Yeastar

Admin Guide

Yeastar S-Series VoIP PBX

Version: 30.7
Updated: April 12, 2018

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Contents

What's New?

Release History of the Yeastar S-Series VoIP PBX.

Version 30.7.0.27

Released Date: April 12, 2018

Optimization

PBX Stability
Improved the stability of the PBX.

ITSP Template
Added the following ITSP templates on VoIP Trunk.

- Germany — Telekom/Deutschland LAN IP Voice
- Germany — Telekom/Deutschland LAN SIP-Trunk

SSH Security
Improved the security of SSH access.

System Logs
Optimized the system logs.

API Interface
Supported for adding an authenticated IP address to get auto recording files from the PBX.

Help Documents
Deleted the Help App, and added a Help button on the task bar; you can click the button to view the online documents.

Fixed Bug

- Fixed the Password setting on SIP Trunk: the password would not take effect if the password length exceeded 63 characters.
- Fixed the video quality issue with Yealink video phones

Version 30.6.0.20

Released Date: January 18, 2018

New Feature

- Added support for Remote Management.

Version 30.6.0.16

Released Date: January 11, 2018

New Feature

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

Optimization

- Updated the following system prompt: English prompt, Korean prompt, and Hebrew prompt.
- Optimized the Web user interface: the PBX's GUI language will be synchronized to your browser language.
- Optimized call recording: the administrator can choose whether to record internal calls.
- Optimized blacklist feature: the administrator can restrict specific extension users from calling the blocked numbers.
- Optimized SLA feature: the administrator can enable or disable the SLA prompt.
- Optimized backup feature: the administrator can choose to backup system configurations, custom prompts or call logs.
- Added support for adding network drive for Windows domains.
- Optimized call forwarding: if an extension is unregistered, incoming calls to the extension will be forwarded to the "No answer" destination.
- Optimized CDR Auto Cleanup: when the CDR number reaches 90% of the Max Number of CDR, the system will send notification.
- Optimized Auto Recording Cleanup: the system will send notification when the usage of the device is about to reach the pre-configured maximum value.
- Optimized LDAP feature: when importing LDAP, the system can automatically add new phonebook nodes.
- Optimized LDAP feature: you can order the contacts by contact names.
- Improved the security of shared files on cHar: the administrator can add permitted IP addresses to access the shared files.
- Optimized billing feature: the administrator can configure whether to disconnect a user's call if the user's balance is insufficient.
- Optimized Distinctive Ringtone setting: you can enter uppercase letters, lowercase letters, numbers, "-", "_" and "+".
- 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.
- Optimized OpenVPN settings: you cannot upload a certification file with incorrect format.
- Optimized Call log searching feature.
- Optimized VLAN Priority setting: the default VLAN priority is 0.
- Optimized SLA feature: Users: users can dial numbers start with "*" through an SLA trunk.
- Optimized voicemail prompts.

Fixed Bug

- Fixed the issue that Mobility Extension user could not change the status of time condition.
- 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.
- Fixed the issue that if you switched the network type of a 4G module, the system would count the cellular data incorrectly.
- Fixed the issue of GSM/3G/4G module: a GSM/3G/4G module could not handle calls and SMS simultaneously.
- Fixed the issue of GSM/3G/4G module: after rebooting the PBX, the GSM/3G/4G modules could not work.
- Fixed "Email to SMS" issue: The system could not send long emails to multiple destination numbers.
- Fixed system email issue: the default TLS port for Sina emails could not work.
- 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.
- Fixed the issue that after receiving a packet of cancelling Register from Panasonic, the Yeastar system would break down.
- Fixed the prompt of Dial by Name.
- Fixed voicemail prompt: if you set the inbound route to voicemail, you could not hear the prompt of “Ask Caller to Dial 5”.
- Fixed All Busy Mode for SIP Forking: the feature only worked for the called extension, not worked for calling extension.
- Fixed ring group issue: if a member of the ring group canceled “Always Forward” by dialing *071; the member still could not receive calls.
- Fixed voicemail feature: if “Ask Caller to Dial 5” was enabled, users could not dial 0 to exit voicemail.
- 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.
- Fixed BRI trunk issue: importing DOD numbers could not work.
- 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.
- Fixed the compatibility issue with Loxone SIP Intercom system.

Version 30.5.0.30

Released Date: September 22, 2017

Upgrade Notes:

- This version has greatly enhanced the security mechanism; we strongly suggest all users upgrade to this new version.
- 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.
- If you are using Hot Standby, please disable it first, and upgrade the two PBXs separately before enabling Hot Standby again. Otherwise, the PBX's extensions will fail to register after the upgrade.

New Feature

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

Optimization

- Reduced background noise when playing system prompts and global recording in Linkus Mobile Clients.

- 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.
- 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.

Fixed Bug

- 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..
- Fixed the issue that DTMF on Polycom Phones did not take effect.
- 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.

Version 30.5.0.8

Released Date: July 12, 2017

New Feature

- 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.
- Added SIP Message by using MESSAGE method communicated between SIP phones.
- 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.
- 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.
- 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.
- 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.
- Added support for counting the duration from "Queue receives call" to "Agent answers call".
- Added 2 VLAN subinterfaces for LAN and WAN.
- Added DTMF pass-through for FXS when connected with door phone.
- Added warning to create backup when trying to upgrade.
- 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.
- Added support for Auto Recording file, One-Touch Recording file, and CDR backup in real time to stand-by device in Hot Standby configuration.
- Added silence detection for FXO, BRI and E1.
- Added new Italian and German Web language.
- Added support for saving Web configuration by pressing Enter.
- Added Local Loop trunk for IP address 127.0.0.1.
- Added BLF monitoring support for Time Condition default feature code like *800.

Optimization

- Optimized SMTP authentication. Account authentication is added.
- Optimized China PRI default configuration to match most of China PRI lines.
- Optimized Web text on Adapt Caller ID page. “Dial Patterns” is replaced by “Adaptation Patterns”.
- Optimized Adapt Caller ID feature. Now it is also available for Ring Group and Queue.
- Optimized Web text in Event Center. “Storage Space Full” is replaced by “Storage Full”.
- 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.
- 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”.
- 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.
- Removed check box of Feature Code header and parking number range.
- Added Italian SLA dial prompt.
- Added prompt for Email to SMS: when country code is not working, try the customized country code.

Optimized PAI and Remote Party ID would be disabled by default.

✔ Fixed Bug

- Fixed Dial by Name issue: the recorded name can be played in the Dial by Name event.
- 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.
- Fixed the “More” button display issue on Notification menu: “More” button would still be clickable when there is no notification.
- Fixed the issue that voicemail couldn’t be found when Caller ID name including “&” on the user Me page.
- Fixed network disk issue that it couldn’t be reconnected when it was disconnected due to unstable network.
- 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.
- Fixed the issue that Ring Group members’ status would still show ringing when the call had been answered by Pick Up.
- Fixed incorrect CPU display when PBX has been up for many days.
- Fixed the issue that lacking External Host Refresh Internal in SIP NAT settings would cause host parsing failure.
- 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.
- 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.
- 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.
- Fixed the issue that content of users.conf would be erased after executing command “astconfig”.
- Fixed the issue that paging initiator would also be paged when he was in the paging group that he paged.
- Fixed the issue that DND status of Hotel APP page would not change when DND was enabled by a hotel extension.

Version 30.4.0.25

Released Date: May 23, 2017

Upgrade Notes

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.

New Feature

- Added schedule reboot feature (Maintenance > Reboot).

Optimization

- 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).
- Optimized network setting: LAN port supports up to 2 IP addresses.
- Optimized firewall feature: users can add firewall rules using domain names.
- Optimized IP Auto Defense feature: users can add a rule for a port range.
- Optimized Local SIP Port: the system has a default IP auto defense rule for the local SIP ports.
- Optimized extension registration security: only the registered extension can dial out, direct IP call is forbidden.
- 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.
- Optimized extension/trunk registration security: the system will not reply OPTIONS packets from unknown extensions or trunks.
- Optimized Forgot Password feature: users need to provide correct extension number and extension email address to retrieve password.
- Optimized extension User Password setting: the User Password can only be changed to a mixture of uppercase, lowercase and numbers.
- Optimized Logs feature: users can download operation logs and event logs.
- Optimized Logs Auto Cleanup feature: users can set the max size of total logs stored in the system.
- Optimized CDR feature: users can choose to show “Caller IP Address” in the CDR logs.
- Optimized System Logs: extension registration failed information will be recorded in system logs.
- Optimized QueueMetrics Live Integration App: the recording files name will contain QueueMetrics UniqueID.

Fixed Bug

- Fixed Operation Logs issue: if users edited extensions in bulk, the operation logs would not contain what settings the users had edited.
- Fixed Queue: the call duration time was incorrect.
- 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.
- 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.

Version 30.4.0.6

Released Date: March 30, 2017

New Feature

- Added Hotel App.
- Added char utile h+ Integration App.
- Added QueueMetrics Live Integration App.

Optimization

- 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.
- Updated French and Italian system Web GUI.
- Updated English, Chinese, Turkish, Spanish, and Hebrew system prompts.
- Optimized CDR and Recordings feature: the system will record the un-answered call if the queue agents didn't answer the call.
- Optimized CDR and Recordings feature: the PBX administrator can delete and download call logs and recording files in batch.
- Optimized CDR and Recordings feature: if an extension has a DOD number, the call log will display the extension number and the DOD number.
- Optimized CDR and Recordings feature: users can click a column heading to sort the table in ascending or descending order.
- Optimized CDR and Recordings feature: users can choose an extension to play recording file on the web interface.
- Optimized Voicemail feature: if voicemails have been played or deleted on Web interface, the IP phone will synchronize the voicemail status.
- Optimized Voicemail feature: users can change the voicemail system's "Busy Prompt" and "Unavailable Prompt" (Settings > PBX > General > Voicemail > Greeting Options).
- 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.
- Optimized User Permission: allow users to delete auto recording files; separate reboot permission and reset permission.
- Optimized DOD Settings: users can import and export DOD numbers.
- 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.
- Optimized SIP DID feature: added support for getting DID number from "Diversion" and "P-Asserted-Identify" (PBX > General > SIP > Advanced).
- Optimized SIP Outbound SIP Registrations Setting: the "Default Incoming/Outgoing Registration Time" is set to 1800 seconds.
- 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.
- Optimized SLA feature: users can choose a failover destination for a SLA trunk.
- Optimized Event Center: added "Network Drive Lost Connection" notification setting.
- Optimized Static Route: users can choose adding static route to LAN gateway or WAN gateway.
- Optimized System Email Password Setting: the password setting are allowed to enter any special characters.
- 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.
- Optimized Auto Cleanup feature: when the CDR reaches over 90% of the "Max number of CDR", the system will send notification.
- Optimized SIP feature: support requests of MESSAGE SIP packet (PBX > General > SIP > Advanced).

- Optimized DID Settings: support DID numbers that start with digit 0 and support DID numbers that contain character “+”.
- Optimized FXO Settings: users can change the FXO trunk’s “DTMF Duration” and “DTMF Gap” (PBX > General).
- Optimized FXO Settings: users can manually release a FXO trunk on web interface (FXO Trunk > Advanced > Other Settings).
- 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.

✔ Fixed Bug

- Fixed the issue that the PBX would continually reboot if the device power was disconnected during firmware upgrade.
- Fixed BLF LED status issue: the BLF LED would stay in Green status after rebooting S-Series IPPBX.PBX.
- Fixed Voicemail issue: the voicemails were not deleted automatically even the option “Delete Voicemail” was enabled.
- Fixed Emergency Call issue: when the PBX reached the max number of concurrent calls, emergency calls could not work.
- Fixed Time Condition issue: if time condition status was force changed by feature code, distinctive ring tones on inbound routes could not work.
- Fixed Time Condition issue: the IP phone BLF key could not correctly indicate the time condition status.
- 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.
- Fixed Email to SMS issue: if the email contains Polish words, sending email to SMS would fail.
- Fixed compatibility issue with some VOIP ISP: making outbound calls with the VoIP trunk, the calls would be disconnected after 30 seconds.
- 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.
- Fixed Hot Standby issue: after enabling Hot Standby feature, the system’s default gateway would automatically be changed to LAN gateway.
- Fixed Hot Standby issue: configurations on the primary server and standby server could not be synchronized.
- 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.
- 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.
- Fixed Voicemail issue: system would reserve the prompt “please leave a message after the tone”, even users had custom “Busy prompt” and “Unavailable prompt”.
- 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.
- 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.
- 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.
- 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.
- Fixed the Outbound route issue: the system could not correctly match outbound route if multiple outbound routes were configured with the same dial pattern.
- Fixed the Backup and Restore issue: restoring a S300 backup to S20 IPPBX, all the S300 extensions would be restored on S20.
- 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.

- Fixed Automatic Upgrade issue: check for update could not work properly.
- Fixed CDR and Recordings issue: pickup calls were not recorded in the call log.
- Fixed CDR and Recordings issue: if the research result includes tens of thousands of records, downloading the research results would fail.
- 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.
- 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.
- Fixed the issue that extension IP Restriction feature could not work properly.
- Fixed the IAX issues: IAX trunk's domain name could not automatically update; using IAX might cause memory leaks.

Version 30.3.0.17

Released Date: January 13, 2017

New Feature

- Added support for French, Italian Web interface.
- Updated Turkish Web interface.

Fixed Bug

- 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.
- Fixed Backup and Restore issue: restored a S300 backup file to a S300 device, the network settings could not be restored.
- 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.
- Fixed importing and exporting extensions issue: exported extensions and deleted extensions on the PBX, then imported the extensions, only one extension would be imported.
- Fixed Inbound Route issue: chose the Fax Destination to an extension, then deleted the extension, the Fax Destination would display an “h”.
- Fixed Time Condition issue: any extension user could reset the time condition.
- Fixed Time Condition issue: if Time Condition Override was disabled, only the first inbound route could work.
- 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.
- 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.
- Fixed Extension Status issue: an extension number was registered on multiple IP phones, and the extension status would display only one IP address.
- Fixed Whitelist/Blacklist issue: a whitelist/blacklist could not be imported to PBX if the numbers contain letters.

Version 30.3.0.10

Released Date: January 4, 2017

New Feature

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

Optimization

- Updated Simplified Chinese, English and Spanish web interface.
- Updated English, Hebrew and Chinese system prompts.
- Upgrade the NTP server version to 4.2.8p8.
- Supports to add a maximum of 5 prompts for an IVR.
- 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.)
- When a user picks up a call, he can see the incoming caller ID on his phone.
- The system will give a prompt when a mobile extension user calls in the S-Series IPPBX.
- Limit the length of outbound route “Dial Pattern” to 63 characters.
- The debug commands under SSH will be recorded in system logs.

Fixed Bug

- 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.
- Fixed the SIP peer trunk issue: the trunk would not work if “Qualify” is disabled.
- Fixed E1 trunk issue: if the E1 trunk signaling is set to mfc/r2, this trunk would be unavailable to use.
- 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.
- 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.

Version 30.2.0.27

Released Date: November 21, 2016

New Feature

- Added Linkus Application.

Optimization

- 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.

Fixed Bug

- 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.

Version 30.2.0.8

Released Date: November 2, 2016

New Feature

- Added VPN Server Application.
- Added Hot Standby feature.
- Added support for deleting used records, keeping missed call records only, and phone numbers digits match for AutoCLIP route.
- Added support for enabling/disabling VoIP trunks.
- Added support for monitoring FXS and GSM/3G channels.
- Added support for changing auto logout time.
- Added IVR invalid key prompt setting.
- Added support for enabling/disabling “Local SIP Port”.
- Added one touch recording prompt settings.
- 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.
- Added support for checking and downloading system prompt files via FTP. The access address is *ftp://PBX's IP address/ysapps/pbxcenter/var/lib/asterisk/sounds/*.
- Added support for changing SSH username and password.
- Added support for Deutsch web interface language.
- Added support for Czech system prompt.

Optimization

- Optimized backup function: the backup file will contain custom prompts and music on hold files.
- Extended the max supported log files captured on Ethernet Capture Tool.
- Time search on CDR and Recording page is accurate to the seconds.
- Increased the maximum number of conference members.
    - S20: 10

    - S50: 25
    - S100: 60
    - S300: 120
- Increased the maximum number of pickup groups.
    - S20: 10
    - S50: 25
    - S100: 64
    - S300: 64
- Limited the SIP UDP port setting: the port value could not fall in the range of local SIP ports.
- Character limit for Trunk name: characters * ? are not allowed.
- Character limit for Inbound route DID number: character + is allowed.
- Character limit for Time Condition Name: characters ! $ ( ) \ / # ; , [ ] \ " = < > & ' ` ^ % @ { } | \ and blank character are not allowed.
- Character limit for system email password: character $ is allowed.
- The system will notify users on web if D30 module is removed.
- When you upload Tiptel phonebook to S-Series PBX, you can choose phone type and upload different phonebooks.
- Notification template could work for both SMS notification and email notification.
- You can sort speed dial numbers.
- Optimized Call Features and Conference Panel Web User Interface.
- Updated Chinese system prompts.
- Updated traditional Chinese web user interface.
- There is no limit for SIP trunk registration attempts by default.

✔ Fixed Bug

- 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.
- Fixed CDR issue: if you change the E1 trunk name, the CDR would display an incorrect E1 trunk name.
- 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.
- Fixed the SD card issue: the system would always warn you that the SD card format was wrong.
- 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.
- Fixed VoIP trunk issue: if you enable proxy server, and disable it again, then save the changes, the VoIP trunk would not work properly.
- 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.
- Fixed custom prompt issue: two prompt files would be created if you made a custom prompt using an extension.
- Fixed conference issue: if you invited users to one conference, the system concurrent calls would be twice of the invited users.
- Fixed SMS to Email issue: emails received in Foxmail would display incorrect received time.
- 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.
- 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.
- Fixed network drive issue: if the shared folder name is a Chinese name, the network drive could not be mounted to PBX.

- Fixed Email to SMS issue: received contents in your mobile phone might be incomplete if you write too many contents in the email.
- Fixed the temporary voicemail message issue.
- Fixed holiday time condition issue: the system computed holiday time incorrectly.
- 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.
- Fixed compatibility issue with IE11 browser: users could not listen and download recordings on IE11.
- 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.
- 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.
- Fixed queue issue: if the ring strategy was set to “Ring All”, some queue members might not ring at the same time.
- 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.
- Fixed VLAN issue: if network mode was DHCP, VLAN could not work.
- Fixed inbound route DID issue: if DID number starts with digit 0, the system would automatically delete digit 0.
- Fixe whitelist/blacklist issue: if you entered a number range, the web interface would display incorrectly.
- Fixed compatibility issue with IE11 browser: checking system information log would cause the page no response.
- Fixed AutoCLIP route issue: if you set “Record Keep Time” to 0, the system CPU would reach 100%.
- Fixed the issue that feature code *8 was not available.
- Fixed Echo Cancellation issue: if you changed module on the PBX, echo cancellation could not work properly.
- Fixed PBX center app issue: updating PBX center would cause all the extension being unavailable.
- 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.
- 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.
- 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.
- Fixed the CDR issue: if an extension user pressed *1 to make one touch recording, he could not check the call log on web.
- Fixed the issue that SIP Registration/Subscription settings could not take effect.
- Fixed the issue that if virtual ring back tone was enabled on GSM/3G trunk, the trunk could not receive calls.
- Fixed that Peer SIP trunk issue.
- Fixed the issue that using wget command could not resolve domain under SSH.
- Fixed the issue that custom configuration file extensions.conf could not work properly.
- Fixed the issue that if an incoming call is forwarded to an external number, this call would not be recorded.
- Fixed VoIP trunk issue: if “Hostname/IP” and “Domain” were set differently, the trunk could not work properly.
- Fixed the issue that adding user permission would fail if there were 300 extension users on the PBX.
- Fixed G729/G723/iLBC codec issue: multiple concurrent calls with the codecs would cause the system freezing up.
- Fixed event center issue: if sending alerts failed, it might cause memory leaks.

Version 30.1.0.16

Released Date: September 19, 2016

Optimization

- On the "Maintenance > Upgrade" page, the length of "HTTP URL" textbox is increased.

Fixed Bug

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

Version 30.1.0.13

Released Date: September 7, 2016

New Feature

- 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

Fixed Bug

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

Version 30.1.0.10

Released Date: August 29, 2016

New Feature

- Added support for Polish Web User Interface language.

Optimization

- Updated Spanish Web User Interface language.
- Updated Russian Web User Interface language.

Fixed Bug

- 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.
- Fixed the Automatic Upgrade issue: the system would give a wrong prompt saying the checked failed because of network issues.

- 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.

Version 30.1.0.7

Released Date: August 23, 2016

New Feature

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

Optimization

- 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.
- Administrator account (admin) Password has an 8-63 character limit; the password must contain uppercase letters, lowercase letters and numbers.
- Extension account Password has a 6-63 character limit; the following characters are not allowed:

  & ; " ' \ < > | ,

- Your S-Series PBX’s system prompts would be updated to the newest version automatically after firmware upgrade.
- If you change the system service port (like FTP, HTTP, SSH etc.), the system will remind you to reconfigure the firewall settings.
- AutoCLIP routing could match incoming numbers with area codes and special character “+”.
- Added “SMS to Email” and “Email to SMS” failure records in Event Log.
- For Logs Auto Cleanup, the “Logs Preservation Duration” setting is for system logs, the “Max Number of Logs” setting is for operation logs.
- The callback list will display each callback’s detailed destination.
- 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
- 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.
- Added CPU temperature information in system logs.

✔ Fixed Bug

- 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.
- 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.
- 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.
- Fixed the compatibility issue with VoIP provider netelip.
- Fixed the issue that sending email to SMS would fail if the content exceeded the length limit.
- Fixed the issue that the system would automatically make outbound calls through FXO trunks.
- Fixed the Event Center issue: WAN port failure was not recorded in the event logs.
- Fixed the dual mode network issue: if one Ethernet port used VLAN, the other Ethernet port could not work properly.
- Fixed the IE11 compatibility issue: users could not play recording files on PBX web GUI.
- Fixed the Holiday issue: the holiday worked based on Time Zone GMT +0.
- Fixed the E1 trunk issue: the call quality was bad when using the E1 trunk.
- Fixed the Event Center issue: the event center repeatedly recorded SIP trunk registration failure.
- 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.

Introduction

Designed with the small and medium sized enterprises in mind, supporting up to 500 users and built using the very latest technology, the Yeastar S-Series delivers exceptional cost savings, productivity and efficiency improvements, delivering power, performance, quality and peace of mind.

The all new S-Series is engineered for the communications needs of today and tomorrow, and with the Yeastar unique modular design future proofs your investment choice.

Appreciate the Easy-to-use Solution

- Intuitive and graphical UI brings point-and-click configuration.
- Convenient Phone Provisioning feature saves you tremendous time.
- Everything can be managed from anywhere with Internet access.

Your Choice of Technologies and Features

- Embedded VoIP capability and analog phone connections.
- Rich external lines options include SIP, PSTN, ISDN BRI, E1/T1/PRI, and cellular networks.
- Concurrent calls and maximum users are expandable with modules.
- App Center integrates features that you can add when you need them.

Telephone System without Risk

- Meanwell power supply featuring MTBF>560Kh.
- High-quality Freescale CPU processor and industry leading TI DSP voice processor.
- Connectors from TE Connectivity with a gold plating layer as thick as 15 µ.
- Lightening protection on analog ports complying with ITU-T K.20/45/21 8/20 µs and GR-1089 standard.

Play Safe and Expect Reliability

- TLS, SRTP, and HTTPS standards for better security.
- Defend against malicious attack with built-in Firewall.
- Monitor system status and behavior and be notified when abnormalities occur.

Learn more about Yeastar S-Series IPPBX

Getting Started

Get started to configure the basics of the Yeastar S-Series VoIP PBX.

Initial System Setup

For the first login, you need to set the PBX according to your local language and location time.

Set Date and Time

Adjust the time of your PBX (including the time zone) to your local time. The PBX will generate call logs with your local time.

1. Go to **Settings** > **System** > **Date & Time**.
2. Select your time zone.
3. (Optional) Configure **Daylight Saving Time**.
4. Click **Save**.
5. On the pop-up window in the bottom right corner, click **Yes** to reboot the system.

Change the System Prompt

The default system prompt on the PBX is US English prompt. The PBX supports multiple language prompts. You can update the system prompt from the cloud server directly or upload system prompt from local PC.

Update system prompt from the cloud server:

1. Go to **Settings** > **PBX** > **Voice Prompt** > **System Prompt**.
2. Click **Download Online Prompt**.
3. In the **Download Online Prompt** window, choose your desired system prompt, click .
   The new downloaded system prompt will appear in the **Prompt List**.
4. In the **Prompt List**, select one system prompt as the default prompt.
5. Click **Save** and **Apply**.

Set up Extensions

To make calls and receive calls, you need to create extensions on your PBX, and register the extensions on your Linkus mobile client or IP phones.

Related information

Work with Linkus on page 28

Change Extension Range

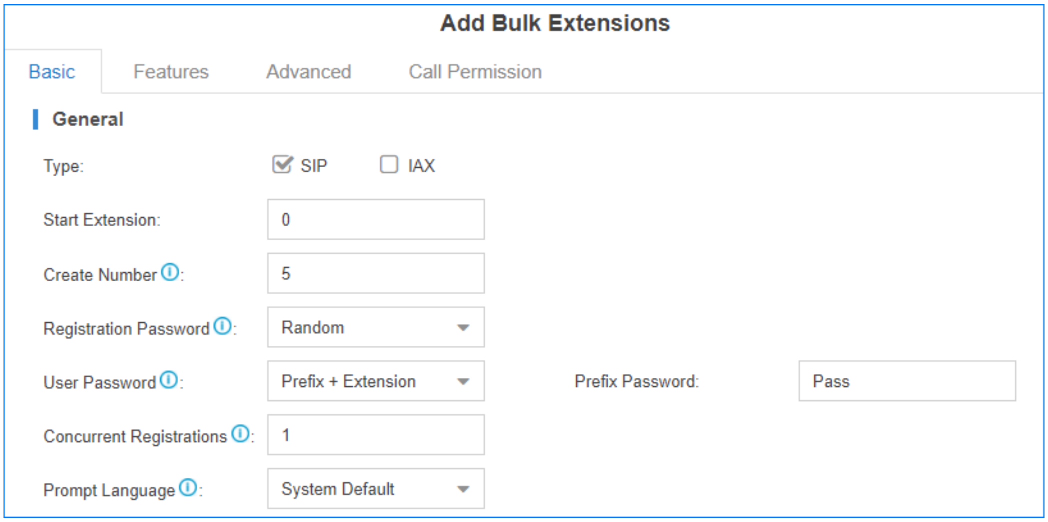
The default extension range is from 1000 to 5999. Before you start to create extensions, you can change the extension range according to your needs.

1. Go to **Settings** > **PBX** > **General** > **Preferences** > **Extension Preferences**.
2. Change the range of **User Extensions**.
3. Click **Save** and **Apply**.

Add Bulk Extensions

You can add bulk SIP/IAX extensions.

1. Go to **Settings** > **PBX** > **Extensions**.
2. Click **Bulk Add**.
3. In the **Basic** page, configure the following settings:



- **Type**: Set the extension type.
- **Start Extension**: Enter the first extension number. The system will create extensions in bulk starting with the extension number.
- **Create Number**: Enter how many extensions to create.
- **Registration Password**: Set the registration password.
- **User Password**: The users can log in the PBX or log in Linkus mobile client by the user password.
- **Concurrent Registrations**: The PBX supports registering one extension on multiple phones. When a call comes to the extension, all the phones will ring. The maximum number of concurrent registrations is 5.
- **Prompt Language**: You can set specific prompt language for the extensions.

4. Optional: Click **Features**, **Advanced**, or **Call Permission** tab to configure other settings.
5. Click **Save** and **Apply**.

Set up Your Phone

To use VoIP extensions, you need to register the extensions on your IP phones or soft phones.

Yeastar provides Linkus mobile client, Linkus helps you set up your phone efficiently and links you and your colleagues and customers anywhere anytime.

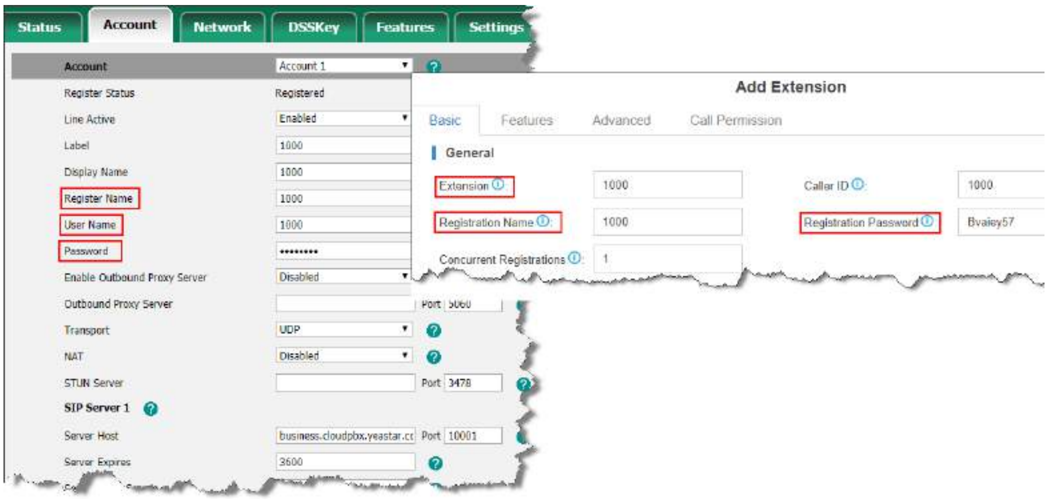
We have tested a number of IP phones and soft phones with Yeastar S-Series VoIP PBX. Below is an example of registering an extension on Yealink phones.

Register an Extension on Yealink Phone

1. Log in the Yealink phone web interface, click **Account** tab.
2. From the **Account** drop-down list, select an available account.
3. Set **Line Active** to **Enabled**.
4. Enter the extension information.

- **Label**: The name you want to display on the phone screen.
- **Display Name**: The name you want to display on another person's phone screen when you are calling the phone.

- **Register Name**: Enter the extension's Registration Name.
- **User Name**: Enter the extension number.
- **Password**: Enter the extension's Registration Password.
- **SIP Server Host**: Enter the PBX's domain or IP address.
- **Port**: Enter the PBX's SIP port.



5. Click **Confirm**.

If the extension is registered, you can see the Register Status shows "Registered" on the **Account** page.

Related information

Work with Linkus on page 28

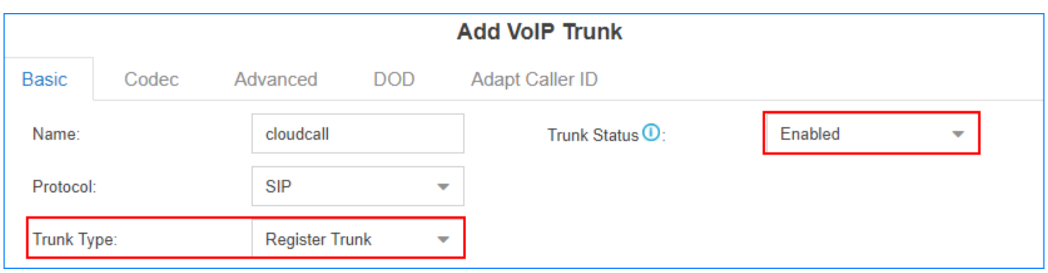
Set up VoIP Trunks

To make and receive calls through a VoIP trunk, you need to buy a VoIP account from the VoIP provider and register the account on the PBX.

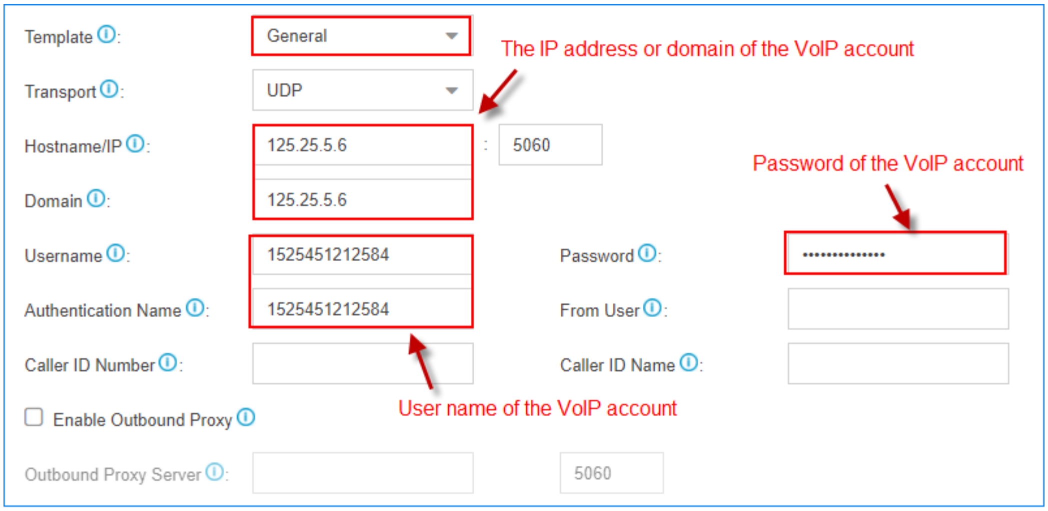
Set up a Register VoIP Trunk

If you bought a VoIP account with user name and password, you need to set up a Register Trunk on the PBX.

1. Log in the PBX, go to **Settings** > **PBX** > **Trunks**, click **Add**.
2. In the **Trunk Name** field, specify a name for the trunk.
3. Set the **Trunk Status** as **Enabled**.
4. Set **Trunk Type** as **Register Trunk**.



5. Set the **Template** as **General**.
6. Enter the trunk information.



- **Transport**: Select the transport according to your VoIP account.
- **Hostname/IP**: Enter the IP address or the domain of the VoIP account.
- **Domain**: Enter the IP address or the domain of the VoIP account.
- **Username**: Enter the user name of the VoIP account.
- **Password**: Enter the password of the VoIP account.
- **Authentication Name**: Enter the user name of the VoIP account.
- **From User**: Keep this filed blank if not needed.
- **Caller ID Number**: This feature requires support from the ITSP. Keep this field blank if not needed.
- **Caller ID Name**: This feature requires support from the ITSP. Keep this field blank if not needed.
- **Enable Outbound Proxy**: Set the outbound proxy if the ITSP needs.

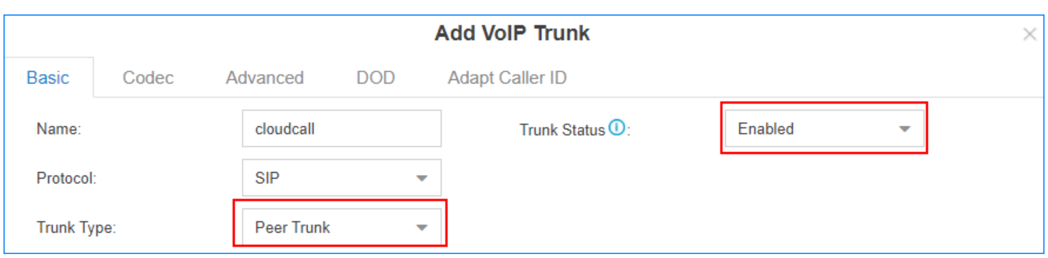
7. In the **Password** field, enter the password of your VoIP account.
8. Optional: Click other tabs on the configuration page, configure the relevant settings according to your VoIP account.
9. Click **Save** and **Apply**.

You can check the trunk status in **PBX Monitor**. If the trunk status shows "Registered", the trunk is ready for use.

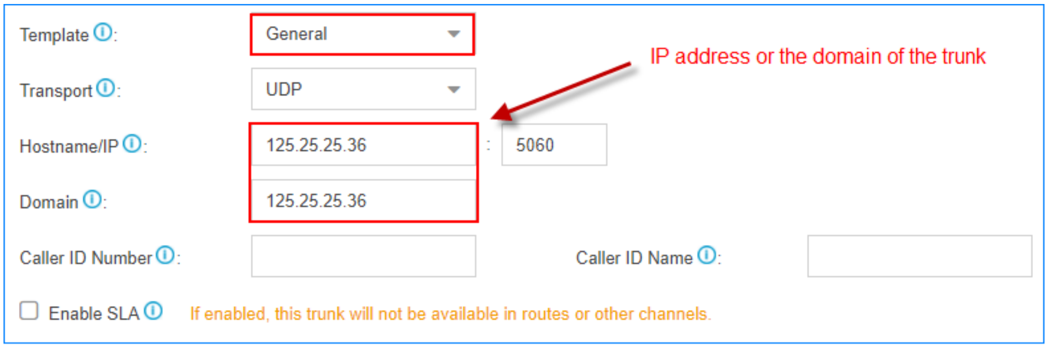
Set up a Peer VoIP Trunk

If the ITSP only provides an IP address or domain for your purchased VoIP account, you need to set up a Peer Trunk on the PBX.

1. Log in the PBX, go to **Settings** > **PBX** > **Trunks**, click **Add**.
2. In the **Trunk Name** field, specify a name for the trunk.
3. Set the **Trunk Status** as **Enabled**.
4. Select **Trunk Type** as **Peer Trunk**.



5. Set the **Template** as **General**.
6. Enter the trunk information.



- **Transport**: Select the transport according to your trunk.
- **Hostname/IP**: Enter the IP address or the domain of the VoIP account.
- **Domain**: Enter the IP address or the domain of the VoIP account.
- **Caller ID Number**: This feature requires support from the ITSP. Keep this field if not needed.
- **Caller ID Name**: This feature requires support from the ITSP. Keep this field if not needed.

7. Click other tabs on the configuration page; configure the relevant settings according to your VoIP account.
8. Click **Save** and **Apply**.

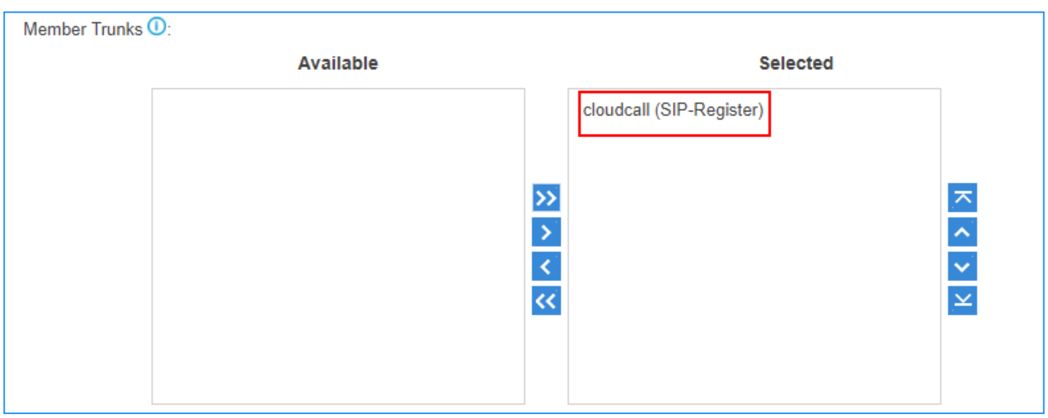
You can check the trunk status in **PBX Monitor**. If the trunk status shows "OK", the trunk is ready for use.

Receive Inbound Calls

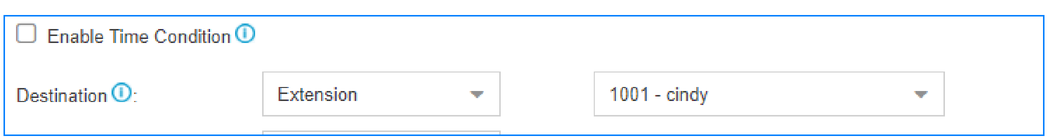
When a call comes into the PBX from outside, PBX needs to know where to direct it. The incoming call can be directed to an extension, a ring group, a queue, or a digital receptionist (IVR) etc. For this purpose, you need to set up inbound routes on the PBX.

Set up an inbound route:

1. Go to **Settings** > **PBX** > **Call Control** > **Inbound Routes**.
2. Click to edit the default inbound route.
3. In the **Member Trunks** field, double click the desired trunk in the **Available** box, the trunk will appear on **Selected** box.



4. In the **Destination** field, select a destination for the inbound route (e.g. extension 1001).



5. Click **Save** and **Apply**.

When a person calls in the selected trunk "cloudcall", the call will be directed to extension 1001.

Make Outbound Calls

Outbound routes are used to specify what numbers are allowed to go out a particular route. When a call is placed, the actual number dialed by the user is compared with the dial patterns in each route (from highest to lowest priority) until a match is found. If no match is found, the call fails. If the number dialed matches a pattern in more than one route, only the rules with the highest priority in the route are used.

The default outbound route (with dial pattern `X.`) on the PBX will allow any outgoing numbers. You can make outbound calls as you usually do on your phone.

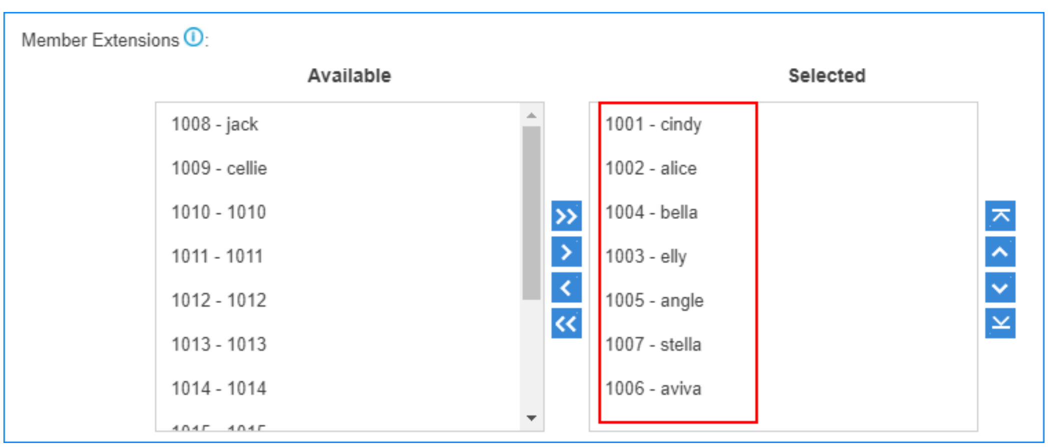
Set up an outbound route:

1. Go to **Settings** > **PBX** > **Call Control** > **Outbound Routes**.
2. Click to edit the default inbound route.
3. In the **Member Trunks** field, double click the desired trunk in the **Available** box, the trunk will appear in **Selected** box.

   The users can only use the selected trunk to make outbound calls through this route.

4. In the **Member Extensions** field, double click the desired extensions in the **Available** box, the extensions will appear on **Selected** box.

   Only the selected extension users can make outbound calls through this route.



5. Click **Save** and **Apply**.

Now, the selected extension users can make outbound calls through this route.

Work with Linkus

Linkus is a VoIP mobile client coordinated with Yeastar S-Series VoIP PBX and Yeastar Cloud PBX. Linkus makes your mobile phone an office extension and links you and your colleagues and customers anywhere anytime.

Download Linkus client (iOS version)

Search "Yeastar" in Apple Store to find Linkus and download the application.

Download Linkus client (Android version)

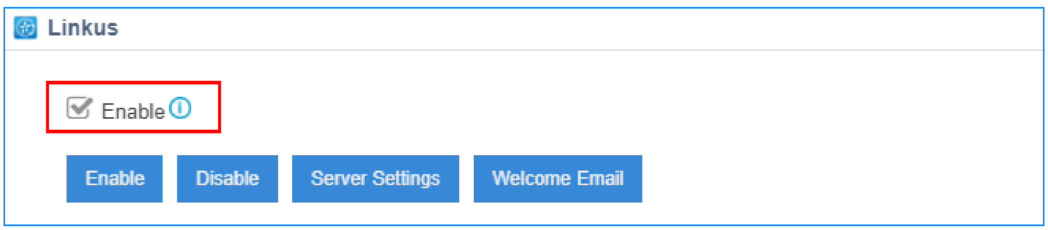
Search "Yeastar" in Google Play to find Linkus and download the application.

Set up Linkus Server

Before users can use Linkus, you need to enable Linkus server and assign the login information of Linkus to the users.

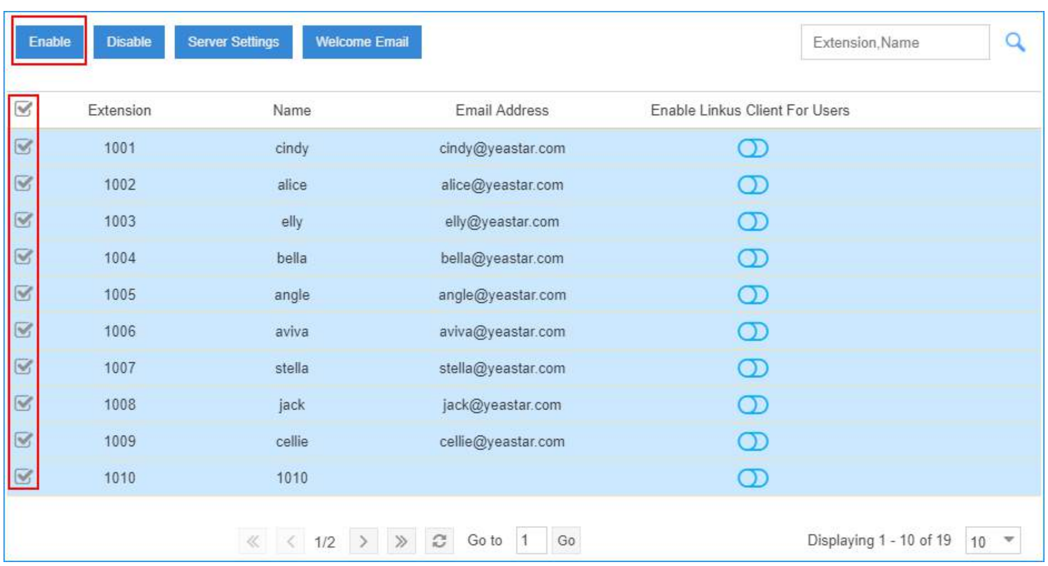
Enable Linkus Server

1. Log in the web interface of Yeastar PBX, go to **Main Menu** > **Linkus**.
2. On the **Linkus** page, select **Enable** option to enable Linkus server application on the PBX.



3. In the pop-up window, click **Yes** to confirm.
4. On the **Linkus** page, select the desired extension users, click **Enable** to enable Linkus service for the users.

Note: If you want to enable Linkus service for all the users, you need to select all the extensions on each page and enable Linkus for the extensions.



5. In the pop-up window, click **Yes** to confirm.
6. Click **Apply**.

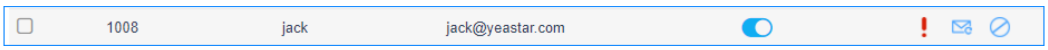
Assign Linkus Login Information

After enabling Linkus clients for the extension users, you can send e-mails to the extension users. The e-mail contains the login information of the Linkus client to the users.

Note: Before assigning login information of Linkus client to the extension users, the extension users must bind to their e-mail addresses.

1. Log in the web interface of Yeastar PBX, go to **Main Menu** > **Linkus**.
2. On the **Linkus** page, click **Welcome Email**.
3. Select extensions, click **Send**.
    - **All Extensions**: send e-mail to all the extension users.
    - **Selected Extensions**: send e-mail to the selected extension users.

    The system starts to send e-mails to the extension users. If the e-mail is failed to be sent, you can see an error sign on the Linkus page.



Log in Linkus Client

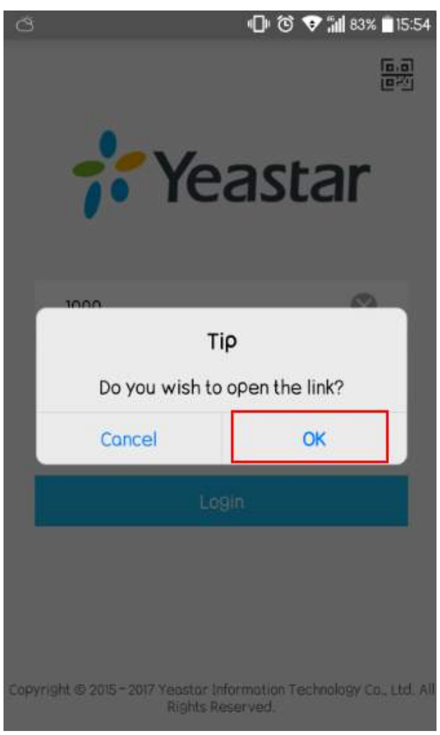
After you send emails to the extension users, the users can quickly log in the Linkus client by QR code or login link.

Note:

- The QR code or login link will expire after 24 hours.
- The QR code and link will expire after the user log in the Linkus client.

Log in Linkus via Login Link

1. Open the e-mail with the subject "Login Linkus Mobile Client" on your mobile phone.
2. Select and copy the Linkus login link which is in the e-mail.
3. Open Linkus mobile client, you will be prompted to log in Linkus with the copied link.
4. Click **OK** to get the login information and log in Linkus.



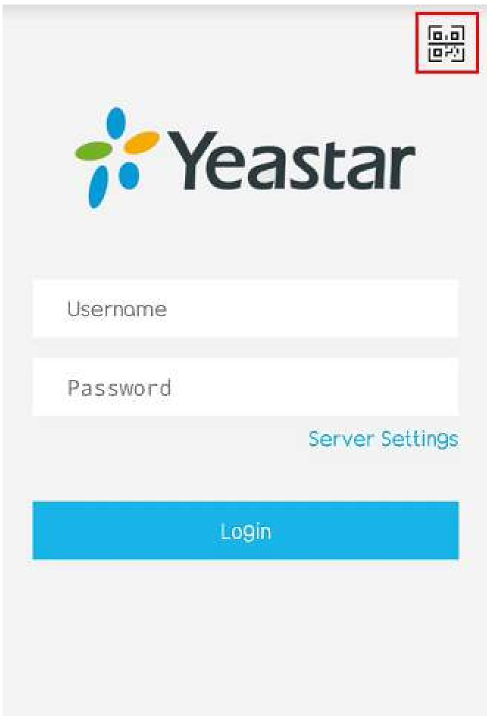
Log in Linkus via QR Code

Note:

For security reason, your email server may intercept the QR code. If you don't receive the QR code, try to log in Linkus by other method.

1. Open the e-mail with the subject "Login Linkus Mobile Client" on your PC.
2. Open Linkus mobile client, tap the QR button, and scan the QR code in the e-mail.

   The Linkus client will get the login information and log in automatically.

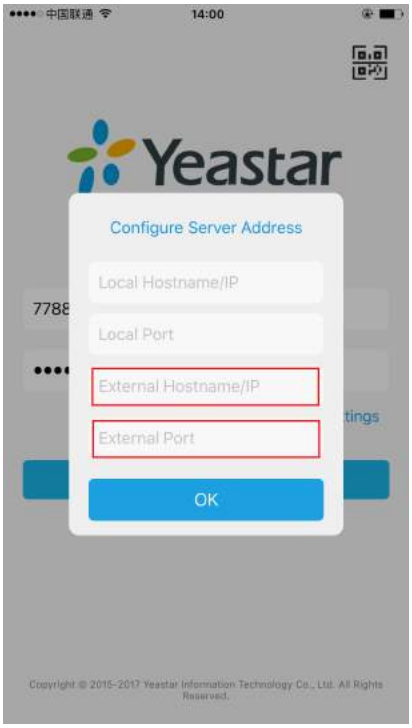


Log in Linkus Manually

The Linkus users can log in Linkus manually. You need to send the Linkus server information and account information to the users. After the users get the information, they can follow the steps below to log in Linkus.

1. Launch Linkus mobile client.
2. Tap **Server Settings**.
3. On the **Configure Server Address** page, enter the external server address and port.

   - **External Hostname/IP**: Enter the external hostname/IP of the PBX.
   - **External Port**: Enter the external port of Linkus server.



4. Tap **OK**.
5. On the Linkus login page, enter the username and password.
    - **Username**: Enter the email address of the extension user.
    - **Password**: Enter the **User Password** of the extension.
6. Tap **Login**.

Extensions & Phones

Manage Extensions

Change Extension Range

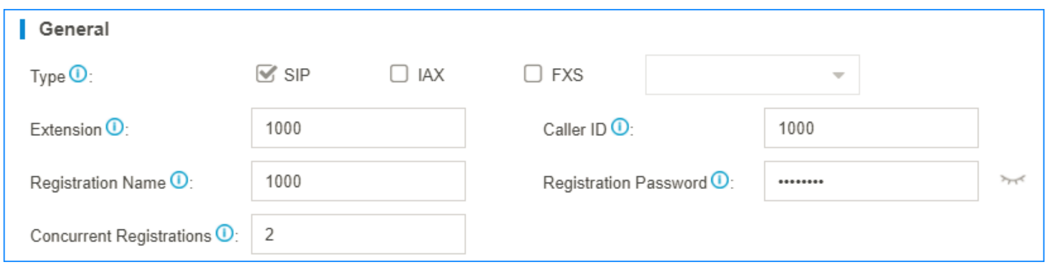
The default extension range is from 1000 to 5999. Before you start to create extensions, you can change the extension range according to your needs.

1. Go to **Settings** > **PBX** > **General** > **Preferences** > **Extension Preferences**.
2. Change the range of **User Extensions**.
3. Click **Save** and **Apply**.

Add a VoIP Extension

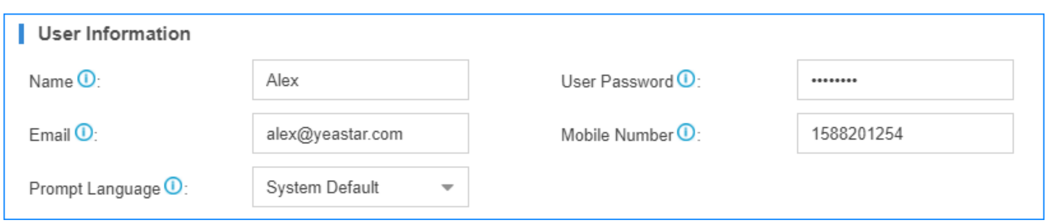
To add a VoIP extension, you don't need insert physical modules to the PBX.

1. Go to **Settings** > **PBX** > **Extensions**, click **Add**.
2. On the extension **Basic** page, set the general settings of the extension.



- **Type**: Select **SIP**, **IAX**, or both.
- **Extension**: The extension number.
- **Caller ID**: Generally, the caller ID is the same with the extension number.
- **Registration Name**: The registration validation for the extension.
- **Registration Password**: The password is used to register the extension.
- **Concurrent Registrations**: The PBX supports registering one SIP extension on multiple phones. When a call comes to the extension, all the phones will ring. The maximum number of concurrent registrations is 5.

3. On the extension **Basic** page, set the user information.



- **Name**: Enter the user's name to identify the extension.
- **User Password**: The user can log in the PBX or log in Linkus mobile client by the user password.
- **Email**: The user can reset his/her login password, receive voice mails, faxes, or receive PBX notifications via this email address.
- **Mobile Number**: The user can receive the PBX notifications or forwarded calls on this mobile number.
- **Prompt Language**: If the user works in a foreign language, you can set a specific system prompt for the user.

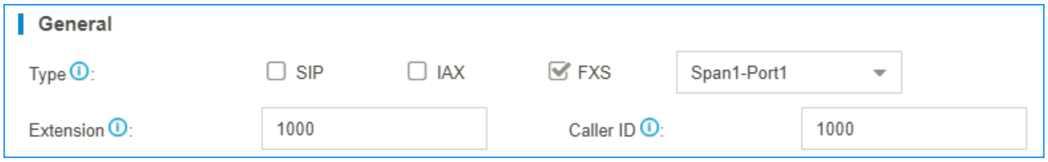
4. Optional: Click **Features**, **Advanced**, or **Call Permission** tab to configure other settings.
5. Click **Save** and **Apply**.

- Auto Provision Your Phones
- Manaually Register Your Phones

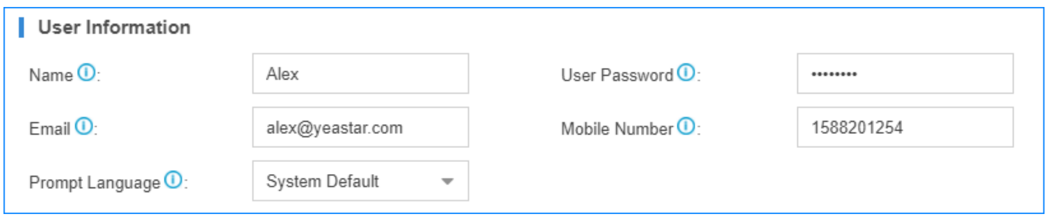
Add an FXS Extension

To add an FXS extension, you need to install an S2 module or SO module on the PBX, then connect an analog phone to the FXS port.

1. Go to **Settings** > **PBX** > **Extensions**, click **Add**.
2. On the extension **Basic** page, set the general settings of the extension.



   - **Type**: Select **FXS** and select an FXS port.
   - **Extension**: The extension number.
   - **Caller ID**: Generally, the caller ID is the same with the extension number.
3. On the extension **Basic** page, set the user information.

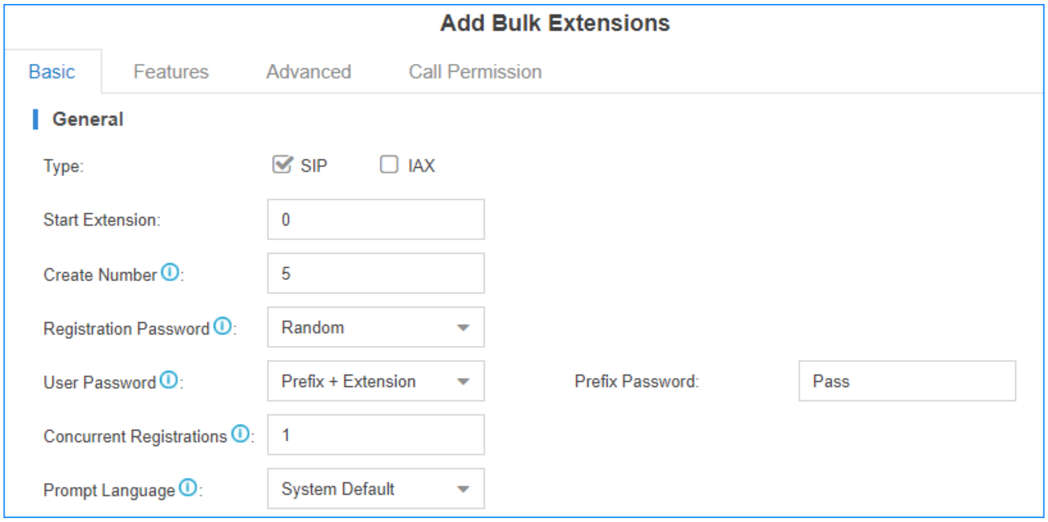


   - **Name**: Enter the user's name to identify the extension.
   - **User Password**: The user can log in the PBX or log in Linkus mobile client by the user password.
   - **Email**: The user can reset his/her login password, receive voice mails, faxes, or receive PBX notifications via this email address.
   - **Mobile Number**: The user can receive the PBX notifications or forwarded calls on this mobile number.
   - **Prompt Language**: If the user works in a foreign language, you can set a specific system prompt for the user.
4. Optional: Click **Features**, **Advanced**, or **Call Permission** tab to configure other settings.
5. Click **Save** and **Apply**.

Add Bulk Extensions

You can add bulk SIP/IAX extensions.

1. Go to **Settings** > **PBX** > **Extensions**.
2. Click **Bulk Add**.
3. In the **Basic** page, configure the following settings:



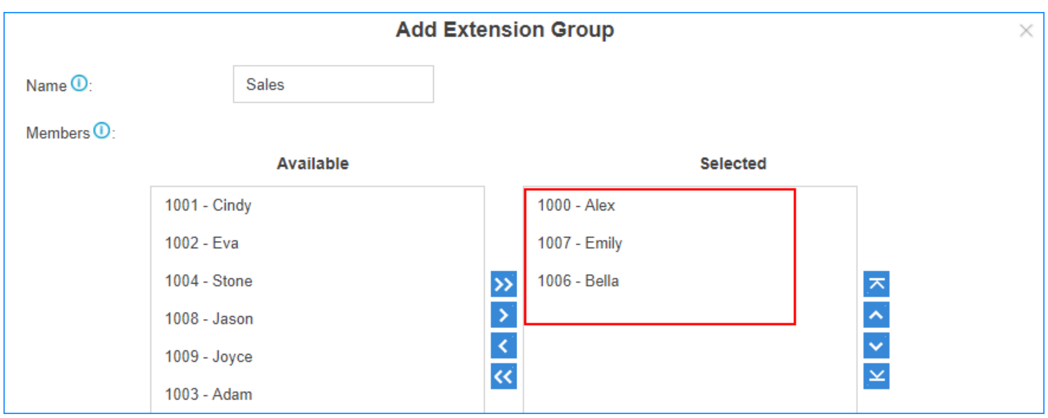
- **Type**: Set the extension type.
- **Start Extension**: Enter the first extension number. The system will create extensions in bulk starting with the extension number.
- **Create Number**: Enter how many extensions to create.
- **Registration Password**: Set the registration password.
- **User Password**: The users can log in the PBX or log in Linkus mobile client by the user password.
- **Concurrent Registrations**: The PBX supports registering one extension on multiple phones. When a call comes to the extension, all the phones will ring. The maximum number of concurrent registrations is 5.
- **Prompt Language**: You can set specific prompt language for the extensions.

4. Optional: Click **Features**, **Advanced**, or **Call Permission** tab to configure other settings.
5. Click **Save** and **Apply**.

Add an Extension Group

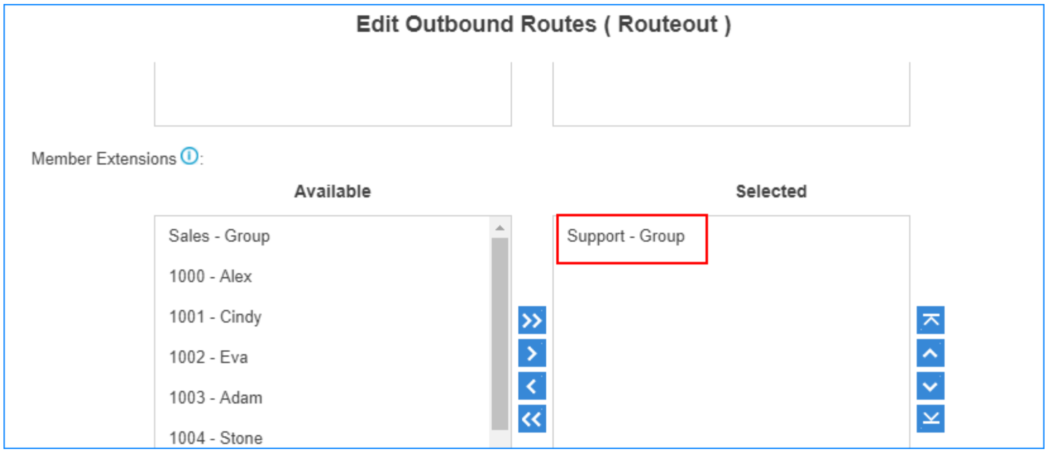
You can assign and categorize extensions in different groups. Extension groups simplify the configuration process.

1. Go to **Settings** > **PBX** > **Extensions** > **Extension Group**, click **Add**.
2. Set the **Name** to help you identify the group.
3. Select the extensions from the left box to the right box.



4. Click **Save** and **Apply**.

You can use the extension groups when you configuring the outbound routes, ring groups, queues, etc.

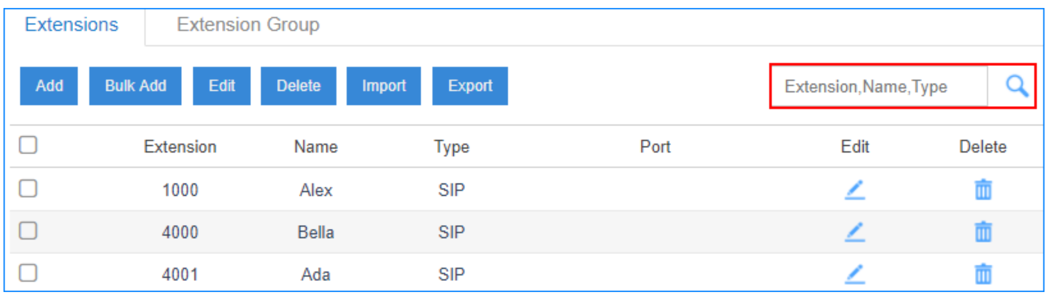


Search Extensions

To quickly find an extension, you can search extensions by extension number or extension name.

1. Go to **Settings** > **PBX** > **Extensions**.
2. In the search field, enter extension number or user name.

   The page automatically displays the matched extensions.



Import/Export Extensions

You can import extensions from a CSV file to the PBX. This will save your time if you need to add a large number of extensions on the PBX. You can also export the PBX's extensions to a CSV file. This will allow you to modify extensions and re-import them to your PBX.

1. Go to **Settings** > **PBX** > **Extensions**.
2. To import extensions, click **Import**.
3. On the **Import Extension** page, click **Browse** to choose a CSV file.

Note:

- The name of the CSV file should not contain special characters.
- The `type`, `username`, `regiserpassword` and `loginpassword` are required fields, your CSV file must contain these column names.
- For `registerpassword` field, if you don't enter the value, PBX will assign random password for the extension.
- For `loginpassword` field, if you don't enter the value, PBX will set the password as "pass" + "extension number".

Below is an example of CSV file.

	A	B	C	D	E	F	G
1	type	username	registerpassword	fullname	email	loginpassword	maxregistrations
2	SIP	1000	1000ssssSSS	Alex	alex@yeastar.com	39a77e244b04959ccc	2
3	SIP	1001	SxHpsw5k	Cindy	cindy@yeastar.com	23605f772f4f15e5fa	2
4	SIP	1002	6Y25Vsi7	Eva	eva@yeastar.com	bc510a4c4299a91f69	2
5							

4. Click **Import** to import the CSV file to the PBX.

5. To export extensions, select the desired extensions or select all the extensions, click **Export** to export the extensions to a CSV file.

Related reference

Extension CSV File Columns on page 39

You can change the column order of the CSV file. The column names in the CSV file are case-sensitive.

Call Monitoring

What is Call Monitoring?

Call Monitoring allows authorized users to monitor another extension user's call in real time. The supervisor can dial "feature codes" + "extension number" to monitor the extension user's call.

The PBX supports the following monitor modes:

- **Listen** (Default code: *90)

  Listen mode allows supervisor to listen to a call in real time.
- **Whisper** (Default code: *91)

  Whisper mode allows the supervisor to speak/coach his staff to help them handle a call.

  The other party could not hear the supervisor's voice.
- **Barge-in** (Default code: *92)

  Barge-in mode allows the supervisor to join a call.

Related tasks

Manage Monitor Permission of Extensions on page 36

Manage Monitor Permission of Extensions

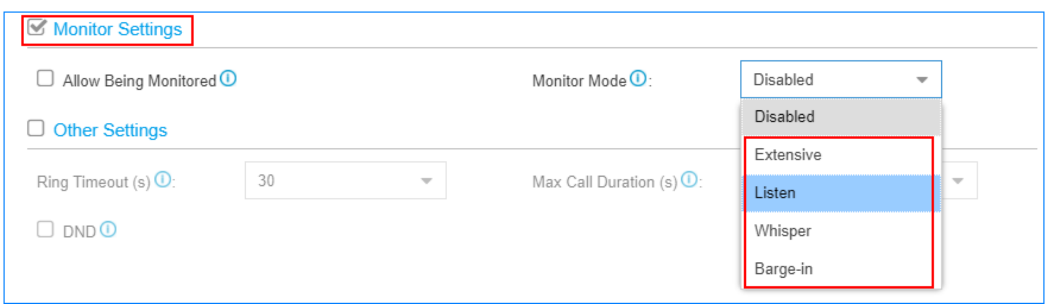
You need to set monitor settings for both the supervisors and the monitored users.

1. Enable and select a monitor mode for the supervisors.

   a) Go to **Settings** > **PBX** > **Extensions**, select the desired extensions, click **Edit** to bulk edit the extensions.

   b) On the configuration page, click **Features**.

   c) Check **Monitor Settings**, select a monitor mode for the users.

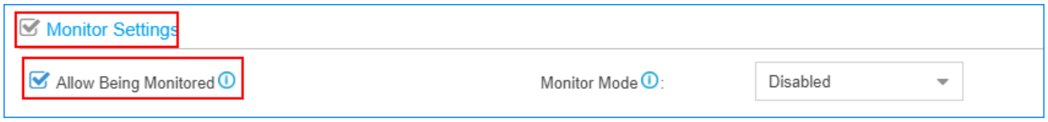


Note: If you choose **Extensive**, the users can use any of the three monitor modes.

   d) Click **Save** and **Apply**.

2. Set the extensions which will be monitored.

a) Go to **Settings** > **PBX** > **Extensions**, select the desired extensions, click **Edit** to bulk edit the extensions.
b) On the configuration page, click **Features**.
c) Check **Monitor Settings**, and check **Allow Being Monitored**.



Related concepts

What is Call Monitoring? on page 36

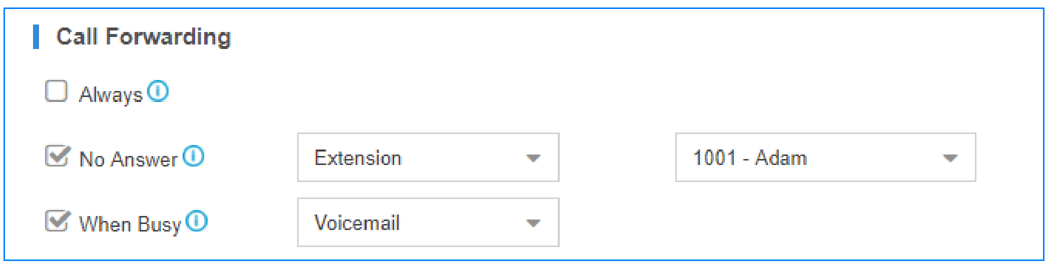
Call Forwarding

Call Forwarding is used to forward calls to other destination based on conditions such as **No Answer**, **When Busy**, and **Always**.

Configure Call Forwarding

You can set call forwarding settings for extension users. The extension users can also change their call forwarding settings on web page or dial feature codes to change the call forwarding conditions and destinations.

1. Go to **Settings** > **PBX** > **Extensions**, select extension(s) to edit.
2. On the configuration page, click **Features** tab, configure the call forwarding settings.



a) Check the condition option.

- **Always**: All the calls will be forwarded regardless of the user's state.
- **No Answer**: Calls will be forwarded to if the user doesn't answer the call.
- **When Busy**: Calls will be forwarded when the user is busy in a call.

b) Select the destination for the forwarding condition.

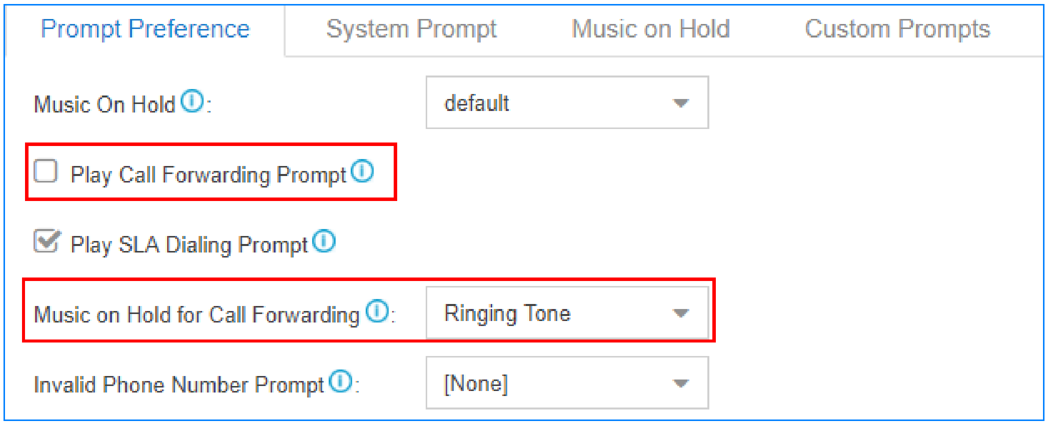
3. Click **Save** and **Apply**.

Set Call Forwarding Prompt

By default, when the PBX is forwarding an incoming call to another number, the PBX will play the call forwarding prompt "please hold when I try to locate the person you are calling", and then play the MoH music when the caller is waiting.

You can disable the call forwarding prompt and change the MoH music to a normal ring tone. In this way, the caller will not realize that the call is forwarded.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Prompt Preference**.
2. Uncheck the option **Play Call Forwarding Prompt**.
3. Set the **Music on Hold for Call Forwarding** to **Ringing Tone**.



4. Click **Save** and **Apply**.

Activate/Deactivate Call Forwarding

Extension usersYou can dial feature codes to activate or deactivate call forwarding.

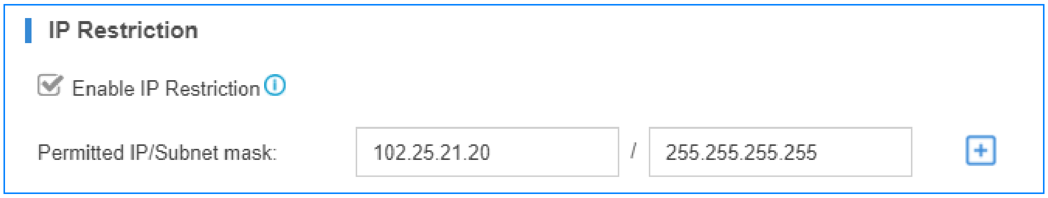
You can check or change the default call forwarding feature codes via **Settings** > **PBX** > **General** > **Feature Code**. Below are the default feature codes and the description of how to use the feature codes.

Code	Action	Example
*71	Activate call forwarding ALWAYS	• Dial *71 to forward all calls to voicemail. • Dial *716000 to forward all calls to extension 6000.
*071	Deactivate call forwarding ALWAYS	• Dial *071 to deactivate call forwarding ALWAYS.
*72	Activate call forwarding WHEN BUSY	• Dial *72 to forward calls (when the user is busy) to voicemail. • Dial *726000 to forward calls (when the user is busy) to extension 6000.
*072	Deactivate call forwarding WHEN BUSY	• Dial *072 to deactivate call forwarding WHEN BUSY.
*73	Activate call forwarding NO ANSWER	• Dial *73 to forward calls (when the user doesn't answer) to voicemail. • Dial *736000 to forward calls ((when the user doesn't answer) to extension 6000.
*073	Deactivate call forwarding NO ANSWER	• Dial *073 to deactivate call forwarding NO ANSWER.

Restrict IP Registration

To enhance the extension security, you can restrict which IP is allowed to register the extension.

1. Go to **Settings** > **PBX** > **Extensions**, click to edit an extension.
2. On the **Edit Extension** page, click **Advanced** tab.
3. Check **Enable IP Restriction**.
4. Enter the IP address and subnet mask.



5. Click to add permitted IP and subnet mask.
6. Click **Save** and **Apply**.

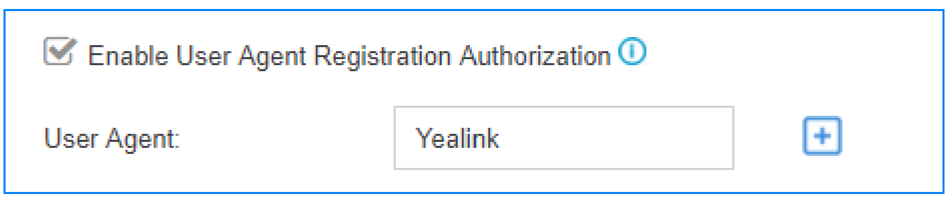
Restrict User Agent Registration

By default, the PBX allows phones to register extensions without user agent limit. To enhance the extension security, you can restrict which user agent is allowed to register the extension.

When a phone is trying to register an extension, the phone will send SIP packets that contain the user agent. If the user agent is not allowed, the registration will fail.

1. Go to **Settings** > **PBX** > **Extensions**, click to edit an extension.
2. On the **Edit Extension** page, click **Advanced** tab.
3. Check **Enable User Agent Registration Authorization**.
4. Enter the user agent according to your phones' setting.

Note: The value is case-sensitive.



5. Click to add another permitted user agent.
6. Click **Save** and **Apply**.

Extension CSV File Columns

You can change the column order of the CSV file. The column names in the CSV file are case-sensitive.

Name	Allowed Values	On Extension Page	Description
type	**SIP**	-	Required column, and needs to fill in the value.
username		Extension	The number should in the extension range.
registerpassword		Registration Password	• The password must contain 3 types of characters: lowercase letters, uppercase letters, and numbers. • At least 6 characters.
fullname		Name	User name.
callerid		Caller ID	The same as the extension number by default.

Name	Allowed Values	On Extension Page	Description
registername		Registration Name	The same as the extension number by default.
vmsecret		Voicemail Access PIN	Numbers only.
hasvoicemail	• **yes** • **no**	Enable Voicemail	-
enablevmtoemail	• **yes** • **no**	Send Voicemail to Email	-
email		Email	-
ringtimeout	• **15** • **30** • **60** • **120** • **300**	Ring Timeout	Unit: second.
mobile		Mobile Number	-
spymode	• **none** — disable call monitor • **wisperspy** — Whisper mode • **normalspy** — Listen mode • **bargespy** — Barge-in mode • **generalspy** — Extensive	Monitor Mode	-
allowbeingspy	• **yes** • **no**	Allow Being Monitored	-
maxduration	• **-1** — Follow the system global max duration. • **0** — no limit • **300** • **600** • **900** • **1800** • **3600** • **6000**	Max Call Duration	Unit: second.
dnd	• **yes** • **no**	DND	-
alwaysforward	• **yes** • **no**	Always	-
atransferto	• **voicemail** • **extension** • **usersmobile** • **customnuber**	-	-
atransfernum		-	-

Name	Allowed Values	On Extension Page	Description
atransferext			
atransferprefix			
noanswerforward	• **yes** • **no**	No Answer	-
atransferto	• **voicemail** • **extension** • **usersmobile** • **customnuber**	-	-
ntransfernum		-	-
ntransferext			
ntransferprefix			
busyforward	• **yes** • **no**	When Busy	-
btransferto	• **voicemail** • **extension** • **usersmobile** • **customnuber**	-	-
btransfernum		-	-
btransferext			
btransferprefix			
loginpassword		User Password	• The password must contain 3 types of characters: lowercase letters, uppercase letters, and numbers. • At least 6 characters.
dtmfmode	• **rfc4733** • **inband** • **info** • **auto**	DTMF Mode	-
enableiprestrict	• **yes** • **no**	Enable IP Restriction	-
permitip		Permitted IP	-
permitmask		Subnet Mask	-
transport	• **tcp** • **udp** • **tls**	Transport	-
enablesrtp		Enable SRTP	-

Name	**Allowed Values**	**On Extension Page**	**Description**
qualify	• **yes** • **no**	Quality	-
language	• **default** — follow by the global system prompt. • **sound-en** — English • **sound-ar** —Arabic • **sound-da** — Dansk(Danish) • **sound-de** — Deutsch (German) • **sound-en_AU** —English (Australia) • **sound-en_BR** — English (British) • **sound-es_LT** — Spanish Latin • **sound-es_MX** — Spanish Mexico • **sound-es** — Spanish • **sound-fa** — Persian • **sound-fi** — Suomi (Finnish) • **sound-fi_CA** — French Canada • **sound-fr** — French • **sound-gr** — Greek • **sound-hu** — Magyar (Hungarian) • **sound-it** — Italian • **sound-iw** — Hebrew • **sound-kr** — Korean • **sound-nl** — Dutch • **sound-no** — Norwegian • **sound-pl** — Polish • **sound-pt_BR** — Portuguese (Brazil) • **sound-pt** — Portuguese • **sound-ru** — Russian • **sound-se** — Swedish • **sound-th** — Thai • **sound-tr** — Turkish • **sound-zh** — Chinese • **sound-za** — Afrikaans • **sound-cz** — Czech • **sound-sk** — Slovak • **sound-jp** — Japanese	Prompt Language	The default value is **default**, if you want to fill in other value, you need to download the relevant system prompt on the PBX via **Settings** > **PBX** > **Voice Prompts** > **System Prompt**.
maxregistrations	digits 1 to 5	Concurrent Registrations	-

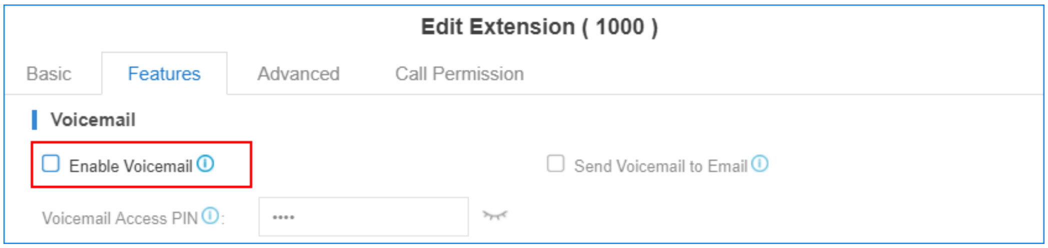
Voicemail

Voicemail function is enabled for each newly created extension user. By default, if an extension user doesn't answer a call, the call will be forwarded to the extension's voicemail.

Enable/Disable Voicemail

By default, the voicemail is enabled for extension users. You can disable the function.

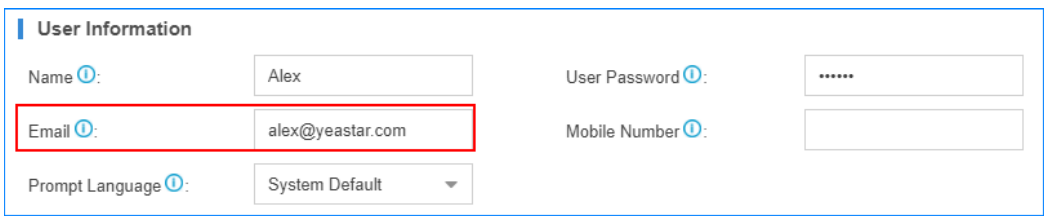
1. Go to **Settings** > **PBX** > **Extensions**, select the desired extension, click .
2. To disable voicemail, uncheck **Enable Voicemail**.



3. To enable voicemail, check **Enable Voicemail**.
4. Click **Save** and **Apply**.

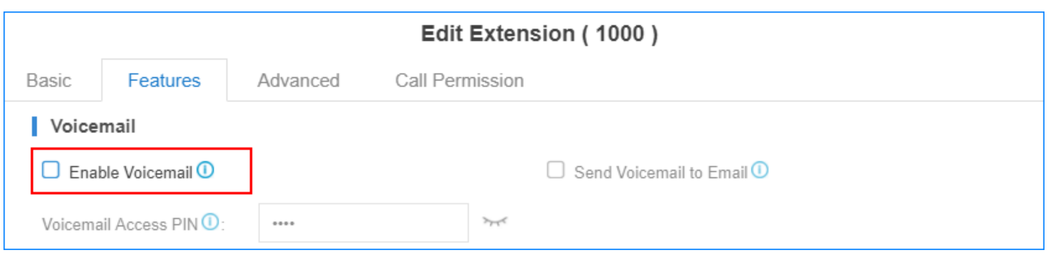
Enable/Disable Voicemail to Email

Voicemail to Email is disabled by default, if the user needs this function, you need to enable it. If **Voicemail to Email** function is enabled, the extension users can receive voicemail messages in their emails.



Note: To send voicemail to email successfully, make sure the system email is working.

1. Go to **Settings** > **PBX** > **Extensions**, select the desired extension, click .
2. To enable voicemail to email, check **Send Voicemail to Email**.



3. To disable voicemail to email, uncheck **Send Voicemail to Email**.
4. Click **Save** and **Apply**.

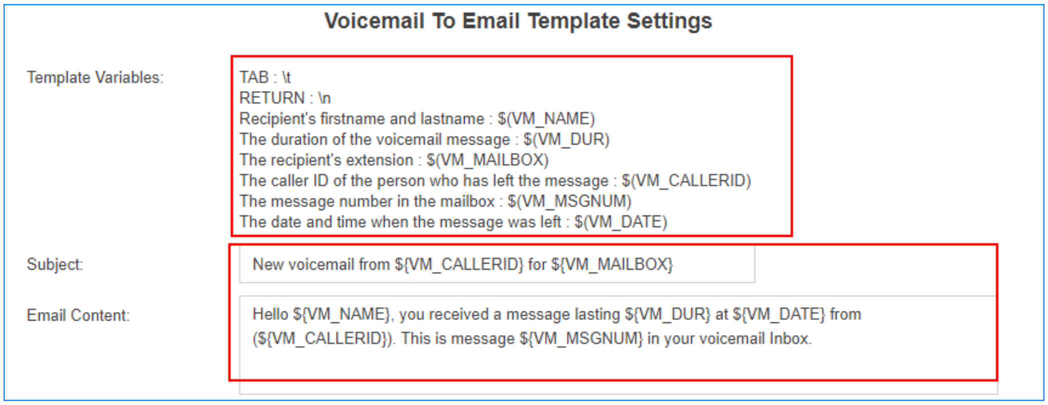
Edit Voicemail to Email Template

The PBX has a default email template for **Voicemail to Email**. You can edit the template according to your needs.

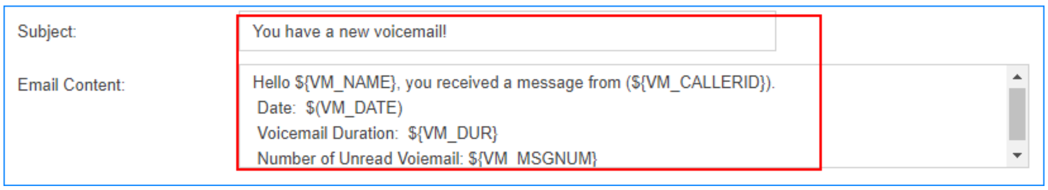
1. Go to **Settings** > **PBX** > **General** > **Voicemail**, click **Voicemail to Email Template Settings**.

You will see the description of variables and the default email contents.

Note: The variables in the email contents are unchangeable.



2. Edit the email subject and email contents.



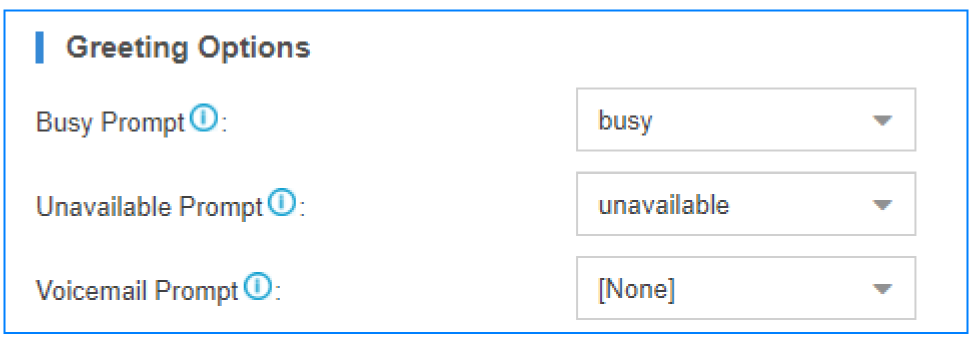
3. Click **Save** and **Apply**.

Change Global Voicemail Greetings

You can change the global voicemail greetings for all the extension users.

Note: Extension users can also log in PBX web interface to change their own voicemail greetings.

1. Go to **Settings** > **PBX** > **General** > **Voicemail**.
2. In the **Greeting Options**, change the voicemail greetings.



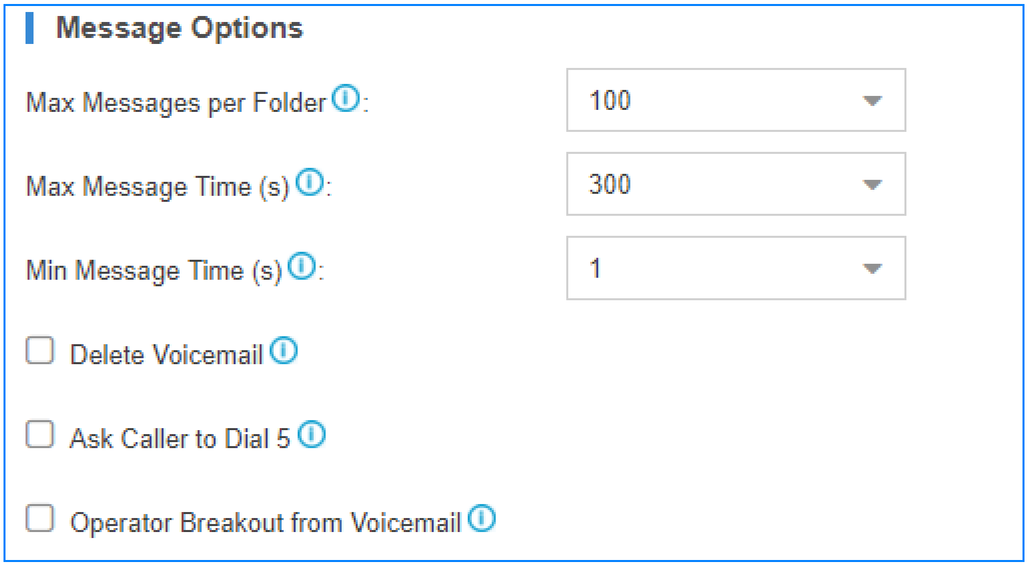
- **Busy Prompt**: Choose your custom prompt. The PBX will play the prompt when the extension user is busy.
- **Unavailable Prompt**: Choose your custom prompt. The PBX will play the prompt when the extension user is unavailable.
- **Voicemail Prompt**: By default, before a user leaves a message, the PBX will play "Please leave your message after the tone; when done, hang up or press the # key". If you set the option to **None**, the user will hear a tone "Di" directly and start to record message.

3. Click **Save** and **Apply**.

Change Voicemail Settings

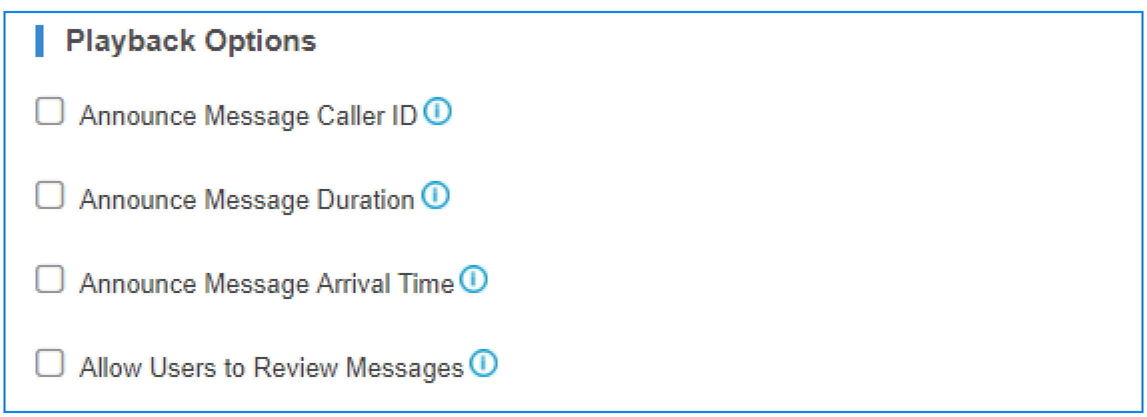
You can change the voicemail message settings, voicemail playback settings according to your needs.

1. Go to **Settings** > **PBX** > **General** > **Voicemail**.
2. In the **Message Options** section, change the message settings.



- **Max Messages per Folder**: Each extension user has a **Read** voicemail folder and an **Unread** folder. You can set the maximum number of messages per folder.
- **Max Message Time**: Set the maximum time of one message.
- **Min Message Time**: Set the minimum time of one message.
- **Delete Voicemail**: This function will work if you enable **Send Voicemail to Email**. If the voicemail is forwarded to the user's email, PBX will delete voicemails from the user's voicemail folder.
- **Ask Caller to Dial 5**: By default, when the caller accesses a user's voicemail, PBX starts to record message automatically. If you want to prompt the caller first, you can enable this option. The caller needs to dial 5 first, then starts to record message.
- **Operator Breakout from Voicemail**: If enabled, the users can dial 0 to exit the voicemail destination of an IVR.

3. In the **Playback Options** section, change the playback settings.



- **Announce Message Caller ID**: If enabled, the PBX will announce who left the message.
- **Announce Message Duration**: If enabled, the PBX will announce the message duration.
- **Announce Message Arrival Time**: If enabled, the PBX will announce when the message was received.
- **Allow Users to Review Messages**: If enabled, the users can review their recorded message, and then send the messages.

4. Click **Save** and **Apply**.

Check Voicemail Messages

Extension users have multiple ways to check their voicemail messages.

Check Voicemail on Linkus

Log in Linkus, go to **Me** > **Voicemail** to check your voicemail.

Check Voicemail on a Phone

- Dial feature code *2 on a phone

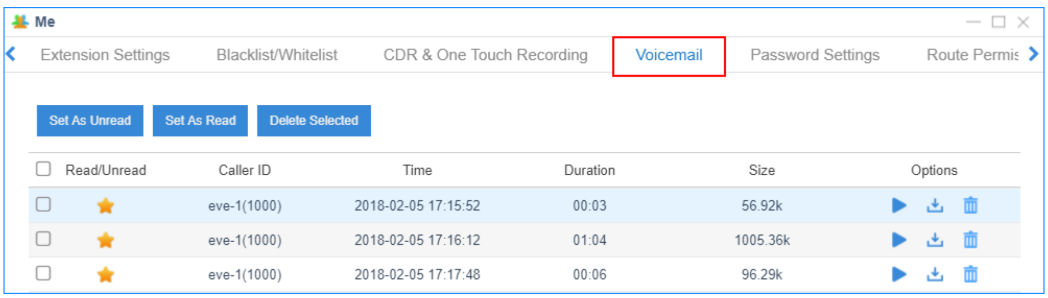
  A user can dial *2 on his own phone to check voicemail.
- Dial feature code *02 on a phone

  A user can dial *02 on other user's phone to enter the voicemail main menu, then enter his/her extension number and voicemail PIN to check voicemail.

Check Voicemail on Web Page

Extension users can log in the PBX web page to check their own voicemails.

- User name: The extension user's email address.
- Password: The extension's **User Password**.



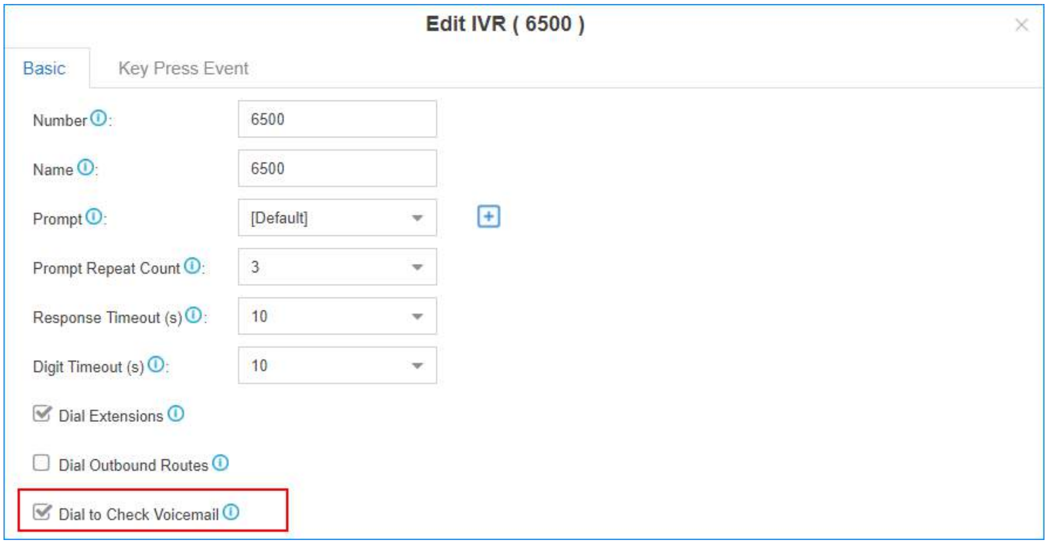
Check Voicemail via Email

If voicemail to email is enabled for an extension user, the user can check voicemails in his/her email box.

Check Voicemail via IVR

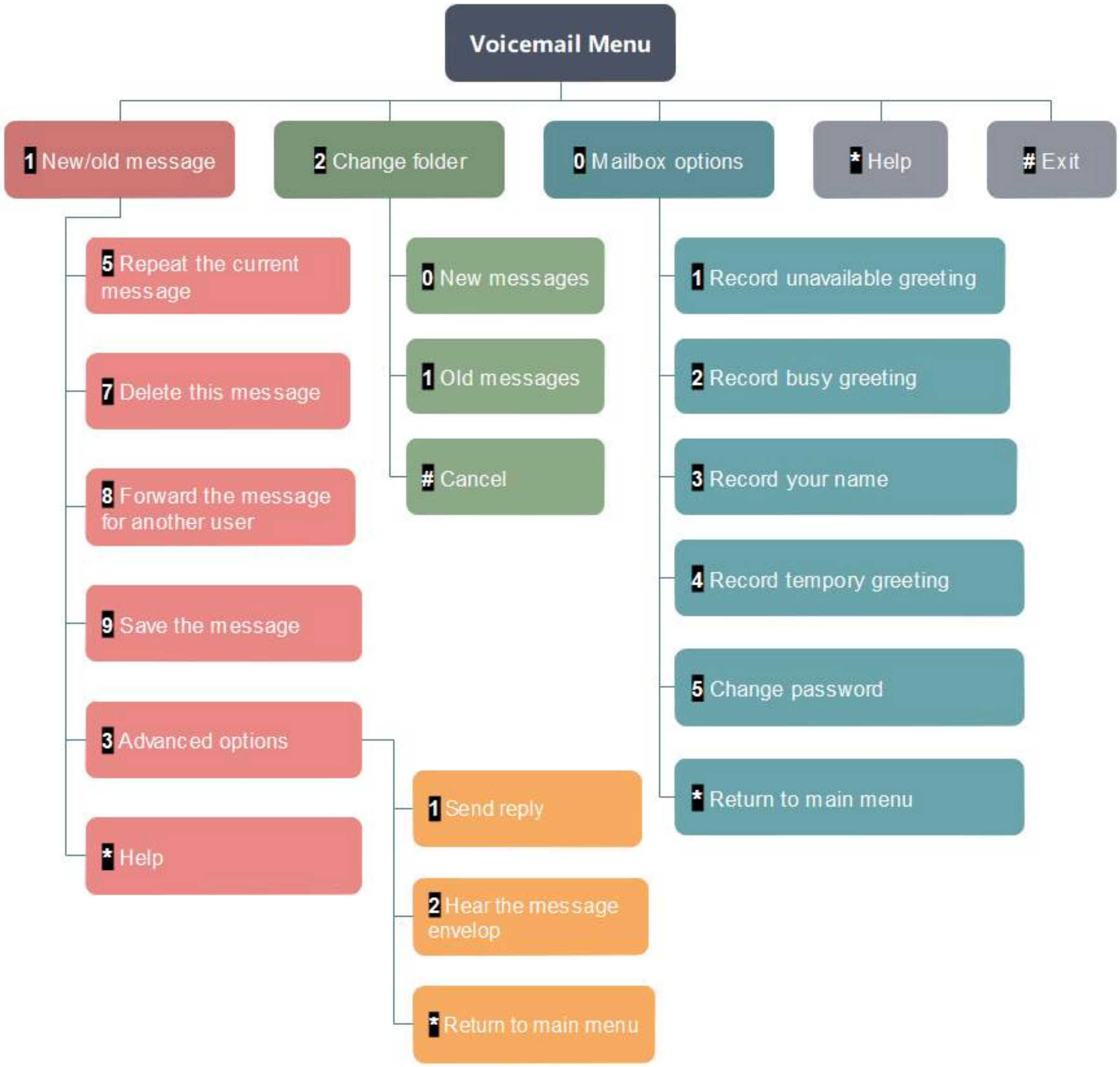
If you check the option **Dial to Check Voicemail** for an IVR; users can access the IVR to check their voicemails. This solution is for the users who are outside the office to check their voicemails.

Tip: If the users are using Linkus, they can dial *2 directly to check their voicemails.



Voicemail Menu

Extension usersYou can dial *2 on their phonesyour phone to access the voicemail menu. Below is the detailed voicemail menu.



Auto Provisioning

What is Auto Provisioning?

Auto Provisioning function helps you set up your IP phones and Yeastar TA VoIP gateways in bulk.

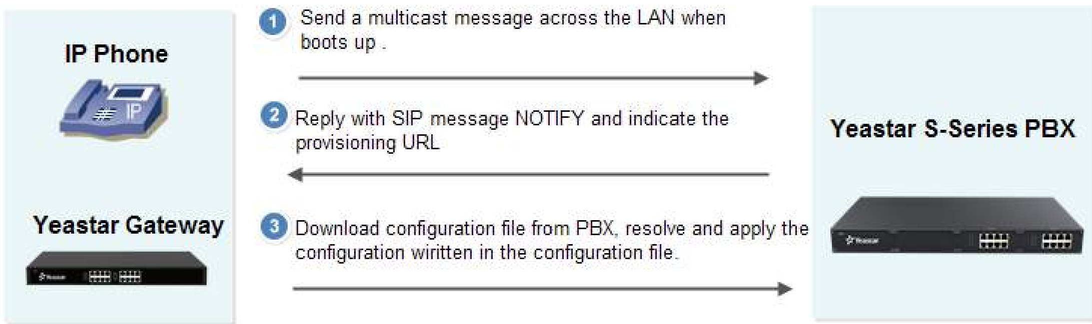
Using auto provisioning, you can instruct phone or Yeastar gateway to retrieve its configuration from Yeastar PBX. Once you provision your phones or gateways, the phones or gateways will automatically configure themselves correctly. In this way, you can manage your phones or gateways centrally without having to configure your phones one by one.

Auto Provisioning Method

Yeastar S-Series VoIP PBX supports 2 provisioning methods:

- **PnP (Plug and Play)** – apply to Yeastar gateways and IP phones that support PnP feature.
- **DHCP** – Yeastar gateways and all supported IP phones could use this method. This method is typically for legacy phones (from a previous PBX installation, e.g. Cisco or Polycom) can be provisioned via DHCP method only.

PnP Method

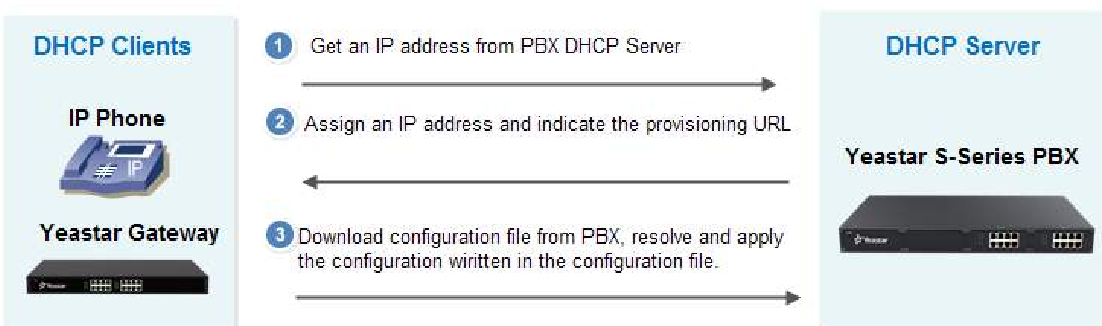


1. Plug the phone/gateway into the same network with the PBX.
2. Make sure that the phone/gateway's PnP feature is enabled.
3. Configure the phone/gateway on the **Auto Provisioning** page.
4. Click **Save** and reboot the phone/gateway.
5. The phone/gateway will send a multicast message across the LAN when boots up.
6. The PBX will send a provisioning URL to the phone.
7. The phone download configuration file from the URL, resolve and apply the settings.

DHCP Method

If your phone does not support plug and play, you can use DHCP mode to do auto provisioning.

Note: For cisco, Polycom phones, you should provision the phones by DHCP mode.



1. Disable DHCP server in your local network.
2. Enable **DHCP Server** in Yeastar S-Series PBX (**Settings** > **System** > **Security** > **Service**).

   Make sure that the PBX is the only DHCP server in your local network.
3. Set the phone network mode as "DHCP".
4. Configure the phone on the **Auto Provisioning** page.
5. Click **Save** and reboot the phone.
6. The PBX will assign an IP and send a provisioning URL to the phone.
7. The phone download configuration file from the URL, resolve and apply the settings.

Related reference

Supported IP Phone Manufactures and Phone Models on page 50

Supported IP Phone Manufactures and Phone Models

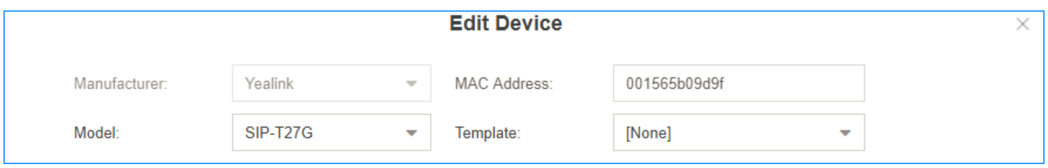
Manufacture	Phone Model	Supported Auto Provision Method
Yealink	• T19P-E2 • T21P-E2, T23G, T23P, T27G, T29G • T40P, T41P, T42G, T46G, T48G, T49G, T41S, T42S, T46S, T48S • T58A, T58V, T56A, T52S, T54S • W52P, W56P, W60B • CP860, CP960, CP920	• PnP • DHCP
Snom	• 320, 710, 715, 720, 725, 760, 765 • D305, D315, D345, D375, D715	• PnP • DHCP
Grandstream	• GXP1100, GXP1105, GXP1160, GXP1165 • GXP1400, GXP1405, GXP1450 • GXP1610, GXP1620, GXP1625, GXP1628, GXP1630 • GXP2130, GXP2135, GXP2140, GXP2160, GXP2170 • GXP2200 • GXV3240, GXV3275	• PnP • DHCP
Fanvil	• C01, C400, C58/C58P, C600 • X3/X3P/X3SP, X5/X5G, X4, X5S, X2, X6 • H2, H3, H5	• PnP • DHCP
Htek	• UC802, UC803, UC804, UC806, UC840, UC842, UC860, UC862 • UC902, UC903, UC923, UC924, UC926	• PnP • DHCP
Tiptel	• 3010, 3020, 3030 • 3110, 3120, 3130 • 3210, 3220, 3230, 3235, 3240, 3245, 3275, 3220XL	• PnP • DHCP
Cisco	• SPA301, SPA303 • SPA501G, SPA502G, SPA504G, SPA508G, SPA509G, SPA512G, SPA514G, SPA525G2	• DHCP
Polycom	• VVX101, VVX201, VVX300, VVX310, VVX400, VVX500, VVX600, VVX601, VVX1500 • IP321, IP331, IP335, IP450, IP550, IP560, IP670	• DHCP
Alcatel	• IP100, IP150, IP151, IP300, IP700G, IP1850, IP2015, IP251G, IP301G, IP701G, IP2115s	• PnP • DHCP
VTech	• VSP600A, VSP610A • VSP715A, VSP716A, VSP725A, VSP726A, VSP735A, VSP736A • VCS754A	• PnP • DHCP

Manufacture	**Phone Model**	**Supported Auto Provision Method**
Akuvox	• SP-R50P, SP-R52P, SP-R53P, SP-R55P, SP-R59P, SP-R55G, SP-R59G • SP-R63G, SP-R67G • VP-R47P, VP-R47G, VP-R48G, VP-R49P	• PnP • DHCP
Panasonic	• HDV100, HDV130, HDV230, HDV330, HDV430	• PnP • DHCP
Escene	• ES330-PEG, ES330-PEN, ES620-PEG, ES280-P, ES206-PN, ES205-PN	• PnP • DHCP
Mitel	• 6863i, 6865i,, 6867i, 6869i, 6873i	• DHCP
Univois	• U6S, U1S, U3S	• PnP • DHCP
Gigaset	• Maxwell 3 • Maxwell Basic	• PnP • DHCP

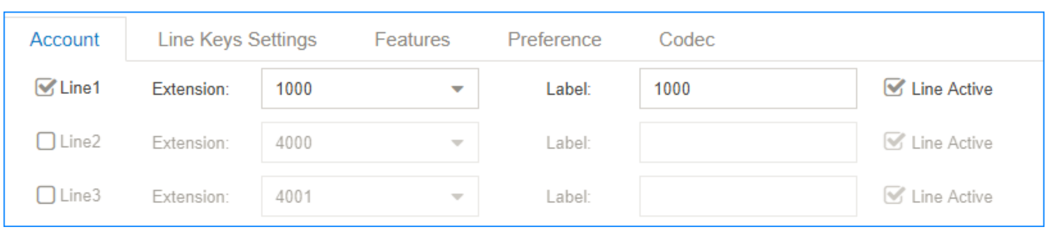
Auto Provision Phones by PnP Method

Important: For the first time to auto provision your phones on the PBX, you need to **RESET** all your phones before auto provisioning, or auto provisioning may not work.

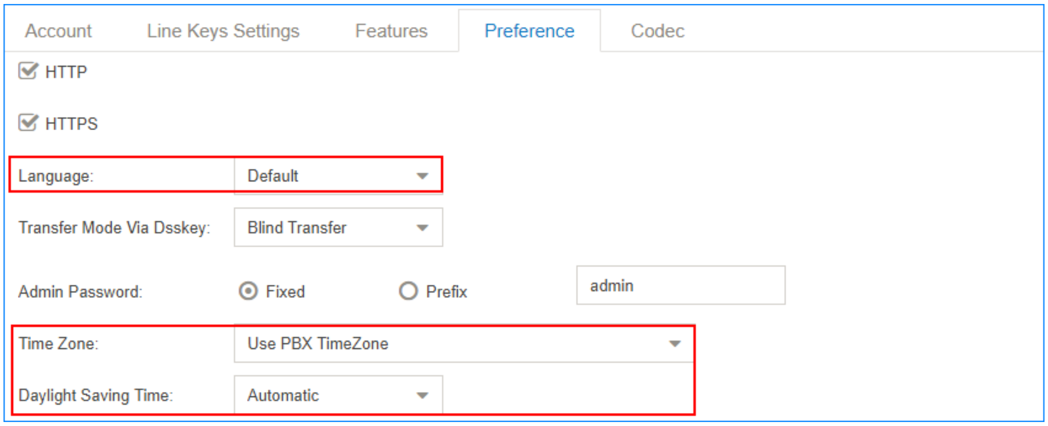
1. Connect your phones to the same network of the PBX.
2. Check your phones' basic information, including phone manufacturer, MAC address, phone model.
3. Enable **PnP** function on your phone.
4. On the PBX **Auto Provisioning** page, scan phones, all the detected phones will appear on the page.
5. Edit or add phones, set your phones' manufacturer, MAC address, model.



6. Set the phone lines.



7. Set the phones' language and time.



8. Optional: Set other phone settings.
9. Save the settings and REBOOT your phones.

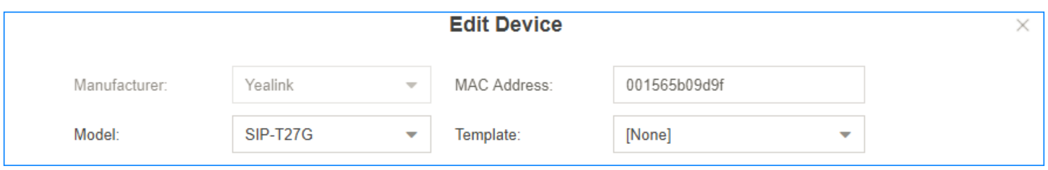
Note:

- If the phones do not reboot automatically, reboot the phone manually to make the configurations take effect.
- After auto provisioned successfully, each time when your phones reboot, they will get and apply configurations (phonebook, language, time, etc.) from the PBX.

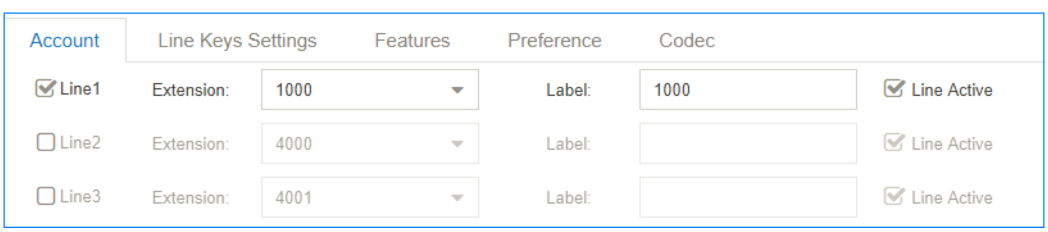
Auto Provision Phones by DHCP Method

Important: For the first time to auto provision your phones on the PBX, you need to **RESET** all your phones before auto provisioning, or auto provisioning may not work.

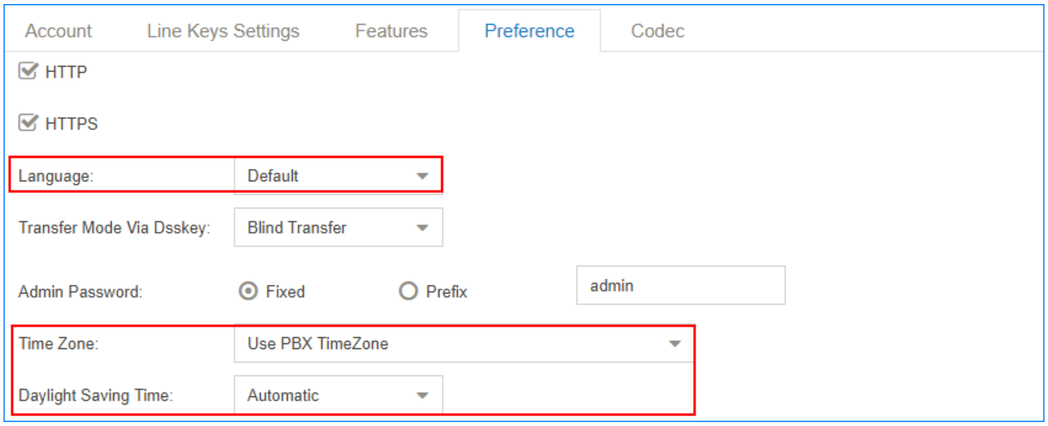
1. Connect your phones to the same network of the PBX.
2. Disable DHCP function on your router. Make sure that the Yeastar S-Series VoIP PBX is the only DHCP server in your local network.
3. Check your phones' basic information, including phone manufacturer, MAC address, phone model.
4. On your phones, enable DHCP function.
5. On the PBX **Settings** > **System** > **Security** > **Service**, enable DHCP Server.
6. On the PBX **Auto Provisioning** page, scan phones, all the detected phones will appear on the page.
7. Edit or add phones, set your phones' manufacturer, MAC address, model.



8. Set the phone lines.



9. Set the phones' language and time.



10. Optional: Set other phone settings.

11. Save the settings and REBOOT your phones.

Note:

- If the phones do not reboot automatically, reboot the phone manually to make the configurations take effect.
- After auto provisioned successfully, each time when your phones reboot, they will get and apply configurations (phonebook, language, time, etc.) from the PBX.

Update Auto Provision Settings

After you finish the auto provision configurations, you can update auto provision settings for specific phones or all the phones.

Note: Only the phone with extension registered to the PBX can update the auto provision settings.

1. Go to **Auto Provisioning** > **Device List**.
2. Select the desired phone(s), click to edit the settings.
3. Click **Save** to save your configurations.
4. On the pop-up window, click **Yes** to reboot the phone(s) and update the configurations.

Note: If the phones do not reboot automatically, reboot the phones automatically to make the configuration take effect.

Auto Provision Phonebook

You can upload a phonebook to your PBX, and auto provisioning the phonebook to all the users' phones.

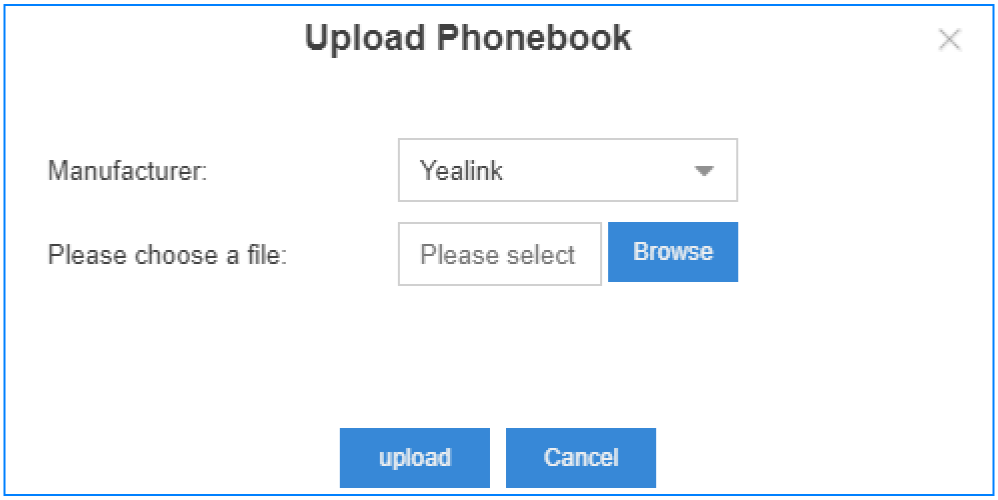
Note:

- After auto provisioning phonebooks to the users' phones, the existing phonebooks on their phones will disappear.
- Auto provisioning phonebook only works for the phones that were registered to the PBX via auto provisioning.

1. Go to **Auto Provisioning** > **Phonebook** > **Phonebook**, click **Upload Phonebook**.
2. On the configuration page, select the phone manufacturer, and click **Browse** to choose an xml file.

Note:

- The file name should not contain special characters.
- To make an xml file, contact your phone manufacturer; you can also log in the phone web interface, export an xml file to start with.



3. Click **Upload** to upload the file to PBX.
4. Reboot the phone(s) and update phonebook

Related tasks

Auto Provision Contacts on page 54

Auto Provision Contacts

You can add contacts on your PBX, and auto provision the contacts to the users' phones.

Note:

- After auto provisioning contacts to the users' phones, the existing contacts on their phones will disappear.
- Auto provisioning contacts only works for the phones that were registered to the PBX via auto provisioning.
- If you have uploaded a phonebook to the PBX, the PBX will auto provision the contacts in the phonebook, and ignore the contacts that you add in the PBX.

1. Go to **Auto Provisioning** > **Phonebook** > **Contact**.
2. Click **Add** to add a contact.
3. On the configurations page, set the contact's information.
4. Click **Save**.
5. Reboot the phone(s) and update contacts.

Related tasks

Auto Provision Phonebook on page 53

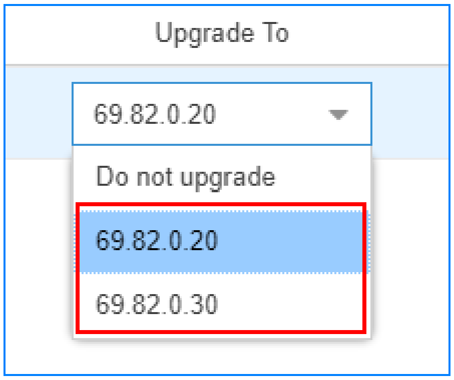
Auto Provision Phone Firmware

You can auto provision firmware of your phones.

Note: Auto provisioning phone firmware only works for the phones that were registered to the PBX via auto provisioning.

1. Go to **Auto Provisioning** > **Firmware Upgrade**, the page displays the auto provisioned phone models.
2. Click **Check for New Version**.
3. Click the **Upgrade To** drop-down menu, check if the phones have new firmware version available.

   If the phone has new firmware version, you can see the version in the drop-down menu.



4. Select a firmware version that you want to upgrade to.
5. Reboot the phone(s) and update the firmware.

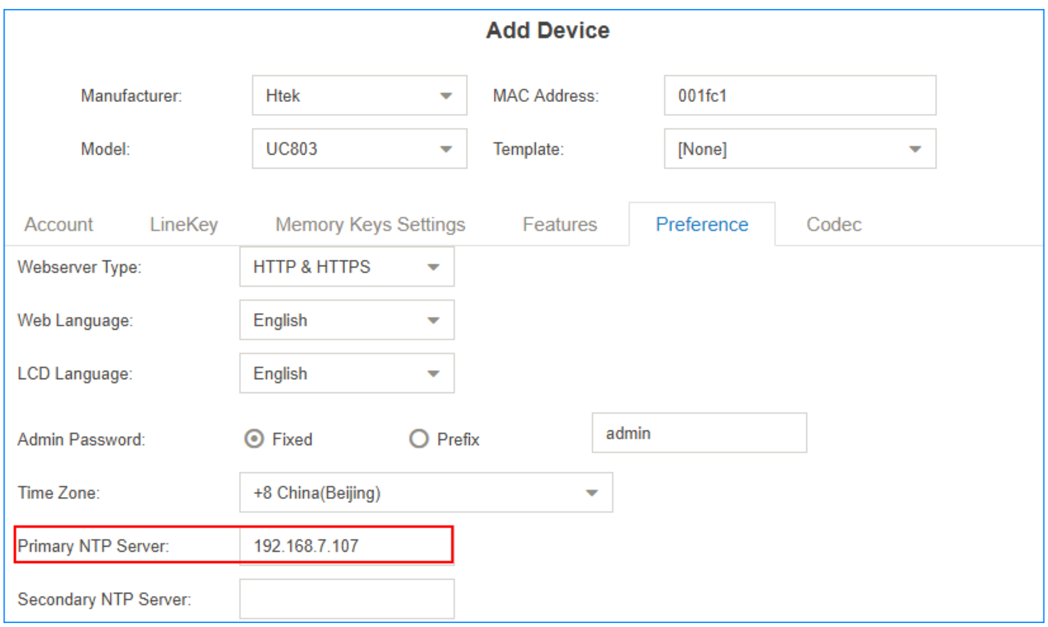
Synchronize Phone Time with the PBX

You can synchronize the users' phone time with the PBX via Auto Provisioning function.

You can also manually set the phone's NTP server as your PBX's time; the phone time will be synchronized to your PBX.

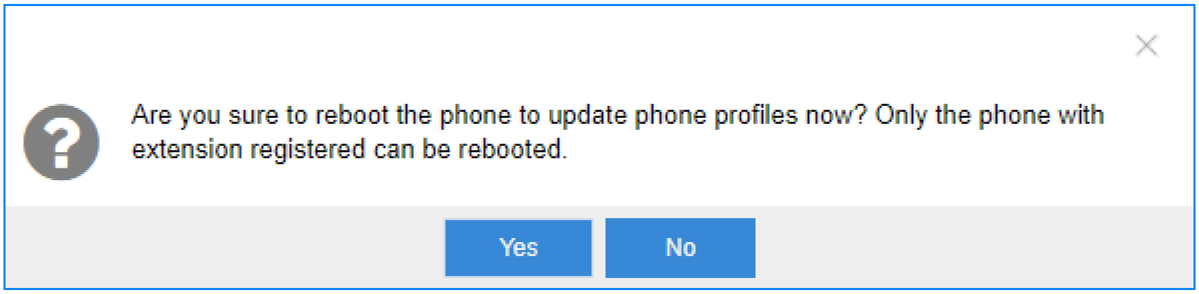
Note: Only the phone that were registered to the PBX via auto provisioning can auto provision the phone time.

1. Go to **Auto Provisioning** > **Device List**, select the desired phone(s) and edit.
2. On the configuration page, click **Preferences** tab.
3. Set the **Primary NTP Server** as IP address of your PBX.



4. Click **Save**.

   The following dialog box appears.



5. Click **Yes** to reboot the phone(s) and synchronize the phone time with the PBX.

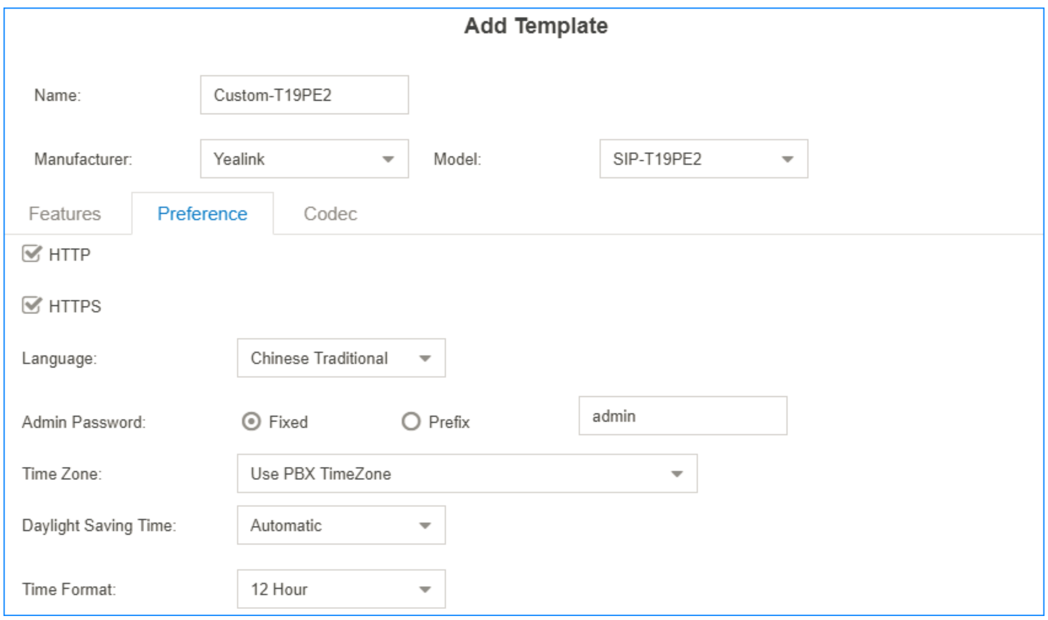
Auto Provisioning Template

Add a Custom Auto Provisioning Template

You can configure a customized auto provisioning template and apply it to your phones. In the customized auto provisioning template, you can set global settings for your phones.

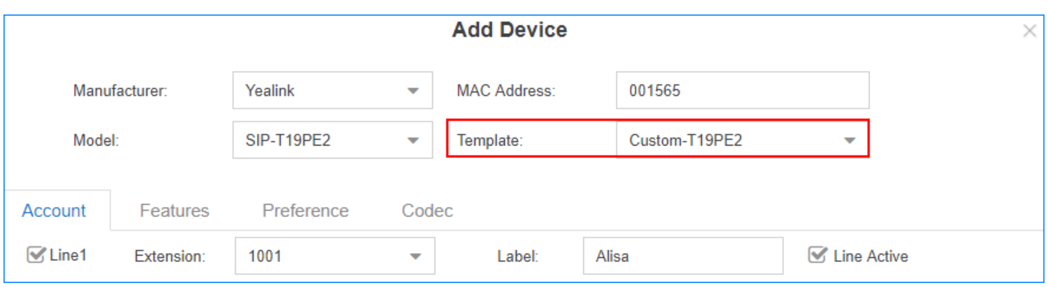
For example, if all your phones use Chinese user interface, you can set the **Language** as **Chinese** in the template.

1. Go to **Auto Provisioning** > **Template**, click **Add**.
2. Set a name for the template.
3. Set the template settings according to your needs.



4. Click **Save**.

When you are configuring auto provisioning devices, you can select the customized template.



Manually Register Your Phones

You can manually register extensions on your IP phones or soft phones.

Register an Extension on ALCATEL Phone

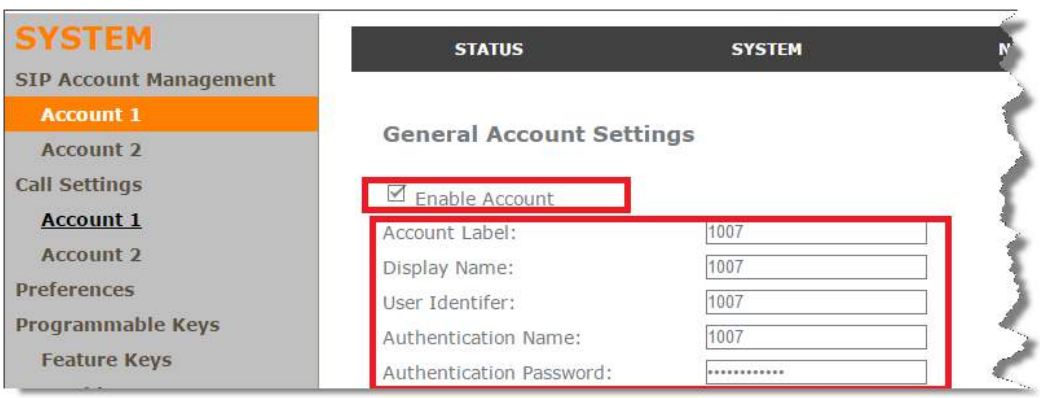
In the following instructions, we take **Temporis IP151** v1. 1. 0. B as an example to explain how to register an extension on ALCATEL IP phone.

Configure the IP address via phone user interface

1. Press **System** > **Network** > **Basic Settings** > **Dual Mode** > **WAN Setting**.
2. Choose **Static IP** and alter the **IP Address, Subnet Mask**, **Preferred DNS Server**, **Alternate DNS Server**.
3. **Apply** it after inputting the correct information.
4. **Reboot** the phone and log in the phone web user interface using the new IP address.
5. Enter the user name and password, click **Log In** to enter the web user interface.
    - User Name: admin
    - Default Password: admin

Account Registration

1. Log in the IP phone, go to **System** > **SIP Account Management**, select one account to configure.
2. Enable the account and fill in the extension information.



    - **Enable Register**: check
    - **Account Label**: The name you want to display on the phone screen.
    - **Display Name**: The name you want to display on another person's phone screen when you are calling the phone.
    - **User Identifier**: Enter the extension's **Caller ID**.
    - **Authentication Name**: Enter the extension's **Registration Name**.
    - **Authentication Password**: Enter the extension's **Registration Password**.
3. In the **SIP Server** section and **Registration** section, fill in your PBX information.

- **SIP Server**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Server Port:** Enter the SIP port of your PBX.
- **Registration**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Port:** Enter the SIP port of your PBX.

4. Click **Apply**.

If the registration is successfully, the register status would show "Registered".

Register an Extension on Cisco Phone

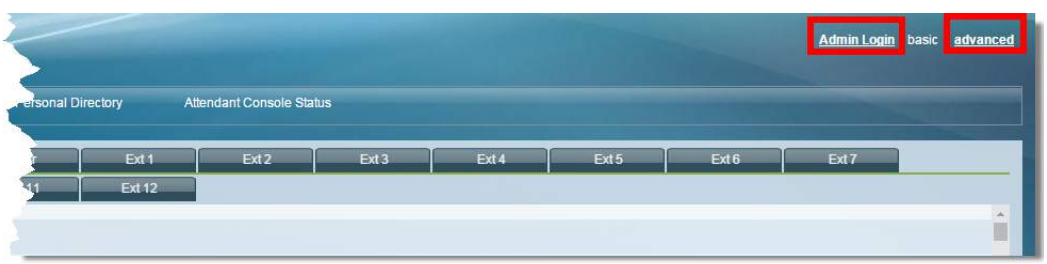
In the following instructions, we take **Cisco SPA509G** as an example to explain how to register an extension on Cisco IP phone.

This guide is applicable to:

- Cisco SPA series: 301, 303, 501G, 502G, 508G, 509G, 512G, 514G, 525G5
- Cisco CP7821

Note: For the IP phone with different firmware version, the web GUI may be different.

1. To check the IP address of the phone, press the menu key, go to **Network**, then press **Select**.
2. Type the phone IP address in your browser, click **Enter** key to access the web page of the IP phone.
3. In the upper-right corner, click **Admin Login**, then click **Advanced** to access the advanced administrator page.



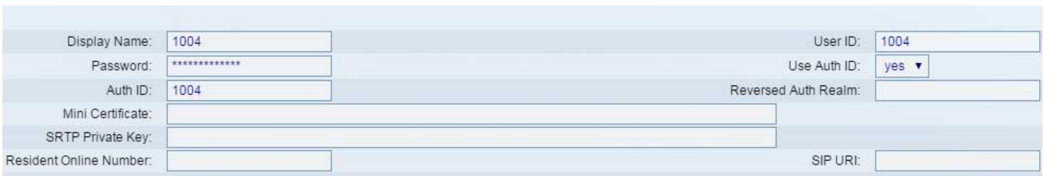
- Choose one account to configure. Here we click **EXT1** to configure account 1. Configure the account as follows:

4. Choose one account to configure. Here we choose **EXT1**.

a) Set the **Line Enable** to **Yes**.

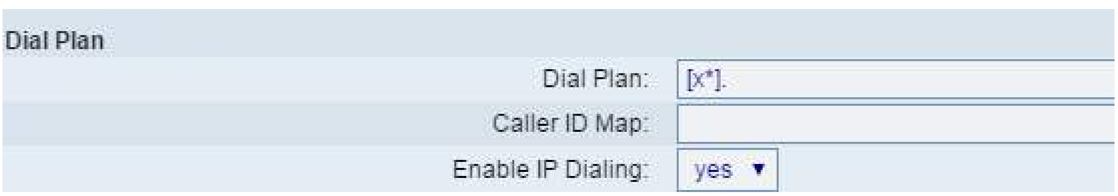
b) In the **Proxy and Registration** section, set the **Proxy** to the domain or IP address of your PBX.

c) In the **Subscriber Information** section, fill in the extension information.

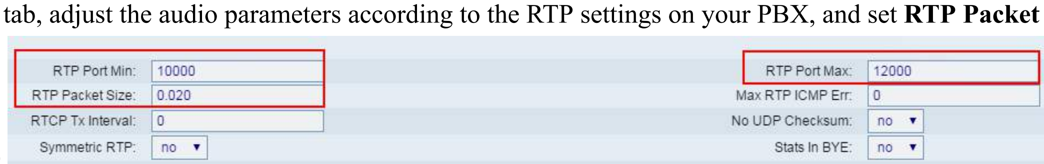


- **Display Name**: Set the name you want to appear on other phone's display when calling other phones.
- **User ID**: Fill in the extension number.
- **Password**: Fill in the extension's **Registration Password**.
- **Use Auth ID**: Set to **Yes**.
- **Auth ID**: Fill in the extension's **Registration Name**.

d) In the **Dial Plan** section, set the **Dial Plan** to `[x*].`.



5. Click **Phone** tab, adjust the audio parameters according to the RTP settings on your PBX, and set **RTP Packet**



Size to **0.020**.

6. In the bottom of the page, click **Submit All Changes**.

The phone will restart. After the phone restarts, check if the extension is registered.

Register an Extension on Grandstream Phone

In the following instructions, we take **GXP2135** as an example to explain how to register an extension on Grandstream IP phone.

This guide is applicable to:

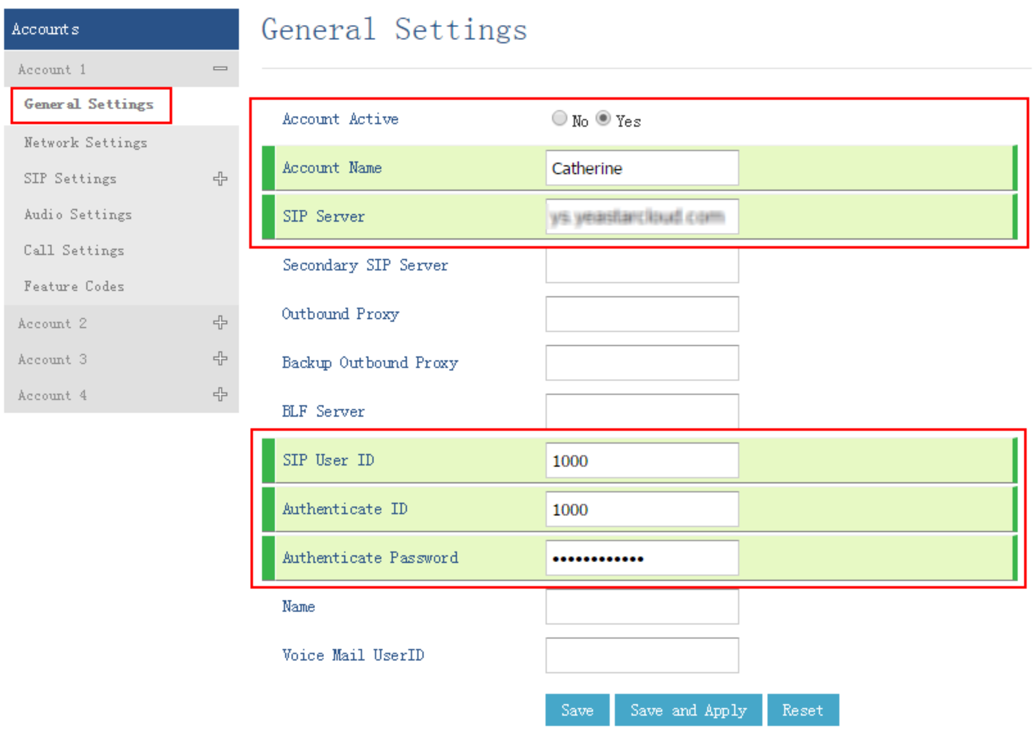
- Grandstream GXP Series 1160, 1165, 1400, 1405, 1450, 1610, 1620, 1625, 1628, 1630, 2130, 2135, 2140, 2160, 2170, 2200, 3240, 3245

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the IP phone.

- **Username**: admin
- **Default Password**: admin

2. Click **Account** tab, choose one account, and configure the general settings.



- **Account Active**: Yes
- **Account Name**: Set a name for the account, the name will be displayed on the phone LCD.
- **SIP Server**: Fill in the domain or IP address of your PBX.
- **SIP User ID**: Fill in the extension number
- **Authenticate ID**: Fill in the extension's **Registration Name**.
- **Authenticate Password**: Fill in the extension's **Registration Password**.

3. Click **Save and Apply**.
4. Go to **Status** > **Account Status** to check the account status.

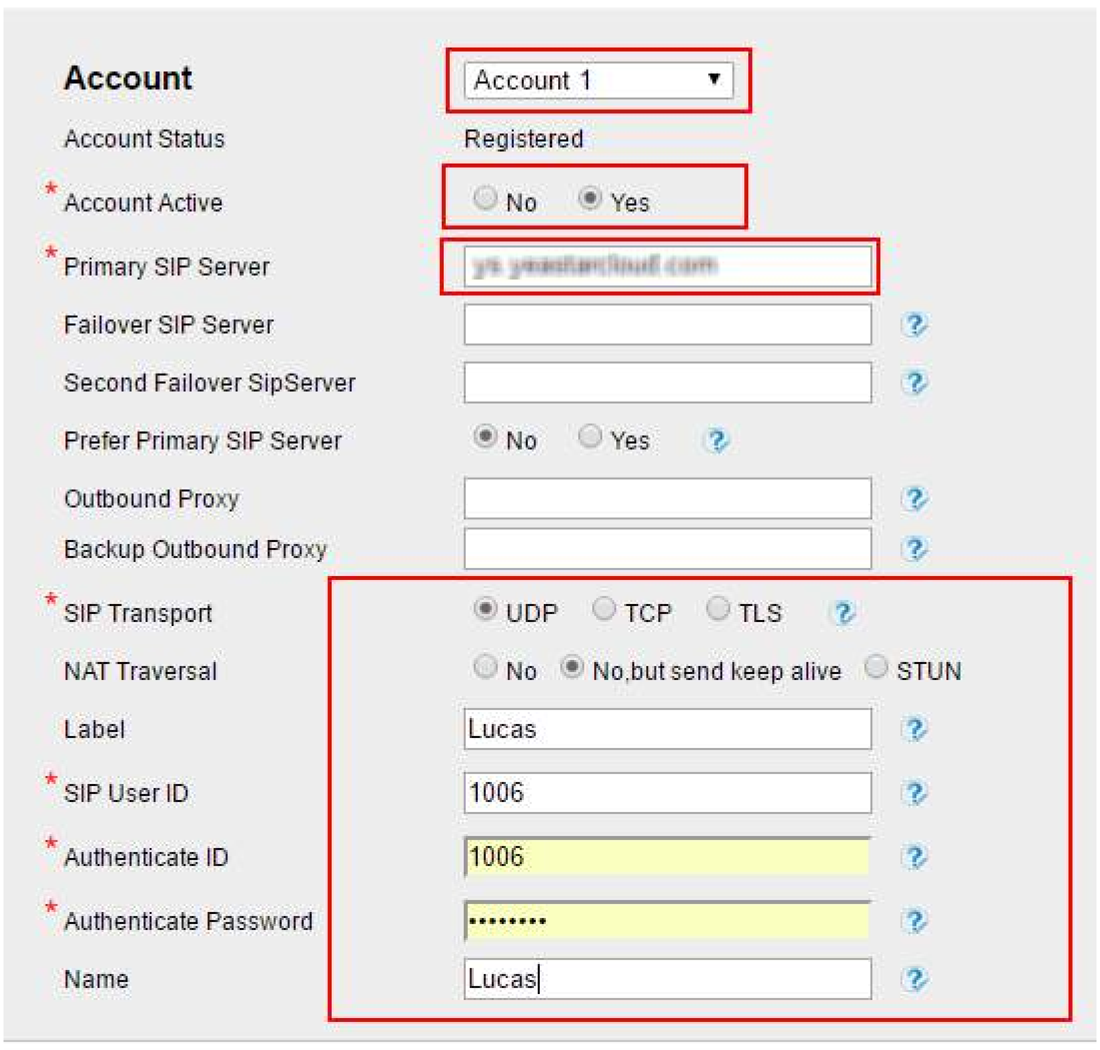
   If the extension is registered, the **SIP Registration** shows "Yes".

Register an Extension on Htek Phone

This guide is applicable to the Htek UC series 802, 803, 804, 840, 842, 806, 862, 902, 903, 923, 924, 926.

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
   - **Username**: admin
   - **Password**: admin
2. Click **Account** tab, choose one account to configure.



- **Account**: Select one account to configure.
- **Account Active**: Yes

- **Primary SIP Server**: Fill in the domain or IP address of your PBX.
- **SIP Transport**: Choose the same transport of the PBX. The default SIP transport on the PBX is UDP.
- **Label**: Set the name you want to appear on the phone screen.
- **SIP User ID**: Fill in the extension number.
- **Authentication ID**: Fill in the extension's **Registration Name**.
- **Authentication Password**: Fill in the extension's **Registration Password**.
- **Name**: The local phone name showing on the other phone when calling out.

3. Click **Save Set**.

If the extension is registered, the page will show "Registered".

Register an Extension on Panasonic Phone

This guide has been tested with Panasonic KX-HDV130 v01.008, and is applicable to the following Panasonic IP Phones.

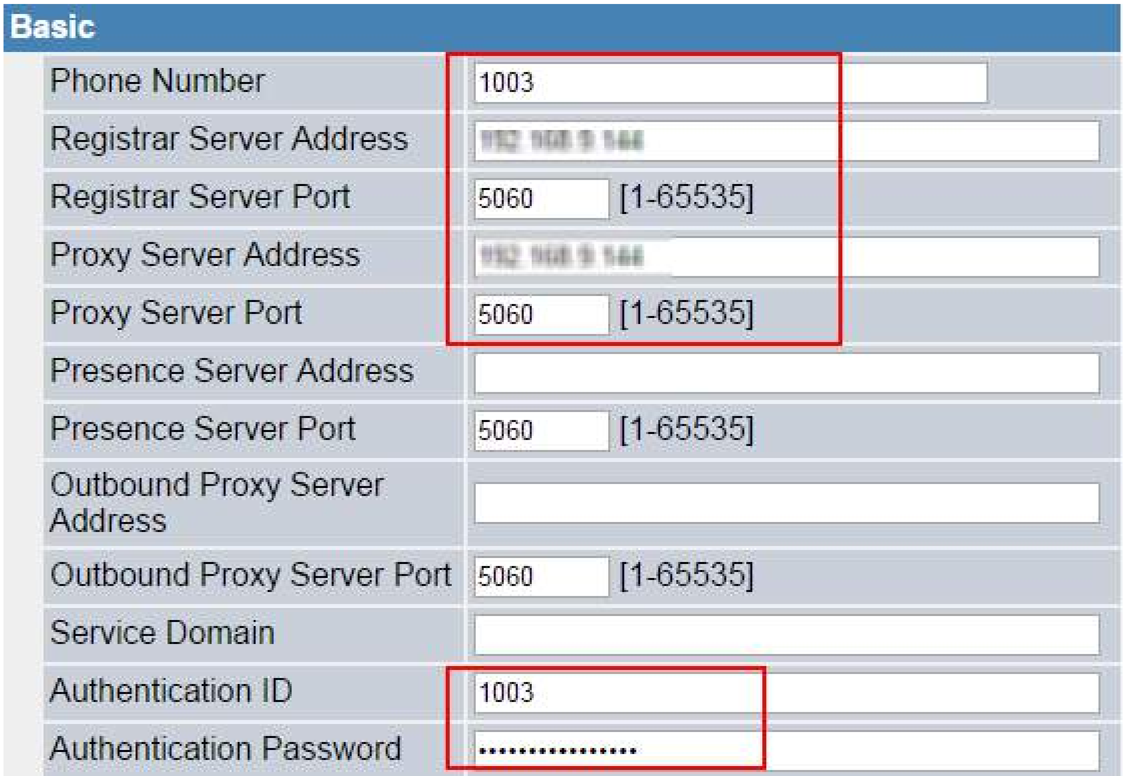
- KX-HDV130
- KX-UT113
- KX-UT123
- KX-UT133
- KX-UT136
- KX-UT248
- KX-UT670
- TGP500
- TGP550

1. Start up the phone and check its IP address.
   a) Press **Menu**.
   b) Go to **System Settings** > **Network Settings** > **IPv4 Settings** > **Static**.

2. Open the web service for the Panasonic phone.
   a) Press **Menu**.
   b) Go to **Basic Settings** > **Other Option** > **Embedded Web**.

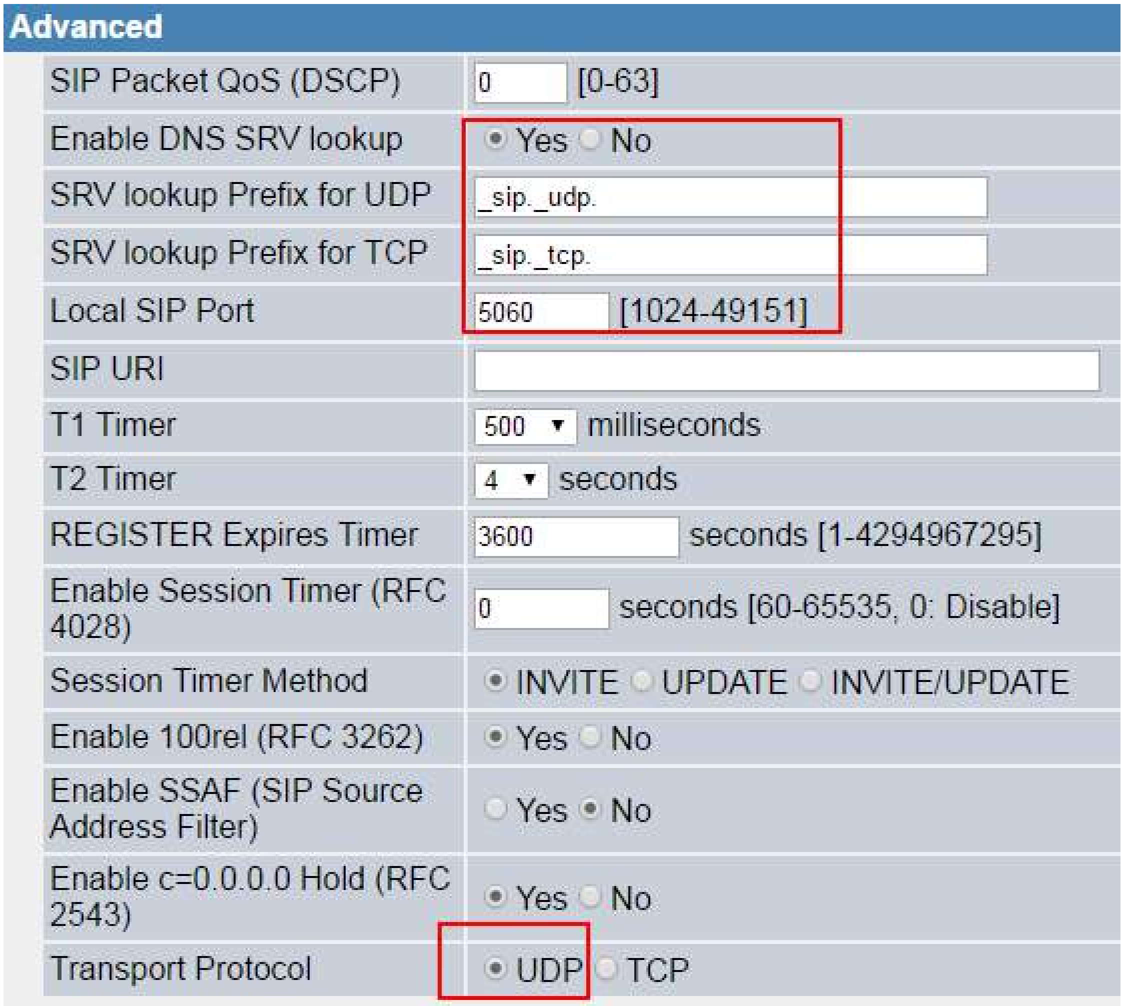
3. Log in the web page of the IP phone.
   - **Username**: admin
   - **Password**: adminpass

4. Click **VoIP**, choose a line to configure.
   a) In the **Basic** section:



- **Phone Number**: Fill in the extension number.
- **Registrar Server Address**: Fill the domain or IP address of your PBX.
- **Registrar Server Port**: Fill in the SIP port of your PBX.
- **Proxy Server Address**: Fill in the domain or IP address of your PBX.
- **Proxy Server Port**: Fill in the SIP port of your PBX.
- **Authentication ID**: Fill in the extension's **Registration Name**.
- **Authentication Password**: Fill in the extension's **Registration Password**.

b) In the **Advanced** section:



- **SRV lookup Prefix for UDP**: Enter `_sip_udp`.
- **SRV lookup Prefix for TCP**: Enter `_sip_tcp`.
- **Local SIP Port**: The SIP port number for each line must be unique, default value: 5060 (for Line 1) and 5070 (for Line 2).
- **Transport Protocol**: Choose the same transport protocol as the PBX.

5. Click **Save**.

   If the extension is registered, you can see the VoIP status shows "Registered".

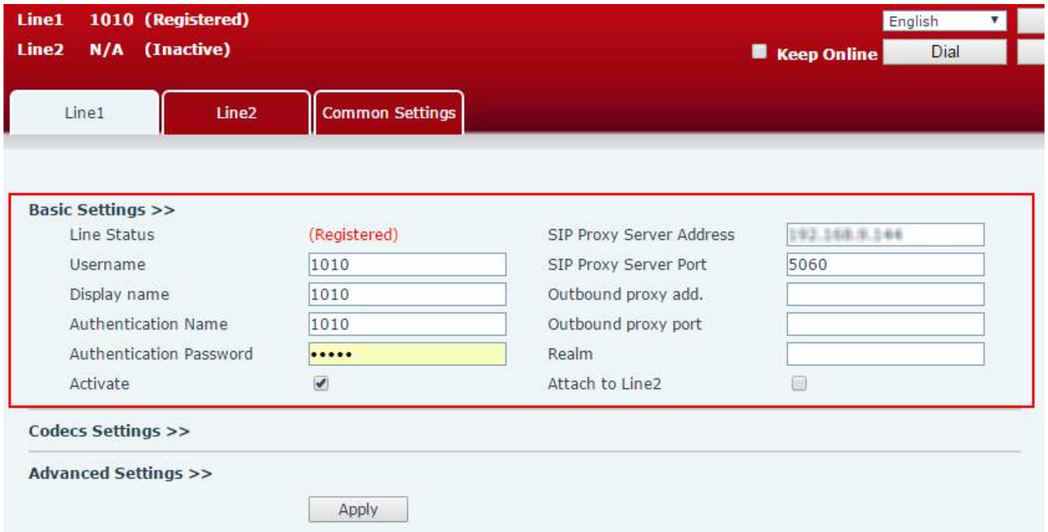
Register an Extension on Fanvil Phone

This guide is applicable to the following phones:

- Fanvil C Series: C01, C58, C58P, C400, C600
- Fanvil X3 Series: X3, X3P, X3SP
- Fanvil X5 Series: X5,X5G

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
   - **User**: admin
   - **Password**: admin
2. Click **Line** and choose a line to configure.



- **User Name**: Fill in the extension number.
- **Display Name**: Set the name you want to appear on other phone's screen when calling other phones.
- **Authentication Name**: Fill in the extension's **Registration Name**.
- **Authentication Password**: Fill in the extension's **Registration Password**.
- **Active**: Check
- **SIP Proxy Server Address**: Fill in the domain or IP address of your PBX.
- **SIP Proxy Server Port**: Fill in the SIP port of your PBX.

3. Click **Apply**.

   If the extension is registered, the **Line Status** will show "Registered".

Register an Extension on Polycom Phone

This guide is applicable to the following phones:

- Polycom VVX Series: 101, 201, 300, 310, 400, 500, 600, 601, 1500
- Polycom SoundPoint Series: IP321, IP331, IP335, IP450, IP550, IP560, IP670

Note: For the IP phone with different firmware version, the web GUI may be different.

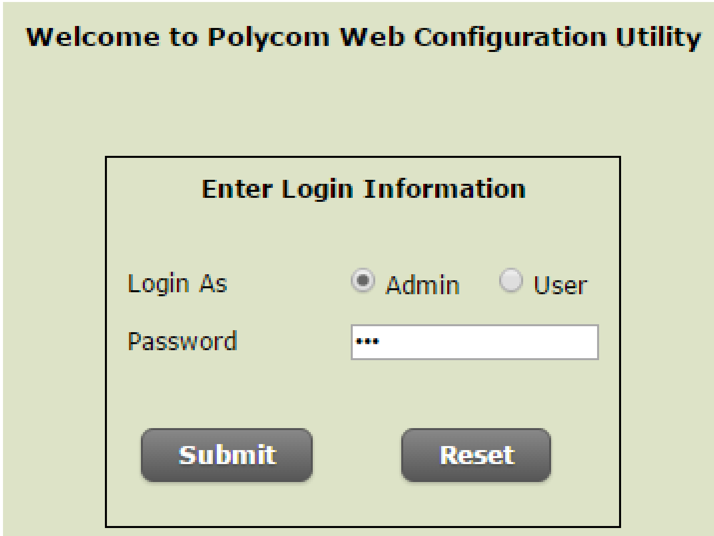
1. To check the IP address of the phone, press **Menu** on the phone, go to **Settings** > **Status** > **Network** > **TCP/IP Parameter**.
2. Enable Web service for the phone.

   a) Press **Menu** on the phone, go to **Settings** > **Advanced**, enter the password 456.

   b) Go to **Administration Settings** > **Web Server Configuration**, configure the following:

      - **Web Server**: Enabled
      - **Web Config Mode**: Choose HTTP or HTTPS
3. Log in the web page of the phone.

   Note: For the firmware version 5.5.0 or later, the phone only supports HTTPS web login. You need use HTTPS to log in the web page. For example, type `https://192.168.6.160` in your web browser to access the phone web page.

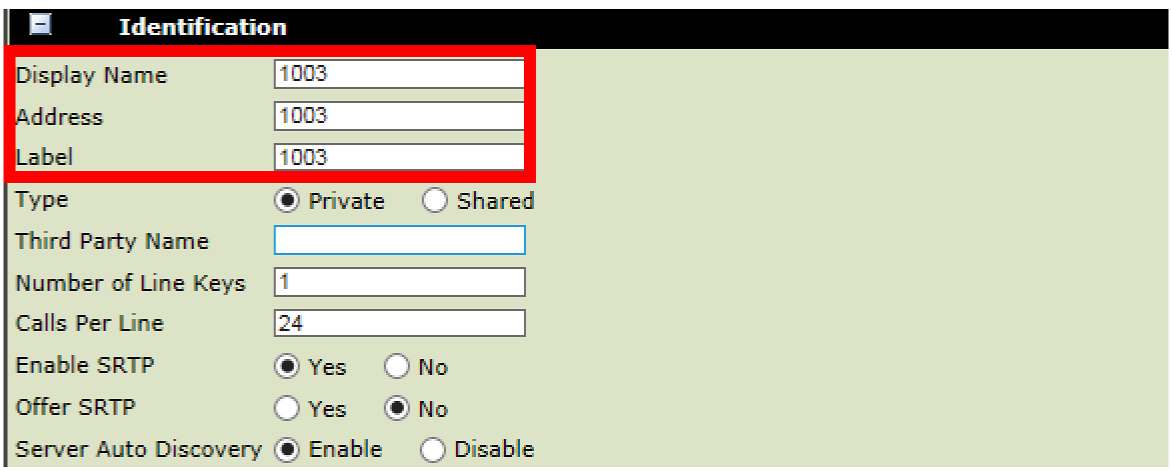


- **Login as**: Admin
- **Password**: 456

4. Go to **Settings** > **Lines**, choose a line to configure.

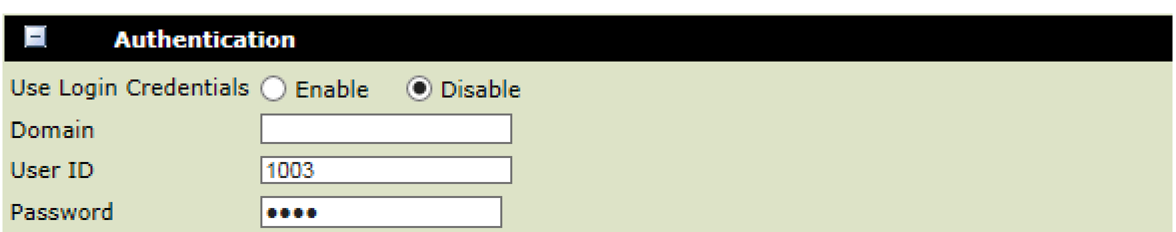
a) Enable **SIP Protocol**.

b) Expand **Identification** option, and set as the following:



- **Display Name**: Set the name you want to appear on other phone's screen when calling other phones.
- **Address**: Fill in the extension number.
- **Label**: Set the name you want to appear on the phone screen.

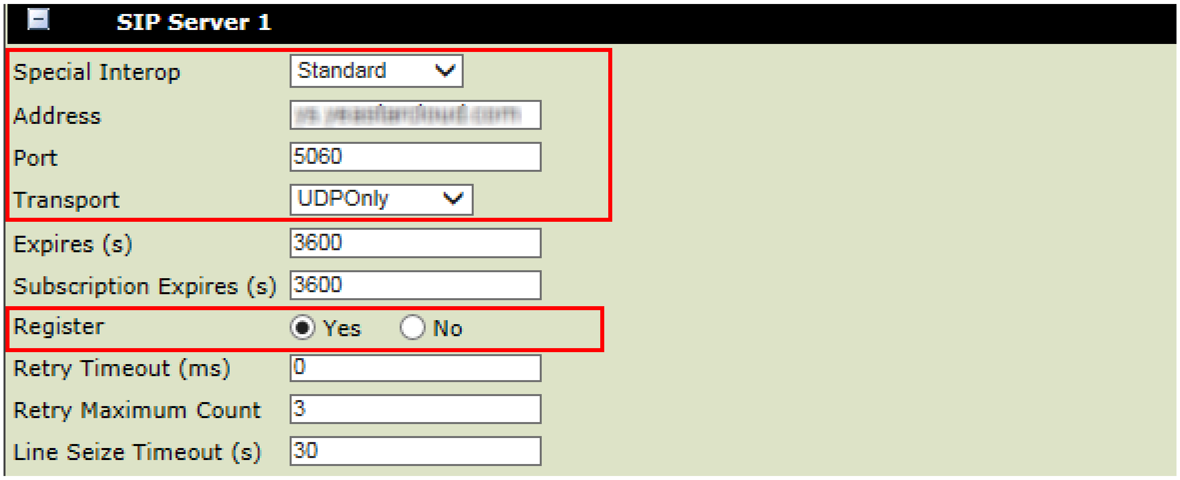
c) Expand **Authentication** option, and set as the following:



- **User Login Credentials**: Disable

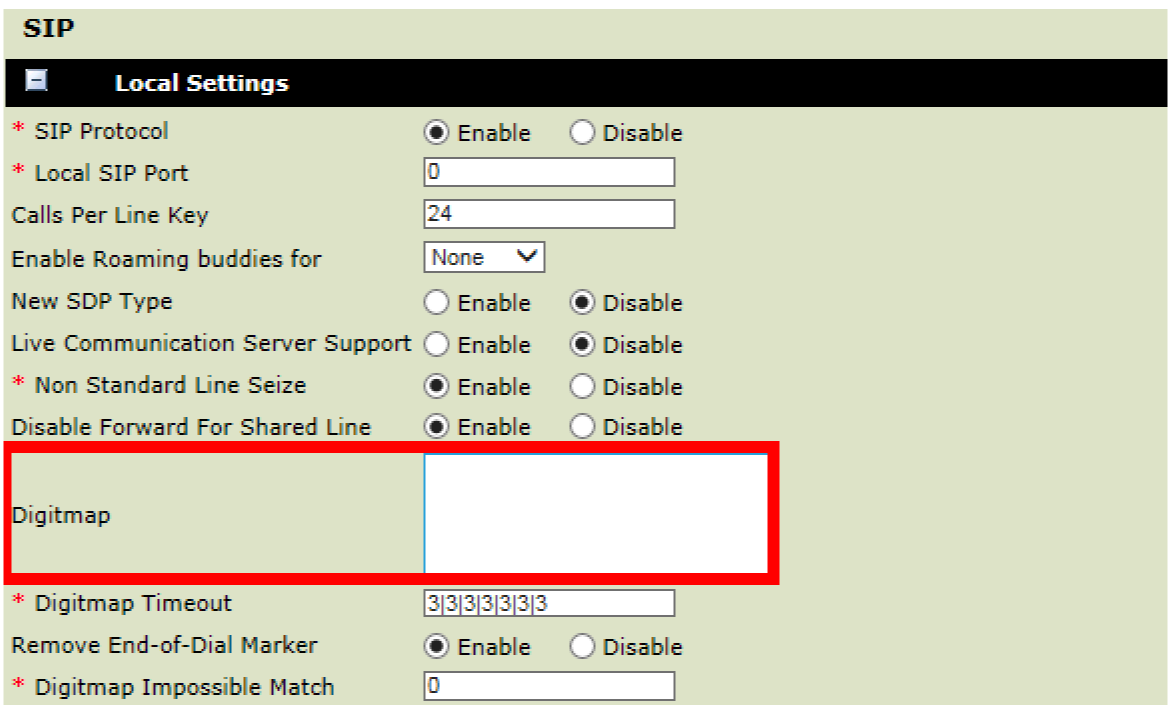
- **User ID**: Fill in the extension's **Registration Name**.
- **Password**: Fill in the extension's **Registration Password**.

d) Expand **SIP Server 1** option, and set as the followings:



- **Special Interop**: Standard
- **Address**: Fill in the domain or IP address of your PBX.
- **Port**: Fill in the same SIP port as the PBX.
- **Transport**: Choose the same transport protocol as the PBX.
- **Register**: Yes

5. Go to **Settings** > **SIP**, set the **Digitmap** to blank. In this way, you can dial any number out.



6. Click **Save**.

If the extension is registered, the **Register Status** will show "Registered".

Register an Extension on Snom Phone

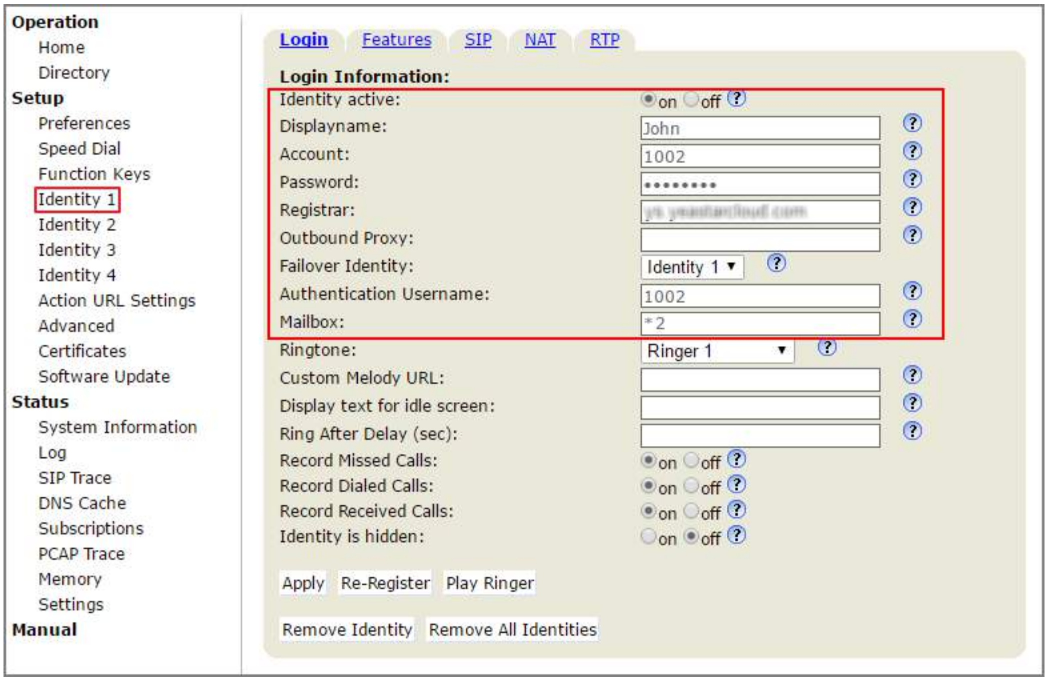
This guide is applicable to the following phones:

- Snom 320, 710, 715, 720, 725, 760, 765

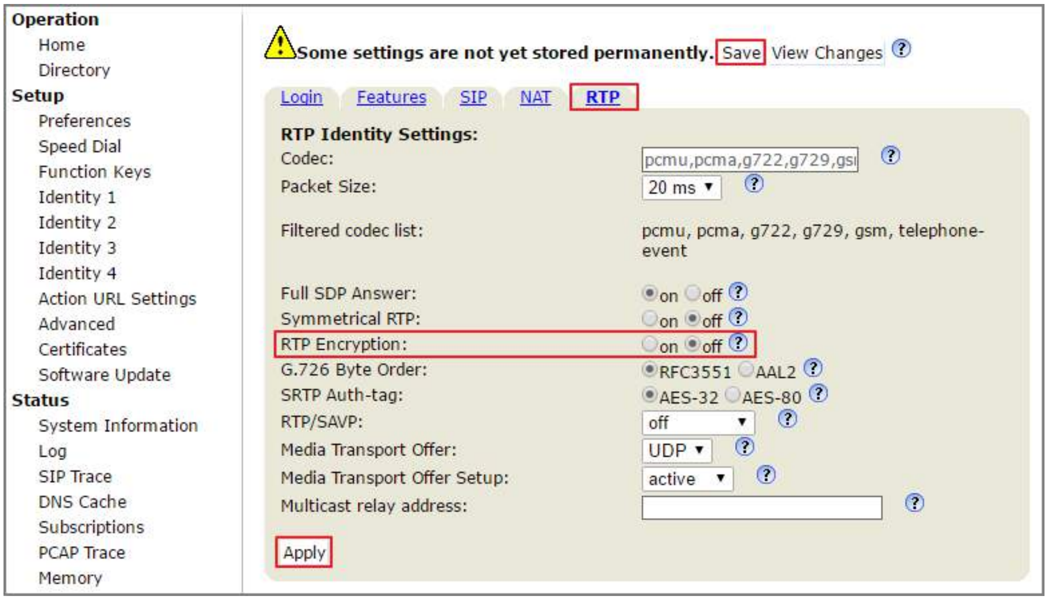
- Snom D Series: 305, 315, 345, 375

Note: For the IP phone with different firmware version, the web GUI may be different.

1. To check the IP address of the phone, press **Settings** > **Information** > **System Info** or press **Menu** > **Information** > **System Info**.
2. Log in the web page of the phone, go to **Setup** > **Identify 1** to configure the account 1.



   - **Identify active**: On
   - **Displayname**: Fill in the name you wish to appear on the phone screen.
   - **Account**: Fill in the extension number.
   - **Password**: Fill in the extension's **Registration Password**.
   - **Registrar**: Fill in the domain or IP address of your PBX.
   - **Authentication Username**: Fill in the extension's **Registration Name**.
   - **Mailbox**: Fill in the feature code of **Check Voicemail** on the PBX. The default code is *2.
3. Click **RTP** tab, set **RTP Encryption** to **Off** if you don't enable SRTP feature for the extension.



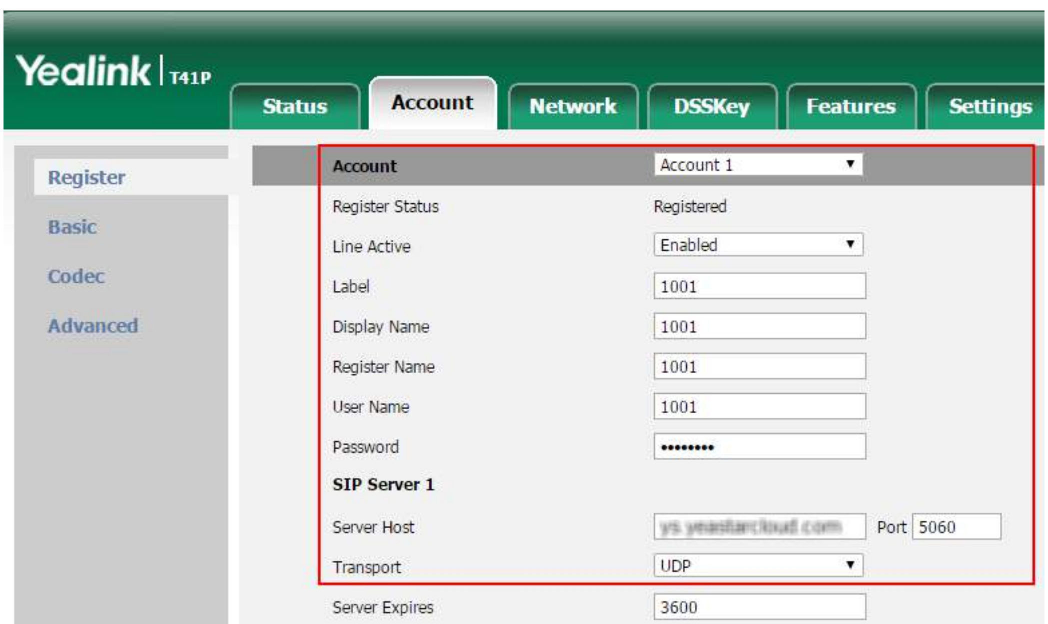
4. Click **Apply**, then click **Save** in the top-right corner.

Register an Extension on Yealink Phone

This guide is applicable to all the Yealink phones.

Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
    - **Username**: admin
    - **Password**: admin
2. Click **Account** tab, and choose one account to configure.

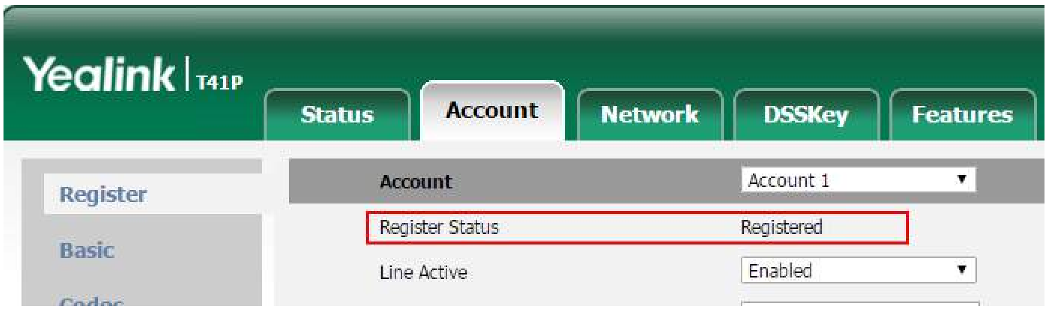


- **Account**: Choose one account.
- **Line Active**: Enabled
- **Label**: Set the name you want to appear on the phone screen.
- **Display Name**: Set the name you want to appear on the other phone's screen when calling out.

- **Register Name**: Fill in the extension's **Register Name**.
- **User Name**: Fill in the extension number.
- **Password**: Fill in the extension's **Registration Password**.
- **Server Host**: Fill in the domain or IP address of your PBX.
- **Port**: Fill in the same SIP port of the PBX.
- **Transport**: Choose the same transport protocol of your PBX.

3. Click **Confirm**.

   If the extension is registered, you can see the **Register Status** shows "Registered".



Register an Extension on Vtech Phone

This guide is based on the Vtech VSP610A v2. 0. 3. 0.

Note: For the IP phone with different firmware version, the web GUI may be different.

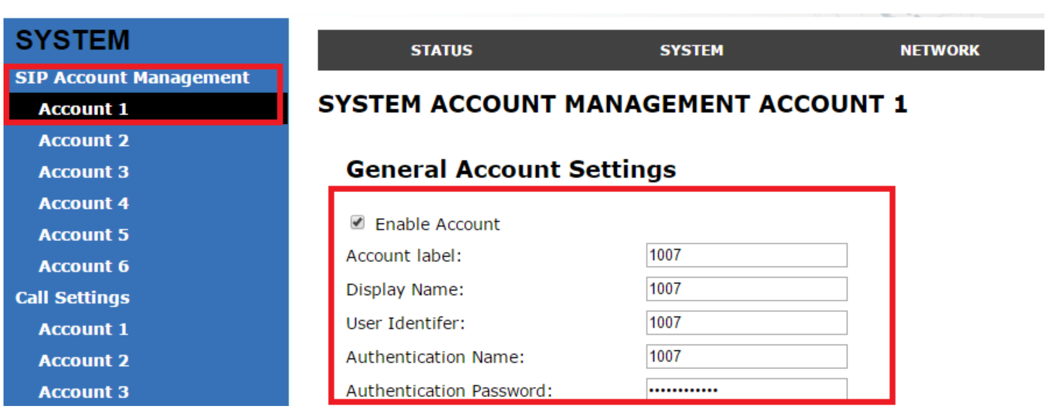
Configure the IP address via phone user interface

1. Press **System** > **Network** > **Basic Settings** > **Dual Mode** > **WAN Setting**.
2. Choose **Static IP** and alter the **IP Address, Subnet Mask**, **Preferred DNS Server**, **Alternate DNS Server**.
3. **Apply** it after input the correct information.
4. **Reboot** the phone and log in the phone web user interface using the new IP address.
5. Enter the user name and password, click **Log In** to enter the web user interface.

   - **User Name**: admin
   - **Default Password**: admin

Account Registration

1. Log in the IP phone, go to **System** > **SIP Account Management**, select one account to configure.
2. Enable the account and fill in the extension information.



- **Enable Register**: check

- **Account Label**: The name you want to display on the phone screen.
- **Display Name**: The name you want to display on another person's phone screen when you are calling the phone.
- **User Identifier**: Enter the extension's **Caller ID**.
- **Authentication Name**: Enter the extension's **Registration Name**.
- **Authentication Password**: Enter the extension's **Registration Password**.

3. In the **SIP Server** section and **Registration** section, fill in your PBX information.

- **SIP Server**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Server Port:** Enter the SIP port of your PBX.
- **Registration**
  - **Server Address:** Enter the domain or IP address of your PBX.
  - **Port:** Enter the SIP port of your PBX.

4. Click **Apply**.

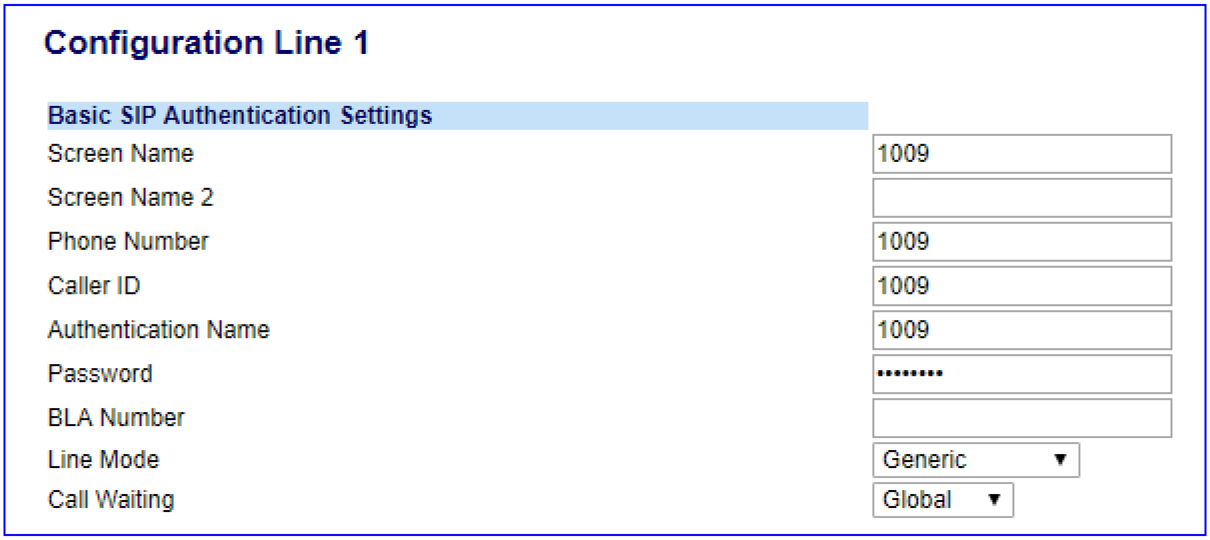
   If the registration is successfully, the register status would show "Registered".

Register an Extension on Mitel Phone

This guide is based on Mitel 6867i v4.1.0.148.

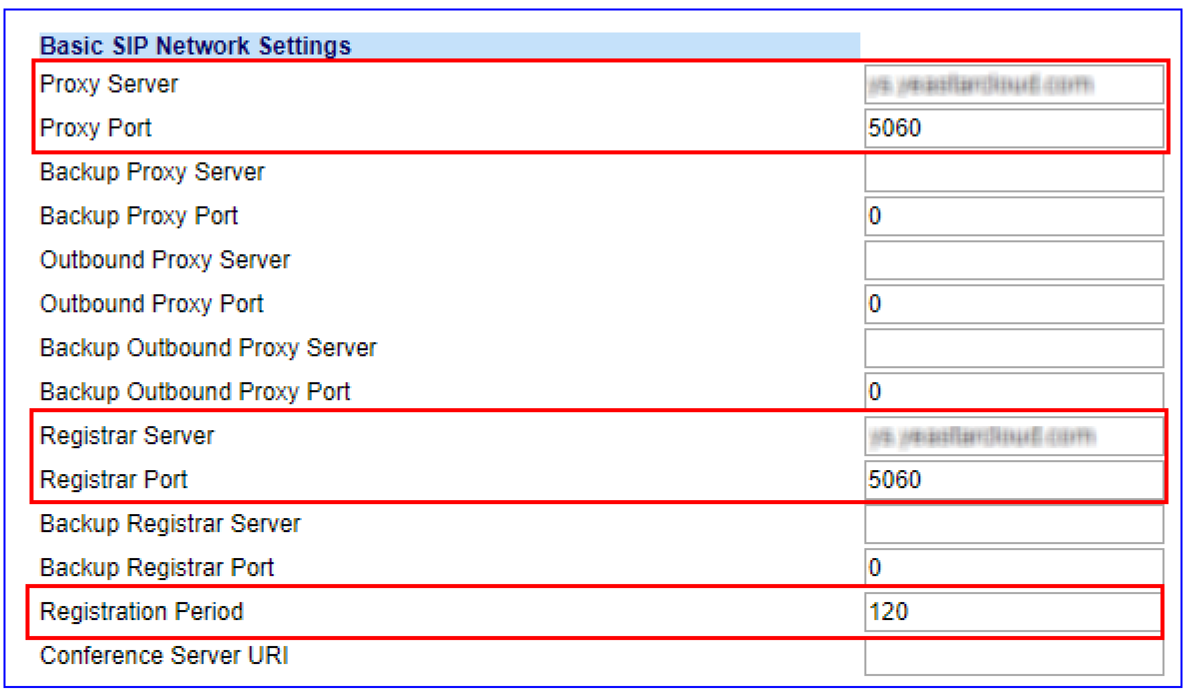
Note: For the IP phone with different firmware version, the web GUI may be different.

1. Log in the web page of the phone.
   - **Username**: admin
   - **Password**: 22222
2. Go to **Advanced** section, choose a line to configure.
   a) In the **Basic SIP Authentication Settings**, enter the extension information.



- **Screen Name**: Set the name that you want to display on the phone screen.
- **Phone Number**: Fill in the extension number.
- **Caller ID**: Fill in the extension's **Caller ID**.
- **Authentication Name**: Fill in the extension's **Registration Name**.
- **Password**: Fill in the extension's **Registration Password**.

b) In the **Basic SIP Network Settings**, enter the PBX information.



- **Proxy Server**: Fill in the domain or IP address of your PBX.
- **Proxy Port**: Fill in the same SIP port of the PBX. The default SIP port on the PBX is 5060.
- **Registrar Server**: Fill in the domain or IP address of your PBX.
- **Registrar Port**: Fill in the same SIP port of the PBX. The default SIP port on the PBX is 5060.
- **Registration Period**: Set the registration period according to the settings on your PBX. The default range of SIP registration time on the PBX is 60-3600 seconds.

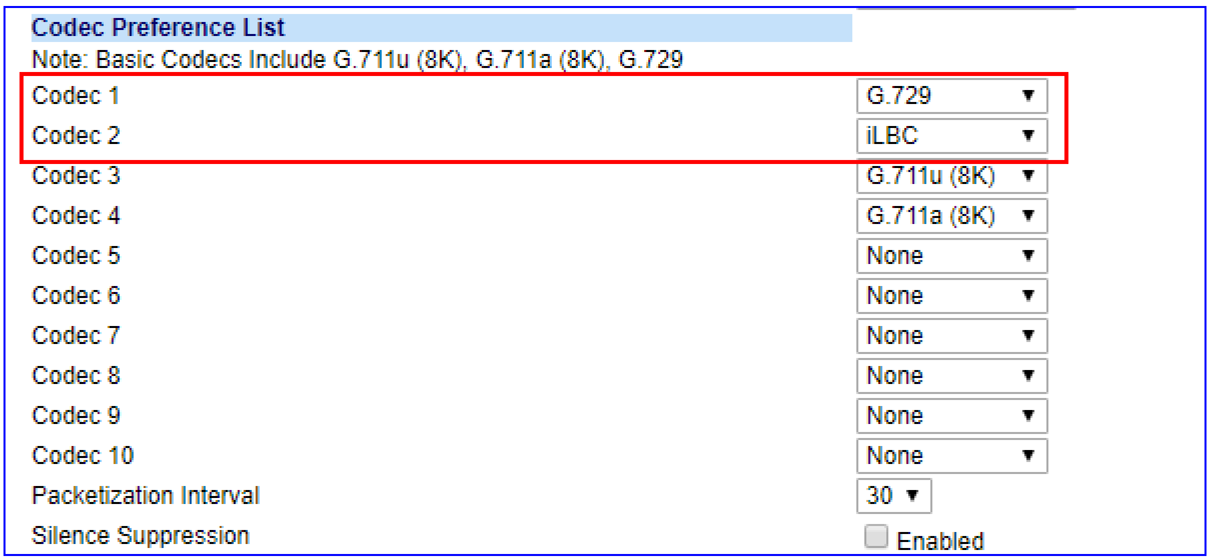
3. Click **Save Settings**.
4. Go to **Advanced** > **Global SIP**, set the RTP settings and codec preferences.

a) In the **RTP Settings** section, configure the RTP according to the settings on your PBX.

- **Force RFC2833 Out-of-Band DTMF**: Enabled
- **DTMF Method**: RTP
- **RTP Encryption**: If you don't enable SRTP for the extension, choose **SRTP Disabled**.

b) In the **Codec Preference List** section, set the codec preferences according your PBX settings.

Note: G729 and iLBC are the default enabled codecs on the PBX, you should enable the G729 codec or the iLBC codec on your phone.



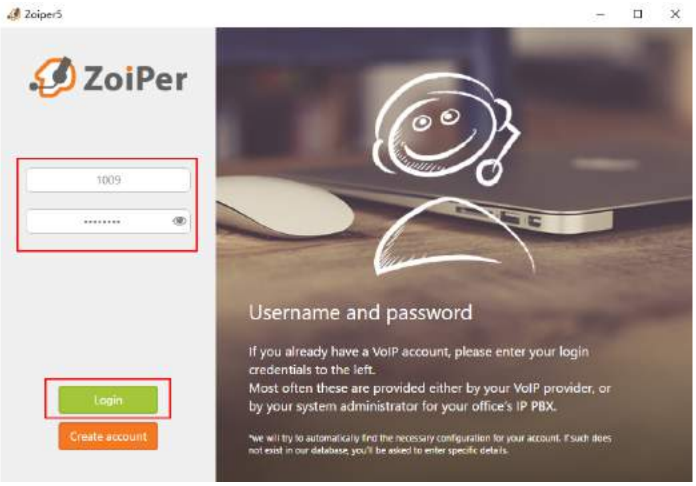
5. Click **Save Settings**.
6. Reboot the phone to make the configuration take effect.

   You can check the extension status via **Status** > **System Information**. If the extension is registered, the status shows "Registered".

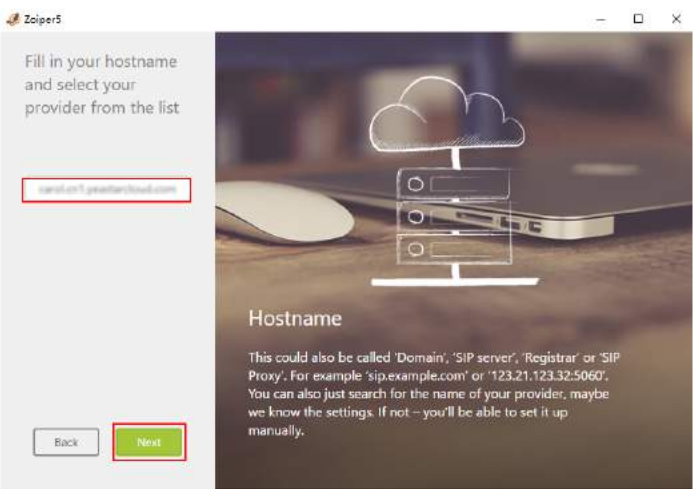
Register an Extension on Zoiper

This guide is based on Zoiper PC client v5.2.12.

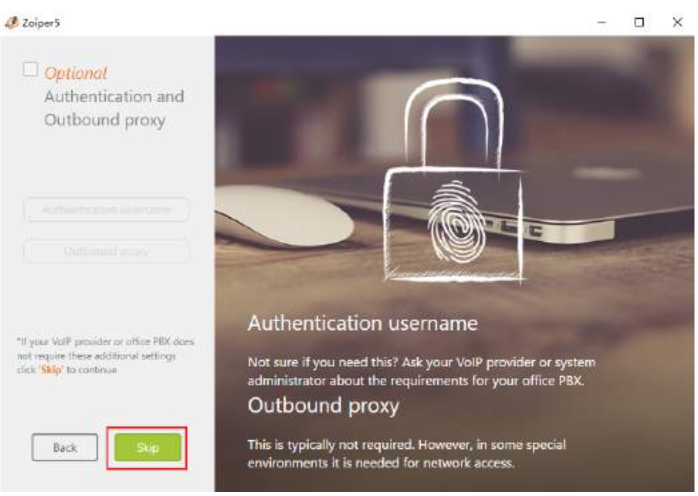
1. Launch Zoiper PC client, enter the extension number and the extension's **Registration Password**, then click **Login**.



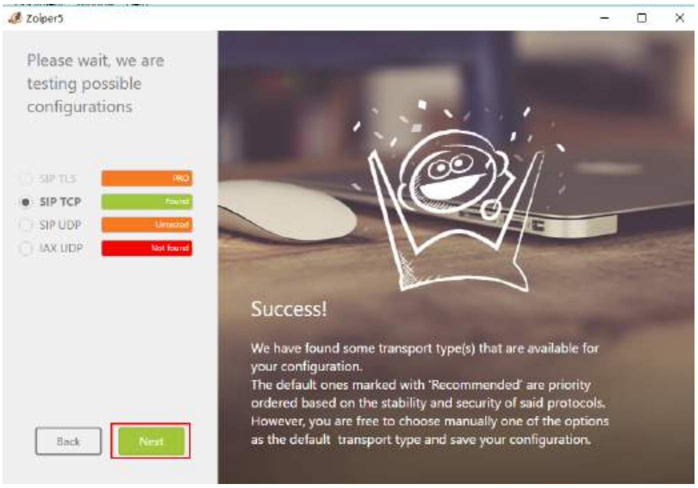
2. Enter the domain or IP address of your PBX, click **Next**.



3. Click **Skip**.

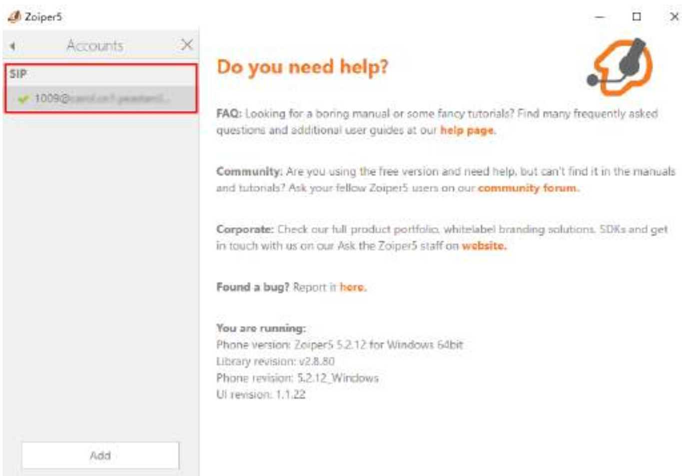


4. Click **Next**.



5. Check the account status.

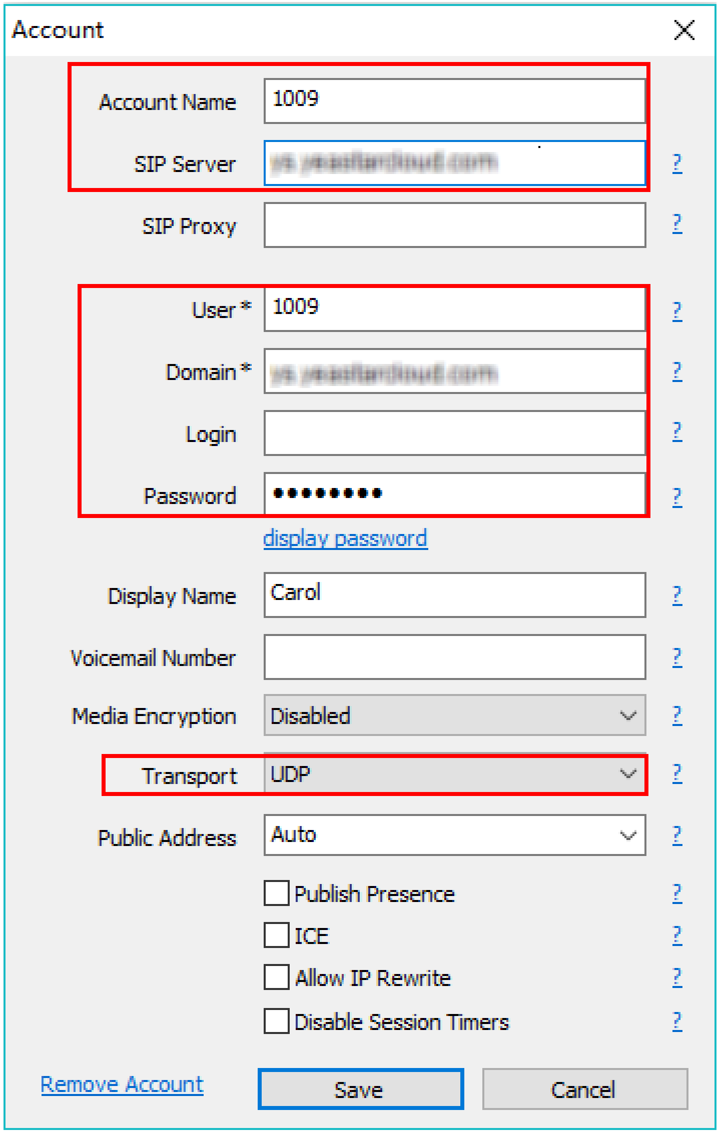
   If the extension is registered, you can see the status shows as the following figure.



Register an Extension on MicroSIP

This guide is based on the MicroSIP v3.17.3.

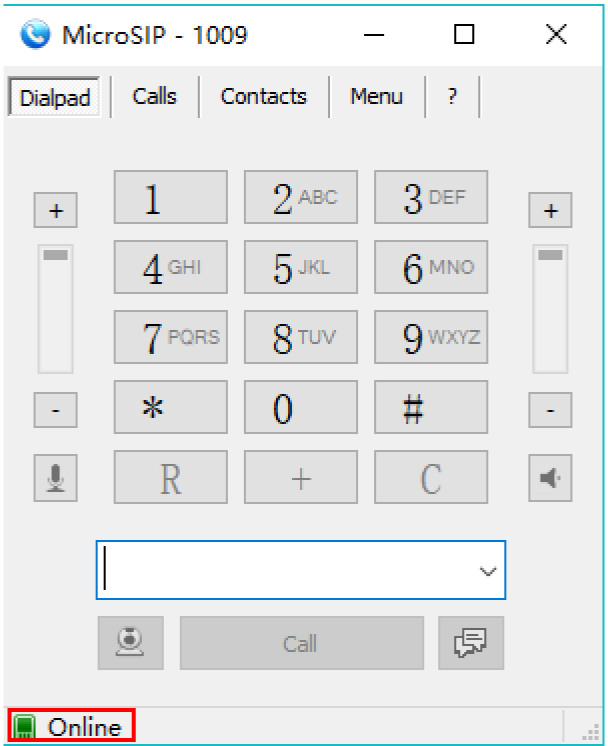
1. Launch MicroSIP, go to **Menu** > **Add Account**, configure the account settings.



- **Account Name**: Set the name that you want to appear on the soft phone screen.
- **SIP Server**: Enter the domain or IP address of your PBX.
- **User**: Enter the extension number.
- **Domain**: Enter the domain or IP address of your PBX.
- **Password**: Enter the extension's **Registration Password**.
- **Display Name**: Set the name you want to appear on the other phone's screen when calling out.
- **Transport**: Choose the same protocol of the PBX. The default protocol on PBX is UDP.

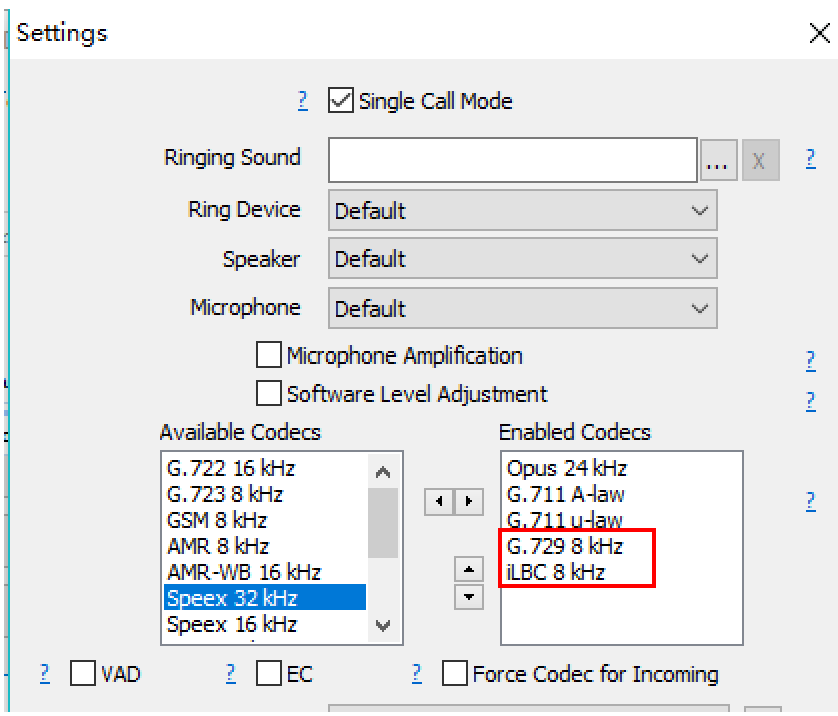
2. Click **Save**.

If the extension is registered, you can see the status shows as below.



3. Go to **Menu** > **Settings**, enable G729 and iLBC codecs.

Note: G729 and iLBC are the default enabled codecs on the PBX. To ensure the call is normal, you need to enable the G729 or iLBC codec on the soft phone.

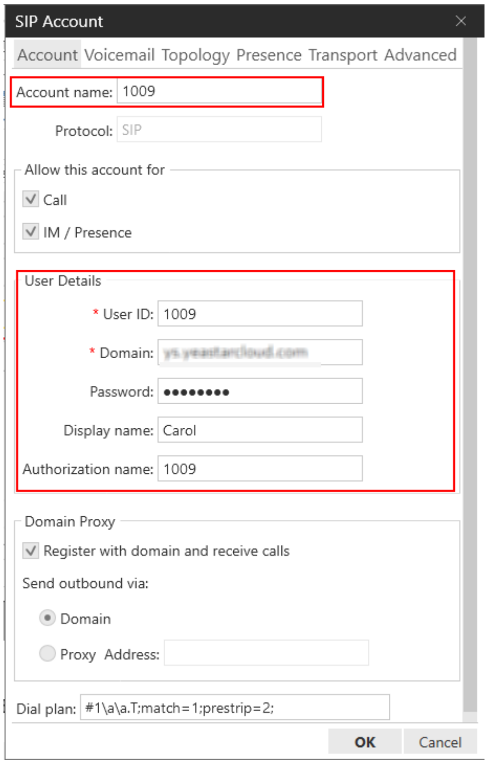


4. Click **Save**.

Register an Extension on X-Lite

This guide is based on X-Lite PC client v5.2.0.

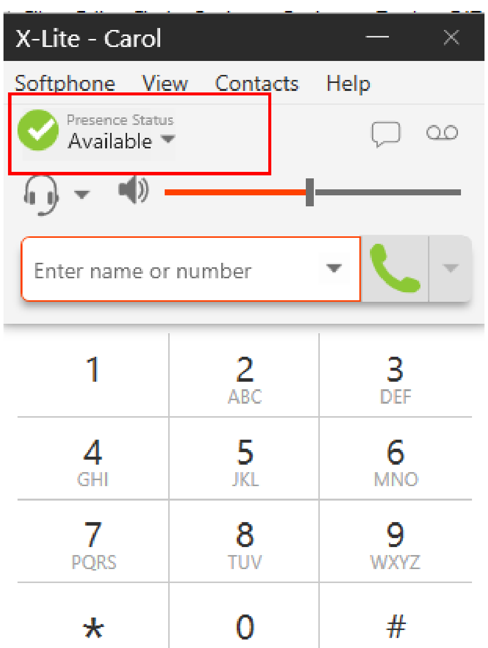
1. Launch X-Lite, go to **Softphone** > **Account Settings**, configure the SIP account.



- **Account name**: Set a name for the account.
- **User ID**: Enter the extension number.
- **Domain**: Enter the domain or IP address of your PBX.
- **Password**: Enter the extension's **Registration Password**.
- **Display name**: Set the name that you want to appear on the soft phone screen.
- **Authorization name**: Enter the extension's **Registration Name**.

2. Click **OK**.

   If the extension is registered, you can see the status shows as below.



Call Routes

To control inbound calls and outbound calls, you need to set up your trunks, inbound routes, and outbound routes.

Manage Outbound Calls

To make outbound calls, you need to set up outbound routes on the PBX.

What is an Outbound Route?

An outbound route works like a traffic cop giving directions to road users to use a predefined route to reach a predefined destination.

Outbound routing is a set of rules that the PBX uses to decide which trunk to use for an outbound call. Having multiple trunks allows you to control cost by routing calls over the least costly trunk for a particular call. Outbound routes are used to specify what numbers are allowed to go out a particular route.

How does an outbound route work?

Every time you dial a number, PBX will do the following in strict order:

- Examine the number you dialed.
- Compare the number with the pattern that you have defined in your route 1 and if matches, it will initiate the call using that trunk. If the route 1 does not match, it will compare the number with the pattern you have defined with route 2 and so on.
- Pass the number to the appropriate trunk to make the call.

Add an Outbound Route

The PBX has a default outbound route with dial pattern `X.` that allows you to dial any outgoing numbers. You can delete the default outbound route, then add a new one to configure settings according to your needs.

1. Go to **Settings** > **PBX** > **Call Control** > **Outbound Routes**, click **Add**.
2. Set the **Name** to specify the route name. The name is usually descriptive, i.e. "local" for local calls, "international" for international calls.
3. Set a dial pattern of the outbound route.
   - **Pattern**

     A dial pattern is a unique set of digits that will select this route and send the call to the designated trunks.

Pattern	**Description**
`X`	Refers to any digit between 0 and 9.
`N`	Refers to any digit between 2 and 9.
[###]	Refers to any digit in the brackets, example [123] is 1 OR 2 OR 3. Ranges of numbers can be specified with a dash ([136-8]) would match the numbers 1,3,6,7 and 8.
`.`	Wildcard `.` matches one or more characters.
`!`	Wildcard `!` is used to initiate call processing as soon as it can be determined that no other matches are possible.
`Z`	Refers to any digit between 1 and 9.

- **Strip** (Optional setting)

  This setting defines how many digits will be stripped from the front of the dialed number before the call is placed.

  Example: If you set **Pattern** as `9.` and set **Strip** as `1`. If a user wants to call number 1588902923, he/she should dial 91588902923. The PBX will strip one digit from the dialed number, and call the number 1588902923.
- **Prepend** (Optional setting)

  The prepend will be added to the beginning of a successful match. If the dialed number matches the **Pattern**, then this will be prepended to the number before placing the call.

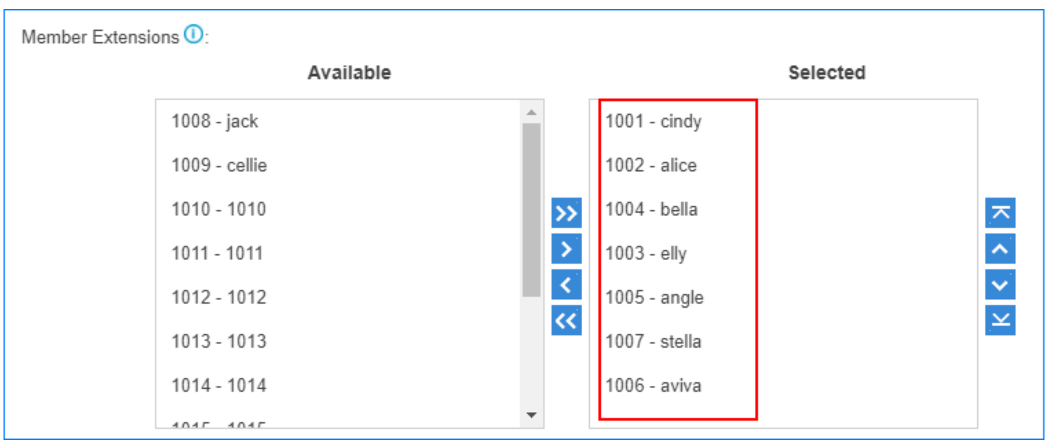
  Example: If a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, this field could be used to prepend a 3-digit area code to all 7-digit phone numbers before the calls are placed.

4. Click [+] to add a dial pattern for the outbound route.
5. In the **Member Trunks** field, select the desired trunk from **Available** box to the **Selected** box.

   The users can only use the selected trunk to make outbound calls through this route.

6. In the **Member Extensions** field, select the desired extensions from **Available** box to the **Selected** box.

   Only the selected extension users can make outbound calls through this route.



7. Optional: Set the other settings.

- **Password**: If password is set, when users try to make outbound calls through this route, they will be asked to enter a password.
- **Rrmemory Hunt**: If this function is enabled, the PBX will remember which trunk was used last time, and then use the next available trunk to call out.
- **Time Condition**: By default, users can call out through the outbound route at any time. You can define when the outbound route is available.

8. Click **Save** and **Apply**.

Note: After you finish the outbound route configurations, you need to check the priority of your outbound routes and adjust the priority if needed.

Related tasks

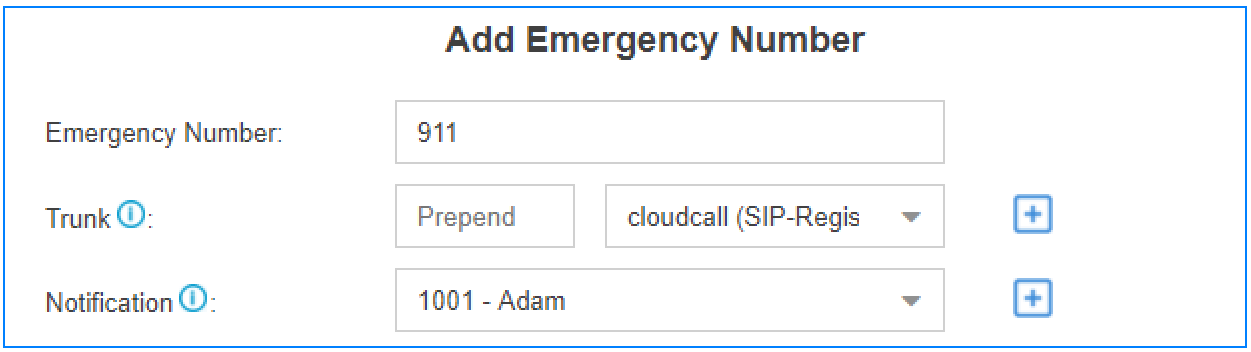
Adjust Priority of Outbound Routes on page 82

Add Emergency Numbers

Emergency calls have the highest priority. Extension users can make emergency calls at any time.

Note: If the trunks used to make emergency calls are busy, the PBX will terminate the ongoing call, then place the emergency call.

1. Go to **Settings** > **PBX** > **Emergency Number**, click **Add**.
2. Set the **Emergency Number**.
3. Set the **Trunk**.



   a) Select a trunk to make the emergency call.
   b) Optional: If your trunk needs a prepended number before the emergency number, you need to set the **Prepend**.

      For example, if your trunk needs a prepended number 0 before the emergency number 911. The users should dial 0911 to make the emergency call. To comply with the user's dialing habit, you can set the **Prepend** as `0`; in this way, the users can dial 911 as they usually do.
   c) Optional: Click to add a new trunk. If the first trunk cannot work properly, the PBX will use the second trunk to make calls.
4. Set the notification contacts.

   If someone makes emergency calls through the PBX, the contacts will receive notification calls on their extensions.
   a) Select an extension from the **Notification** drop-down menu.
   b) Optional: Click to add another contact.
5. Click **Save** and **Apply**.

Outbound Route Examples

Below are sample configurations that will help you understand outbound route dial patterns.

Route Name: Domestic

Local numbers in XiaMen, China are all 7-digit numbers and the numbers do not start with 0, such as 5503305.

For long-distance calls, you need to dial the 4-digit Area Code then local numbers, such as 0595-87588666. The Area Code in China is in the format 0ZXX, the first digit is 0, and the second digit cannot be 0.

Pattern	Strip	Prepend	Description
`90ZXX.`	1		This is for long-distance call, dial 9 before the long-distance numbers. ***Example:*** To call number 059587588123, the user should dial 9059587588123.
`9ZXXXXXX`	1		Users will dial XiaMen local numbers as 9 + local numbers, PBX will strip the leading 9 before the call is placed. ***Example:*** To call number 5503301, the user should dial 95503301.

Route Name: Mobile

All the mobile phone numbers in China are 11-digit numbers and start with digit 1, such as 15880260666.

Pattern	Strip	Prepend	Description
`1XXXXXXXXXX`			The users can dial the mobile numbers as they usually do. ***Example:*** To call number 15880260666, dial 15880260666.

Route Name: International_Call

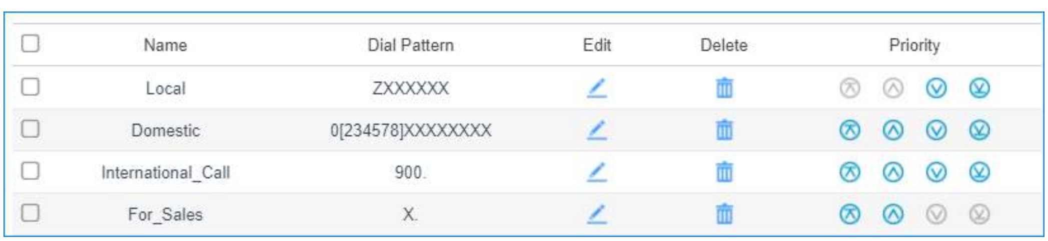
All the international numbers start with digits 00.

Pattern	Strip	Prepend	Description
`00.`			Numbers start with digits 00 will go through this outbound route.

Adjust Priority of Outbound Routes

The PBX will search for a matching dial pattern from the top route. If a route with higher priority is matched, the PBX will not continue going down the route list to search a "better" route. Therefore, the route priority is important, especially if there is some overlap. For example, the number 5503305 will match both a dial pattern of `ZXXXXXX` and `X.`.

1. Go to **Settings** > **PBX** > **Call Control** > **Outbound Routes**.
2. Click the buttons to adjust the priority of your outbound routes.



Name	Dial Pattern	Edit	Delete	Priority
Local	ZXXXXXX			
Domestic	0[234578]XXXXXXXX			
International_Call	900.			
For_Sales	X.			

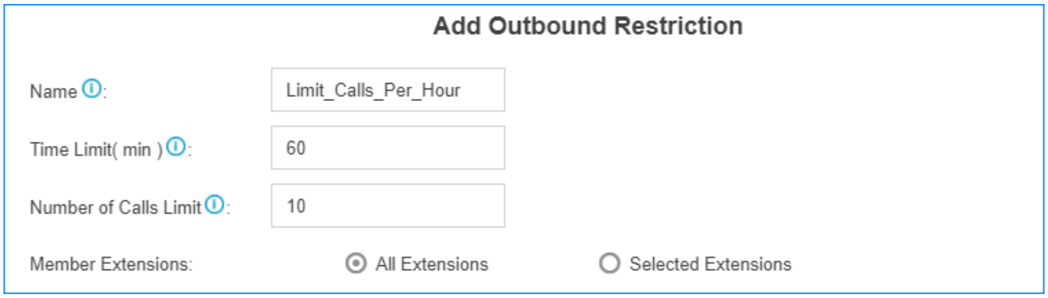
Restrict Outbound Calls

The PBX has a default rule to limit how many times a user can make outbound calls during a certain time period. The default rule limits users to make maximum 5 outbound calls in 1 minute. If a user makes outbound calls over the limit, the extension will be locked and prohibited from making outbound calls.

You can edit the default rule or add new rules.

1. Go to **Settings** > **PBX** > **Call Control** > **Outbound Restriction**, click **Add**.
2. On the configuration page, set the rule name.
3. Set the **Time Limit** and **Number of Calls Limit**.

   This will limit how many times a user can make outbound calls during the configured time period.

4. Select **All Extensions** to apply the rule to all the extensions, or select **Selected Extensions**, and select the desired extensions to apply the rule the users.
5. Click **Save** and **Apply**.

Manage Inbound Calls

What is an Inbound Route?

An inbound route is used to tell the PBX where to route inbound calls based on the phone number or DID dialed. Inbound routes are often used in conjunction with time conditions and an IVR.

DID Routing & Caller ID Routing

The PBX allows two specific types of inbound routing: DID Routing and Caller ID Routing. You can set both DID routing and Caller ID routing for an inbound route, or set one of the routing types.

If you don't specify which DID numbers and Caller ID numbers on the inbound route, the inbound route will matches all calls.

Add an Inbound Route

The PBX has a default inbound route. When users call to the selected trunk, the PBX will route the calls to an IVR. You can delete the default inbound route, then add a new one to configure settings according to your needs.

1. Go to **Settings** > **PBX** > **Call Control** > **Inbound Routes**, click **Add**.
2. Set the **Name** to specify the route name.
3. Optional: Set **DID Pattern** to route calls based on DID numbers.

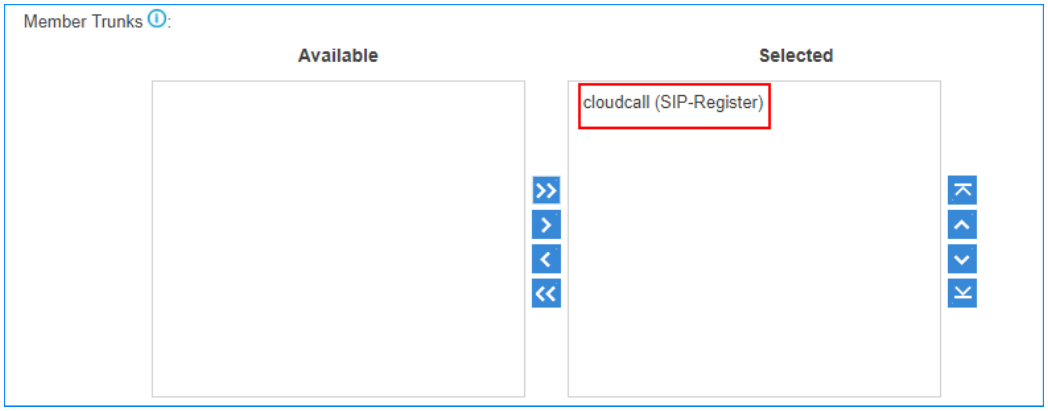
   The PBX will route the call only when the caller dials the matched numbers.

   Note: Leave this blank to match calls with any or no DID info.
4. Optional: Set **Caller ID Pattern** to route calls based on caller IDs.

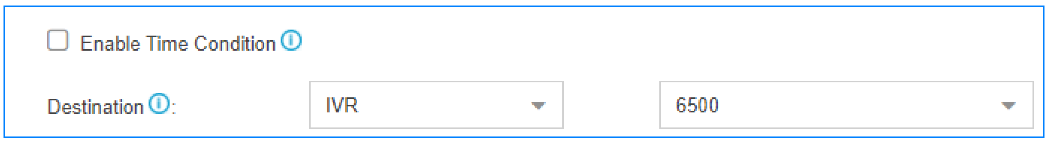
   The PBX will route the call only when the caller ID number of the caller matches the **Caller ID Pattern**.

   Note: Leave this blank to match calls with any or no caller ID info.
5. In the **Member Trunks** field, select the desired trunk from **Available** box to the **Selected** box.

   The PBX will route the incoming calls when the caller calls the number of the selected trunk.

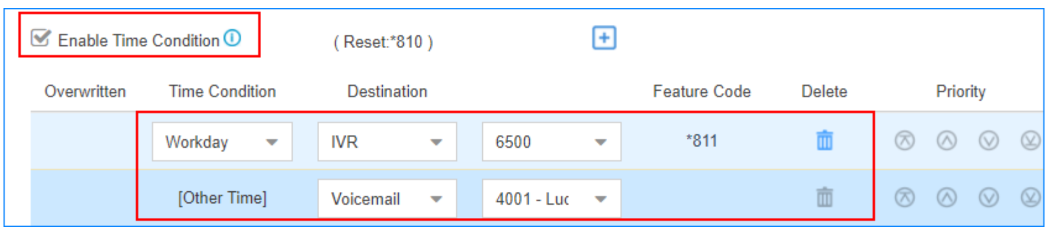


6. If you will allow the incoming calls being routed to a desired destination without time limit:



a) Uncheck **Enable Time Condition**.
b) Set the **Destination**.

7. If you will route incoming calls to different destinations based on time condition:



a) Check **Enable Time Condition**.
b) Set the destination for **Other Time**.

   If a call comes outside the time condition, PBX will route the call to the selected destination.

c) Click, select your time condition and the destination.

   If a call comes in the time condition, PBX will route the call to the selected destination.

d) Click to set another destination.

8. Optional: Set other settings.
   - **Distinctive Ringtone**: This feature needs support from the phones. Distinctive ring tones help users recognize where the call is from.
   - **Enable Fax Detection**: If fax detection is enabled, the PBX will send fax to the fax destination if a fax tone is detected.
9. Click **Save** and **Apply**.

Inbound Route Examples

Below are sample configurations that will help you understand inbound route DID setting and Caller ID setting.

Note: The following examples ignore time condition, you can set time condition according to your needs.

Inbound Route without Limit

Any user call in the selected trunk will be routed to the inbound route destination.

- **Name**: Set a name according to your preference.
- **Member Trunk**: Choose desired trunk(s).
- **Destination**: Set the destination.

  Leave all other fields blank.

Inbound Route Based on DID Numbers

If a trunk has multiple DID numbers, you can add multiple inbound routes that based on different DID numbers. When users dial different DID numbers, they will be routed to different destinations.

The following example shows inbound route based on DID number 5503301.

- **Name**: Set a name according to your preference. For DID routes, you can set the name as the DID number, which help you identify the route.
- **DID Pattern**: 5503301
- **Member Trunk**: Choose the trunk that has the DID number.
- **Destination**: Set the destination.

  Leave all other fields blank.

Inbound Route Based on DID Number Range

If a trunk has multiple DID numbers that are in a range, you can quickly set the DID number range in an inbound route to route calls to different destinations based on the DID numbers.

The following example shows inbound route based on DID range 5503301-5503305, that will route calls to extension 1001-1005.

- **Name**: Set a name according to your preference.
- **DID Pattern**: 5503301-5503305
- **Member Trunk**: Choose the trunk that has the DID number.
- **Destination**: Choose Extension Group, and enter the extension range 1001-1005.

  Leave all other fields blank.

Inbound Route Based on Caller ID

The PBX will route calls only when the users' caller ID numbers match the Caller ID Pattern.

The following example inbound route will route caller ID numbers that start with digit 1 to the destination. If a user's number, for example, 532352584, that doesn't start with digit 1 could not call in this inbound route.

- **Name**: Set a name according to your preference.
- **Caller ID Pattern**: `1.`
- **Member Trunk**: Choose desired trunk(s).
- **Destination**: Choose a destination.

  Leave all other fields blank.

Inbound Route Base on Caller ID and DID Numbers

The inbound calls should match both caller ID numbers and DID numbers, or the call will fail.

The following example inbound route allows the numbers that start with digit 1 to call in; the users should dial the number 5503301.

- **Name**: Set a name according to your preference.
- **Caller ID Pattern**: `1.`
- **DID Pattern**: 5503301
- **Member Trunk**: Choose desired trunk(s).

- **Destination**: Choose a destination.

  Leave all other fields blank.

Distinguish Inbound Calls

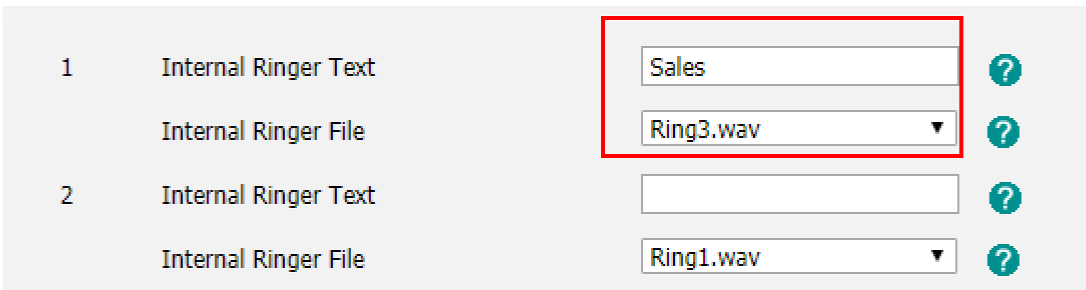
You can set distinctive ring tone, distinctive caller ID, or DNIS names to distinguish where the calls come from.

Distinguish Inbound Calls by Ring Tones

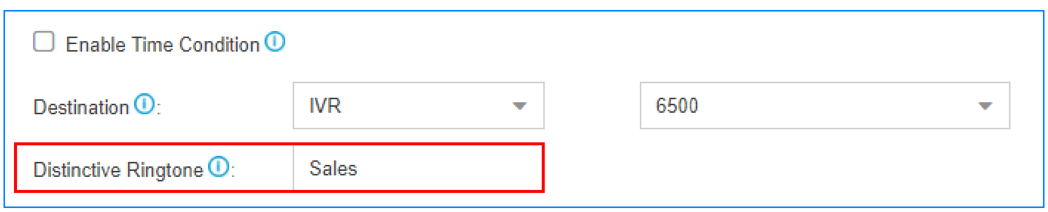
Distinctive ring tone distinguishes calls from different inbound routes. You can set distinctive ring tone on different inbound routes. When a user hears the ring tone of the incoming call, he/she may notice the intention of the incoming call.

Note: This feature needs support from the IP phones. We take Yealink phone as an example.

1. Log in the phone web page, go to **Settings** > **Ring**, select a ringtone and set the name.



2. Click **Confirm** to save the settings.
3. Log in PBX web page, go to **Settings** > **PBX** > **Call Control** > **Inbound Routes**, select an inbound route to edit.
4. Enter the ringtone name in the **Distinctive Ringtone** field.



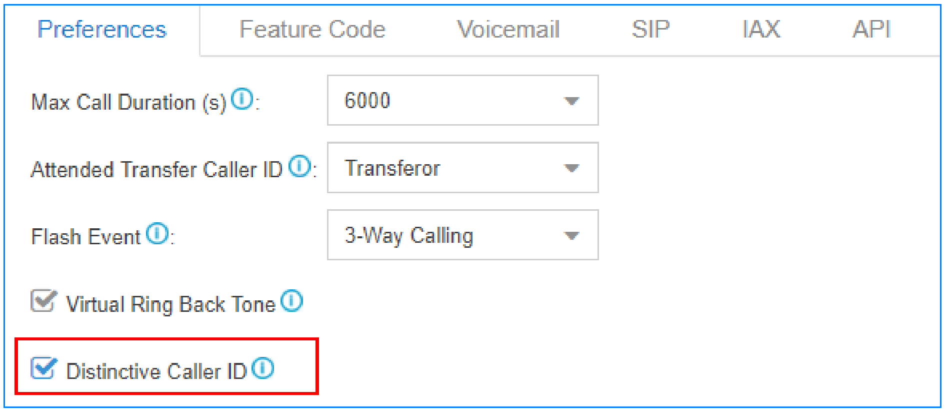
5. Click **Save** and **Apply**.

When a call comes through the inbound route, the phone will play the `Ring3.wav` file.

Distinguish Inbound Calls by Caller ID

When incoming calls are routed from a ring group, a queue or an IVR, the PBX can display the name of ring group/ queue/IVR. When the extension user receive a call from the ring group/queue/IVR, he/she may notice the intention of the incoming call.

1. Go to **PBX** > **General** > **Preference** , check

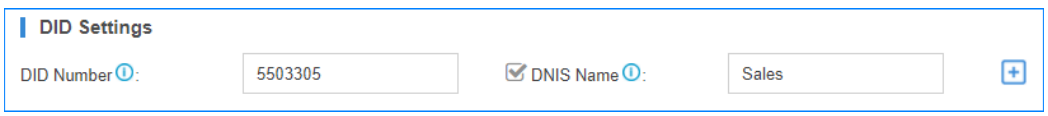


2. Click **Save** and **Apply**.

Distinguish Inbound Calls by DNIS Name

DNIS (Dialed Number Identification Service) is a service sold by the trunk service provider to let users decide which trunk number was dialed by a customer. This is useful to decide how to answer an inbound call.

1. Go to **Settings** > **PBX** > **Trunks**, select a trunk to edit.
2. On the trunk edit page, click **Advanced** tab.
3. Enter the DID number of the trunk, and set the **DNIS Name** for the DID number.



4. If the trunk has another DID number, click to add a DNIS name.
5. Click **Save** and **Apply**.

Time Conditions

Time conditions are used to control call flow based upon time and date.

A Time Condition contains a time group. You can enable Time Conditions on an inbound route. When a call arrives at the Time Condition destination, the PBX will check the current system time and date against the time groups in the Time Condition.

You can also apply Time Conditions to an outbound route to limit when the users can use the outbound route.

Example

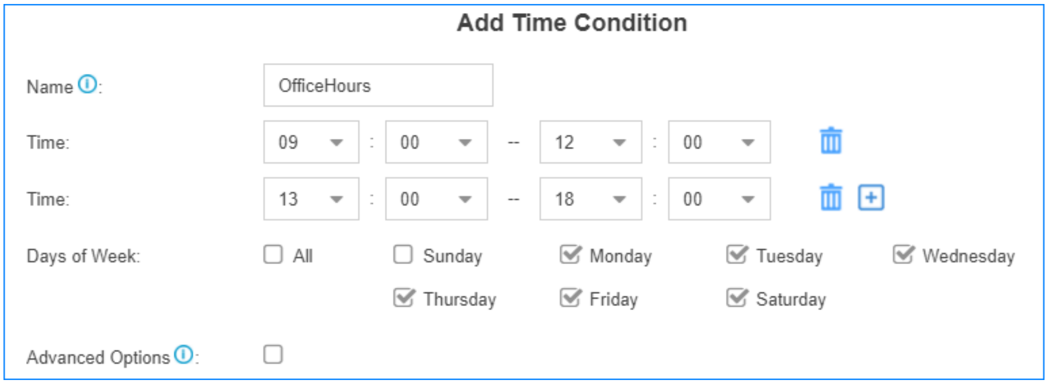
1. Create a Time Condition called "LunchBreak" that starts at 12:00 p.m and end at 1:00 p.m. In your inbound route, set the Time Condition destination to voicemail.
2. Create a Time Condition called "Business Hours" that defines your normal office hours. Apply the Time Condition to your inbound route. You can also apply the Time Condition in your outbound route to prevent employees from making outbound calls outside the office hours.

Add a Time Condition

A Time Condition defines periods of time that can be applied to outbound routes or inbound routes.

1. Go to **Settings** > **PBX** > **Call Control** > **Time Conditions**, click **Add**.
2. On the configuration page, set the **Name**.
3. Define the time period for the Time Condition.

4. Click to add another time period.
5. Choose days of week.



6. If you want apply the time period(s) to specific dates, check **Advanced Options**, and set the month and the days of month.

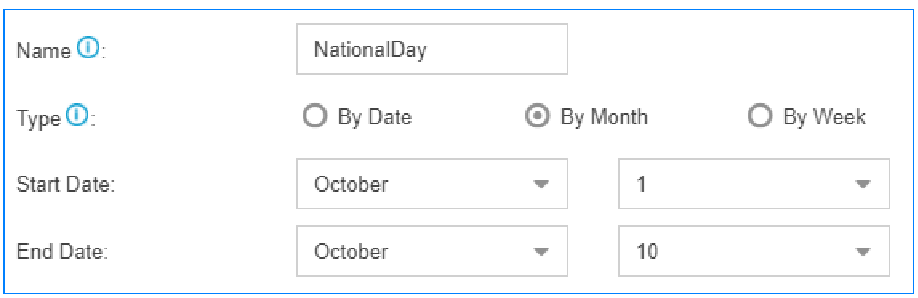
   Note: Advanced Options are disabled by default that means that the time period(s) will apply to every day of the month, every month of the year.

7. Click **Save** and **Apply**.

Add a Holiday

You can add a group of holidays and set a Time Condition destination for the holidays on your inbound route. When a customer calls to your company during holidays, the PBX will route the call to the pre-configured destination.

1. Go to **Settings** > **PBX** > **Call Control** > **Time Conditions** > **Holiday**, click **Add**.
2. On the configuration page, set the **Name** of the holiday.
3. Choose the **Type**.

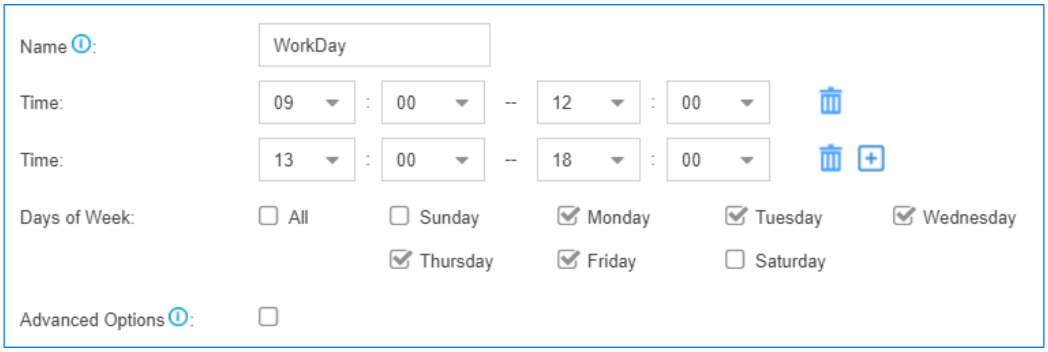


- **By Date**: If the holiday such as Chinese Spring Festival varies very year, choose this type.
- **By Month**: If the holiday such Chinese National Day always fall on the same calendar date, choose this type.
- **By Week**: If the holiday such as Thanks Giving Day always fall on the same week, choose this type.

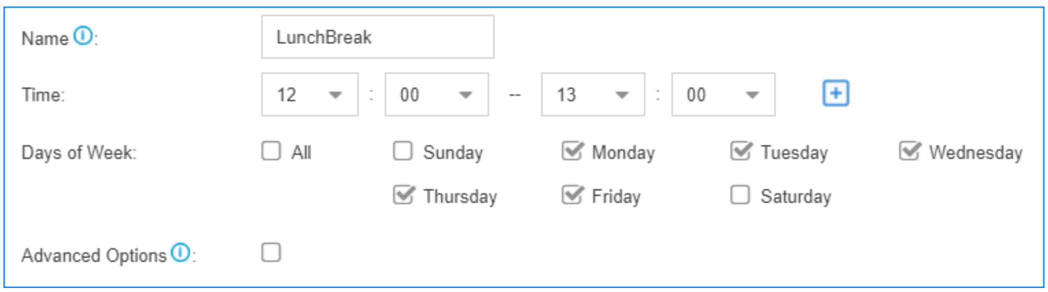
4. Set the date of the holiday.
5. Click **Save** and **Apply**.

Time Condition Examples

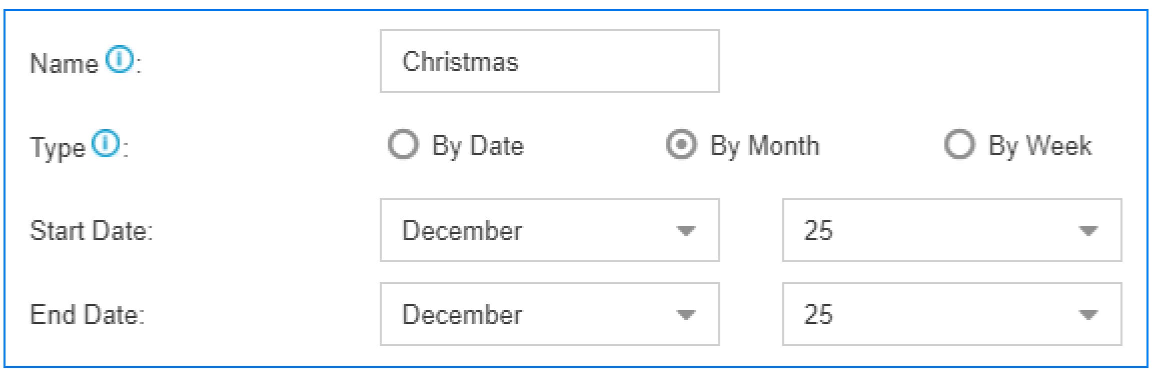
Time Condition: Work Day



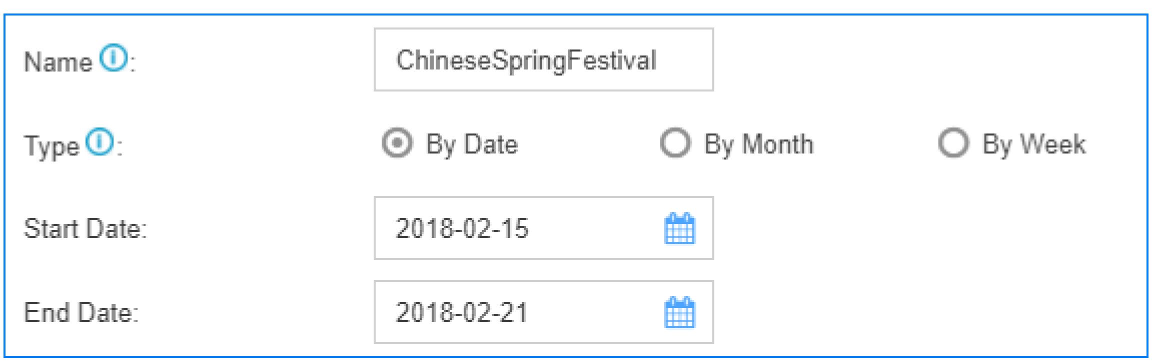
Time Condition: Lunch Break



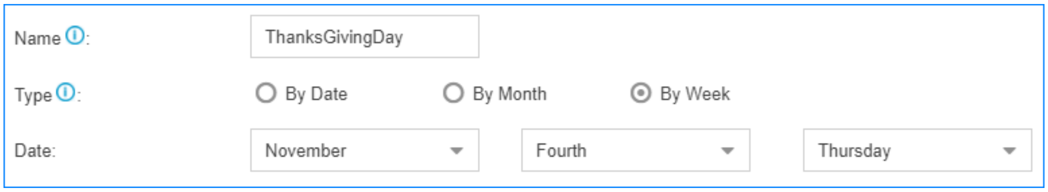
Holiday: Christmas



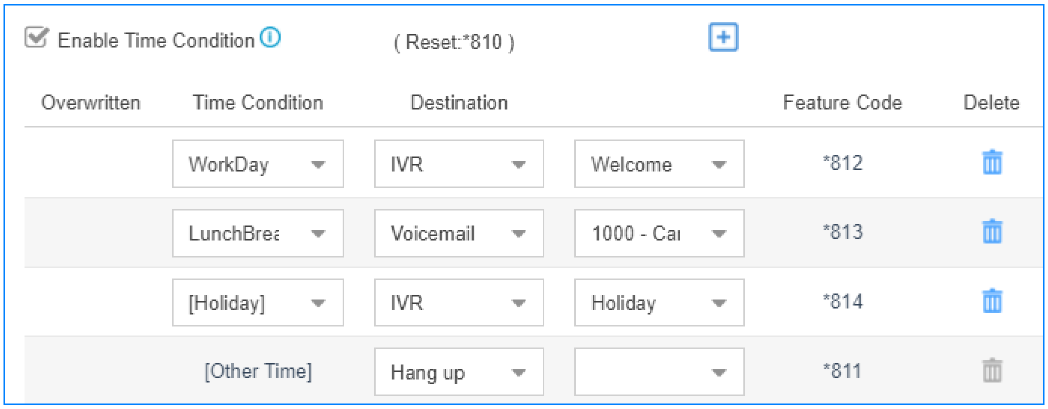
Holiday: Chinese Spring Festival



Holiday: Thanks Giving Day



Apply Time Conditions on Inbound Route



Time Condition Override

The Time Condition Override function is used to switch the incoming call routing against the Time Condition.

For example, during office hours, incoming calls go to ring group; after office hours, incoming calls go to voicemail. Users can override the time condition to ring group if they are in the office after office hours.

Voice Prompts

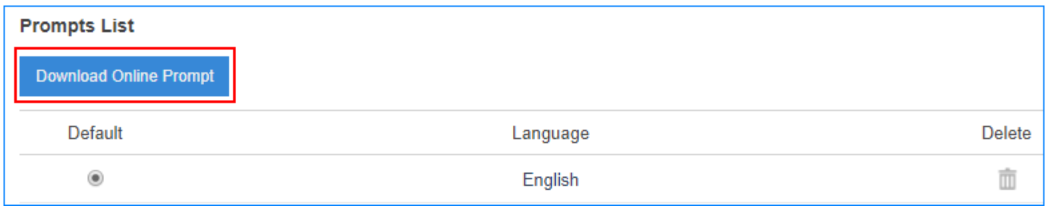
Change System Prompt

The default system prompt language is English. You can change the global system prompt, and if an extension user works in a foreign language, you can set a different system prompt for the user.

Change to an Online System Prompt

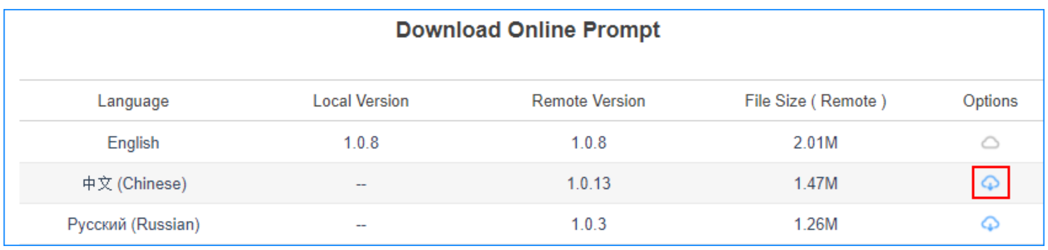
Yeastar have stored all the supported system prompts online. You can check the supported system prompts on the PBX web page, and download an online system prompt file, then change to the desired system prompt.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **System Prompt**.
2. Click **Download Online Prompt**.



3. On the **Download Online Prompt** page, select your desired system prompt, click to download the file.

   After the file is downloaded, you can see the system prompt in **Prompt List**.



4. Set the downloaded system prompt as the default system prompt.

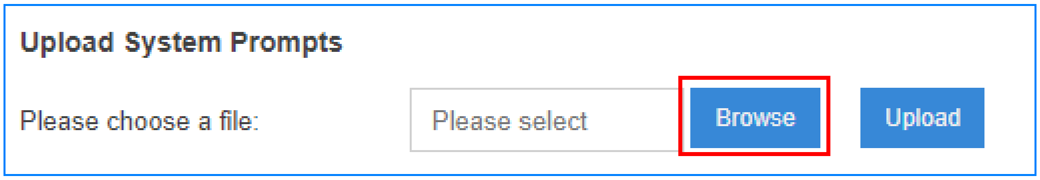
5. Click **Save** and **Apply**.

Change to a Custom System Prompt

You can customize your system prompts, and upload the system prompt file from your local PC to the PBX, then change the default system prompt to the custom system prompt.

Note: To customize a system prompt, please contact Yeastar.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **System Prompt**.
2. In the **Upload System Prompts** section, click **Browse** to choose the system prompt file (.tar).



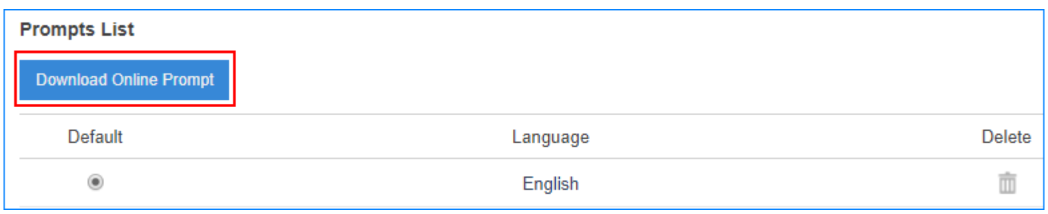
3. **Click Upload.**
   If the file is uploaded successfully, you can see the prompt file in the **Prompt List**.
4. Set the uploaded system prompt as the default system prompt.

5. Click **Save** and **Apply**.

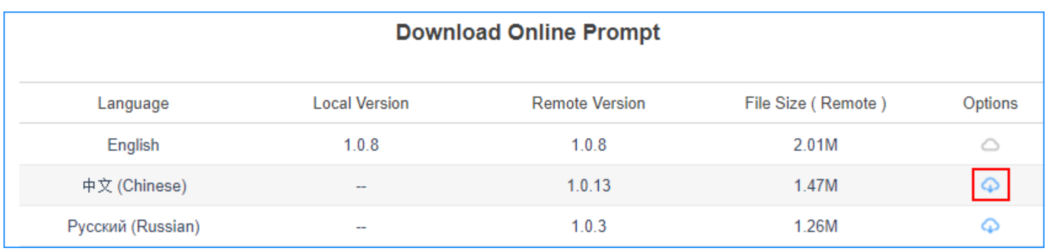
Change an Extension's System Prompt

If a user works in a foreign language, you can set a different system prompt for the extension user.

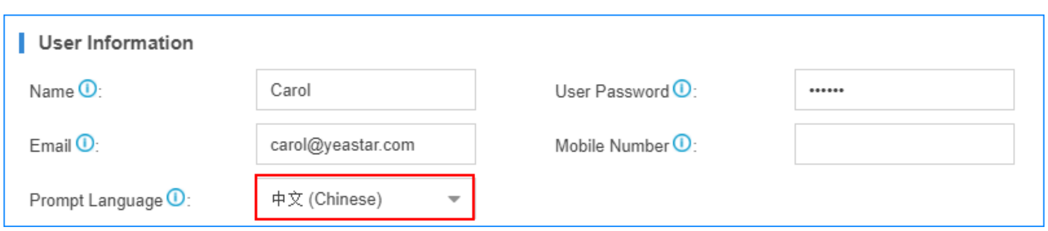
1. Download a system prompt for the extension user.
   a) Go to **Settings** > **PBX** > **Voice Prompts** > **System Prompt**.
   b) Click **Download Online Prompt**.



   c) On the **Download Online Prompt** page, select your desired system prompt, click to download the file.
   After the file is downloaded, you can see the system prompt in **Prompt List**.



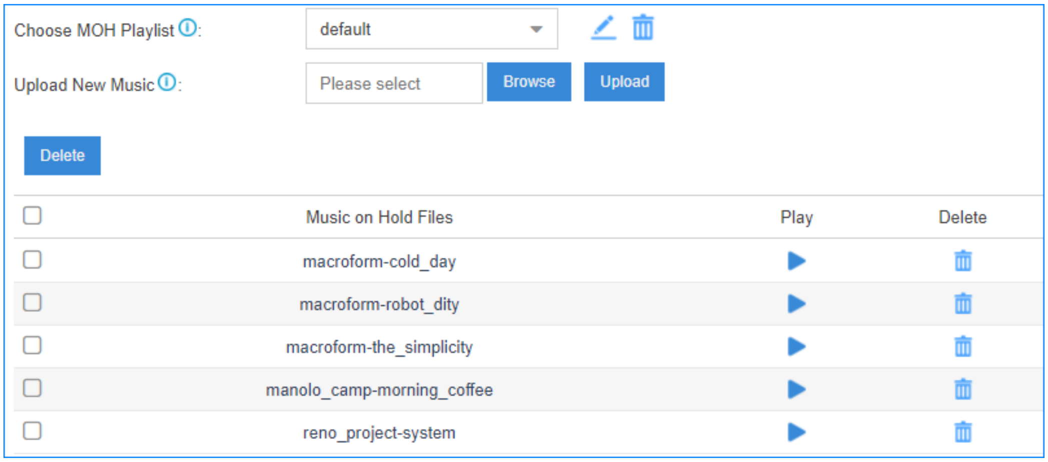
2. Go to **Settings** > **PBX** > **Extensions**, select the desire extension, click .
3. On the **Basic** page, set the **Prompt Language**.



Music on Hold (MoH)

Music on hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by callers who have been placed on hold.

The PBX has a default MoH playlist, you can add MoH playlists and upload music files to the PBX.

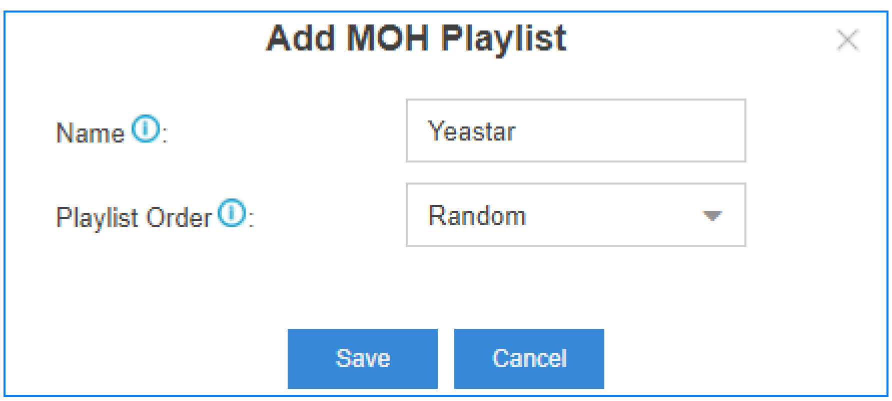


Notice: The default MoH files are distributed under the Creative Commons Attribution-ShareAlike3.0 license through explicit permission from their authors.

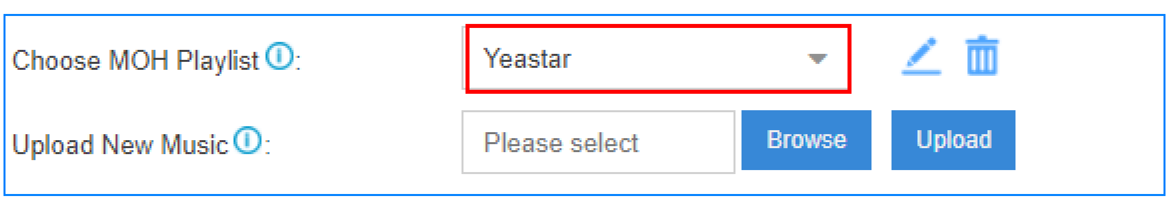
Add a Custom MoH Playlist

You can add a custom MoH playlist and upload your audio files to the PBX.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Music on Hold**, click **Create New Playlist**.
2. On the configuration page, set the playlist name and the playlist order, click **Save**.



3. On the **Music On Hold** page, choose the new created playlist.

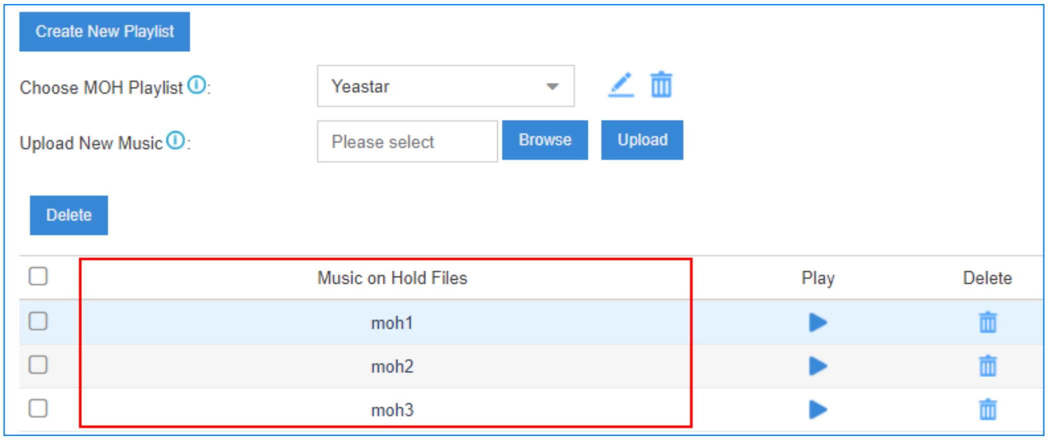


4. Click **Browse** to choose an audio file from your local PC, then click **Upload**.

   Note: The uploaded file should meet the audio file requirements.

5. Repeat the step 4 to add another audio file.

   You can see the uploaded audio files in the MoH list.



Related concepts

Requirements of Custom Audio Files on page 94

Related tasks

Change the MoH Playlist on page 94

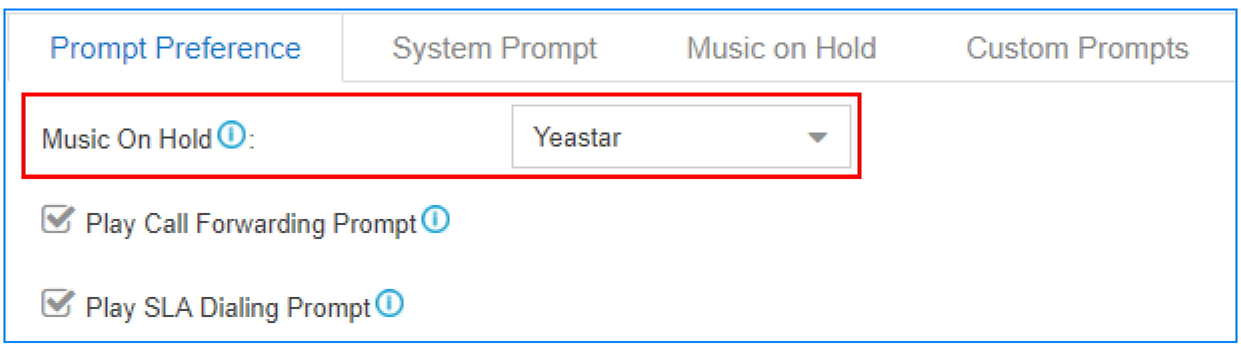
Covert Audio Files via Online Tool on page 97

Convert Audio Files via WavePad on page 97

Change the MoH Playlist

To change the MoH playlist, you need to first add a MoH playlist and upload your audio files to the PBX.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Prompt Preference**.
2. Select a MoH playlist from the drop-down menu of **Music On Hold**.



The PBX will play the select MoH playlist when a user is held in a call.

Related tasks

Add a Custom MoH Playlist on page 93

Custom Prompt

The default voice prompts and announcements in the system are suitable for almost every situation.

However, you may want to use your own voice prompt to make it more meaningful and suitable for your case. In this case, you need to upload a custom prompt to the system or record a new prompt and apply it to the place you want to change.

Requirements of Custom Audio Files

You can upload your audio file to the PBX, the audio file should meet the following requirements.

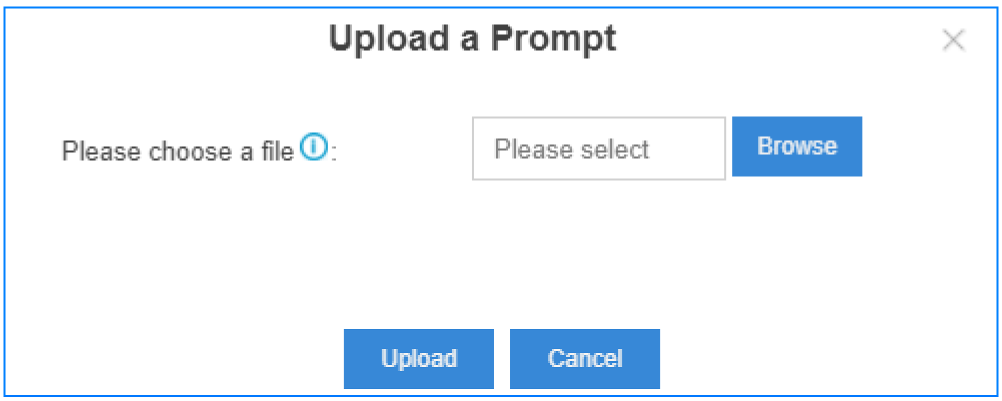
Option	Requirement
File Format	`WAV`, `wav`, or `gsm` file. • gsm 6.10 8kHz, Mono, 1Kb/s • alaw 8kHz, Mono, 1Kb/s • ulaw 8kHz, Mono, 1Kb/s • pcm 8kHz,Mono,16Kb/s
File Name	Should NOT contain special characters.
File Size	Smaller than 8MB.

Related tasks

Covert Audio Files via Online Tool on page 97
Convert Audio Files via WavePad on page 97

Upload a Custom Prompt

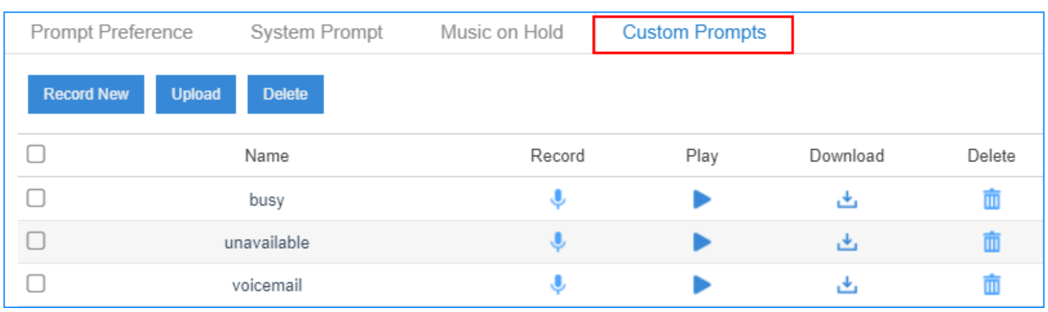
1. Go to **Settings** > **PBX** > **Voice Prompts** > **Custom Prompts**, click **Upload**.
2. On the configuration page, click **Browse** to choose your audio file.



Note: The uploaded file should meet the audio file requirements.

3. Click **Upload** to start uploading the file.

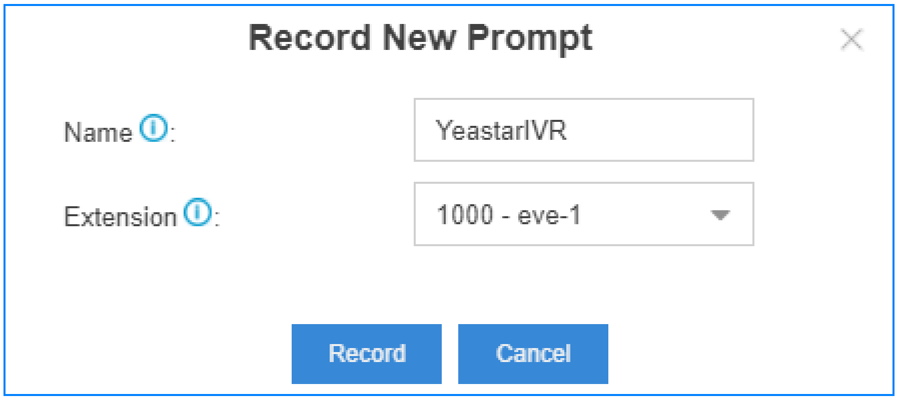
After the file is uploaded, you can see the file on the **Custom Prompts** page.



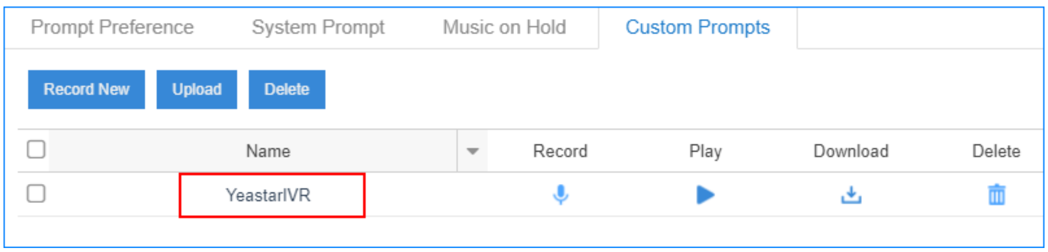
Record a Custom Prompt

You can use an extension to record custom prompts.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Custom Prompts**, click **Record New**.
2. On the configuration page, set the prompt name and select an extension to record the prompt.



3. Click **Save**.
   The selected extension will ring.
4. Record your prompt on the phone. When done, press the # key or hang up.
5. Refresh the **Custom Prompts** page, you can see the saved prompt file.



You can click to play the prompt, and decide whether to save it or not. If you are not satisfied with the prompt, click to record again.

Related tasks

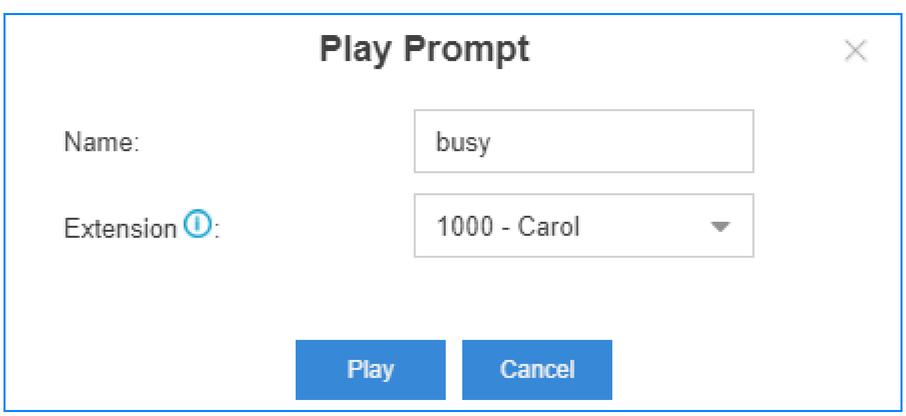
Play a Custom Prompt on page 96

Play a Custom Prompt

After you upload a custom prompt or record a custom prompt, you can select an extension to play the prompt.

Note: We recommend that you play your custom prompts before you apply the custom prompts to IVR, MoH, or other places.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Custom Prompts**.
2. In the Custom Prompts list, choose a prompt, click .
3. On the configuration page, choose an extension to play the prompt.



4. Click **Play**.

The selected extension will ring.

5. Pick up the phone to listen to the prompt.

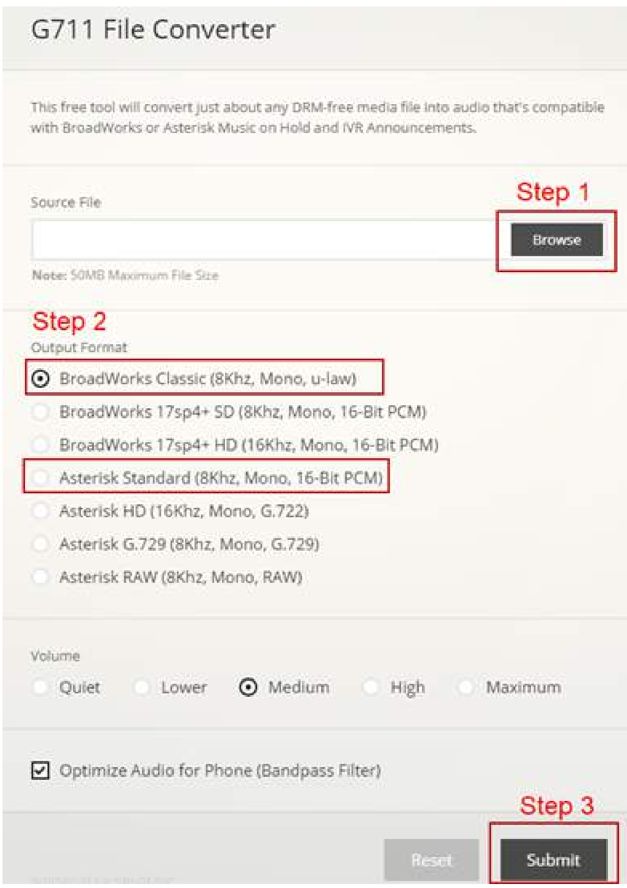
Related tasks

Upload a Custom Prompt on page 95
Record a Custom Prompt on page 95

Covert Audio Files via Online Tool

You can quickly convert your audio files via G711 File Converter online.

1. Visit g711.org.
2. Click **Browse** to upload your audio file.
3. Set the **Output Format**.
   We recommend **BroadWorks Classic** or **Asterisk Standard**.
4. Click **Submit** to start converting the file.

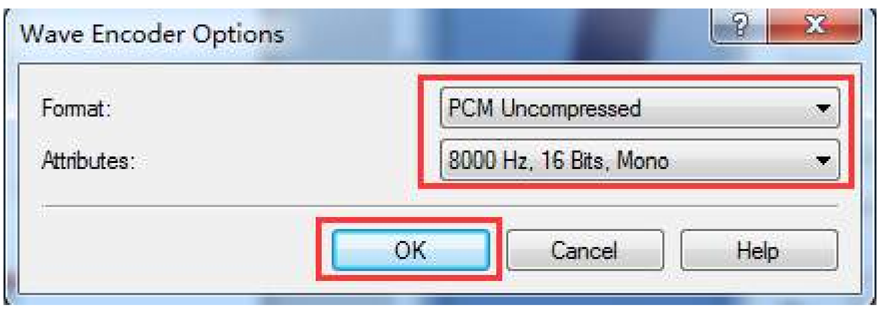


Convert Audio Files via WavePad

WavePad is audio editing software, you can convert audio files via WavePad, then upload the audio files to your PBX.

1. Launch WavePad, open your audio file.
2. Click **File** > **Save File As**.

3. Set the **Save as type** to `.wav` or `.gsm`, click **Save**.
4. For the `.wav` type, set the encoder options according to the requirements of custom audio files, click **OK**.



Related concepts

Requirements of Custom Audio Files on page 94

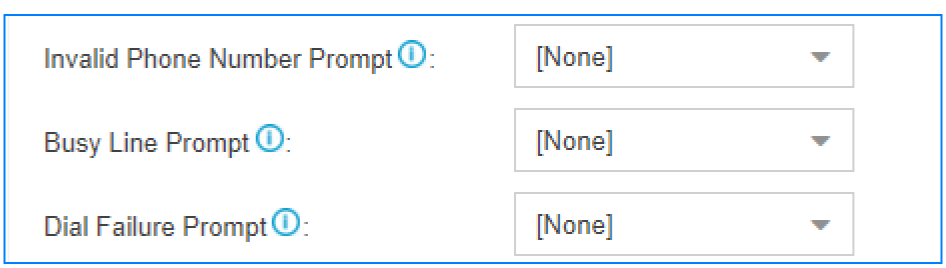
Related tasks

Covert Audio Files via Online Tool on page 97

Set Prompts for Failed Calls

A user may fail to make outbound calls due to many reasons, such as the trunk is busy, no trunk available, or invalid number. You can set different prompts to inform the user why the call fails.

1. Go to **Settings** > **PBX** > **Voice Prompts** > **Prompt Preference**.
2. Set the prompts for different type of failed calls.



- **Invalid Phone Number Prompt**: The PBX will play the prompt when the dialed number is invalid.
- **Busy Line Prompt**: The PBX will play the prompt when the trunk used is busy.

- **Dial Failure Prompt**: The PBX will play the prompt if no trunk is available to call out.

Multisite Interconnect

You can connect multi-site Yeastar S-Series VoIP PBX easily by **Multisite Interconnect** feature. After connecting different branch PBXs to the headquarter PBX, you can achieve:

- Making internal calls between each branch and the headquarter.
- Making internal calls between each two branches.

Video Tutorial

Compatibility

Yeastar S-Series VoIP PBX version 30.6 or later.

Main Steps to Interconnect Yeastar S-Series VoIP PBX

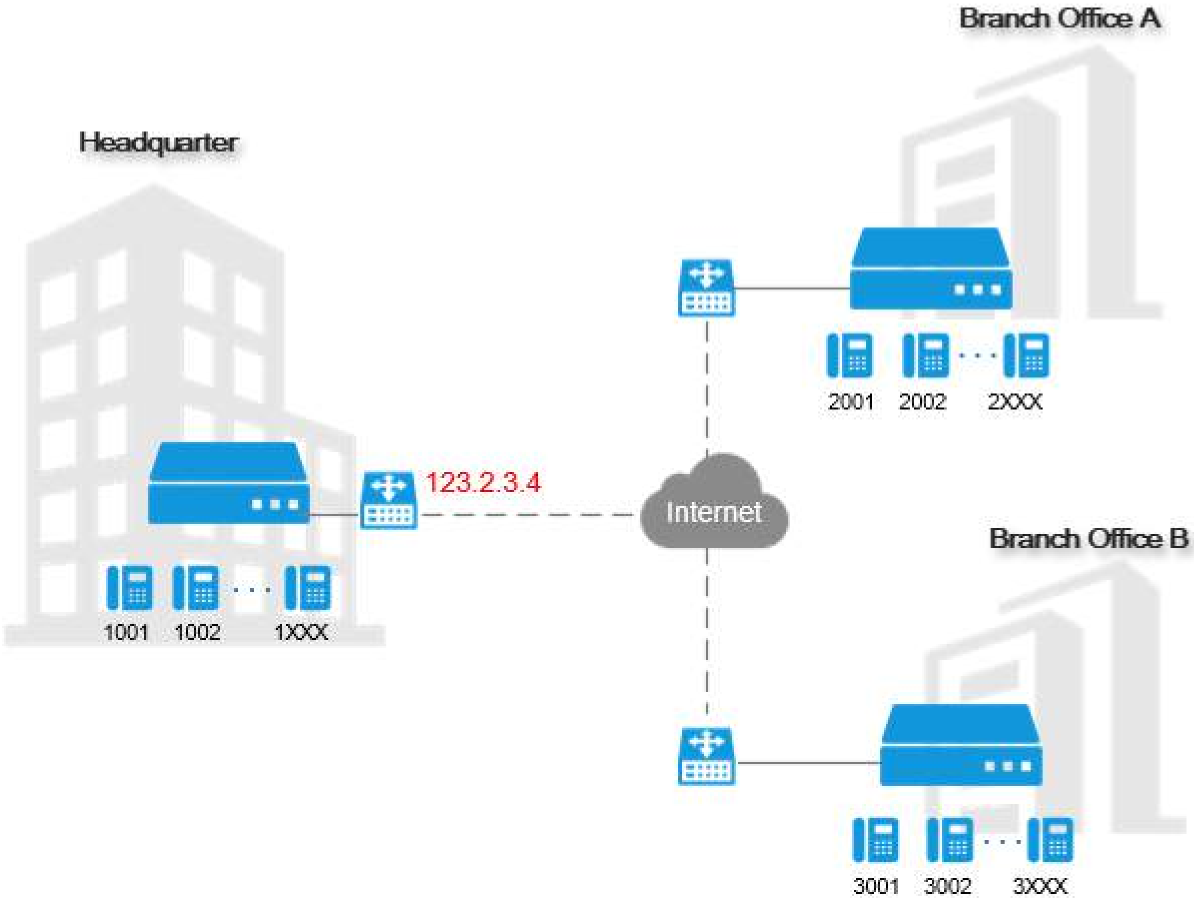
1. Plan and Assign Extensions
2. Forward Ports for Headquarter PBX
3. Set up the Headquarter PBX
4. Connect BranchA PBX and Headquarter PBX
5. Connect BranchB PBX and Headquarter PBX

Interconnect Multiple S-Series VoIP PBXs

In this section, we introduce how to interconnect 3 Yeastar S-Series VoIP PBXs. The 3 PBXs are located in different places. You can connect more Yeastar S-Series IPPBXs in the same way.

After interconnecting the 3 PBXs:

- Users in Branch A and Branch B can make internal calls.
- Users in Branch A and the Headquarter can make internal calls.
- Users in Branch B and the Headquarter can make internal calls.



Plan and Assign Extensions

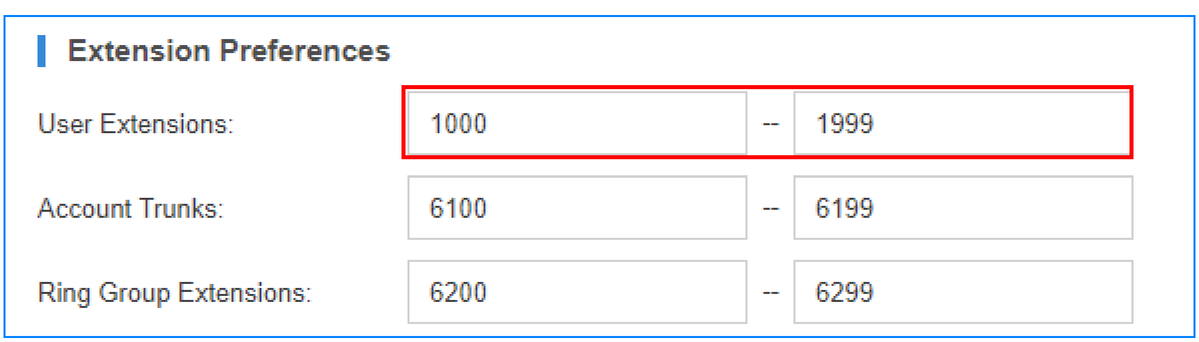
Before you start to connect the Yeastar S-Series VoIP PBXs, you need to plan and assign extensions for the headquarter and the branch offices.

In our example, we assign extensions as bellow:

- **Headquarter PBX**: 1XXX (extension number starts with 1)
- **BranchA PBX**: 2XXX (extension number starts with 2)
- **BranchB PBX**: 3XXX (extension number starts with 3)

Plan and assign the extensions on the headquarter PBX and the branch PBXs.

1. Log in the PBX web interface, go to **Settings** > **PBX** > **General** > **Preferences**.
2. Change the **User Extensions** to the desired range.



3. Click **Save** and **Apply**.
4. Go to **Settings** > **PBX** > **Extensions**, create extensions.

Forward Ports for Headquarter PBX

If the headquarter PBX is behind a router, you need to forward ports for the headquarter PBX.

Note: You don't need to forward ports for the branch PBXs.

1. Log in the router that is connected to the headquarter PBX.
2. Forward ports for the PBX.

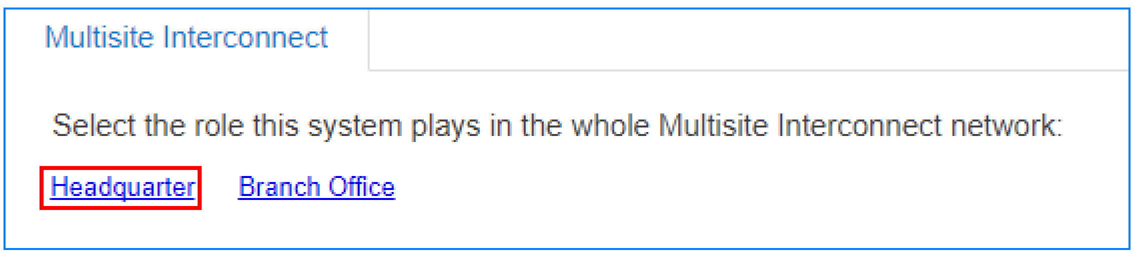
   Below are the default ports you need to forward:

   - **SIP Registration Port**: UDP 5060
   - **RTP Ports**: UDP 10000-12000

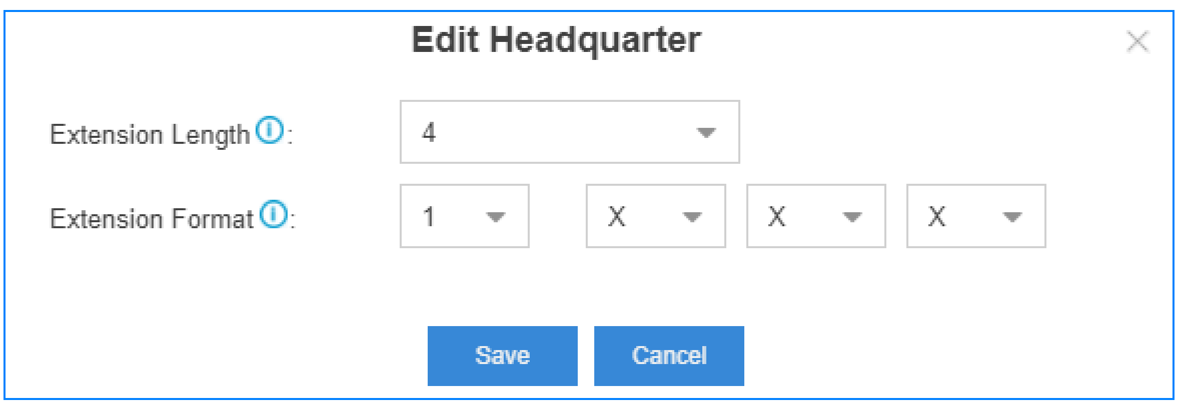
Set up Headquarter PBX

Set the role of the headquarter PBX as **Headquarter**, and create interconnections for the branch PBXs.

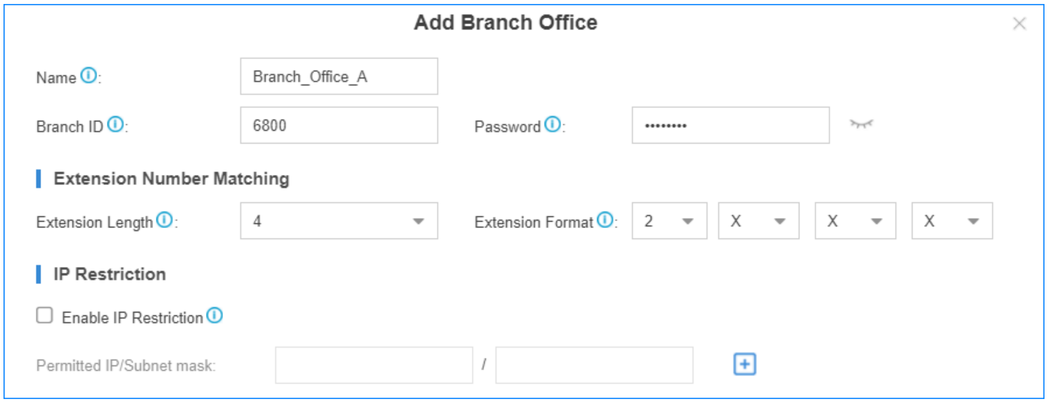
1. Log in the web interface of the headquarter PBX, go to **Settings** > **Multisite Interconnect**.
2. Set the PBX as a Headquarter system.
   a) Click **Headquarter** to set the role of the PBX as headquarter.



   b) Click to edit the Headquarter according to the extensions length and extension format of this IPPBX.



   c) Click **Save** and **Apply**.
3. Create an interconnection for branch office A.
   a) Click **Add** to create an interconnection.



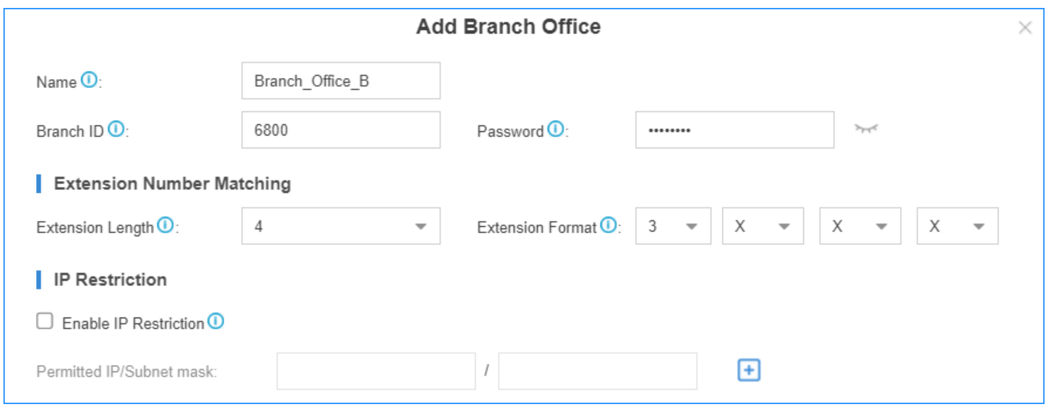
b) Set the name, branch ID and password.

> **Note:** You need to use the branch ID and password to connect the branch office A and the headquarter later.

c) Set the extension length and the extension format of the branch office A.
d) Optional: Enable IP restriction and enter the permitted IP and subnet mask. Only the permitted IP address can connect to the system with this branch ID.
e) Click **Save** and **Apply**.

4. Create an interconnection for branch office B.

a) Click **Add** to create an interconnection.



b) Set the name, branch ID and password.

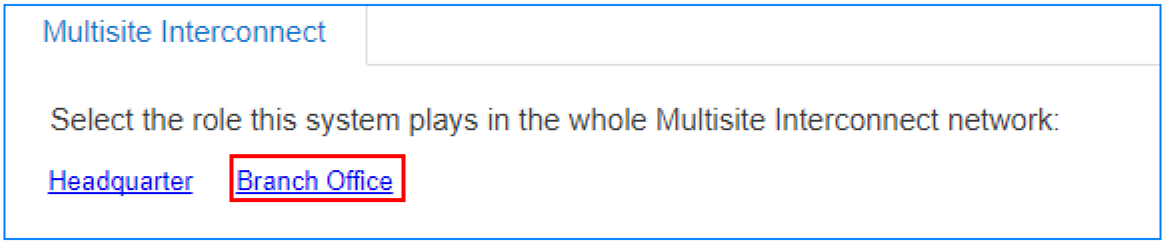
> **Note:** You need to use the branch ID and password to connect the branch office B and the headquarter later.

c) Set the extension length and the extension format of the branch office B.
d) Optional: Enable IP restriction and enter the permitted IP and subnet mask. Only the permitted IP address can connect to the system with this branch ID.
e) Click **Save** and **Apply**.

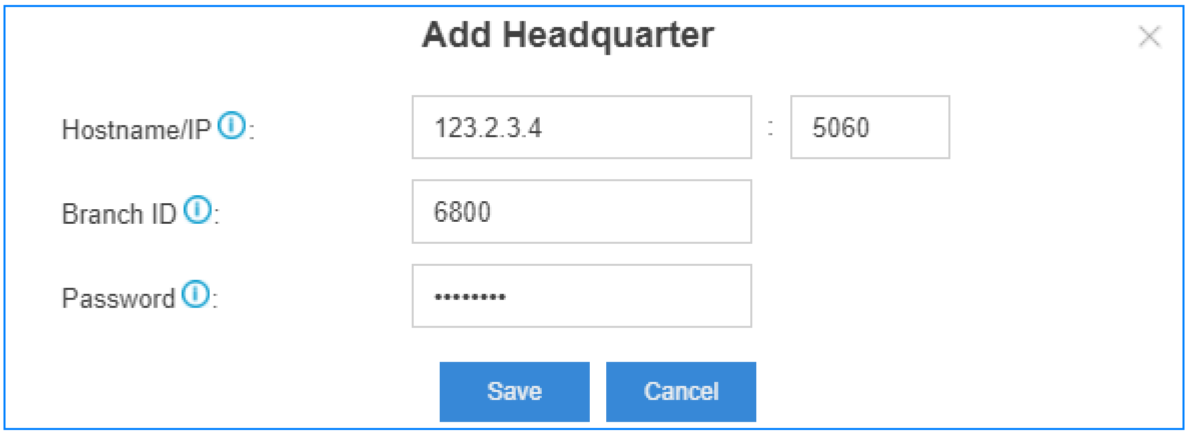
Connect BranchA PBX and Headquarter PBX

Set the role of the BranchA PBX as **Branch**, and connect to the headquarter PBX. Users can make internal calls between the branch office A and the headquarter.

1. Log in the web interface of the BranchA PBX, go to **Settings** > **Multisite Interconnect**.
2. Click **Branch Office** to set the PBX as a branch PBX.



3. Create an interconnection to the headquarter PBX.
   a) Click **Add** to add an interconnection.



   b) Enter the IP/domain and SIP port of the headquarter PBX.
   c) Enter the branch ID and password that were set on the headquarter PBX for branch office A.
4. Click **Save** and **Apply**.

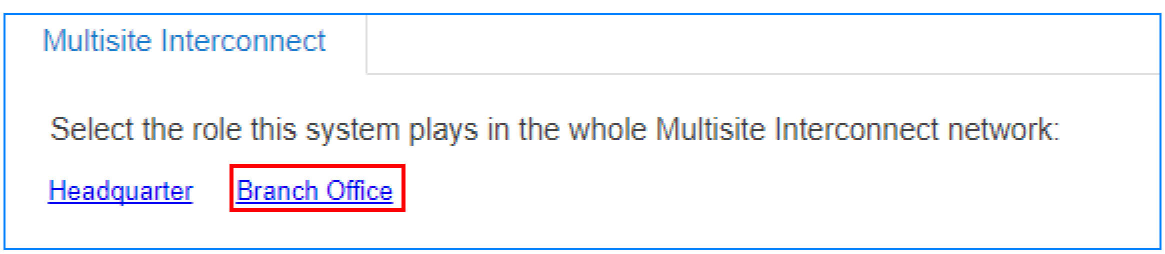
   If the interconnection status shows , the branch PBX is connected to the headquarter PBX.

   The users in Branch Office A and the Headquarter Office can make internal calls.

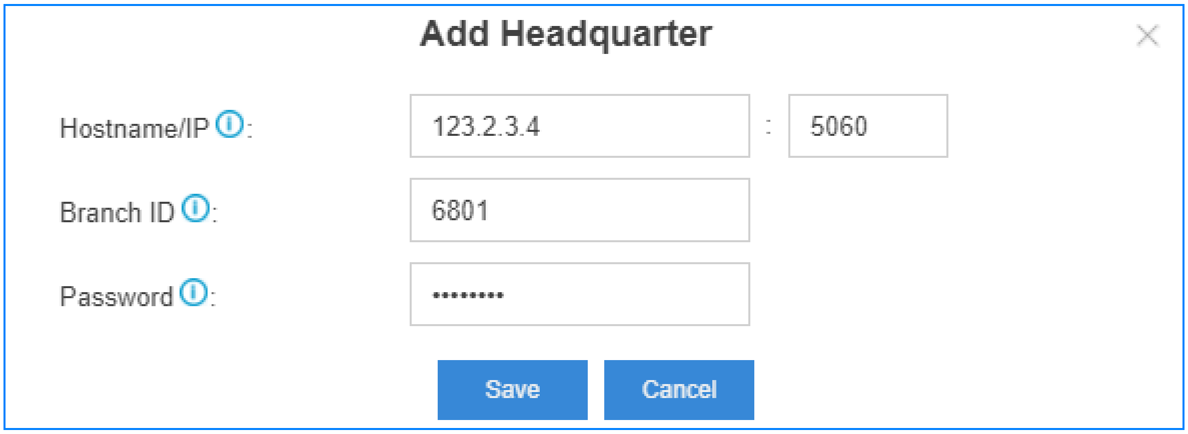
Connect BranchB PBX and Headquarter PBX

Set the role of the BranchB PBX as **Branch**, and connect to the headquarter PBX. Users can make internal calls between the branch office B and the headquarter.

1. Log in the web interface of the BranchB PBX, go to **Settings** > **Multisite Interconnect**.
2. Click **Branch Office** to set the PBX as a branch PBX.



3. Create an interconnection to the headquarter PBX.
   a) Click **Add** to add an interconnection.



b) Enter the IP/domain and SIP port of the headquarter PBX.
c) Enter the branch ID and password that were set on the headquarter PBX for branch office B.

4. Click **Save** and **Apply**.

If the interconnection status shows , the branch PBX is connected to the headquarter PBX.

As branch A and branch B are both connected to the headquarter PBX.

The users in Branch A, Branch B and Headquarter can make internal calls between each other.

Maintenance

Upgrade Firmware

You have multiple ways to update the PBX firmware. We suggest you back up the PBX configurations before you start to update the PBX firmware.

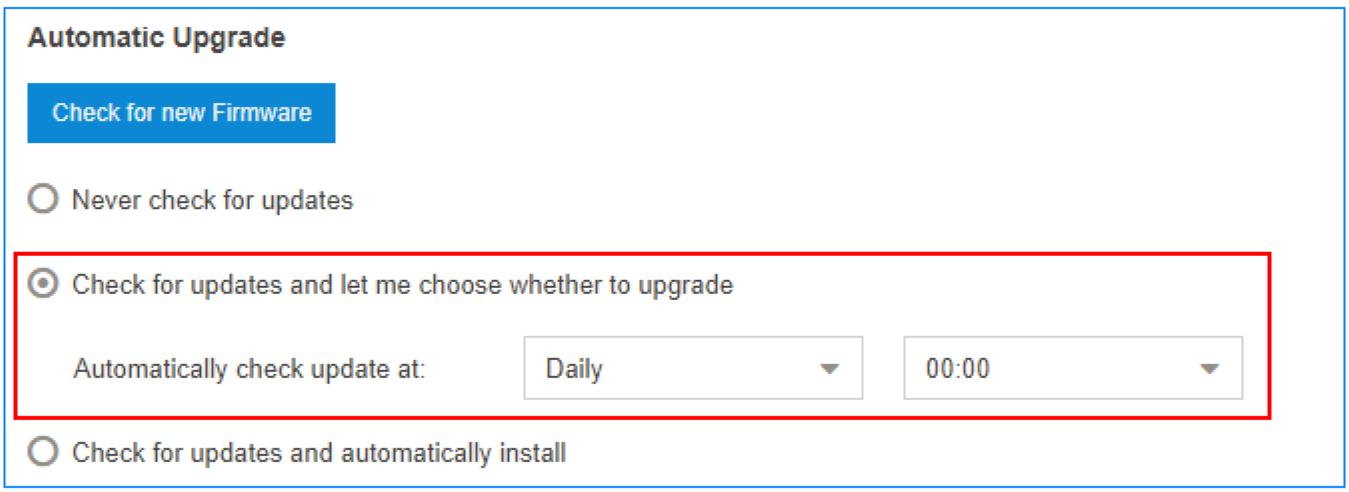
One-Click Upgrade

You can set the PBX to check new firmware version automatically, if the PBX has a new released version, you can do one-click upgrade.

Note: Make sure that the PBX can access the Internet, or the upgrade will fail.

1. Go to **Maintenance** > **Upgrade** > **Automatic Upgrade**.
2. Check the option **Check for updates and let me choose whether to upgrade** and set when the PBX should check the update.

   If the PBX has new firmware version, the new firmware version will appear on this page.

3. Click **New** to check what's new for the new version.
4. Click **Upgrade Now** to upgrade the PBX to the new version.

   Note:

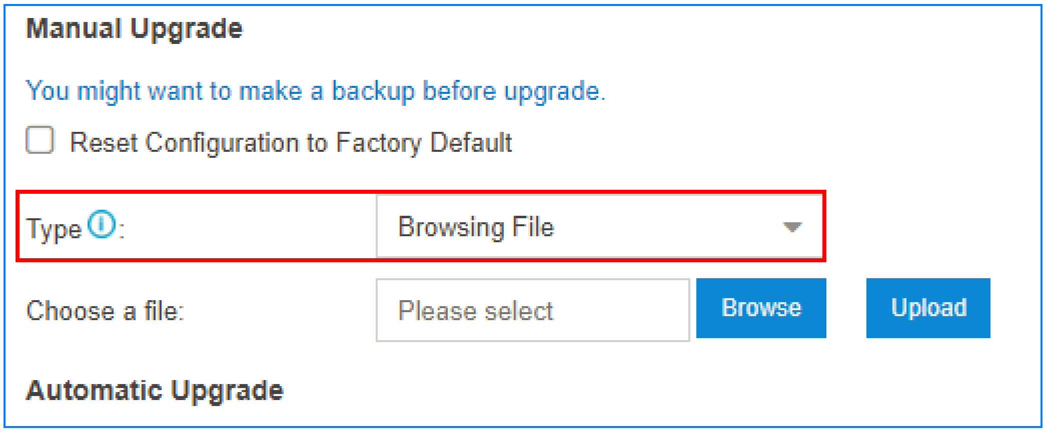
   - When the PBX is updating the firmware, do NOT turn off the power. Or the system will get damaged.

Browse a Local File to Upgrade

You can download the latest firmware file from Yeastar Firmware Download center, and upgrade the PBX firmware by uploading the firmware file from your local PC.

This upgrade method is suitable when the PBX cannot access the Internet.

1. Go to **Maintenance** > **Upgrade** > **Manual Upgrade**.



2. If you want to reset the configuration to factory defaults, check the option **Reset Configuration to Factory Default**.

   Important: If you check this option, all your PBX configurations will be erased. We don't suggest you reset the PBX before upgrade firmware.

3. Set **Type** to **Browing File**.
4. Click **Browse** to choose your local firmware file.

   Note: The firmware file format should be `.bin`, and the file name should not have special characters.

5. Click **Upload**.
   The PBX will start uploading the file and upgrading the firmware automatically.

   Note:

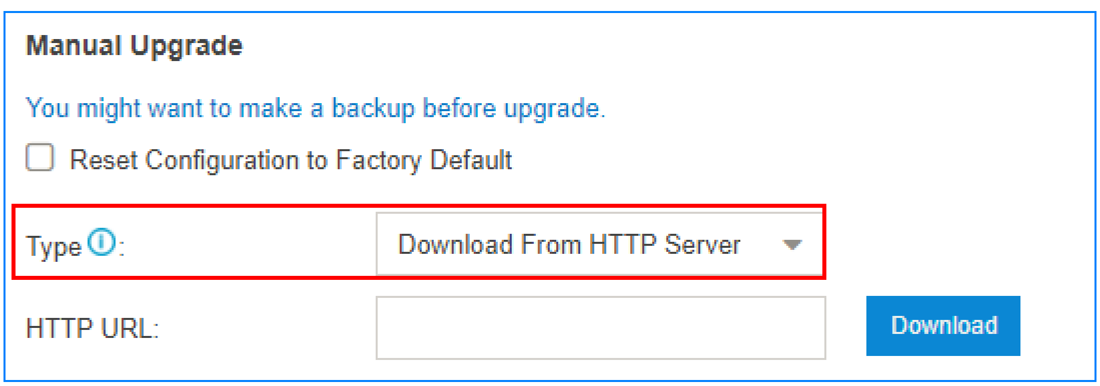
   - When the PBX is updating the firmware, do NOT turn off the power. Or the system will get damaged.

Upgrade Firmware by HTTP Method

You can get the firmawre download link from Yeastar Firmware Download center, and enter the link to the PBX web interface to upgrade firmware.

Make sure that the PBX could access the Internet, or the upgrade will fail.

1. Go to **Maintenance** > **Upgrade** > **Manual Upgrade**.



2. If you want to reset the configuration to factory defaults, check the option **Reset Configuration to Factory Default**.

   Important: If you check this option, all your PBX configurations will be erased. We don't suggest you reset the PBX before upgrade firmware.

3. Set **Type** to **Download From HTTP Server**.
4. Enter the firmware download link in the **HTTP URL** field.

Note: The URL should be a download link for a `.bin` file. For example, `http://www.yeastar.com/download/S_Series/30.7.0.27.bin`.

5. Click **Download**.
   The PBX will start downloading file from the HTTP server, and upgrading the firmware automatically.

   Note:

   - When the PBX is updating the firmware, do NOT turn off the power. Or the system will get damaged.

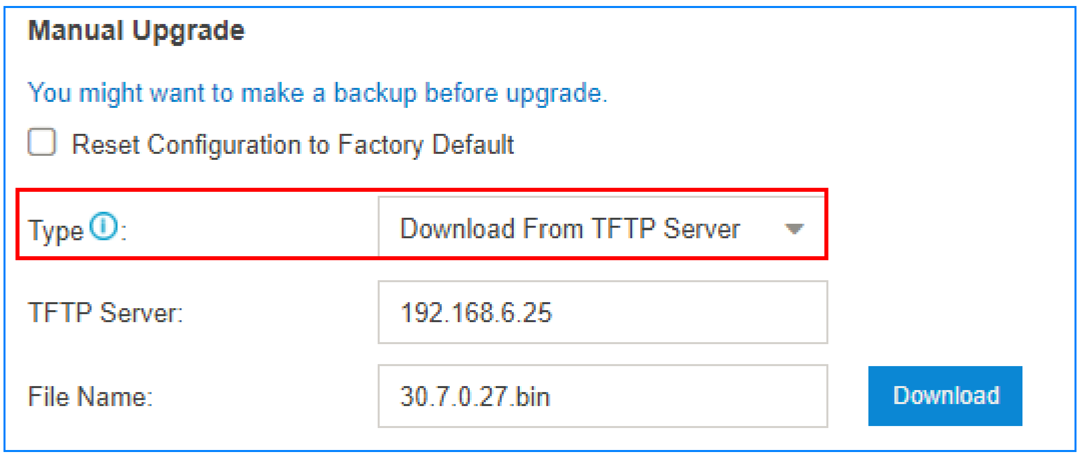
Upgrade Firmware by TFTP Method

You can download the latest firmware file from Yeastar Firmware Download center to your TFTP server. Then upgrade the PBX firmware from the TFTP server.

If the TFTP server is your local PC, this upgrade method is suitable when the PBX cannot access the Internet.

1. Go to **Maintenance** > **Upgrade** > **Manual Upgrade**.
2. If you want to reset the configuration to factory defaults, check the option **Reset Configuration to Factory Default**.

   Important: If you check this option, all your PBX configurations will be erased. We don't suggest you reset the PBX before upgrade firmware.
3. Set **Type** to **Download From TFTP Server**.



4. In the **TFTP Server** field, enter the IP address of the TFTP server.
5. In the**File Name** field, enter the firmware file name.

   Note: The file should be a `.bin` file. For example, `30.7.0.27.bin`.
6. Click **Download**.
   The PBX will start downloading file from the TFTP server, and upgrading the firmware automatically.

   Note:

   - When the PBX is updating the firmware, do NOT turn off the power. Or the system will get damaged.

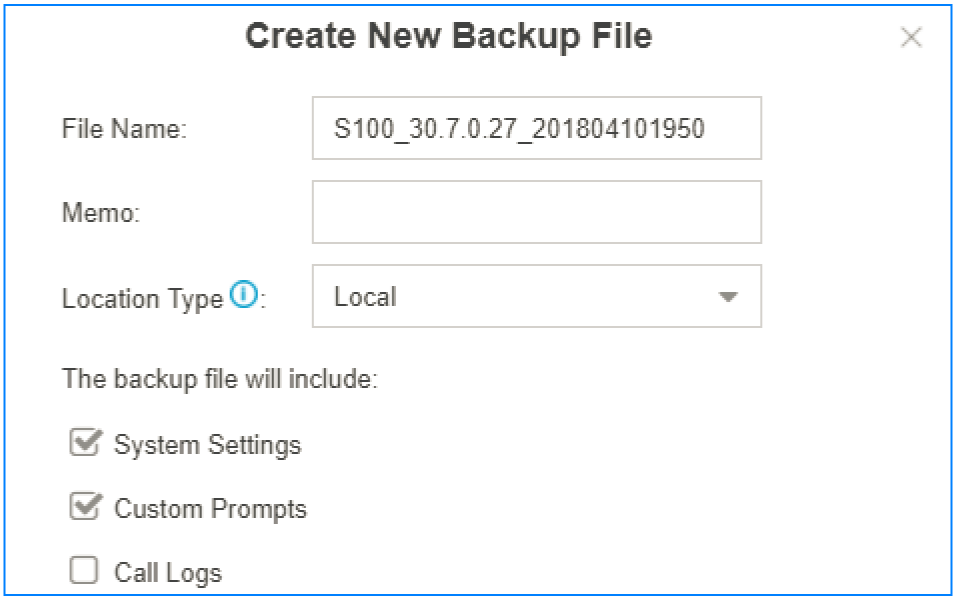
Backup and Restore

Create a Backup File

You can create a backup file of the PBX settings on the PBX web interface.

Note: The backup file do not contain auto recording files, one-touch recording files, and voicemail files.

1. Go to **Maintenance** > **Backup and Restore**, click **Backup**.



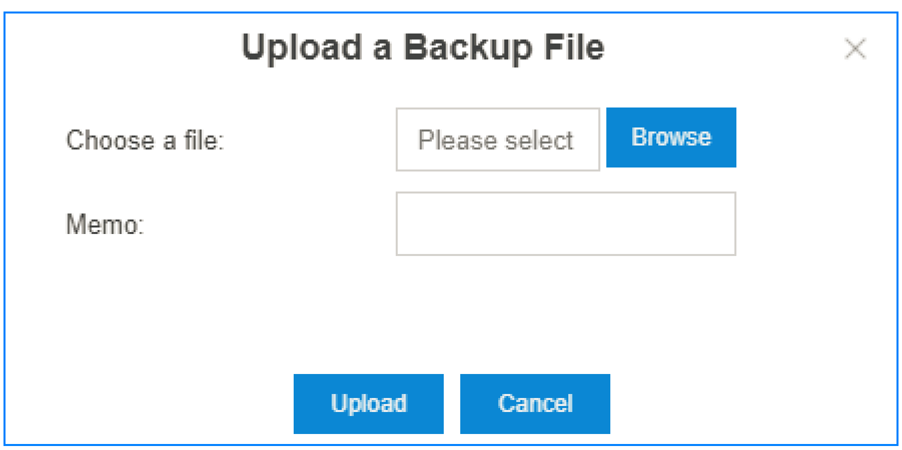
2. Set the **File Name**.
   The default file name contains the PBX model, firmware version, and backup date.
3. In the **Memo** field, enter notes for the backup file.
4. Select where to store the backup file.
5. Select which configurations and files to back up.
6. Click **Save**.
   The created backup file will appear on the **Backup and Restore** page.

Upload a Backup File

You can select a backup file from your local PC, and upload the file to the PBX.

Note: The file format is `.bak` and the file name should not contain special characters.

1. Go to **Maintenance** > **Backup and Restore**, click **Backup**.



2. Click **Browse**, and select your backup file to upload.
3. In the **Memo** field, enter notes for the backup file.
4. Click **Upload**.
   The uploaded backup file will appear on the **Backup and Restore** page.

Restore a Backup File

After restore a backup file, the current configurations on your PBX will be OVERWRITTEN with the backup data.

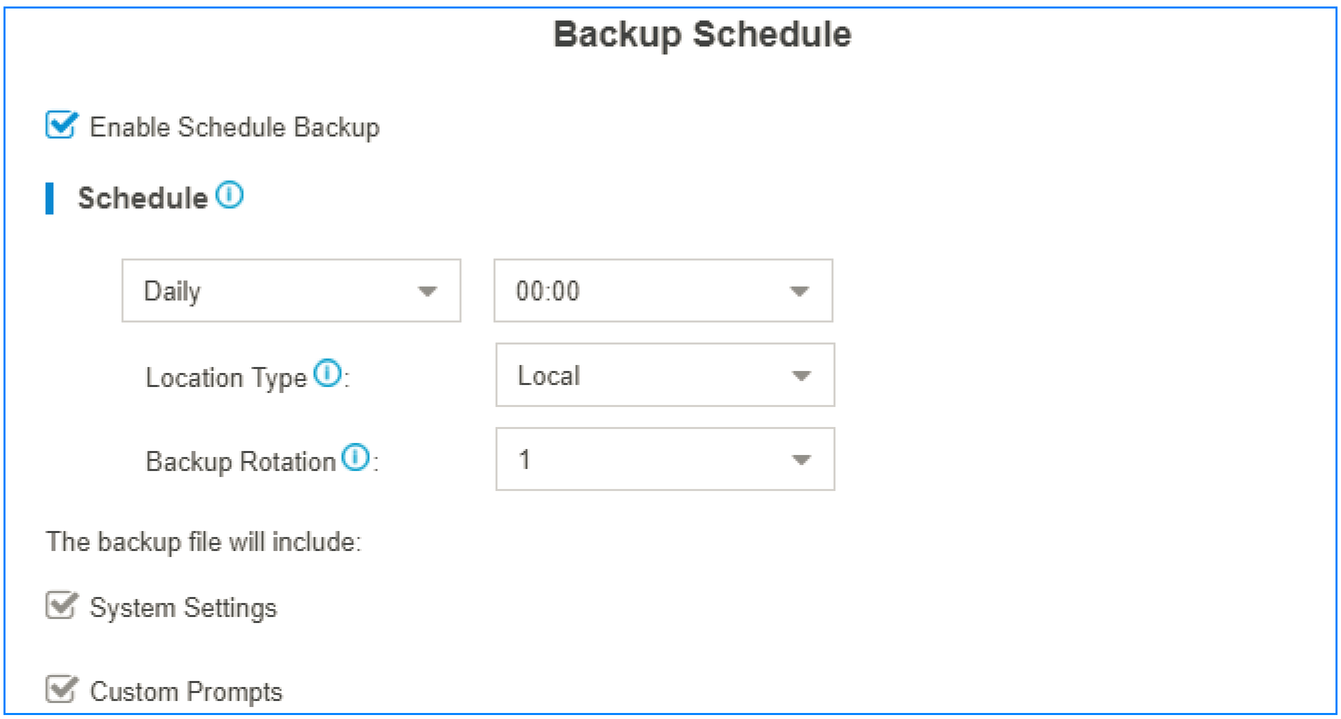
Note:

- If the backup file version is higher than the PBX version, the backup file would not work.
- You cannot restore a backup file that is downloaded from a different PBX model.

1. Go to **Maintenance** > **Backup and Restore**.
2. Choose a backup file, click .
   A pop-up window will apear at the bottom-right of the web page.
3. Click **Yes** to reboot the PBX.
   The PBX starts to restore data from the backup file.

Schedule Auto Backup

1. Go to **Maintenance** > **Backup and Restore**, click **Backup Schedule**.



2. Check the option **Enable Schedule Backup**.
3. Set the backup **Schedule** settings.
   - Frequency and time: Select the backup frequency and when to make the backup.
   - **Location Type**: Select where to store the backup file.
   - **Backup Rotation**: Set the maxmuim number of backup files that is stored in the selected location. When the number of backup files excedes the set value, the oldest file will be replaced with the newest.
4. Set which files to back up.
5. Click **Save**.

Reboot the PBX

Reboot the PBX Immediately

Reboot the PBX immediately via the PBX web interface.

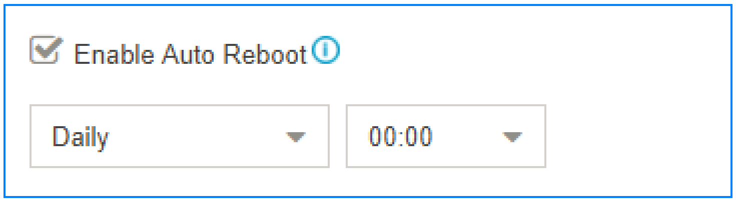
Note: When the PBX is rebooting, all the on-going calls will be terminated.

Go to **Maintenance** > **Reboot**, click **Reboot**.

Schedule Auto Reboot

You can schedule auto reboot to keep the system running smoothly.

1. Go to **Maintenance** > **Reboot**, check the option **Enable Auto Reboot**.

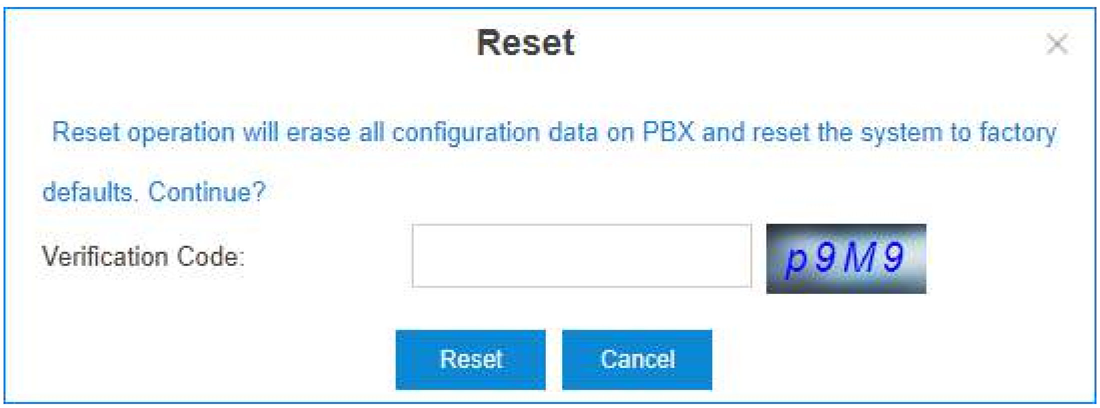


2. Set the frequency and time of auto reboot.
3. Click **Save**.

Reset the PBX

If you want to erase all the configurations on your PBX, you can reset the PBX to the factory defaults.

1. Go to **Maintenance** > **Reset**, click **Reset**.



2. Enter the verification code.
3. Click **Reset**.