonitored" on the PBX monitor page and calls are allowed to make.
9. Removed check box of Feature Code header and parking number range.
10. Added Italian SLA dial prompt.
11. Added prompt for Email to SMS: when country code is not working, try the customized country code.
12. Optimized PAI and Remote Party ID would be disabled by default.

BUG FIXES

1. Fixed Dial by Name issue: the recorded name can be played in the Dial by Name event.
2. Fixed Booking issue for Hotel App: when the room type 1 has been booked and room type 1 has one last room, this room still can be checked in by other person.
3. Fixed the "More" button display issue on Notification menu: "More" button would still be clickable when there is no notification.
4. Fixed the issue that voicemail couldn't be found when Caller ID name including "&" on the user Me page.
5. Fixed network disk issue that it couldn't be reconnected when it was disconnected due to unstable network.
6. Fixed the issue that users could still call into Queue when Join Empty and Leave When Empty is both checked. Note that users can still call into the Queue if any extension in the Queue has enabled Mobility Extension and Simultaneous Ring.
7. Fixed the issue that Ring Group members' status would still show ringing when the call had been answered by Pick Up.
8. Fixed incorrect CPU display when PBX has been up for many days.
9. Fixed the issue that lacking External Host Refresh Internal in SIP NAT settings would cause host parsing failure.
10. Fixed the issue that Only Keep Missed Call Records of Auto CLIP was not available for PSTN line. Note that this would work only when Answer Detection checked as Polarity as well as the line provider should deliver polarity signal.
11. Fixed the call recording file playback problem that call recoding file would often be played by the first extension number of PBX rather than the extension number selected.
12. Fixed no video problem caused by codec payload negotiation that video phone would include different payload. Note that currently the payload is fixed to 99 for PBX.
13. Fixed the issue that content of users.conf would be erased after executing command "astconfig".
14. Fixed the issue that paging initiator would also be paged when he was in the paging group that he paged.
15. Fixed the issue that DND status of Hotel APP page would not change when DND was enabled by a hotel extension.

FIRMWARE VERSION 30.4.0.25

DATE: May 23, 2017

NOTE

In this new firmware version, we enhanced the IPPBX security. This security update has many features that will make your Yeastar S-Series IPPBX more secure and protect it from attacks. We recommend you update the firmware as soon as possible.

CHANGES SINCE FIRMWARE RELEASE 30.4.0.6

NEW FEATURES

1. Added schedule reboot feature (Maintenance > Reboot).

OPTIMIZATION

1. Optimized outbound route security: the administrator can limit how many outbound calls the extension users can make during a specific time (Settings > PBX > Call Control > Outbound Restriction).
2. Optimized network setting: LAN port supports up to 2 IP addresses.
3. Optimized firewall feature: users can add firewall rules using domain names.
4. Optimized IP Auto Defense feature: users can add a rule for a port range.
5. Optimized Local SIP Port: the system has a default IP auto defense rule for the local SIP ports.
6. Optimized extension registration security: only the registered extension can dial out, direct IP call is forbidden.
7. Optimized extension registration security: added support for user agent registration authorization (Settings > General > SIP > Advanced > SIP > User Agent Registration Authorization). Fill in user agent prefixes in the User Agent Registration Authorization field, only the matched user agent can register extension to the IPPBX.
8. Optimized extension/trunk registration security: the system will not reply OPTIONS packets from unknown extensions or trunks.
9. Optimized Forgot Password feature: users need to provide correct extension number and extension email address to retrieve password.
10. Optimized extension User Password setting: the User Password can only be changed to a mixture of uppercase, lowercase and numbers.
11. Optimized Logs feature: users can download operation logs and event logs.
12. Optimized Logs Auto Cleanup feature: users can set the max size of total logs stored in the system.
13. Optimized CDR feature: users can choose to show “Caller IP Address” in the CDR logs.
14. Optimized System Logs: extension registration failed information will be recorded in system logs.
15. Optimized QueueMetrics Live Integration App: the recording files name will contain QueueMetrics UniqueID.

BUG FIXES

1. Fixed Operation Logs issue: if users edited extensions in bulk, the operation logs would not contain

what settings the users had edited.

2. Fixed Queue: the call duration time was incorrect.
3. Fixed Queue: “Ring in Use” was disabled, and the queue strategy wasn’t “Ring All”, but the agent who was in a call still could receive a new call.
4. Fixed the Ring Group: if there were more than 50 members in the group, and the ring strategy was “Ring All”, the system would not ring all the members if a call reached the ring group.

FIRMWARE VERSION 30.4.0.6

DATE: March 30, 2017

CHANGES SINCE FIRMWARE RELEASE 30.3.0.17

NEW FEATURES

1. Added Hotel App.
2. Added char utile h+ Integration App.
3. Added QueueMetrics Live Integration App.

OPTIMIZATION

1. Changed all the default system music on hold files to the Asterisk version 13 music on hold files. The music on hold files are authorized to use.
2. Updated French and Italian system Web GUI.
3. Updated English, Chinese, Turkish, Spanish, and Hebrew system prompts.
4. Optimized CDR and Recordings feature: the system will record the un-answered call if the queue agents didn't answer the call.
5. Optimized CDR and Recordings feature: the PBX administrator can delete and download call logs and recording files in batch.
6. Optimized CDR and Recordings feature: if an extension has a DOD number, the call log will display the extension number and the DOD number.
7. Optimized CDR and Recordings feature: users can click a column heading to sort the table in ascending or descending order.
8. Optimized CDR and Recordings feature: users can choose an extension to play recording file on the web interface.
9. Optimized Voicemail feature: if voicemails have been played or deleted on Web interface, the IP phone will synchronize the voicemail status.
10. Optimized Voicemail feature: users can change the voicemail system's "Busy Prompt" and "Unavailable Prompt" (Settings > PBX > General > Voicemail > Greeting Options).
11. Optimized Voicemail feature: when users call in the PBX's IVR, they can dial *02 to check voicemails. To achieve this function, you need to enable "Dial to Check Voicemail" on the IVR edit page.
12. Optimized User Permission: allow users to delete auto recording files; separate reboot permission and reset permission.
13. Optimized DOD Settings: users can import and export DOD numbers.
14. Optimized SIP Caller ID feature: different SIP trunks can get Caller ID from different SIP fields. There is a global "Get Caller ID From" setting on "PBX > General > SIP > Advanced" page. You can set a different "Get Caller ID From" for a specific SIP trunk on the trunk edit page.
15. Optimized SIP DID feature: added support for getting DID number from "Diversion" and "P-Asserted-Identify" (PBX > General > SIP > Advanced).
16. Optimized SIP Outbound SIP Registrations Setting: the "Default Incoming/Outgoing Registration Time" is set to 1800 seconds.

17. Optimized Call Transferring: “Send Diversion ID” (PBX > General > SIP > Advanced) will work for extension “Call Forwarding” and transfer feature codes (*03, *3). When an incoming call is forwarded or when using *03 or *3 to transfer a call, the original caller’s number will be displayed.
18. Optimized SLA feature: users can choose a failover destination for a SLA trunk.
19. Optimized Event Center: added “Network Drive Lost Connection” notification setting.
20. Optimized Static Route: users can choose adding static route to LAN gateway or WAN gateway.
21. Optimized System Email Password Setting: the password setting are allowed to enter any special characters.
22. Optimized Auto Cleanup settings: the default “Max Number of CDR” on S20/S50/S100 is 200000, the default “Max Number of CDR” on S300 is 50000. The default value of “CDR Preservation Duration”, “Files Preservation Duration” and “Recordings Preservation Duration” are set to 0, meaning no limit.
23. Optimized Auto Cleanup feature: when the CDR reaches over 90% of the “Max number of CDR”, the system will send notification.
24. Optimized SIP feature: support requests of MESSAGE SIP packet (PBX > General > SIP > Advanced).
25. Optimized DID Settings: support DID numbers that start with digit 0 and support DID numbers that contain character “+”.
26. Optimized FXO Settings: users can change the FXO trunk’s “DTMF Duration” and “DTMF Gap” (PBX > General).
27. Optimized FXO Settings: users can manually release a FXO trunk on web interface (FXO Trunk > Advanced > Other Settings).
28. Optimized FXO Hangup Detection feature: support hang up a call by “Loop Current Disconnect” method. This method is used when connecting the S-Series IPPBX to a traditional PBX, and the traditional PBX uses this method for signaling call termination.

BUG FIXES

1. Fixed the issue that the PBX would continually reboot if the device power was disconnected during firmware upgrade.
2. Fixed BLF LED status issue: the BLF LED would stay in Green status after rebooting S-Series IPPBX.PBX.
3. Fixed Voicemail issue: the voicemails were not deleted automatically even the option “Delete Voicemail” was enabled.
4. Fixed Emergency Call issue: when the PBX reached the max number of concurrent calls, emergency calls could not work.
5. Fixed Time Condition issue: if time condition status was force changed by feature code, distinctive ring tones on inbound routes could not work.
6. Fixed Time Condition issue: the IP phone BLF key could not correctly indicate the time condition status.
7. Fixed the SMS issue: if an email contains Greek words, sending this email to SMS, the SMS contents would appear garbled words; Sending SMS to email also had this problem.
8. Fixed Email to SMS issue: if the email contains Polish words, sending email to SMS would fail.
9. Fixed compatibility issue with some VOIP ISP: making outbound calls with the VoIP trunk, the calls would be disconnected after 30 seconds.
10. Fixed DID number matching issue on inbound route: the system could not match DID number

correctly if there were multiple inbound routes configured with the same Peer-to-Peer SIP trunk and different DID numbers.

11. Fixed Hot Standby issue: after enabling Hot Standby feature, the system's default gateway would automatically be changed to LAN gateway.
12. Fixed Hot Standby issue: configurations on the primary server and standby server could not be synchronized.
13. Fixed Paging Group issue: if a user called the PBX's FXO trunk, and the call reached a paging group, all the group members hung up the call, but call was still connected.
14. Fixed the Auto Defense issue: if an IP was listed in blacklist due to SSH accessing failure, this IP would have no authority to access the PBX web interface.
15. Fixed Voicemail issue: system would reserve the prompt "please leave a message after the tone", even users had custom "Busy prompt" and "Unavailable prompt".
16. Fixed the Voicemail issue: the S-Series IPPBX would not send RTP packets when users were leaving voicemail messages, this would cause some ISP consider the PBX is not alive and disconnect the call.
17. Fixed File Share issue: Win 10 users could not access the shared files; Window users need to change the PC LAN Manager authentication level, or they cannot access the shared files.
18. Fixed the File Share issue: if the user first enabled file share feature then installed the external storage device, the file share feature would not work.
19. Fixed the issue that if ring group member or queue agent's extension was not registered, the extension and the mobility extension could not ring simultaneously.
20. Fixed the Outbound route issue: the system could not correctly match outbound route if multiple outbound routes were configured with the same dial pattern.
21. Fixed the Backup and Restore issue: restoring a S300 backup to S20 IPPBX, all the S300 extensions would be restored on S20.
22. Fixed the User Permission issue: a user with permission to check and download CDR and recordings could not check and download his/her own CDR and recordings.
23. Fixed Automatic Upgrade issue: check for update could not work properly.
24. Fixed CDR and Recordings issue: pickup calls were not recorded in the call log.
25. Fixed CDR and Recordings issue: if the research result includes tens of thousands of records, downloading the research results would fail.
26. Fixed NAT issue: if NAT method was set to a domain, and when the domain could not be resolved, the NAT page would be unreachable, debugging Asterisk also would fail.
27. 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.
28. Fixed the issue that extension IP Restriction feature could not work properly.
29. Fixed the IAX issues: IAX trunk's domain name could not automatically update; using IAX might cause memory leaks.

FIRMWARE VERSION 30.3.0.17

DATE: January 13, 2017

CHANGES SINCE FIRMWARE RELEASE 30.3.0.10

NEW FEATURES

4. Added support for French, Italian Web interface.
5. Updated Turkish Web interface.

BUG FIXES

10. Fixed Backup and Restore issue: if CDR was stored in hard disk, USB, or TF/SD card, the backup file could not be restored normally.
11. Fixed Backup and Restore issue: restored a S300 backup file to a S300 device, the network settings could not be restored.
12. Fixed the Storage File Share issue: enabled File Share feature and stored a backup file to hard disk, USB, or TF/SD card, the backup file could not be shared.
13. Fixed importing and exporting extensions issue: exported extensions and deleted extensions on the PBX, then imported the extensions, only one extension would be imported.
14. Fixed Inbound Route issue: chose the Fax Destination to an extension, then deleted the extension, the Fax Destination would display an “h”.
15. Fixed Time Condition issue: any extension user could reset the time condition.
16. Fixed Time Condition issue: if Time Condition Override was disabled, only the first inbound route could work.
17. Fixed CDR issue: the extensions users who have privilege to download other user’s CDR might be unable to check or download the relevant call logs.
18. Fixed the Call Transfer issue: if a user pressed the Transfer key on IP phone to transfer a call, and the other party didn’t answer the call, he/she would receive a “Recall” call.
19. Fixed Extension Status issue: an extension number was registered on multiple IP phones, and the extension status would display only one IP address.
20. Fixed Whitelist/Blacklist issue: a whitelist/blacklist could not be imported to PBX if the numbers contain letters.

FIRMWARE VERSION 30.3.0.10

DATE: January 4, 2017

CHANGES SINCE FIRMWARE RELEASE 30.2.0.27

NEW FEATURES

1. Added support for Czech and Slovak system prompts.
2. Added support for Portuguese and Serbian web interface.
3. Added Billing App.
4. Added Emergency Number feature.
5. Added support for installing or updating App by uploading App installer package to S-Series IPPBX.
6. Added support for logging in Web user interface by user email address.
7. Added Backup Schedule feature.
8. Added support for sending Email to SMS to multiple mobile phone numbers.
9. Extension users can configure their own whitelists to allow only trusted numbers to call their extensions.
10. Added support for adapting caller ID.
11. Added support for sharing hard disk / TF card / SD card / USB drive.
12. Added “Delete Voicemail” and “Dial 5 Prompt” settings for Voicemail.
13. Added support for displaying the original caller ID when you do attended transfer.
14. Added support for limiting VoIP trunk’s maximum channels.
15. Added “DID Number” setting for PSTN/GSM/3G trunk.

OPTIMIZATION

22. Updated Simplified Chinese, English and Spanish web interface.
23. Updated English, Hebrew and Chinese system prompts.
24. Upgrade the NTP server version to 4.2.8p8.
25. Supports to add a maximum of 5 prompts for an IVR.
26. Only permitted extensions can override the time condition. (Go to “Settings > PBX > General > Feature Code > Time Condition”, click “Set Extension Permission” to select the extensions.)
27. When a user picks up a call, he can see the incoming caller ID on his phone.
28. The system will give a prompt when a mobile extension user calls in the S-Series IPPBX.
29. Limit the length of outbound route “Dial Pattern” to 63 characters.
30. The debug commands under SSH will be recorded in system logs.

BUG FIXES

19. Fixed call log issue: when the incoming call reached a ring group, and a user in the group answered the call, this call log would not be record in the system.
20. Fixed the SIP peer trunk issue: the trunk would not work if “Qualify” is disabled.
21. Fixed E1 trunk issue: if the E1 trunk signaling is set to mfc/r2, this trunk would be unavailable to use.

22. Fixed the issue that using a BLF key to monitor a trunk, and pressed the key to dial out, the system might use a wrong trunk to call out.
23. Fixed SLA issue: using a BLF key to monitor a SLA trunk, and pressed the key to use the trunk, the system would hang up.

NEW FEATURES (INSTRUCTION)

1. **Added Billing App.**

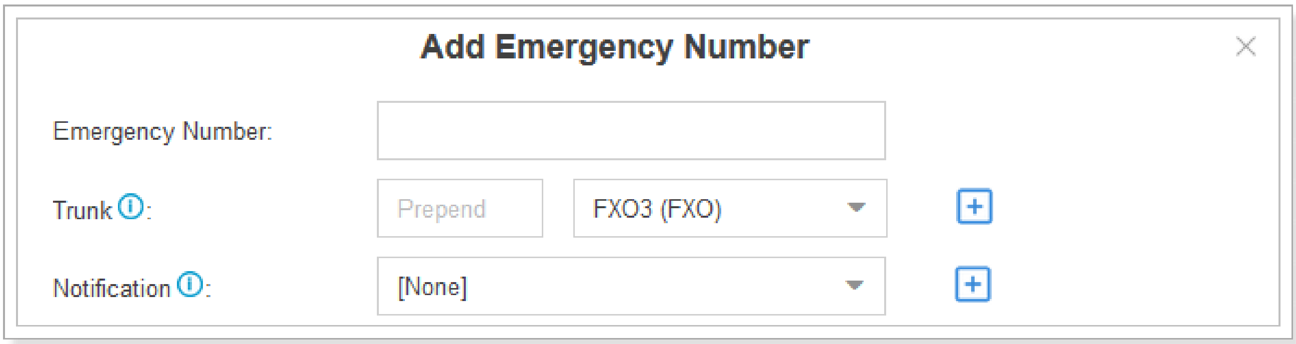
   Instruction: use the Billing App to allocate call credit and top up extensions, and conduct call analysis. Both prepaid and postpaid payments are supported. Rate can be set according to extensions, time periods, call duration, prefix number and number length. Real-time top-up history and statistics are all recorded.

   Go to "App Center" to install the Billing App and go to the Billing "General Settings" page, enable the billing feature, you will have 30-day free trial.

2. **Added Emergency Number feature.**

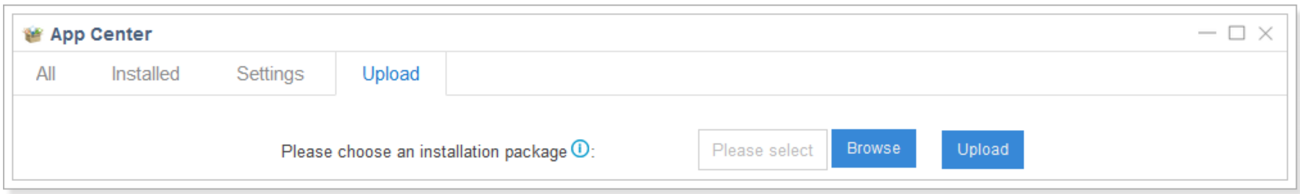
   Path: Settings > PBX > Emergency Number

   Instruction: enter the emergency number, and select a trunk to call the emergency number. You can also set a notification extension. When the emergency number is dialed, the system will make a notification call to the selected extension with a prompt.



3. **Added support for installing or updating App by uploading App installer package to S-Series IPPBX.**

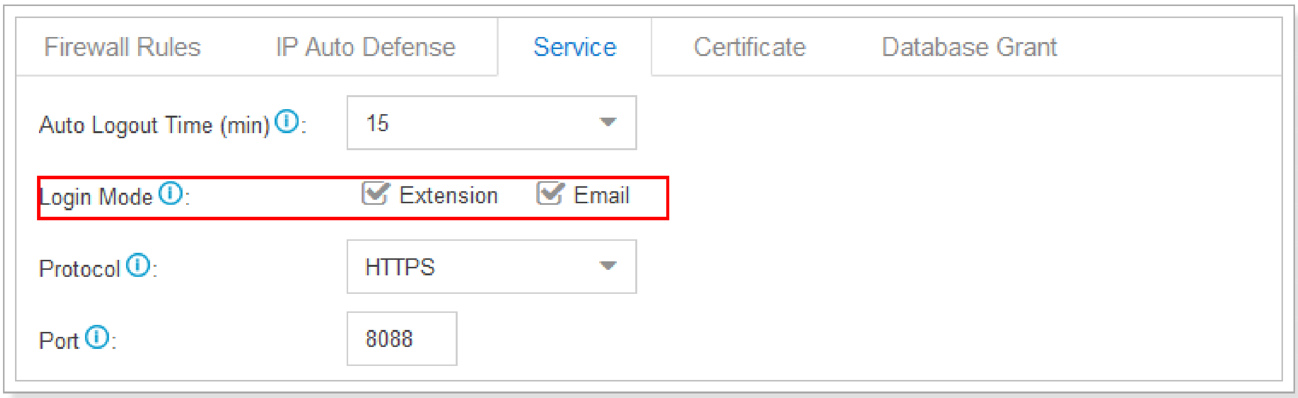
   Instruction: if the S-Series IPPBX cannot connect to the Internet, you can choose this method to install or update App. You can request the installer package from Yeastar support.



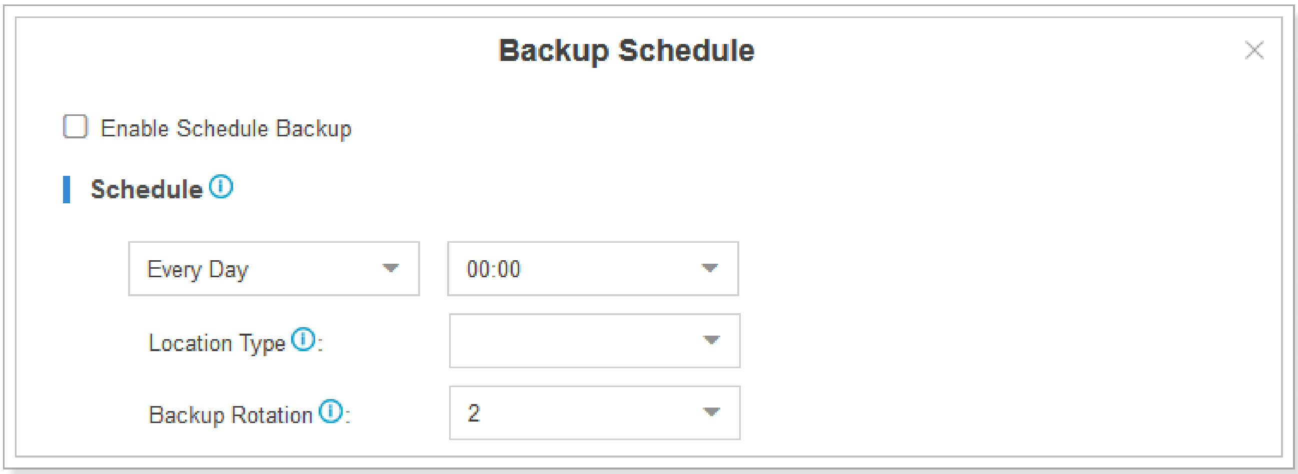
4. **Added support for logging in Web user interface by user email address.**

   Path: Settings > System > Security > Service

   Instruction: by default, the extension users need to log in Web user interface by their extension numbers. If checking the option "Email", then the extension users could log in by their bound email address.

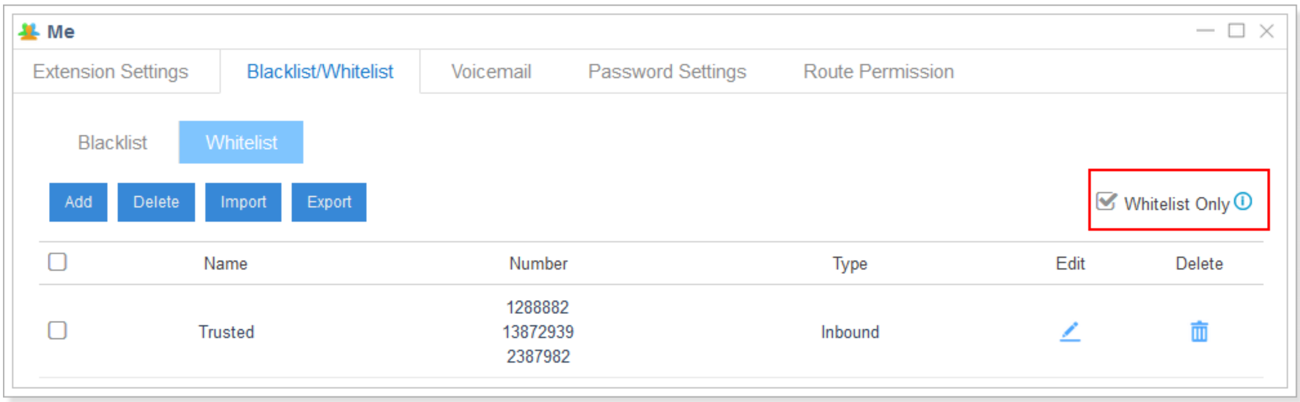


5. **Added Backup Schedule feature.**
   Path: Maintenance > Backup and Restore
   Instruction: check the option “Enable Schedule Backup” and set the Schedule details.

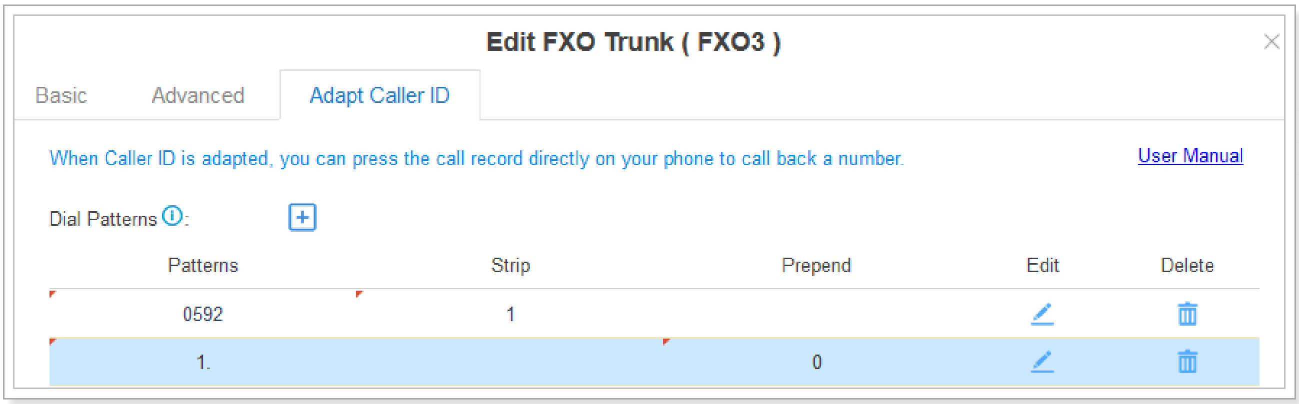


6. **Added support for sending Email to SMS to multiple mobile phone numbers.**
   Instruction: you can send email to SMS to a maximum of 5 mobile phone numbers, separate numbers with commas.

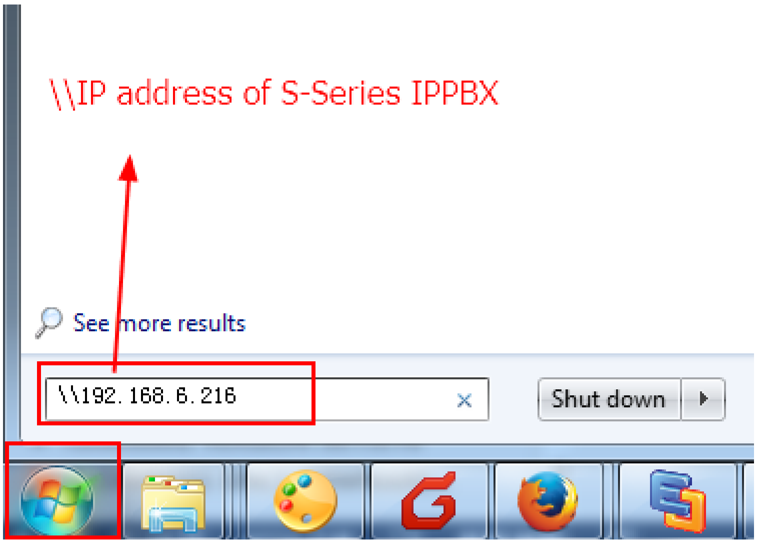
7. **Extension users can configure their own whitelists to allow only trusted numbers to call their extensions.**
   Instruction: log in S-Series IPPBX Web user interface by extension accounts, and go to configure the extension’s whitelist. Add trusted numbers in the whitelist, check the option “Whitelist Only”, then only the trusted numbers can call the extension. For example, if you add number 1000 in the whitelist, and the type is Inbound, then only 1000 can dial in and reach this extension.



8. **Added support for adapting caller ID.**
   Instruction: all the trunks on S-Series IPPBX support this feature. When Caller ID is adapted, you can press the call record directly on your phone to call back a number.
   <u>Examples:</u>
   a) If the incoming caller ID is 05925503302, but you need to dial number 5925503302 to call back. In this way, set Patterns to “0592.”, Strip 1 digit. When the incoming call reaches your phone, the caller ID shows 5925503302. You can use the number to call back directly.
   b) If the incoming caller ID is 15880278099, but you need to dial digit 0 before the number to call back. In this way, set Patterns to “1.”, set Prepend to “0”. When the incoming call reaches your phone, the caller ID shows 015880278099. You can use the number to call back directly.



9. **Added support for sharing hard disk / TF card / SD card / USB drive.**
   Path: Settings > System > Storage > File Share
   Instruction: if the hard disk / TF card / SD card / USB drive is being shared, the local users could access the shared folder to view and download the files.

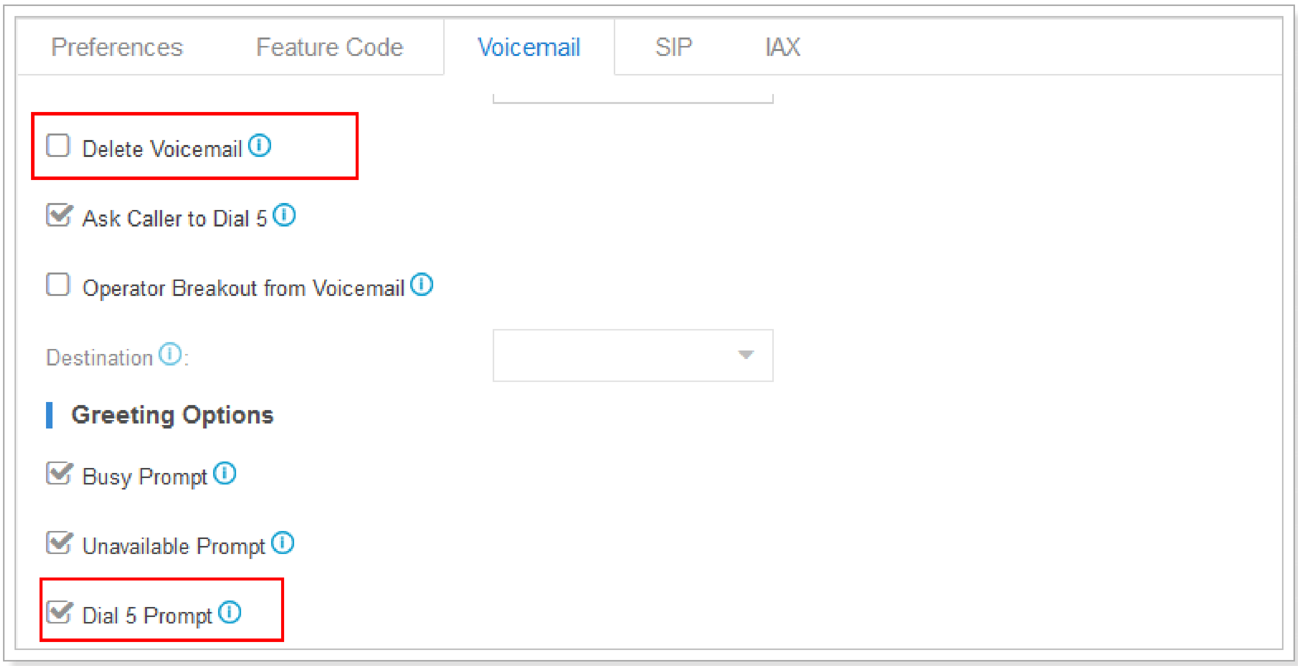


10. Added “Delete Voicemail” and “Dial 5 Prompt” settings for Voicemail.

Path: Settings > PBX > General > Voicemail

Instruction:

- **Delete Voicemail:** if enabled, the system will delete the voicemails that have been forwarded to email.
- **Dial 5 Prompt:** if the option is enabled, the prompt will be played to ask the caller to dial 5 before leaving a message.



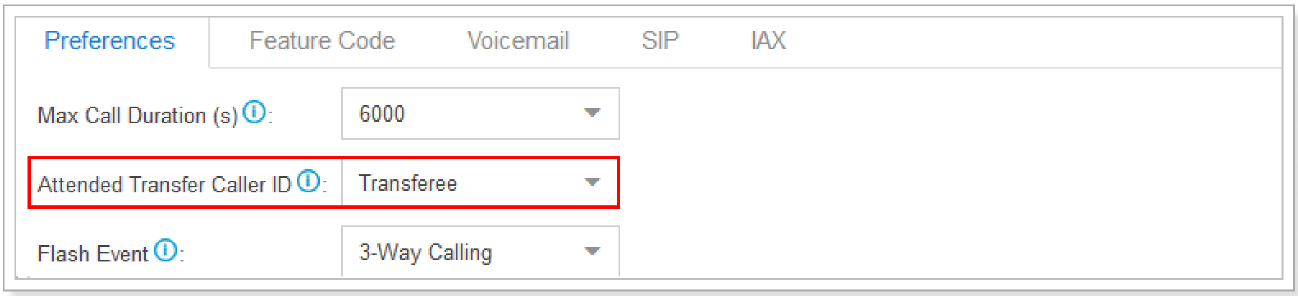
11. Added support for displaying the original caller ID when you do attended transfer.

Path: Settings > PBX > General > Preferences > Attended Transfer Caller ID

Instruction: User A and User B is in a call, User A makes an attended transfer to User C, when the call is established between User B and User C:

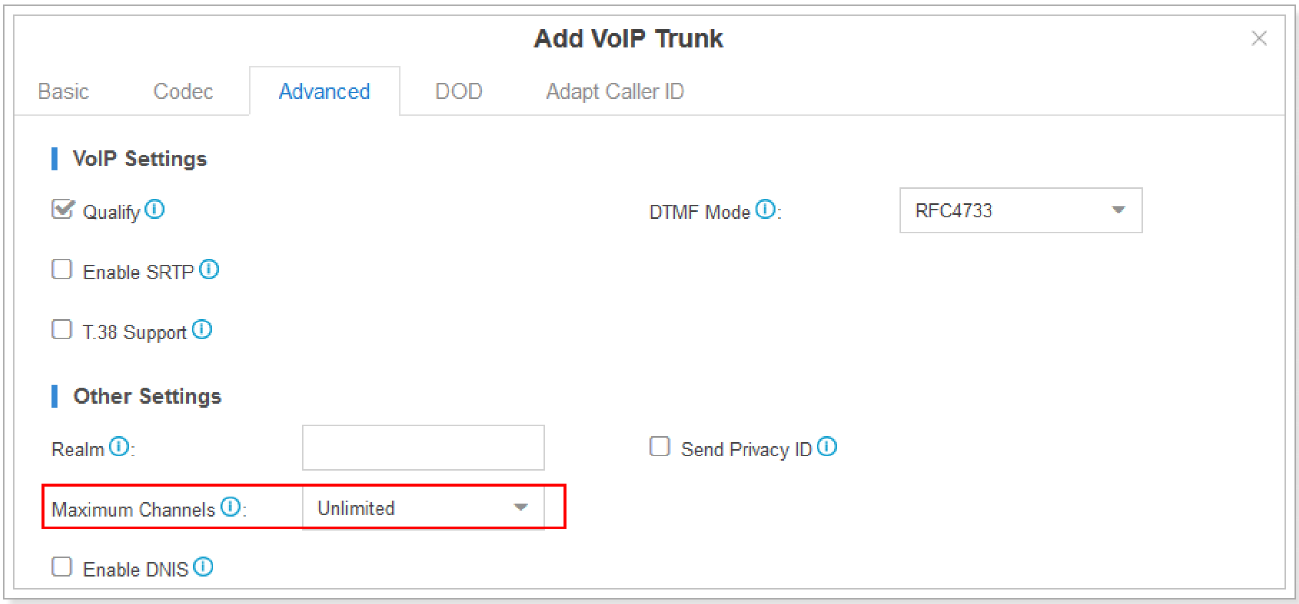
- If the Attended Transfer Caller ID is set to “Transferor”, User C’s phone will display User A’s number.

- If the Attended Transfer Caller ID is set to “Transferee”, User C’s phone will display User B’s number.



12. Added support for limiting VoIP trunk’s maximum channels.

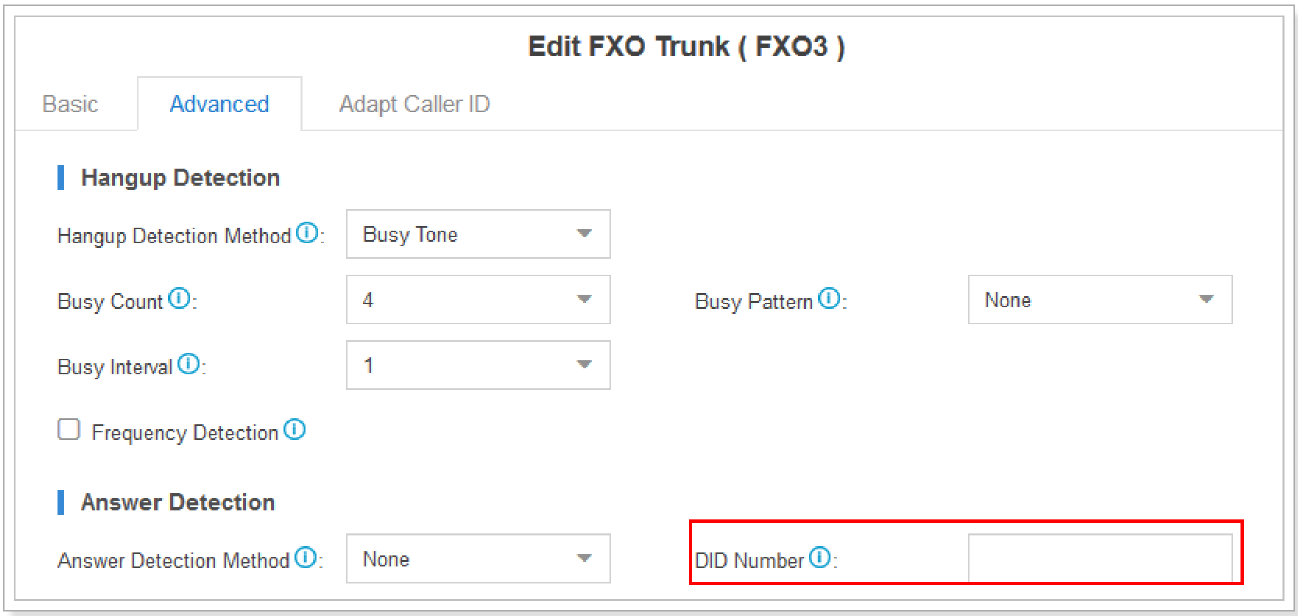
Instruction:



13. Added “DID Number” setting for PSTN/GSM/3G trunk.

Instruction: PSTN/GSM/3G trunk do not have DID numbers. This setting is a solution for transferring incoming GSM/3G/PSTN calls to an external number (Inbound-to-Outbound feature).

To make inbound-to-outbound feature work on GSM/3G/PSTN trunk, you need to set the DID number as the trunk number.


Edit FXO Trunk (FXO3)
Basic
Advanced
Adapt Caller ID
Hangup Detection
Hangup Detection Method:
Busy Tone
Busy Count:
4
Busy Pattern:
None
Busy Interval:
1
Frequency Detection
Answer Detection
Answer Detection Method:
None
DID Number:

FIRMWARE VERSION 30.2.0.27

DATE: November 21, 2016

CHANGES SINCE FIRMWARE RELEASE 30.2.0.8

NEW FEATURES

1. Added Linkus Application.

OPTIMIZATION

1. Optimized IVR: users could set Invalid Key event to a custom prompt. When the caller presses an invalid key, he/she could hear the invalid key prompt.

BUG FIXES

1. Fixed one touch recording issue: if one touch recording prompt was enabled, and the two parties both dialed *1 to finish one touch recording, that would cause the system to break down.

NEW FEATURES (INSTRUCTION)

1. **Added Linkus Application.**

   Instruction: Linkus is a Mobile Client that makes your mobile phone an office IP extension and links you and your colleagues and customers anywhere anytime. Make and receive calls through corporate phone network to slash call costs and enhance efficiency with consistent in-office experience.

   To use Linkus, the administrator need to install Linkus in S-Series PBX App Center, and enable Linkus feature.

   The extension users could install Linkus App in their phones and log in Linkus.
   [Learn more about Linkus and download Linkus App.](#)

FIRMWARE VERSION 30.2.0.8

DATE: November 2, 2016

CHANGES SINCE FIRMWARE RELEASE 30.1.0.16

NEW FEATURES

1. Added VPN Server Application.
2. Added Hot Standby feature.
3. Added support for deleting used records, keeping missed call records only, and phone numbers digits match for AutoCLIP route.
4. Added support for enabling/disabling VoIP trunks.
5. Added support for monitoring FXS and GSM/3G channels.
6. Added support for changing auto logout time.
7. Added IVR invalid key prompt setting.
8. Added support for enabling/disabling “Local SIP Port”.
9. Added one touch recording prompt settings.
10. Extension users could check their outbound route privileges on S-Series web user interface. Notice that the extension users could only check but could not edit outbound routes.
11. Added support for checking and downloading system prompt files via FTP. The access address is *ftp://S-Series’s IP address/*ysapps/pbxcenter/var/lib/asterisk/sounds/.
12. Added support for changing SSH username and password.
13. Added support for Deutsch web interface language.
14. Added support for Czech system prompt.

OPTIMIZATION

1. Optimized backup function: the backup file will contain custom prompts and music on hold files.
2. Extended the max supported log files captured on Ethernet Capture Tool.
3. Time search on CDR and Recording page is accurate to the seconds.
4. Increased the maximum number of conference members.
    - S20: 10
    - S50: 25
    - S100: 60
    - S300: 120
5. Increased the maximum number of pickup groups.
    - S20: 10
    - S50: 25
    - S100: 64
    - S300: 64

6. Limited the SIP UDP port setting: the port value could not fall in the range of local SIP ports.
7. Character limit for Trunk name: characters * ? are not allowed.
8. Character limit for Inbound route DID number: character + is allowed.

9. Character limit for Time Condition Name: characters ! $ ( ) \ / # ; , [ ] \ " = < > & ' ` ^ % @ { } | \ and blank character are not allowed.
10. Character limit for system email password: character $ is allowed.
11. The system will notify users on web if D30 module is removed.
12. When you upload Tiptel phonebook to S-Series PBX, you can choose phone type and upload different phonebooks.
13. Notification template could work for both SMS notification and email notification.
14. You can sort speed dial numbers.
15. Optimized Call Features and Conference Panel Web User Interface.
16. Updated Chinese system prompts.
17. Updated traditional Chinese web user interface.
18. There is no limit for SIP trunk registration attempts by default.

BUG FIXES

1. Fixed event center issue: when the PBX used Dual network mode, if you disconnected network cables for both LAN and WAN port, the system would not send notification.
2. Fixed CDR issue: if you change the E1 trunk name, the CDR would display an incorrect E1 trunk name.
3. Fixed the issue that if an extension user download call logs from web, he would download all the call logs on the system even he didn't have the privilege.
4. Fixed the SD card issue: the system would always warn you that the SD card format was wrong.
5. Fixed event center issue: if you use GSM/3G trunk to call out and the called party declined, the system would treat this call as a failed call and send you a notification.
6. Fixed VoIP trunk issue: if you enable proxy server, and disable it again, then save the changes, the VoIP trunk would not work properly.
7. Fixed conference issue: if a conference moderator extension is in an extension group, the moderator would be required to enter a password to enter the conference.
8. Fixed custom prompt issue: two prompt files would be created if you made a custom prompt using an extension.
9. Fixed conference issue: if you invited users to one conference, the system concurrent calls would be twice of the invited users.
10. Fixed SMS to Email issue: emails received in Foxmail would display incorrect received time.
11. Fixed one touch recording issue: if you changed extension A's default Caller ID, and use extension A call to extension B, press *1 to make one touch record; this call would not be recorded in CDR.
12. Fixed one touch recording issue: in an internal call, if the two extension users both press *1 to make one touch record; this call would not be recorded in CDR.
13. Fixed network drive issue: if the shared folder name is a Chinese name, the network drive could not be mounted to PBX.
14. Fixed Email to SMS issue: received contents in your mobile phone might be incomplete if you write too many contents in the email.
15. Fixed the temporary voicemail message issue.
16. Fixed holiday time condition issue: the system computed holiday time incorrectly.
17. Fixed LDAP phonebook issue: edit the default LDAP phonebook, set the settings as one new created phonebook settings, then all the phonebooks would be changed to the same phonebook

automatically.

18. Fixed compatibility issue with IE11 browser: users could not listen and download recordings on IE11.
19. Fixed queue issue: when you called in a queue, if you pressed any key on your phone when you were waiting for answer, the call would be disconnected automatically.
20. Fixed SIP trunk issue: for a registered and active SIP trunk, if the SIP provider always requires an Expires field in REGISTER packets, the SIP trunk would turn into an unavailable trunk.
21. Fixed queue issue: if the ring strategy was set to “Ring All”, some queue members might not ring at the same time.
22. Fixed PPPoE issue: if you set store system logs to one network drive, and changed network mode to PPPoE, the system could not obtain network.
23. Fixed VLAN issue: if network mode was DHCP, VLAN could not work.
24. Fixed inbound route DID issue: if DID number starts with digit 0, the system would automatically delete digit 0.
25. Fixe whitelist/blacklist issue: if you entered a number range, the web interface would display incorrectly.
26. Fixed compatibility issue with IE11 browser: checking system information log would cause the page no response.
27. Fixed AutoCLIP route issue: if you set “Record Keep Time” to 0, the system CPU would reach 100%.
28. Fixed the issue that feature code *8 was not available.
29. Fixed Echo Cancellation issue: if you changed module on the PBX, echo cancellation could not work properly.
30. Fixed PBX center app issue: updating PBX center would cause all the extension being unavailable.
31. Fixed user permission issue: if you set the privilege as “Custom”, and granted “voice prompts” privilege for the user, the privilege would be invalid for the user.
32. Fixed recording prompt issue: if you set inbound route to one callback destination, the system did not play the recording prompt for the incoming calls.
33. Fixed connection issue between two S-Series PBXs: PBX1 set NAT mode to STUN, PBX2 set NAT mode to external IP address, internal calls between the two PBXs could not work properly.
34. Fixed the CDR issue: if an extension user pressed *1 to make one touch recording, he could not check the call log on web.
35. Fixed the issue that SIP Registration/Subscription settings could not take effect.
36. Fixed the issue that if virtual ring back tone was enabled on GSM/3G trunk, the trunk could not receive calls.
37. Fixed that Peer SIP trunk issue.
38. Fixed the issue that using wget command could not resolve domain under SSH.
39. Fixed the issue that custom configuration file extensions.conf could not work properly.
40. Fixed the issue that if an incoming call is forwarded to an external number, this call would not be recorded.
41. Fixed VoIP trunk issue: if “Hostname/IP” and “Domain” were set differently, the trunk could not work properly.
42. Fixed the issue that adding user permission would fail if there were 300 extension users on the PBX.
43. Fixed G729/G723/iLBC codec issue: multiple concurrent calls with the codecs would cause the system freezing up.
44. Fixed event center issue: if sending alerts failed, it might cause memory leaks.

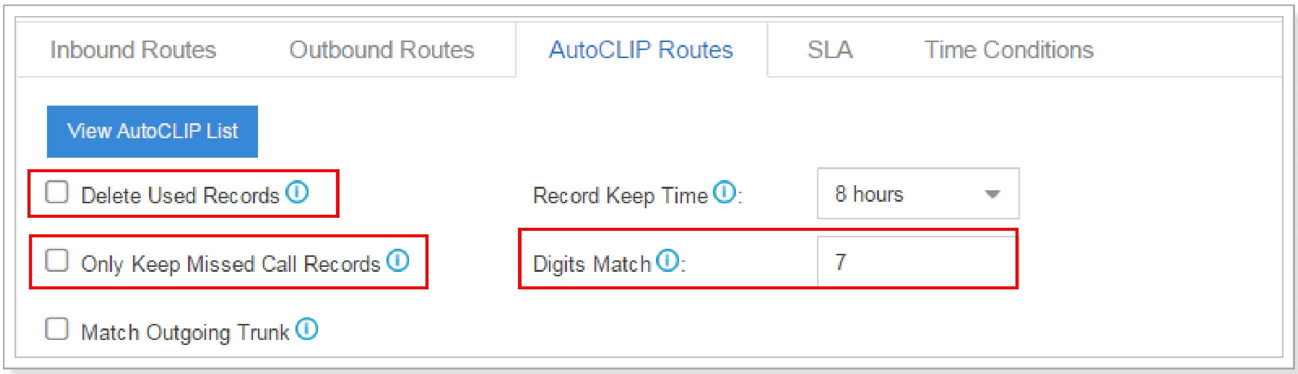
New Feature (Instruction)

2. **Added VPN Server Application.**
   Instruction: VPN Server application provides an easy VPN solution that turns your S-Series PBX into a VPN server. You can connect S-Series PBX at a remote location using the virtual VPN IP. The traffic between the PBX and remote network is encrypted, which improves the security of S-Series PBX. Enter App Center to download VPN server app, and you can see the app appear in main menu.

3. **Added Hot Standby feature.**
   Path: Settings > System > Hot Standby
   Instruction:
   The Hot Standby solution offers you the ability to provide high system availability, helping to prevent the unnecessary business loss caused by unexpected server failure.

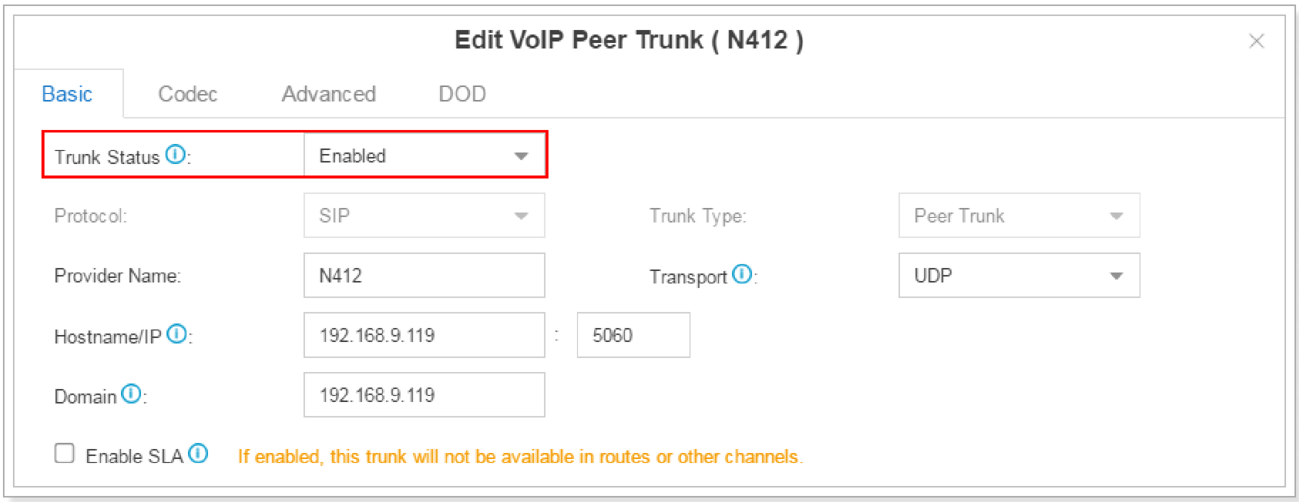
   You can set one S-Series PBX as a primary server, set another S-Series PBX as a secondary server. The hot-standby server (secondary server) can automatically and instantly take over if the primary server goes down. Callers still reach the people they need and business can continue as usual.

4. **Added support for deleting used records, keeping missed call records only, and phone numbers digits match for AutoCLIP route.**
   Path: Settings > PBX > Call Control > AutoCLIP Routes
   Instruction:

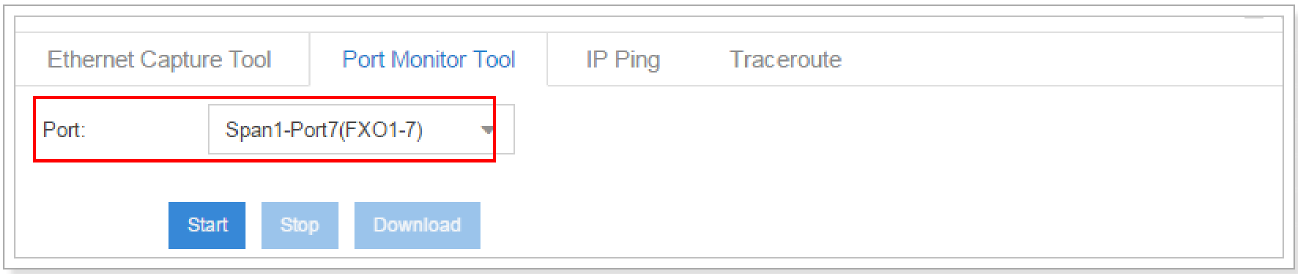


   - **Delete Used Records:** if enabled, when an AutoCLIP record is matched, it will be automatically deleted afterwards.
   - **Only Keep Missed Call Records:** if enabled, the system will only keep records of calls that are not answered by the called party in the AutoCLIP list. Note that PSTN line will keep records of all calls whether this option is enabled or disabled.
   - **Digits Match:** define how many digits from the last digit of the incoming phone number will be used to match the AutoCLIP record.

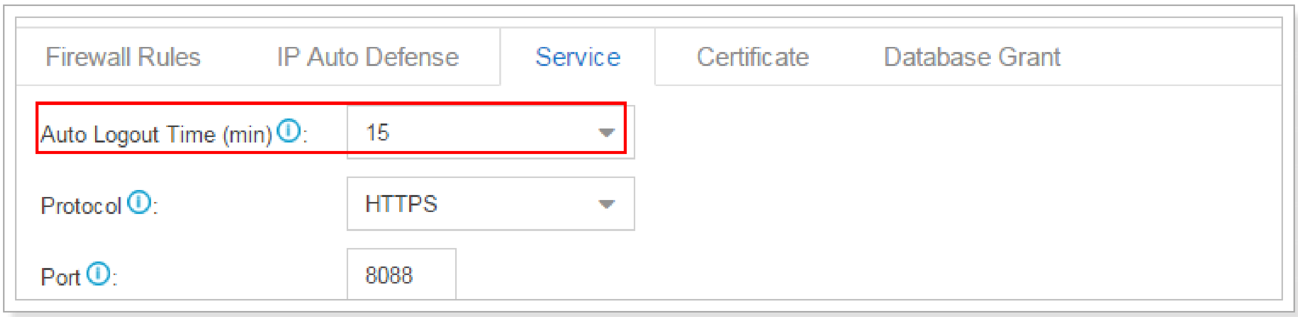
5. **Added support for enabling/disabling VoIP trunks.**
   Path: Settings > PBX > Trunk > VoIP Trunk
   Instruction: if the trunk is disabled, it is unavailable in outbound routes and inbound routes.



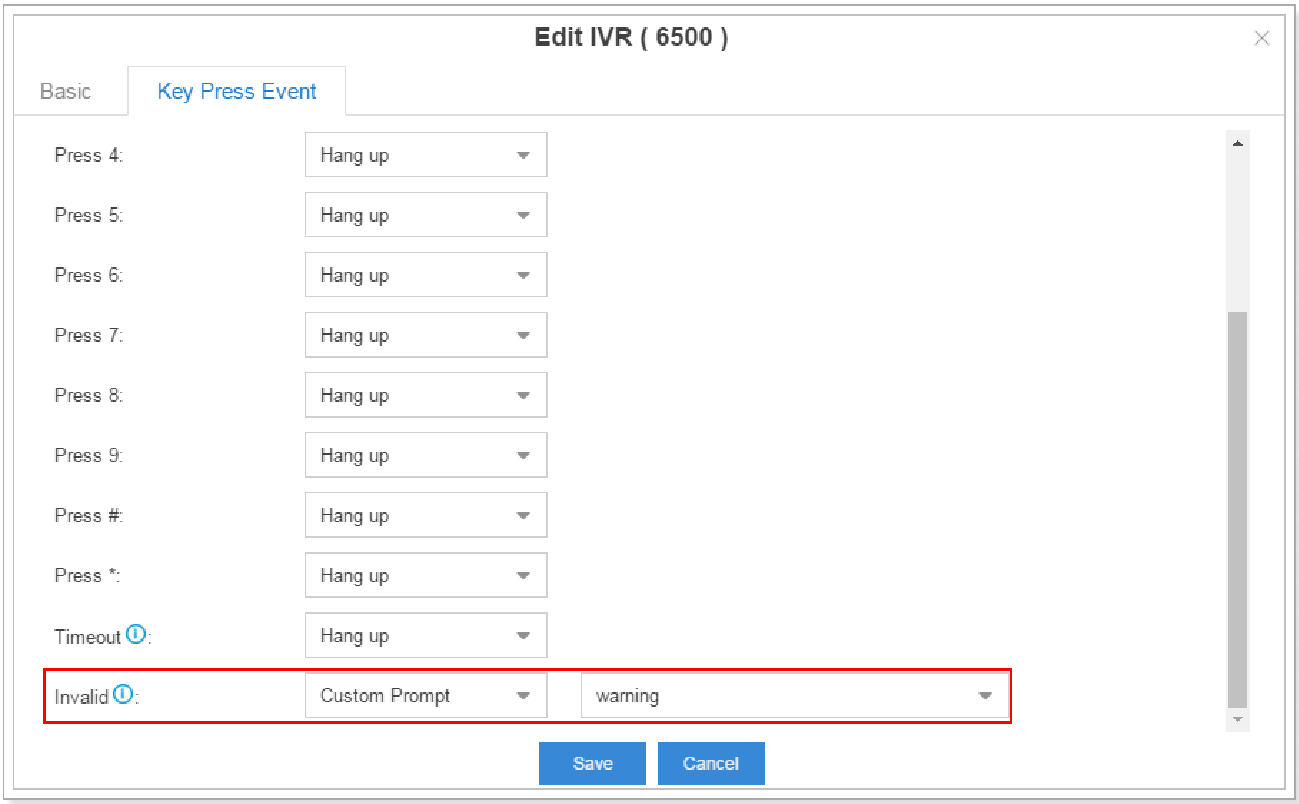
6. **Added support for monitoring FXS and GSM/3G channels.**
   Path: Maintenance > Troubleshooting > Port Monitor Tool
   Instruction:



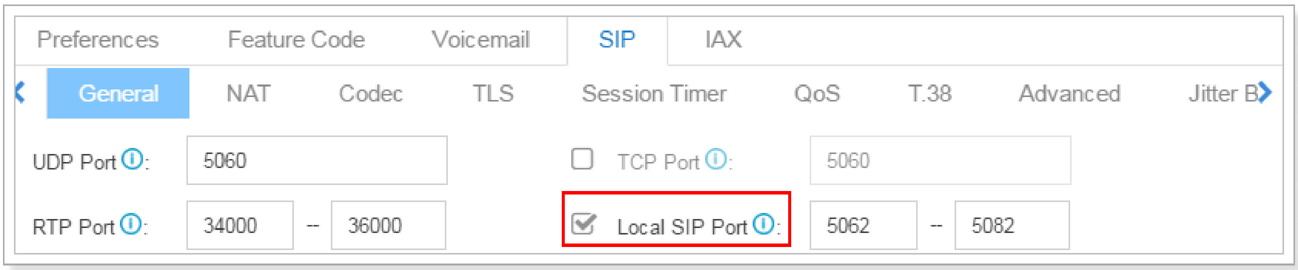
7. **Added support for changing auto logout time.**
   Path: Settings > System > Security > Service
   Instruction:



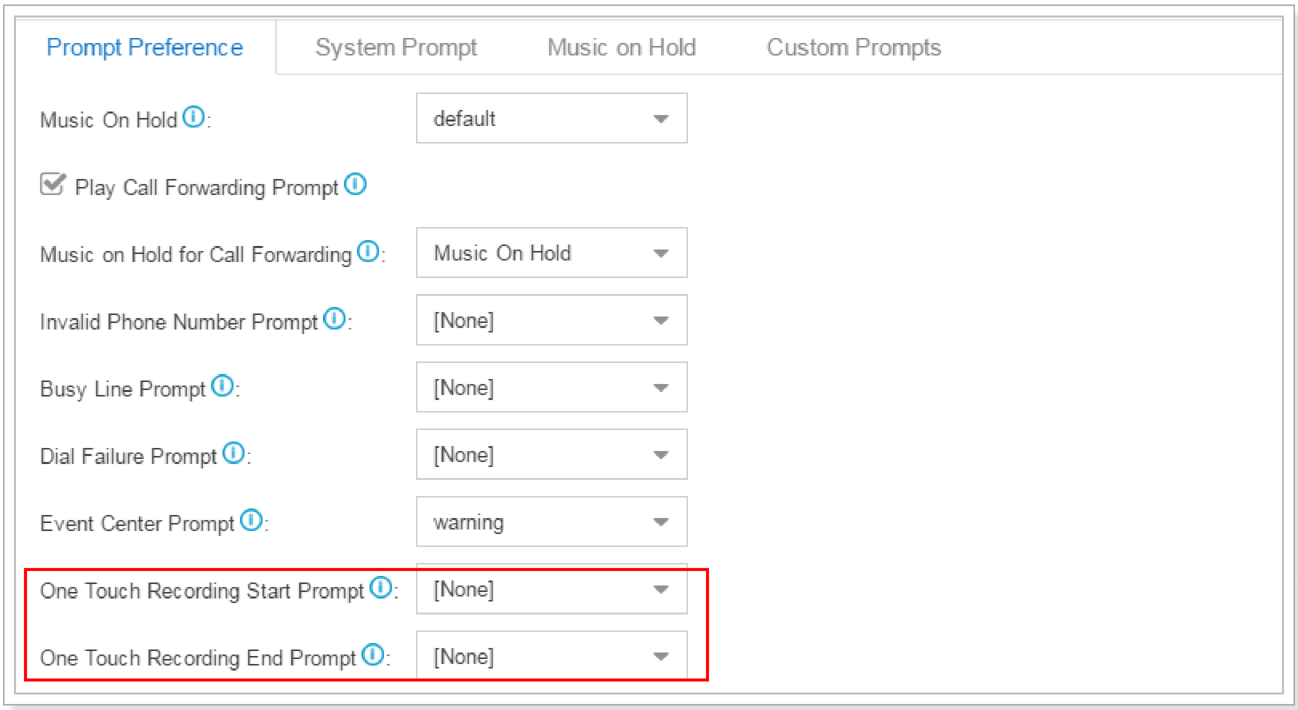
8. **Added IVR invalid key prompt setting.**
   Path: Settings > PBX > Call Features > IVR > Key Press Events
   Instruction:
   If you set invalid key event to a custom prompt, when the caller presses an invalid key, he will hear the prompt.



9. **Added support for enabling/disabling "Local SIP Port".**
   Path: Settings > PBX > General > SIP
   Instruction: some VoIP providers use a specific SIP port for registration but not a random SIP port, you need to disable this option to make S-Series PBX work with these VoIP providers.



10. **Added one touch recording prompt settings.**
    Path: Settings > PBX > Voice Prompts > Prompt Preferences
    Instruction: by default, if you use one touch record, the system doesn't prompt the call is being recorded. You can set once touch recording start prompt and end prompt here.


Prompt Preference
System Prompt
Music on Hold
Custom Prompts
Music On Hold:
default
Play Call Forwarding Prompt
Music on Hold for Call Forwarding:
Music On Hold
Invalid Phone Number Prompt:
[None]
Busy Line Prompt:
[None]
Dial Failure Prompt:
[None]
Event Center Prompt:
warning
One Touch Recording Start Prompt:
[None]
One Touch Recording End Prompt:
[None]

FIRMWARE VERSION 30.1.0.16

DATE: September 19, 2016

CHANGES SINCE FIRMWARE RELEASE 30.1.0.13

OPTIMIZATION

1. On the “Maintenance > Upgrade” page, the length of “HTTP URL” textbox is increased.

BUG FIXES

1. Fixed App Uninstallation issue: the pop-up confirmation dialog appeared twice when you tried to uninstall one app.

FIRMWARE VERSION 30.1.0.13

DATE: September 7, 2016

CHANGES SINCE FIRMWARE RELEASE 30.1.0.10

NEW FEATURES

1. Added support for "Auto Provisioning" App version 1.2.1. The new version added support for the following phones:
   - Cisco SPA301, SPA303, SPA501G, SPA502G, SPA504G, SPA508G, SPA509G, SPA512G, SPA514G, SPA525G2, CP7821
   - Polycom VVX101, VVX201, VVX300, VVX310, VVX400, VVX500, VVX600, VVX601, VVX1500, IP321, IP331, IP335, IP450, IP550, IP560, IP670
   - Htek UC902, UC903, UC923

BUG FIXES

1. Fixed SIM800 issue: PBX could not identify a new SIM800 module.
2. Fixed G729 Codec issue: making a call with G729 codec would cause latency.

FIRMWARE VERSION 30.1.0.10

DATE: August 29, 2016

CHANGES SINCE FIRMWARE RELEASE 30.1.0.7

NEW FEATURES

1. Added support for Polish Web User Interface language.

OPTIMIZATION

1. Updated Spanish Web User Interface language.
2. Updated Russian Web User Interface language.

BUG FIXES

1. Fixed the call parking issue: if you monitored a parking extension using BLF key on your phone, the BLF LED did not turn red when the extension was being used.
2. Fixed the Automatic Upgrade issue: the system would give a wrong prompt saying the checked failed because of network issues.
3. Fixed the FXO trunk issue: if you set the “Caller ID Start” as “After Ring”, set “Caller ID Signaling” as “DTMF”, incoming calls could not reach the FXO trunk.

FIRMWARE VERSION 30.1.0.7

DATE: August 23, 2016

CHANGES SINCE FIRMWARE RELEASE 30.0.0.40

NEW FEATURES

1. Added new App Conference Panel.
2. Added SLA feature.
3. Added new network mode “Single”.
4. Added Notifications on taskbar.
5. Added web alert when the PBX power is off or the PBX is off-line.
6. Added SIP “Jitter Buffer” settings.
7. Added support for importing and exporting extensions.
8. Added support for the following Tone Regions: Turkey, Korea, Serbia, Panama.
9. Added support for Spanish Web User Interface language.
10. Added conference status on PBX Monitor panel.
11. Added “Echo Cancellation” setting on GSM/3G trunk and FXO trunk.
12. Added support for registering SIP trunk with random SIP port.

OPTIMIZATION

1. If you do not change login password and set up an email, the system will remind you to do the configuration every time you log in.
2. Administrator account (admin) Password has an 8-63 character limit; the password must contain uppercase letters, lowercase letters and numbers.
3. Extension account Password has a 6-63 character limit; the following characters are not allowed: & ; " ' \ < > | ,
4. Your S-Series PBX’s system prompts would be updated to the newest version automatically after firmware upgrade.
5. If you change the system service port (like FTP, HTTP, SSH etc.), the system will remind you to reconfigure the firewall settings.
6. AutoCLIP routing could match incoming numbers with area codes and special character “+”.
7. Added “SMS to Email” and “Email to SMS” failure records in Event Log.
8. For Logs Auto Cleanup, the “Logs Preservation Duration” setting is for system logs, the “Max Number of Logs” setting is for operation logs.
9. The callback list will display each callback’s detailed destination.
10. Limit the Network Drive number:
    - S20 Maximum Network Drive: 2
    - S50 Maximum Network Drive: 2
    - S100 Maximum Network Drive: 4
    - S300 Maximum Network Drive: 4
11. The system will log you off from the web if you do not do any operation on the S-Series PBX web GUI in half an hour.

12. Added CPU temperature information in system logs.

BUG FIXES

1. Fixed the DNS SRV Lookup issue: if the DNS SRV analysis result was a domain, a VoIP trunk registered using the domain would not work.
2. Fixed VoIP 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 queue ring strategy issue: if a queue agent has registered his own extension number on multiple phones, incoming calls could only reach one of the agent’s phones.
4. Fixed the compatibility issue with VoIP provider netelip.
5. Fixed the issue that sending email to SMS would fail if the content exceeded the length limit.
6. Fixed the issue that the system would automatically make outbound calls through FXO trunks.
7. Fixed the Event Center issue: WAN port failure was not recorded in the event logs.
8. Fixed the dual mode network issue: if one Ethernet port used VLAN, the other Ethernet port could not work properly.
9. Fixed the IE11 compatibility issue: users could not play recording files on PBX web GUI.
10. Fixed the Holiday issue: the holiday worked based on Time Zone GMT +0.
11. Fixed the E1 trunk issue: the call quality was bad when using the E1 trunk.
12. Fixed the Event Center issue: the event center repeatedly recorded SIP trunk registration failure.
13. Fixed the Call Log issue: if you configured DOD numbers on a SIP trunk, and made outbound calls through the SIP trunk, the “Call From” value would be wrong in call logs.

INSTRUCTION (NEW FEATURES)

1. **Added new App Conference Panel.**
   Instruction: enter App Center, you will find the new App Conference Panel. This Conference Panel could realize all conference management based on Web. You can initiate and administrate a conference on Web.

2. **Added SLA feature.**
   Path: Settings > PBX > Call Control > SLA
   Instruction: the Shared Line Appearance (SLA) feature helps users share VoIP trunks and FXO trunks. It also helps monitor the status of the shared line. When an outgoing call is made by the user, all members in the SLA group are informed about the call and will be blocked from this line appearance until the line goes back to idle state or the call is put on hold. When an incoming call is received, all the members are informed of it and may join it depending on the line appearance linked with the SLA extension.

3. **Added new network mode “Single”.**
   Path: Settings > System > Network > Basic Settings
   Instruction: if you choose Single mode, only LAN port will be used for uplink connection. WAN port is disabled. The default network mode is Single.

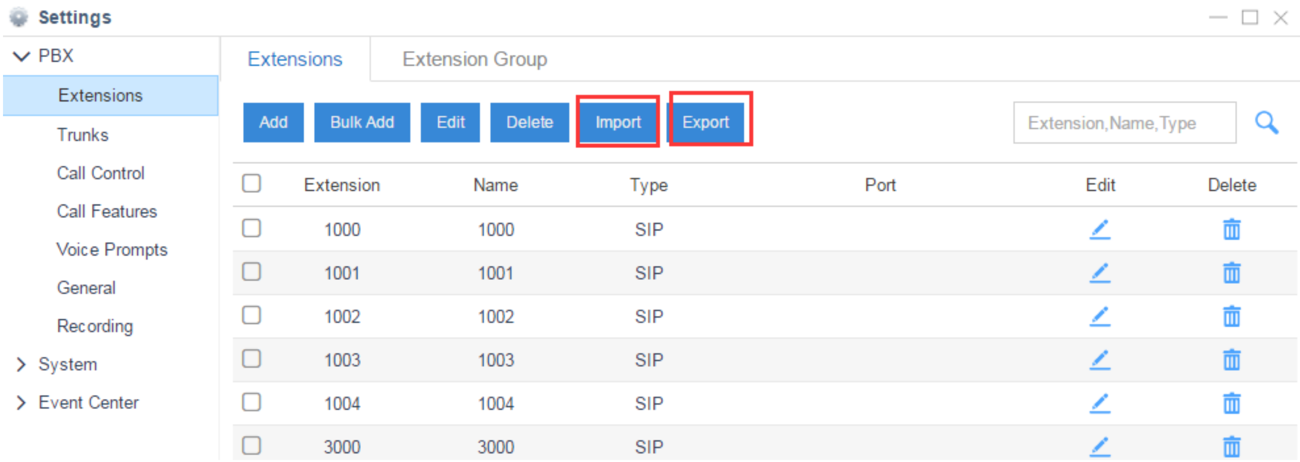
4. **Added Notifications on taskbar.**
   Instruction: on the upper-right corner, you can see the new added Notification icon and Recourse Monitor icon. Click to check news and alerts, click to check the PBX information, network

status and storage status.

5. **Added Web alert when the PBX power is off or the PBX is off-line.**
   Instruction: if the PBX power is off or any problem of the system network, a dialog will display on the webpage to inform you.

6. **Added SIP "Jitter Buffer" settings.**
   Path: Settings > PBX > General > SIP > Jitter Buffer
   Instruction: jitter is the variation in the time between packets arriving on a VoIP system. These variations can be caused by network congestion, timing drift or route changes. Jitter buffers are used to counter delay or latency, dropped packets, and jitter. They temporarily store arriving packets to minimize jitter and discard packets that arrive too late.
   Configure the Jitter Buffer settings on S-Series PBX will improve the call quality through VoIP.

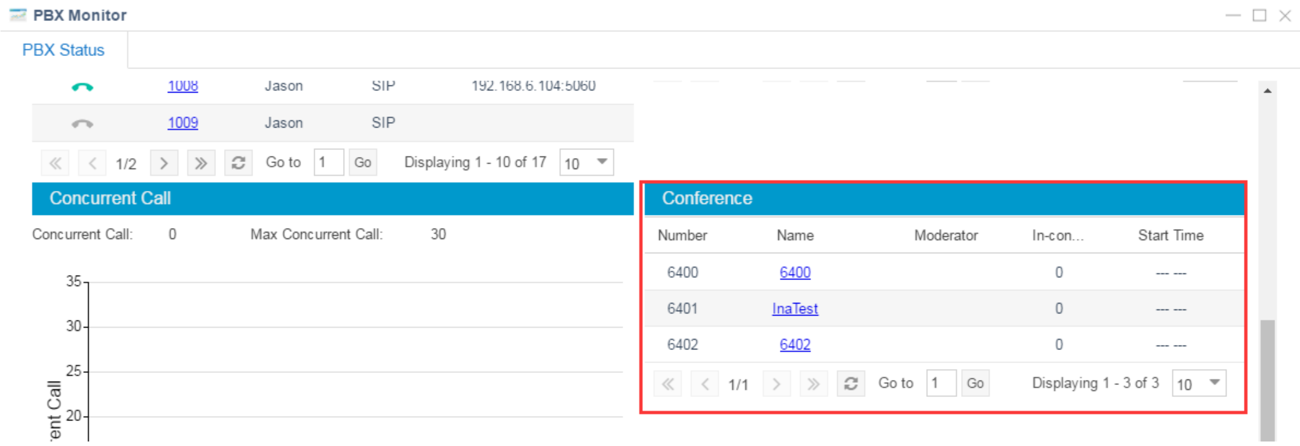
7. **Added support for importing and exporting extensions.**
   Path: Settings > PBX > Extensions
   Instruction: you can export an extension file from the PBX and use it as a sample to start with. The file format is csv.



8. **Added support for the following Tone Regions: Turkey, Korea, Serbia, Panama.**
   Path: Settings > PBX > General > Preferences
   Instruction: select the tone region according to the S-Series PBX location.

9. **Added support for Spanish Web User Interface language.**
   Instruction: users can switch the web language to Spanish.

10. **Added conference status on PBX Monitor panel.**
    Instruction: click PBX Monitor, here you can monitor the conference status.



11. Added "Echo Cancellation" setting on GSM/3G trunk and FXO trunk.

Path: Settings > PBX > Trunks > GSM Trunk / FXO Trunk

Instruction: check the option "Echo Cancellation" to enable echo cancellation for the trunk.

12. Added support for registering SIP trunk with random SIP port.

Path: Settings > PBX > General > SIP > General > Local SIP Port

Instruction: the random port in the port range will be used when sending packets to SIP server. The default range is 5062-5082.