Grandstream Networks, Inc.

GCC6020 - User Manual - IPPBX

GCC6020 - User Manual - IPPBX

INTRODUCTION

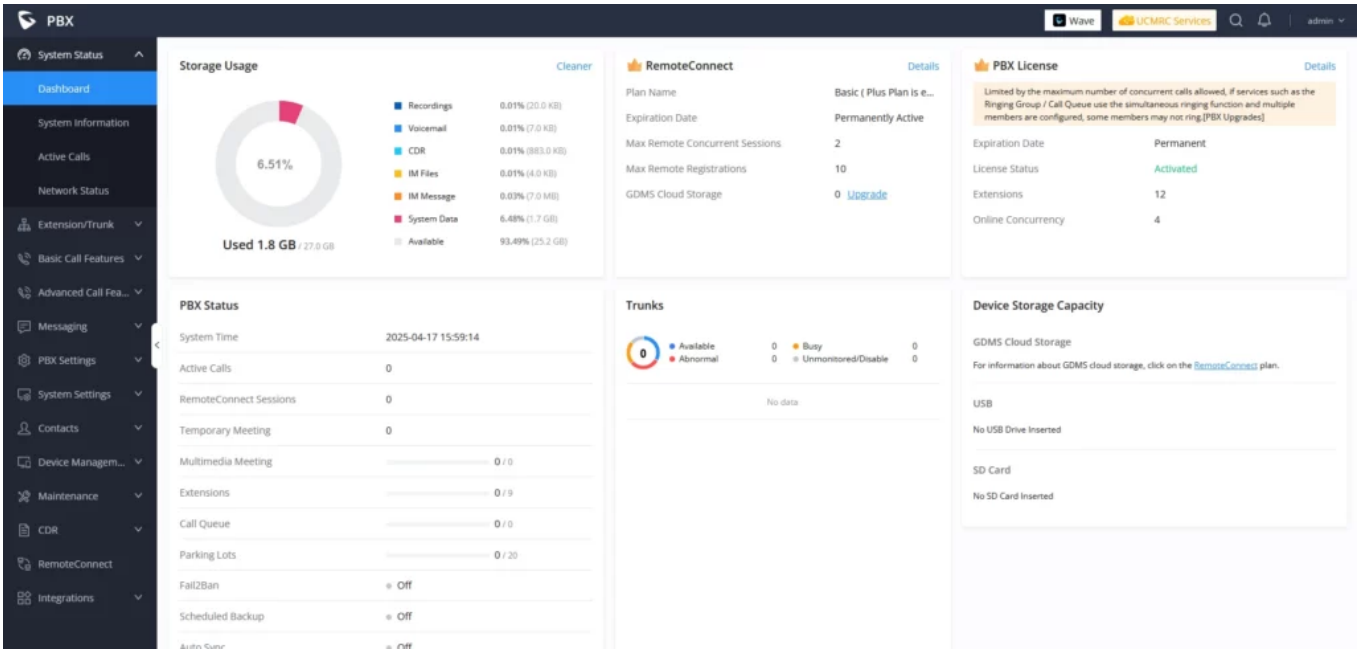
This document explains all the IPPBX features offered by Grandstream GCC6020 All-in-One devices. This document does not include the management features or settings related to the device overall or the other modules of the device.

Grandstream offers expansion package and add-ons for the IPPBX module in the GCC6020 Series. For more details, please refer to the following link: https://ucmrc.gdms.cloud/pbx/plans

SYSTEM STATUS

Dashboard

- **Storage Usage**
- **RemoteConnect**
- **PBX License**
- **PBX Status**
- **Trunks**



GCC602x PBX Module Dashboard

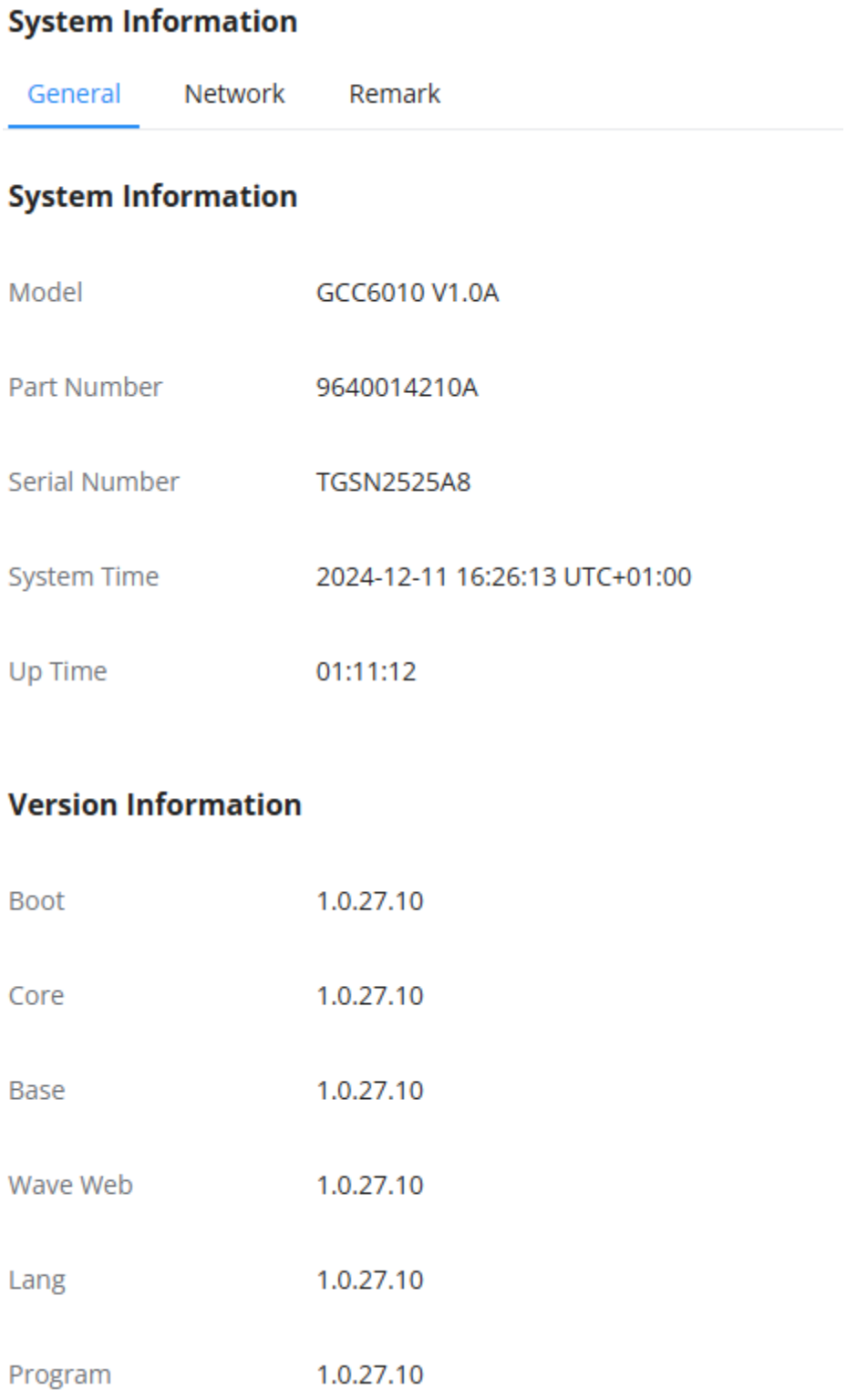
System Information

In this category, the user can view all the information regarding the GCC device hardware and software. In addition to the networking information which is assigned to the device when connected to the network.

General

In this page, the user can see the hardware and software information about the the GCC device. This page includes information about the model of the device, the hardware version, the part number, the serial number, the system time, and the duration of the operation of the device. In addition to the version numbers of the

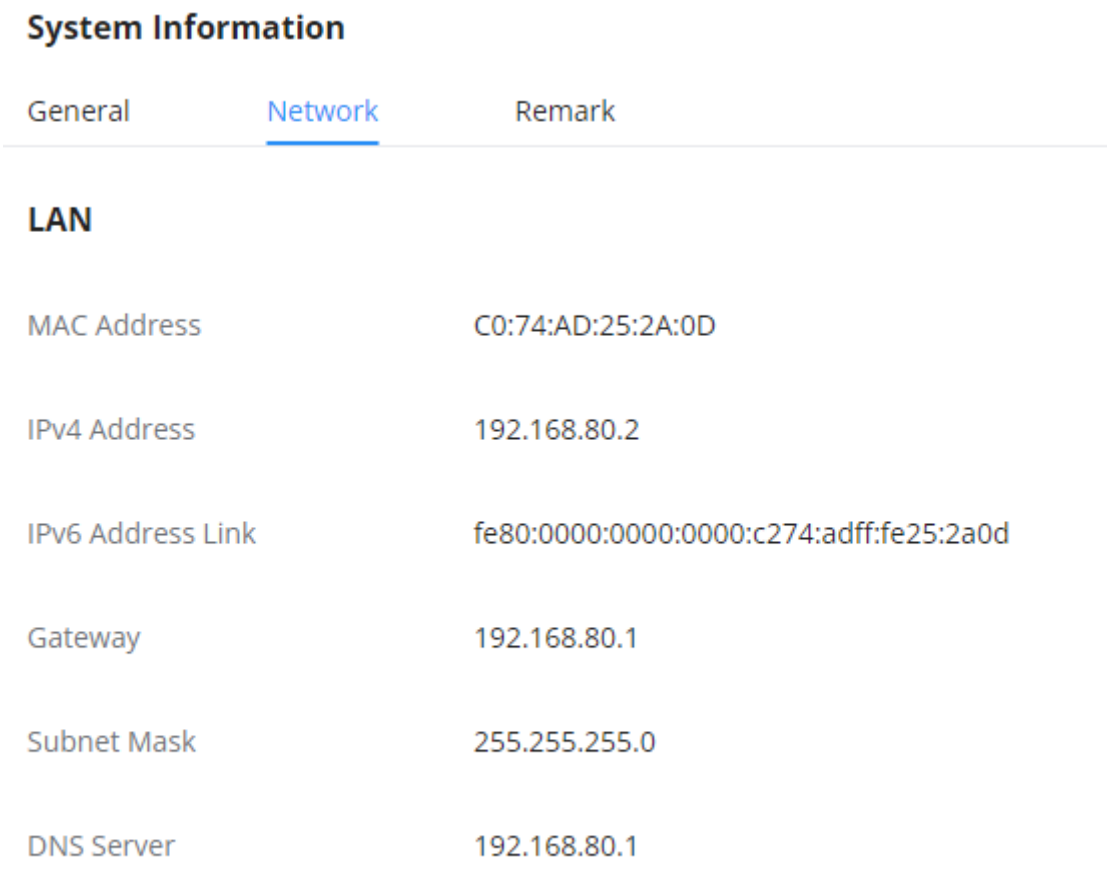
different modules of the device's software.



PBX Module System Information

Network Settings

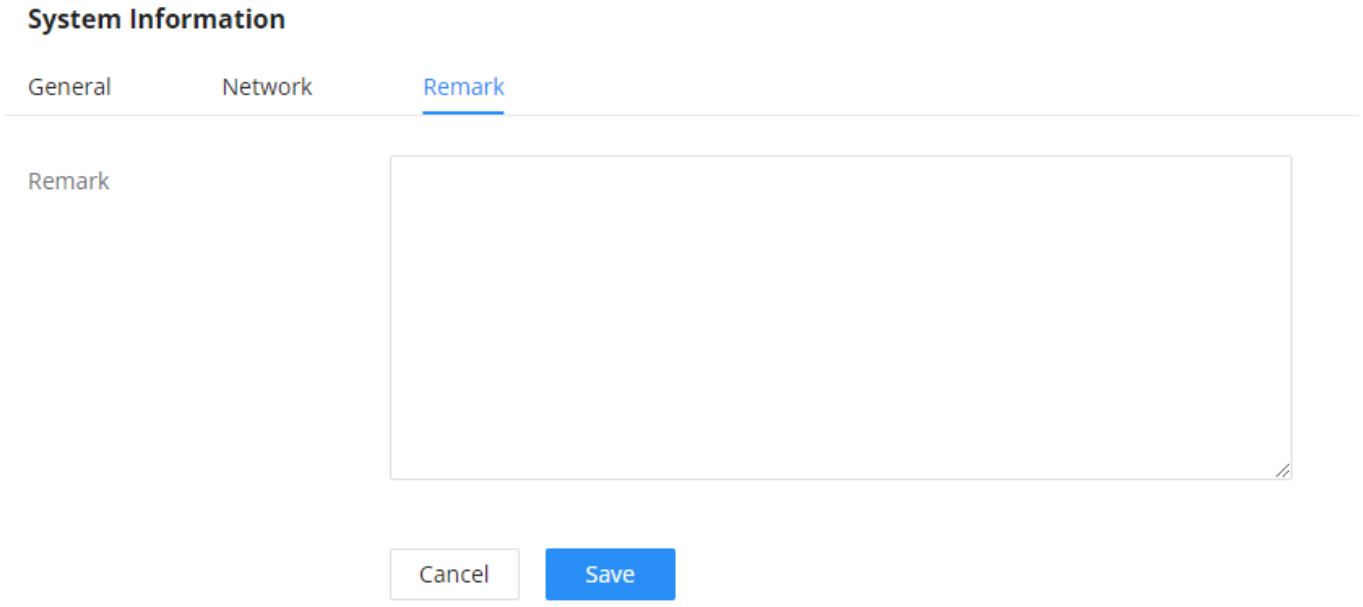
This page displays the network configuration and information of the device. The information displayed include the MAC address, IPv4 address, IPv6 address, the gateway, the subnet mask, and the preferred DNS server.



Network

Remark

In this page, the user can enter specific information about the device to help easy identification.



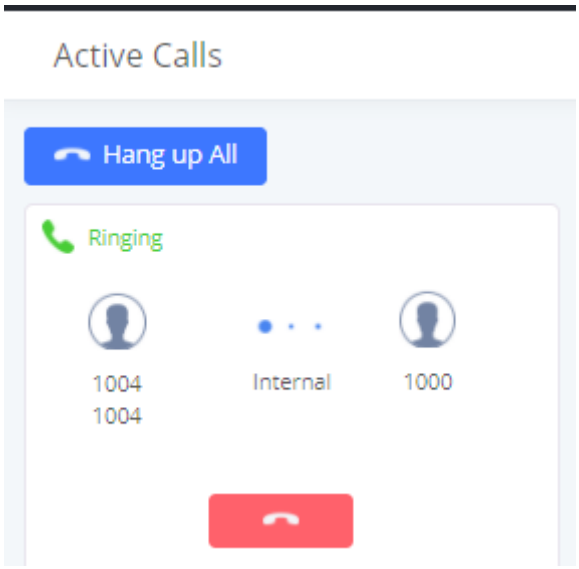
Remark

Active Calls

The active calls on the IPPBX are displayed in the Web GUI→**System Status**→**Active Calls** page. Users can monitor the status, hang up a call, and barge in the active calls in a real-time manner.

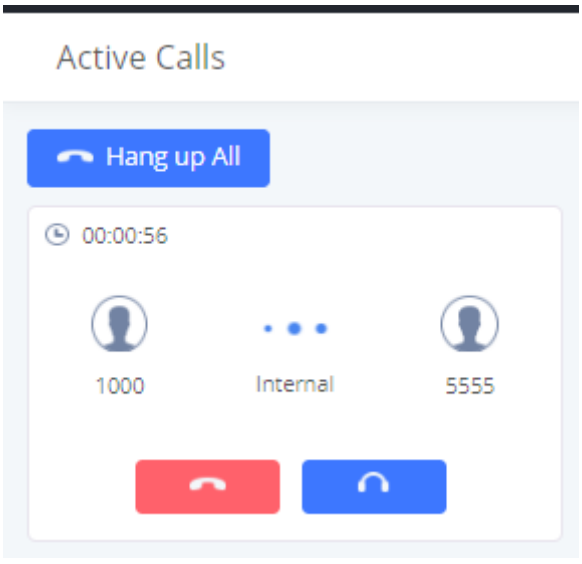
Active Calls Status

To view the status of active calls, navigate to Web GUI→**System Status**→**Active Calls**. The following figure shows extension 1004 is calling 1000. 1000 is ringing.



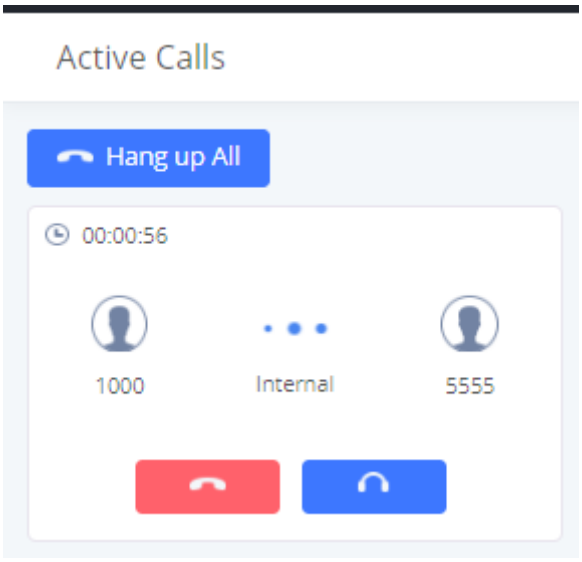
Active Calls Ringing Status

The following figure shows the call between 1000 and 5555 is established.



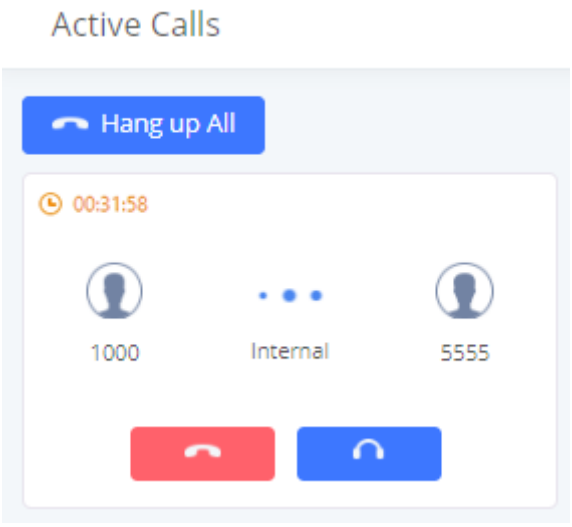
Active Call Established Status

The gray color of the active call means the connection of call time is less than half an hour. It means this call is normal.



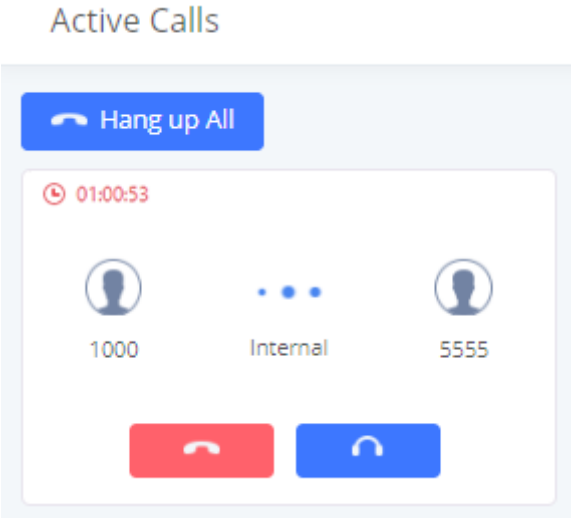
Active Call Short Duration

The orange color of the active call means the connection of call time is greater than half an hour but less than one hour. It means this call is a bit long.



Active Call Long Duration

The red color of the active call means the connection of call time is more than one hour. It means this call could be abnormal.



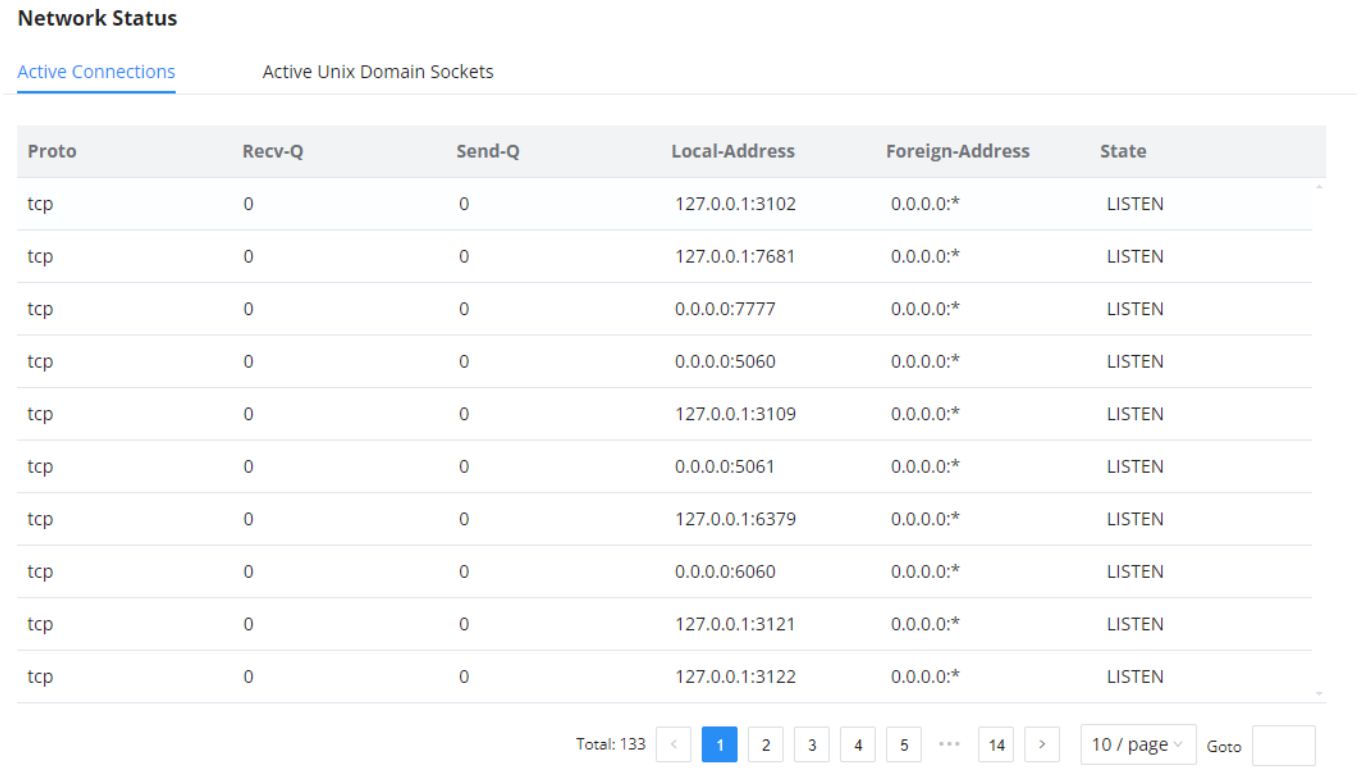
Active Call Very Long Duration

Network Status

This section shows the overall network status of the IPPBX module like the network services which are running on the IPPBX and the active unix domain sockets.

Active Connections

This page shows all the active connections of the IPPBX module with internal modules or devices using TCP/IP.



Active Connections

Active Unix Domain Sockets

This page shows the processes which are open and communicating with each other.

Network Status

Active Connections | Active Unix Domain Sockets

Proto	RefCnt	Flags	Type	State	I-Node
unix	3	[]	STREAM	CONNECTED	47409
unix	3	[]	STREAM	CONNECTED	47270
unix	3	[]	STREAM	CONNECTED	38807
unix	3	[]	STREAM	CONNECTED	48551
unix	3	[]	STREAM	CONNECTED	47510
unix	3	[]	STREAM	CONNECTED	49336
unix	2	[]	DGRAM		47507
unix	3	[]	STREAM	CONNECTED	46336
unix	3	[]	STREAM	CONNECTED	47271
unix	2	[]	DGRAM		42410

Total: 43 < 1 2 3 4 5 > 10 / page Goto

Active Unix Domain Sockets

EXTENSION/TRUNK

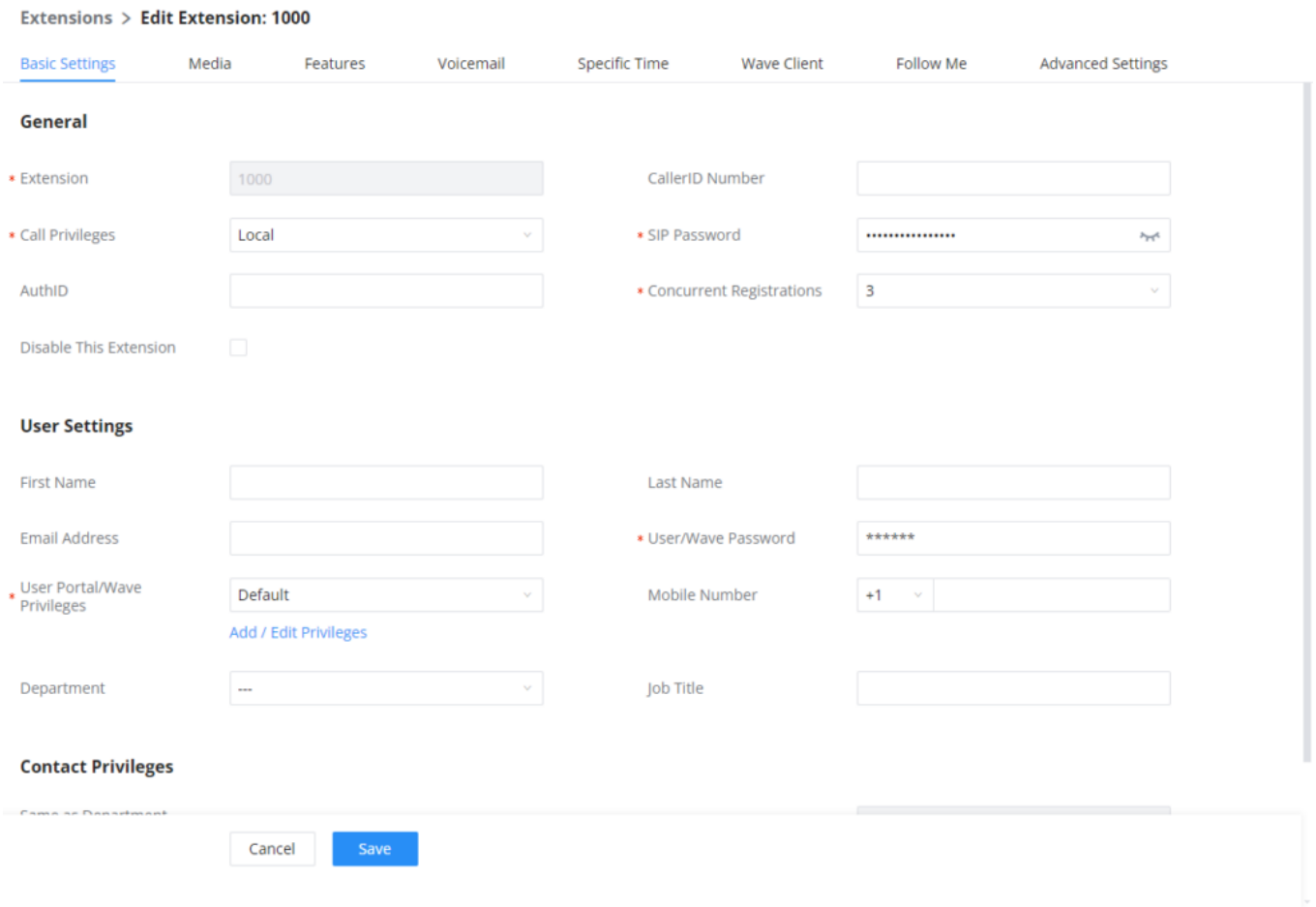
This section combines settings related to extensions, trunks, and inbound/outbound routes.

Extensions

In this page the user can create, view, and configure all the extensions that exist on the PBX.

SIP Extension

To manually create a new SIP user, go to Web GUI→**Extension/Trunk→Extensions**. Click on "Add" and a new window will show for users to fill in the extension information.



Create New Extension

Extension options are divided into five categories:

- Basic Settings
- Media
- Features
- Specific Time
- Wave
- Follow me
- Advanced Settings

The configuration parameters are as follows.

General	
Extension	The extension number associated with the user.
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user.Note:The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.
Call Privileges	Assign permission level to the user. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal".Note: Users need to have the same level as or higher level than an outbound rule's privilege to make outbound calls using this rule.
SIP/IAX Password	Configure the password for the user. A random secure password will be automatically generated. It is recommended to use this password for security purposes.Note: The password should contain at least 10 characters, with at least one lower-case letter, one upper-case letter, one numeral, and one special character.The password must not contain an ascending or descending consecutive numeric sequence, or a consecutive sequence of the same number/letter.

Auth ID	Configure the authentication ID for the user. If not configured, the extension number will be used for authentication.
Voicemail	Configure Voicemail. There are three valid options, and the default option is "Enable Local Voicemail". Disable Voicemail: Disable Voicemail. Enable Local Voicemail: Enable voicemail for the user. Enable Remote Voicemail: Forward the notify message from the remote voicemail system for the user, and the local voicemail will be disabled. Note: Remote voicemail feature is used only for Infomatec (Brazil).
Voicemail Password	Configure voicemail password (digits only) for the user to access the voicemail box. A random numeric password is automatically generated. It is recommended to use the randomly generated password for security purposes.
Skip Voicemail Password Verification	When a user dials voicemail code, the password verification IVR is skipped. If enabled, this would allow one-button voicemail access. By default, this option is disabled.
Send Voicemail Email Notification	Configures whether to send emails to the extension's email address to notify of a new voicemail.
Attach Voicemail to Email	Configures whether to attach a voicemail audio file to the voicemail notification emails.Note: When set to "Default", the global settings in Basic Call Features > Voicemail > Voicemail Email Settings will be used.
Keep Voicemail after Emailing	Whether to keep the local voicemail recording after sending them. If set to "Default", the global settings will be used.Note: When set to "Default", the global settings in Basic Call Features > Voicemail > Voicemail Email Settings will be used.
Enable Keep-alive	If enabled, an empty SDP packet will be sent to the SIP server periodically to keep the NAT port open. The default setting is "No".
Keep-alive Frequency	Configure the Keep-alive interval (in seconds) to check if the host is up. The default setting is 60 seconds.
Enable SCA	If enabled, (1) Call Forward, Call Waiting, and Do Not Disturb settings will not work, (2) Concurrent Registrations can be set only to 1, and (3) Private numbers can be added in Advanced Call Features > SCA page.
Emergency CID Name	CallerID name that will be used for emergency calls and callbacks.
Disable This Extension	If selected, this extension will be disabled on the PBX.Note: The disabled extension still exists on the PBX but cannot be used on the end device.
Sync Contact	If enabled, this extension number will be displayed in the Wave contact, otherwise, it will not be displayed, and it cannot be found in the chat, but the user can still dial this number.
User Settings	
First Name	Configure the first name of the user. The first name can contain characters, letters, digits, and _.
Last Name	Configure the last name of the user. The last name can contain characters, letters, digits, and _.
Email Address	Fill in the Email address for the user. Voicemail will be sent to this Email address.
User/Wave Password?	Configure the password for user portal access. A random password is automatically generated. It is recommended to use the randomly generated password for security purposes.The password must respect the following criteria: At least one lower case letter At least one upper case letter At least one numeral At least one special character At least 10 total characters
User Portal/Wave Privileges	Change User Portal/Wave Privileges. Default: Setting this privilege will grant the standard Wave privileges. Wave Administrator: Setting this privilege will grant the user access to Management Portal on Wave.

Multi-Factor Authentication	When this option is enabled, the extension user will be prompted to configure multi-factor authentication on their account when they log in from the User Portal or Wave. The user can choose to use an OTP device or application, or they can use their email as a second factor of authentication.To ensure enabling Wave multi-factor authentication is successful, please refer to the minimum versions required below for Wave on each platform. Wave Web: 1.0.27.12 and later Wave Desktop: 1.27.10 and later Wave App for Android(TM): 1.0.27.18 and later Wave App for iOS(TM): 1.27.24 and later
Language	Select the voice prompt language to be used for this extension. The default setting is "Default" which is the selected voice prompt language under PBX Settings > Voice Prompt > Language Settings. The dropdown list shows all the currently available voice prompt languages on the PBX. To add more languages to the list, please download the voice prompt package by selecting "Check Prompt List" under PBX Settings > Voice Prompt > Language Settings.
Concurrent Registrations	The maximum endpoints which can be registered to this extension. For security concerns, the default value is 3.Note: The user can configure up to 10 registrations per extensions.
Mobile Phone Number	Configure the phone number for the extension, user can type the related star code for the phone number followed by the extension number to directly call this number.For example, the user can type *881000 to call the mobile number associated with extension 1000.
Department	Configure the user's department. The department can be configured in User Management > Address Book Management > Department Management. Job Title: The user's department position.
Contact Privileges	
Same as Department Contact Privileges	When enabled, The extension will inherit the same privilege attributed to the department it belongs to.
Contact View Privileges	Select the privileges regarding the contact view in SIP endpoints and Wave.

SIP Extension Configuration Parameters > Basic Settings

General	
NAT	Use NAT when the PBX is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is a one-way audio issue, usually it is related to NAT configuration or the Firewall's support of SIP and RTP ports. The default setting is enabled.
Enable Direct Media	By default, the PBX will route the media streams from SIP endpoints through itself. If this option is enabled, the PBX will attempt to redirect the RTP media streams to bypass the PBX and to go directly between caller and callee.Note: It is not always possible for the PBX to negotiate endpoint-to-endpoint media routing.
DTMF Mode	Select DTMF mode for the user to send DTMF. The default setting is "RFC4733". If "Info" is selected, the SIP INFO message will be used. If "Inband" is selected, a-law or u-law are required. When "Auto" is selected, RFC4733 will be used if offered, otherwise "Inband" will be used.Note: The default DTMF mode selected is RFC4733.
TEL URI	If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". The "User=Phone" parameter will be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel" will be used instead of "SIP" in the SIP request.
Alert-Info	When present in an INVITE request, the alert-Info header field specifies an alternative ring tone to the UAS.
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.

TURN Relay	Enable this option if the following are true:PBX is deployed on a private network.There are remote endpoints outside the PBX's network registering to it via its public IP address.The network's firewall is not configured for media port forwarding.Media NAT penetration is required.Once a TURN server is configured, media will be forwarded to it. This configuration does not affect endpoints that are registered via the PBX's RemoteConnect address.
Codec Preference	Select audio and video codec for the extension. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G,726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p, RTX and VP8.
QoS	
Jitter Buffer	Select the jitter buffer method.Disable: Jitter buffer will not be used.Fixed: Jitter buffer with a fixed size (equal to the value of "jitter buffer size")Adaptive: Jitter buffer with an adaptive size (no more than the value of "max jitter buffer").NetEQ: Dynamic jitter buffer via NetEQ.
Packet Loss Retransmission	Configure to enable Packet Loss Retransmission.NACKNACK+RTX(SSRC-GROUP)OFF
Video FEC	Check to enable Forward Error Correction (FEC) for Video.
Audio FEC	Check to enable Forward Error Correction (FEC) for Audio.
Silence Suppression	If enabled, the PBX will send CN packets for silence suppression after a successful CN negotiation in the SIP SDP. If the client endpoint's OPUS codec supports the reception of DTX packets, the PBX will send DTX packets instead.
FECC	Configure to enable Remote Camera Management.
RTP Encryption	
SRTP	Enable SRTP for the call. The default setting is disabled.Disabled Enabled and Enforced: SRTP will be necessary to transmit media traffic. If the IP phone of this extension has SRTP disabled, calls cannot be established. Optional: The PBX will negotiate whether to use SRTP or not. If the SIP endpoint has SRTP enabled, SRTP will be used. If it is disabled, SRTP will not be used.
SRTP Crypto Suite	SRTP encryption suite used by the PBX for outbound calls. Priority is based on order of configuration.The following encryption alogrithms can be used to encrypt an RTP stream.AES_CM_128_HMAC_SHA1_80 (This is the default algorithm used)AES_256_CM_HMAC_SHA1_80AEAD_AES_128_GCMAEAD_AES_256_GCM
ZRTP Encryption Mode	ZRTP, also known as Media Path Key Agreement for Secure RTP, is an encryption protocol which allows negotiating the encryption key for RTP traffic. ZRTP uses Diffie-Hellman exchange to establish an encrypted and secure connection between the PBX and the SIP endpoint. If the SIP endpoint has both SRTP and ZRTP enabled, ZRTP will always be prioritized.

SIP Extension Configuration Parameters > Media

CC Settings	
Enable CC	If enabled, the PBX will automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason. By default, it is disabled.
CC Mode	Two modes for Call Completion are supported: Normal: This extension is used as an ordinary extension.For Trunk: This extension is registered from a PBX.The default setting is "Normal".
Call Settings	
Call Duration Limit	Block calls for the configured duration.
Maximum Call Duration (s)	The maximum allowed duration of a call in seconds. Default value is 0 (no limit).
The Maximum Number of Call Lines	The maximum number of simultaneous calls that the extension can have. 0 indicates no limit.

Outgoing Call Frequency Limit	If enabled, if the number of outbound calls exceed the configured threshold within the specified period, further outbound will be not be allowed.
Period (m)	The time window for the configured outbound call frequency limit. Supported value range is 1 to 120 minutes. Default value is 1.
Max Number of Calls	The maximum allowed number of outbound calls within the specified period. Supported value range is 1 to 20. Default value is 5.
Ring Simultaneously	
Ring Simultaneously	Enable this option to have an external number ring simultaneously along with the extension. If a register trunk is used for outbound, the register number will be used to be displayed for the external number as the caller ID number.
External Number	Set the external number to ring simultaneously. '-' is the connection character that will be ignored. This field accepts only letters, numbers, and special characters + = * #.
Time Condition for Ring Simultaneously	Ring the external number simultaneously along with the extension based on this time condition.
Use callee DOD on FWD or RS	Use the DOD number when calls are being diverted/forwarded to external destinations or when ring simultaneous is configured.
Monitor Privilege Control	
Allowed to be Monitored	When disabled, this user's calls cannot be listened to by anyone. When enabled, this user's calls are allowed to be listened to through this extension or the opposite extension.
Call Montoring Allowlist	Add members from "Available Extensions" to "Selected Extensions" so that the selected extensions can spy on the used extension using feature code.
Allow Operator Panel Monitoring	Configure whether this extension can be monitored by the Operator Panel administrator.
Seamless transfer privilege control	
Allowed to seamless transfer	Any extensions on the PBX can perform a seamless transfer. When using the Pickup Incall feature, only extensions available on the "Selected Extensions" list can perform a seamless transfer to the edited extension.
PMS Remote Wakeup Whitelist	
Select the extensions that can set wakeup service for other extensions	Selected extensions can set a PMS wakeup service for this extension via feature code.
Other Settings	
Ring Timeout	Configure the number of seconds to ring the user before the call is forwarded to voicemail (voicemail is enabled) or hang up (voicemail is disabled). If not specified, the default ring timeout is 60 seconds on the PBX. The valid range is between 5 seconds and 600 seconds.Note: If the end point also has a ring timeout configured, the actual ring timeout used is the shortest time set by either device.
Auto Record	Enable automatic recording for the calls using this extension. The default setting is disabled. The recordings can be accessed under CDR > Recording Files.
Skip Trunk Auth	If set to "yes", users can skip entering the password when making outbound calls.If set to "By Time", users can skip entering the password when making outbound calls during the selected time condition.If set to "No", users will be asked to enter the password when making outbound calls.
Time Condition for Skip Trunk Auth	If 'Skip Trunk Auth' is set to 'By Time', select a time condition during which users can skip entering the password when making outbound calls.
Dial Trunk Password	Configure personal password when making outbound calls via the trunk.

Support Hot-Desking Mode	Check to enable Hot-Desking Mode on the extension. Hot-Desking allows using the same endpoint device and logs in using extension/password combination. This feature is used in scenarios where different users need to use the same endpoint device during a different time of the day for instance. If enabled, SIP Password will accept only alphabet characters and digits. Auth ID will be changed to the same as Extension.
Enable LDAP	If enabled, the extension will be added to the LDAP Phonebook PBX list. Default is enabled.
Use MOH as IVR ringback tone	If enabled, when the call to the extension is made through the IVR, the caller will hear MOH as a ringback tone instead of the regular ringback tone.
Music On Hold	Specify which Music On Hold class to suggest to the bridged channel when putting them on hold.
Max Number of Calls	Set the maximum number of outgoing calls in a period. The valide tange is from 1 to 20. The default value is 5.
Enable Auto-Answer Support	If enabled, the extension will support auto-answer when indicated by Call-info/Alert-info headers.
Call Waiting	Allows calls to the extension even when it is already in a call. This only works if the caller is directly dialing the extension. If disabled, the CC service will take effect only for unanswered and timeout calls.
Stop Ringing	If enabled, when the extension has concurrent registrations on multiple devices, upon incoming call or meeting invite ringing, if one end device rejects the call, the rest of the devices will also stop ringing. By default, it's disabled.
Email Missed Call Log	If enabled, the log of missed calls will be sent to the extension's configured email address.
Missed Call Type	If Email Missed Calls enabled, users can select the type of missed calls to be sent via email, the available types are:Default: All missed calls will be sent in email notifications.Missed Internal Call: Only missed local extension-to-extension calls will be sent in email notifications.Missed External Call: Only missed calls from trunks will be sent in email notifications.
Enable SCA (Share Call Appearance)	If enabled, (1) Call Forward, Call Waiting and Do Not Disturb settings will not work, (2) Concurrent Registrations can be set only to 1, and (3) Private numbers can be added in Advanced Call Features > SCA page.
Emergency CID	CallerID name and number that will be used when making emergency calls and receiving direct callbacks. If ELIN subnet mapping has been configured, and the extension is registered to a device in a mapped subnet, the configured ELIN will be used for CID number instead.
Language	Select voice prompt language for this extension. If set to "Default", the global setting for voice prompt language will be used. If a language is selected, the voice prompt will be in the corresponding language when calling this specific extension.

SIP Extension Configuration Parameters > Features

Specific Time	
Time Condition	Click to add Time Condition to configure a specific time for this extension.

SIP Extension Configuration Parameters > Specific Time

Normal	
Enable Wave	Enable Wave for the specific extension.
Allow Concurrent Logins from the Same Client Type	Enables/disables the ability to login to Wave from different sessions on the same type of client Note: This option is disabled by default.
Wave Welcome Email	Wave Welcome Email template.

Wave Permission Settings	Clicking the path will direct you to Wave Permission configuration.
Wave	
Download Link?	https://fw.gdms.cloud/wave/download/

SIP Extension Configuration Parameters > Wave

Follow Me	
Enable	Configure to enable or disable Follow Me for this user.
Skip Trunk Auth	If the outbound calls need to check the password, we should enable this option or enable the option "Skip Trunk Auth" of the Extension. Otherwise, this Follow Me cannot call out.
Music On Hold Class	Configure the Music On Hold class that the caller would hear while tracking the user.
Confirm When Answering	If enabled, call will need to be confirmed after answering.
Enable Destination	Configure to enable destination.
Default Destination	The call will be routed to this destination if no one in the Follow Me answers the call.
Use Callee DOD for Follow Me	Use the callee DOD number as CID if configured Follow Me numbers are external numbers.
Play Follow Me Prompt	If enabled, the Follow Me prompt tone will be played.
New Follow Me Number	Add a new Follow Me number which could be a "Local Extension" or an "External Number". The selected dial plan should have permissions to dial the defined external number.
Dialing Order	This is the order in which the Follow Me destinations will be dialed to reach the user.

SIP Extension Configuration Parameters > Follow Me

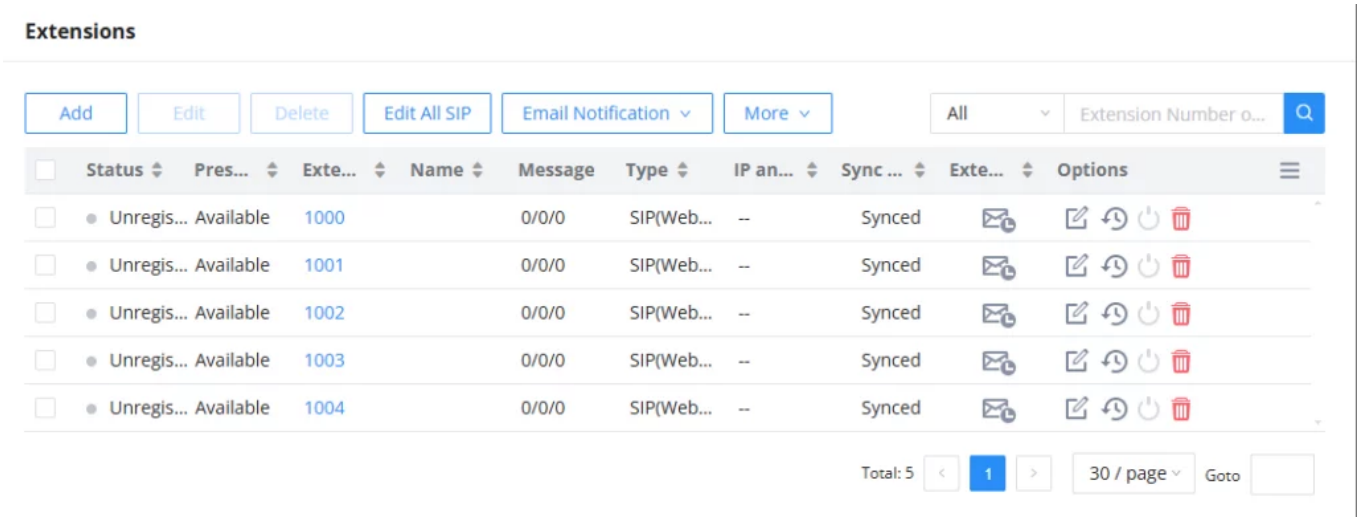
SIP Settings?	
Send PCPID Header	If enabled, this extension's SIP INVITE messages will contain the P-Called-Party-ID (PCPID) header if the callee is a SIP device.
Enable Auto-Answer	If enabled, the extension will support auto-answer when indicated by Call-info/Alert-info headers.
Enable Keep-alive	If enabled, the PBX will regularly send SIP OPTIONS to check if host device is online.
Keep-alive Frequency	Configure the keep-alive interval (in seconds) to check if the host is up.
TEL URI	If "Enabled" option is selected, TEL URI and Remove OBP from Route cannot be enabled at the same time. If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". A "User=Phone" parameter will then be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request.
ACL Policy	Access Control List manages the IP addresses that can register to this extension.Default: Follows Geo-IP Access ControlAllow Whitelist IP Address: Only whitelisted IP addresses in the configured network segments can register to this extension.Allow Whitelist IP Address + RemoteConnect: Registrations from whitelisted IP address network segments and RemoteConnect registrations to this extension will be allowed.
Fax	

Fax Mode	Configure fax mode. The following options are available:None: Disable fax support. This is the default option.Fax Detect: During a call, the fax signal from the user/trunk will be detected, and the received fax will be sent to the user's configured email address. If the destination does not have a configured email address, the fax will be sent to the Default Email Address configured in the Call Features > Fax/T.38 > Fax Settings page.
Fax to Email	If set to "Yes", the fax will be sent to the user-configured email address. If no user email address is found, the fax will be sent to the default email address configured in Fax/T.38 > Fax Settings.

SIP Extension Configuration Parameters > Advanced Settings

Search and Edit Extension

All the IPPBX extensions are listed under Web GUI→**Extension/Trunk→Extensions**, with status, Extension, CallerID Name, IP, and Port. Each extension has a checkbox for users to "Edit" or "Delete". Also, options "Edit" , "Reboot" and "Delete" are available per extension. Users can search for an extension by specifying the extension number to find an extension quickly.



Manage Extensions

- **Status**

Users can see the following icon for each extension to indicate the SIP status.

Green: Idle

Blue: Ringing

Yellow: In Use

Grey: Unavailable (the extension is not registered or disabled on the PBX)

- **Edit single extension**

Click on

to start editing the extension parameters.

- **Reset single extension**

Click on
A close up of a logo Description generated with high confidence
to reset the extension parameters to default (except concurrent registration).

Other settings will be restored to default in **Maintenance→User Management→User Information** except for username and permissions and delete the user voicemail prompt and voice messages.

This is the expected behavior when you reset an extension:

- All the data and configuration on the user side will be deleted. That includes user information, call history, call recordings, faxes, voice mails, meeting schedules, and recordings, as well as chat history. However, the data related to the user will be kept on the IPPBX side.
- The extension will be removed from group chats and the messages sent previously by the extension will be kept. However, only other users can search through those messages while the new user of the extension cannot.
- If the extension was in a meeting schedule, the meeting will still be present. The extension will be removed from the meeting and will not be notified about the meeting.

- **Reboot the user**

Click on to send NOTIFY reboot event to the device that has a IPPBX extension already registered. To successfully reboot the user, "Zero Config" needs to be enabled on the IPPBX Web GUI**→Device Management→Zero Config→Zero Config Settings**.

- **Delete single extension**

Click on to delete the extension. Or select the checkbox of the extension and then click on "Delete Selected Extensions".

This is the expected behavior when you delete an extension:

- The system will delete all the data of the extension except the CDR and meetings records. All the data on the user side will be erased.
- The extension will be removed from group chats and the messages sent previously by the extension will be kept. However, only other users can search through those messages while the new user of the extension cannot.
- If the extension was in a meeting schedule, the meeting will still be present. The extension will be removed from the meeting and will not be notified about the meeting.

- **Modify selected extensions**

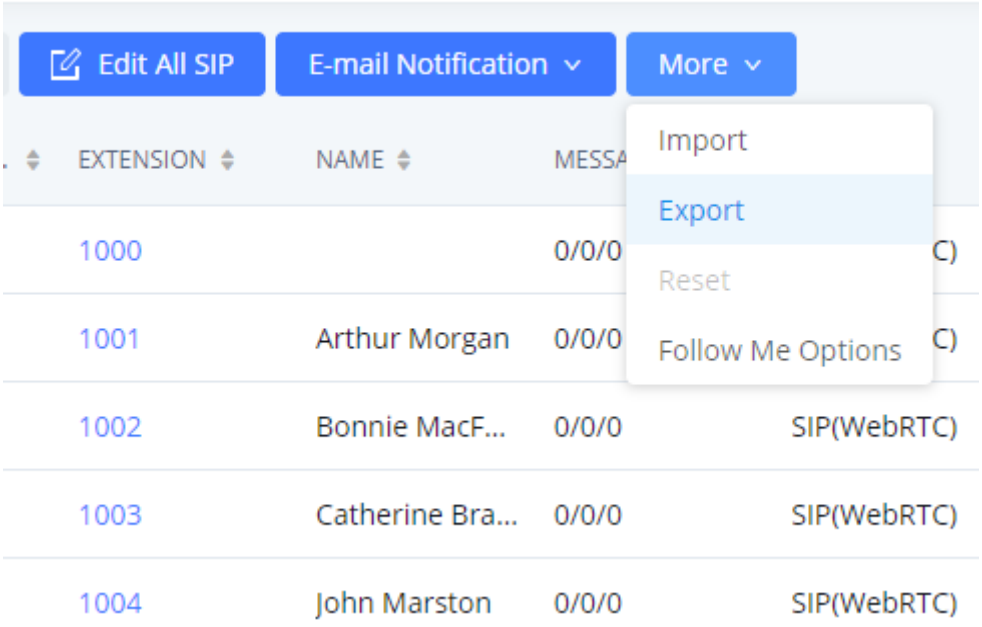
Select the checkbox for the extension(s). Then click on "Edit" to edit the extensions in a batch.

- **Delete selected extensions**

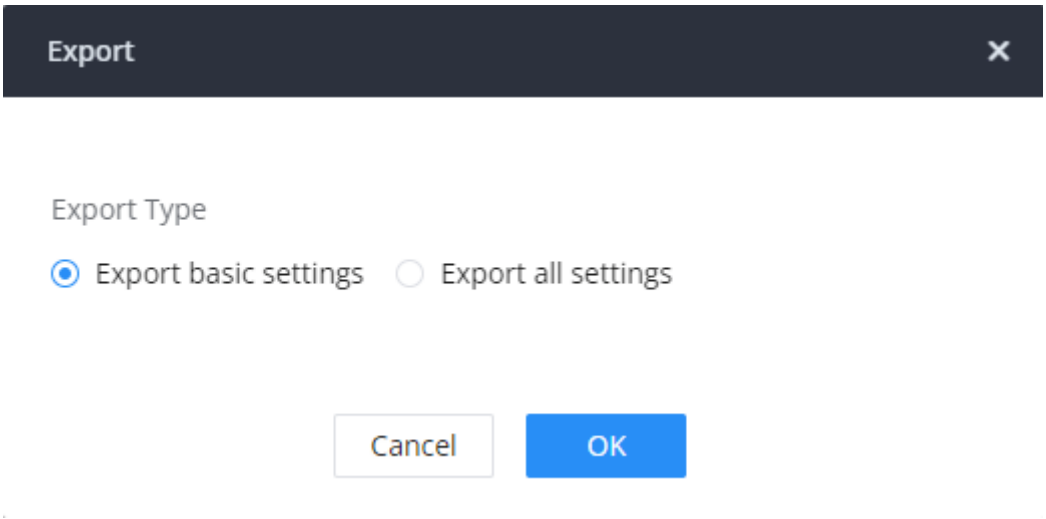
Select the checkbox for the extension(s). Then click on "Delete " to delete the extension(s).

Export Extensions

The extensions configured on the IPPBX can be exported to a CSV format file. Click on the "Export Extensions" button and select technology in the prompt below.



Export Extensions



Export Basic Settings

Export Basic Information includes:

- Extension
- CallerID Number
- Privilege
- SIP Password
- AuthID
- Voicemail
- Voicemail Password
- Sync Contact
- First Name
- Last Name
- Email Address
- User/Wave Password

If importing extensions with no values for settings, the following will occur:

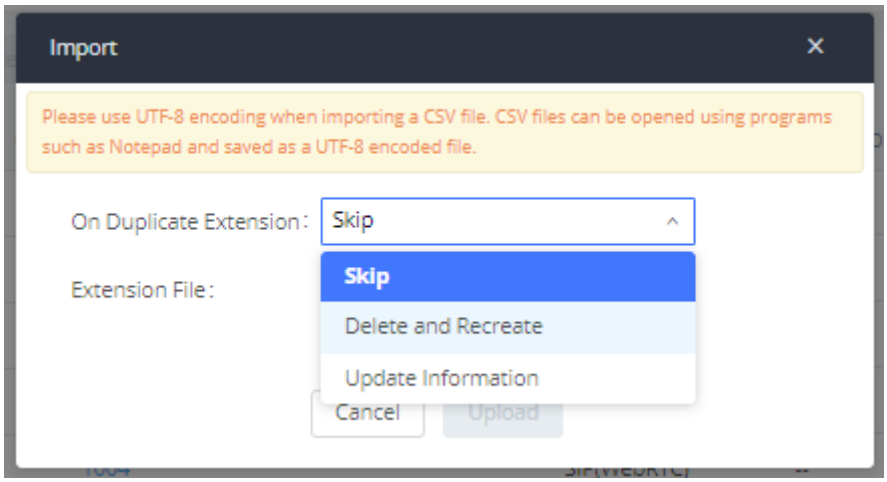
- If importing new extensions, or if **Replace** is selected as the duplicate import option, the default values for those settings will be used.
- If **Update** is selected as the duplicate import option, no changes will be made to the existing settings.

The exported CSV file can serve as a template for users to fill in desired extension information to be imported to the IPPBX.

Import Extensions

The capability to import extensions to the IPPBX provides users the flexibility to batch-add extensions with similar or different configurations quickly into the PBX system.

1. Export the extension CSV file from the IPPBX by clicking on the "Export Extensions" button.
2. Fill up the extension information you would like in the exported CSV template.
3. Click on the "Import Extensions" button. The following dialog will be prompted.



Import Extensions

4. Select the option in "On Duplicate Extension" to define how the duplicate extension(s) in the imported CSV file should be treated by the PBX.

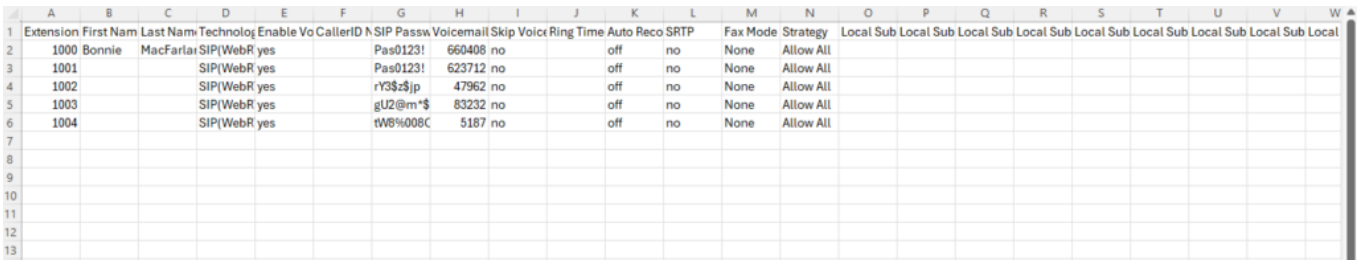
- **Skip:** Duplicate extensions in the CSV file will be skipped. The PBX will keep the current extension information as previously configured without change.

- **Delete and Recreate:** The current extension previously configured will be deleted and the duplicate extension in the CSV file will be loaded to the PBX.

- **Update Information:** The current extension previously configured in the PBX will be kept. However, if the duplicate extension in the CSV file has a different configuration for any options, it will override the configuration for those options in the extension.

5. Click on "Choose file to upload" to select a CSV file from a local directory on the PC.
6. Click on "Apply Changes" to apply the imported file on the IPPBX.

Example of a file to import:



	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	Q	R	S	T	U	V	W
1	Extension	First Nam	Last Nam	Technolog	Enable Vo	CallerID N	SIP Passw	Voicemail	Skip Voic	Ring Time	Auto Reco	SRTP	Fax Mode	Strategy	Local Sub	Local Sub	Local Sub	Local Sub	Local Sub	Local Sub	Local Sub	Local Sub	Local
2	1000	Bonnie	MacFarla	SIP(WebR	yes		Pas0123!	660408	no		off	no	None	Allow All									
3	1001			SIP(WebR	yes		Pas0123!	623712	no		off	no	None	Allow All									
4	1002			SIP(WebR	yes		rY3$z$jp	47962	no		off	no	None	Allow All									
5	1003			SIP(WebR	yes		gU2@m*$	83232	no		off	no	None	Allow All									
6	1004			SIP(WebR	yes		tW8%008(	5187	no		off	no	None	Allow All									
7																							
8																							
9																							
10																							
11																							
12																							
13																							

Import File

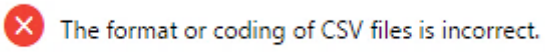
Field	Supported Values
Extension	Digits
Technology	SIP/SIP(WebRTC)
Enable Voicemail	yes/no/remote
CallerID Number	Digits

SIP Password	Alphanumeric characters
Voicemail Password	Digits
Skip Voicemail Password Verification	yes/no
Ring Timeout	Empty/ 3 to 600 (in second)
SRTP	yes/no
Skip Trunk Auth	yes/no/bytime
Codec Preference	PCMU,PCMA,GSM,G.726,G.722,G.729,H.264,ILBC,AAL2-G.726-32,ADPCM,G.723,H.263,H.263p,vp8,opus
Permission	Internal/Local/National/International
DTMF Mode	RFC4733/info/inband/auto
Insecure	Port
Enable Keep-alive	Yes/no
Keep-alive Frequency	Value from 1-3600
AuthID	Alphanumeric value without special characters
TEL URI	Disabled/user=phone/enabled
Call Forward Busy	Digits
Call Forward No Answer	Digits
Call Forward Unconditional	Digits
Support Hot-Desking Mode	Yes/no
Dial Trunk Password	Digits
Disable This Extension	Yes/no
CFU Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
CFN Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
CFB Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Music On Hold	Default/ringbacktone_default
CC Agent Policy	If CC is disabled use: never If CC is set to normal use: generic If CC is set to trunk use: native
CC Monitor Policy	Generic/never
CCBS Available Timer	3600/4800
CCNR Available Timer	3600/7200
CC Offer Timer	60/120
CC Max Agents	Value from 1-999
CC Max Monitors	Value from 1-999
Ring simultaneously	Yes/no
External Number	Digits
Time Condition for Ring Simultaneously	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Time Condition for Skip Trunk Auth	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Enable LDAP	Yes/no

Enable T.38 UDPTL	Yes/no
Max Contacts	Values from 1-10
Enable Wave	Yes/no
Alert-Info	None/Ring 1/Ring2/Ring3/Ring 4/Ring 5/Ring 6/Ring 7/ Ring 8/Ring 9/Ring 10/bellcore-dr1/bellcore-dr2/ bellcore-dr3/ bellcore-dr4/ bellcore-dr5/custom
Do Not Disturb	Yes/no
DND Time Condition	All time/Office time/out of office time/holiday/out of holiday/out of office time or holiday/specific time
Custom Auto answer	Yes/no
Do Not Disturb Whitelist	Empty/digits
User Password	Alphanumeric characters.
First Name	Alphanumeric without special characters.
Last Name	Alphanumeric without special characters.
Email Address	Email address
Language	Default/en/zh
Phone Number	Digits
Call-Barging Monitor	Extensions allowed to call barging
Seamless Transfer Members	Extensions allowed to seamless transfer

SIP extensions Imported File Example

The CSV file should contain all the above fields, if one of them is missing or empty, the IPPBX will display the following error message for missing fields.

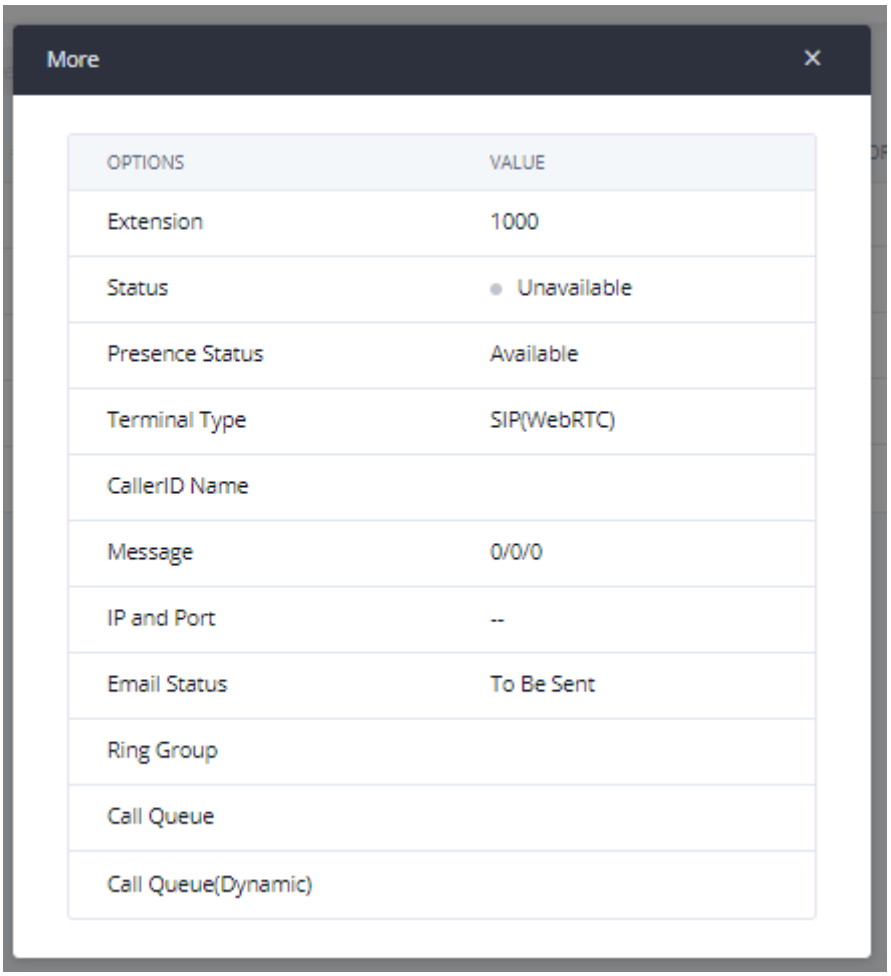


Import Error

Extension Details

Users can click on an extension number in the Extensions list page and quickly view information about it such as:

- **Extension**: This shows the Extension number.
- **Status**: This shows the status of the extension.
- **Presence status**: Indicates the Presence Status of this extension.
- **Terminal Type**: This shows the type of the terminal using this extension
- **Caller ID Name**: Reveals the Caller ID Name configured on the extension.
- **Messages**: Shows the messages' stats.
- **IP and Port**: The IP address and the ports of the device using the extension.
- **Email status**: Show the Email status (sent, to be sent...etc.).
- **Ring Group**: Indicates the ring groups that this extension belongs to.
- **Call Queue**: Indicates the Cal Queues that this extension belongs to.
- **Call Queue (Dynamic)**: Indicates the Call Queues that this extension belongs to as a dynamic agent.

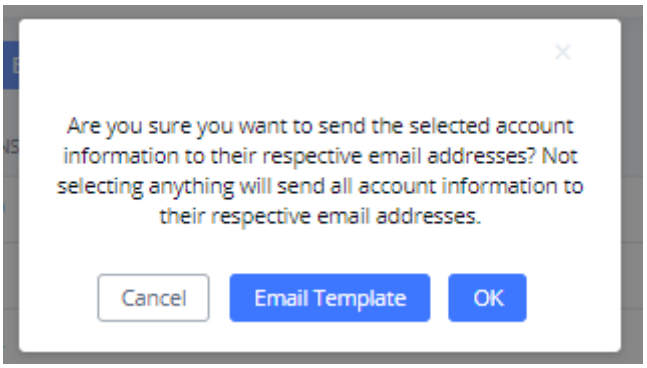


Extension Details

E-mail Notification

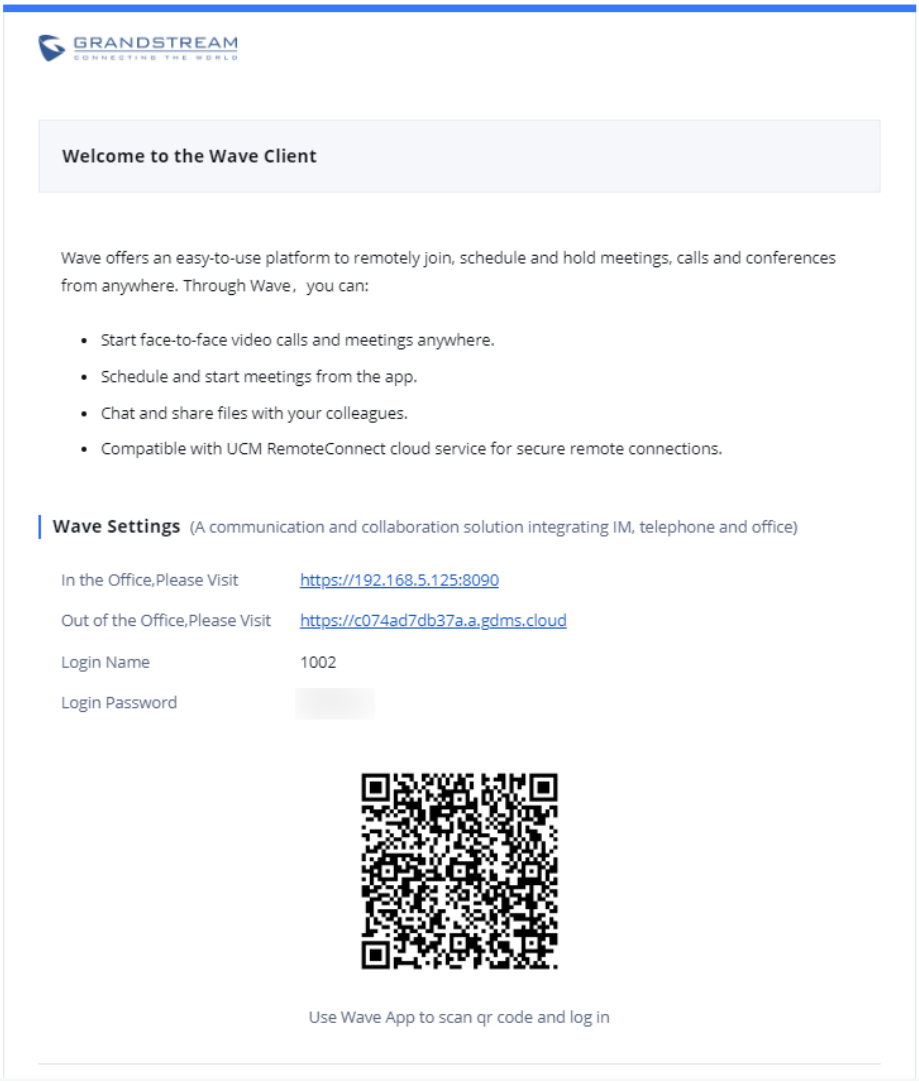
Once the extensions are created with Email addresses, the PBX administrator can click on the button "E-mail Notification" to send the account registration and configuration information to the user. Please make sure the Email setting under Web GUI→**System Settings→Email Settings** is properly configured and tested on the IPPBX before using "E-mail Notification".

When clicking on "More" > "E-mail Notification" button, the following message will be prompted on the web page. Click on OK to confirm sending the account information to all users' Email addresses.



E mail Notification Prompt Information

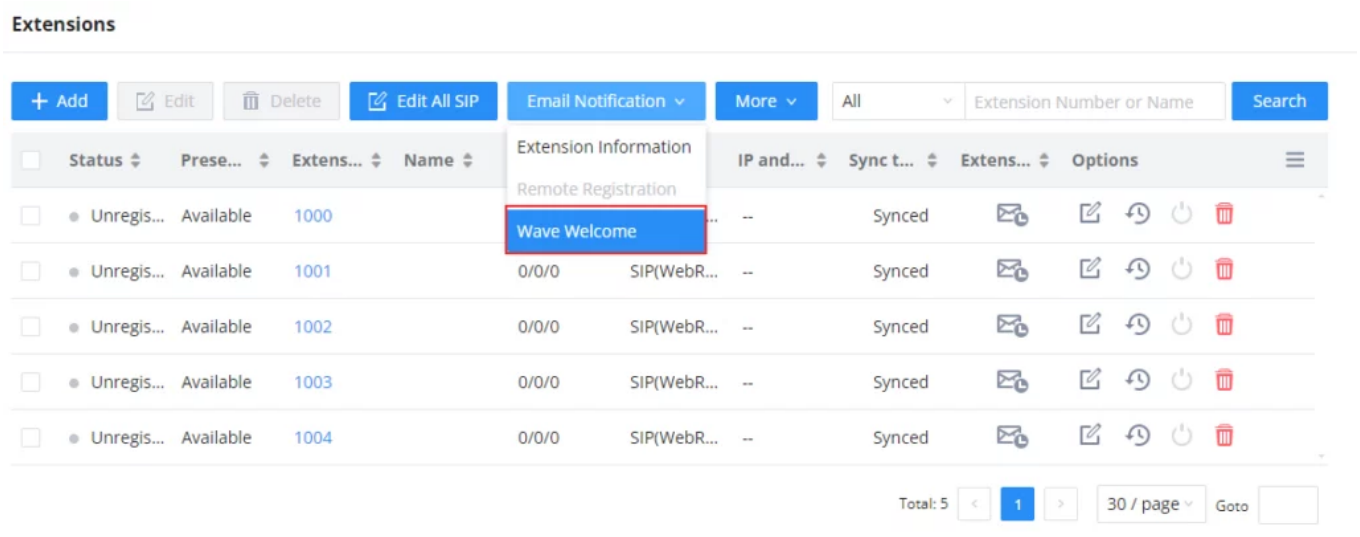
The user will receive an Email including account registration information as well as the Wave Settings with the QR code:



Wave Settings and QR Code

For security and confidentiality reasons, it is highly advisable for the user to change the Wave login extension after the first time log in.

The PBX admin can also send "Extension Information" mail and "Wave Welcome" mail as the figure below shows



Send Email Notification

SMS Message Support

The IPPBX provides built-in SIP SMS message support. For SIP end devices such as Grandstream GXP or GXV phones that support SIP messages, after a IPPBX account is registered on the end device, the user can send and receive SMS messages. Please refer to the end device documentation on how to send and receive SMS messages.

Extension Groups

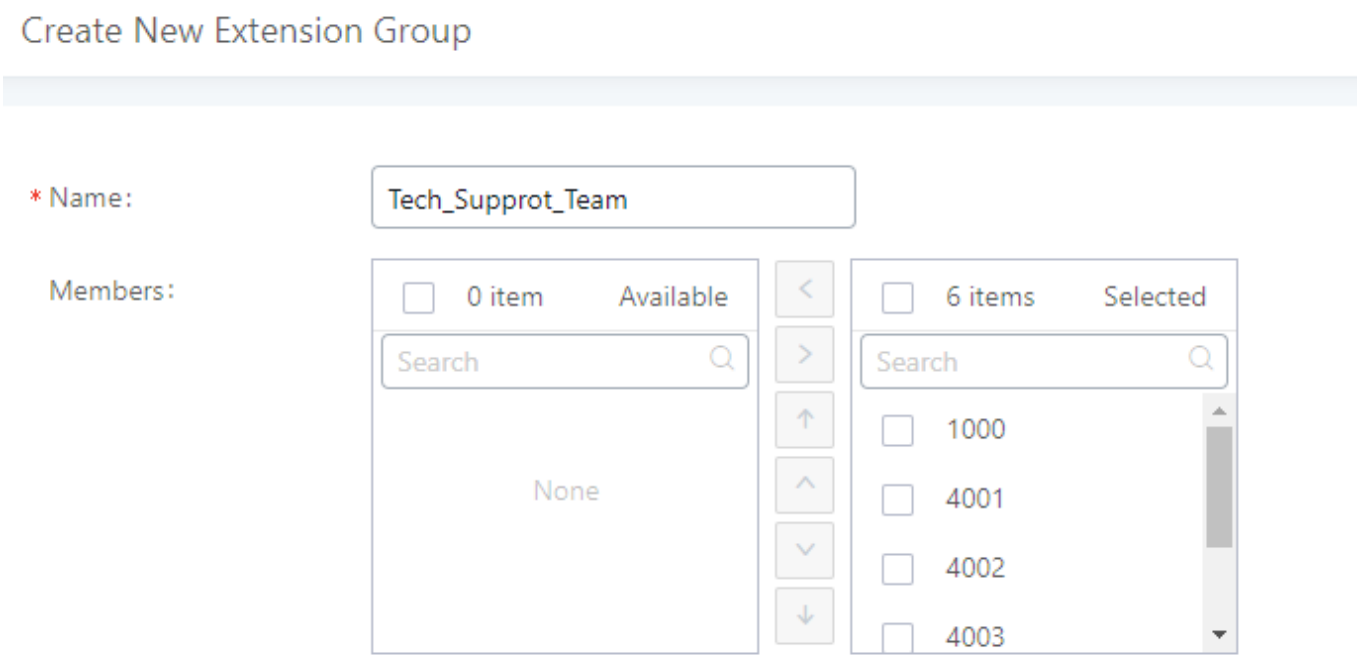
The IPPBX extension group feature allows users to assign and categorize extensions in different groups to better manage the configurations on the IPPBX. For example, when configuring the "Enable Source Caller ID Whitelist", users could select a group instead of each person's extension to assign. This feature simplifies the configuration process and helps manage and categorize the extensions for a business environment.

Configure Extension Groups

Extension groups can be configured via Web GUI→**Extension/Trunk→Extension Groups**.

- Click on to create a new extension group.
- Click on to edit the extension group.
- Click on to delete the extension group.

Select extensions from the list on the left side to the right side.

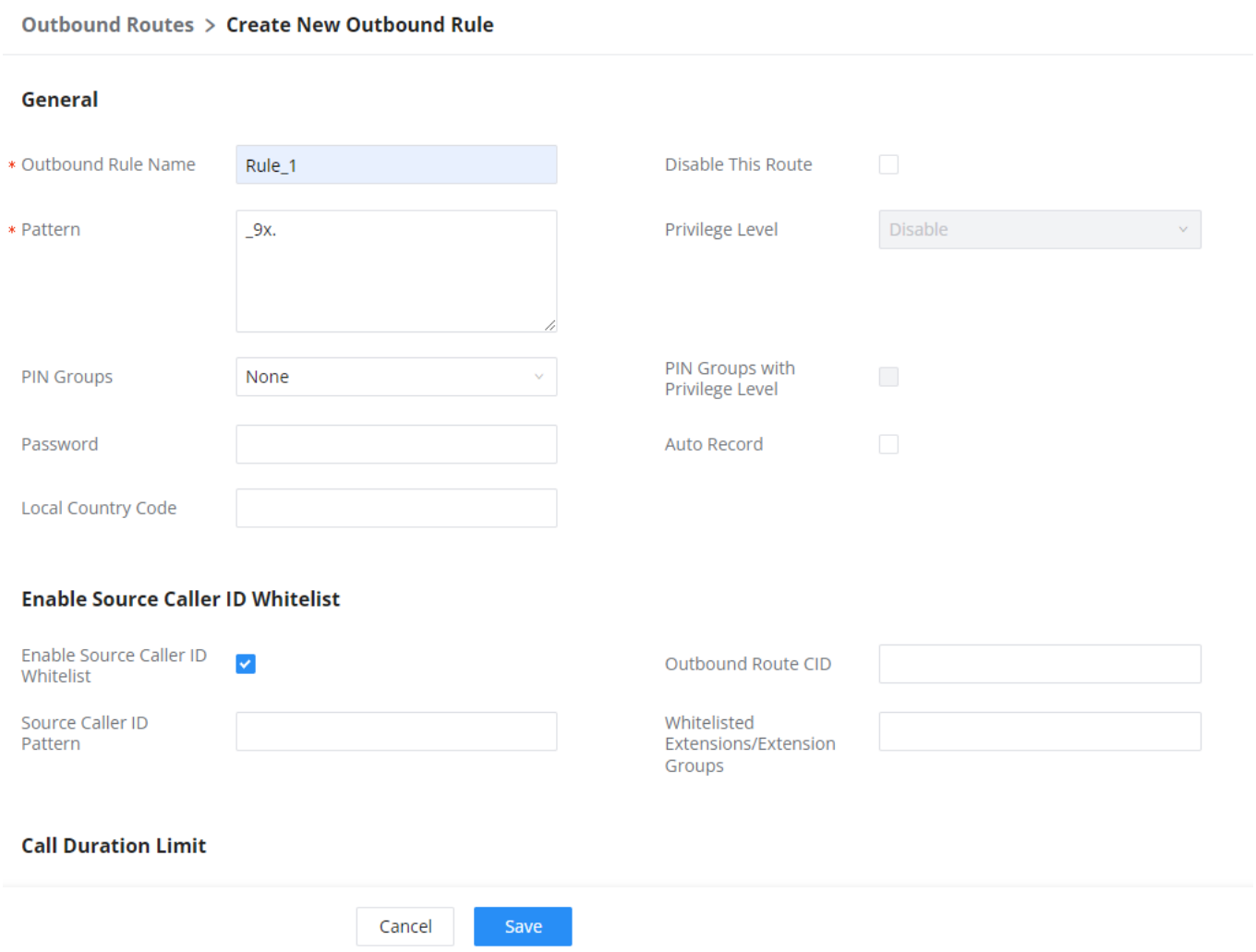


Edit Extension Group

Click on to change the ringing priority of the members selected on the group.

Using Extension Groups

Here is an example where the extension group can be used. Go to Web GUI→**Extension/Trunk→ Outbound Routes** and select "Enable Source Caller ID Whitelist". Both single extensions and extension groups will show up for users to select.



Select Extension Group in Outbound Route

VoIP Trunks

In this section, the user can configure VoIP trunks to establish the connection with an ITSP or a remote PBX over Internet. The user can create a peer trunk, register trunk, or account SIP trunk.

Please note that GCC60XX Series support creating up to 200 SIP trunks.

VoIP Trunk Configuration

VoIP trunks can be configured in the PBX under Web GUI→**Extension/Trunk→VoIP Trunks**. Once created, the VoIP trunks will be listed with the Provider Name, Type, Hostname/IP, Username, and Options to edit/detect the trunk.

- Click on "Add SIP Trunk" to add a new VoIP trunk.
- Click on to configure detailed parameters for the VoIP trunk.
- Click on to configure Direct Outward Dialing (DOD) for the SIP Trunk.
- Click on to start LDAP Sync.
- Click on to delete the VoIP trunk.

The VoIP trunk options are listed in the table below.

Disable This Trunk	Check this box to disable this trunk.

Type	Select the VoIP trunk type. Peer SIP Trunk: A direct IP-to-IP connection between the PBX and another SIP server or device, without requiring registration. Register SIP Trunk: A trunk that requires the PBX to register with the SIP server or provider using credentials (username and password). Account SIP Trunk: A trunk where the PBX acts as the registrar, allowing remote devices or endpoints to register with it.
Provider Name	Configure a unique label (up to 64 characters) to identify this trunk when listed in outbound rules, inbound rules, etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.
Dedicated VLAN	After selecting the corresponding VLAN, the traffic related to the relay will go through the VLAN interface.
Transport	Select the transport protocol to use. UDP: if selected, then the option Enabe UDP should be checked, under PBX Settings > SIP Settings > Transport Protocol. TCP: If selected, then the option TCP Enable should be checked under PBX Settings > SIP Settings > Transport Protocol. TLS: The default Transport protocol.
Trunk Mode	Set the trunk mode for incoming calls. In certain scenarios, service providers do not include a domain in "To" SIP header. In other scenarios, the service providers do not accept SIP INVITE messages from a different port than 5060. The trunk mode options allow to resolve such issues. DID Access: When a domain is not included in "To" SIP header, the user can configure a DID which will be used to verify incoming calls. Port Access: Choose this option to allow outbound SIP traffic to be sent from port 5060. Choosing this option will change the port used to receive SIP requests for this specific trunk to 6040, this should taken into consideration when interconnecting two PBXs Note: This option is available only for Register Trunk and Peer Trunk when using UDP as transport protocol.
DID Number	Enter the DID number which will be included in the "To" SIP header. This option is mandatory when Trunk Mode is set to DID Access.
Server Address	Use the indicated address to register/peer trunks when configuring other PBXs with the CloudUCM.
Keep Original CID	Keep the CID from the inbound call when dialing out. This setting will override the "Keep Trunk CID" option. Please make sure that the peer PBX at the other side supports to match user entry using the "username" field from the authentication line.
Keep Trunk CID	If enabled, the trunk CID will not be overridden by the extension's CID when the extension has CID configured. The default setting is "No".
NAT	Enable this setting if the PBX is using public IP and communicating with devices behind NAT. Note 1: This setting will overwrite the Contact header of received messages, which may affect the ability to establish calls when behind NAT. Consider changing setttings in PBX Settings > SIP Settings > NAT instead.Note 2: If one is experiencing one-way audio issues, please check the NAT configuration and SIP/RTP ports in the firewall.
TEL URI	If "Enabled" option is selected, TEL URI and Remove OBP from Route cannot be enabled at the same time. If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". A "User=Phone" parameter will then be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request.
Need Registration	Defines Whether to register the trunk on the external server. Enabled by default. Note: This option appears when the Type is set to Register Trunk.
Allow outgoing calls if registration fails	Disable to block outgoing calls if registration fails. If "Need Registration" option is disabled, this setting will be ignored. This option is enabled by default. Note: This option appears when the Type is set to Register Trunk.

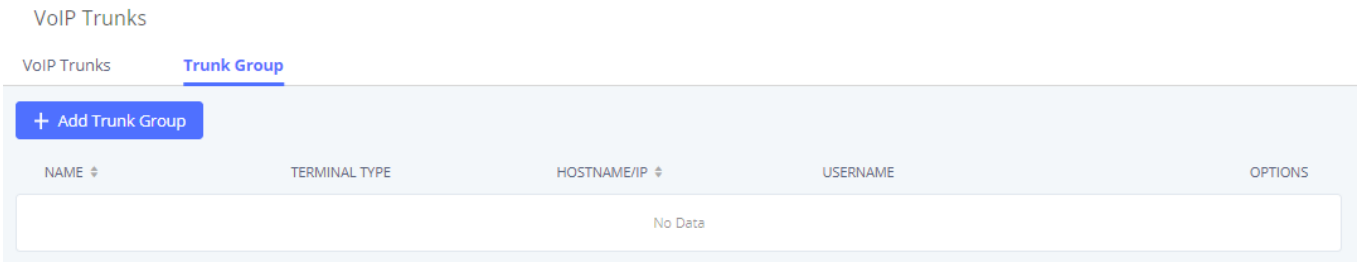
Caller ID Number	Configure the Caller ID. This is the number that the trunk will try to use when making outbound calls. For some providers, it might not be possible to set the CallerID with this option and this option will be ignored.Important Note: When making outgoing calls, the following priority order rule will be used to determine which CallerID will be set before sending out the call: From the user (Register Trunk Only) ? CID from inbound call (Keep Original CID Enabled) ? Trunk Username/CallerID (Keep Trunk CID Enabled) ? DOD ? Extension CallerID Number ? Trunk Username/CallerID (Keep Trunk CID Disabled) ? Global Outbound CID.
CallerID Name	Configure the new name of the caller when the extension has no CallerID Name configured.
Username	The number or username used for registration and authentication with the service provider. Note: You can configure this option for "Account SIP Trunk" and "Register SIP Trunk only"
Password	The password used for registration and authentication with the service provider. Note: You can configure this option for "Account SIP Trunk" and "Register SIP Trunk only"
Auth ID	Enter the Authentication ID for the "Register SIP Trunk" type.
AuthTrunk	If enabled, the PBX will send a 401 response to the incoming call to authenticate the trunk.
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded. Note: the recording functionality is not available on the startup plan.
Direct Callback	Allows external numbers the option to get directed to the extension that last called them.For Example, User 2002 has dialed external number 061234575 but they were not reachable, once they have received missed call notification on their phone, they would mostly call back the number, if the option "Direct Callback" is enabled then they will be directly bridged to user 2002 regardless of the configured inbound destination.
Domain Connection Mode	If enabled, the following options will be automatically configured: TLS transport, From Domain, Enable Heartbeat Detection and ICE Support. Please ensure that the trunk host name is a GDMS-assigned address and supports TLS.
Monitor Concurrent Calls	If enabled and when the number of concurrent calls exceeds any trunk's configured concurrent call thresholds, an alarm notification will be generated. Note: Please make sure the system alert event "Trunk Concurrent Calls" is enabled.
Concurrent Call Threshold	Threshold of all incoming and outgoing concurrent calls through this trunk.
Outgoing Concurrent Calls Threshold	Threshold of all outgoing concurrent calls passing through this trunk.
Incoming Concurrent Calls Threshold	Threshold of all incoming concurrent calls passing through this trunk.
Total Time Limit For Outbound Calls	
Enable Total Time Limit For Outgoing Calls	When this setting is activated, the user can set a time limit before calls cannot be initiated through this trunk
Period	This setting defines how long until the time allowed for outgoing calls is reset. Monthly: The time allowed will reset every month. Quarterly: The time allowed will reset every 3 months. Example: If the time limit has been set to 4320 minutes, the allowed time will always revert back to 4320 after a month or 3 month based on the period configured.
Total Time	Total time allowed in minutes.
Advanced Settings	
Codec Preference	Select audio and video codec for the VoIP trunk. The available codecs are: PCMU, PCMA, GSM, AAL2-G.726-32, G.726, G.722, G.729, G.723, iLBC, ADPCM, H.264, H.263, H.263p and VP8.
Packet Loss Retransmission	Configure to enable Packet Loss Retransmission.
Audio FEC	Configure to enable Forward Error Correction (FEC) for audio.

Video FEC	Configure to enable Forward Error Correction (FEC) for video.
ICE Support	Toggles ICE support. For peer trunks, ICE support will need to be enabled on the other end.
FECC	Configure to enable Far-end Camera Control
Silence Suppression	If enabled, the PBX will send CN packets for silence suppression after a successful CN negotiation in the SIP SDP. If the client endpoint's OPUS codec supports the reception of DTX packets, the PBX will send DTX packets instead.
SRTP	Enable SRTP for the VoIP trunk. The default setting is "No".
SRTP Crypto Suite	SRTP encryption suite used by PBX for outbound calls. Priority is based on order of configuration.
ZRTP Encryption Mode	If disabled, the PBX will not support ZRTP encryption. Otherwise, ZRTP will be supported, and if the registered endpoint supports both ZRTP and SRTP, ZRTP will be used first.
IPVT Mode	Similar to Enable Direct Media. The PBX will attempt to redirect the RTP media streams to bypass the PBX and to go directly between caller and callee. Primarily for use with trunks to IPVT.
Enable T.38 UDPTL	Enable or disable T.38 UDPTL support.
Include Special Attributes	If enabled, this trunk's SIP SDP will contain ssrc/msid/mid/as/tias/record attributes. These attributes may cause incompatibility when connecting to other devices and services.
Send PPI Header	If checked, the invite message sent to trunks will contain PPI (P-Preferred-Identity) Header.
Send PAI Header	If checked, the INVITE, 18x and 200 SIP messages sent to trunks will contain P-Asserted-Identity (PAI) header. It is not possible to send both PPI and PAI headers. If both Send PAI Header and Passthrough PAI Header are enabled, the following will occur: On incoming calls, the Passthrough PAI Header value will be preferred for this PBX's 18x and 200 SIP messages to the caller. On outbound calls, the Send PAI Header value will be preferred for this PBX's INVITE SIP message to the callee.
Passthrough PAI Header	If enabled and "Send PAI Header" is disabled, PAI headers will be preserved as calls pass through the PBX.
Send PANI Header	If checked, the INVITE sent to the trunk will contain P-Access-Network-Info header.
Send Anonymous	If checked, the "From" header in outgoing INVITE message will be set to anonymous.
DID Mode	Configure to obtain the destination ID of an incoming SIP call from SIP Request-line or To header.
DTMF Mode	Configures the mode for sending DTMF. RFC4733 (default): DTMF is transmitted as audio in the RTP stream but is encoded separately from the audio stream. Backward-compatible with RFC2833. Inband: DTMF is transmitted as audio and is included in the audio stream. Requires alaw/ulaw codecs.List Item 2 Info: DTMF is transmitted separely from the media streams. RFC4733_info: DTMF is transmitted through both RFC4733 and SIP INFO. Auto: DTMF mode will be negotiated with the remote peer, only supports RFC4733 and inband. RFC4733 will be used by default unless the remote peer does not indicate support.
Enable Heartbeat Detection	If enabled, the PBX will regularly send SIP OPTIONS to check if the device is online.
Max Outgoing Calls	The number of current outgoing calls over the trunk at the same time. The default value 0 means no limit.
Max Incoming Calls	The max allowed number of concurrent incoming calls through the trunk. Default is 0 (no limit).
Sync LDAP Enable	Automatically sync local LDAP phonebooks to a remote peer (SIP peer trunk only). To ensure successful syncing, the remote peer must also enable this service and set the same password as the local PBX. Port 873 is used by default.

STIR/SHAKEN	Block Spam Calls. Disabled: Disable STIR/SHAKEN. Outgoing Attest: Enable STIR/SHAKEN attestation for outgoing calls. Incoming Verify: Enable STIR/SHAKEN verification for incoming calls. Both: Enable STIR/SHAKEN for both outgoing and incoming calls.
Enable CC	Check this box to allow the system to automatically alert this extension when a called party is available, given that a previous call to that party failed for some reason.

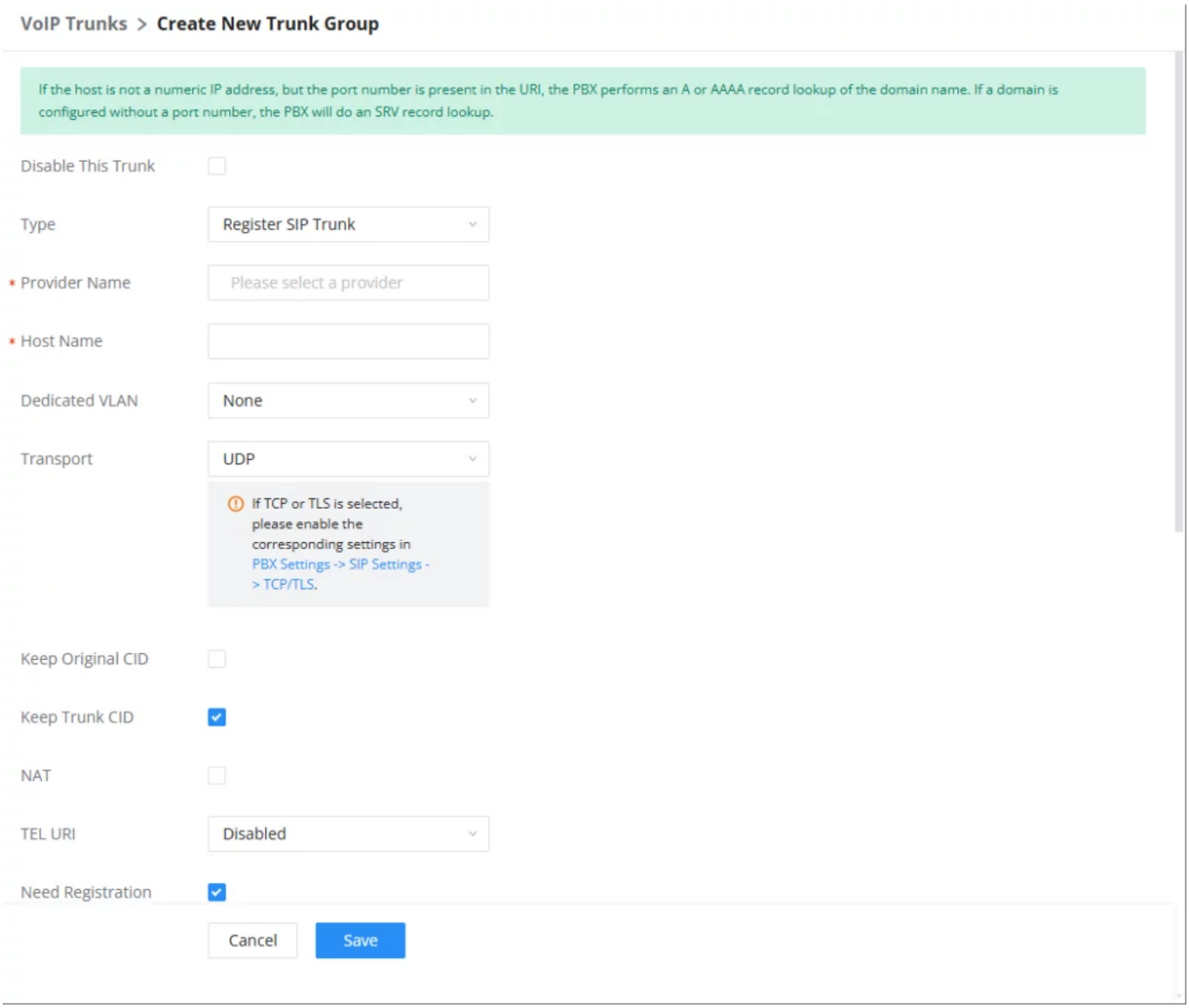
Trunk Group

Users can create VoIP Trunk Groups to register and easily apply the same settings on multiple accounts within the same SIP server. This can drastically reduce the amount of time needed to manage accounts for the same server and improve the overall cleanliness of the web UI.



Trunk Group

Once creating the new trunk group and configuring the SIP settings, users can add multiple accounts within the configured SIP server by pressing the button and configuring the username, password, and authentication ID fields.



Trunk Group Configuration

Type	Register Trunk
Provider Name	Configure a unique label to identify the trunk when listed in outbound rules and incoming rules.
Host Name	Enter the IP address or hostname of the VoIP provider's server.
Transport	Configure the SIP Transport method. Using TCP requires local TCP support; using TLS requires local TLS support.
Keep Original CID	Keep CID from the inbound call when dialing out even if option "Keep Trunk CID" is enabled. Please make sure the peer PBX at the other end supports matching user entry using the "username" field from the authentication line.
Keep Trunk CID	Always use trunk CID if specified even if extension has DOD number or CID configured.
NAT	Enable this setting if the IPPBX is using public IP and communicating with devices behind NAT. Note 1: This setting will overwrite the Contact header of received messages, which may affect the ability to establish calls when behind NAT. Consider changing settings in PBX Settings ? SIP Settings ? NAT instead.
Disable This Trunk	Check this box to disable this trunk

TEL URI	if "Enabled" option is selected, TEL URI and remove OBP from Route cannot be enabled at the same time. If the phone has an assigned PSTN telephone number, this field should be set to "User=Phone". A "User=Phone" parameter will the be attached to the Request-Line and "TO" header in the SIP request to indicate the E.164 number. If set to "Enable", "Tel:" will be used instead of "SIP:" in the SIP request.
Need Registration	Whether to register on the external server.
Allow outgoing calls if registration fails	Uncheck to block outgoing cakks if registration fails. If "Need Registration" option is unchecked, this settting will be ignored.
CallerID Name	To display the caller ID name of the trunk, you must configure the caller ID number of the trunk.
Trunk Registration Number	The number used to register with the provider server, and the VoIP provider will authenticate the number based on the trunk registration number.
Line Selection Strategy	Linear: Select lines in list order and make Outbound calls. Round Robin: Rotary line selection with memory and making Outboun calls.
AuthTrunk	If enabled, the IPPBX will send a 401 response to the incming call to authenticate the trunk.
Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Direct Callback	Allows external numbers the option to get directed to the extension that last called them.
RemoteConnect Mode	If enabled, RemoteConnect-related options will be automatically configured. Please confirm the trunk has a GDMS-assigned address or supports TLS.
Monitor Concurrent Calls	If enabled, the number of concurrent calls on this trunk will be monitored. If the "Trunk Concurrent Calls" system alert is enabled, alert notifications will be generated if the number of concurrent calls exceeds this trunk's configured concurrent call thresholds.
Concurrent Call Threshold	Threshold of all incoming and outgoing concurrent calls in this trunk.
Outgoing Concurrent Call Threshold	Threshold of all outgoing concurrent calls passing through this trunk.
Incoming Concurrent Call Threshold	Threshold of all incoming concurrent calls passing through this trunk.
Enable Total Time Limit For Outbound Calls	If enabled, a limit will be placed on the cumulative duration of outbound calls within a specific period. Once this limit has been reached, further outbound calls from this trunk will not be allowed.

Direct Outward Dialing (DOD)

The IPPBX provides Direct Outward Dialing (DOD), which is a service of a local phone company (or local exchange carrier) that allows subscribers within a company's PBX system to connect to outside lines directly.

Example of how DOD is used:

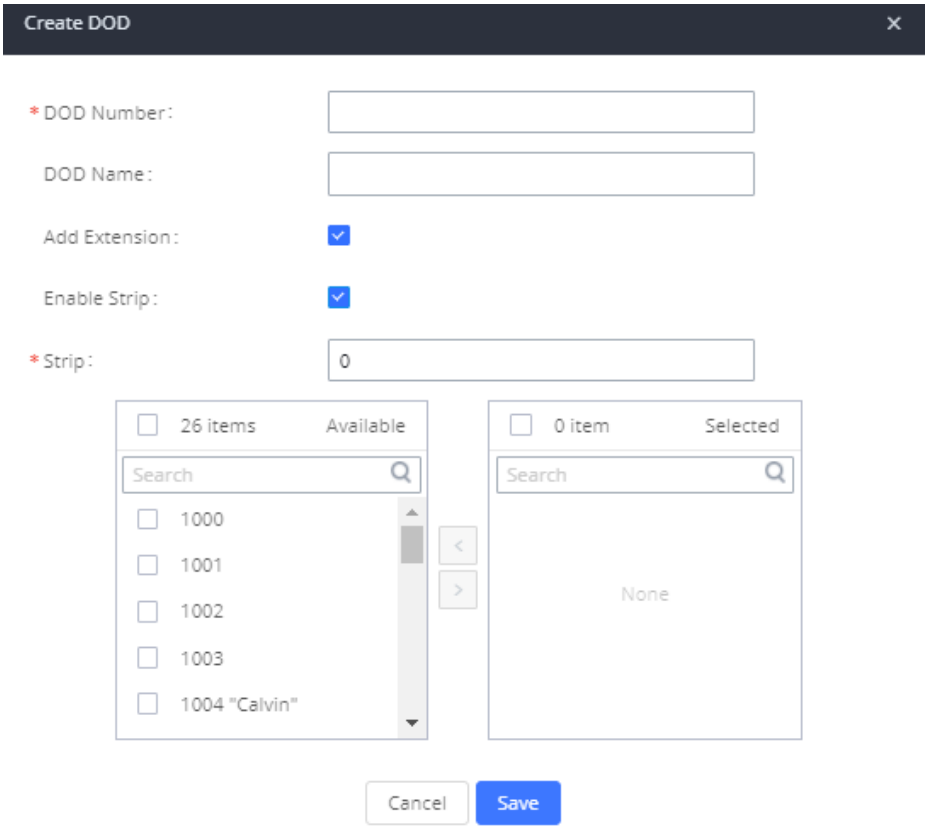
Company ABC has a SIP trunk. This SIP trunk has 4 DIDs associated with it. The main number of the office is routed to an auto attendant. The other three numbers are direct lines to specific users of the company. Now when a user makes an outbound call their caller ID shows up as the main office number. This poses a problem, as the CEO would like their calls to come from their direct line. This can be accomplished by configuring DOD for the CEO's extension.

Steps to configure DOD:

1. To setup DOD go to IPPBX Web GUI→**Extension/Trunk→VoIP Trunks** page.
2. Click to access the DOD options for the selected SIP Trunk.
3. Click "Add DOD" to begin your DOD setup
4. Enter a SIP trunk DID number in the "DOD number" field. In this example, ABC company has a total of 4

DID numbers. Enter the phone number used by the CEO here.

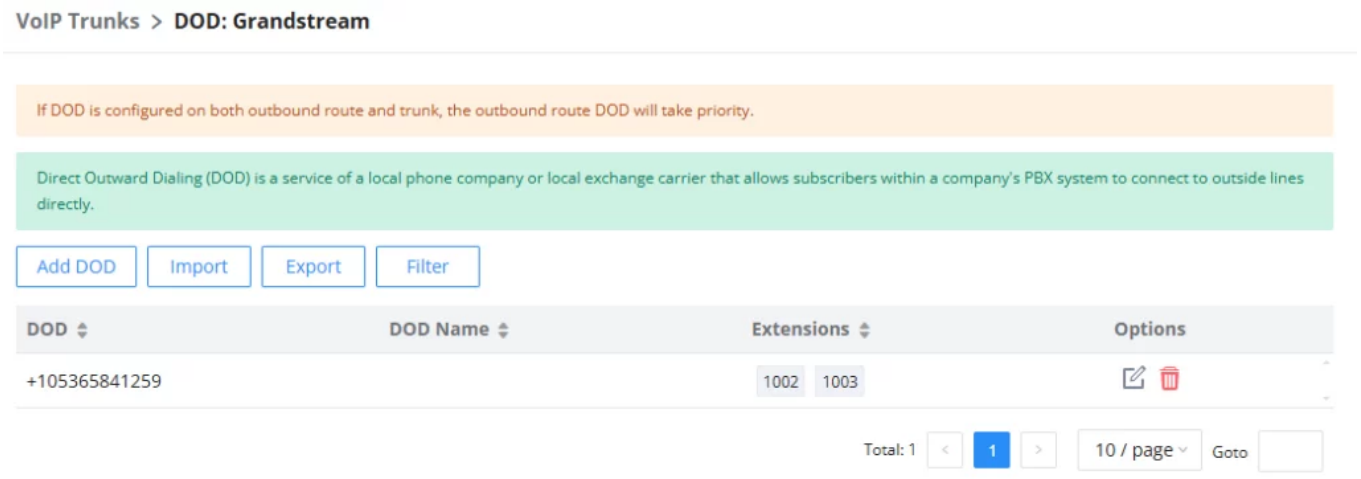
5. When adding extensions, you can choose whether to "Enable Strip" according to your needs. If it is enabled, you can configure the number (0-64) that will be stripped from the extension number before being added to the DOD number. For example, if the entered digit is 2, and the DOD number for extension 4002 is 1122, then dialing out from 4002, 112202 will be used as the caller ID (DOD).
6. Select an extension from the "Available Extensions" list. Users have the option of selecting more than one extension. In this case, Company ABC would select the CEO's extension. After making the selection, click on the button to move the extension(s) to the "Selected Extensions" list.



DOD extension selection

7. Click "Save" at the bottom.

Once completed, the user will return to the **EDIT DOD** page which shows all the extensions that are associated with a particular DOD.



Edit DOD

: Add a DOD.
: Import DODs using a csv file.
: Export the DODs using a csv file.
: Filter DODs by number or name.

For DOD importing, please refer to the screenshot below for the template used.

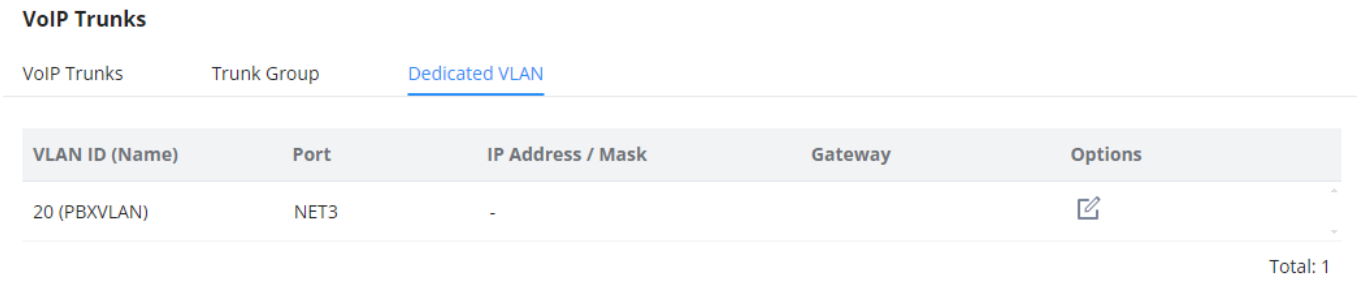
	A	B	C	D	E	F	G
1	DOD Number	DOD Name	Add Extension	Local Members	Ldap Members	Enable Strip	Strip Number
2	2.12555E+11	DOD1	no	1,002,100,310,051,000		no	0
3							
4							
5							
6							
7							

DOD CSV file Template

Dedicated VLAN

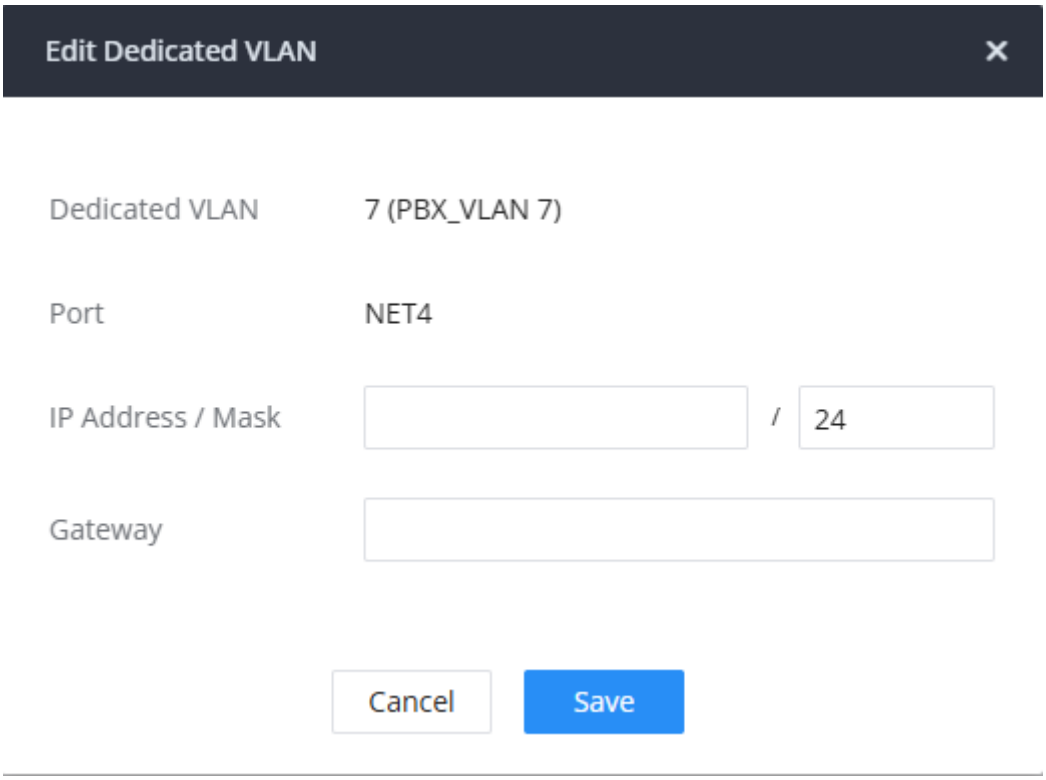
In this section the user can configure the IP address for the dedicated VLAN for the IPPBX operations. This section can be configured after creating the module on the networking module. For more information, please refer to the following link:
https://documentation.grandstream.com/knowledge-base/gcc60xx-networking-user-manual/#pbx-vlan

To edit the dedicated VLAN configuration, please click on the button



Dedicated VLAN

Then enter the IP address, mask, and the IP address of the gateway. If you are connecting using a SIP trunk provided by your ITSP, you may request these information from them.



Edit Dedicated VLAN

WebRTC Trunks

WebRTC, Web Real-Time Communication, is a real-time audio/video chatting framework that allows real-time audio/video chatting through the web browser. WebRTC usually does not refer to the web application itself but to the set of protocols and practices bundled with a graphical interface. Our IPPBX supports creating WebRTC trunks to use exclusively with web applications, this allows the users to join calls and meetings just by clicking a link to a web page.

Below is a figure that shows the options to configure when setting up this feature:

WebRTC Trunk Configuration

Trunk Name	Create a unique label to easily identify the trunk for inbound route configuration.
Disable This Trunk	Check this box to disable this trunk.

Auto Record	If enabled, calls handled with this extension/trunk will automatically be recorded.
Monitor Concurrent Calls	If enabled, the number of concurrent calls on this trunk will be monitored. If the "Trunk Concurrent Calls" system alert is enabled, alert notifications will be generated if the number of concurrent calls exceeds this trunk's configured concurrent call thresholds.
Incoming Concurrent Call Threshold	Threshold of all incoming concurrent calls passing through this trunk.
WebRTC Inbound Link Address	This link can be embedded onto a web page. Clicking the link will connect to a pre-configured WebRTC trunk destination. You can also enter this link in the browser address bar to directly access and test WebRTC calls.

Outbound Routes

In the following sections, we will discuss the steps and parameters used to configure and manage outbound rules in IPPBX, these rules are the regulating points for all external outgoing calls initiated by the IPPBX through SIP trunks.

Configuring Outbound Routes

In the IPPBX, an outgoing calling rule pairs an extension pattern with a trunk used to dial the pattern. This allows different patterns to be dialed through different trunks. Users can also set up a fail-over trunk to be used when the primary trunk fails.

Go to Web GUI→**Extension/Trunk→Outbound Routes** to add and edit outbound rules.

- Click on to add a new outbound route.
- Click the "Import" button to upload the outgoing route in .CSV format.
- Click the "Export" button to generate outgoing routes in .CSV format.
- Click to edit the outbound route.
- Click to delete the outbound route.

On the IPPBX, the outbound route priority is based on the "Best matching pattern". For example, the IPPBX has outbound route A with pattern 1xxx and outbound route B with pattern 10xx configured. When dialing 1000 for an outbound call, outbound route B will always be used first. This is because pattern 10xx is a better match than pattern 1xxx. Only when there are multiple outbound routes with the same pattern configured.

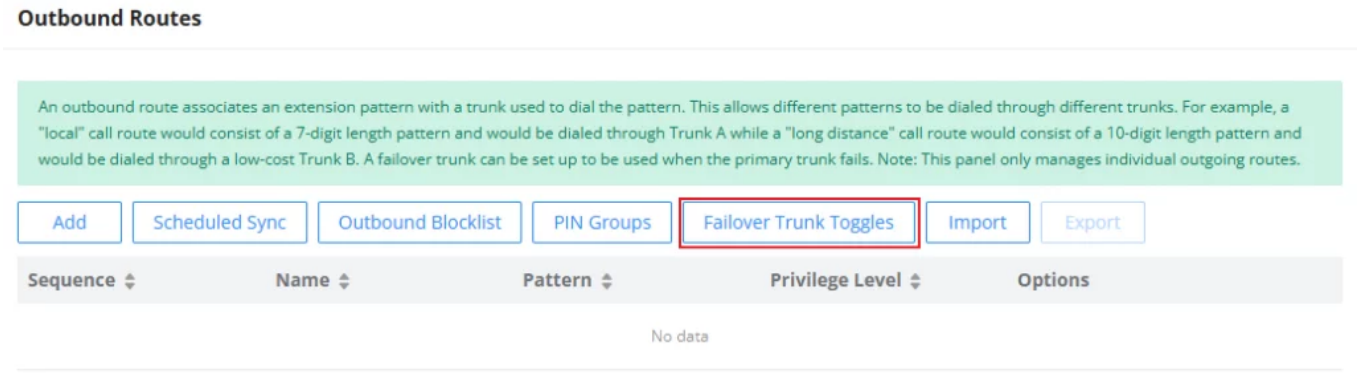
Outbound Rule Name	Configure the name of the calling rule (e.g., local, long_distance, etc.). Letters, digits, _ and - are allowed.
Pattern	All patterns are prefixed by the "_" character, but please do not enter more than one "_" at the beginning. All patterns can add comments, such as "_pattern /* comment */". In patterns, some characters have special meanings: [12345-9] ... Any digit in the brackets. In this example, 1,2,3,4,5,6,7,8,9 is allowed. N ... Any digit from 2-9. . ... Wildcard, matching one or more characters. ! ... Wildcard, matching zero or more characters immediately. X ... Any digit from 0-9. Z ... Any digit from 1-9. - ... Hyphen is to connect characters and it will be ignored. [] Contain special characters ([x], [n], [z]) represent letters x, n, z.
Disable This Route	After creating the outbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in IPPBX. Users can enable it again when it is needed.
Password	Configure the password for users to use this rule when making outbound calls.
Local Country Code	If your local country code is affected by the outbound blacklist, please enter it here to bypass the blacklist.

Call Duration Limit	Enable to configure the maximum duration for the call using this outbound route.
Maximum Call Duration	Configure the maximum duration of the call (in seconds). The default setting is 0, which means no limit.
Warning Time	Configure the warning time for the call using this outbound route. If set to x seconds, the warning tone will be played to the caller when x seconds are left to end the call.
Auto Record	If enabled, calls using this route will automatically be recorded.
Warning Repeat Interval	Configure the warning repeat interval for the call using this outbound route. If set to X seconds, the warning tone will be played every x seconds after the first warning.
PIN Groups	Select a PIN Group
PIN Groups with Privilege Level	If enabled and PIN Groups are used, Privilege Levels and Filter on Source Caller ID will also be applied.
Privilege Level	Select the privilege level for the outbound rule. Internal: The lowest level required. All users can use this rule. Local: Users with Local, National, or International levels can use this rule. National: Users with National or International levels can use this rule. International: The highest level required. Only users with the international level can use this rule. Disable: The default setting is "Disable". If selected, only the matched source caller ID will be allowed to use this outbound route. Please be aware of the potential security risks when using the "Internal" level, which means all users can use this outbound rule to dial out from the trunk.
Enable Source Caller ID Allowlist	When enabled, users could specify extensions allowed to use this outbound route. "Privilege Level" is automatically disabled if using "Enable Source Caller ID Allowlist". The following two methods can be used at the same time to define the extensions as the source caller ID. Select available extensions/extension groups from the list. This allows users to specify arbitrary single extensions available in the PBX. Custom Dynamic Route: define the pattern for the source caller ID. This allows users to define extension range instead of selecting them one by one. All patterns are prefixed with the "_". Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9] - Any digit from 1 to 9. Note: Multiple patterns can be used. Patterns should be separated by a comma ",". Example: _X. , _NNXXNXXXXX, _818X.
Outbound Route CID	Attempt to use the configured outbound route CID. This CID will not be used if DOD is configured.
Send This Call Through Trunk	
Trunk	Select the trunk for this outbound rule.
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. Example: The users will dial 9 as the first digit of long-distance calls. In this case, 1 digit should be stripped before the call is placed.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Use Failover Trunk	
Failover Trunk	Failover trunks can be used to make sure that a call goes through an alternate route when the primary trunk is busy or down. If "Use Failover Trunk" is enabled and "Failover trunk" is defined, the calls that cannot be placed via the regular trunk may have a secondary trunk to go through. IPPBX supports up to 10 failover trunks. Example: The user's primary trunk is a VoIP trunk, and the user would like to use the PSTN when the VoIP trunk is not available. The PSTN trunk can be configured as the failover trunk of the VoIP trunk.

Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk. Example: The users will dial 9 as the first digit of long-distance calls. In this case, 1 digit should be stripped before the call is placed.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Time Condition	
Time Condition Mode	Use Main Trunk or Failover Trunk: Use the Main Trunk and its settings during the configured time conditions. If the main trunk is unavailable, the Failover Trunk and its settings will be used instead. Use Specific Trunks: Use specific trunks during the configured time conditions. The Strip and Prepend settings of the Main Trunk will be used. If a trunk is unavailable during its time condition, no failover trunks will be used.

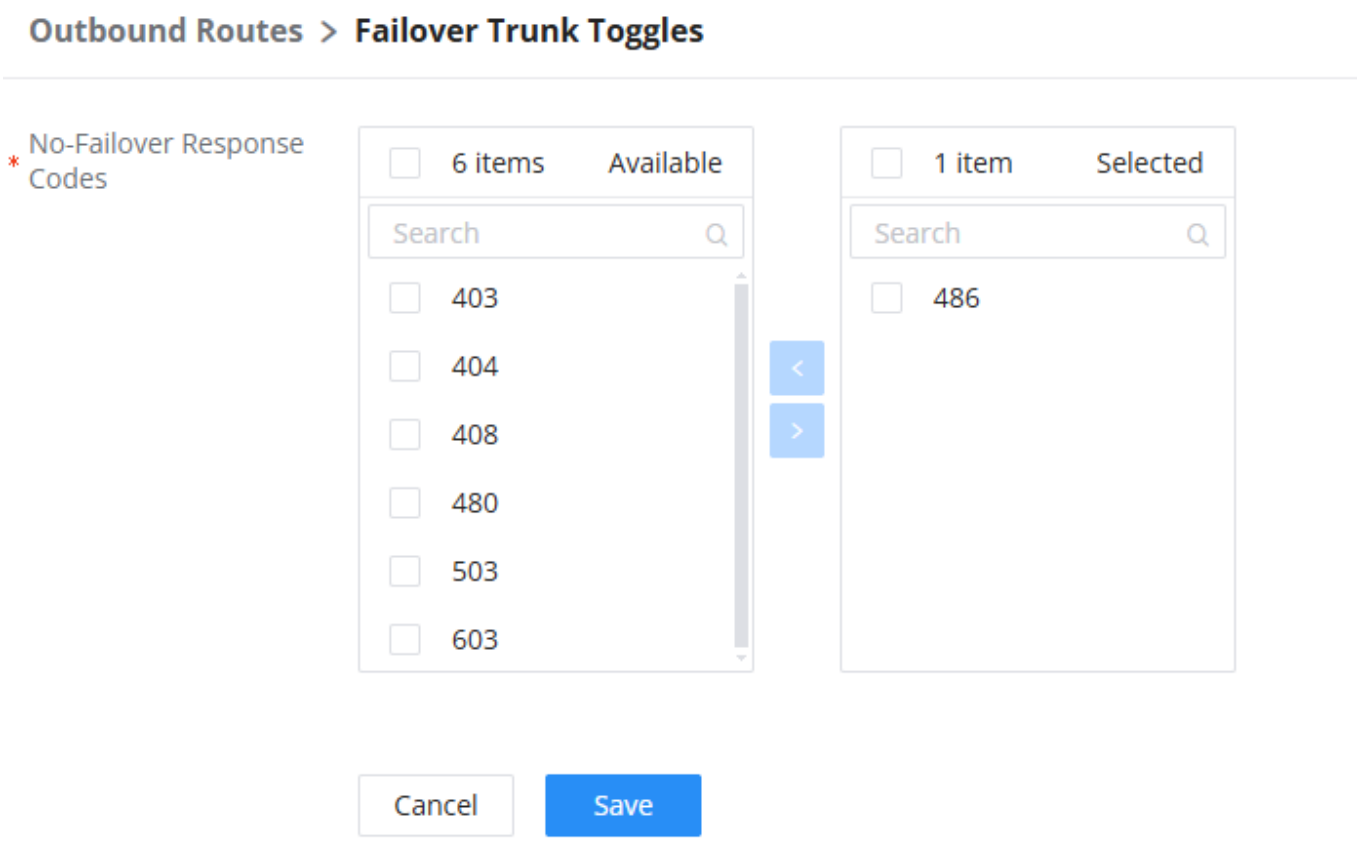
Outbound Route Configuration Parameters

Failover Trunk Toggles



Inbound Routes

This option controls whether failover trunks will be used if receiving specific responses to outgoing calls.



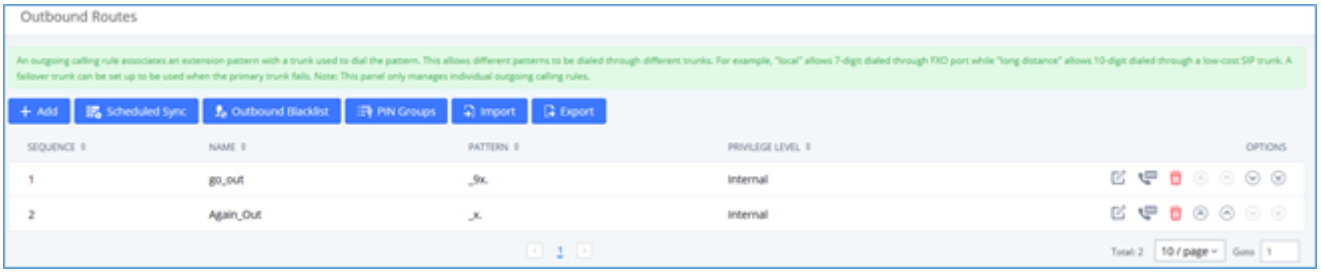
Failover Trunk Toggles

If a call receives the selected response codes, the IPPBX will redirect it to the call route's failover trunk.

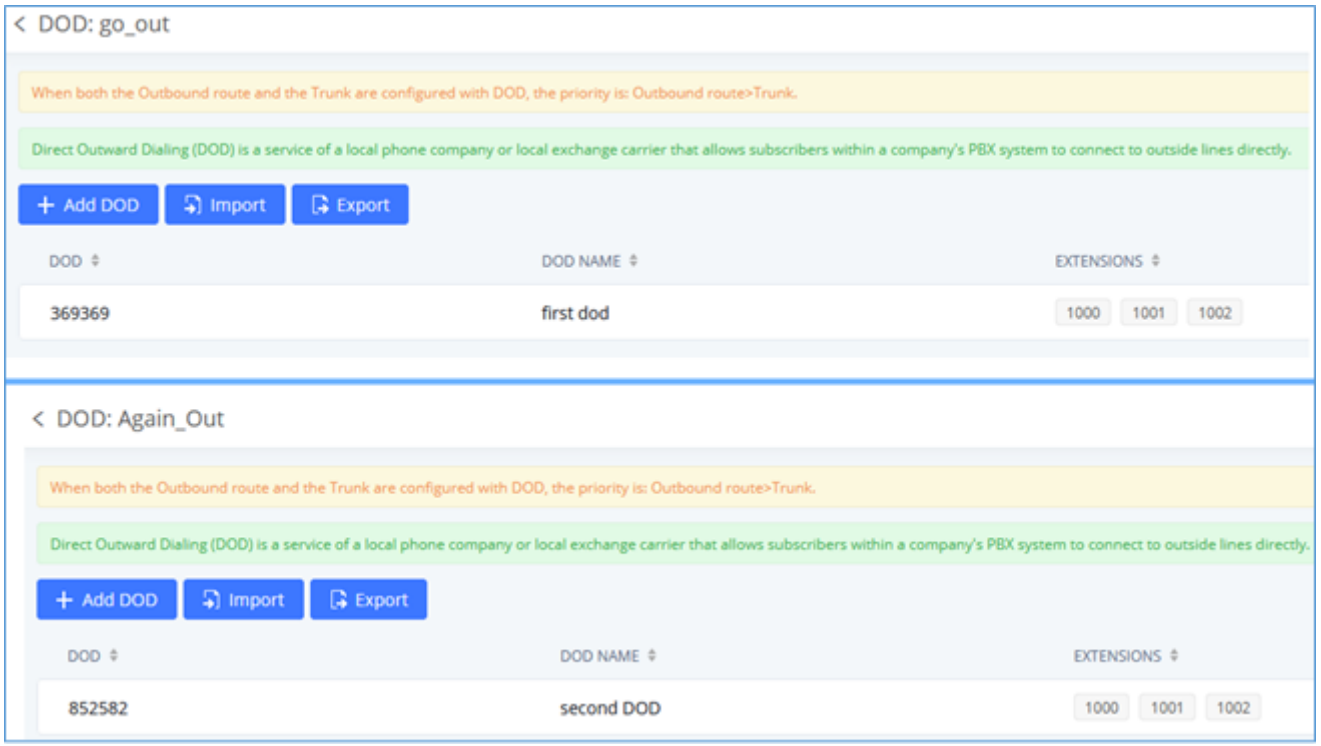
Due to the addition of this option, the **Enable 486 to Failover Trunks** option under **PBX Settings > General Settings** page has been removed.

Outbound Routes DOD

It is possible to specify the DOD number based on the Outbound Route, as displayed in the screenshot below. For each outbound route.



Outbound Routes Page



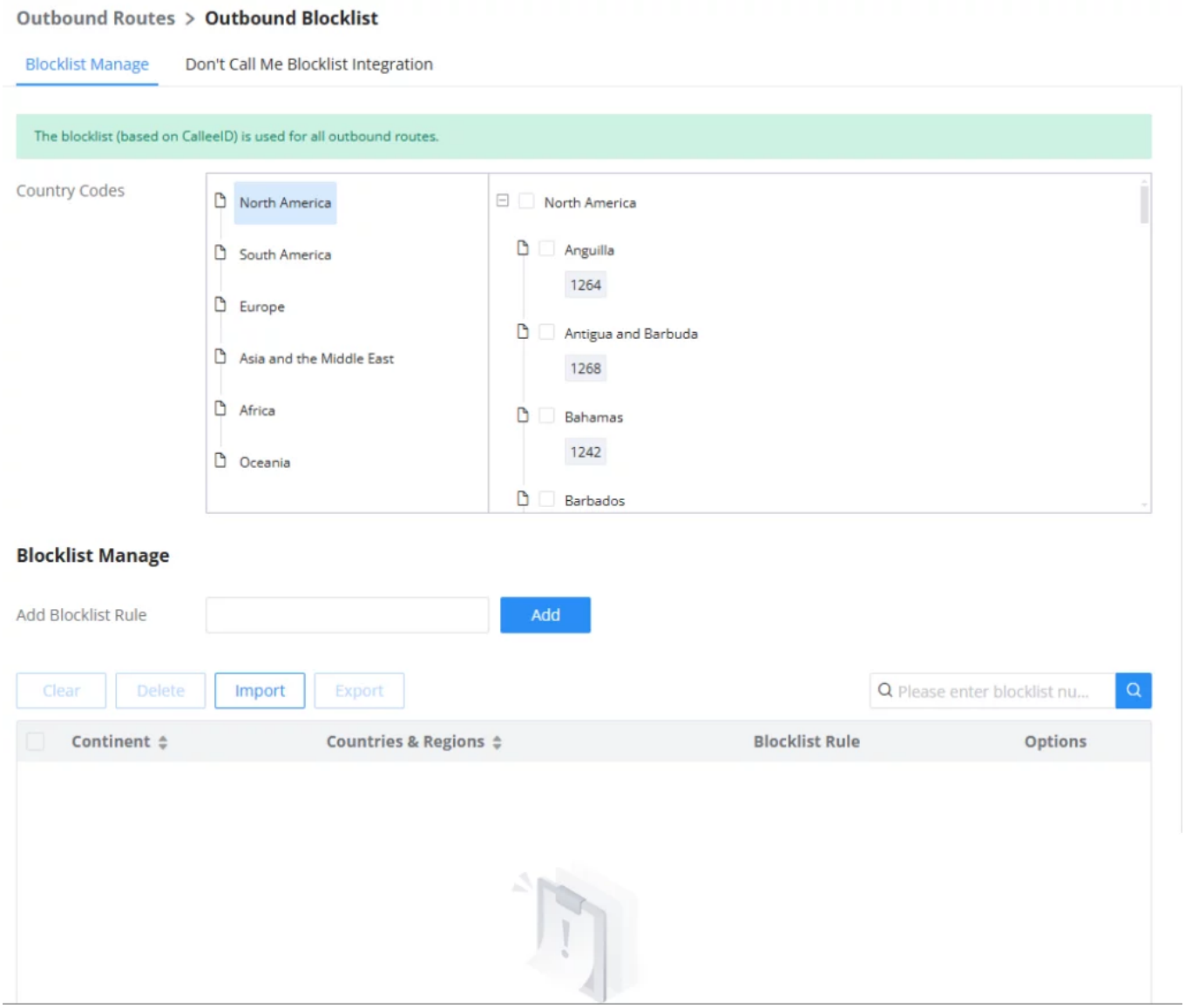
DOD Configuration by Outbound Route

Outbound Blocklist

The IPPBX allows users to configure a blacklist for outbound routes. If the dialing number matches the blacklist numbers or patterns, the outbound call will not be allowed. The outbound blacklist can be configured under IPPBX Web GUI **> Extension/Trunk > Outbound Routes**: Outbound Blocklist.

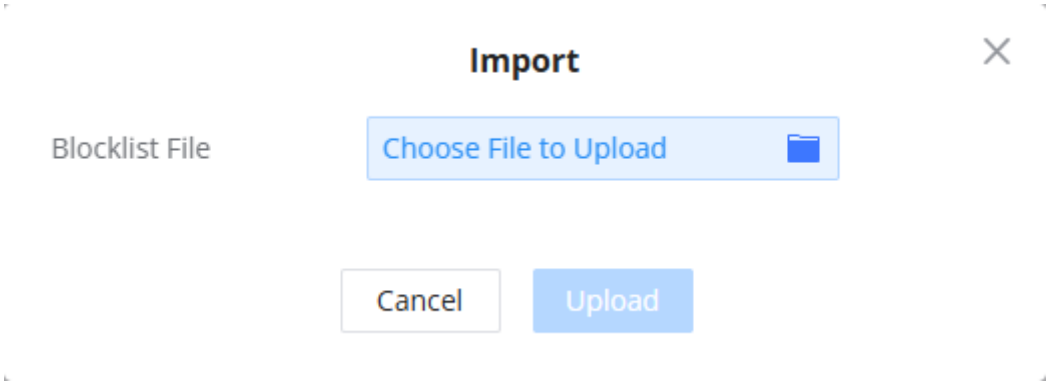
Blocklist Manage

Users can configure numbers, patterns or select country code to add to the blacklist. Please note that the blacklist settings apply to all outbound routes.



Country Codes

Users can export outbound route blacklists and delete all blacklist entries. Additionally, users can also import blacklists for outbound routes.



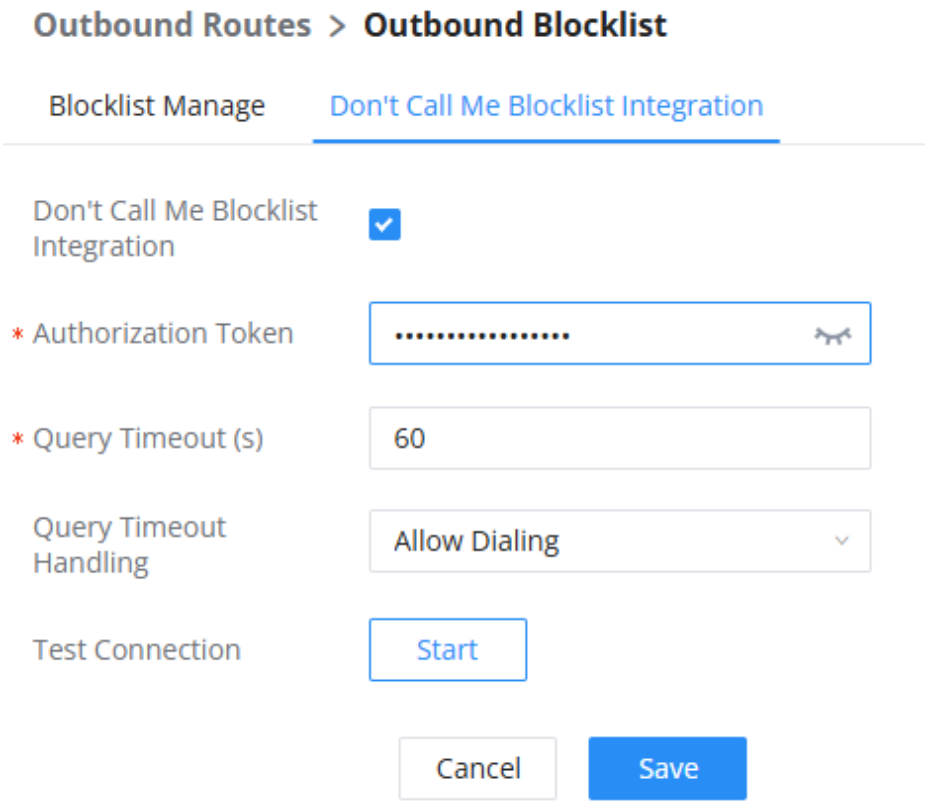
Blocklist Import

Don't Call Me Blocklist Integration

GCC IPPBX module supports integration with Vercel's Don't Call Me, a large public database containing numbers that have been registered and opted to not receive unsolicited telemarketing calls. This allows the

PBX to query dialed numbers via API and deny outgoing calls if they are found in database to comply with some countries' laws.

To access the integration page, please navigate to **Extension/Trunk > Outbound Routes,** then click on "Outbound Blocklist" button and click on **Integrate Don't Call Me Blocklist**.



Integrate Dont Call Me Blocklist

Parameter	Description
Integrate Don't Call Me Blacklist	Enable or disable Don't Call Me integration
Authorization Token	Enter the authorization token generated by the Don't Call Me database.
Query Timeout (s)	Enter the duration after which the query is considered timed out.
Query Timeout Handling	Select the action to perform after the query timeout. Allow Dialing: If the query times out, the call will be allowed. Prohibit Dialing: If the query times out, the call will be prohibited.
Test Connection	Click on this button to test that the integration is working as intended. Note: If the database or Internet access is momentarily down, this test will fail.

PIN Groups

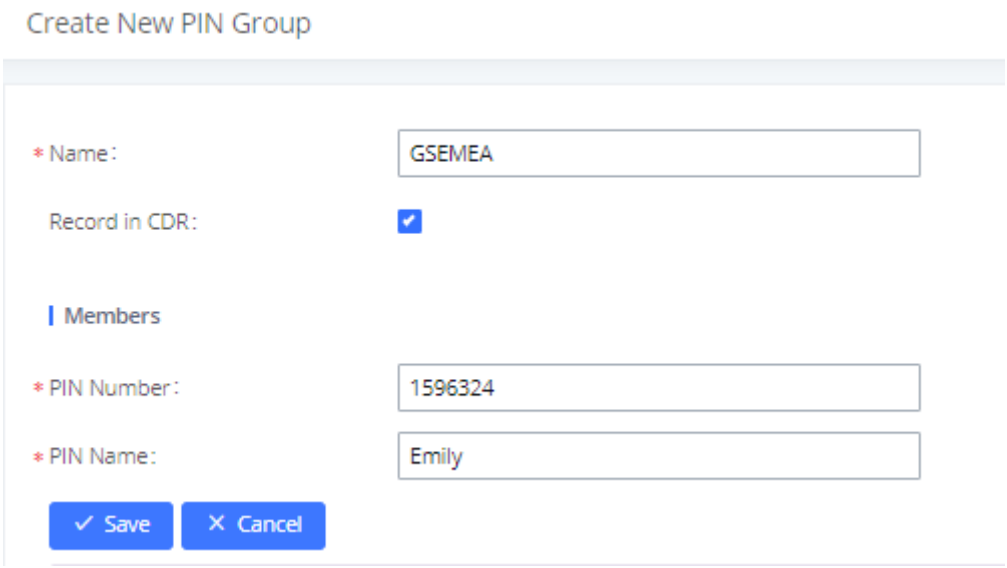
The IPPBX supports the pin group. Once this feature is configured, users can apply pin groups to specific outbound routes. When placing a call on pin-protected outbound routes, the caller will be asked to input the group PIN, this feature can be found on the Web GUI **> Extension/Trunk > Outbound Routes > PIN Groups.**

Name	Specify the name of the group
Record In CDR	Specify whether to enable/disable the record in CDR
PIN Number	Specify the code that will be asked once dialing via a trunk
PIN Name	Specify the name of the PIN

Outbound Routes/PIN Group

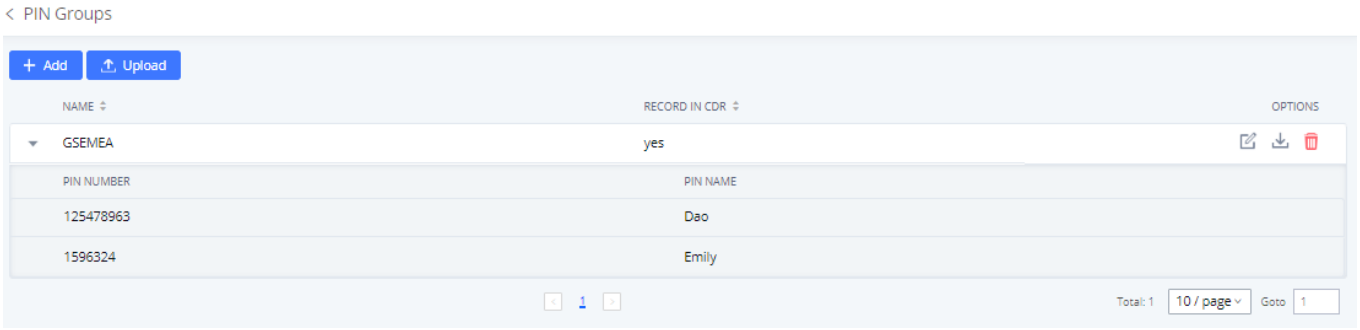
Once the user clicks

, the following figure shows to configure the new PIN.



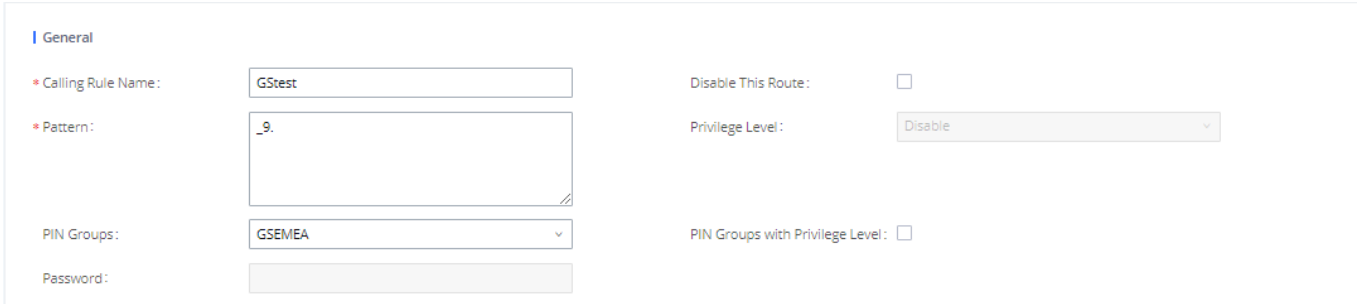
Create a New PIN Group

The following screenshot shows an example of created PIN Groups and members:



PIN Members

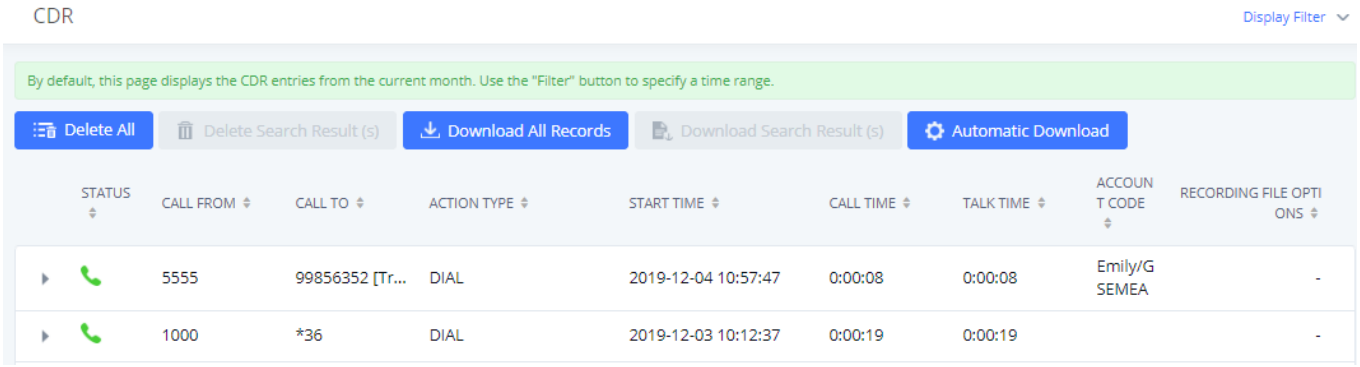
If the PIN group is enabled on the outbound route level, the password, privilege level and enable the filter on source caller ID will be disabled, unless you check the option "PIN Groups with Privilege Level" where you can use the PIN Groups and Privilege Level or PIN Groups and Enable Source Caller ID Allowlist.



Outbound PIN

If PIN group CDR is enabled, the call with PIN group information will be displayed as part of CDR under the

Account Code field.

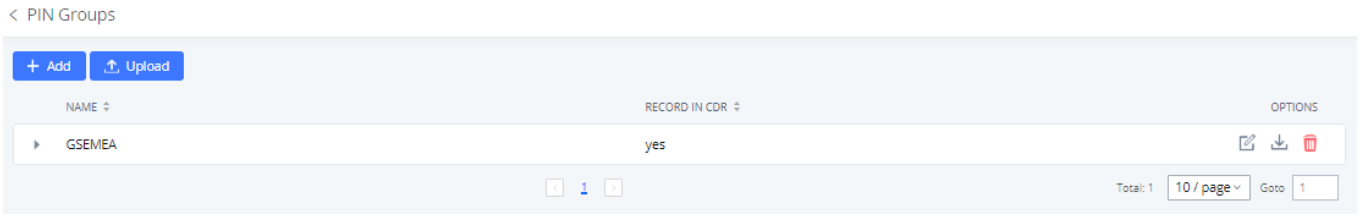


CDR Record

- **Importing PIN Groups from CSV files:**

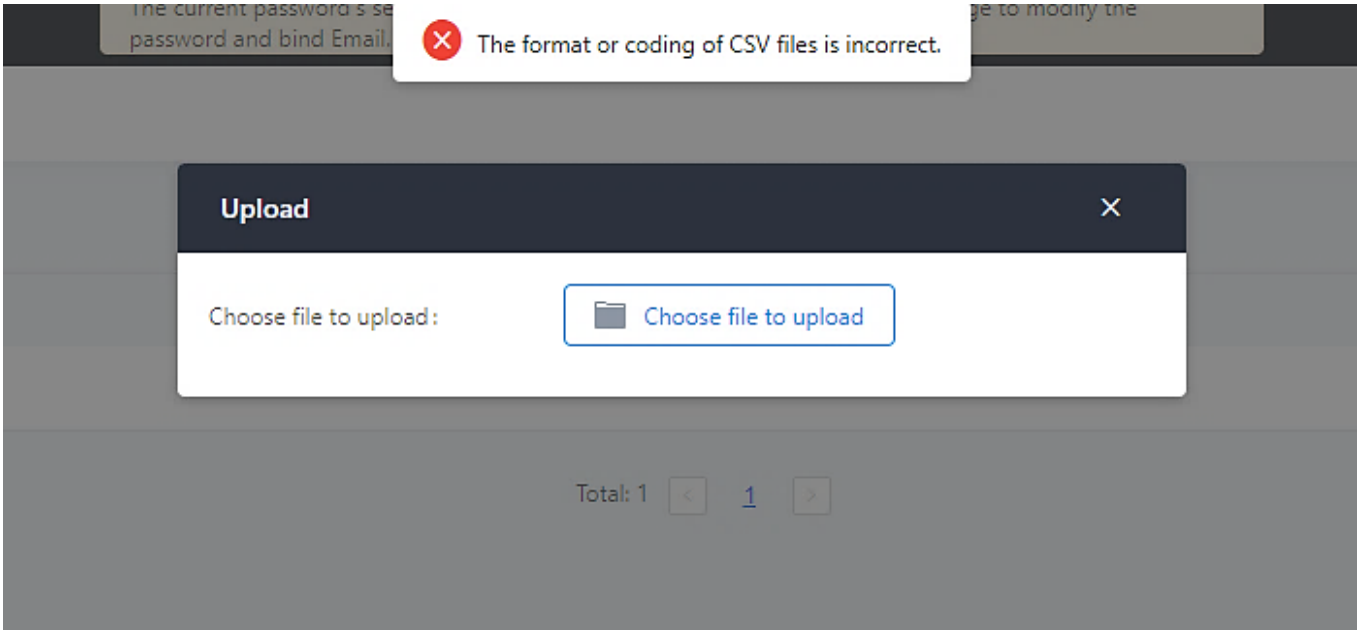
Users can also import PIN Groups by uploading CSV files for each group. To do this:

1. Navigate to **Extension/Trunk→Outbound Routes→PIN Groups** and click on the "Upload" button.



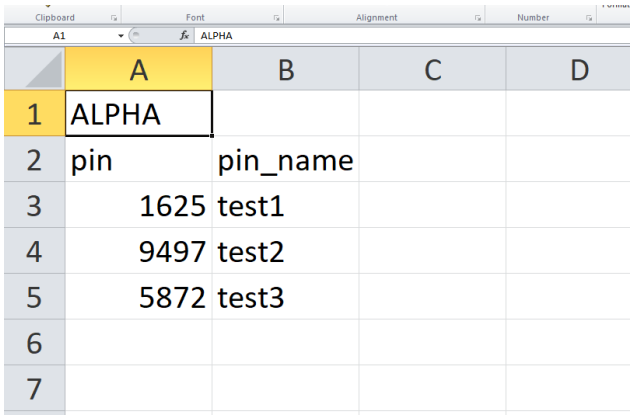
Importing PIN Groups from CSV files

2. Select the CSV file to upload. Incorrect file formats and improperly formatted CSV files will result in error messages such as the one below:



Incorrect CSV File

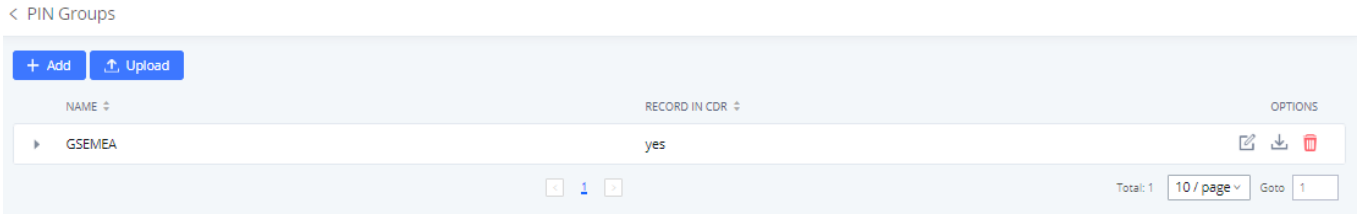
3. To ensure a successful import, please follow the format in the sample image below



	A	B	C	D
1	ALPHA			
2	pin	pin_name		
3	1625	test1		
4	9497	test2		
5	5872	test3		
6				
7				

CSV File Format

- The top-left value (A1) is the PIN Group name. In this case, it is "ALPHA".
- Row 2 contains the labels for the modifiable fields: pin and pin_name. These values should not be changed and will cause an upload error otherwise.
- Rows 3+ contain the user-defined values with Column A holding the PINs and Column B holding the PIN names. PIN values must consist of at least four digits.
- Once the file is successfully uploaded, the entry will be added to the list of PIN Groups.



CSV File Successful Upload

Inbound Routes

Inbound routes can be configured via Web GUI→**Extension/Trunk→Inbound Routes**.

- Click on to add a new inbound route.
- Click on "Blacklist" to configure the blacklist for all inbound routes.
- Click on to edit the inbound route.
- Click on to delete the inbound route.

Inbound Route Configuration

Trunks	Select the trunk to configure the inbound rule.
Inbound Route Name	Configure the name of the Inbound Route. For example, "Local", "LongDistance" etc.
Pattern	All patterns are prefixed with the "_".Special characters:X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately. Example: [12345-9] - Any digit from 1 to 9.Notes: Multiple patterns can be used. Each pattern should be entered in a new line. Example: _X. _ NNXXNXXXXX /* 10-digit long distance */
Disable This Route	After creating the inbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in the PBX. Users can enable it again when it is needed.

CID Source	Configures the source of the CID to match with the configured CallerID Pattern. None: CID is not obtained from any source. Only applicable if no CallerID Pattern is configured. DiversionID: CID is obtained from the Diversion header. Only applicable to SIP trunks. CallerID: If the call is from a SIP trunk, the CID is obtained from the From header. Otherwise, the CID will be obtained from other related signaling.
Seamless Transfer Whitelist	Allows the selected extension to use this function. If an extension is busy, and a mobile phone is bound to that extension, the mobile phone can pick up calls to that extension.
Ringback tone	Choose the custom ringback tone to play when the caller reaches the route.
Auto Record	If enabled, calls using this route will automatically be recorded.
Block Collect Call	If enabled, collect calls will be blocked.Note: Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".
Alert-Info	Configure the Alert-Info, when the PBX receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
Fax Detection	If enabled, fax signals from the trunk during a call will be detected.
Fax Destination	Configures the destination of faxes. Extension: send the fax to the designated SIP extension (fax machine) or a FAX extension. Fax to Email: send the fax as an email attachment to the designated extension's email address. If the selected extension does not have an associated email address, it will be sent to the default email address configured in the Advanced Call Features->Fax/T.38->Fax Settings page. Note: please make sure the sending email address is correctly configured in System Settings->Email Settings.
Auto Answer	If enabled, the PBX will automatically answer calls and receive faxes through the inbound route. If disabled, the PBX will not receive a fax until after the call has been answered. Enabling this option will slow down the answering of non-fax calls on the inbound route. The alert tone heard during the detection period can be customized.
Block Collect Calls	If enabled, collect calls will be blocked. Note: Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call". Note: This is affected by Block Set Calls on the SIP Settings -> General Settings page.
Prepend Trunk Name	If enabled, the trunk name will be added to the caller id name as the displayed caller id name.
Set Caller ID Info	Manipulates Caller ID (CID) name and/or number within the call flow to help identify who is calling. When enabled two fields will show allowing to manipulate the CalleID Number and the Caller ID Name.
CallerID Number	Configure the pattern-matching format to manipulate the numbers of incoming callers or to set a fixed CallerID number for calls that go through this inbound route. ${CALLERID(num)}: Default value which indicates the number of an incoming caller (CID). The CID will not be modified. ${CALLERID(num):n}: Skips the first n characters of a CID number, where n is a number. ${CALLERID(num):-n}: Takes the last n characters of a CID number, where n is a number. ${CALLERID(num):s:n}: Takes n characters of a CID number starting from s+1, where n is a number and s is a character position (e.g. ${CALLERID(num):2:7} takes 7 characters after the second character of a CID number). n${CALLERID(num)}: Prepends n to a CID number, where n is a number.
CallerID Name	The default string is ${CALLERID(name)},which means the name of an incoming caller, it is a pattern-matching syntax format. A${CALLERID(name)}B means Prepend a character 'A' and suffix a character 'B' to ${CALLERID(name)}.Not using pattern-matching syntax means setting a fixed name to the incoming caller.
Enable Route-Level Inbound Mode	Gives uses the ability to configure inbound mode per individual route. When enabled two fields will show allowing to set the Inbound mode and the Inbound mode Suffix.Note: Global inbound mode must be enabled before users can configure route-level inbound mode.

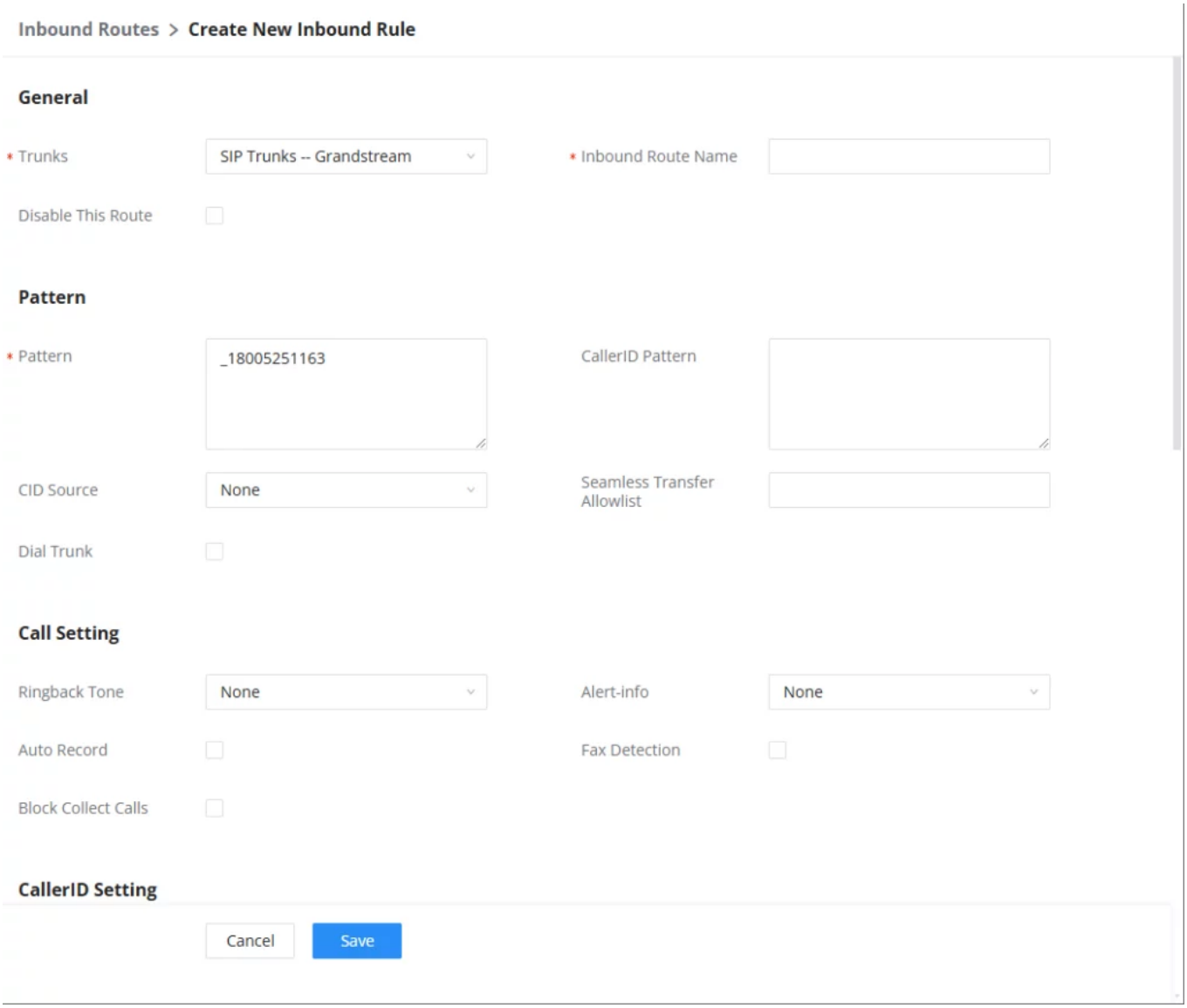
Inbound Mode	Choose the inbound mode for this route.Note: Toggling the global inbound mode will not affect routes that have Route-level Inbound Mode enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.
Inbound Mode Suffix	Dial "Global Inbound Mode feature code + Inbound Mode Suffix" or a route's assigned suffix to toggle the route's inbound mode.The BLF subscribed to the inbound mode suffix can monitor the current inbound mode.
Inbound Multiple Mode	Multiple mode allows users to switch between destinations of the inbound rule by feature codes. Configure related feature codes as described in [Inbound Route: Multiple Mode]. If this option is enabled, the user can use feature code to switch between different modes/destinations.
CallerID Name Lookup	If enabled, the callerID will be resolved to a name through local LDAP. Note, if a matched name is found, the original callerID name will be replaced. The name lookup is performed before other callerID or callerID name modifiers (e.g., Inbound Route's Set CallerID Info or Prepend Trunk Name). Note: Name lookup may impact system performance.
Dial Trunk	This option shows up only when "By DID" is selected. If enabled, the external users dialing into the trunk via this inbound route can dial outbound calls using the PBX's trunk.
Privilege Level	This option shows up only when "By DID" is selected. Disable: Only the selected Extensions or Extension Groups are allowed to use this rule when enabled Filter on Source Caller ID. Internal: The lowest level required. All users are allowed to use this rule, checking this level might be risky for security purposes. Local: Users with Local level, National or International level are allowed to use this rule. National: Users with National or International Level are allowed to use this rule. International: The highest level required. Only users with an international level are allowed to use this rule.
Allowed DID Destination	This option shows up only when "By DID" is selected. This controls the destination that can be reached by the external caller via the inbound route. The DID destination is: Extension Multimedia Meeting Call Queue Ring Group Paging/Intercom Groups IVR Voicemail Groups Dial By Name All
Default Destination	Select the default destination for the inbound call. Extension Voicemail Multimedia Meeting Call Queue Ring Group Paging/Intercom Voicemail Group DISA IVR External Number By DID When "By DID" is used, the PBX will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, and voicemail groups as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through. Dial By Name Callback
Strip	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Time Condition	
Start Time	Select the start time "hour:minute" for the trunk to use the inbound rule.
End Time	Select the end time "hour:minute" for the trunk to use the inbound rule.
Date	Select "By Week" or "By Day" and specify the date for the trunk to use the inbound rule.
Week	Select the day in the week to use the inbound rule.

Destination	Select the destination for the inbound call under the defined time condition. Extension Voicemail Meeting Room Call Queue Ring Group Paging/Intercom Voicemail Group DISA IVR By DID When "By DID" is used, the PBX will look for the destination based on the number dialed, which could be local extensions, conference, call queue, ring group, paging/intercom group, IVR, and voicemail groups as configured in "DID destination". If the dialed number matches the DID pattern, the call will be allowed to go through.Configure the number of digits to be stripped in the "Strip" option. Dial By Name External Number Callback

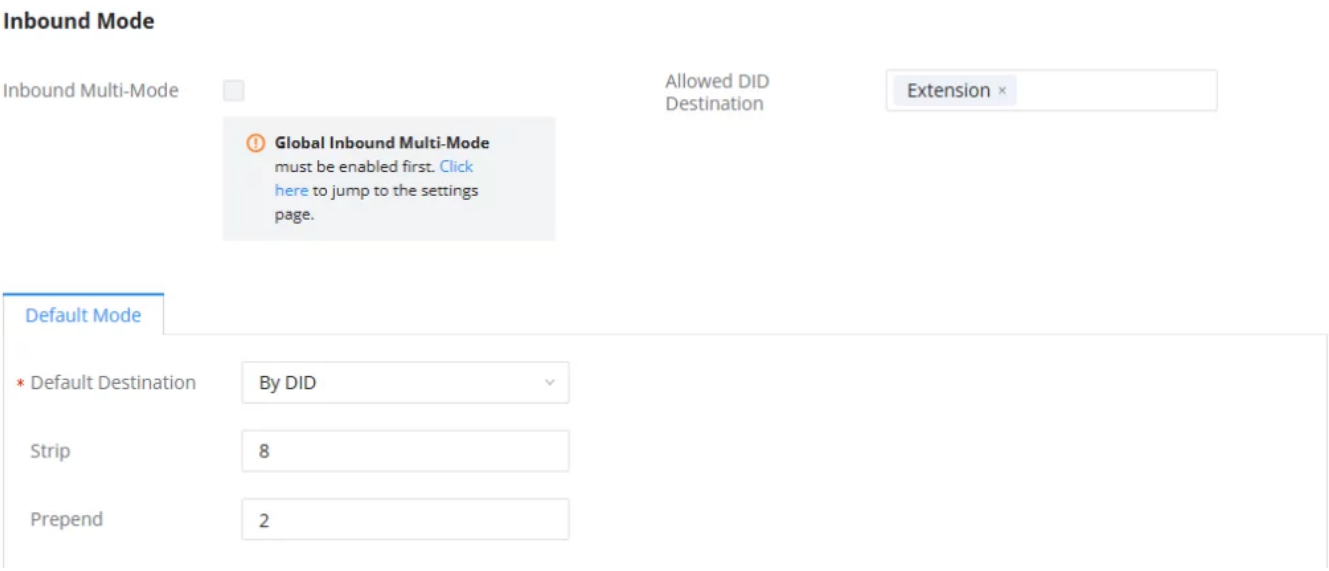
Inbound Route: Prepend Example

The IPPBX allows users to prepend digits to an inbound DID pattern, with strip taking precedence over prepend. With the ability to prepend digits in the inbound route DID pattern, the user no longer needs to create multiple routes for the same trunk to route calls to different extensions. The following example demonstrates the process:

1. If Trunk provides a DID pattern of 18005251163.
2. If **Strip** is set to 8, IPPBX will strip the first 8 digits.
3. If **Prepend** is set to 2, IPPBX will then prepend a 2 to the stripped number, now the number becomes 2163.
4. The IPPBX will forward the incoming call to extension 2163.



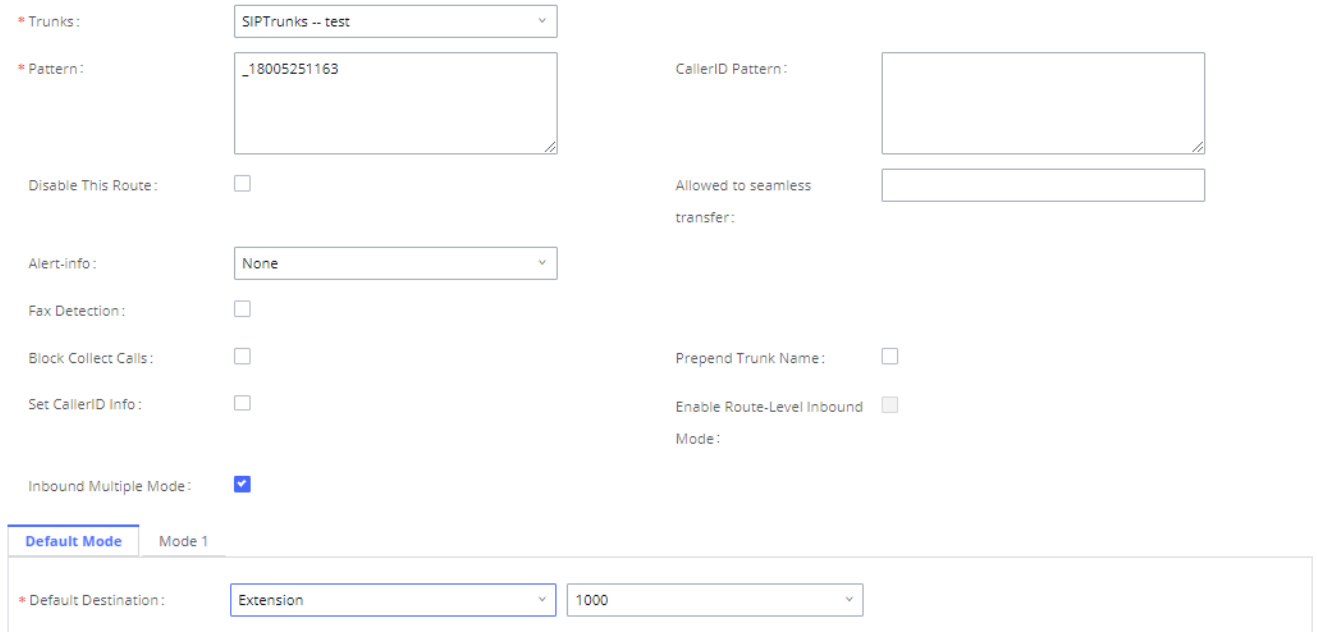
Inbound Route feature Pattern



Inbound Route Strip Prepend

Inbound Route: Multiple Mode

In the IPPBX, the user can configure an inbound route to enable multiple mode to switch between different destinations. The inbound multiple mode can be enabled under Inbound Route settings.



Inbound Route Multiple Mode

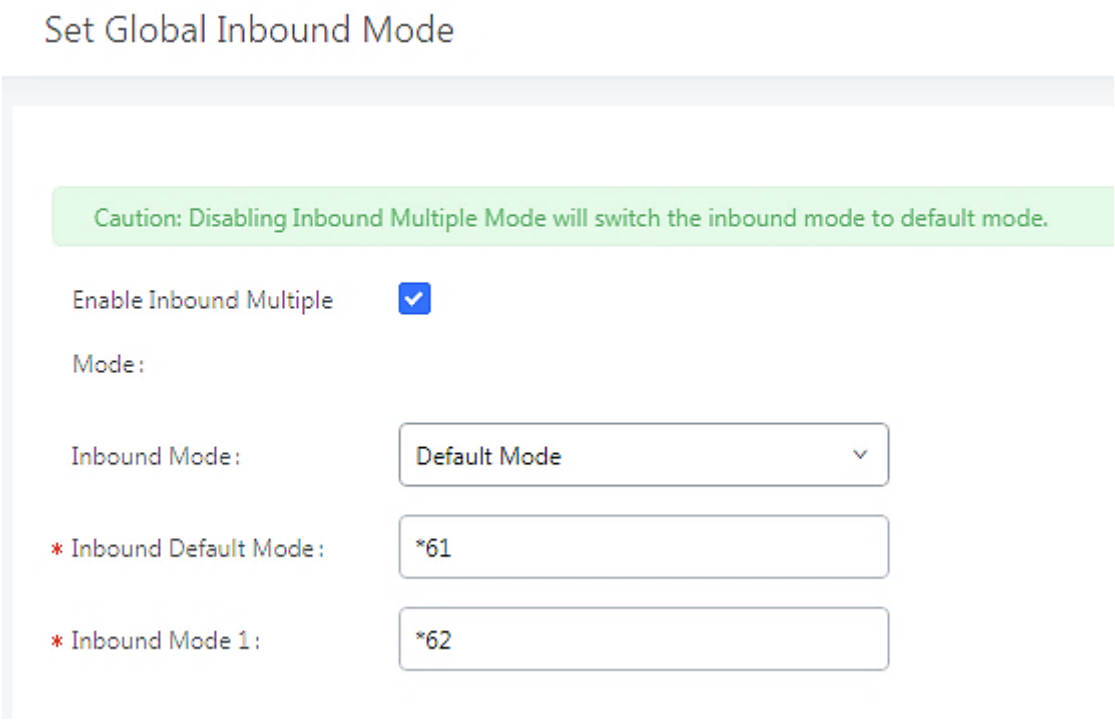
When Multiple Mode is enabled for the inbound route, the user can configure a "Default Destination" and a "Mode 1" destination for all routes. By default, the call coming into the inbound routes will be routed to the default destination.

SIP end devices that have registered on the IPPBX can dial feature code *62 to switch to the inbound route "Mode 1" and dial feature code *61 to switch back to "Default Destination". Switching between different modes can be easily done without a Web GUI login.

For example, the customer service hotline destination has to be set to a different IVR after 7 PM. The user can dial *62 to switch to "Mode 1" with that IVR set as the destination before off work.

To customize feature codes for "Default Mode" and "Mode 1", click on under the "Inbound Routes" page, check the "Enable Inbound Multiple Mode" option, and change "Inbound Default Mode" and "Inbound Mode

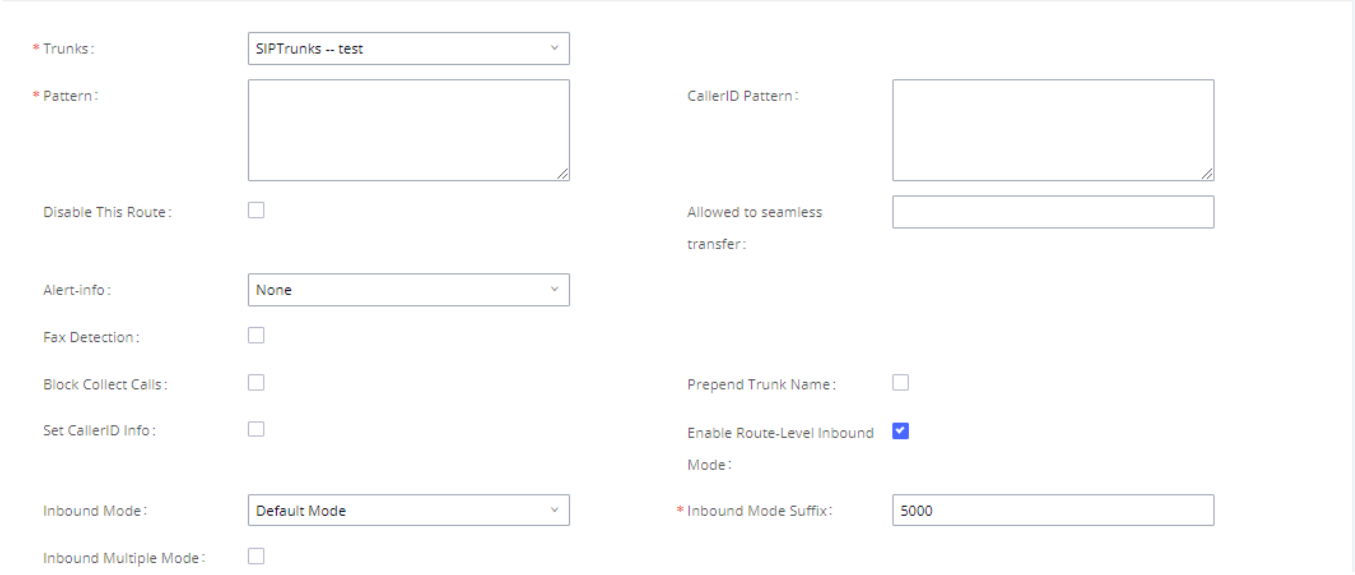
1" values (By default, *61 and *62 respectively).



Inbound Route Multiple Mode Feature Codes

Inbound Route: Route-Level Mode

In the IPPBX, users can enable Route-Level Inbound Mode to switch between different destinations for each inbound route. The inbound Route-Level mode can be enabled under Inbound Route settings.



Inbound Route Route Level Mode

The global inbound mode must be enabled before configuring Route-Level Inbound Mode. Additionally, Mode 1 must be configured as well.

When Route-Level Inbound Mode is enabled, the user can configure a "Default Destination" and a "Mode 1" destination for each specific route. By default, the call coming into this specific inbound route will be routed to the default destination.

Users can toggle the route's inbound mode by dialing "Global Inbound Mode feature code + Inbound Mode Suffix" and the current inbound route can be monitored by subscribing a BLF to the Inbound Mode Suffix.

For example, the Inbound Default Mode feature code is set to *61 and the Inbound Mode suffix for route 1 is set to 1010. To switch the mode of route 1 to Default Mode, users can dial *611010.

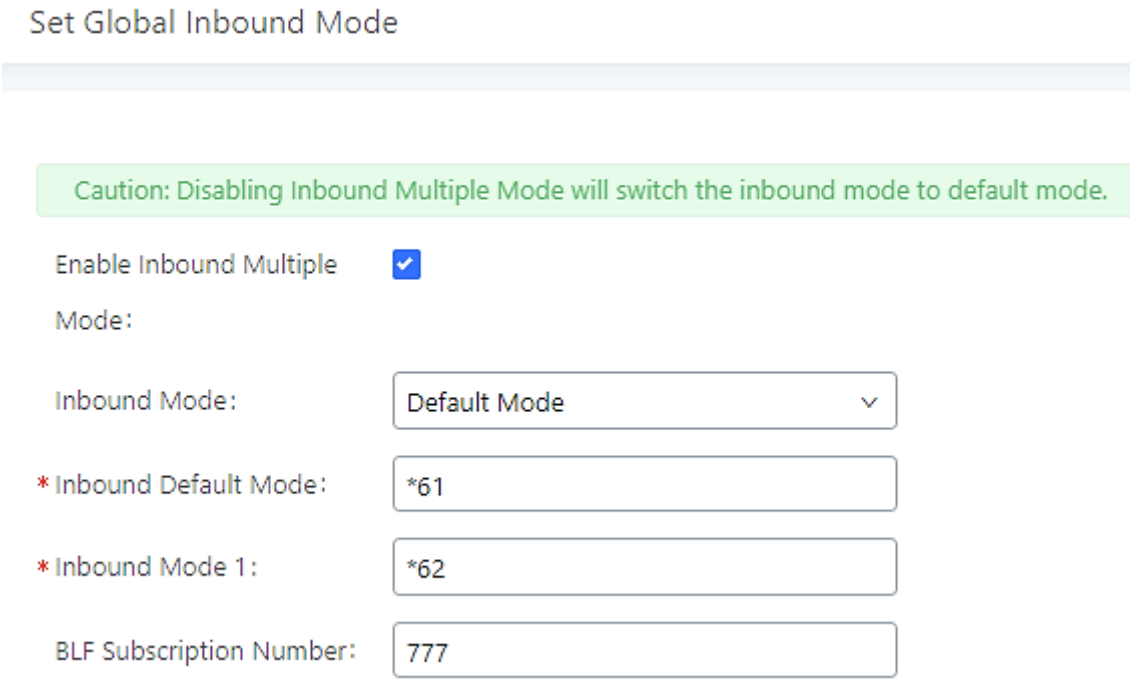
Note: Toggling the global inbound mode will not affect routes that have Route-level Inbound Mode enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.

Inbound Route: Inbound Mode BLF Monitoring

Users can assign MPKs and VPKs to monitor and toggle the current global inbound mode of the IPPBX.

To do this, please refer to the following steps:

1. Access the IPPBX web GUI and navigate to Extension/Trunk→Inbound Routes.
2. Click on the button and enable Inbound Multiple Mode.
3. Edit the subscribe number field to the desired BLF value.



Global Inbound Mode

4. Configure the BLF value on a phone's MPK/VPK. As an example, a GXP2140 with the BLF configured will show the Inbound Mode status on its screen once configured. The 777 BLF is lit green, indicating that the current inbound mode is "Default Mode".

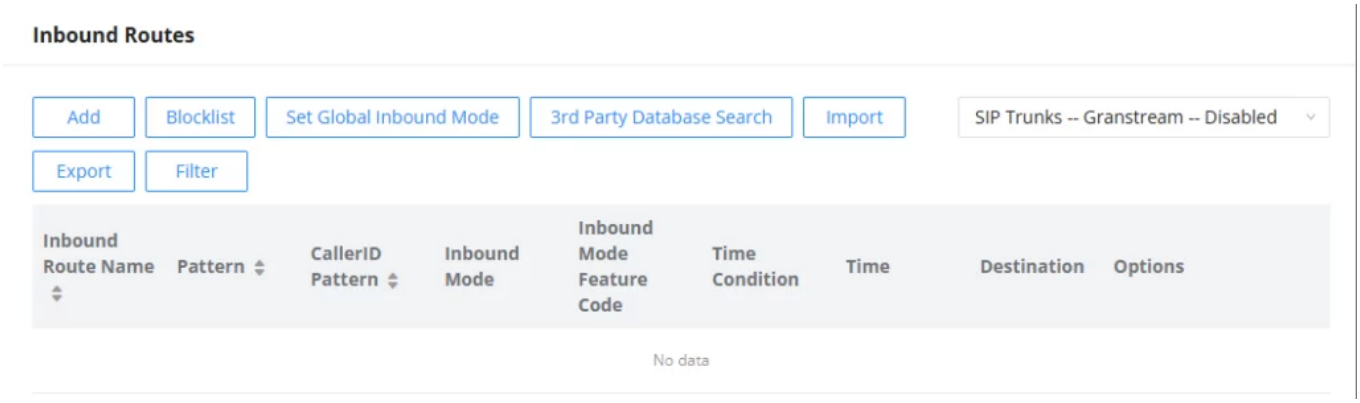
Inbound Mode Default Mode

5. Pressing the key will toggle the inbound mode to "Mode 1", and the button's color will change to red.

Inbound Mode Mode 1

Inbound Route: Import/Export Inbound Route

Users can now import and export inbound routes to quickly set up inbound routing on a IPPBX or to back up an existing configuration. An exported inbound route configuration can be directly imported without needing any manual modifications.



ImportExport Inbound Route

The imported file should be in CSV format and using UTF-8 encoding, the imported file should contain the below columns, and each column should be separated by a comma (It is recommended to use Notepad++ for the imported file creation):

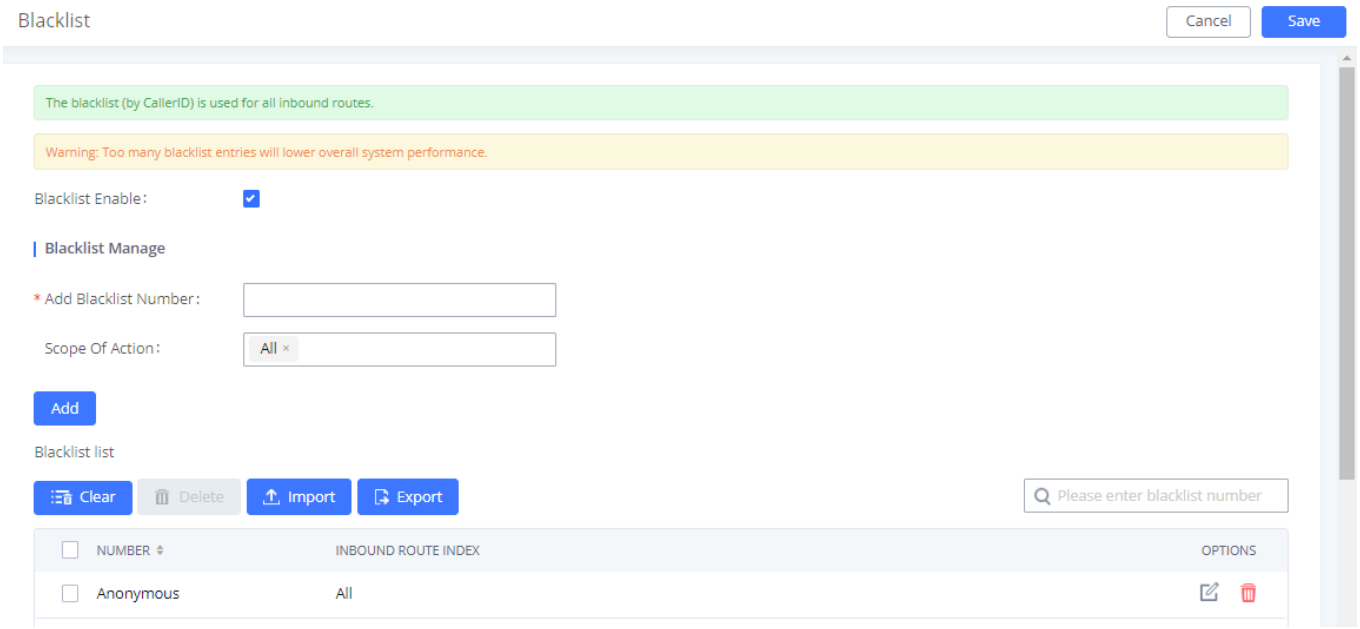
- Disable This Route: Yes/No.
- Pattern: Always prefixed with _
- CallerID Pattern: Always prefixed with _

- Prepend Trunk Name: Yes/No.
- Prepend User Defined Name Enable: Yes/No.
- Prepend User Defined Name: A string.
- Alert-info: None, Ring 1, Ring 2... The user should enter an Alert-info string following the values we have in the Inbound route Alert-Info list.
- Allowed to seamless transfer: [Extension_number]
- Inbound Multiple Mode: Yes/No.
- Default Destination: By DID, Extension, Voicemail... Users should enter a Default Destination string following the values we have in the Inbound route Default Destination list.
- Destination: An Extension number, Ring Group Extension...
- Default Time Condition.
- Mode 1: By DID, Extension, Voicemail... Users should enter a Default Destination string following the values we have in the mode 1 Default Destination list.
- Mode 1 Destination: An Extension number, Ring Group Extension...
- Mode 1 Time Condition.

Blacklist Configurations

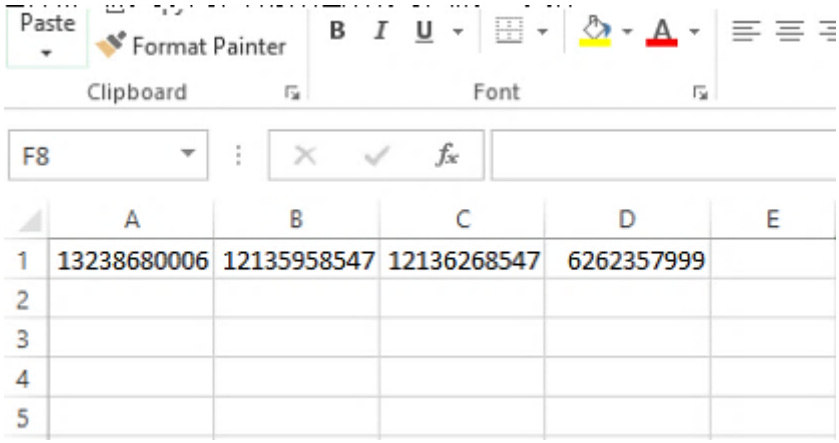
In the IPPBX, Blacklist is supported for all inbound routes. Users could enable the Blacklist feature and manage the Blacklist by clicking on "Blacklist".

- Select the checkbox for "Blacklist Enable" to turn on the Blacklist feature for all inbound routes. The blacklist is disabled by default.
- Enter a number in the "Add Blacklist Number" field and then click "Add" to add to the list. Anonymous can also be added as a Blacklist Number by typing "Anonymous" in Add Blacklist Number field.
- To remove a number from the Blacklist, select the number in the "Blacklist list" and click on or click on the" Clear" button to remove all the numbers on the blacklist.
- Users can also export the inbound route blacklist by pressing the button.



Blacklist Configuration Parameters

- To add blacklisted numbers in batch, click on "Import" to upload the blacklist file in CSV format. The supported CSV format is as below.



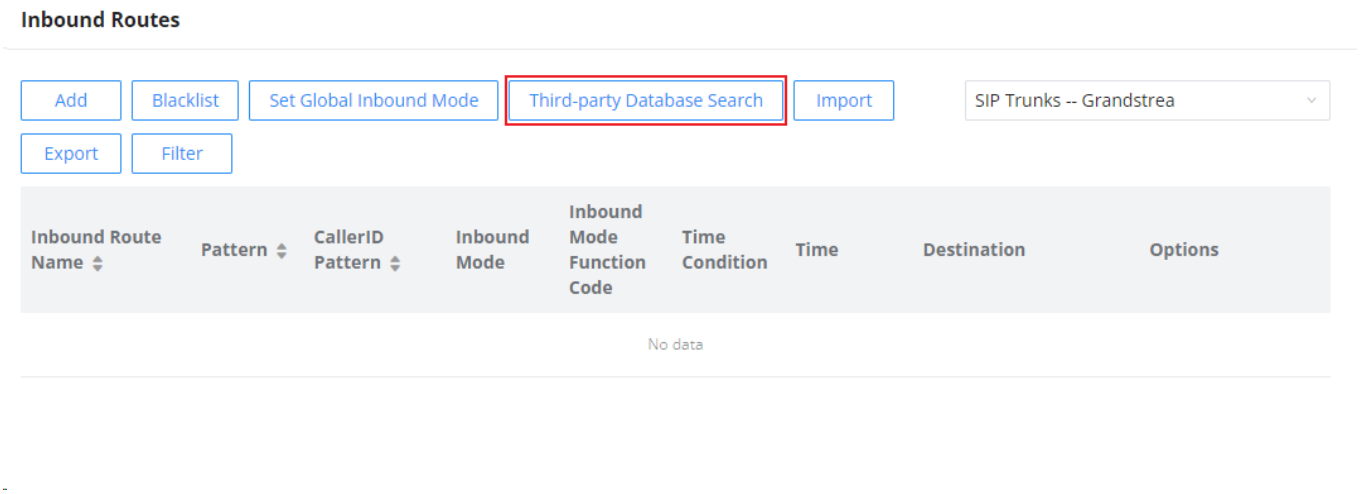
Blacklist CSV File

Users could also add a number to the Blacklist or remove a number from the Blacklist by dialing the feature code for "Blacklist Add' (default: *40) and "Blacklist Remove" (default: *41) from an extension. The feature code can be configured under **Web GUI > Basic Call Features > Feature Codes**.

Inbound Route: Third-party Database Search

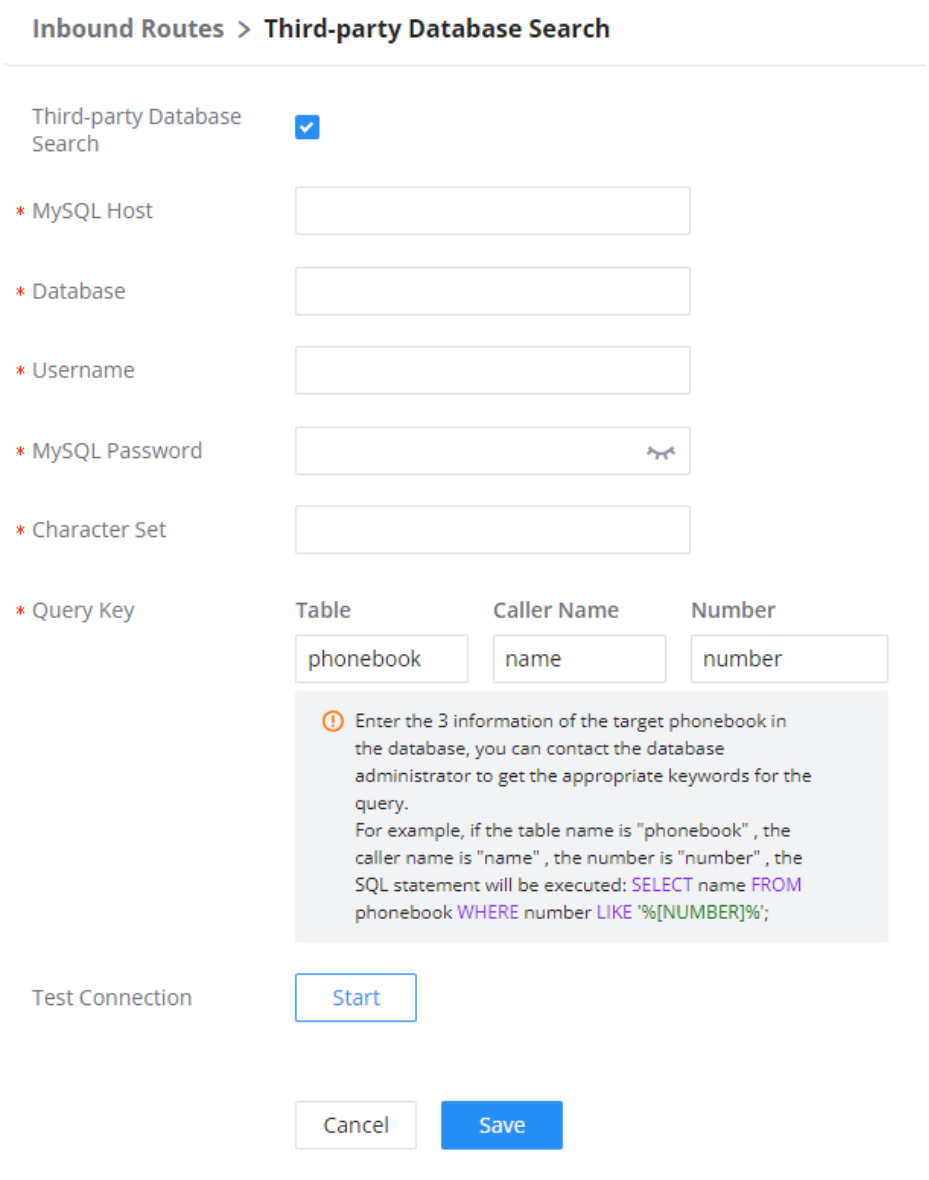
This feature allows the user to enter to integrate the PBX with a third-party database which contains the phone numbers and their matching names. When a call is received on a specific inbound route, the callerID will be checked against the database, if it's found, then the corresponding name will be displayed.

This feature uses MySQL queries, therefore, it will function only with MySQL databases.



Inbound Routes

Once the user clicks on "Third-party Database Search", it will open the configuration page, as seen in the figure below.



Third-Party Database Integration Configuration

Third-party Database Search	Enable or disable the feature.
MySQL Host	Specifies the hostname or IP address of the MySQL server.
Database	The name of the MySQL database that stores caller information.
Username	Enter the username used to connect to the MySQL database.
MySQL Password	Enter the password for the specified MySQL username.
Character Set	Specifies the character set for MySQL connections.
Query Key	Enter the 3 information of the target phonebook in the database, you can contact the database administrator to get the appropriate keywords for the query. For example, if the table name is "phonebook" , the caller name is "name" , the number is "number" , the SQL statement will be executed: SELECT name FROM phonebook WHERE number LIKE '%[NUMBER]%';
Test Connection	Test the connection to the database

BASIC CALL FEATURES

Multimedia Meeting

The IPPBX supports multimedia meeting room allowing multiple rooms used at the same time.

The multimedia meeting room configurations can be accessed under Web GUI→**Basic Call Features**→ **Multimedia Meeting**. On this page, users can create, edit, view, invite, manage the participants, and delete multimedia meeting rooms. The multimedia meeting room status and meeting call recordings (if recording is enabled) will be displayed on this web page as well.

For video meeting, which is based on WebRTC, participants can join the meeting from a PC without installing extra plug-ins or software.

The IPPBX admin can create multiple multimedia meeting rooms for users to dial in.

Meeting room specifications affect user participation to a certain extent. IPPBX supports the forecasting of meeting resources. There will be corresponding judgments and adjustments in the following scenarios:

1. When meeting resources are used up, scheduled meeting members cannot join the meeting in advance.
2. When a point-to-point call is transferred to a conference, the conference resources are used up.
3. When meeting resources are used up, do not join a group IM chat when you initiate a meeting.
4. When meeting resources are used up, do not join an instant meeting.
5. Close other instant meetings or scheduled meetings that have timed out to ensure that invited members can join the scheduled meeting.
6. In an ongoing meeting, if the number of invited members exceeds the upper limit, members cannot be invited to join the meeting.
7. Enable flow control for videos and presentations in the conference room.

The multimedia meeting room supports up to 4 video calls and one video presentation.

- The administrator can set the number of videos to 9 parties. The increase in the number of videos will take up more system resources and affect the overall performance of the IPPBX system. Please set it according to your needs.
- During a meeting, when the system detects that another scheduled meeting is about to be held, it will remind the meeting members that the subsequent meeting room has been reserved, please end the meeting in advance.
- The use of video in the meeting will take up system resources and may cause performance problems when used.
- The maximum meeting duration is 12 hours. If it exceeds 12 hours, the system will remind the current meeting and the host can continue to extend the meeting.

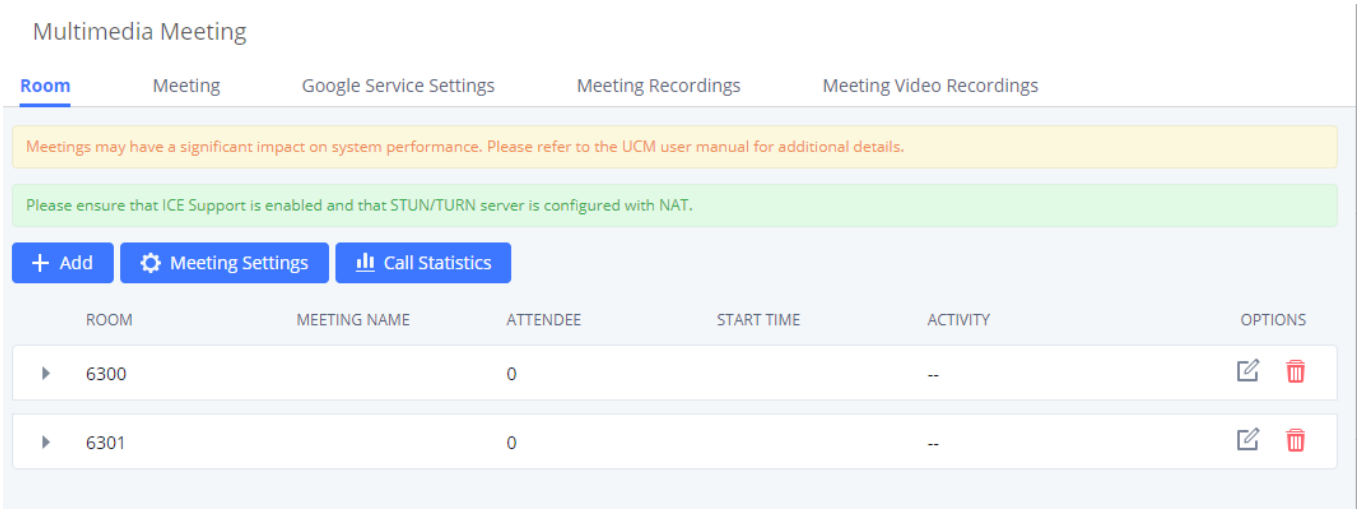
Multimedia Room Configurations

- Click on "Add" to add a new meeting room.
- Click on to edit the meeting room.
- Click on to delete the meeting room.

Meeting Settings contains the following options:

Log in to the IPPBX Web GUI and open the **Basic Call Features → Multimedia Meeting** page to manage the conference room. Users can create, edit, view, invite, manage meeting members, and delete meeting

rooms. The conference room status and conference call recording (if the recording function is enabled) will be displayed on the page. The meeting rooms in the list include public meeting rooms and random meeting rooms. For temporary meeting room administrators, only the "batch kicking people" function is supported. The temporary meeting room has no meeting password or host code. The member who initiates the group meeting is the host, and ordinary members have the right to invite.



Multimedia Meeting

Meetings Settings

To edit the general settings of the meeting rooms created in the IPPBX, the user can click on "Meetings Settings" button under the **Room** tab.

Multimedia Meeting Call Operations

Join a Meeting Call

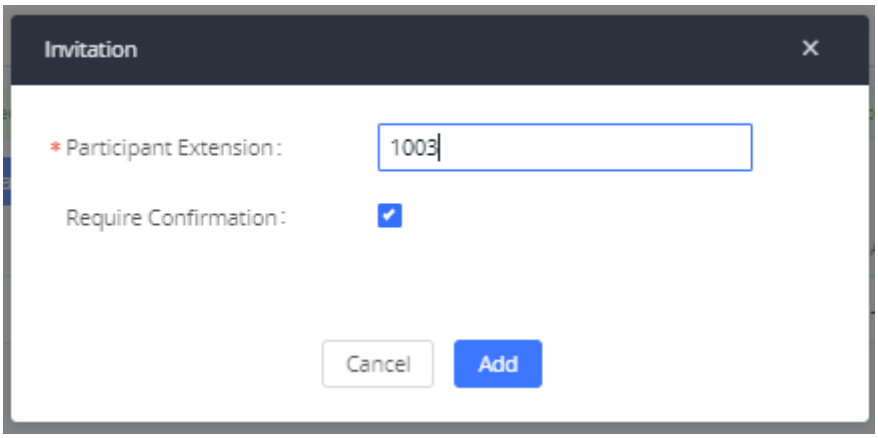
Users could dial the meeting room extension to join the meeting. If the password is required, enter the password to join the meeting as a normal user, or enter the admin password to join the meeting as an administrator.

Invite Other Parties to Join a Meeting

When using the IPPBX meeting room., there are two ways to invite other parties to join the meeting.

- **Invite from Web GUI.**

For each meeting room in PBX Web GUI→**Basic Call Features**→ **Multimedia Meeting**, there is an icon for option "Invite a participant". Click on it and enter the number of the party you would like to invite. Then click on "Add". A call will be sent to this number to join the conference.



Meeting Invitation from Web GUI

- **Invite by dialing 0 or 1 during a conference call.**

A meeting participant can invite other parties to the meeting by dialing from the phone during the meeting call. Please make sure the option "Enable User Invite" is turned on for the meeting room first. Enter 0 or 1 during the meeting call. Follow the voice prompt to input the number of the party you would like to invite. A call will be sent to this number to join the meeting.

0: If 0 is entered to invite another party, once the invited party picks up the invitation call, permission will be asked to "accept" or "reject" the invitation before joining the conference.

1: If 1 is entered to invite another party, no permission will be required from the invited party.

Conference administrators can always invite other parties from the phone during the call by entering 0 or 1. To join a conference room as an administrator, enter the admin password when joining the conference. A conference room can have multiple administrators.

During The Meeting

During the meeting call, users can manage the conference from Web GUI or IVR.

- **Manage the meeting call from Web GUI.**

Log in IPPBX Web GUI during the meeting call, and the participants in each meeting room will be listed.

1. Click on to kick a participant from the meeting.
2. Click on to mute the participant.
3. Click on to lock this meeting room so that other users cannot join it anymore.
4. Click on to invite other users into the meeting room.
5. Click on to Invite meeting rooms or Invite contact groups.

- **Manage the meeting call from IVR.**

Please see the options listed in the table below.

Meeting Administrator IVR Menu	
1	Mute/unmute yourself.
2	Lock/unlock the conference room.
3	Kick the last joined user from the conference.
4	Decrease the volume of the conference call.
5	Decrease your volume.

6	Increase the volume of the conference call.
7	Increase your volume.
8	More options. 1: List all users currently in the conference call. 2: Kick all non-administrator participants from the conference call. 3: Mute/Unmute all non-administrator participants from the conference call. 4: Record the conference call. 8: Exit the caller menu and return to the conference.
Meeting User IVR Menu	
1	Mute/unmute yourself.
4	Decrease the volume of the conference call.
5	Decrease your volume.
6	Increase the volume of the conference call.
7	Increase your volume.
8	Exit the caller menu and return to the conference.

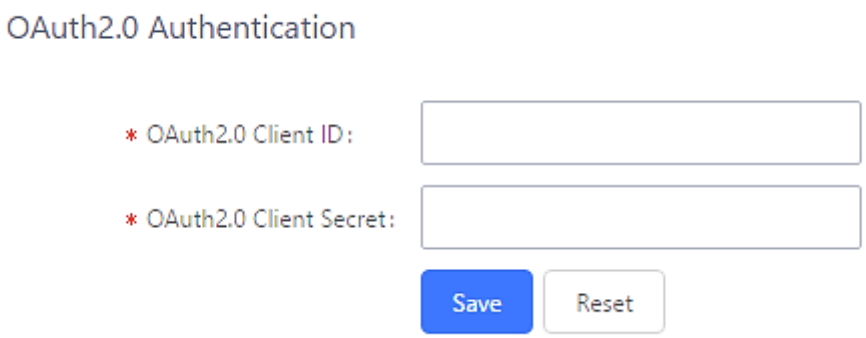
Meeting Caller IVR Menu

When there is a participant in the meeting, the meeting room configuration cannot be modified.

Google Service Settings Support

PBX supports Google OAuth 2.0 authentication. This feature is used for supporting the PBX meeting scheduling system. Once OAuth 2.0 is enabled, the PBX conference system can access Google Calendar to schedule or update conference.

Google Service Settings can be found under Web GUI→**Basic Call Features→ Multimedia Meeting → Google Service Settings→Google Service Settings**.

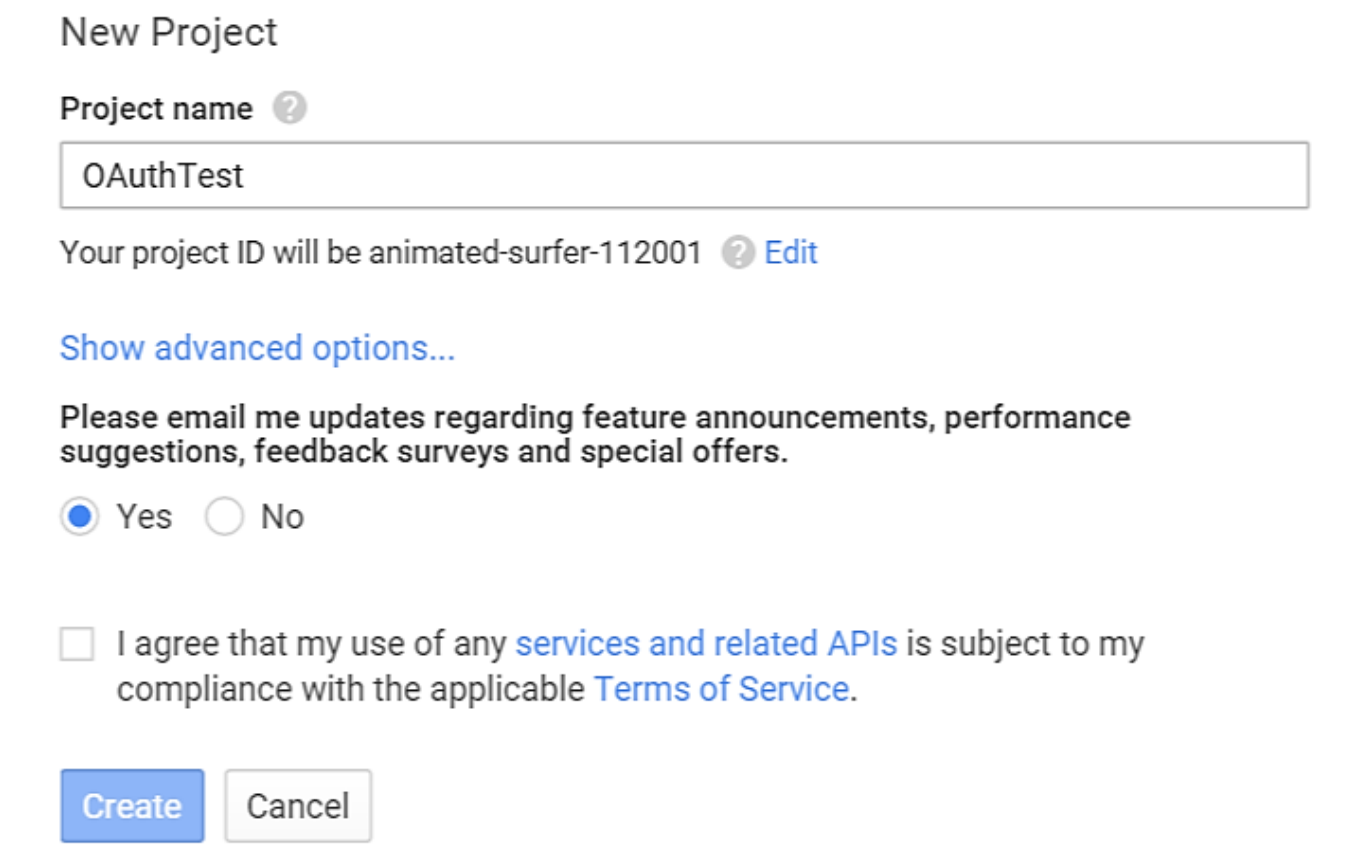


Google Service Settings→OAuth20 Authentication

If you already have an OAuth2.0 project set up on the **Google Developers** web page, please use your existing login credentials for "OAuth2.0 Client ID" and "OAuth2.0 Client Secret" in the above figure for the PBX to access Google Service.

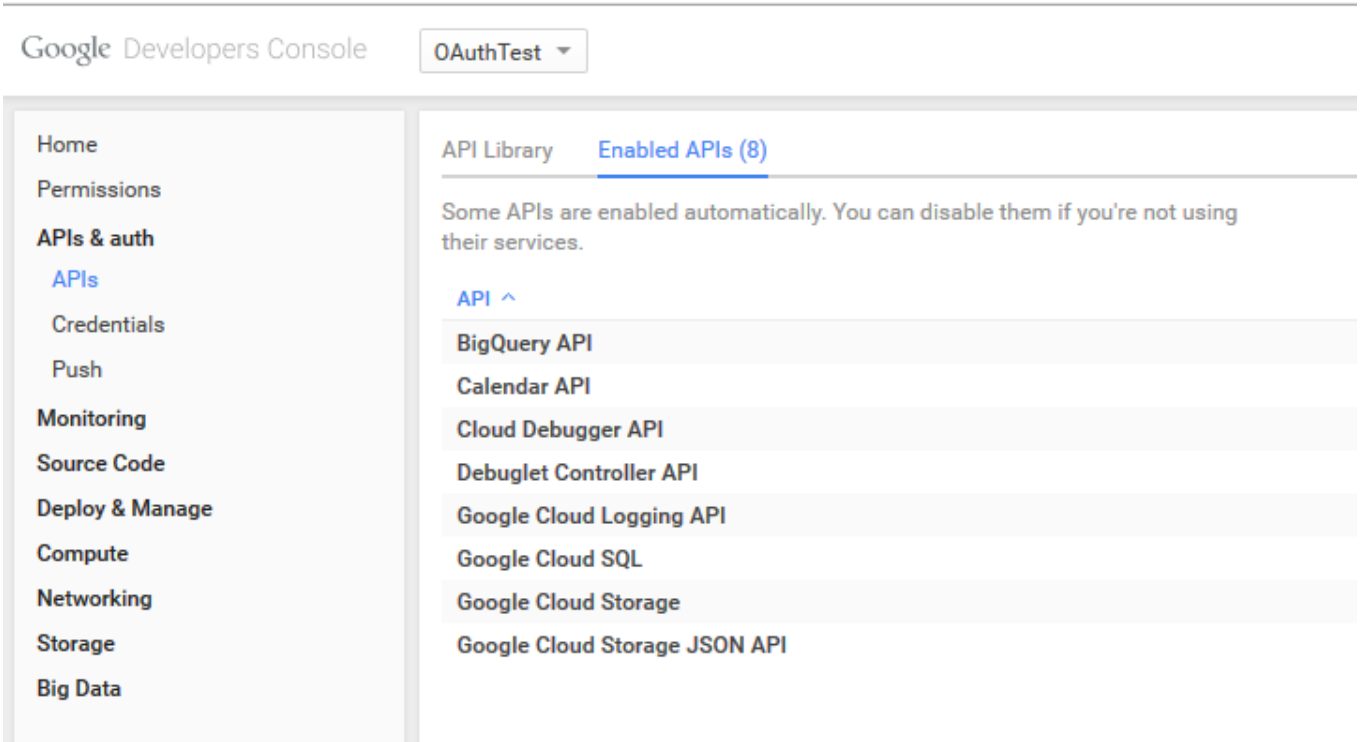
If you do not have the OAuth2.0 project set up yet, please follow the steps below to create a new project and obtain credentials:

1. Go to the Google Developers page https://console.developers.google.com/start Create a New Project on the Google Developers page.



Google Service→New Project

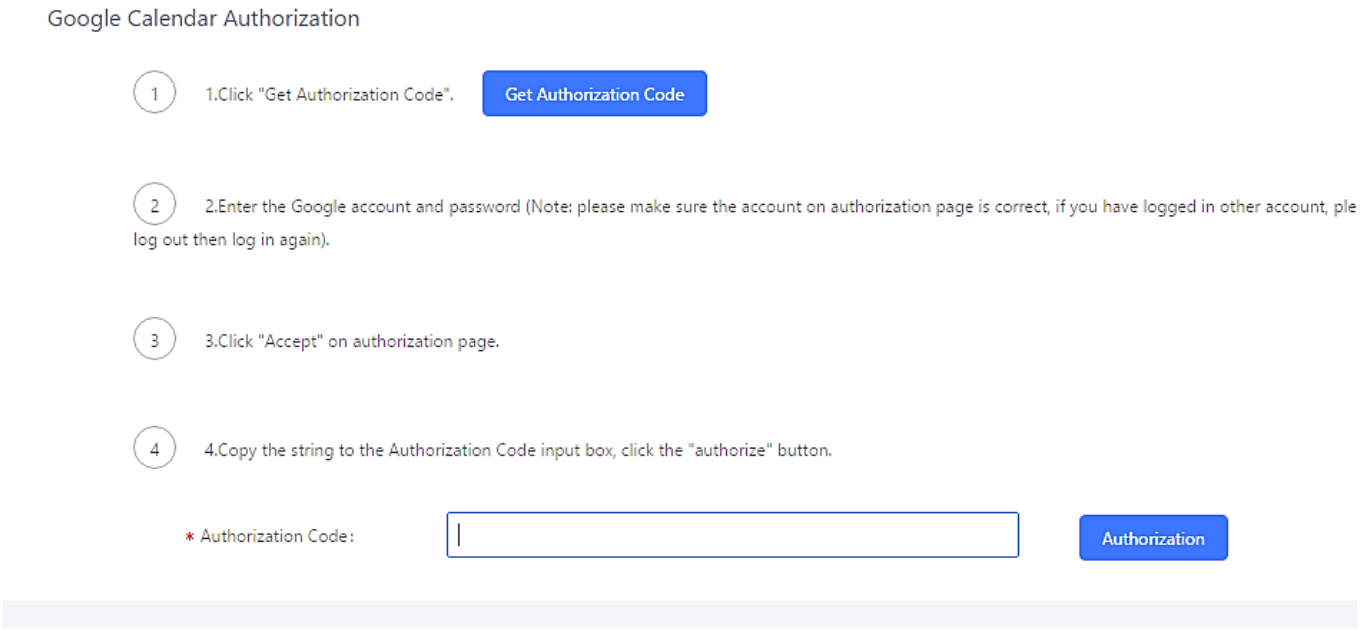
2. Enable Calendar API from API Library.
3. Click "Credentials" on the left drop-down menu to create new OAuth2.0 login credentials.



Google Service→Create New Credential

4. Use the newly created login credential to fill in "OAuth2.0 Client ID" and "OAuth2.0 Client Secret".

5. Click "Get Authentication Code" to obtain an authentication code from Google Service.



Google Service→OAuth20 Login

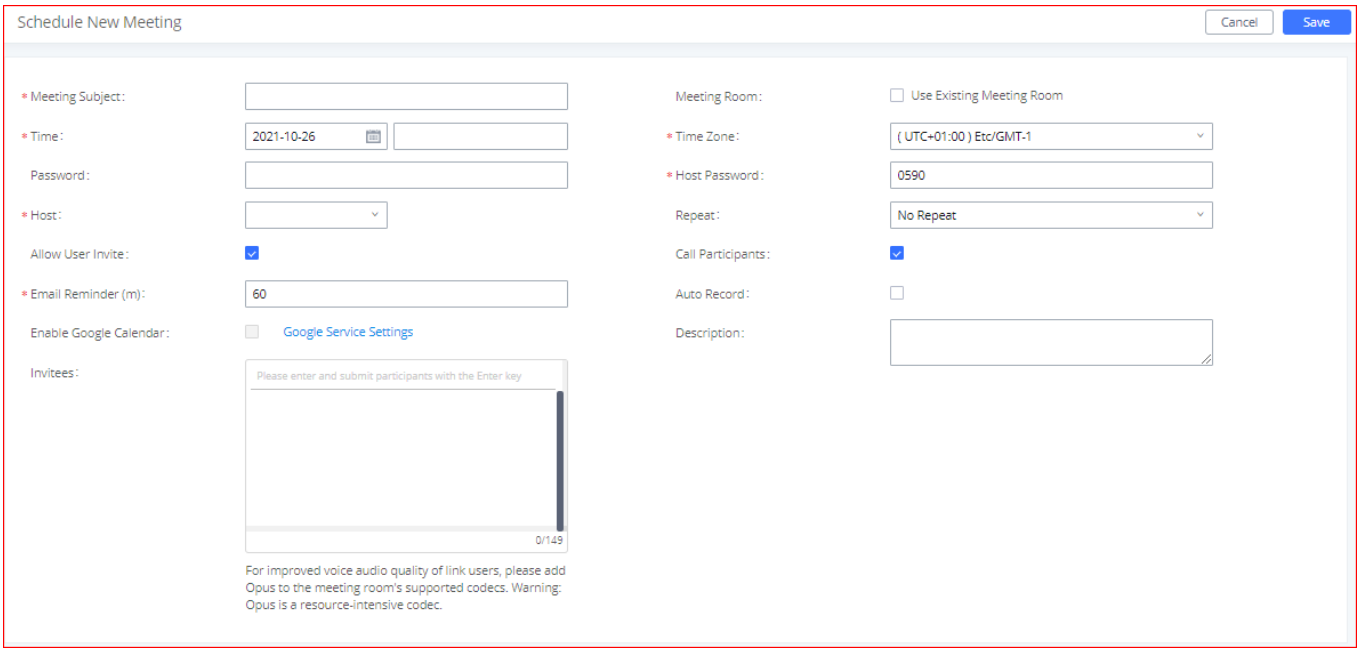
6. Once this has been done, the PBX will connect to Google services.

You can also configure the Status update, which automatically refreshes your Google Calendar with the configured time (m). **Note:** Zero means disable.

Schedule Meeting

Log in to the IPPBX Web GUI, open the **Basic Call Features → Multimedia Meeting → Meeting Schedule** page, and you can manage the reservation management of the meeting room. Users can create, edit, view, and delete meeting room reservation records. The following is a set meeting room reservation, which shows the ongoing and pending reservations. Once the conference room is reserved, all users will be removed from the conference room at the start time, and extensions will no longer be allowed to enter the conference room. At the scheduled meeting time, IPPBX will send invitations to the extensions that have been selected to participate in the meeting. At the same time, it supports users to enter the meeting 10 minutes in advance. If the current meeting is occupied, enter the waiting room and wait (members joining the meeting in advance occupy global member resources, but it will be released after the scheduled meeting starts); otherwise, you can join the meeting directly and the meeting will be held in advance. After the meeting ends, the reservation record is transferred to the historical meeting list. History meeting displays the information of the ended and expired meetings.

- Click the button "Schedule Meeting" to edit the meeting room reservation.

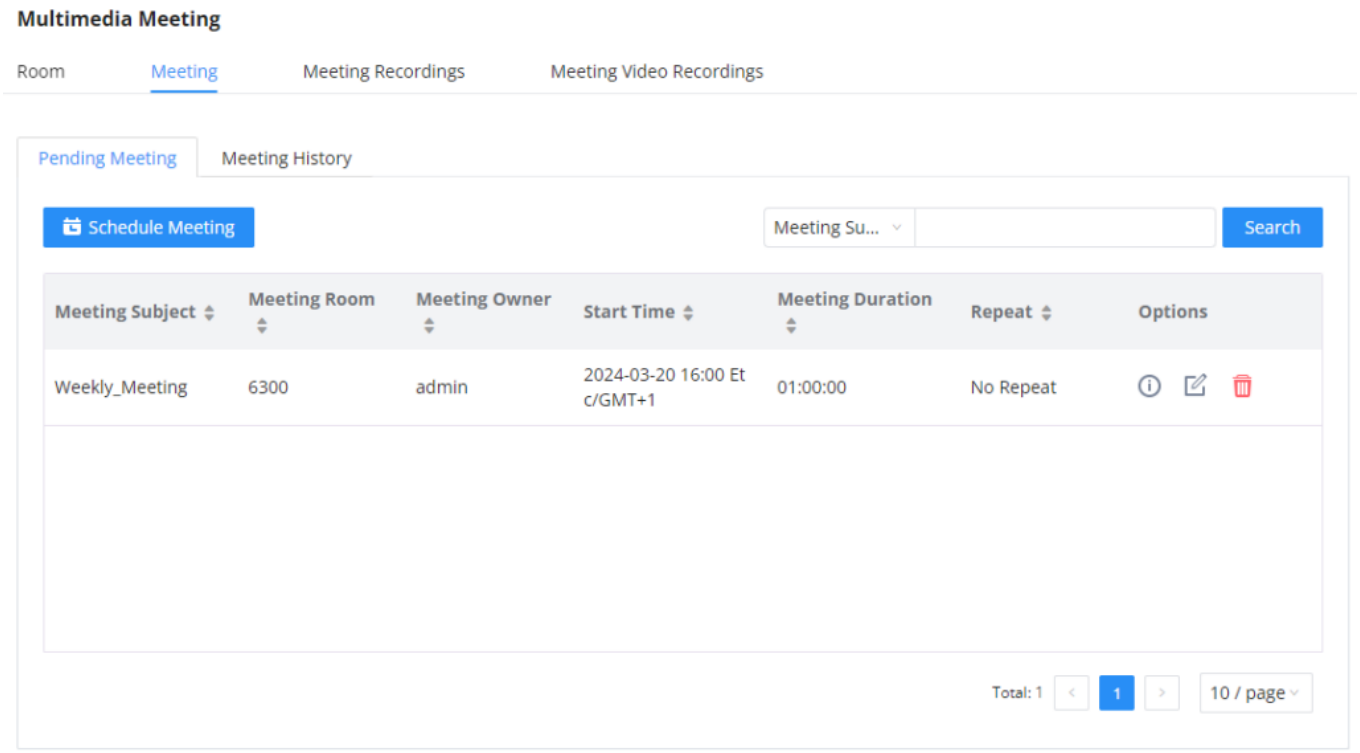


Schedule meeting Interface

Schedule Options	
Meeting Subject	Configure the name of the scheduled meeting. Letters, digits, some special characters are also supported. Characters ` % # ? < > @ $ ^ ~ | are not allowed.
Meeting Room	Choose which room to have this scheduled meeting. If this option has been enabled, please select an existing room for this meeting. If this option has not been enabled, a new meeting room will be created.
Time	Configure the meeting date and time.
Time Zone	Select the meeting time zone.
Password	Configure the meeting's login password.
Host Password	Configure the Host Password.Note: It is randomly generated when first creating a new meeting Schedule.
Host	Configure Host.
Repeat	Choose when to repeat a scheduled meeting. No Repeat Every Day Weekly Monthly Custom: it specifies how often the meeting is repeated per days/weeks. E.g., every 3 days/weeks.
Allow User Invite	If enabled, participants will be able to invite others to the meeting by pressing 1 on their keypad or by clicking the Participants->Invite option.
Call Participant	If enabled, the invited participants will be called upon meeting start time.
Allowed to Override Host Mute	If enabled, participants will be able to unmute themselves if they have been muted by the host.
Email Reminder (m)	Check to enable scheduled meeting email reminder. Email reminders will be sent out x minutes prior to the start of the meeting. Valid range is 5-120. 60 is the default value.Note: After editing the time of a single recurrence of a scheduled meeting, a cancellation email will be sent out followed by a meeting update email.
Auto Record	If selected, the meeting will be recorded and saved as either a .WAV or .MKV file. The default filename is meeting-${Meeting Number}-${UNIQUEID}. Recordings can be downloaded from either the Meeting Recordings or the Meeting Video Recordings page. Video recordings require external storage to be available. When recording a screen share, only the screen share and meeting audio will be recorded.

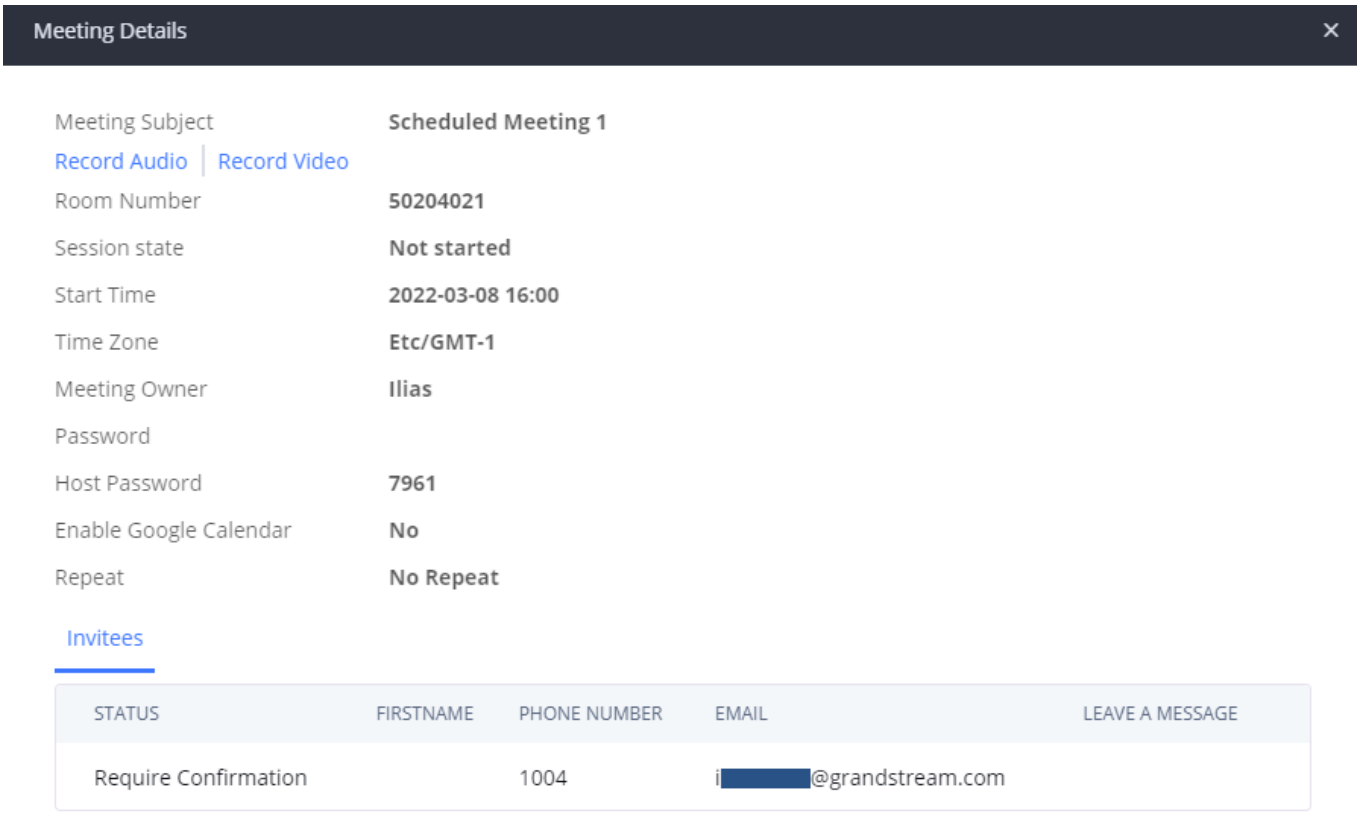
Enable Google Calendar	Select this option to sync scheduled meeting with Google Calendar.Note: Google Service Setting OAuth2.0 must be configured on the IPPBX. Please refer to section [Google Service Settings Support].
Meeting Agenda	Enter information about the meeting, e.g., the purpose of the meeting or the subjects that will be discussed in the meeting.
Invitees	Local extensions, remote extensions, and special extensions are supported.

Once the Meeting Schedule is configured, the scheduled meeting will be displayed as the below figure.



Meetings Schedule

- Click the button to view the meeting details in the Meeting room. The meeting details of Meeting History include actual participant information.

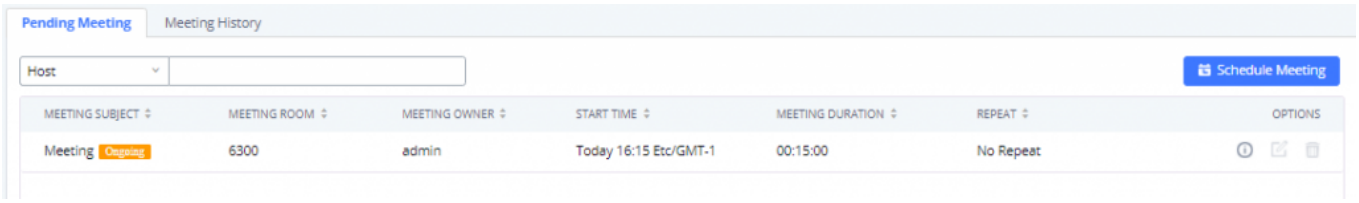


Meeting details

- Click on to edit the Meeting Schedule.
- Click on to delete the Meeting Schedule.

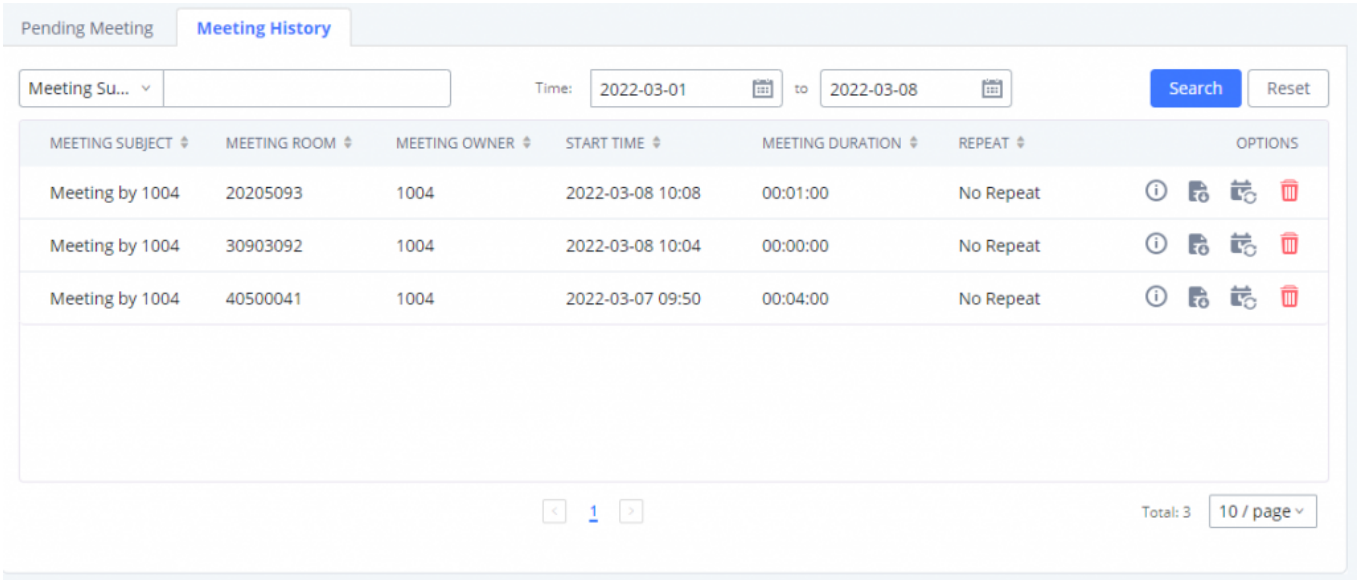
At the scheduled meeting time, PBX will send INVITE to the extensions that have been selected for the conference.

Once the meeting starts, it will be displayed under **Pending Meeting** with an "Ongoing" status, as displayed below:



Meeting Scheduled Ongoing

Once the conference is finished, the conference will be displayed under Historical Meeting as below:

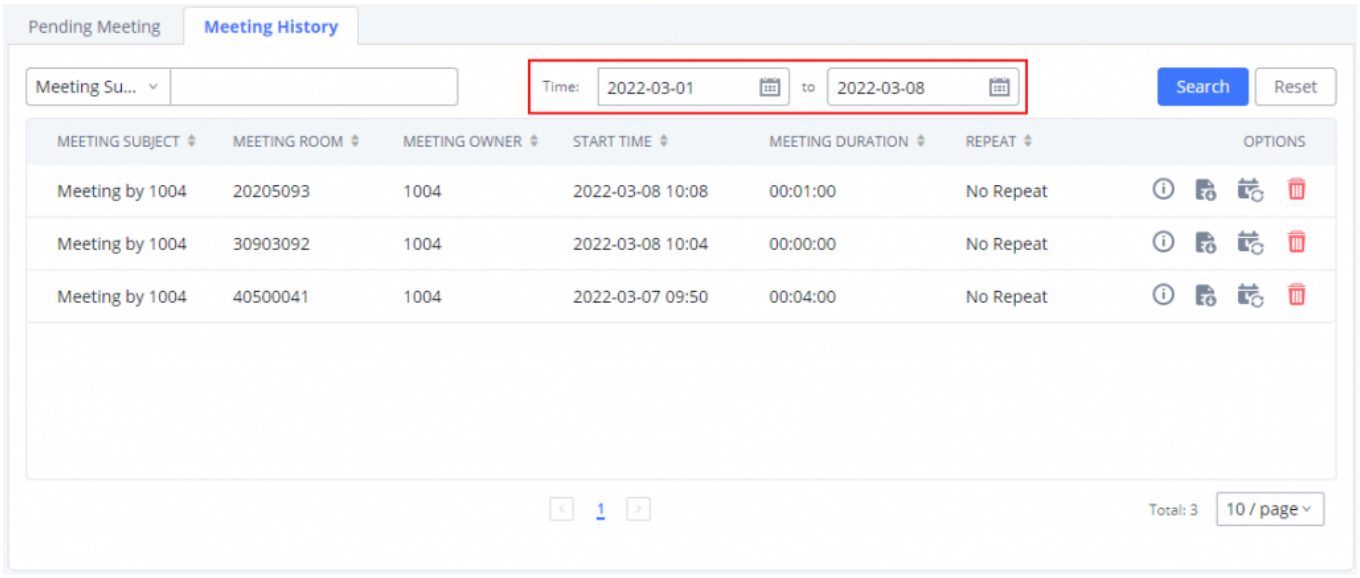


Meeting Schedule Completed

- Click the button to download the Meeting Report of the meeting.
- Click the button to reschedule the Meeting.

In addition, once the meeting ends, the system will send a meeting report email to the host including PDF file where he/she can view the meeting, participant information, device type and trend graph of participant levels.

You can also choose to display the meetings that took place in a specific time frame. Please see the screenshot below:



Meeting History

Please make sure that the outbound route is properly configured for remote extensions to join the meeting.

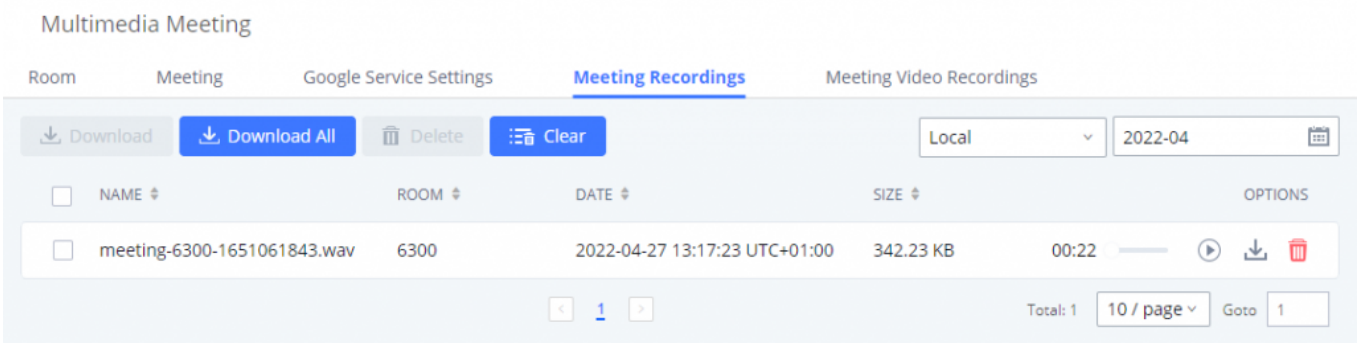
Meeting Recordings

The PBX allows users to record the audio of the meeting call and retrieve the recording from Web GUI→ **Basic Call Features→ Multimedia Meeting→ Meeting Recordings**.

To record the Meeting call, when the meeting room is idle, enable "Auto Record" from the meeting room

configuration dialog. Save the setting and apply the change. When the meeting call starts, the call will be automatically recorded in .wav format.

The recording files will be listed below once available. Users could click on to download the recording or click on to delete the recording. Users could also delete all recording files by clicking on "Delete All Recording Files" or delete multiple recording files at once by clicking on "Delete" after selecting the recording files.



Meeting Recordings

Meeting Video Recordings

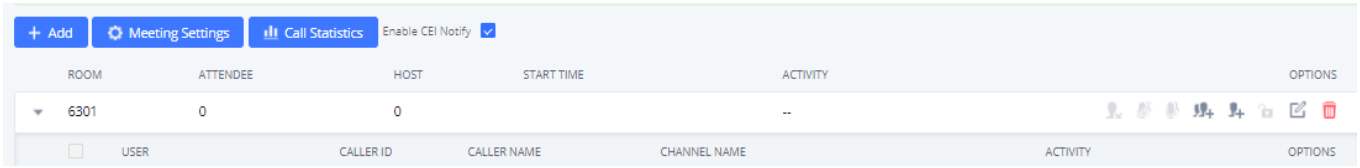
The PBX allows users to record the audio and video of the meeting call and retrieve the recording from Web GUI→**Basic Call Features→ Multimedia Meeting→ Meeting Recordings**.

To record the Meeting call, when the meeting room is idle, enable "Auto Record" from the meeting room configuration dialog. Save the setting and apply the change. When the meeting call starts, the call will be automatically recorded in .mkv format.

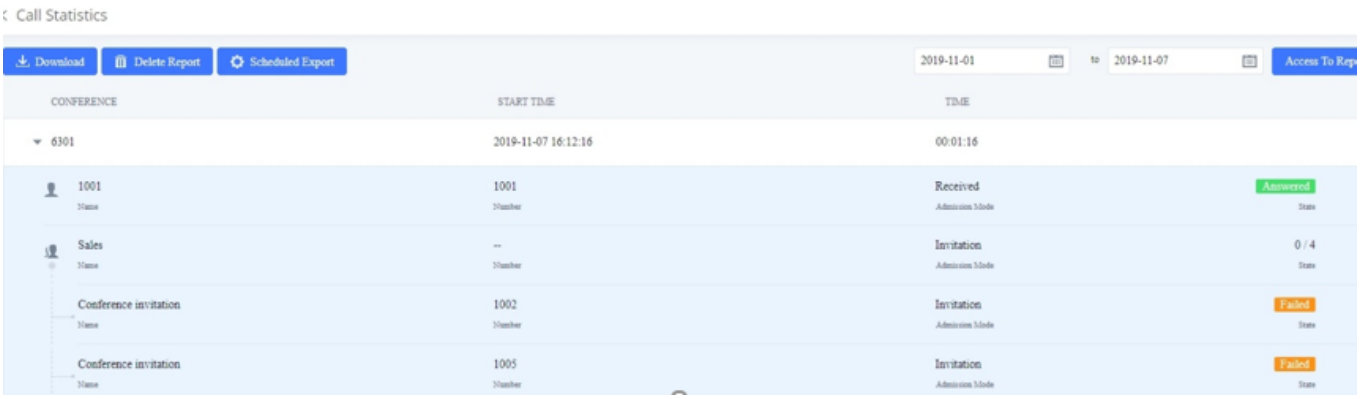
The recording files will be listed below once available. Users could click on to download the recording or click on to delete the recording. Users could also delete all recording files by clicking on "Delete All Recording Files" or delete multiple recording files at once by clicking on "Delete" after selecting the recording files.

Call Statistics

Meeting reports will now be generated after every conference. These reports can be exported to a .CSV file for offline viewing. The conference report page can be accessed by clicking on the Call Statistics button on the main Conference page.



Meeting Call Statistics



Meeting Report on Web

	A	B	C	D	E	F	G
1	Room	Start Time	Duration Time				
2	6301	11/7/2019 16:12	0:01:16				
3	Contact Number	Name	Way	Status	Contact Group Name		
4	1002	Conference invitatio	INVITE	FAILURE	Sales		
5	1005	Conference invitatio	INVITE	FAILURE	Sales		
6	1004	Conference invitatio	INVITE	FAILURE	Sales		
7	1003	Conference invitatio	INVITE	FAILURE	Sales		
8	1001	1001	CALLIN	ANSWER			

Meeting Report on CSV

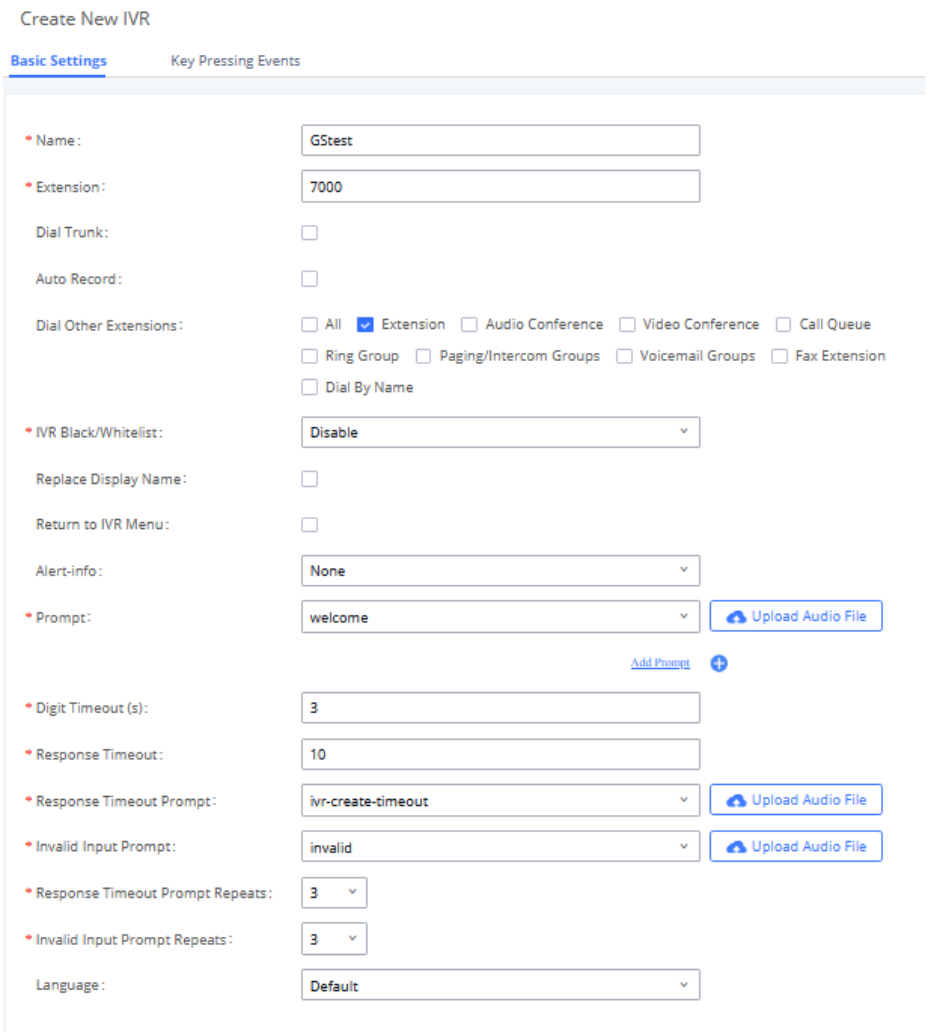
IVR

IVR (Interactive Voice Response) is a feature that enables triggering certain actions by using DTMF. The recorded prompt of the IVR instructs the numbers and the corresponding actions to the caller, when the caller press a certain number, the PBX will trigger the equivalent action.

Configure IVR

IVR configurations can be accessed under the PBX Web GUI→**Basic Call Features→IVR**. Users could create, edit, view, and delete an IVR.

- Click on "Add" to add a new IVR.
- Click on to edit the IVR configuration.
- Click on to delete the IVR.

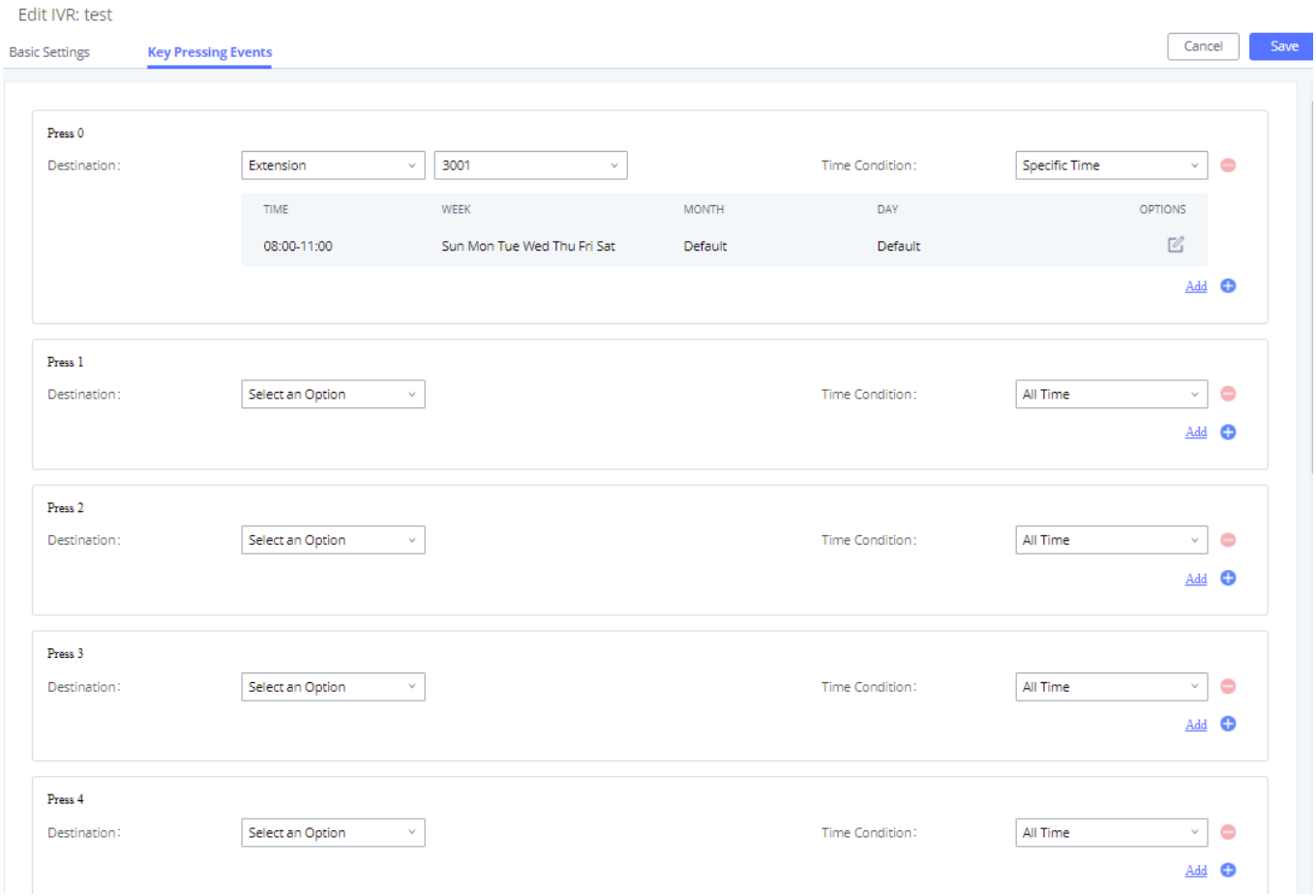


Create New IVR

Basic Settings	
Name	Configure the name of the IVR. Letters, digits, _ and - are allowed.
Extension	Enter the extension number for users to access the IVR.
Dial Trunk	If enabled, all callers to the IVR can use the trunk. The permission must be configured for the users to use the trunk first. The default setting is "No".
Auto Record	If enabled, calls to this IVR will automatically be recorded.
Permission	Assign permission level for outbound calls if "Dial Trunk" is enabled. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the IVR, the PBX will compare the IVR's permission level with the outbound route's privilege level. If the IVR's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.
Dial Other Extensions	This controls the destination that can be reached by the external caller via the inbound route. The available destinations are: Extension Conference Call Queue Ring Group Paging/Intercom Groups Voicemail Groups Dial by Name All
IVR Black/Whitelist	If enabled only numbers inside of the Whitelist or outside of the Blacklist can be called from IVR.
Internal Black/Whitelist	Contain numbers, either of Blacklist or Whitelist.

External Black/Whitelist	This feature can be used only when Dial Trunk is enabled, it contains external numbers allowed or denied calling from the IVR, the allowed format is the following: Number1, number2, number3...
Replace Display Name	If enabled, the PBX will replace the caller display name with the IVR name.
Return to IVR Menu	If enabled and if a call to an extension fails, the caller will be redirected to the IVR menu.
Alert Info	When present in an INVITE request, the alert-Info header field specifies an alternative ring tone to the UAS.
Prompt	Select an audio file to play as the welcome prompt for the IVR. Click on "Prompt" to add an additional audio file under Web GUI→PBX Settings→Voice Prompt→Custom Prompt.
Digit Timeout	Configure the timeout between digit entries. After the user enters a digit, the user needs to enter the next digit within the timeout. If no digit is detected within the timeout, the PBX will consider the entries complete. The default timeout is 3s.
Response Timeout	After playing the prompts in the IVR, the PBX will wait for the DTMF entry within the timeout (in seconds). If no DTMF entry is detected within the timeout, a timeout prompt will be played. The default setting is 10 seconds.
Response Timeout Prompt	Select the prompt message to be played when the timeout occurs.
Invalid Input Prompt	Select the prompt message to be played when an invalid extension is pressed.
Response Timeout Prompt Repeats	Configure the number of times to repeat the prompt if no DTMF input is detected. When the loop ends, it will go to the timeout destination if configured, or hang up. The default setting is 3.
Invalid Input Prompt Repeats	Configure the number of times to repeat the prompt if the DTMF input is invalid. When the loop ends, it will go to the invalid destination if configured, or hang up. The default setting is 3.
Language	Select the voice prompt language to be used for this IVR. The default setting is "Default" which is the selected voice prompt language under Web GUI→PBX Settings→Voice Prompt→Language Settings. The dropdown list shows all the currently available voice prompt languages on the PBX. To add more languages in the list, please download the voice prompt package by selecting "Check Prompt List" under Web GUI→PBX Settings→Voice Prompt→Language Settings.
Key Pressing Events	
Key Press Event: Press 0 Press 1 Press 2 Press 3 Press 4 Press 5 Press 6 Press 7 Press 8 Press 9 Press *	Select the event for each key pressing for 0-9, *, Timeout, and Invalid. The event options are: Extension Voicemail Multimedia Meeting Voicemail Group IVR Ring Group Queues Page Group Custom Prompt Hangup DISA Dial by Name External Number Callback For each key event, time conditions can be configured. At the configured time condition, this IVR key event can be triggered. Office time, holiday time, or specific time can be configured for time conditions. Up to 5 time conditions can be added for each key. The available time conditions are 'All', 'Office Time', 'Out of Office Time', 'Holiday', 'Out of Holiday', 'Out of Office Time or Holiday', 'Office Time and Out of Holiday', and 'Specific Time'. If 'Specific Time' is selected, a new window will prompt for admin to configure start time, end time, and frequency.
Timeout	When exceeding the number of defined answer timeout, IVR will enter the configured event when timeout. If not configured, then it will hang up.
Invalid	Configure the destination when the Invalid Repeat Loop is done.
Time Condition	Configure the time condition for each key press event, so that it goes to the corresponding destination within a specified time.

IVR Configuration Parameters



Key Pressing Events

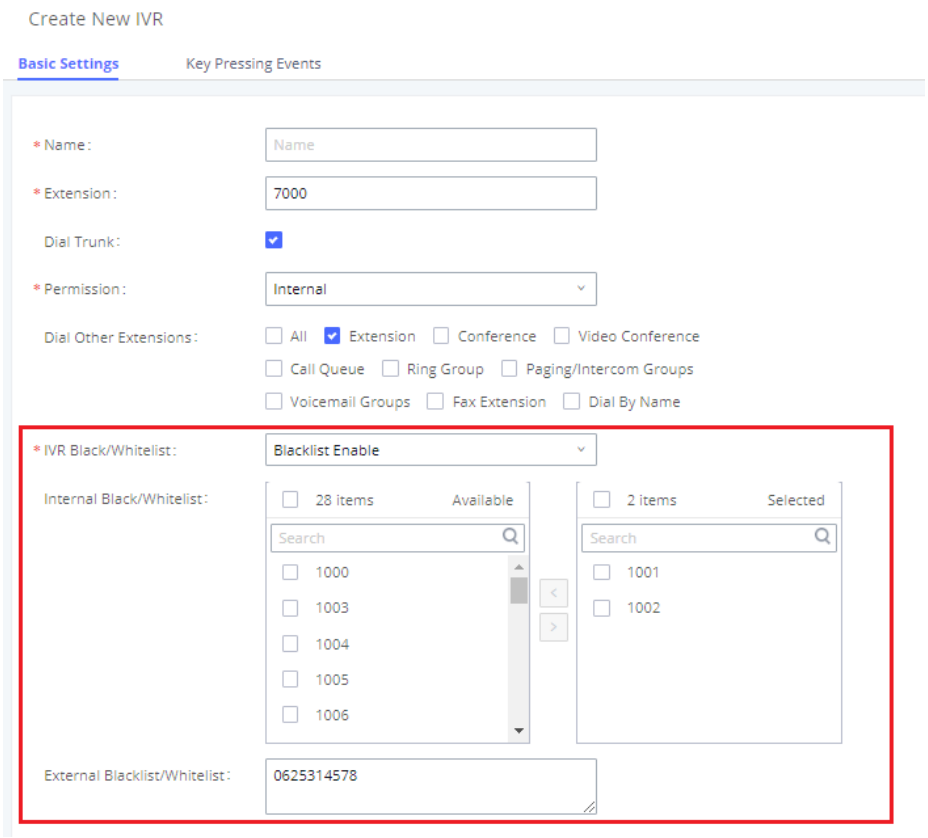
Black/Whitelist in IVR

In some scenarios, the IPPBX administrator needs to restrict the extensions that can be reached from IVR. For example, the company CEO and directors prefer only receiving calls transferred by the secretary, and some special extensions are used on IP surveillance endpoints which should not be reached from external calls via IVR for privacy reasons. The PBX has now added blacklist and whitelist in IVR settings for users to manage this.

Up to 500 extensions are allowed on the back/whitelist.

To use this feature, log into the PBX Web GUI and navigate to **Basic Call Features→IVR→**Create/Edit IVR: IVR Black/Whitelist.

- If the user selects "Blacklist Enable" and adds an extension to the list, the extensions in the list will not be allowed to be reached via IVR.
- If the user selects "Whitelist Enable" and adds an extension to the list, only the extensions in the list can be allowed to be reached via IVR.



BlackWhitelist

Create Custom Prompt

To record a new IVR prompt or upload IVR prompt to be used in IVR, click on "Upload Audio File" next to the "Welcome Prompt" option and the users will be redirected to the Custom Prompt page. Or users could go to Web GUI→**PBX Settings→Voice Prompt→Custom Prompt** page directly.

Click on Prompt to Create IVR Prompt

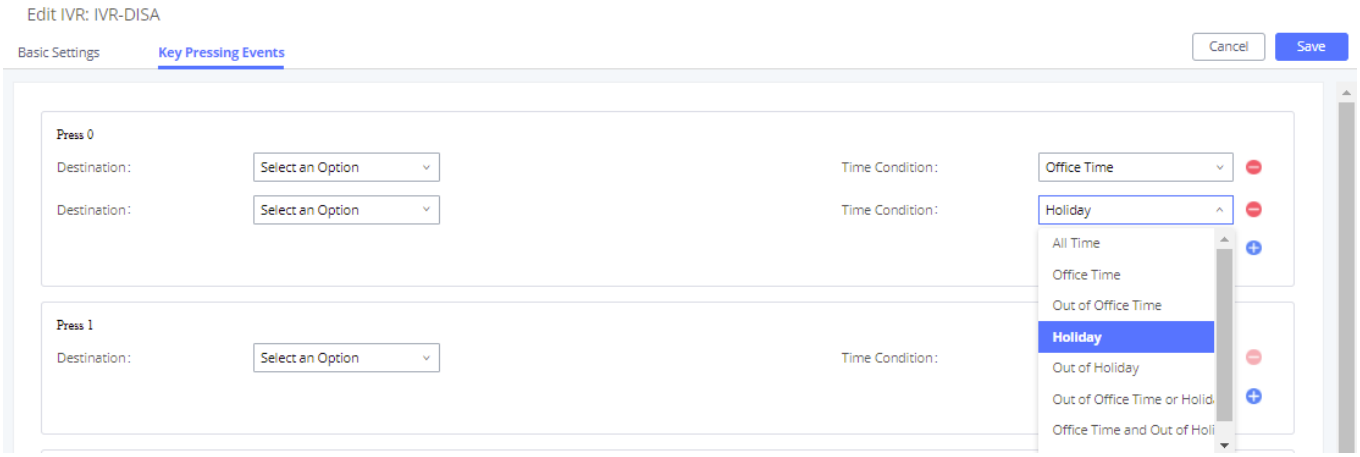
Once the IVR prompt file is successfully added to the PBX, it will be added to the prompt list options for users to select in different IVR scenarios.

Key Pressing Events

Standard Key Event

The PBX supports adding time conditions for different key events so that each key event of the IVR goes to the corresponding destination within a specified time.

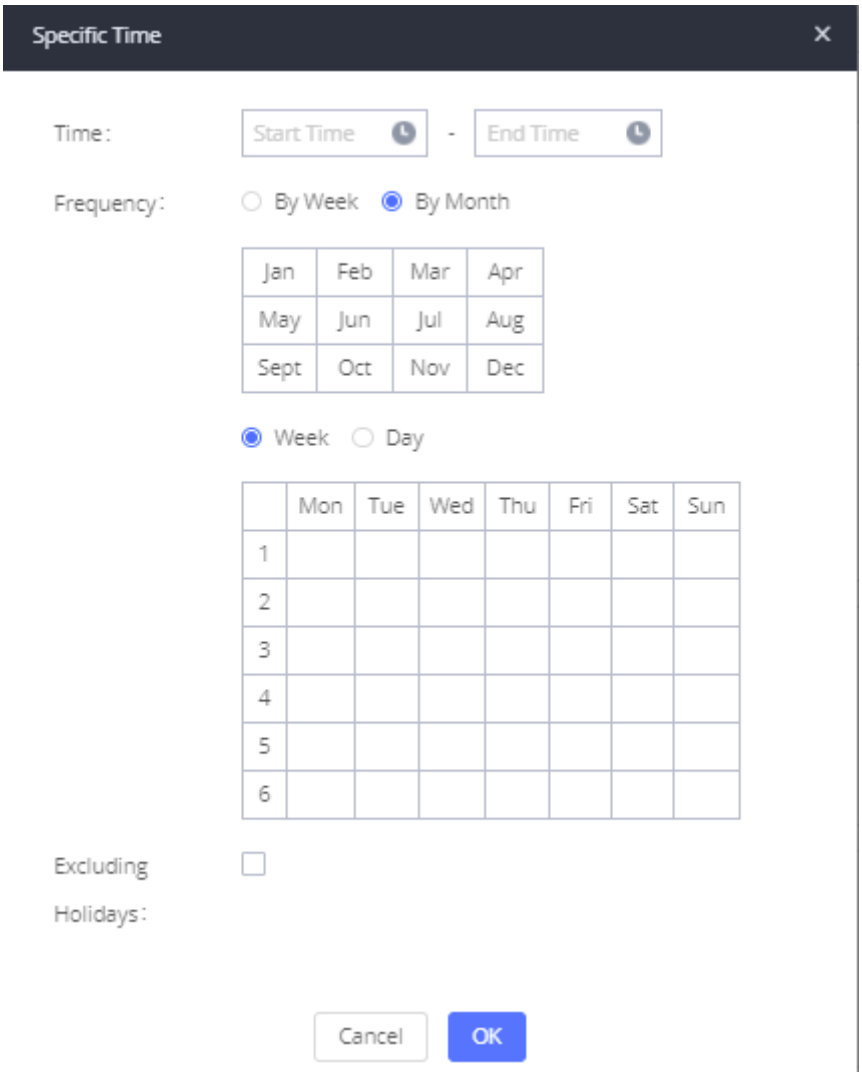
Each key event supports up to five time conditions, the options available are: All time, Office Time, Out of Office Time, Holiday, Out of Holiday, Out of Office Time or Holiday, Office Time and Out Of Holiday, Specific time.



Key Pressing Events

If you select "Specific time", you need to select the start time and the end time.

The frequency supports two options: By week and By Month, by default, the specific time does not include the holidays.



Specific Time

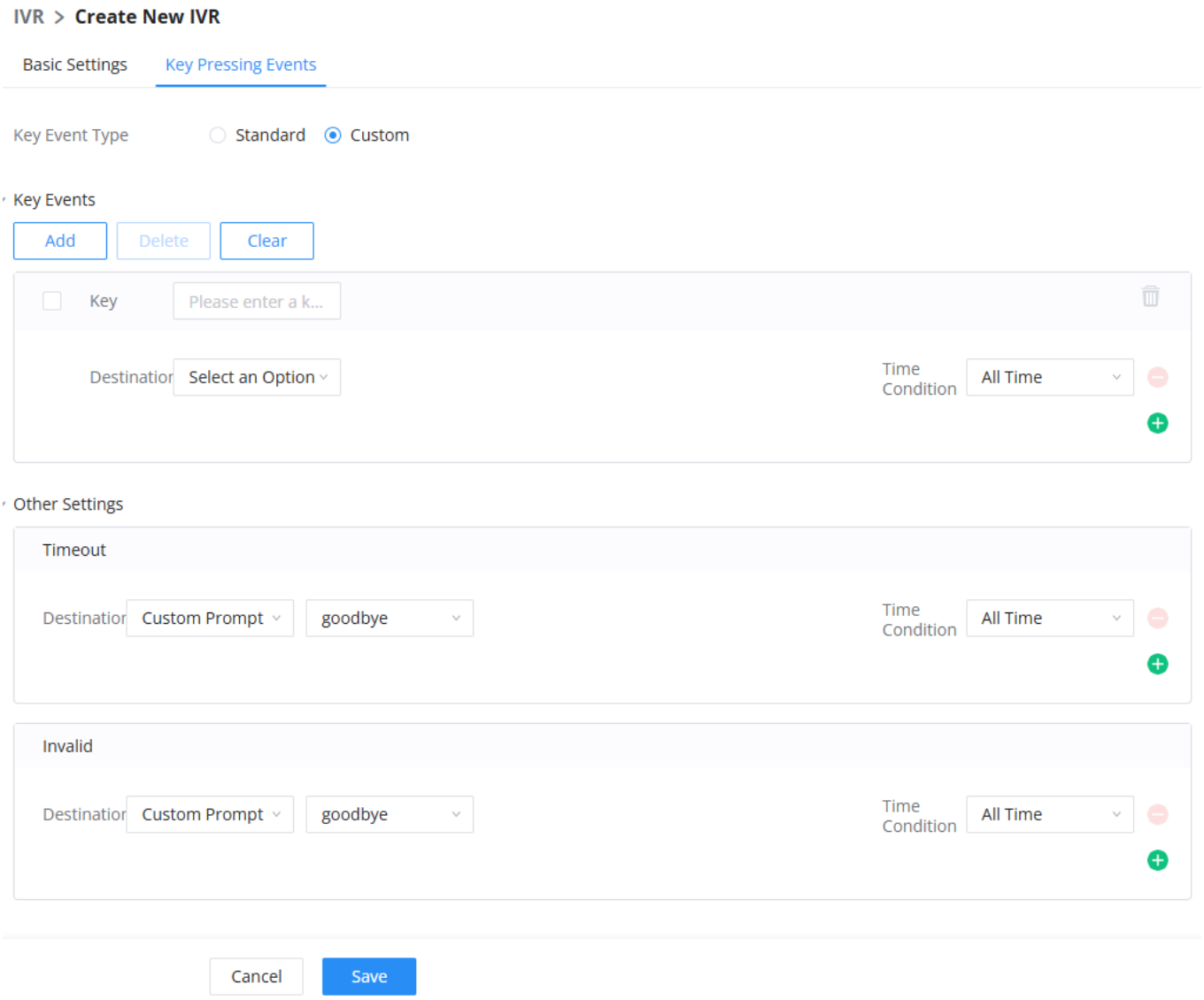
The following table describes available options that can be assigned for Standard Key Pressing Events:

Key Event Type	Select Standard to configure single-key DTMF events (0-9, *). This mode supports time-based routing per key.

Key (0-9, *)	Each section corresponds to a specific DTMF key (for example, Key 0, Key 2). Configuration under a key defines how calls are routed when that key is pressed.
Destination	Select the destination type and target for the key event. Supported destinations include:ExtensionMeetingVoicemailVoicemail GroupIVR (Maximum 64 nested IVRs)Ring GroupQueuesPaging/Intercom GroupFaxCustom PromptHang-upDISADial By NameExternal NumberCallbackAnnouncement
Time Condition	Assign a time condition to each destination entry. The IVR evaluates the time conditions in order and routes the call to the first matching rule. Common options include All Time, Office Time, Out of Office Time, Holiday, and Specific Time. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for the same key, or the (-) icon to remove an existing rule.
Timeout	Configures the destination when the Timeout Repeat Loop is done. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for this option, or the (-) icon to remove an existing rule.
Invalid	Configures the destination when the Invalid Repeat loop is done. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for this option, or the (-) icon to remove an existing rule.

Custom Key Event

Users can create custom IVR key press events, vastly increasing the options a business can provide to its customers and improving customer relations and accessibility.



Key Pressing Events

This new feature supports the following:

- Up to 100 custom key press events
- Each key combination can contain up to 8 characters (numbers and star (*) only)
- Supports Time Conditions
- Different custom keys can have the same Destination and Time Condition

IVR option **Dial Other Extensions** will be disabled if using custom IVR keys.

The following options are available for Custom Key Pressing Events:

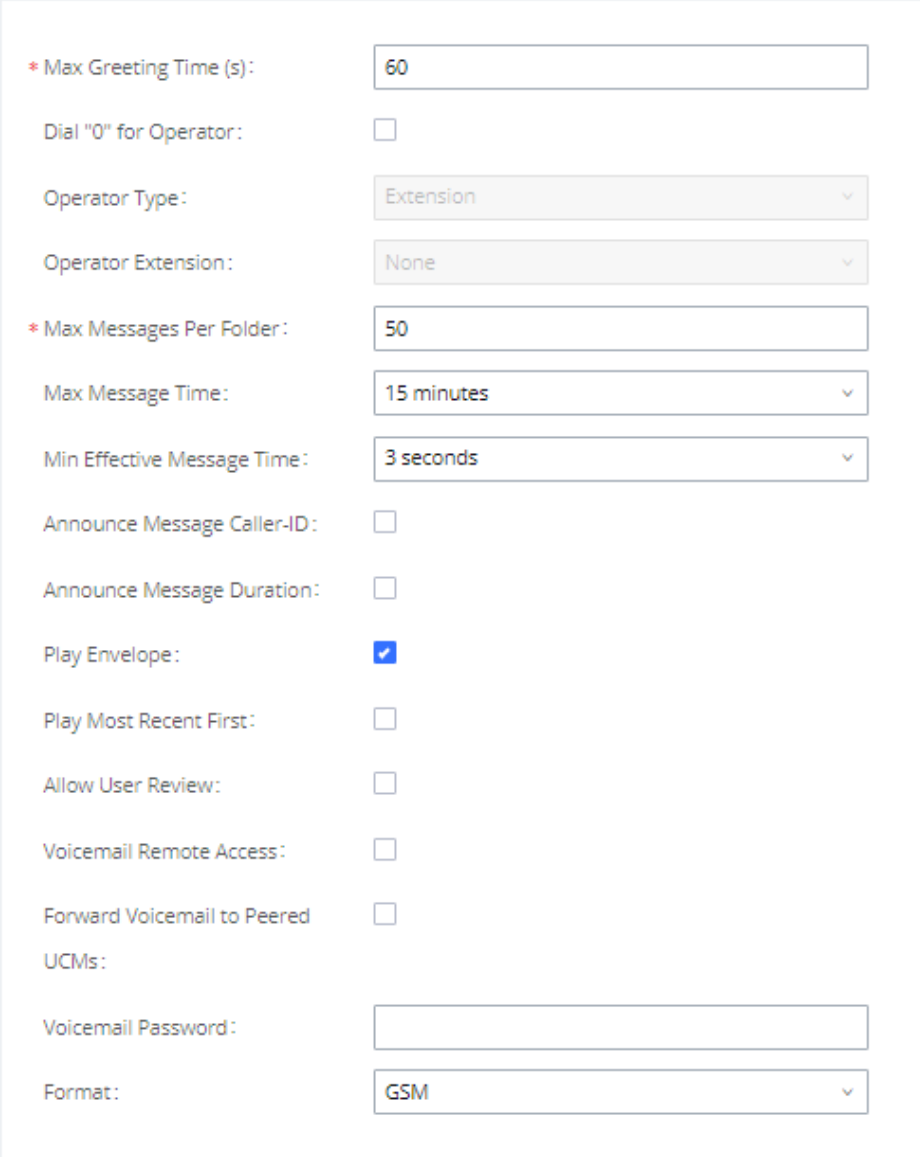
Key Event Type	Select "Custom" to configure multi-digit DTMF key combinations. This mode supports time-based routing and advanced IVR logic.
Key	Defines the DTMF key combination that triggers the event. Notes:Each key combination can contain up to 8 charactersSupported characters: 0-9 and *Up to 100 custom key events are supported per IVR
Destination	Select the destination type and target for the custom key event. Supported destinations include:ExtensionMeetingVoicemailVoicemail GroupIVR (Maximum 64 nested IVRs)Ring GroupQueuesPaging/Intercom GroupFaxCustom PromptHang-upDISADial By NameExternal NumberCallbackAnnouncement

Time Condition	Assign a time condition to each destination entry. The IVR evaluates the time conditions in order and routes the call to the first matching rule. Common options include All Time, Office Time, Out of Office Time, Holiday, and Specific Time. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for the same key, or the (-) icon to remove an existing rule.
Timeout	Configures the destination when the Timeout Repeat Loop is done. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for this option, or the (-) icon to remove an existing rule.
Invalid	Configures the destination when the Invalid Repeat loop is done. Note: A single key can contain multiple destination entries with different time conditions (up to five). Users can click the (+) icon to add another destination with a different time condition for this option, or the (-) icon to remove an existing rule.

Voicemail

Configure Voicemail

If the voicemail is enabled for PBX extensions, the configurations of the voicemail can be globally set up and managed under **Basic Call Features→Voicemail**.



Voicemail Settings

Max Greeting Time (s)	Configure the maximum number of seconds for the voicemail greeting. The default setting is 60 seconds.
Dial '0' For Operator	If enabled, the caller can press 0 to exit the voicemail application and connect to the configured operator's extension.
Operator Type	Configure the operator type; either an extension or a ring group.
Operator Extension	Select the operator extension, which will be dialed when users press 0 to exit the voicemail application. The operator extension can also be used in IVR.
Max Messages Per Folder	Configure the maximum number of messages per folder in users' voicemail. The valid range is 10 to 1000. The default setting is 50.
Max Message Time	Select the maximum duration of the voicemail message. The message will not be recorded if the duration exceeds the maximum message time. The default setting is 15 minutes. The available options are: 1 minute 2 minutes 5 minutes 15 minutes 30 minutes Unlimited
Min Effective Message Time	Configure the minimum duration (in seconds) of a voicemail message. Messages will be automatically deleted if the duration is shorter than the Min Message Time. The default setting is 3 seconds. The available options are: No minimum 1 second 2 seconds 3 seconds 4 seconds 5 seconds Note: Silence and noise duration are not counted in message time.
Announce Message Caller-ID	If enabled, the caller ID of the user who has left the message will be announced at the beginning of the voicemail message. The default setting is "No".
Announce Message Duration	If enabled, the message duration will be announced at the beginning of the voicemail message. The default setting is "No".
Play Envelope	If enabled, a brief introduction (received time, received from, etc.) of each message will be played when accessed from the voicemail application. The default setting is "Yes".
Play Most Recent First	If enabled, it will play the most recent message first.
Allow User Review	If enabled, users can review the message following the IVR before sending.
Voicemail Remote Access	If enabled, external callers routed by DID and reaching VM will be prompted by the PBX with 2 options: Press 1 to leave a message. To leave a message for the extension reached by DID. Press 2 to access the voicemail management system. This will allow the caller to access any extension VM after entering the extension number and its VM password. Note: This option applies to inbound calls routed by DID only. The default setting is "Disabled".
Forward Voicemail to Peered IPPBX	Enables the forwarding of voicemail to remote extensions on peered SIP trunks. The default setting is "Disabled".
Voicemail Password	Configures the default voicemail password that will be used when an extension is reset.
Format	Warning: WAV files take up significantly more storage space than GSM files.

Voicemail Settings

Resetting an extension will reset Voicemail Password, Send Voicemail to Email, and Keep Voicemail after Emailing values to default. Previous custom voicemail prompts and messages will be deleted.

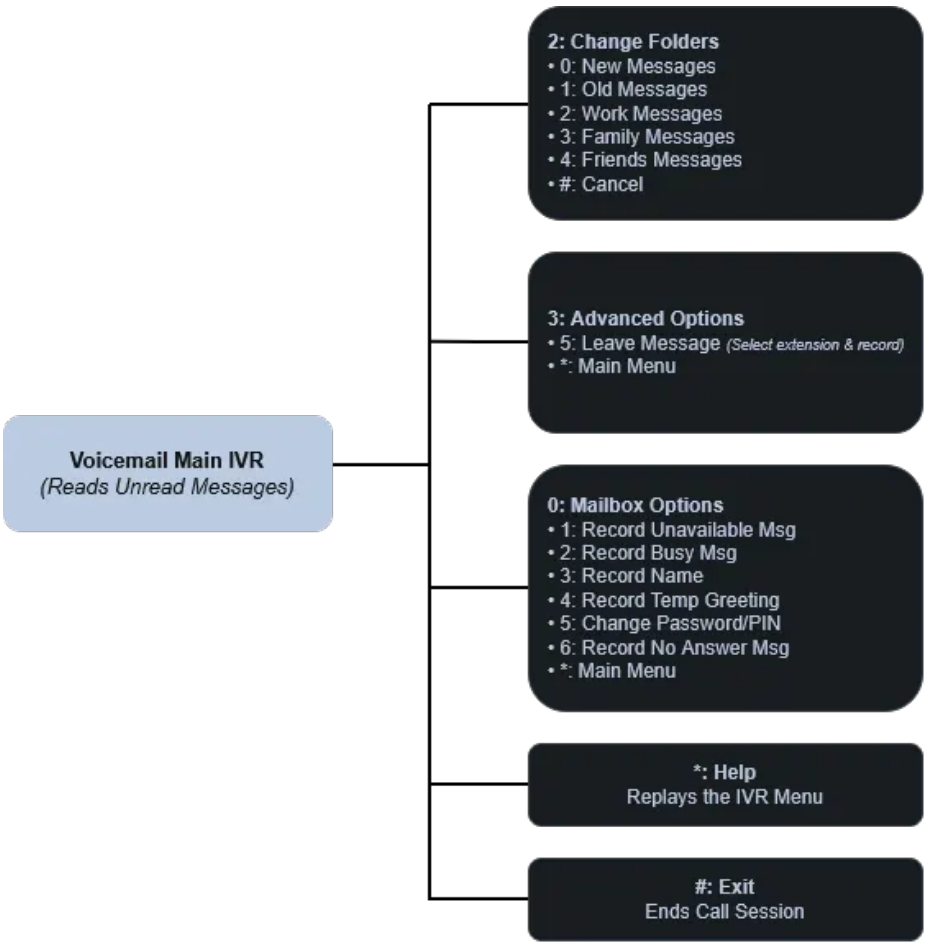
Access Voicemail

When voicemail is enabled for an extension, users can access voicemail using two distinct feature codes:

- **My Voicemail:** Dialing the dedicated feature code for the user's own extension (by default ***97**) directly accesses the user's mailbox without requiring the extension number to be entered. The user only needs to enter their voicemail password/PIN.
- **Voicemail Access:** Dialing the general voicemail feature code (by default ***98**) allows accessing any extension's mailbox. Users can dial ***98** alone and enter the target extension number when prompted, or dial ***98** directly followed by the target extension number (e.g., *981005), followed by the mailbox password/PIN.

After authenticating, both feature codes connect to the same interactive Voicemail IVR menu, which first announces unread voice messages.

The diagram below describes the structure of the Voicemail IVR Menu:



- While listening to the voicemail, press * or # to rewind and forward the voice message, respectively. Each press will forward or rewind 3 seconds.
- Rewind can go back to the beginning of the message while forward will not work when there are 3 seconds or less left in the voice message.
- Voice guidance will be automatically played when the voicemail is done playing.

Leaving Voicemail

If an extension has voicemail enabled under basic settings "**Extension/Trunk → Extensions → Basic Settings**" and after a ring timeout or the user is not available, the caller will be automatically redirected to the

voicemail to leave a message on which case they can press # to submit the message.

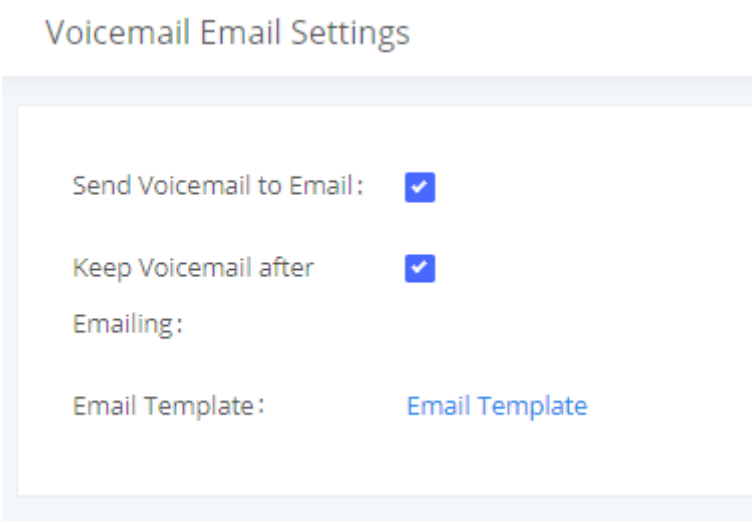
In case the caller is calling from an internal extension, they will be directly forwarded to the extension's voicemail box. But if the caller is calling from outside the system and the incoming call is routed by DID to the destination extension, then the caller will be prompted with the choice to either press 1 to access voicemail management or press 2 to leave a message for the called extension. This feature could be useful for remote voicemail administration.

Voicemail Email Settings

The PBX can be configured to send the voicemail as an attachment to the Email. Click on the "Voicemail Email Settings" button to configure the Email attributes and content.

Send Voicemail to Email	If enabled, voicemail will be sent to the user's email address. Note: SMTP server must be configured to use this option.
Keep Voicemail after Emailing	Enable this option if you want to keep recording files after the Email is sent. The default setting is Enable.
Email Template	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the user. The template variables are: t: TAB ${VM_NAME}: Recipient's first name and last name ${VM_DUR}: The duration of the voicemail message ${VM_MAILBOX}: The recipient's extension ${VM_CALLERID}: The caller ID of the person who has left the message ${VM_MSGNUM}: The number of messages in the mailbox ${VM_DATE}: The date and time when the message is left. (Format: MM/dd/yyyy hh:mm:ss)

Voicemail Email Settings

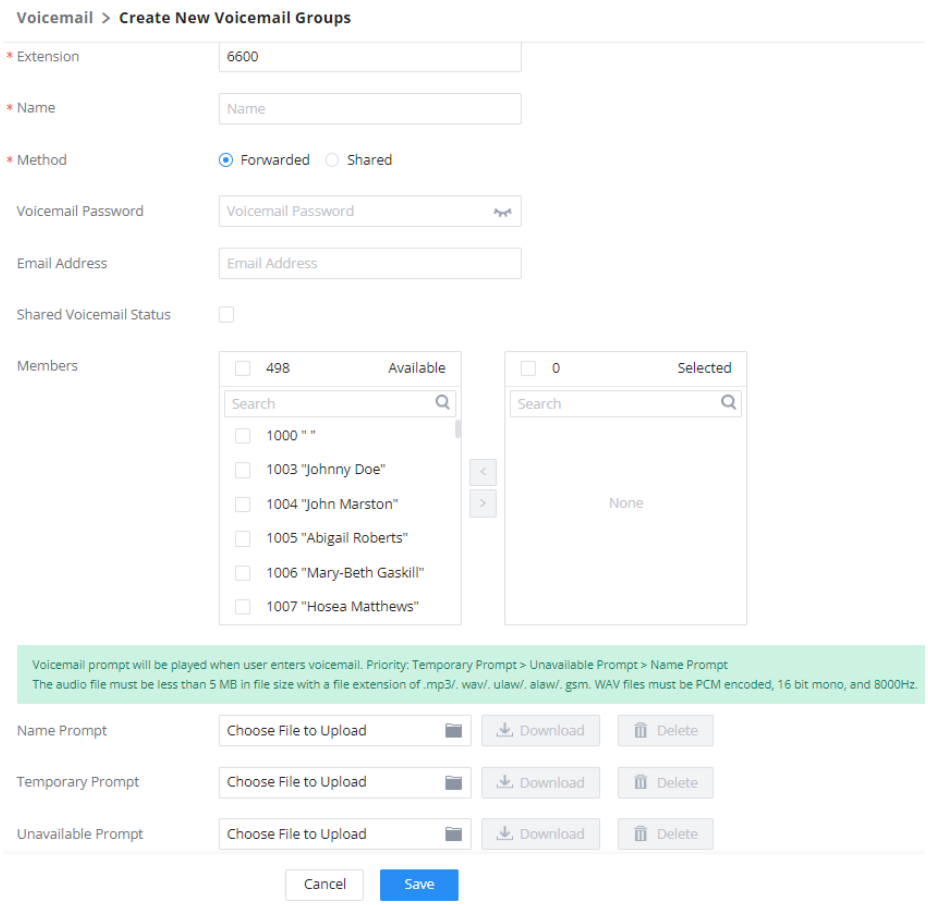


Voicemail Email Settings

Click on the "Email Template" button to view the default template as an example.

Configure Voicemail Group

The PBX supports voicemail group and all the extensions added in the group will receive the voicemail to the group extension. The voicemail group can be configured under **Basic Call Features → Voicemail → Voicemail Group**. Click on "Add" to configure the group.



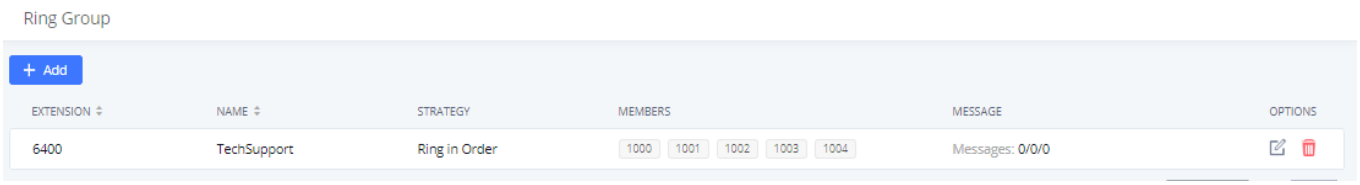
Voicemail Group

Ring Groups

The PBX supports ring group feature with different ring strategies applied to the ring group members. This section describes the ring group configuration on the PBX.

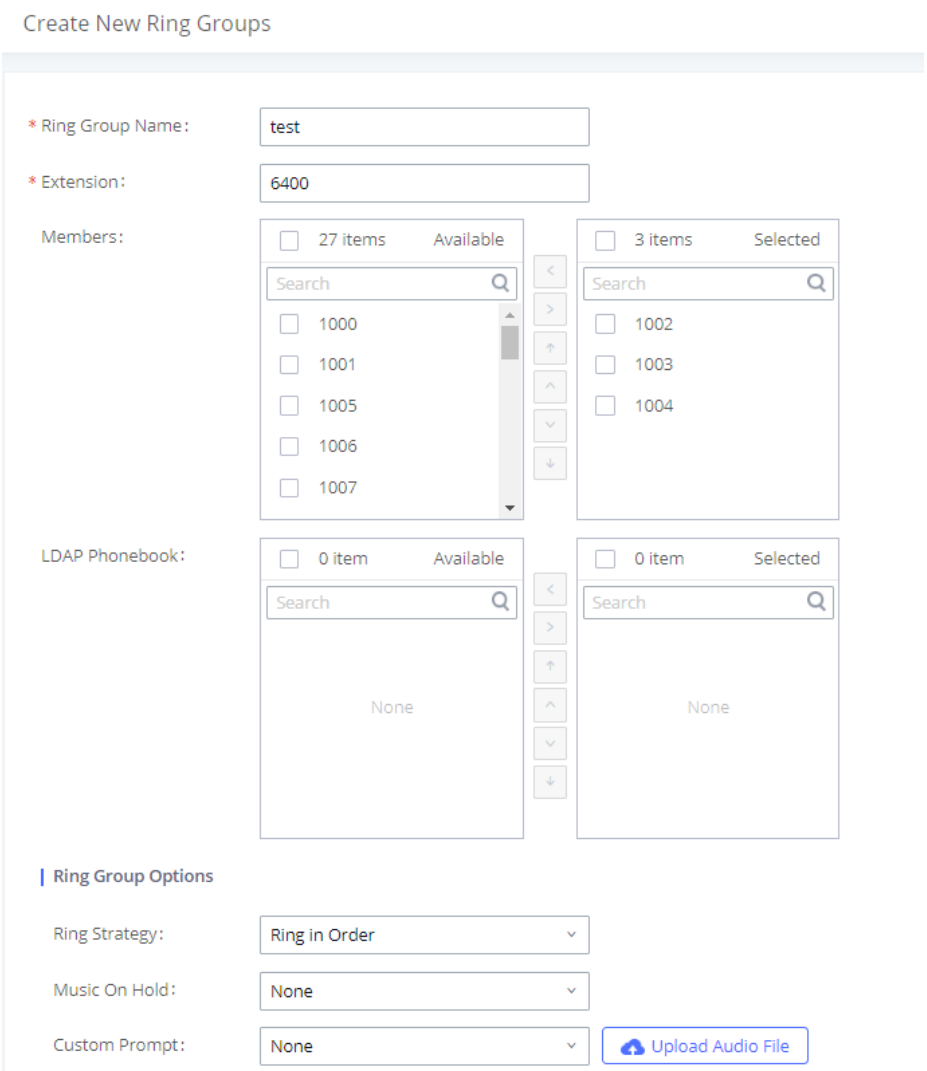
Configure Ring Group

Ring group settings can be accessed via **Basic Call Features→Ring Group**.



Ring Group

- Click on to add ring group.
- Click on to edit the ring group. The following table shows the ring group configuration parameters.
- Click on to delete the ring group.

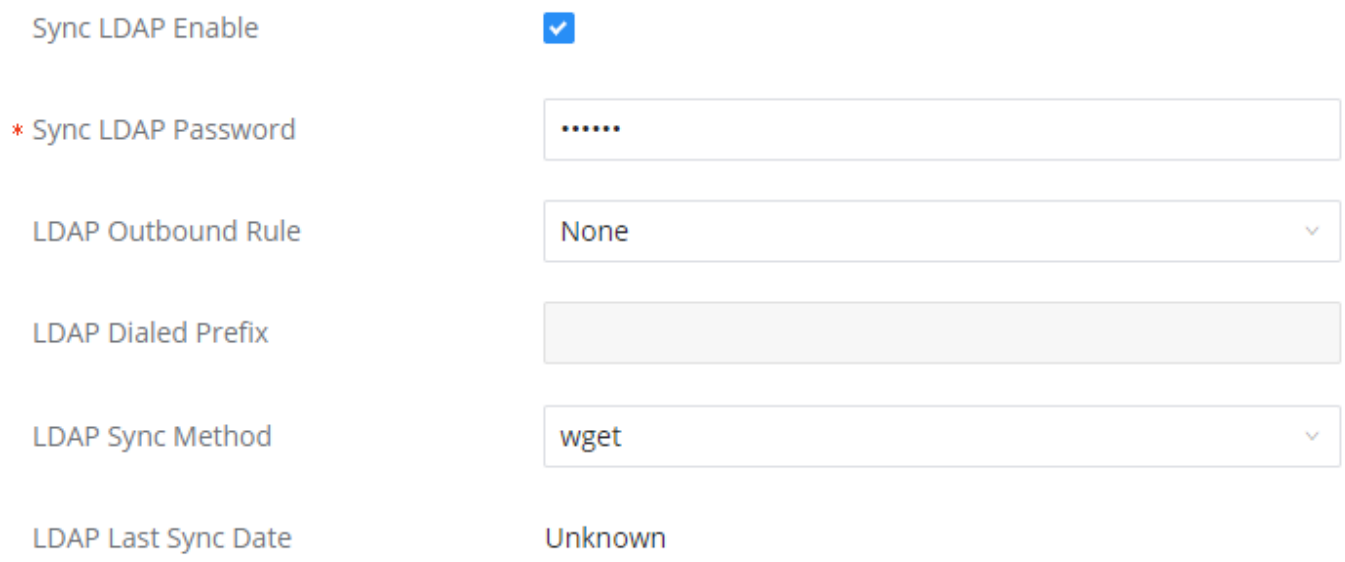


Ring Group Configuration

Remote Extension in Ring Group

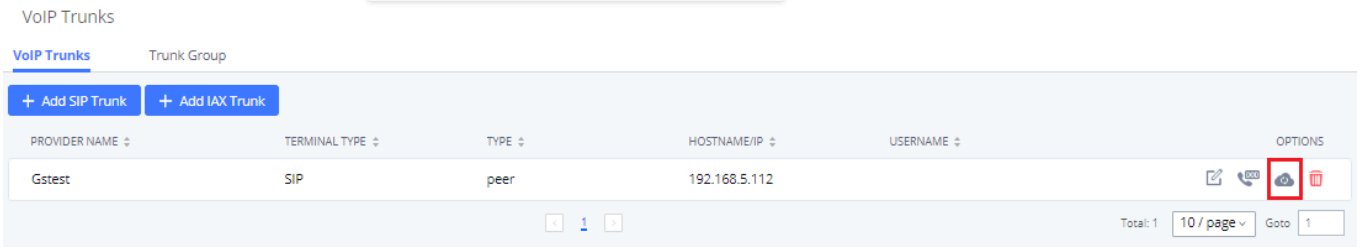
Remote extensions from the peer trunk of a remote IPPBX can be included in the ring group with local extension. An example of Ring Group with peer extensions is presented in the following:

1. Creating SIP Peer Trunk between both IPPBX_A and IPPBX_B. **SIP Trunk** can be found under Web GUI→**Extension/Trunk→VoIP Trunks.** Also, please configure their Inbound/Outbound routes accordingly.
2. Click edit button in the menu , and check if **Sync** LDAP **Enable** is selected, this option will allow IPPBX_A update remote LDAP server automatically from peer IPPBX_B. In addition, **Sync LDAP Password** must match for IPPBX_A and IPPBX_B to sync LDAP contact automatically. Port number can be anything between 0~65535, and use the outbound rule created in step 1 for the **LDAP Outbound Rule** option**.**



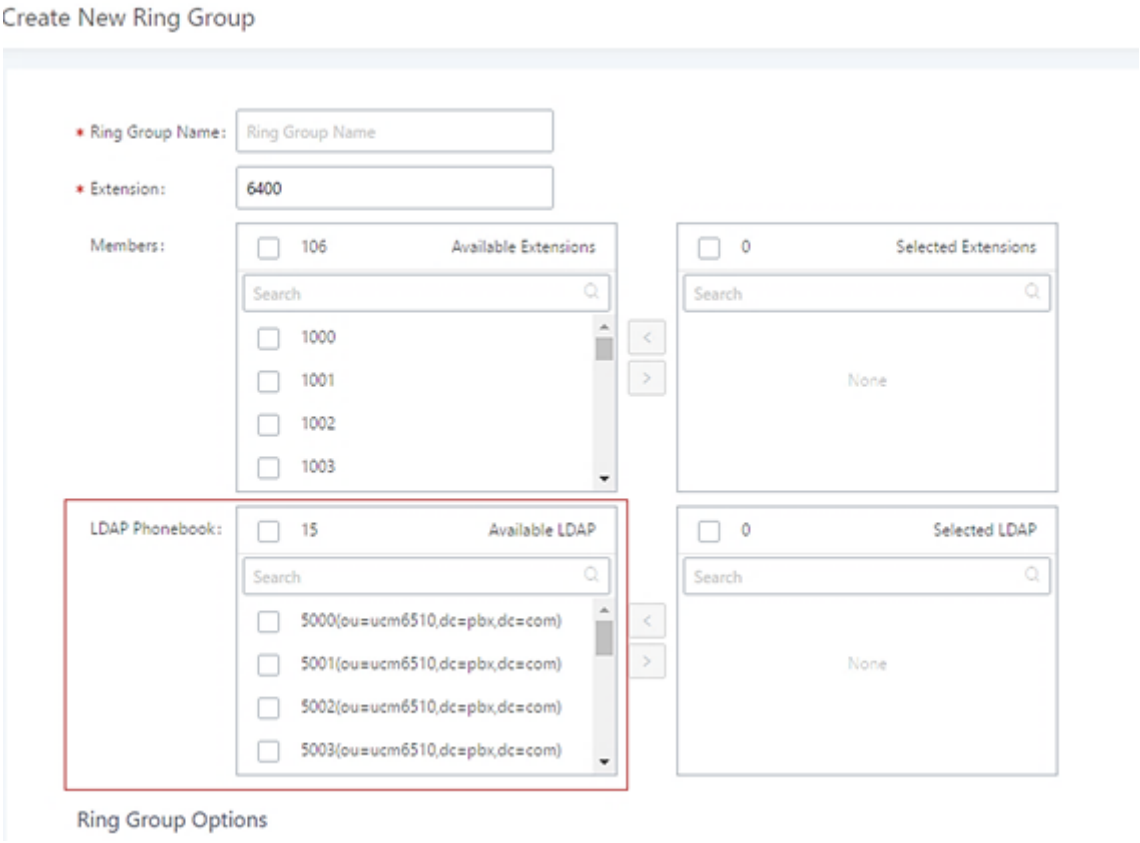
Sync LDAP Server Options

3. In case if LDAP server does not sync automatically, user can manually sync LDAP server. Under **VoIP Trunks** page, click sync button shown in the following figure to manually sync LDAP contacts from peer IPPBX.



Manually Sync LDAP Server

4. Under **Ring Groups** setting page, click "Add". **Ring Groups** can be found under **Basic Call Features→ Ring Groups**.
5. If LDAP server is synced correctly, **Available LDAP Numbers** box will display available remote extensions that can be included in the current ring group. Please also make sure the extensions in the peer IPPBX can be included into that IPPBX's LDAP contact.



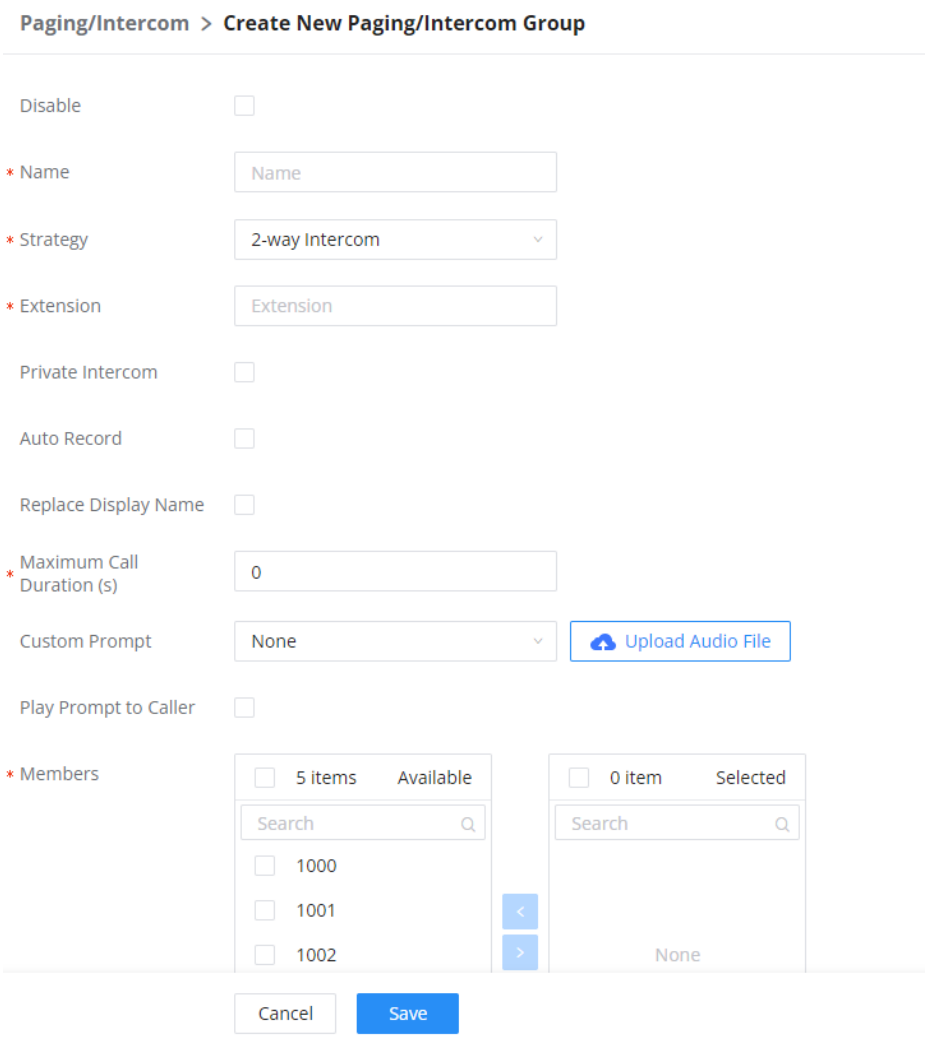
Ring Group Remote Extension

Paging/Intercom

Paging and Intercom Group can be used to make an announcement over the speaker on a group of phones. Targeted phones will answer immediately using speaker. The IPPBX paging and intercom can be used via feature code to a single extension or a paging/intercom group. This section describes the configuration of paging/intercom group under **Basic Call Features→Paging/Intercom**.

Paging/Intercom Groups

2-way Intercom



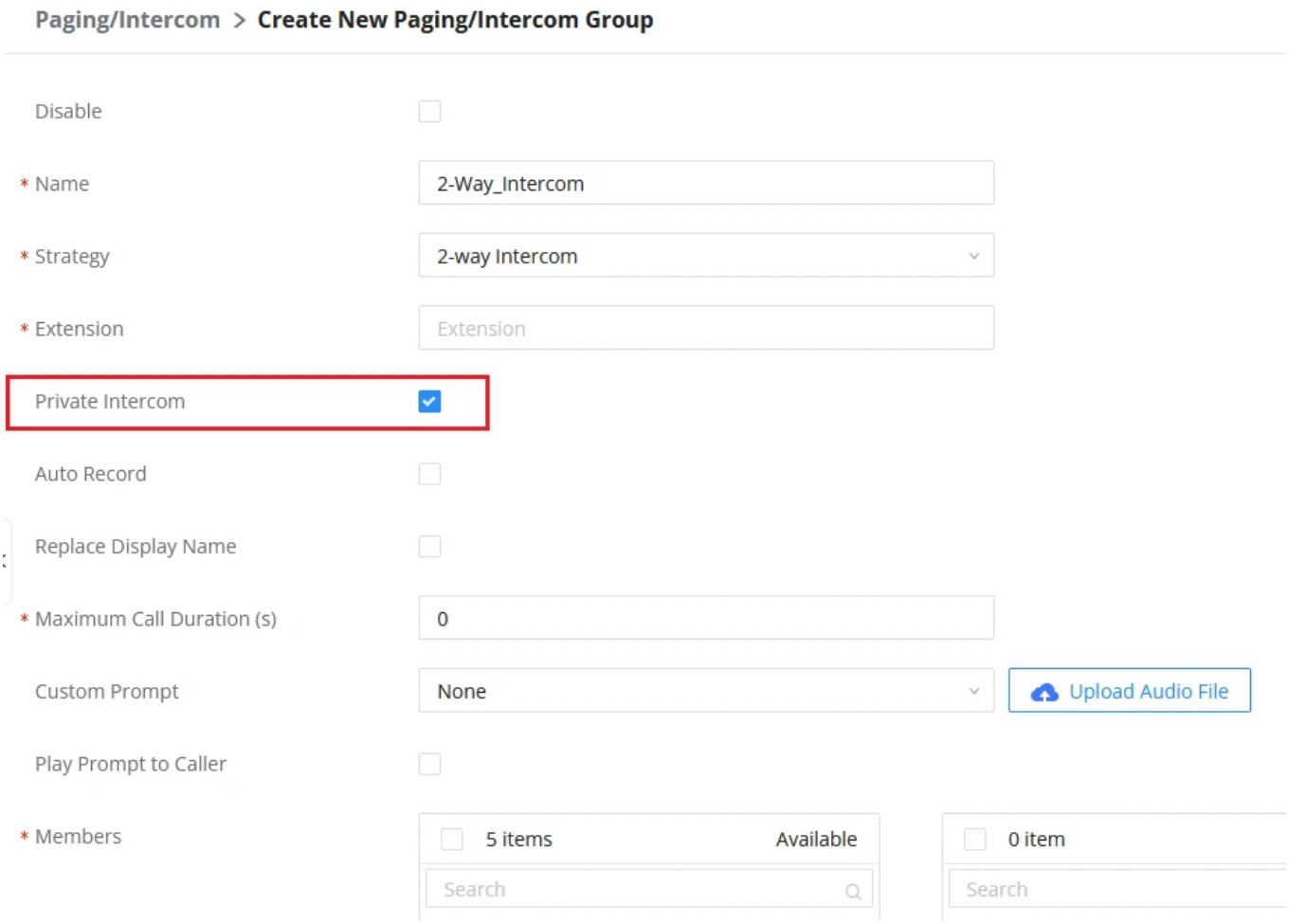
2-way Intercom

Parameter	Description
Disable	If disabled, the real-time and scheduled intercom will not be triggered.
Name	Configure intercom group name.
Strategy	Select "2-way Intercom".
Extension	Configure the intercom group extension.
Private Intercom	If enabled, members can only hear the voice of the initiator and cannot hear the voice of other members. The initiator can hear the voice of all members.
Auto Record	Enable this option to record in WAV format.
Replace Display Name	If enabled, the PBX will replace the caller display name with Intercom name.
First Answer Termination	Enabling this option will result in halting other phones from ringing once the intercom is answered.
Maximum Call Duration	Specify the maximum call duration in seconds. The default value 0 means no limit.
Custom Prompt	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Upload Audio File', it will direct the users to upload the customized voice prompts. Note: Users can also refer to the page PBX Settings?Voice Prompt?Custom Prompt, where they could record new prompt or upload prompt files.
Play Prompt to Caller	Play the prompt to the caller.

Members	Select available users from the left side to the paging/intercom group member list on the right.
Paging/Intercom Whitelist	Select which extensions are allowed to use the paging/intercom feature for this paging group.

Private Intercom

Private intercom allows the user to initiate an intercom to many endpoints. Whichever endpoint microphone has detected sound input first, only the intercom initiator and the responder will be able to hear each other. Once the first responder has finished talking, the second responder can start talking. To configure private intercom, the user can follow the steps mentioned above in 2-way intercom and while creating the intercom, the user may tick the option "Private Intercom" as indicated in the screenshot below.

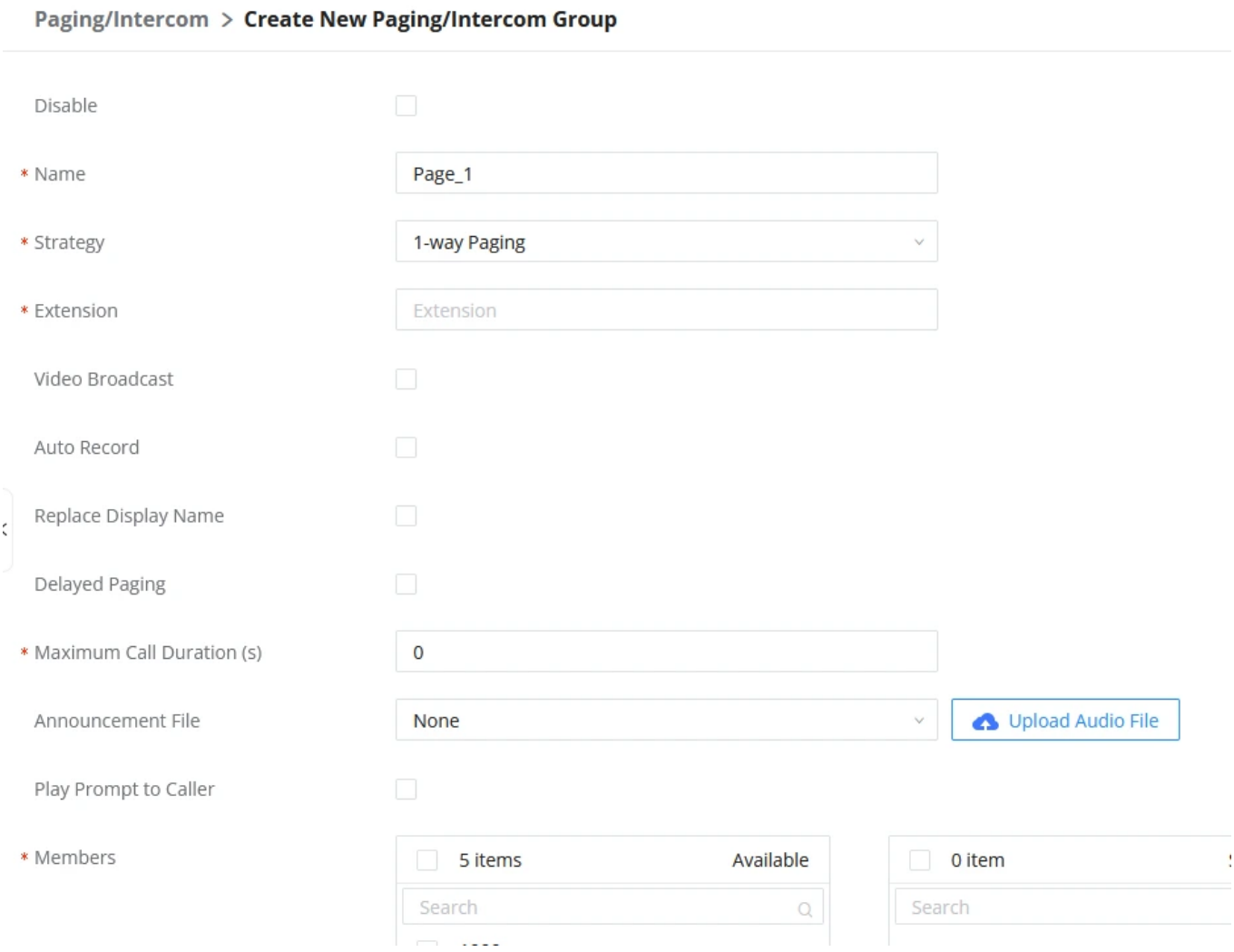


Private Intercom

Parameter	Configuration
Name	Enter a name for the intercom
Strategy	Choose "Private Intercom".
Extension	Configure the intercom group extension.
Auto Record	Enable this option to record in WAV format.
Replace Display Name	If enabled, the PBX will replace the caller display name with Paging/Intercom name.
Maximum Call Duration (s)	The maximum allowed duration of a call in seconds. Default value is 0 (no limit).
Disable	If disabled, the real-time and scheduled paging/intercom will not be triggered.

Custom Prompt	This option sets a custom prompt to be used as an announcement to the person receiving a paging/intercom call. The file can be uploaded from the page "Custom Prompt". Click "Upload Audio File" to add additional record.
Members	Selected members will receive paging/intercom calls to this paging/intercom group.
Paging/Intercom Whitelist	Only selected extensions will be able to use this paging /intercom group. If none is selected, all extensions will be able to use this paging/intercom group.

1-way Paging



1-way Paging

Parameter	Description
Name	Configure paging/intercom group name.
Strategy	Select "1-way Paging".
Extension	Configure the paging/intercom group extension.
Video Broadcast	If checked, video paging will be supported. If the caller sends a video page, the paging group members will be able to receive and view the video.
Auto Record	Enable this option to record in WAV format (audio) and MKV format (video).
Replace Display Name	If enabled, the PBX will replace the caller display name with Paging/Intercom name.

First Answer Termination	Enabling this option will result in halting other phones from ringing once the paging is answered.
Delayed Paging	If enabled, a caller can enter *82 before the paging group extension to start a delayed paging call. In a delayed paging call, the system will prompt the caller to record a message. Once the messaging is recorded and saved, and the configured delay has passed, the paging call will be sent out. When a paging group member answers the call, the prerecorded message will be played, and the call will end after it is finished playing.
Maximum Call Duration	Specify the maximum call duration in seconds. The default value 0 means no limit.
Disable	If disabled, the real-time and scheduled paging/intercom will not be triggered.
Local Members	Selected members will receive paging/intercom calls to this paging/intercom group.
LDAP Members	Select a number from the LDAP phonebook. The LDAP number can use only the main trunk and specified trunks to make outbound calls. Other trunks cannot be used.
Announcement File	Configures an audio/video file to play to the paging members. This can be used to play preconfigured audio/video at the beginning of paging calls or to simply notify members that it is a paging/intercom call.
Play Prompt to Caller	Play the prompt to the caller.
Paging/Intercom Whitelist	Select which extensions are allowed to use the paging/intercom feature for this paging group.

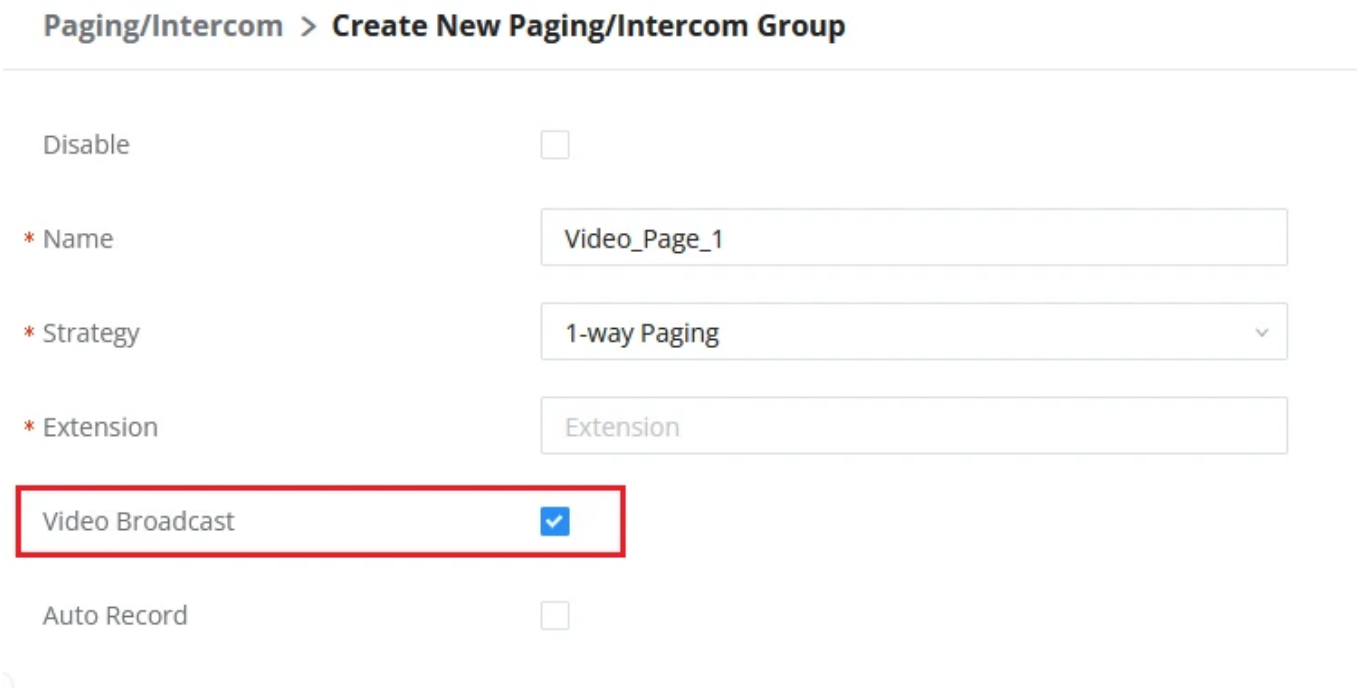
In case the user wants to broadcast a video, these requirements should be respected.

- H.264 video encoding
- .mkv or .tar/.tgz/tar.gz format
- MKV files must be 30 MB file or less
- Compressed files (.tar/.tgz/tar.gz) must be 50 MB or less.
- File name can only contain alphanumeric characters, hyphens (-) and period (.)

If Auto Record is enabled, recorded video pages will be saved in MKV file format.Saved recordings can be found in the CDR→Recordings→Video Recordings page.

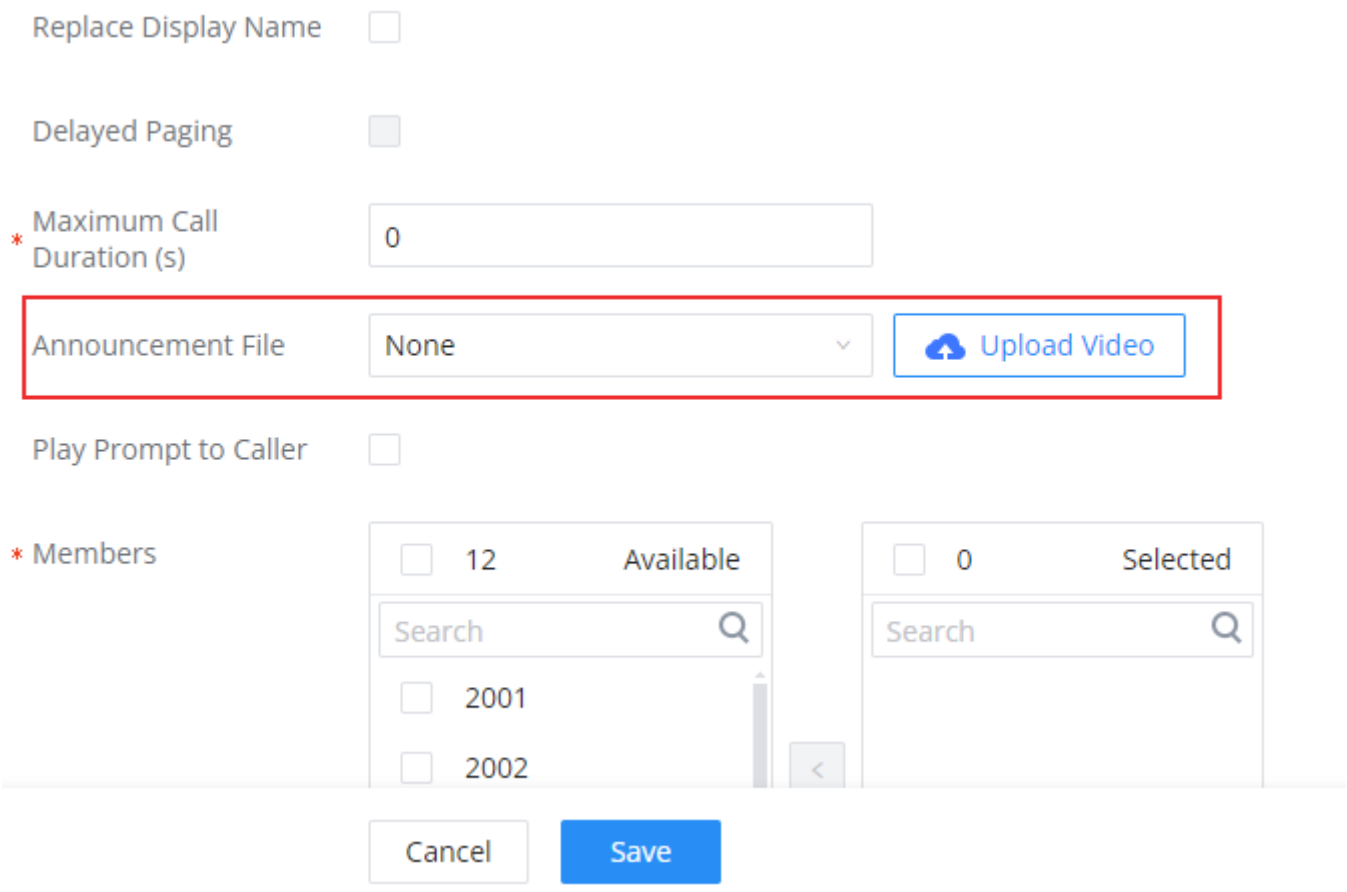
Video Broadcast

Using this feature, the user can send a video to the SIP endpoints which support streaming video to notify the users of the beginning of a paging. To configure Video Broadcast before a paging, the user can create a 1-way paging following the steps mentioned in the previous section. Then the user can enable "Video Broadcast" in the settings.



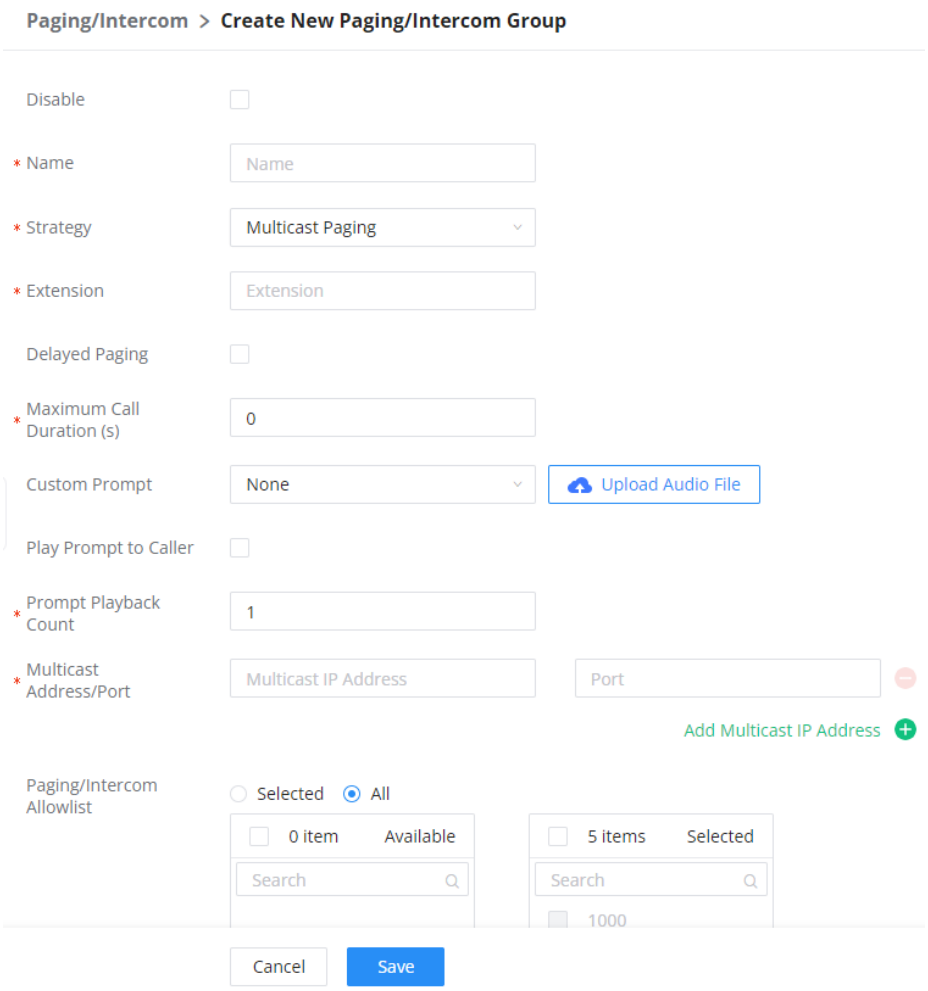
Video Broadcast

The user can upload the video in the "Announcement File" section as shown in the screenshot below.



Video Announcement File Upload

Multicast Paging



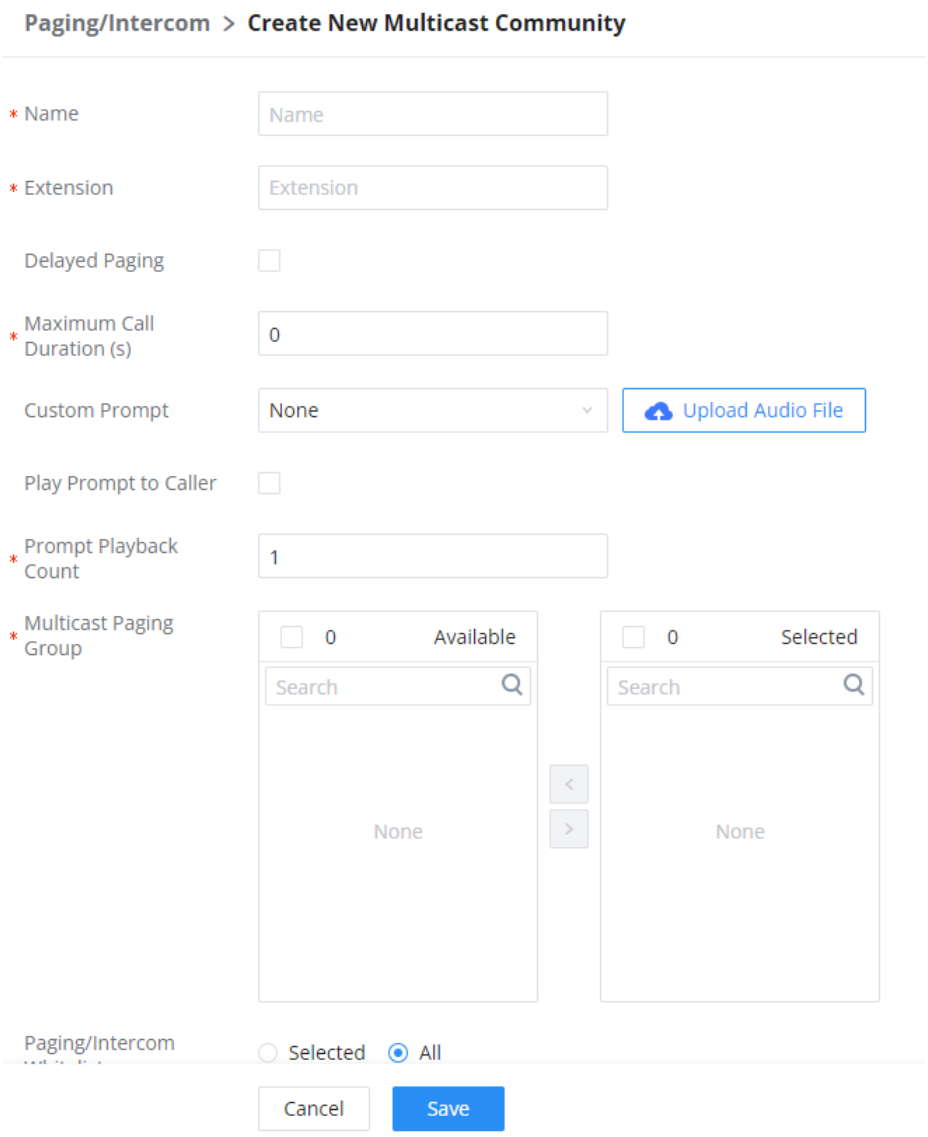
Multicast Paging

Parameter	Description
Name	Configure paging/intercom group name.
Strategy	Select "Multicast Paging".
Extension	Configure the paging/intercom group extension.
Maximum Call Duration	Specify the maximum call duration in seconds. The default value 0 means no limit.
Disable	If disabled, the real-time and scheduled paging/intercom will not be triggered.
Multicast IP Address/Port	The allowed multicast IP address range is 224.0.1.0 - 238.255.255.255.Note: You can add up to 30 multicast addresses.
Paging/Intercom Allowlist	Select the extension which can initiate this paging. If none is selected, all extensions will be able to use this paging/intercom group.
Opening Prompt	Select the opening prompt for the paging.
Custom Prompt	This option is to set a custom prompt for a paging/intercom group to announce to caller. Click on 'Prompt', it will direct the users to upload the customized voice prompts.Note: Users can also refer to the page PBX Settings?Voice Prompt?Custom Prompt, where they could record new prompt or upload prompt files.
Play Prompt to Caller	Play the prompt to the caller.
Prompt Playback Count	Sets the number of times the prompt is played during the page/intercom. To ensure the prompt is played the specified number of times, please set an appropriate max call duration.

Background Sound	Enable background sound.
Background Sound Source	If an extension is selected as the audio source, the audio from that extension will be automatically used as the background sound when the paging/intercom is initiated.Custom Background SoundExtension Shared Audio
Custom Background Sound	Choose or upload a custom background sound.
Announcement Command	While background sound is playing, callers can start and end an announcement by pressing *1.
Start/End Announcement Cue	If enabled, after pressing the Announcement command (*1) to start and end the announcement, the system will play this prompt. Otherwise, only a beep will play.
Mute Background Sound	If enabled, the ongoing background sound will be muted during the announcement.
Multicast Audio Format	Choose the format/codec of the multicast audio.
Delayed Paging	If enabled, a caller can enter *82 before the paging group extension to start a delayed paging call. In a delayed paging call, the system will prompt the caller to record a message. Once the messaging is recorded and saved, and the configured delay has passed, the paging call will be sent out. When a paging group member answers the call, the recorded message will be played, and the call will end after it is finished playing.
Delay (s)	Configure the amount of delay in seconds after a message is recorded to send out the delayed paging call. Default is 5 seconds.

Multicast Community

Multicast community allows creating an extension, which when dialed, can send a preconfigured prompt as a multicast paging to a group of extensions. The user should create first a **Paging/Intercom Group** with **Multicast Paging** as the **Strategy** selected. Please see previous section for more information.



Multicast Community Parameters

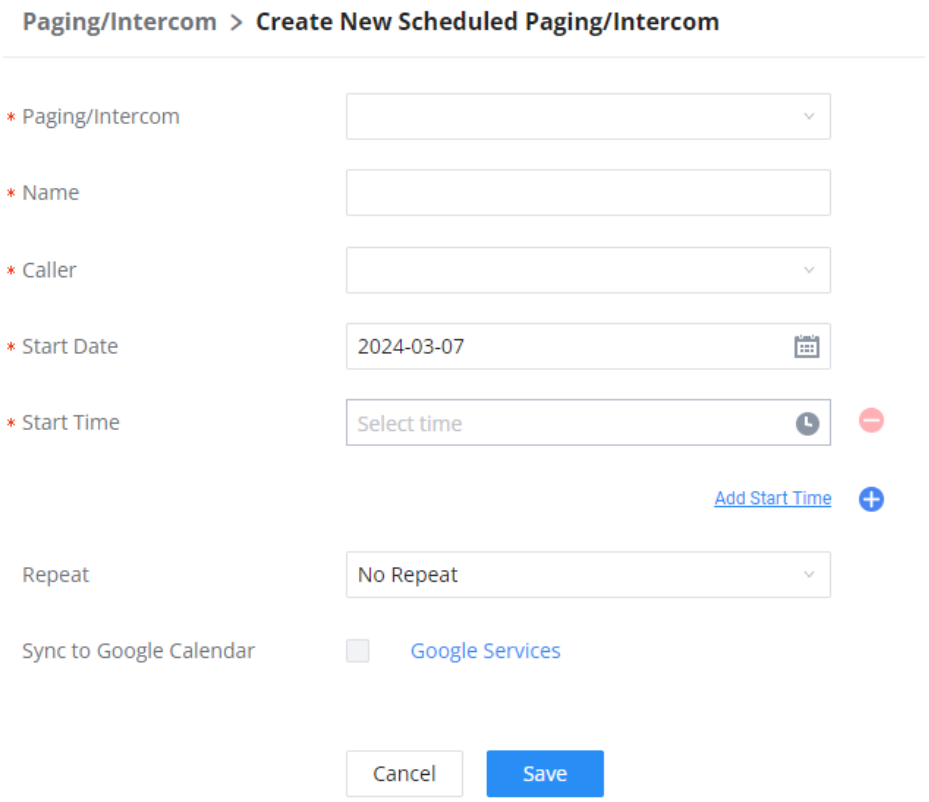
Name	Enter the name of the multicast community.
Extension	Configure the extension number for the paging/intercom group. When this number is dialed, the paging/intercom will be initiated.
Delayed Paging	If enabled, a caller can enter *82 before the paging group extension to start a delayed paging call. In a delayed paging call, the system will prompt the caller to record a message. Once the messaging is recorded and saved, and the configured delay has passed, the paging call will be sent out. When a paging group member answers the call, the precorded message will be played, and the call will end after it is finished playing. Delay (s): Configure the amount of delay in seconds after a message is recorded to send out the delayed paging call. Default is 5 seconds.
Maximum Call Duration (s)	The maximum allowed duration of a call in seconds. Default value is 0 (no limit).Note: Please note that the call duration that can be configured can be within the range 0 - 86400.

Custom Prompt	Choose the custom prompt to play for the callees at the beginning of the paging. The user can also directly upload a prompt file directly on this page. Note: When uploading the custom prompt file, please make sure it respects the following requirements: The audio file must be less than 5 MB in file size with a file extension of .mp3/. wav/. ulaw/. alaw/. gsm. WAV files must be PCM encoded, 16 bit mono, and 8000Hz. If uploading a compressed file, the file extension must be .tar/.tgz/.tar.gz, and the file size must not exceed 50MB. File name can only contain alphanumeric characters and special characters -_
Play Prompt to Caller	When this option is enabled, the prompt will be played back on the caller's phone.
Prompt Playback Count	Sets the number of times the prompt is played during the page/intercom. To ensure the prompt is played the specified number of times, please set an appropriate max call duration.Note: The minimum number of playcounts which can be configured is 1 and the maximum is 10.
Multicast Paging Group	Configures multicast paging groups within a community. When dialing the extension number of the community, users may simultaneously initiate paging to linked multicast paging groups.
Paging/Intercom Whitelist	Selected: Only selected extension will be allowed to initiate paging/intercom. All: Allow all extensions and configured services to initiate paging/intercom.

Scheduled Paging/Intercom

Pending Paging/Intercom

In this page, the user can create scheduled intercom/paging to be played automatically when the time scheduled arrives.

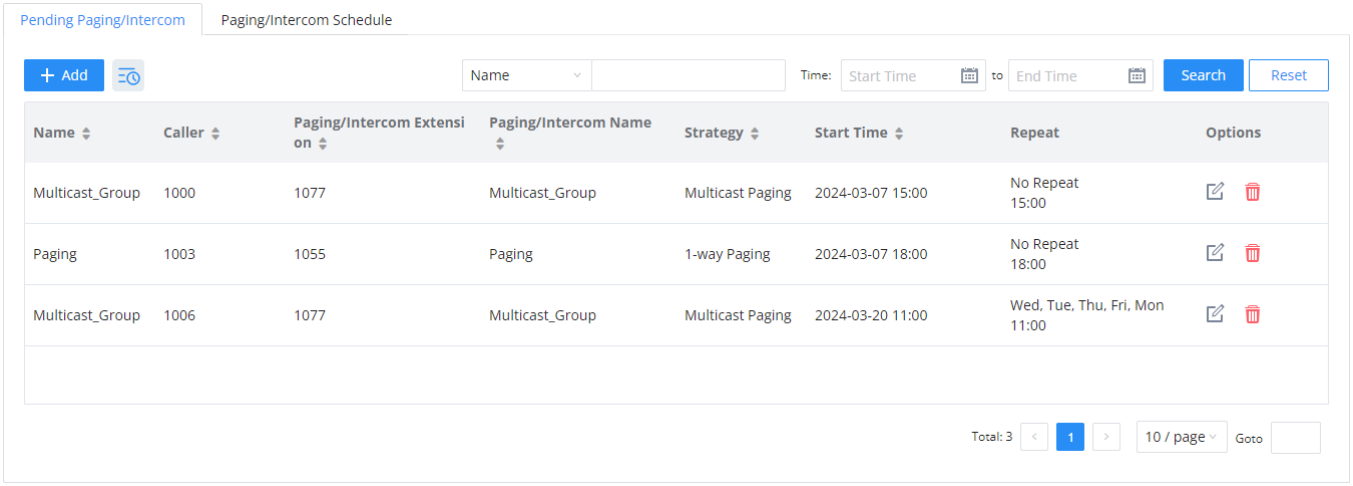


Create New Scheduled PagingIntercom

Paging/Intercom	Select existing paging/intercom groups and multicast communities.
Name	Enter the name of the scheduled Intercom/Paging.

Caller	Once a caller is selected, and the specified start time is reached, the system will contact the caller. If this call is rejected, the page/intercom will be cancelled. If caller is set to None, the system will call all group members and play the configured prompt.
Start Date	Select the date of the start of the paging/intercom.
Start Time	Select the start time of the paging/intercom.
Repeat	Select the repeat interval of the paging/intercom. No Repeat: The intercom/paging will play once on the scheduled date and time. Everyday: The intercom/paging will play daily starting from the scheduled day and on the time scheduled every day. Weekly: The intercom/paging will play weekly on the selected day(s) of the week. Monthly: The intercom/paging will play monthly on the selected date of the month.
Include Holidays	When Every day, Weekly, or Monthly is selected. This option will appear. If enabled, scheduled pages/intercoms will run during holidays. Otherwise, scheduled pages/intercoms displayed on the calendar will not actually be run.
Sync to Google Calendar	This feature cannot be used if Google Services have not been authorized. Please resolve this in the Integrations > Google Services page.

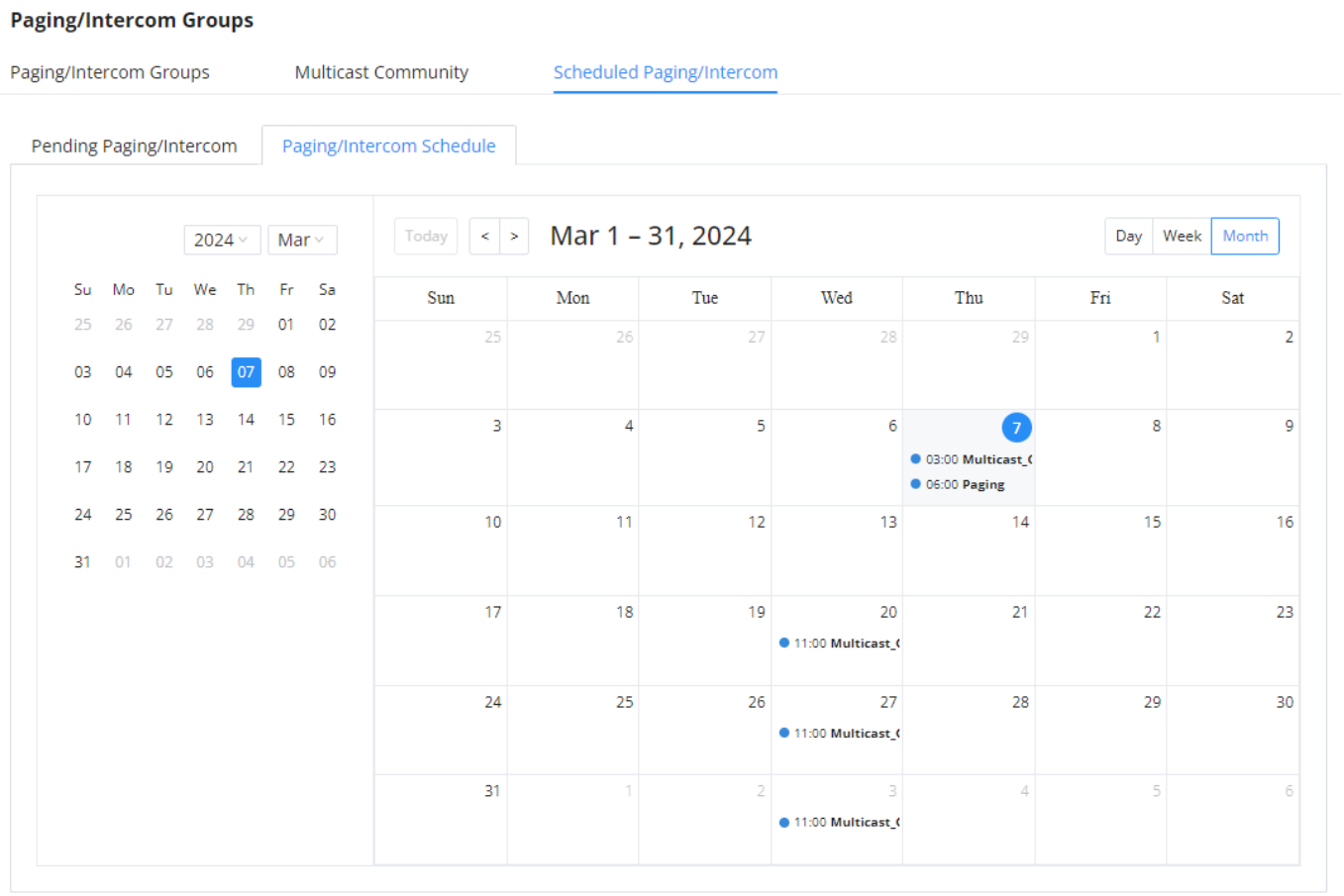
Once the paging and intercom has been created, it can be viewed on the same page.



Pending PagingIntercom

Paging/Intercom Schedule

This section displays the schedule of the paging/intercom which have been scheduled. The user can choose to display per day, week, or per month.



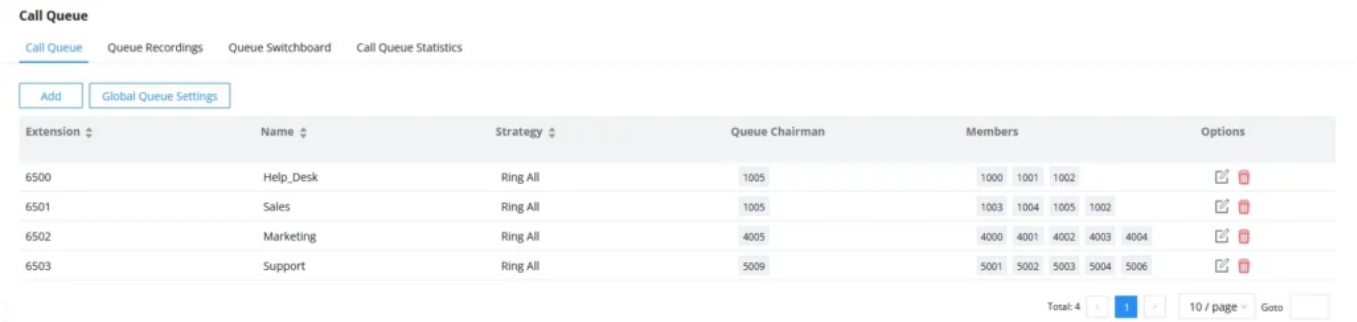
Scheduled PagingIntercom

Call Queue

GCC IPPBX supports call queues using static agents or dynamic agents. The call queue system can accept more incoming calls than the number of available agents, allowing calls to be held until an agent becomes available. This section describes how to configure call queues under **Basic Call Features-> Call Queue**.

Add Call Queue

Call queue settings can be accessed via **Basic Call Features->Call Queue**.



Call Queue Page

- Click on "Add" to add call queue.
- Click the **edit icon** to modify an existing call queue.
- Click the **delete icon** to remove a call queue.

The call queue configuration parameters are listed in the table below.

Static and Dynamic Agents

There are two types of agents that can be part of a call queue:

- **Static Agents:** manually added during queue configuration.
- **Dynamic Agents:** can log in and out of a queue at any time using preset login/logout codes.

Both static and dynamic agents are included in the total number of agents allowed for the system.

Static Agents

GCC PBX allows a limited number of total agents based on the package plan. The total number of agents (including both static and dynamic) is determined according to the number of extension units in the package, following the table below:

Model	Maximum Number of Agents in Call Queue
GCC6020	50
GCC6021	50

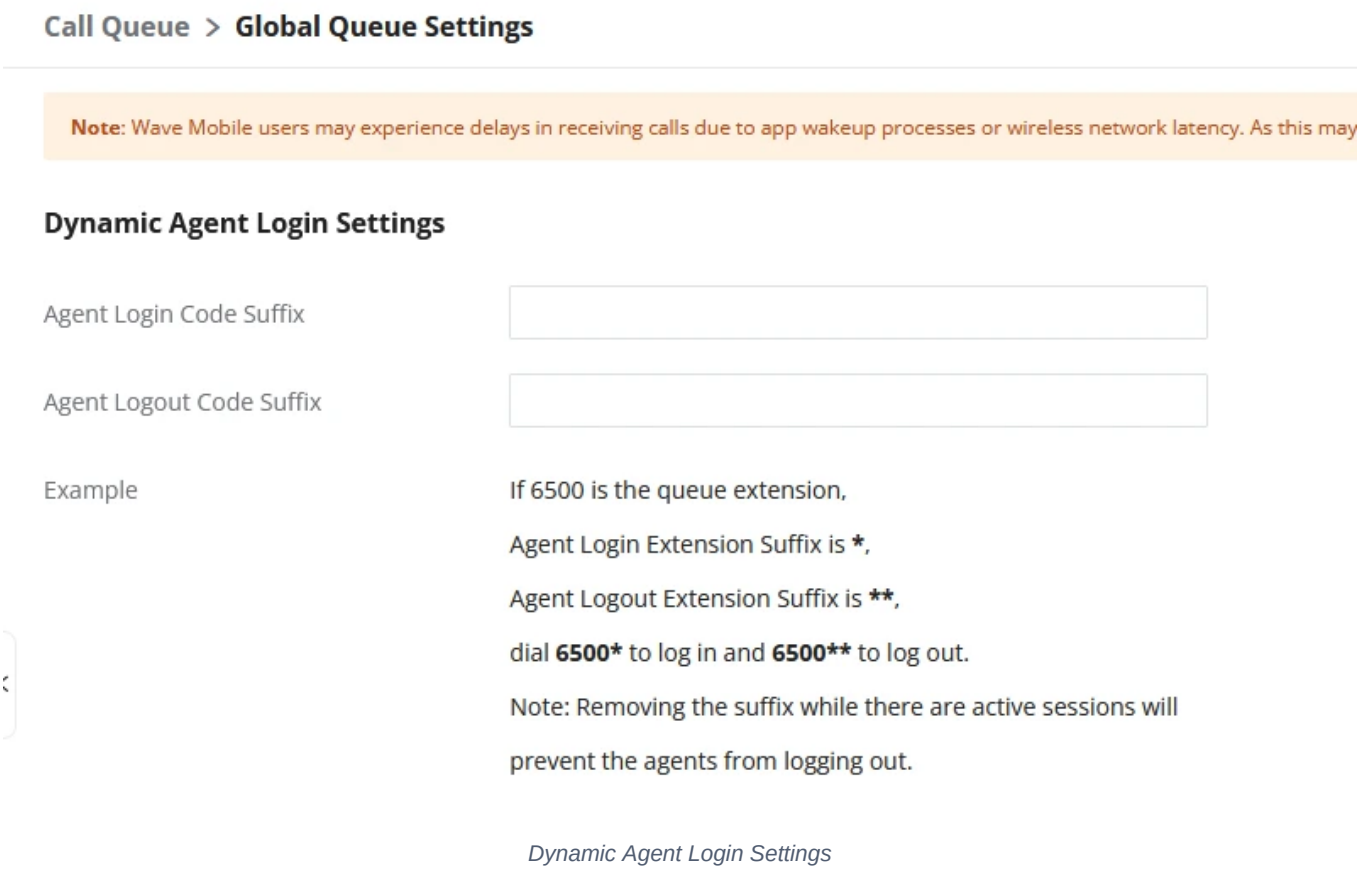
Static agents can be configured on **Basic Call Features**->**Call Queue**->**Edit Queue**->**Agents**.

- The total number of agents here refers to the entirety of the GCC PBX system, including all queues, not just a specific queue.
- Users cannot exceed the total agent limit by mixing static and dynamic agents.

Dynamic Agents

Dynamic agents are not permanently assigned to a queue. Instead, they can **log in or log out** at any time using specific dial codes.

These codes can be configured under **Basic Call Features**->**Call Queue**->**Global Queue Settings**->**Dynamic Agent Login Setting** as shown below:



Dynamic Agent Login Settings

When configuring the call queue settings, checking the option "Enable Agent Login" will cause the dynamic agents to be unavailable.

Call Queue Feature Codes

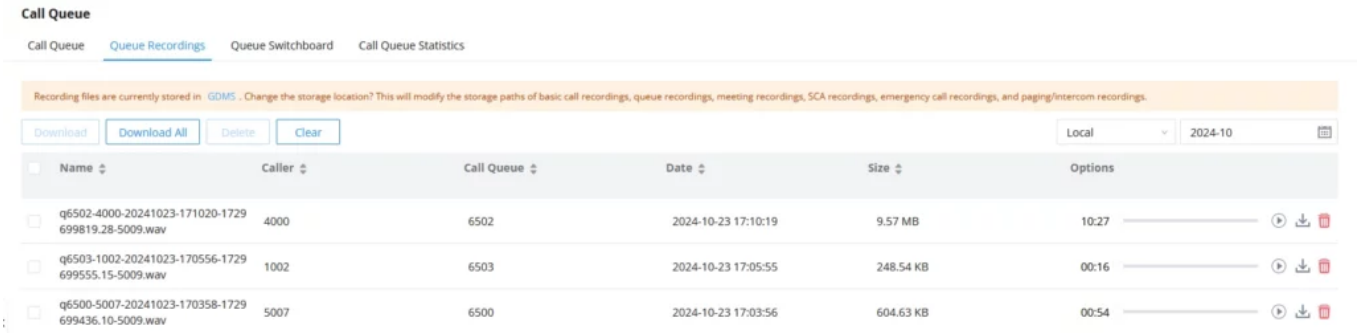
Users can leverage feature codes to perform different call queue related actions by accessing **Basic Call Features**->**Feature Codes**->**Feature Codes**, below is the description of each code along with their default values:

- **Agent Pause (*83):** Allows agents to pause their activity in all queues for a specific reason. Once the code is dialed, the agent will be prompted to enter a pause reason represented by a digit from 0 to 9. Another way to do this is to dial the feature code and the reason code number together. (e.g. dialing *831 to directly set the pause status as "Lunchtime" (1)).
- **Agent Unpause (*84):** This code is used by the agent to resume activity in all queues.
- **Dynamic Agent Logout (*85):** Agents can dial this code to logout from all queues.

Users can configure up to 10 pause reasons when the Agent Pause Reason Settings are set to "Custom". However, there only 5 pause reasons on the default settings: (1) Lunch, (2) Hourly Break, (3) Backoffice, (4) Email, and (5) Wrap.

Queue Recordings

Queue recordings are shown under **Basic Call Features**->**Call Queue**->**Queue Recordings**.



Queue Recordings Page

- Click on

to download the recording file in .wav format; and on

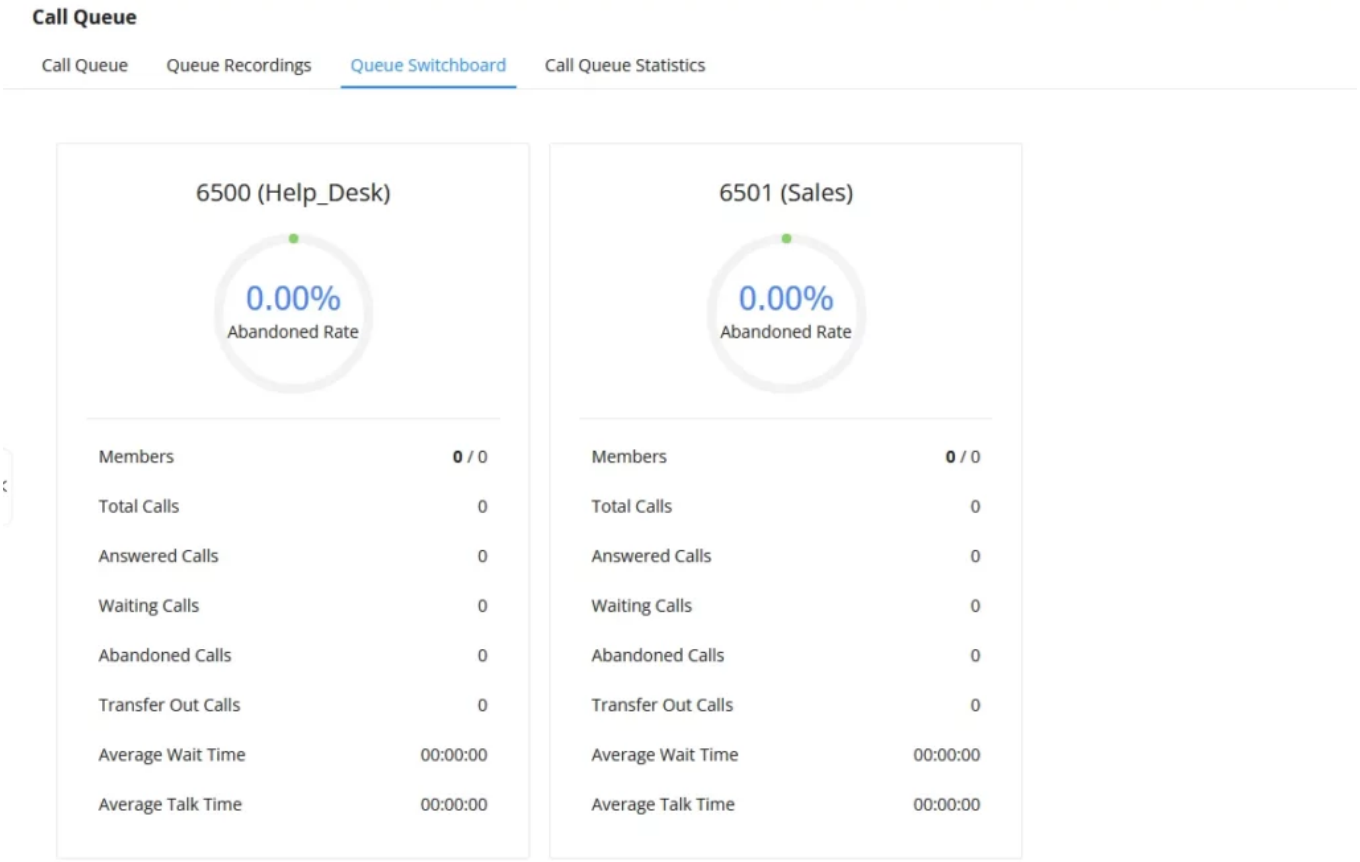
to delete the recording file.

If file data/file encryption is enabled for recording files, the Decryption Tool will be required to play the downloaded file. Instructions for using the tool can be found in this guide.

Queue Switchboard

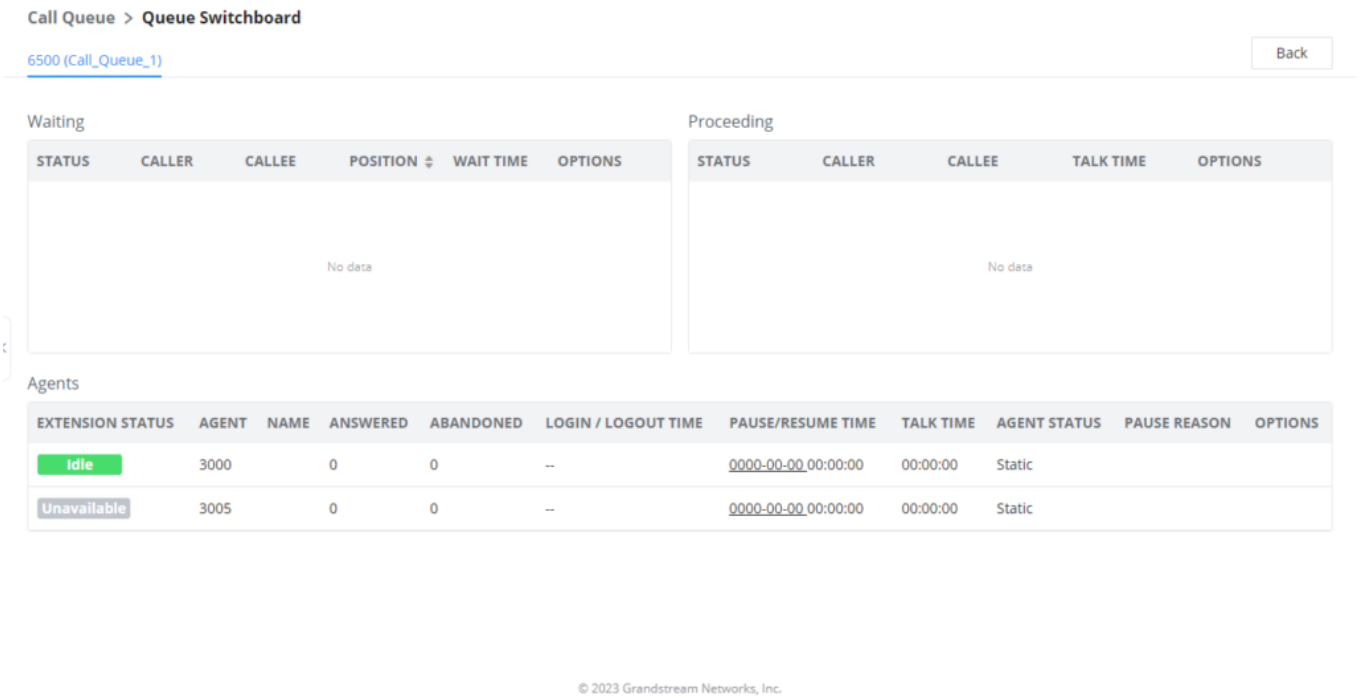
Switchboard is a Web GUI tool for call queue monitoring and management, admin can access to it from the menu **Basic Call Features**->**Call Queue** then press "Switchboard".

Following page will be displayed:



Switchboard Summary

Page above summarizes the available queues statistics and if one of the queues is clicked the user will be directed to page below:



Call Queue Switchboard

The table below gives a brief description for the main menus:

Waiting	Shows the current waiting calls along with the caller ID and the option to hang-up call.
Proceeding	Shows the current established calls along with the caller ID and the callee (agent) as well as the option to hang-up, transfer, add conference or barge-in the call.
Agents	Displays some basic call statistics and agent's mode (static or dynamic) along with the list of agents in the queue and the extension status: Idle Ringing In Use Paused Unavailable

Queue Switchboard Privilege

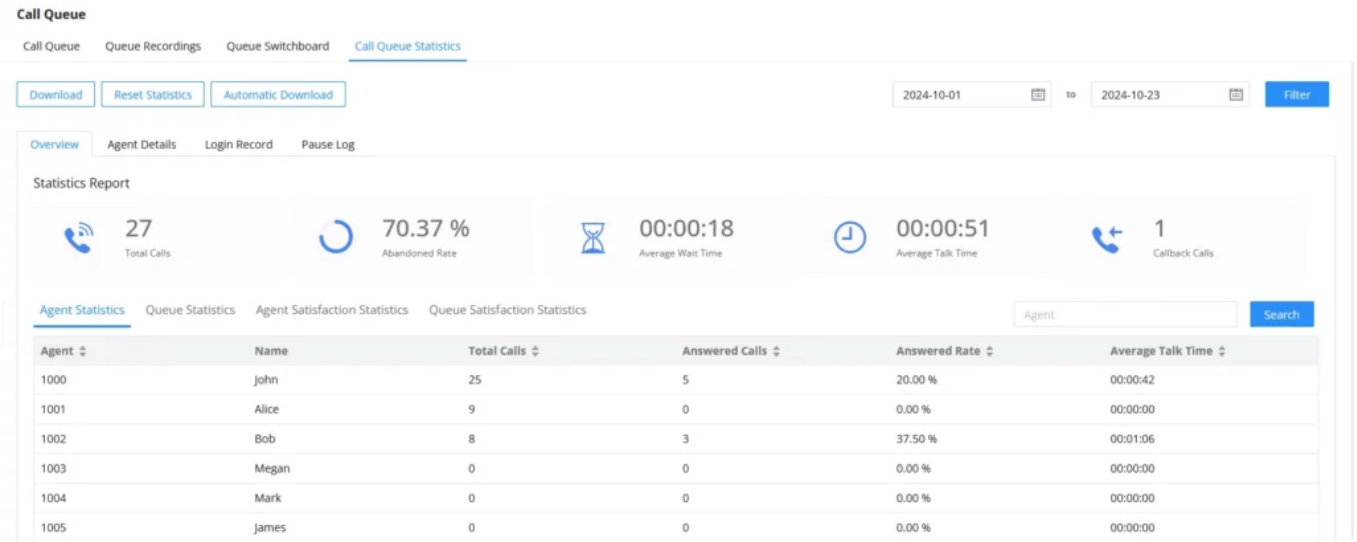
There are three different privilege levels for Call Queue management from the switchboard: Super Admin, Queue Chairman, and Queue Agent.

Super Admin	Default admin of the PBX. Call queue privileges include:Viewing and edit all queue agents.Monitor and execute actions for incoming/outgoing calls for all queues.Generate Call Queue reports to track performance.
Queue Chairman	User appointed by Super Admin to monitor and manage an assigned queue extension via Switchboard.The Queue Chairman can log into the PBX user portal with their extension number and assigned user password.Call Queue privileges include:Viewing status and information related to all agents in the assigned queue. Perform actions on calls such as hang-up, transfer or barge-in. Manual Login/Logout of static and dynamic agents.Viewing CDR
Queue Agent	User appointed by Super Admin to be a member of a queue extension.A queue agent can log into the PBX user portal with their extension number and assigned user password to manage their calls only.

Call Queue Statistics

Along with call center features, users can also gather detailed call queue statistics allowing them to make better changes/decision to manage better the call distribution and handling based on time, agent, and queue.

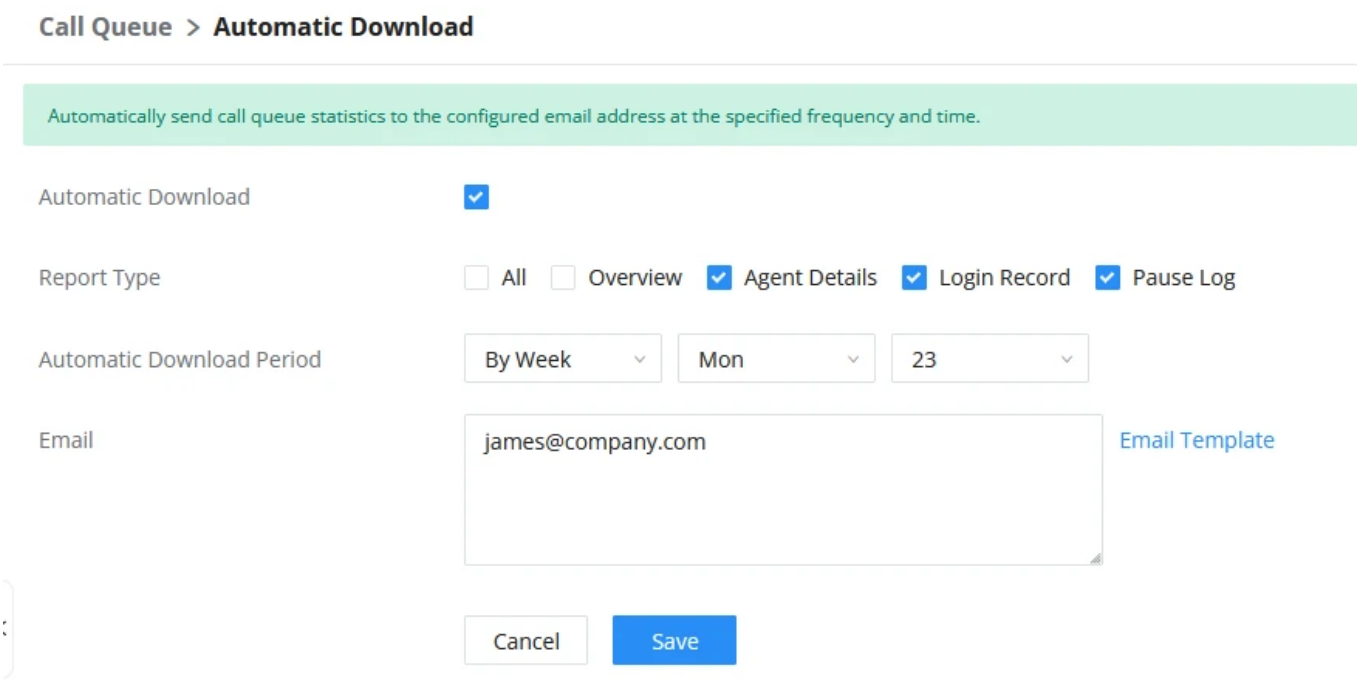
To access call queue statistics, go to **Basic Call Features**->**Call Queue** and click on "**Call Queue Statistics** ", the following page will be displayed:



Call Queue Statistics

Users can download statistics in CSV format by clicking the button. They can also set up automatic email deliveries of these statistics at regular intervals by clicking on

Additionally, users can clear the statistics using the button.



Call Queue Statistics Automatic Download

This section provides a detailed description of each tab on the **Basic Call Features→Call Queue→Call Queue Statistics** page.

Overview Tab

The overview page shows shows the following information:

- **Agent statistics:** shows the number of calls and call-related information of agents.
- **Queue Statistics**: counts the number of calls in the queue and information such as calls, waiting, and callback.
- **Agent satisfaction statistics**: used for user's rating of agents;
- **Queue satisfaction statistics:** counts the score survey statistics.

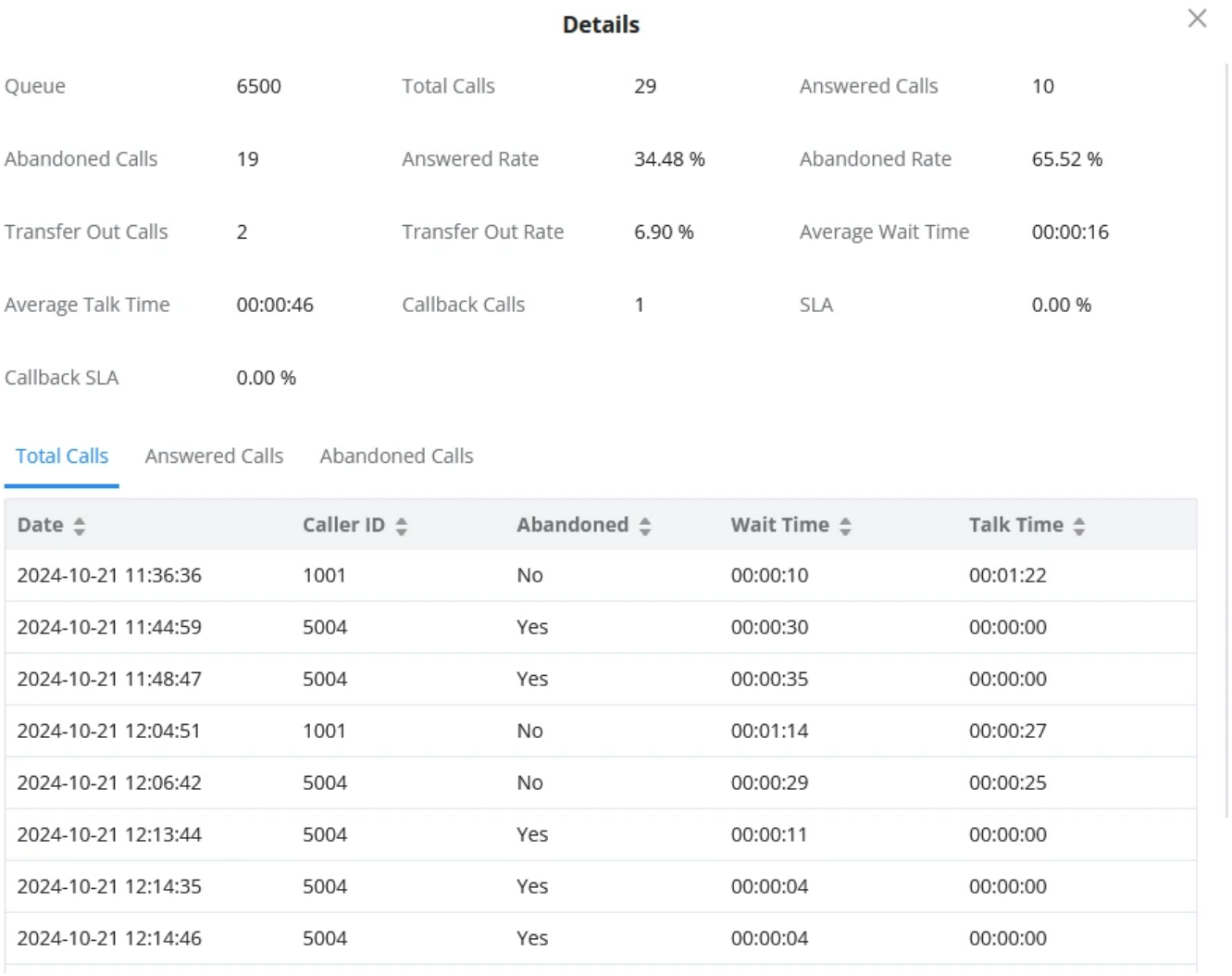
By selecting a time interval, administrators can get detailed statistics for agents such as total calls, answered calls etc, as well as queue statistics like abandoned calls, transferred calls and SLA.

Abandoned Call Calculation

When a call queue is configured with a failover destination pointing to another queue, the abandonment statistics are calculated independently for each queue.

- If a call enters CQ1 and is redirected to CQ2 (failover destination) and the call is answered, CQ1 records the call as abandoned, and CQ2 marks the call as answered.
- If the call is not answered in either queue, both call queues consider the call as abandoned.

A more detailed version of the queue statistics (as shown in image below) can be found under **Call Queue->Call Queue Statistics->Overview->Queue Statistics->Options->Details**.

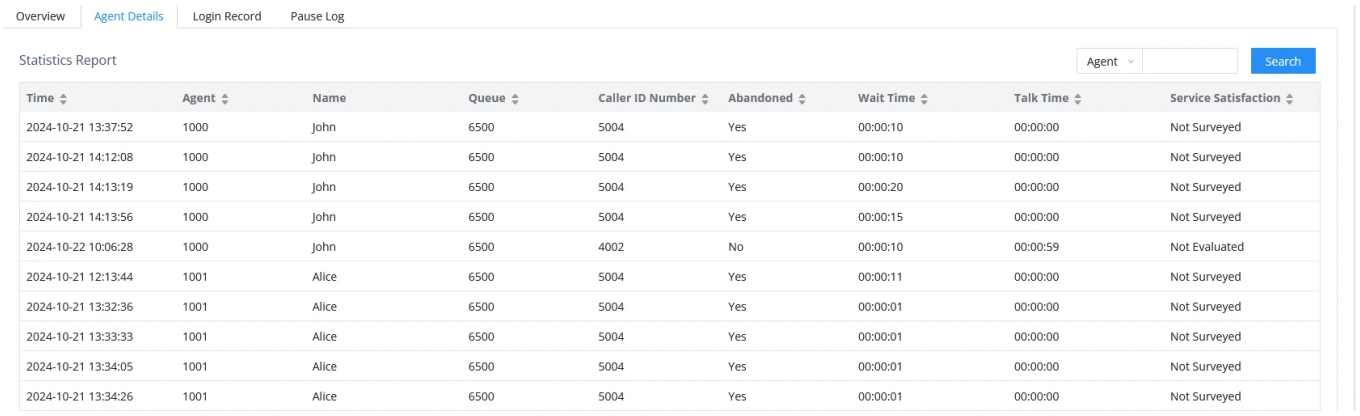


Queue Statistics Details

Agent Details Tab

Agent Detail**s** is a call log that shows every call to each individual agent from all queues. The following information is available:

- **Time:** the date and time the call was received.
- **Agent:** the agent extension that was rung for the call.
- **Name:** the name of to agent that received the call.
- **Queue:** the queue that the call went to.
- **Caller ID Number:** the CID of the caller
- **Abandoned:** indicates whether the call was picked up or not by that specific agent. If the call rang several agents simultaneously, and this specific agent did not pick up the call, the call will be considered abandoned even if a different agent in the same queue picked it up.
- **Wait Time:** the amount of time that the call was waiting in queue after dialing in.
- **Talk Time:** the duration of the call after it was picked up by agent.
- **Service Satisfaction:** Indicates whether the call was not surveyed, not evaluated, or provides the satisfaction score result.

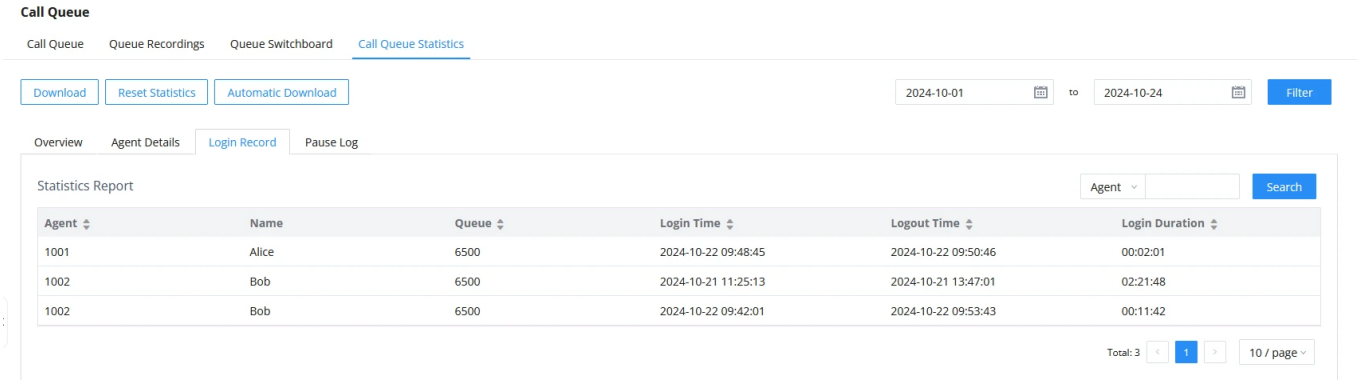


Agent details

Login Record Tab

Login Record is a report that shows the timestamps of dynamic agent logins and logouts and calculates the login duration. The following information is available:

- **Agent:** the extension that logged in and out.
- **Queue:** the queue that the extension logged in and out of.
- **Login Time:** the time that the extension logged into the queue.
- **Logout Time:** the time that the extension logged out of the queue.
- **Login Duration:** the total length of time that the extension was logged in.

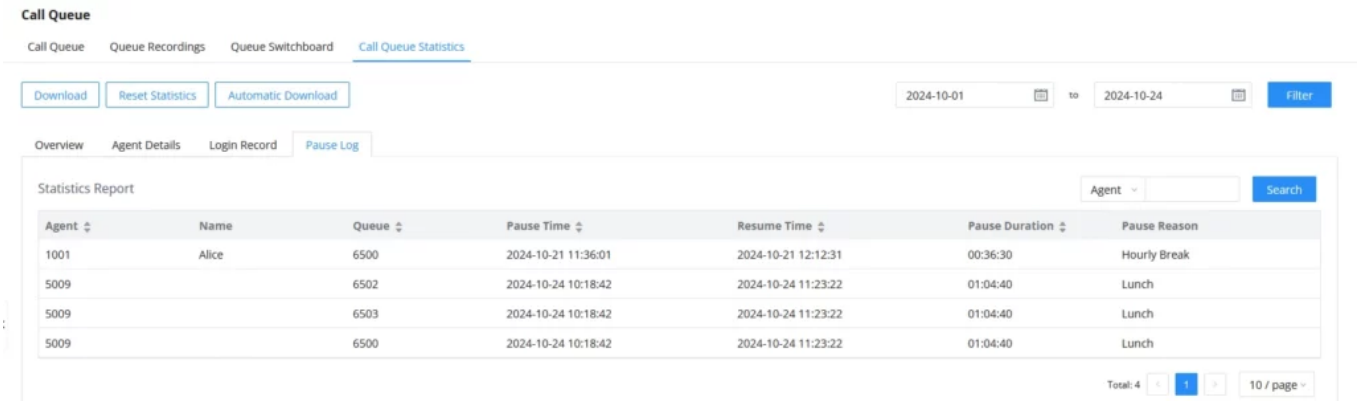


Login Record

Pause Log Tab

Pause Log is a report that shows information related to agent pauses. An entry will only be created after an agent unpauses. The following information is available:

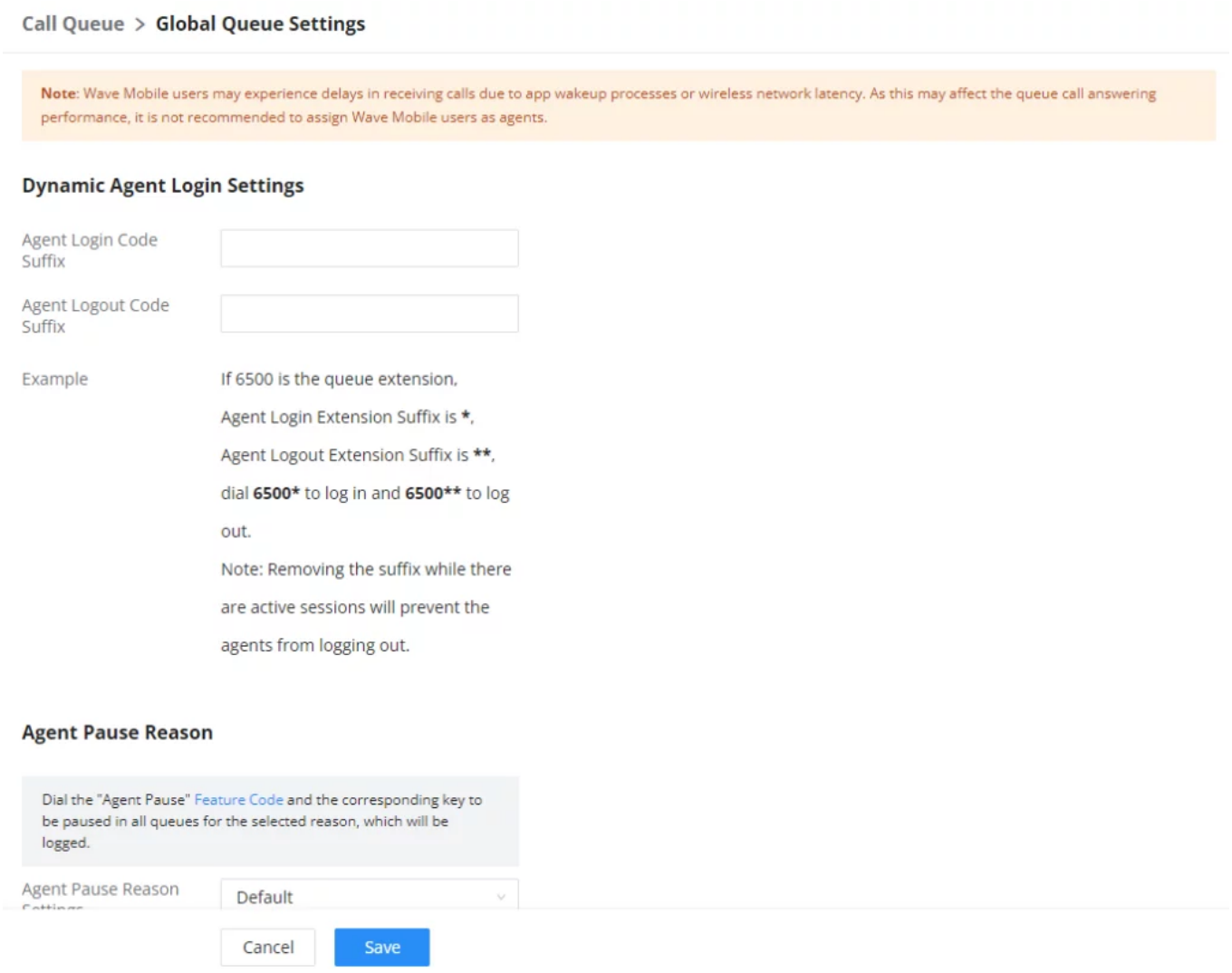
- **Agent**: The extension that paused/unpaused.
- **Name**: The name of the agents that paused/unpaused.
- **Queue**: The queue that the agent is in.
- **Pause Time**: The time when the agent paused.
- **Resume Time**: The time when the agent unpaused.
- **Pause Duration**: The total length of time the agent was paused for.
- **Pause Reason**: The reason of the pause (e.g., lunch, coffee break, etc...)



Pause Log

Global Queue Settings

As explained before, under this section users can configure the feature codes for Dynamic agent login and logout, and also can now customize the keys for virtual queue options like shown below.



Global Queue Settings

Dynamic Agent Login Settings	
Agent Login Code Suffix	Configure the code to dial after the queue extension to log into the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log in.

Agent Logout Code Suffix	Configure the code to dial after the queue extension to log out of the queue (i.e. queue extension + suffix). If no suffix is configured, dynamic agents will not be able to log out.
Agent Pause Reason	
Agent Pause Reason Settings	Select the agent pause reason settings. Default: Use the default settings for the agent pause reason. Custom: User the custom settings. These settings should be configured using they corresponding key for each status. The user can upload a custom prompt which will be played for the agent once they set the pause reason.
Key Settings	Enter which key to press to set the different pause reasons.
Virtual Queue Callback Key Settings	
Enable	Select whether to enable or disable virtual queue callback feature. By default it's disabled.
Call Back Current Number	Press the feature key configured to set your current number as callback number.
Custom Callback Number	Press these feature key configured to set a custom callback number.
Continue Waiting	Press the feature key configured to continue waiting.

Speed Dial

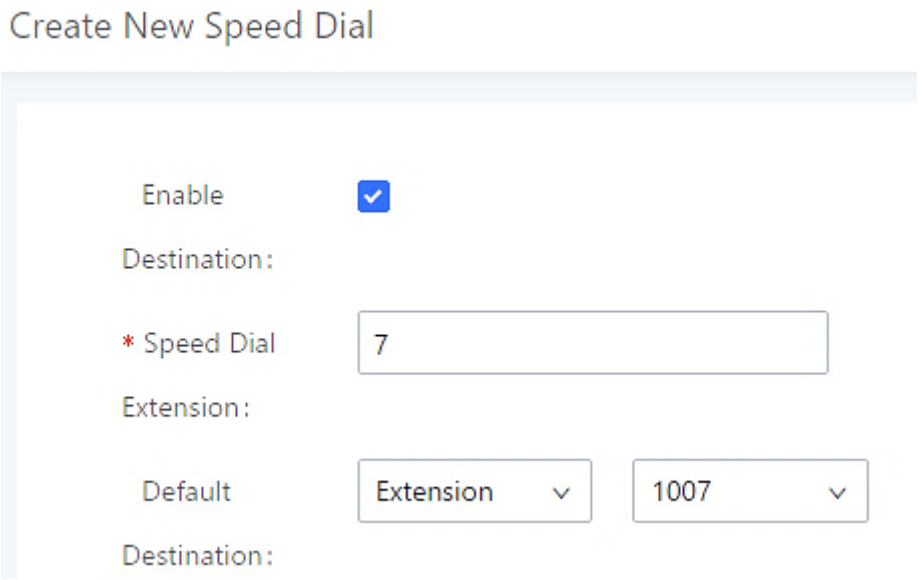
The PBX supports Speed Dial feature that allows users to call a certain destination by pressing one or four digits on the keypad. This creates a system-wide speed dial access for all the extensions on the PBX.

Add Speed Dial

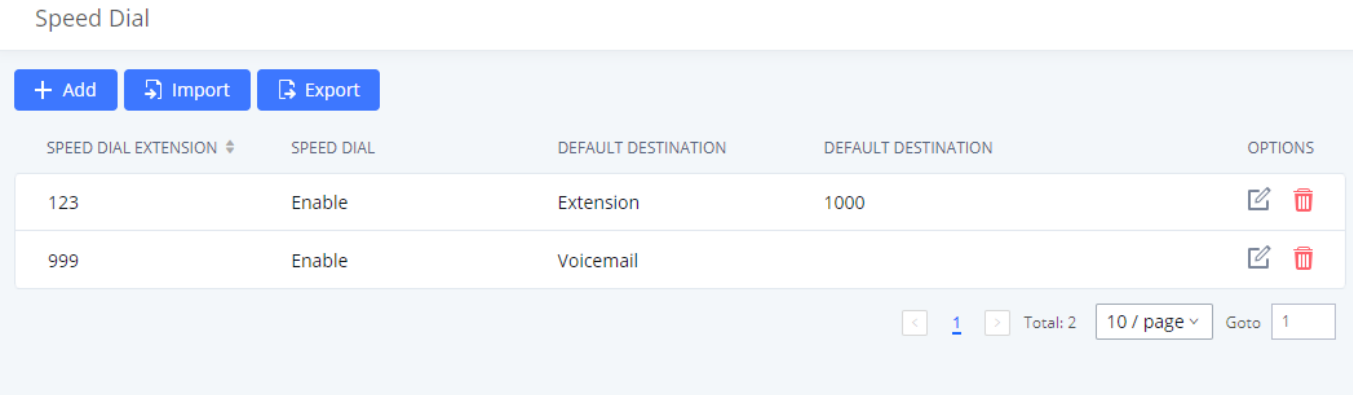
To enable Speed Dial, on the PBX Web GUI, go to page **Basic Call Features→Speed Dial**.

User should first click on . Then decide from one digit up to four digits combination used for Speed Dial and select a dial destination from "Default Destination". The supported destinations include extension, voicemail, conference room, voicemail group, IVR, ring group, call queue, page group, DISA, Dial by Name and external number.

The maximum number of speed dial entries which can be configured is 1000 speed dial entries.



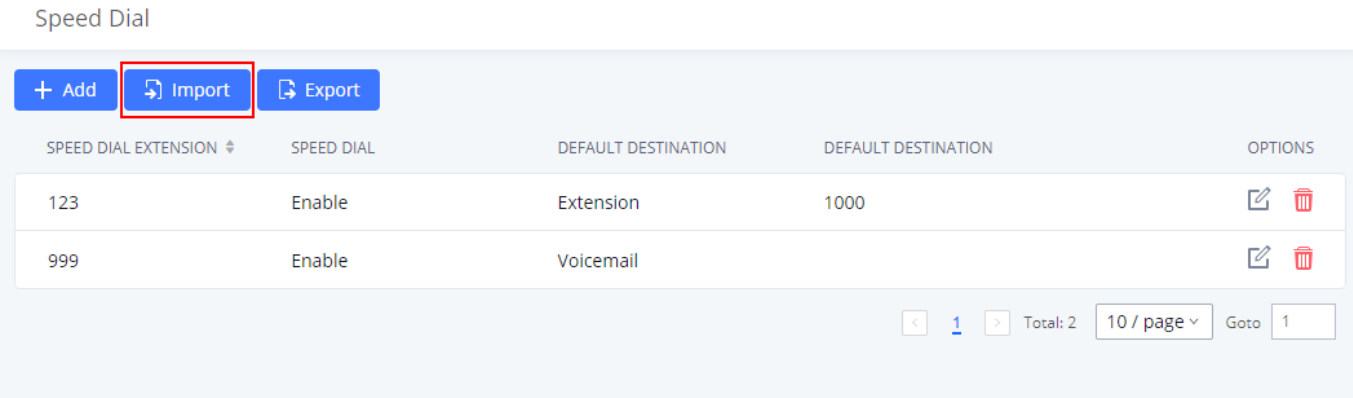
Speed Dial Destinations



List of Speed Dial

Import Speed Dial

The user can import speed dial entries from a csv file, this reduces the amount of configuring the same speed dial entries on different IPPBXs. To do this, please click on "**Import**" as the figure below shows.



Import Dial Speed

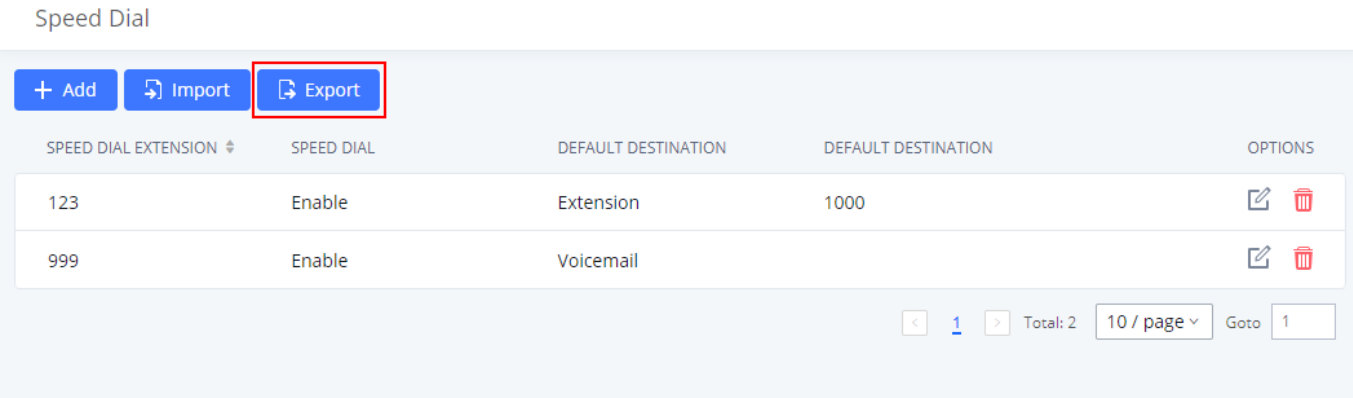
Then select the csv file of the speed dial entries and click

Please use UTF-8 encoding when importing a CSV file. CSV files can be opened using programs such as Notepad and saved as a UTF-8 encoded file.

Importing speed dial entries will overwrite the existing speed dials, if you wish to import new speed dial entries to the already existing ones, you will have to export them then combine them together in one file before you import it.

Export Speed Dial

To export speed dial entries, please click on export as the screenshot below shows, then choose the location where to save the csv file.

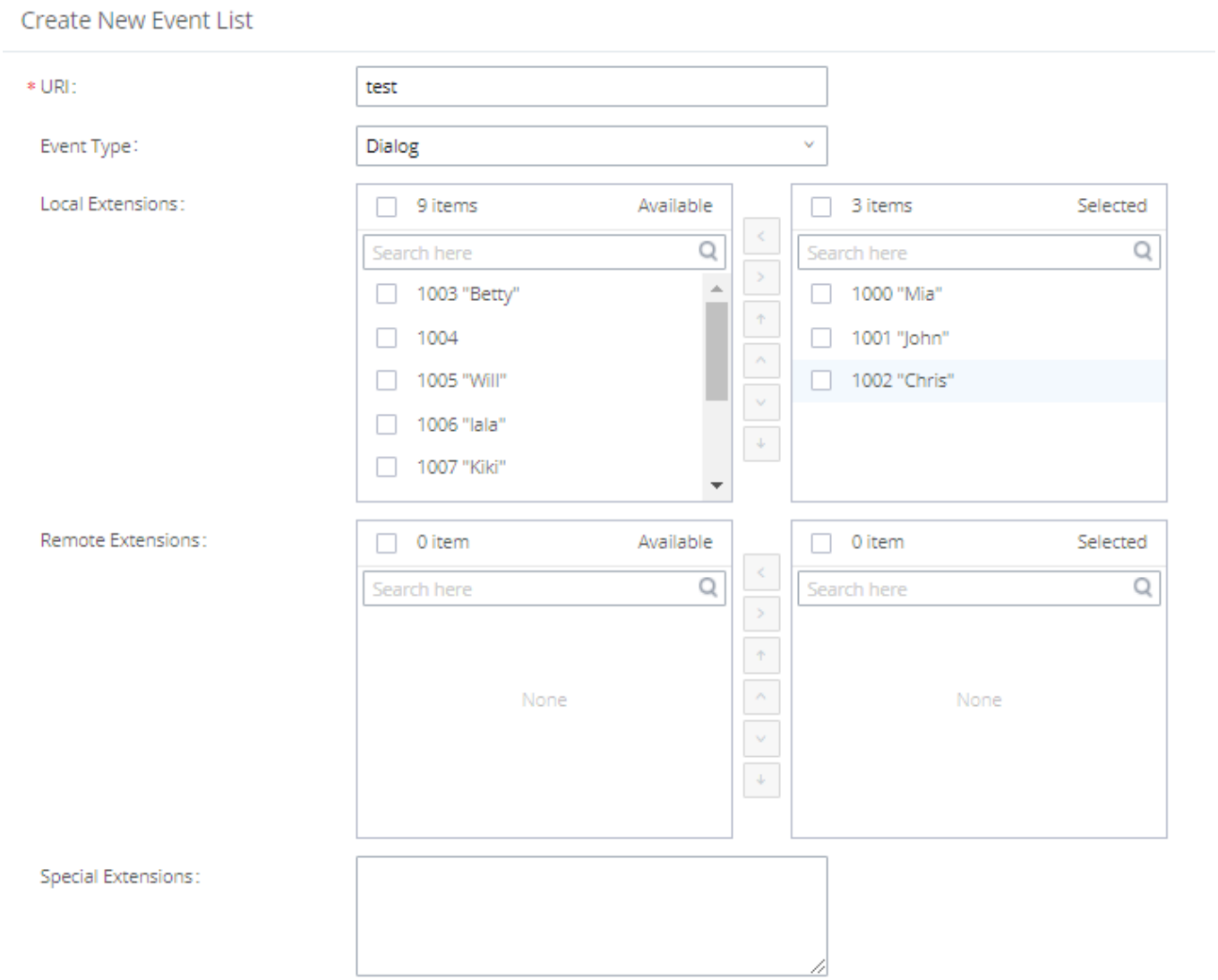


Export Dial Speed

Event List

Besides BLF, users can also configure the phones to monitor event list. In this way, both local extensions on the same PBX and remote extensions on the VOIP trunk can be monitored. The event list setting is under **Basic Call Features→Event List**.

- Click on "Add" to add a new event list.
- Sort selected extensions manually in the Eventlist
- Click on to edit the event list configuration.
- Click on to delete the event list.



Create New Event List

Remote extension monitoring works on the IPPBX via event list BLF, among Peer SIP trunks or Register SIP trunks (register to each other). Therefore, please properly configure SIP trunks on the IPPBX first before using remote BLF feature. Please note the SIP end points need support event list BLF in order to monitor remote extensions.

When an event list is created on the IPPBX and remote extensions are added to the list, the IPPBX will send out SIP SUBSCRIBE to the remote IPPBX to obtain the remote extension status. When the SIP end points register and subscribe to the local IPPBX event list, it can obtain the remote extension status from this event list. Once successfully configured, the event list page will show the status of total extension and subscribers for each event list. Users can also select the event URI to check the monitored extension's status and the subscribers' details.

- To configure LDAP sync, please go to PBX Web GUI→Extension/Trunk→VoIP Trunk. You will see "Sync LDAP Enable" option. Once enabled, please configure password information for the remote peer IPPBX to connect to the local IPPBX. Additional information such as port number, LDAP outbound rule, LDAP Dialed Prefix will also be required. Both the local IPPBX and remote IPPBX need enable LDAP sync option with the same password for successful connection and synchronization.
- Currently LDAP sync feature only works between two IPPBXs.
- (Theoretically) Remote BLF monitoring will work when the remote PBX being monitored is

non-Grandstream PBX. However, it might not work the other way around depending on whether the non-Grandstream PBX supports event list BLF or remote monitoring feature.

URI	Configure the name of this event list (for example, office_event_list). Please note the URI name cannot be the same as the extension name on the IPPBX. The valid characters are letters, digits, _ and -.
Local Extensions	Select the available extensions/Extension Groups listed on the local IPPBX to be monitored in the event list.
Remote Extensions	If LDAP sync is enabled between the IPPBX and the peer IPPBX, the remote extensions will be listed under "Available Extensions". If not, manually enter the remote extensions under "Special Extensions" field.
Special Extensions	Manually enter the remote extensions in the peer/register trunk to be monitored in the event list. Valid format: 5000,5001,9000

Feature Codes

Features codes allow performing certain actions using IP phones. The codes can be directly dialed on the IP phone. Certain feature codes require IVR input, therefore, the user needs to follow the IVR instructions and use DTMF to respond accordingly.

The PBX also allows user to one click enable / disable specific feature code as shown below:

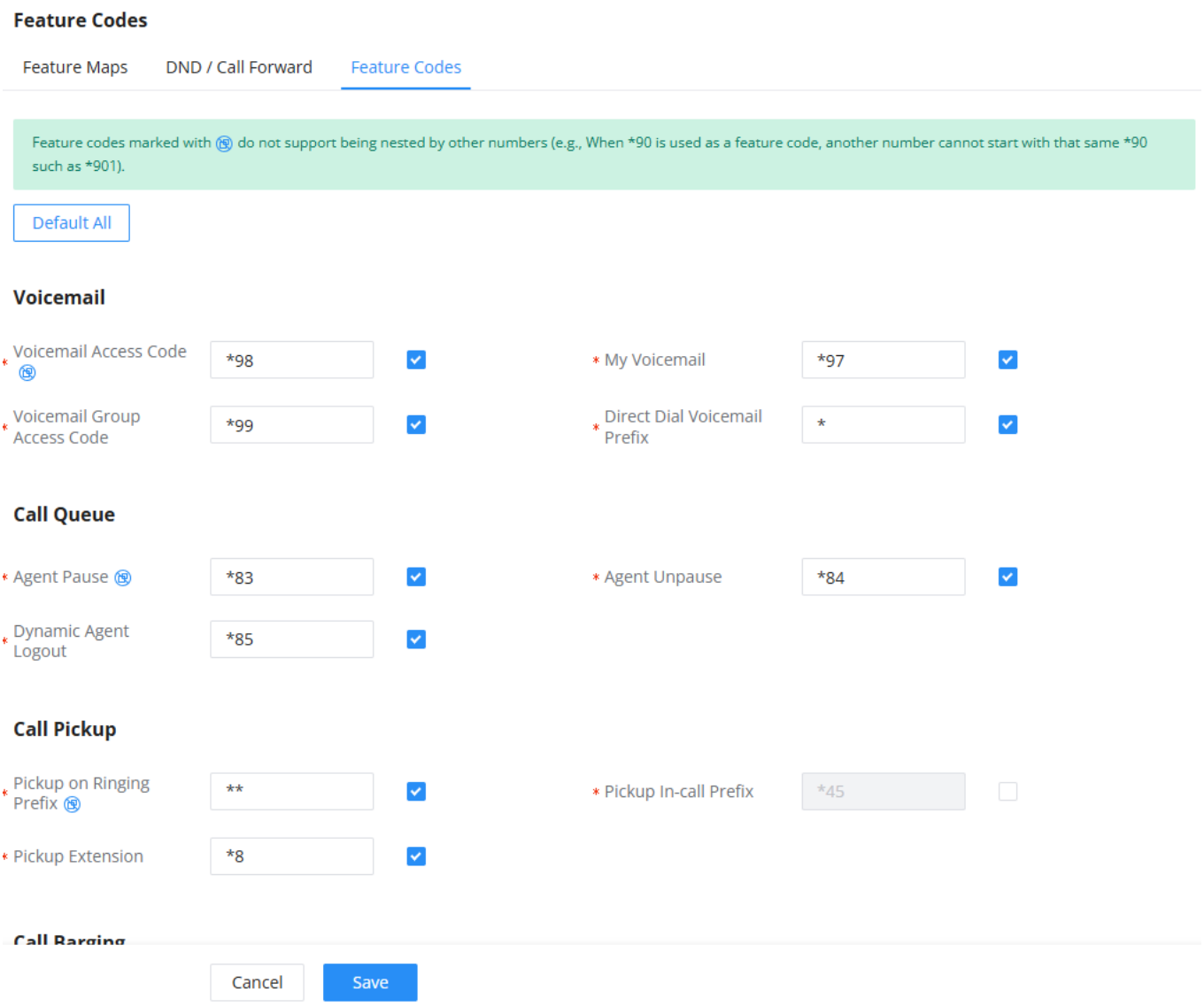
Feature Maps	
Blind Transfer	- Default code: #1- Enter the code during active call. After hearing "Transfer", you will hear dial tone. Enter the number to transfer to. Then the user will be disconnected, and transfer is completed. - Options: Disable Allow Caller: Enable the feature code on caller side only. Allow Callee: Enable the feature code on callee side only. Allow Both: Enable the feature code on both caller and callee.
Attended Transfer	- Default code: *2 - Enter the code during active call. After hearing "Transfer", you will hear the dial tone. Enter the number to transfer to and the user will be connected to this number. Hang up the call to complete the attended transfer. In case of the called party does not answer, users could press *0 to cancel the call and retrieve the first call leg. - Options: Disable Allow Caller: Enable the feature code on caller side only. Allow Callee: Enable the feature code on callee side only. Allow Both: Enable the feature code on both caller and callee.
Transfer Dialing Timeout Period (s)	Configures the dial timeout period of blind and attended transfers.
Seamless Transfer	Default code: *44 (Disabled by default). Seamless Transfer allows user to perform blind transfer using PBX feature code without having music on hold presented during the transfer process, it minimizes the interruption during transfer, making the process smooth and simple. During an active call use the feature code (*44 by default) followed by the number you want to transfer to in order to perform the seamless transfer. (This feature code cannot be nested by other feature codes)
Start/Stop AI Call Transcription	Allows authorized extension users to manually start or stop AI call transcription dynamically during an active call session. The default feature code for this option is *6(This feature code cannot be nested by other feature codes)
Disconnect	- Default code: *0 - Enter the code during active call. It will disconnect the call. - Options: Disable Allow Caller: Enable the feature code on caller side only. Allow Callee: Enable the feature code on callee side only. Allow Both: Enable the feature code on both caller and callee.
Call Park	- Default code: #72- Enter the code during active call to park the call. - Options: Disable Allow Caller: Enable the feature code on caller side only. Allow Callee: Enable the feature code on callee side only. Allow Both: Enable the feature code on both caller and callee.
Feature Code Input Timeout (ms)	Configure the maximum interval (ms) between digits for feature code activation.

Start/Stop Call Recording	-Default code: *3 - Enter the code followed by # or SEND to start recording the audio call and the PBX will mix the streams natively on the fly as the call is in progress. - Options: Disable Allow Caller: Enable the feature code on caller side only. Allow Callee: Enable the feature code on callee side only. Allow Both: Enable the feature code on both caller and callee.
Enable Recording Whitelist	Enable the Recording Whitelist feature
Recording Operation Whitelist	Select extension in the whitelist that can use the *3 recording function.
Feature Code Digits Timeout	Set the maximum interval (ms) between digits for feature code activation
DND/Call Forward	
Call Forward Setting Type	Basic Advanced If Advanced is selected, call forwarding can be set for all calls, internal calls and external calls. To do this, users can dial one of the feature codes below and then dial 0, 1 or 2. The feature code modifiers are as follow: ? 0 corresponds to "All calls".? 1 refers to "Internal calls". (calls that came within the PBX)? 2 is used for "External calls". (calls from outside the PBX)
Do Not Disturb (DND) Activate	Default code: *77Activate DND feature to ignore any incoming calls.
Do Not Disturb (DND) Deactivate	Default code: *78Deactivate DND Feature.
Call Forward Busy Enable	Default Code: *90 Enables Call Forward Busy (CFB) for the dialing extension.Assuming feature code is xxx, the following call forward setting methods are available :Method 1: Dial xxx and follow the system prompts.Method 2: Dial xxx + target extension (e.g., xxx6000).Method 3: Dial xxx + 0/1/2 + target extension (e.g., xxx16000).Methods 1 and 2 are supported in Basic.Methods 1 and 3 are supported in Advanced.(This feature code cannot be nested by other feature codes)
Call Forward Busy Disable	Default Code: *91Disables Call Forward Busy (CFB) for the dialing extension.Assuming feature code is xxx, use one of the following methods based on your Call Forward Setting Type: Basic: Dial xxx. Advanced: Dial xxx + 0/1/2.
Call Forward No Answer Enable	Default Code: *92 Enables Call Forward No Answer (CFNA) for the dialing extension.Assuming feature code is xxx, the following call forward setting methods are available :Method 1: Dial xxx and follow the system prompts.Method 2: Dial xxx + target extension (e.g., xxx6000).Method 3: Dial xxx + 0/1/2 + target extension (e.g., xxx16000).Methods 1 and 2 are supported in Basic.Methods 1 and 3 are supported in Advanced.(This feature code cannot be nested by other feature codes)
Call Forward No Answer Disable	Default Code: *93Disables Call Forward No Answer (CFNA) for the dialing extension.Assuming feature code is xxx, use one of the following methods based on your Call Forward Setting Type: Basic: Dial xxx. Advanced: Dial xxx + 0/1/2.
Call Forward Always Enable	Default Code: *72Enables Call Forward Always (CFA) for the dialing extension.Assuming feature code is xxx, the following call forward setting methods are available :Method 1: Dial xxx and follow the system prompts.Method 2: Dial xxx + target extension (e.g., xxx6000).Method 3: Dial xxx + 0/1/2 + target extension (e.g., xxx16000).Methods 1 and 2 are supported in Basic.Methods 1 and 3 are supported in Advanced. (This feature code cannot be nested by other feature codes)
Call Forward Always Disable	Default Code: *73Disables Call Forward Always (CFA) for the dialing extension.Assuming feature code is xxx, use one of the following methods based on your Call Forward Setting Type: Basic: Dial xxx. Advanced: Dial xxx + 0/1/2.
Remote Call Forward Enable	Enable this option and configure the Remote Call Forward Whitelist below to allow specific extensions to dial the remote call forwarding feature codes to set call forwarding for any extension.
Remote DND / Call Forward Settings	
Enable	Enable this option and configure the Whitelist below to allow specific extensions to dial feature codes to set DND or call forwarding for any extension.

Remote Call Forward Busy Enable	Default code: *65 Configures and enables CFB for any extension.
Remote Call Forward No Answer Enable	Default code: *66Configures and enables CFNA for any extension.
Remote Call Forward Always Enable	Default code: *67Configures and enables CFU for any extension.
Remote DND Enable	Default code: *68Enables Do Not Disturb for any extension.
Remote Call Forward Busy Disable	Default code: *651Disables CFB for any extension.
Remote Call Forward No Answer Disable	Default code: *661Disables CFNA for any extension.
Remote Call Forward Always Disable	Default code: *671Disables CFU for any extension.
Remote DND Disable	Default code: *681Disables Do Not Disturb for any extension.
Whitelist	Extensions in this whitelist can configure DND or call forwarding for any extension via feature codes.
Feature Codes	
Voicemail	
Voicemail Access Code	- Default code: *98 - Enter *98 and follow the voice prompt. Or dial *98 followed by the extension and # to access the entered extension's voicemail box.(This feature code cannot be nested by other feature codes)
My Voicemail	- Default code: *97 - Press *97 to access the voicemail box.
Voicemail Group Access Code	Default code: *99Dial this code to access group voicemail. If password is required, enter password followed by the pound (#) key.If Allow Members to Modify Voice Prompts is enabled under Voicemail Group settings, group members dialing *99 can also record, update, or customize voice prompts and greetings for the Voicemail Group via DTMF audio menus.
Direct Dial Voicemail Prefix	Prefix used to dial directly to voicemail.
Call Queue	
Agent Pause	Default code: *83Pause the agent in all call queues.(This feature code cannot be nested by other feature codes)
Agent Unpause	Default code: *84Unpause the agent in all call queues.
Dynamic Agent Logout	Default code: *85 Log the dynamic agent out of all queues.
Call Pickup	
Pickup on Ringing Prefix	Picks up a ringing call for another extension. Example: If the prefix is **, and there is a call ringing ext 1008, dial **1008 from a different extension to pick up the call to 1008. (This feature code cannot be nested by other feature codes)
Pickup In-call Prefix	Picks up an ongoing call for another extension. Example: If the feature code is *45, and ext 1008 is in a call, dialing *45 and then 1008 following the prompt will take that call. Note: The feature code user must be in the extension's Allowed to seamless transfer list to pick up calls for it.
Pickup Extension	This is the feature code to pick up incoming calls for other extensions in the same pickup group. The default setting is *8.
Call Barging	
Enable Spy	Check this box to enable spy feature codes.

Listen Spy	This is the feature code to listen in on a call to monitor performance. Your line will be muted, and neither party will hear you. The default setting is *54.(This feature code cannot be nested by other feature codes)
Barge Spy	This is the feature code to join in on the call to assist both parties. The default setting is *56.(This feature code cannot be nested by other feature codes)
Whisper Spy	This is the feature code to speak to only one party in the call. For example, you could whisper to employees to help them handle a call. Only an employee on your account will be able to hear you. The default setting is *55.(This feature code cannot be nested by other feature codes)
PMS	
PMS Wakeup Service	Dial this feature code to access PMS Wakeup Service. You can add, update, activate or deactivate PMS Wakeup Service.
PMS Remote Wakeup Service	Dial this code to add, update, activate, and deactivate PMS wakeup service for other extensions.
Update PMS Room Status	2 methods are available:1. Dial the room status feature code + housekeeper code, listen to the prompt and then the dial the appropriate key for the desired room status. Example: The housekeeper with housekeeper code 0001 dials *230001, listens to the room status options prompt, and then dials 1 to change room status to Available.2. Dial room status feature code*housekeeper code*desired room status option key to quickly change the room status without needing to go through the system voice prompts. Example: Housekeeper with Housekeeper code 0001 dials *23*0001*1 to change room status Available.(This feature code cannot be nested by other feature codes)
Misc	
Paging Prefix	Configure the paging prefix for paging. For example, if the Paging Prefix is set to *81, dial *816000 to initiate a paging call to extension 6000.(This feature code cannot be nested by other feature codes)
Intercom Prefix	Configure the intercom prefix for intercom calls. For example, if the Intercom Prefix is set to *80, dial *806000 to initiate an intercom call to extension 6000.(This feature code cannot be nested by other feature codes)
Blacklist Add	Follow the voice prompt to add a caller ID to blacklist.
Blacklist Last Caller	Add the last inbound caller ID number to blacklist.
Blacklist Remove	Follow the voice prompt to remove a caller ID from blacklist.
Direct Dial Mobile Phone Prefix	If calling mobile phone numbers is permitted, use this prefix plus the extension number to dial the mobile phone number of this extension directly.(This feature code cannot be nested by other feature codes)
Call Completion Request	If the caller wants to use CC to complete a call, he/she can dial this code. After the CC has been registered successfully, the system will start to monitor the status of the callee. The system will call back the caller when the callee's extension is available.

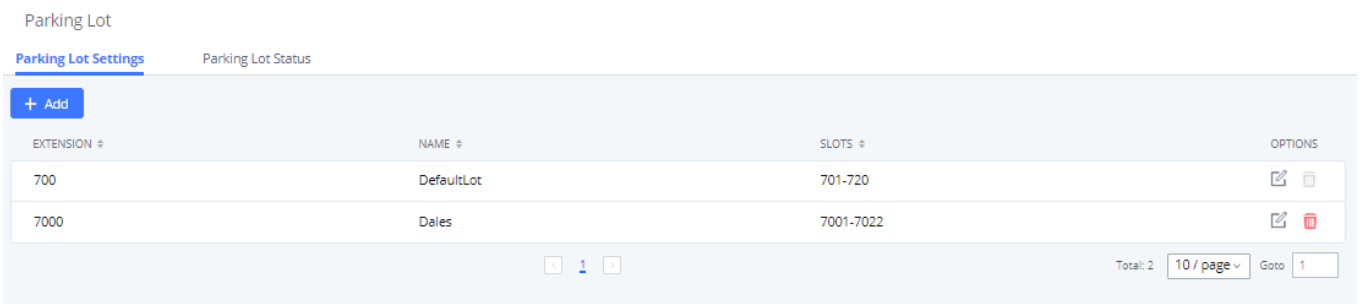
Call Completion Cancel	If the caller has requested CC successfully, and he/she doesn't need to call back anymore, he/she can dial this code to cancel the request.
Presence Status	Dial this feature code to set the presence status of the extension.
Call Flip	- Default code: *46- Dial this code to move the call of this extension from another device to the current device.
Wakeup Service	Dial this feature code to access PBX Wakeup Service. You can add, update, activate or deactivate PBX Wakeup Service.
Remote Extension Privilege Update	Whitelisted extensions will be able to use the Remote Extension Privilege Update feature code to remotely change any extension's outgoing call privilege.Note: After this function has been enabled, the extension is the whitelist can set the privilege for outgoing calls of any extension by dialing the feature code.
Remote Extension Privilege Update Whitelist	Remote Extension Privilege Update WhitelistProcedure: Dial *26 on the whitelisted extension, hear the prompt "Change extension's outgoing permission level, please enter the phone number, then enter # key." After the process, voice will prompt "Press 1 to set to internal, press 2 to set to local, press 3 to set to national, press 4 to set to international." After selecting, it will prompt "Change extension XXXX outgoing permission to XXX", and hang up.
Privileged Call	Dial the feature code + extension number, for example *001002 to hang up the current call of extension 1002 with privilege, and then call extension 1002.
Privileged Call Whitelist	The extensions in this white list can use the feature code of the privileged call.
Priority Call	Makes a high priority call to a specified extension by dialing the feature + extension number. If the specified extension is an ongoing call, it will be forcibly hung up to allow the caller to ring the extension. (This feature code cannot be nested by other feature codes)
Priority Call Whitelist	Extension in the whitelist will be allowed to use this feature code to make priority calls.



EnableDisable Feature codes

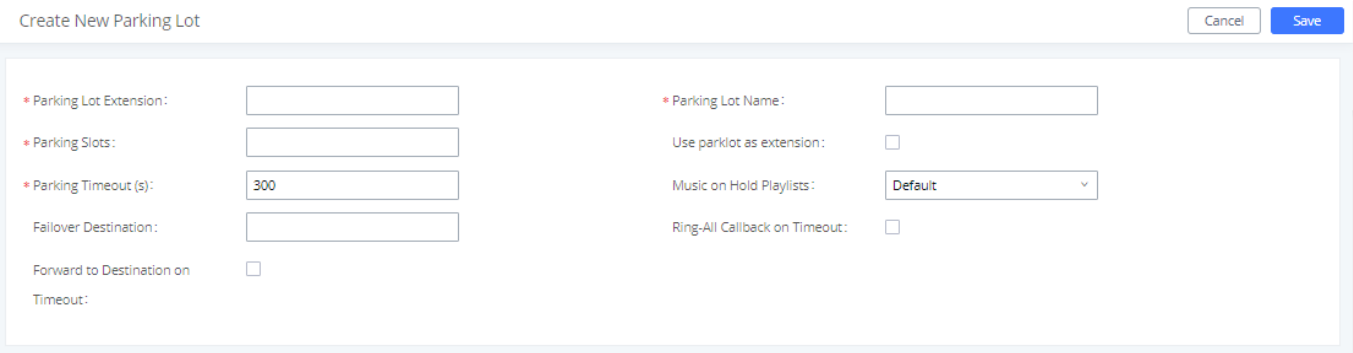
Parking Lot

User can create parking lots and their related slots under Web GUI **Basic Call Features > Parking Lot.** In the Parking Lot page, users can create lots of their own. This allows different groups within an organization to have their own parking lots instead of sharing one large parking lot with others. While creating a new parking lot, users can assign it a range that they think is appropriate for the group that will use the parking lot.



Parking Lot

User can create a new Parking lot by clicking on button "Add" :



New Parking Lot

Parking Lot Extension	Default Extension: 700 During an active call, initiate blind transfer and then enter this code to park the call.
Parking Lot Name	Set a name to the parking lot
Parked Slots	Default Extension: 701-720 These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.
Use Parklot as Extension	If checked, the parking lot number can be used as extension. The user can transfer the call to the parking lot number to park the call. Please note this parking lot number range might conflict with extension range.
Parking Timeout (s)	Default setting is 300 seconds, and the maximum limit is 99.999 seconds. This is the timeout allowed for a call to be parked. After the timeout, if the call is not picked up, the extension who parks the call will be called back.
Music On Hold Classes	Select the Music on Hold Class.
Failover Destination	Configures a callback failover destination when the extension that is called back is busy. The call will be routed to the destination number and this reduces the chance of dropping parked calls.
Ring All Callback on Timeout	If enabled, all registered endpoints of the extension will ring when callback occurs. Otherwise, only the original endpoint will be called back.
Forward to destination on timeout	If enabled, the call will be routed to the configured destination upon timeout. Otherwise, the call will be routed back to the original caller.
Timeout Destination	This option appears once Forward to Destination on Timeout is enabled. Upon park timeout, the call will be routed to the configured destination.
Parking Lot Timeout Alert-Info	Adds an Alert-Info header to parking lot callbacks after the Parking Timeout has been reached.

Parking Lot

Call Park

The PBX provides call park and call pickup features via feature code.

Park a Call

There are two feature codes that can be used to park the call.

- **Feature Maps→Call Park (Default code #72)**

During an active call, press #72 and the call will be parked. Parking lot number (default range 701 to 720) will be announced after parking the call.

- **Feature Misc→Call Park (Default code 700)**

During an active call, initiate blind transfer (default code #1) and then dial 700 to park the call. Parking lot number (default range 701 to 720) will be announced after parking the call.

Retrieve Parked Call

To retrieve the parked call, simply dial the parking lot number and the call will be established. If a parked call is not retrieved after the timeout, the original extension who parks the call will be called back.

Monitor Call Park CID Name Information (GXP21xx, GRP261x Phones Only)

Users can see the CID name information of parked calls. VPK/MPKs must be configured as "Monitored Call Park" with the desired parking lot extension. The display will alternate between displaying the parking lot extension and the call's CID name. There is no need to configure anything on the IPPBX.

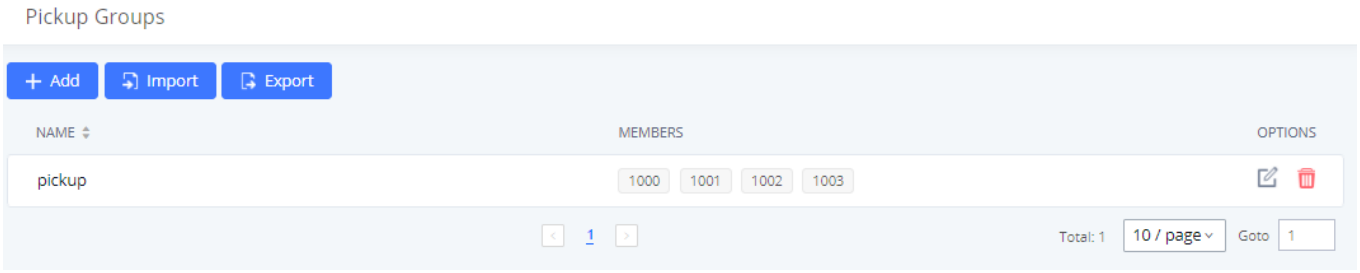
Monitored Call Park CID name

ADVANCED CALL FEATURES

Pickup Groups

The PBX supports pickup group feature which allows users to pick up incoming calls for other extensions if they are in the same pickup group, by dialing "Pickup Extension" feature code (by default *8).

Configure Pickup Groups

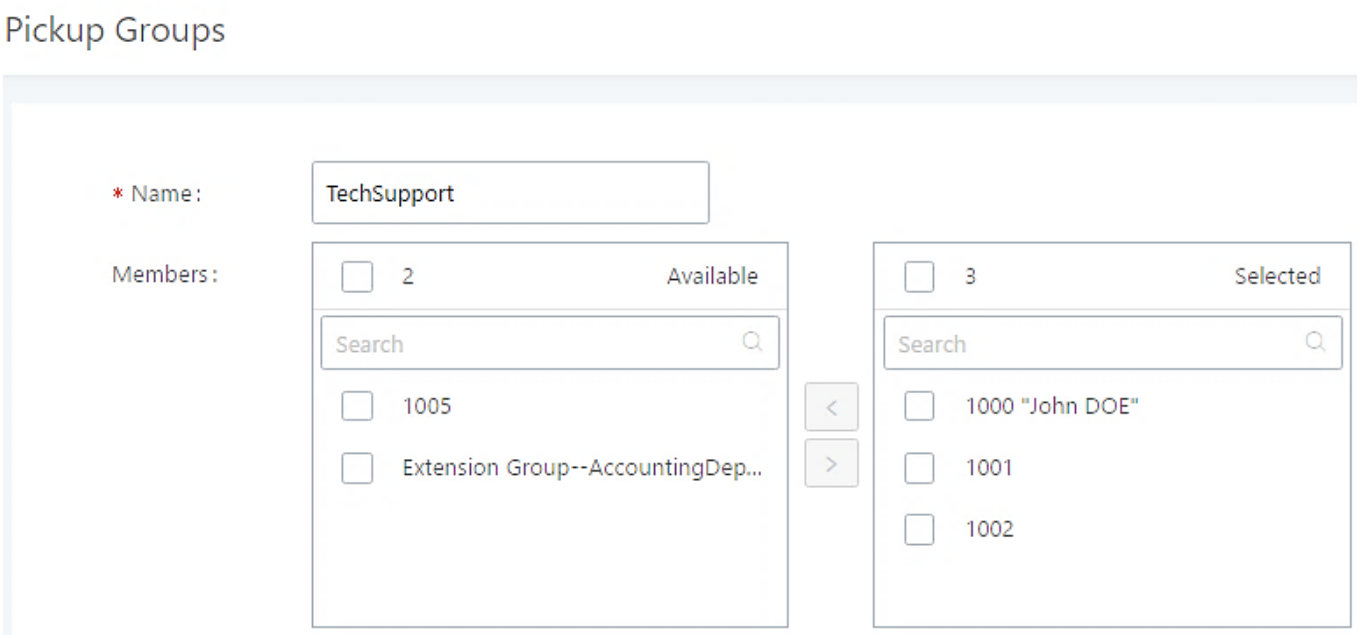


Pickup Groups interface

Pickup groups can be configured via Web GUI→**Advanced Call Features**→**Pickup Groups**.

- Click on to create a new pickup group.
- Click the button to upload the pickup group information in CSV format.
- Click the button to generate pickup group information in .CSV format.
- Click on to edit the pickup group.
- Click on to delete the pickup group.

Select extensions from the list on the left side to the right side.

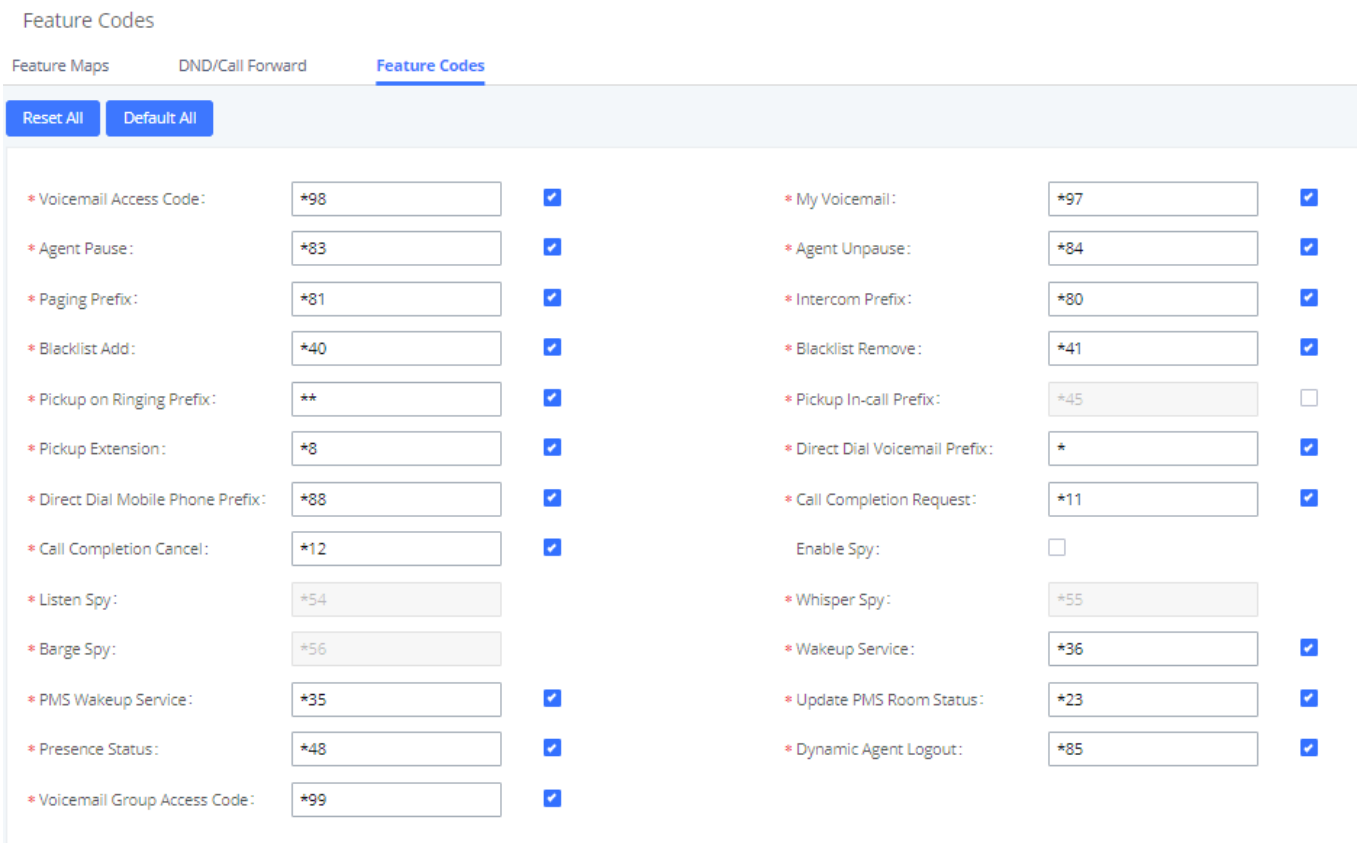


Edit Pickup Group

Configure Pickup Feature Code

When picking up the call for the pickup group member, the user only needs to dial the pickup feature code. It is not necessary to add the extension number after the pickup feature code. The pickup feature code is configurable under **Basic Call Features→Feature Codes**.

The default feature code for call pickup extension is *8, otherwise if the person intending to pick up the call knows the ringing extension they can use ** followed by the extension number in order to perform the call pickup operation. The following figure shows where you can customize these features codes



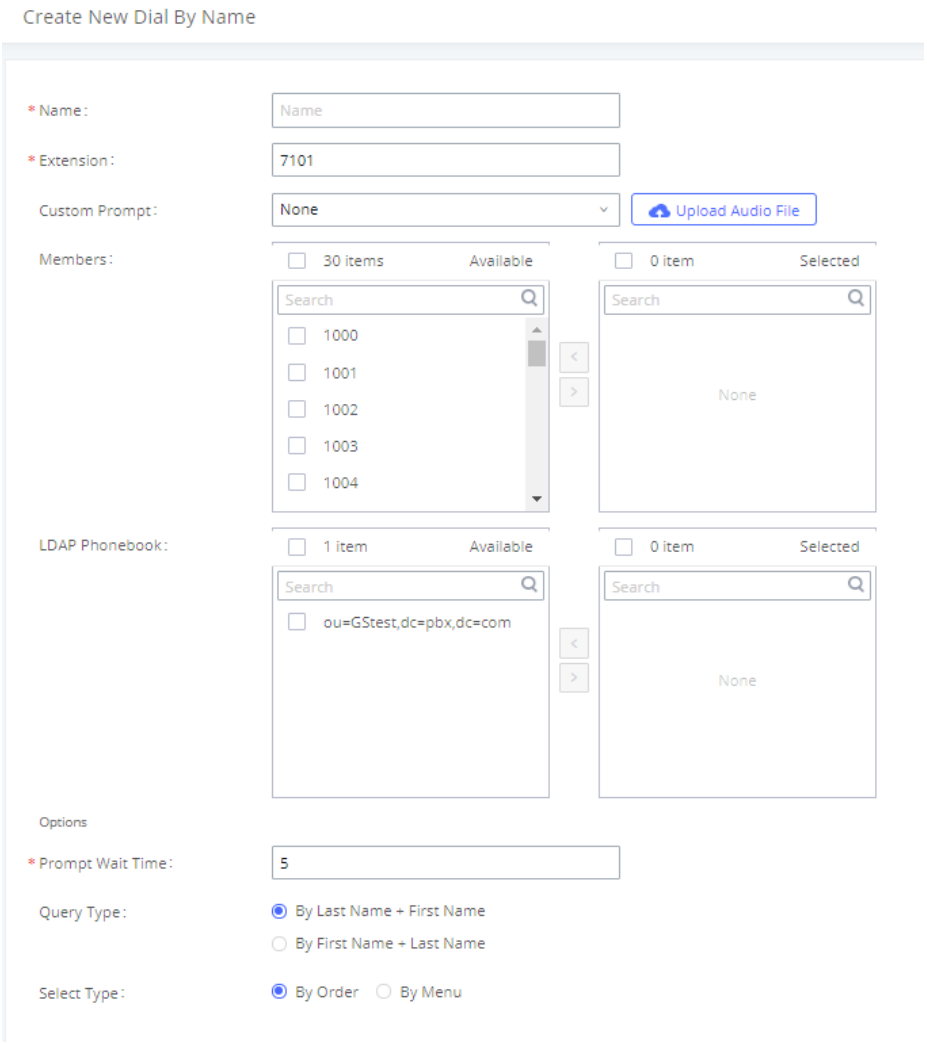
Edit Pickup Feature Code

Dial By Name

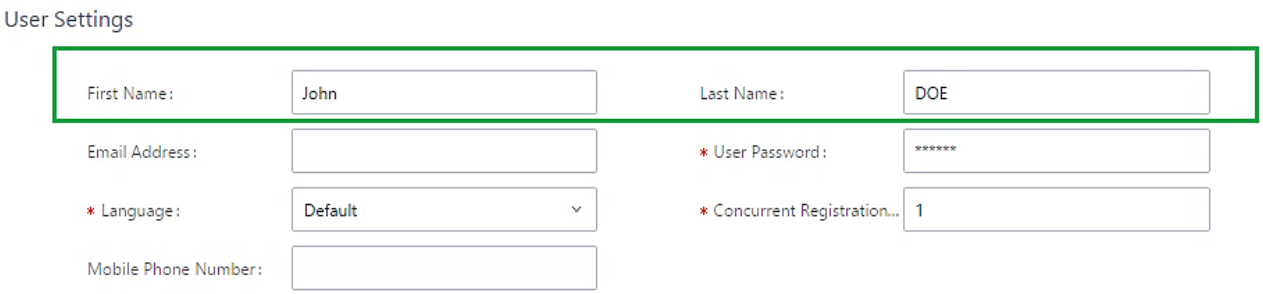
Dial by Name is a feature on the PBX that allows caller to search a person by first or last name via his/her phone's keypad. The administrator can define the Dial by Name directory including the desired extensions in the directory and the searching type by "first name" or "last name". After dialing in, the PBX IVR/Auto Attendant will guide the caller to spell the digits to find the person in the Dial by Name directory. This feature allows customers/clients to use the guided automatic system to contact the enterprise employees without having to know the extension number, which brings convenience and improves business image for the enterprise.

Dial by Name Configuration

The administrators can create the dial by name group under **Advanced Call Features→Dial By Name**.



Create Dial by Name Group



Configure Extension First Name and Last Name

1. **Name**

   Enter a Name to identify the Dial by Name group.

2. **Extension**

   Configure the direct dial extension for the Dial By Name group.

3. **Custom Prompt**

   This option sets a custom prompt for directory to announce to a caller. The file can be uploaded from the page "Custom Prompt". Click "Upload Audio File" to add additional record.

4. **Available Extensions/Selected Extensions**

   Select available extensions from the left side to the right side as the directory for the Dial By Name group. Only the selected extensions here can be reached by the Dial By Name IVR when dialing into this group. The extensions here must have a valid first name and last name configured under Web GUI→ **Extension/Trunk→Extensions** in order to be searchable in Dial By Name directory through IVR. By specifying the extensions here, the administrators can make sure unscreened calls will not reach the company employee if he/she does not want to receive them directly.

5. **Prompt Wait Time**

   Configure "Prompt Wait Time" for Dial By Name feature. During Dial By Name call, the caller will need to input the first letters of First/Last name before this wait time is reached. Otherwise, timeout will occur, and the call might hang up. The timeout range is between 3 and 60 seconds.

6. **Query Type**

   Specify the query type. This defines how the caller will need to enter to search the directory.
   By First Name: enter the first 3 digits of the first name to search the directory.
   By Last Name: enter the first 3 digits of the last name to search the directory.

7. **Select Type**

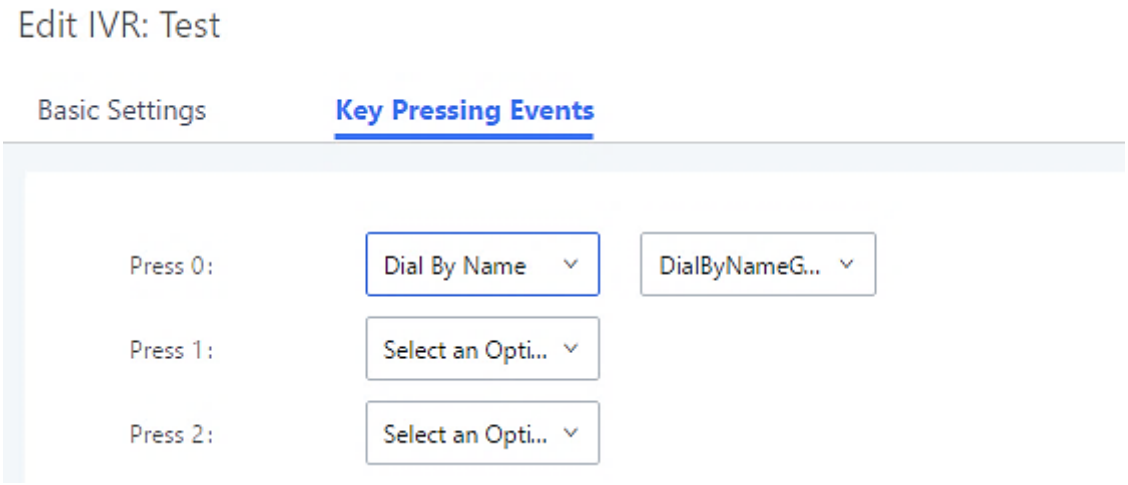
   Specify the select type on the searching result. The IVR will confirm the name/number for the party the caller would like to reach before dialing out.

   By Order: After the caller enters the digits, the IVR will announce the first matching party's name and number. The caller can confirm and dial out if it is the destination party, or press * to listen to the next matching result if it is not the desired party to call.

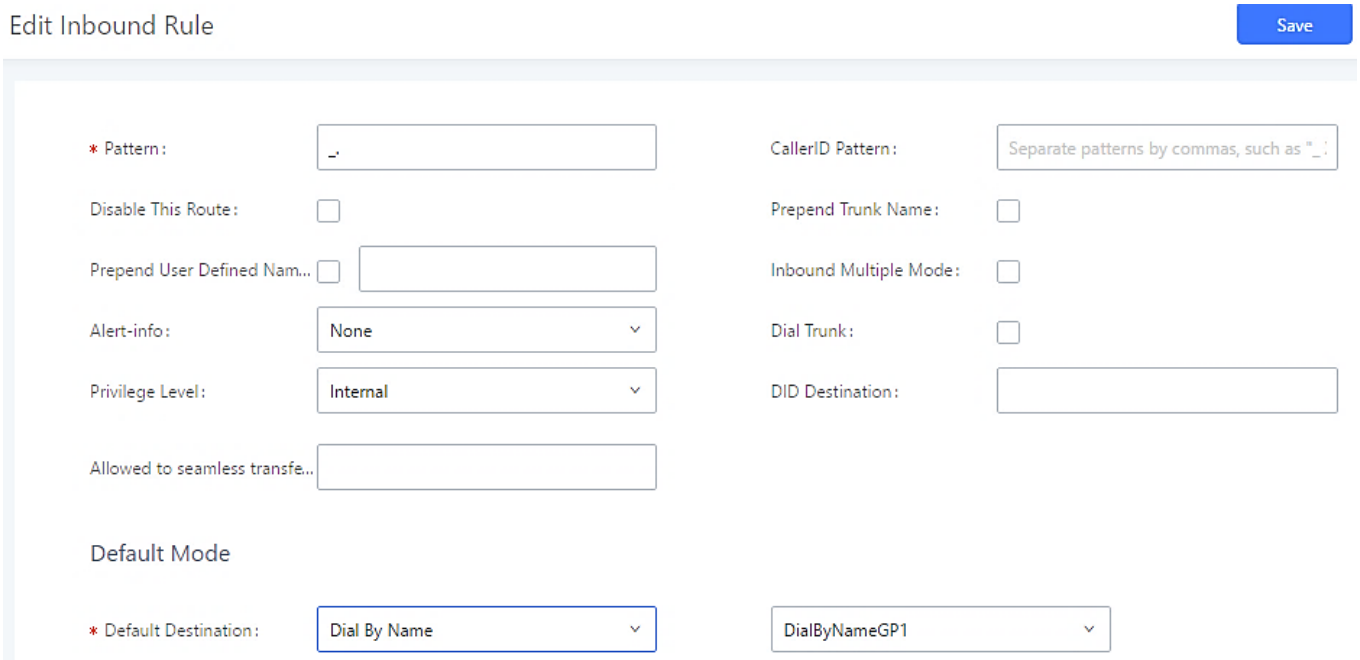
   By Menu: After the caller enters the digits, the IVR will announce 8 matching results. The caller can press

number 1 to 8 to select and call or press 9 for results in next page.

The Dial by Name group can be used as the destination for inbound route and key pressing event for IVR. The group name defined here will show up in the destination list when configuring IVR and inbound route. If Dial by Name is set as a key pressing event for IVR, user could use '*' to exit from Dial by Name, then re-enter IVR and start a new event. The following example shows how to use this option.



Dial By Name Group In IVR Key Pressing Events



Dial by Name Group In Inbound Rule

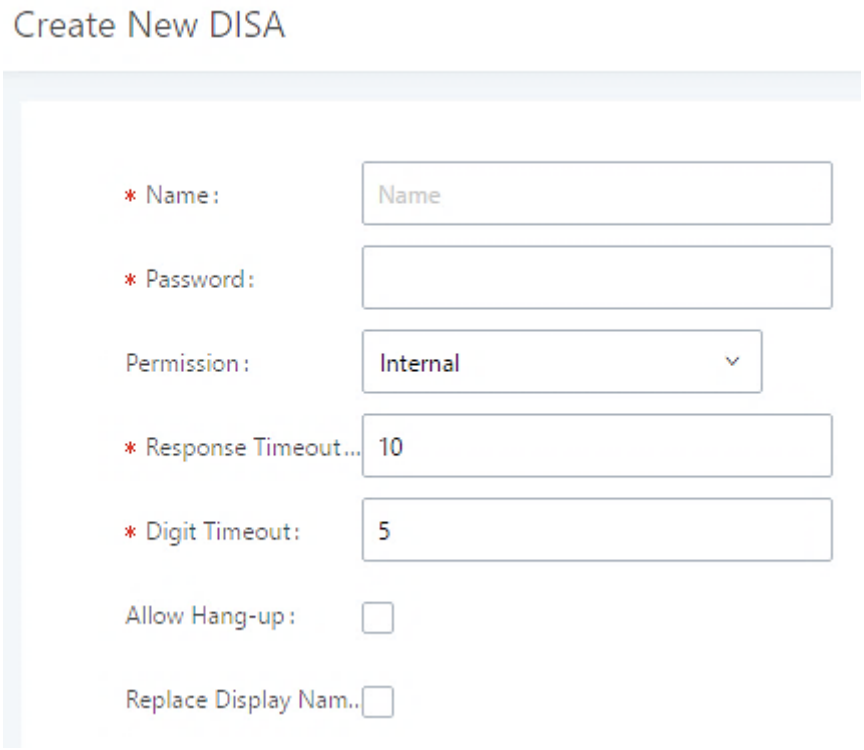
Please refer to [Username Prompt] for Username Prompt customization.

DISA

In many situations, the user will find the need to access his own IP PBX resources, but he is not physically near one of his extensions. However, he does have access to his own cell phone. In this case, we can use what is commonly known as DISA (Direct Inward System Access). Under this scenario, the user will be able to call from the outside, whether it is using his cell phone, pay phone, etc. After calling into the PBX, the user can then dial out via the SIP trunk connected to the PBX as it is an internal extension.

The PBX supports DISA to be used in IVR or inbound route. Before using it, create new DISA under Web **Advanced Call Features→DISA**.

- Click on to add a new DISA.
- Click on to edit the DISA configuration.
- Click on to delete the DISA.



Create New DISA

The following table details the parameters to set and configure DISA feature on PBX.

Name	Configure DISA name to identify the DISA.
Password	Configure the password (digit only) required for the user to enter before using DISA to dial out.Note: The password must be at least 4 digits.
Permission	Configure the permission level for DISA. The available permissions are "Internal", "Local", "National" and "International" from the lowest level to the highest level. The default setting is "Internal". If the user tries to dial outbound calls after dialing into the DISA, the IPPBX will compared the DISA's permission level with the outbound route's privilege level. If the DISA's permission level is higher than (or equal to) the outbound route's privilege level, the call will be allowed to go through.
Response Timeout	Configure the maximum amount of time the IPPBX will wait before hanging up if the user dials an incomplete or invalid number. The default setting is 10 seconds.
Digit Timeout	Configure the maximum amount of time permitted between digits when the user is typing the extension. The default setting is 5 seconds.
Allow Hangup	If enabled, during an active call, users can enter the IPPBX Hangup feature code (by default it is *0) to disconnect the call or hang up directly. A new dial tone will be heard shortly for the user to make a new call. The default setting is "No".
Replace Display Name	If enabled, the IPPBX will replace the caller display name with the DISA name.

Once successfully created, users can configure the inbound route destination as "DISA" or IVR key event as "DISA". When dialing into DISA, users will be prompted with password first. After entering the correct password, a second dial tone will be heard for the users to dial out.

Callback

Callback is designed for users who often use their mobile phones to make long distance or international calls which may have high service charges. The callback feature provides an economic solution for reduce the cost from this.

The callback feature works as follows:

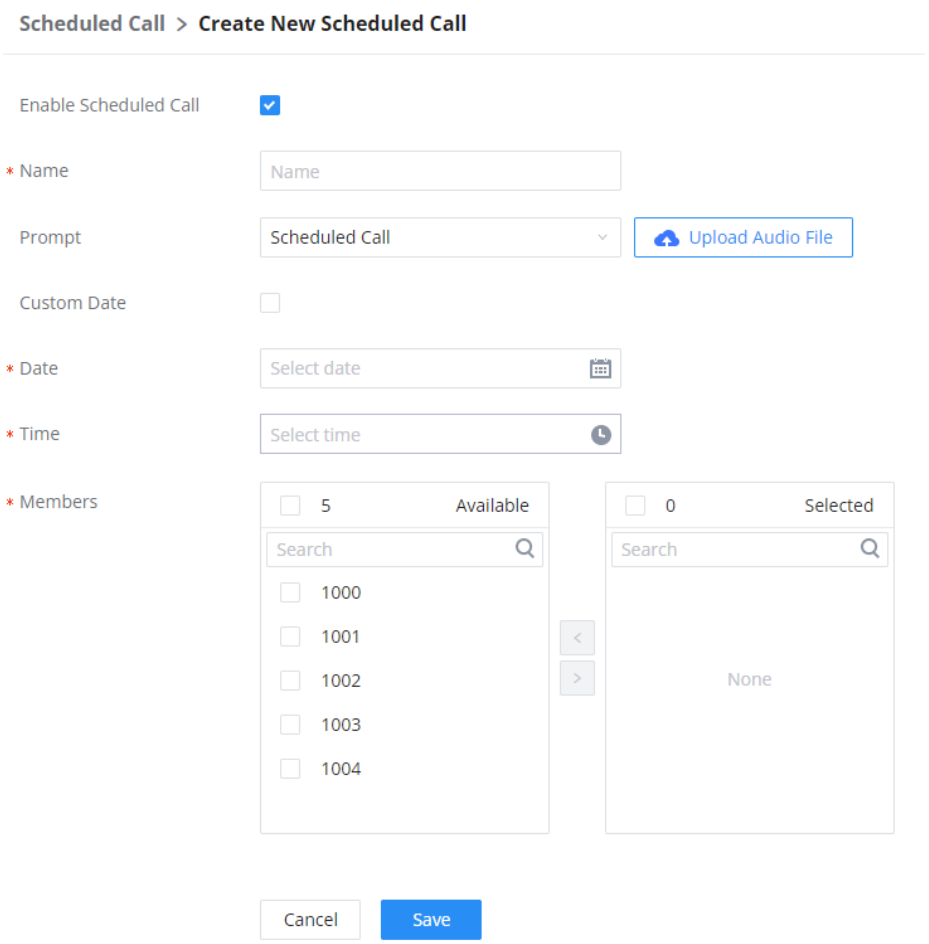
1. Configure a new callback on the PBX.
2. On the PBX, configure destination of the inbound route for callback.
3. Save and apply the settings.
4. The user calls number of the PBX using the mobile phone, which goes to callback destination as specified in the inbound route.
5. Once the user hears the ringback tone from the mobile phone, hang up the call on the mobile phone.
6. The PBX will call back the user.
7. The user answers the call.
8. The call will be sent to DISA or IVR which directs the user to dial the destination number.
9. The user will be connected to the destination number.

In this way, the calls are placed and connected through trunks on the PBX instead of to the mobile phone directly. Therefore, the user will not be charged on mobile phone services for long distance or international calls.

To configure callback on the PBX, go to Web GUI→**Advanced Call Features**→**Callback** page and click on . Configuration parameters are listed in the following table.

Scheduled Call

Call scheduler feature allows the user to schedule a wakeup call to a specific extension. The user can choose the time and date of the call, and when the time arrives, a call will be initiated to designated extension(s). When the call is answered, the chosen prompt will be played.

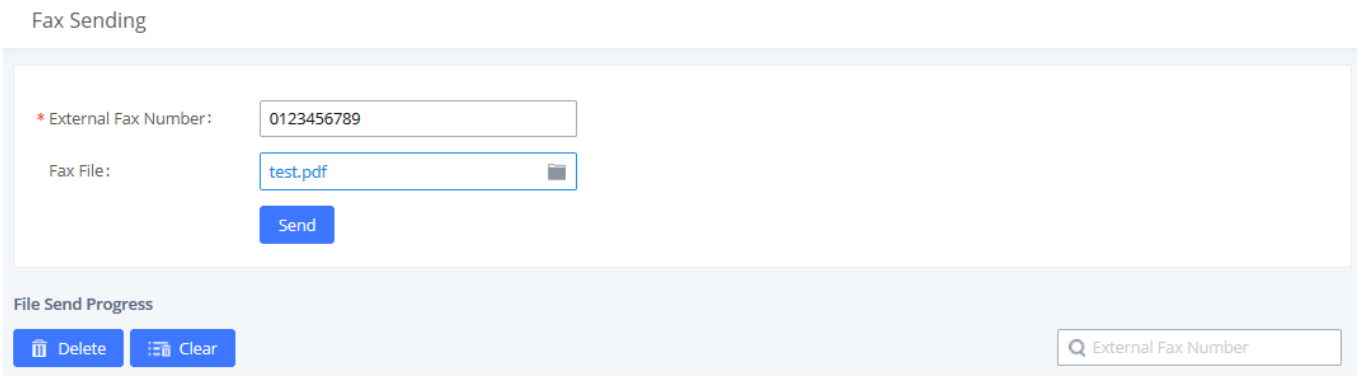


Scheduled Call

Fax Sending

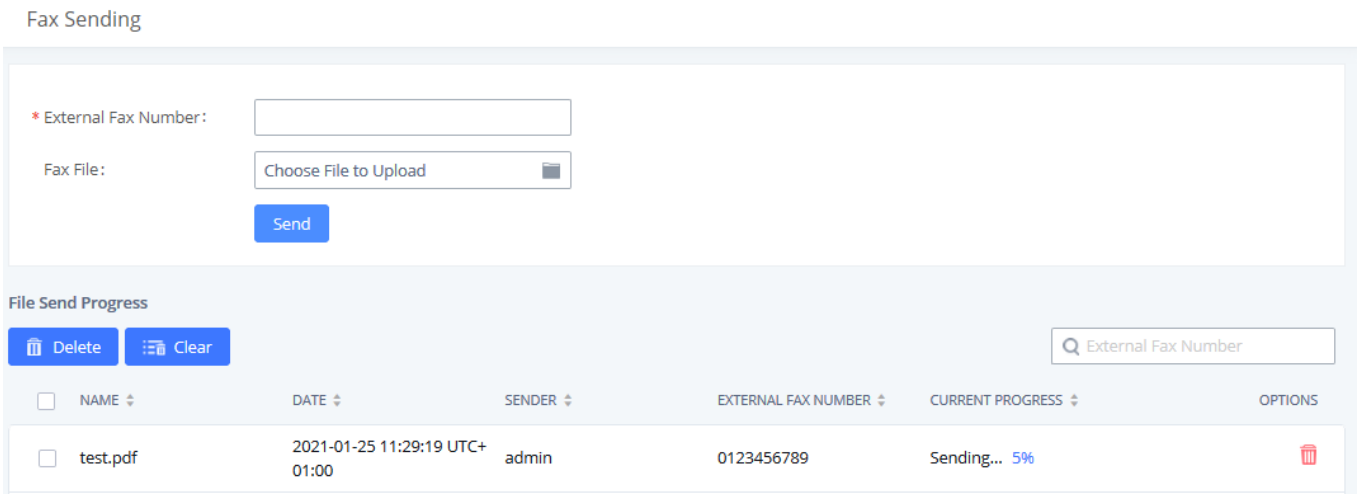
The IPPBX supports sending Fax via Web GUI . This feature can be found on **Advanced Call Features→ Fax Sending** page. The user can enter the number of the destination, then the fax will be sent to a fax gateway on the receiving end.

After entering the fax number, please upload the pdf file that you wish to send as a fax.



Fax Sending in Web GUI

After that you can see the ongoing sending operation on the progress bar.

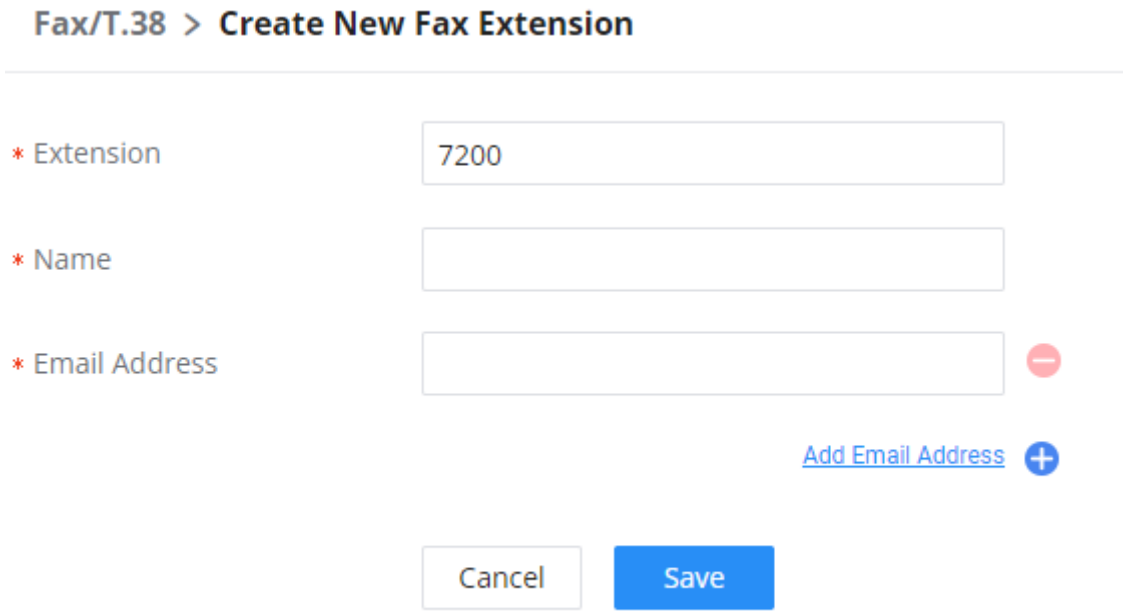


Fax Send Progress

Only A3, A4, and B4 paper sizes are supported for the Fax Sending.

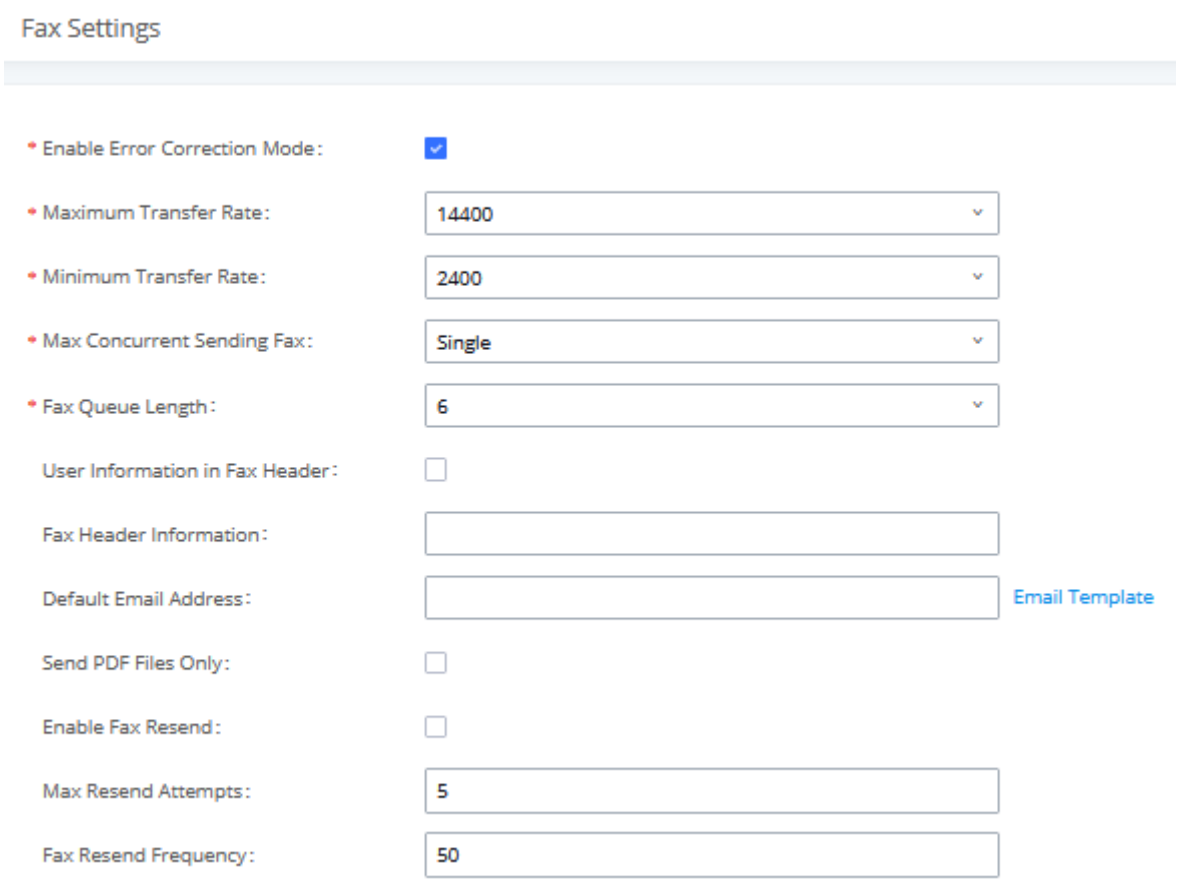
Fax/T.38

The PBX supports T.38 Fax It can convert the received Fax to PDF format and send it to the configured Email address. Fax/T.38 settings can be accessed via **Advanced Call Features→FAX/T.38**. The list of received Fax files will be displayed on the same web page for users to view, retrieve and delete.



Create New Fax Extension

- Click on "Create New Fax Extension". In the popped-up window, fill the extension, name, and Email address to send the received Fax to.
- Click on **"Fax Settings"** to configure the Fax parameters.
- Click on to edit the Fax extension.
- Click on to delete the Fax extension.



Fax Settings

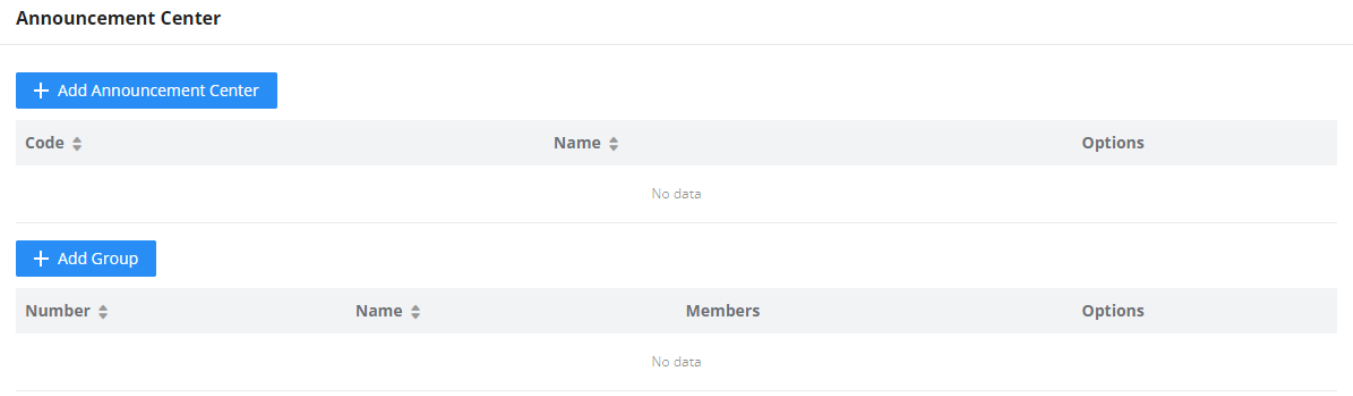
Enable Error Correction Mode	Configure to enable Error Correction Mode (ECM) for the Fax. The default setting is "Yes".
Maximum Transfer Rate	Configure the maximum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000, and 14400. The default setting is 14400.
Minimum Transfer Rate	Configure the minimum transfer rate during the Fax rate negotiation. The possible values are 2400, 4800, 7200, 9600, 12000, and 14000. The default setting is 2400.
Max Concurrent Sending Fax	Configure the concurrent fax that can be sent by PBX. Two modes "Only" and "More" are supported. Only Under this mode, the PBX allows only a single user to send a fax at a time. More Under this mode, the PBX supports multiple concurrent faxes sending by the users. By default, this option is set to "only".
Fax Queue Length	Configure the maximum length of Fax Queue from 6 to 10. The default setting is 6.
User Information in Fax Header	If enabled, this will give users the option to send a special header in SIP fax messages.
Fax Header Information	Adds fax header into the fax file.
Default Email Address	Configure the Email address to send the received Fax to if the user's Email address cannot be found. Note: The extension's Email address or the Fax's default Email address needs to be configured to receive Fax from Email. If neither of them is configured, Fax will not be received from email.
Template Variables	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending the Fax to the users. The template variables are: ${CALLERIDNUM} : Caller ID Number ${CALLERIDNAME} : Caller ID Name ${RECEIVEEXTEN} : The extension to receive the Fax ${FAXPAGES} : Number of pages in the Fax ${VM_DATE} : The date and time when the Fax is received. (Format: MM/dd/yyyy hh:mm:ss)

Send PDF Files Only	If enabled, fax emails will no longer attach TIFF files. Only PDF files will be attached.
Enable Fax Resend	Enables the fax resend option which allows the IPPBX to keep attempting to send faxes up to a specified amount of times. Additionally, if fax still fails to send, a Resend button will appear in the File Send Progress list in Advanced Call Features→Fax Sending to allow manual resending.
Max Resend Attempts	Configures the number of the maximum attempts to resend the fax. The default value is set to 5.
Fax Resend Frequency	Configures the Fax Resend Frequency. The default value is set to 50.

FAX/T.38 Settings

Announcement Center

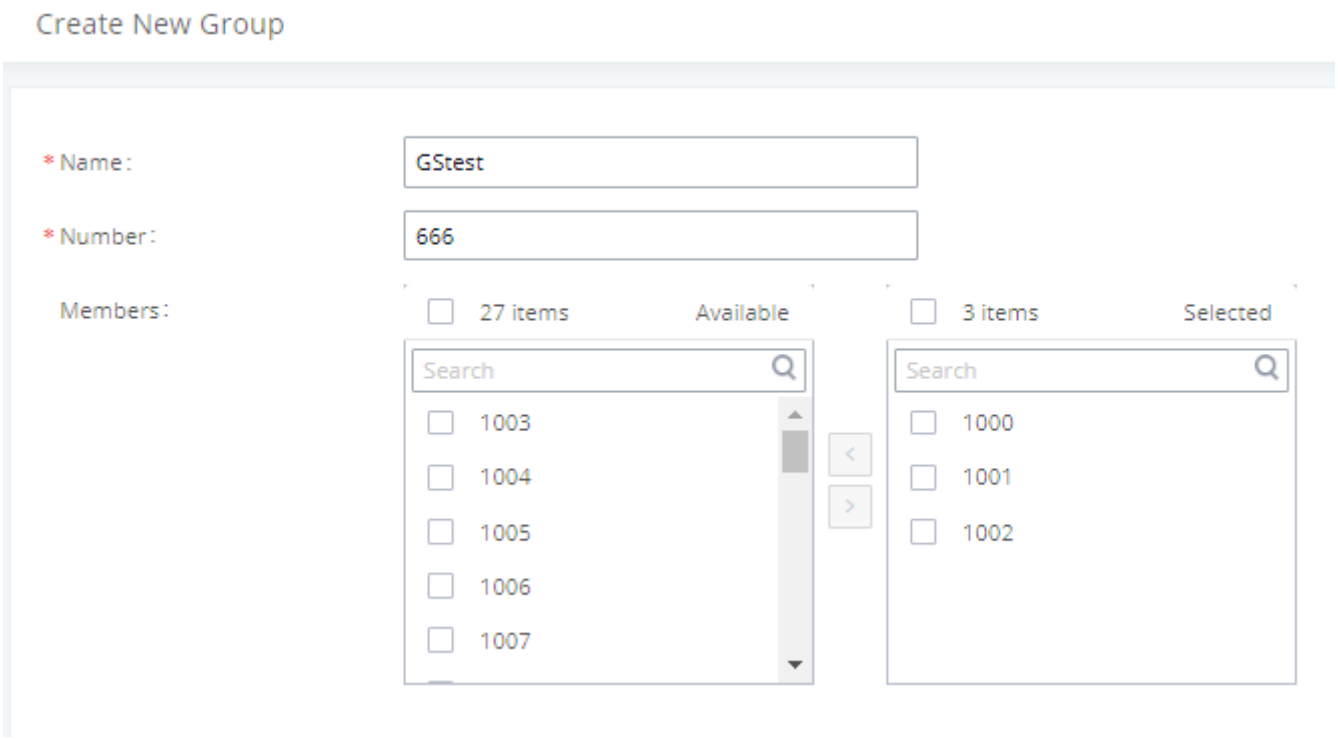
The IPPBX supports Announcement Center feature which allows users to pre-record and store voice message into the IPPBX with a specified code. The users can also create group with specified extensions. When the code and the group number are dialed together in the combination of **code + group number**, the specified voice message is sent to all group members and only extensions in the group will hear the voice message.



Announcement Center

Announcement Center feature can be found under **Advanced Call Features→Announcement Center**. The following example demonstrates the usage of this feature.

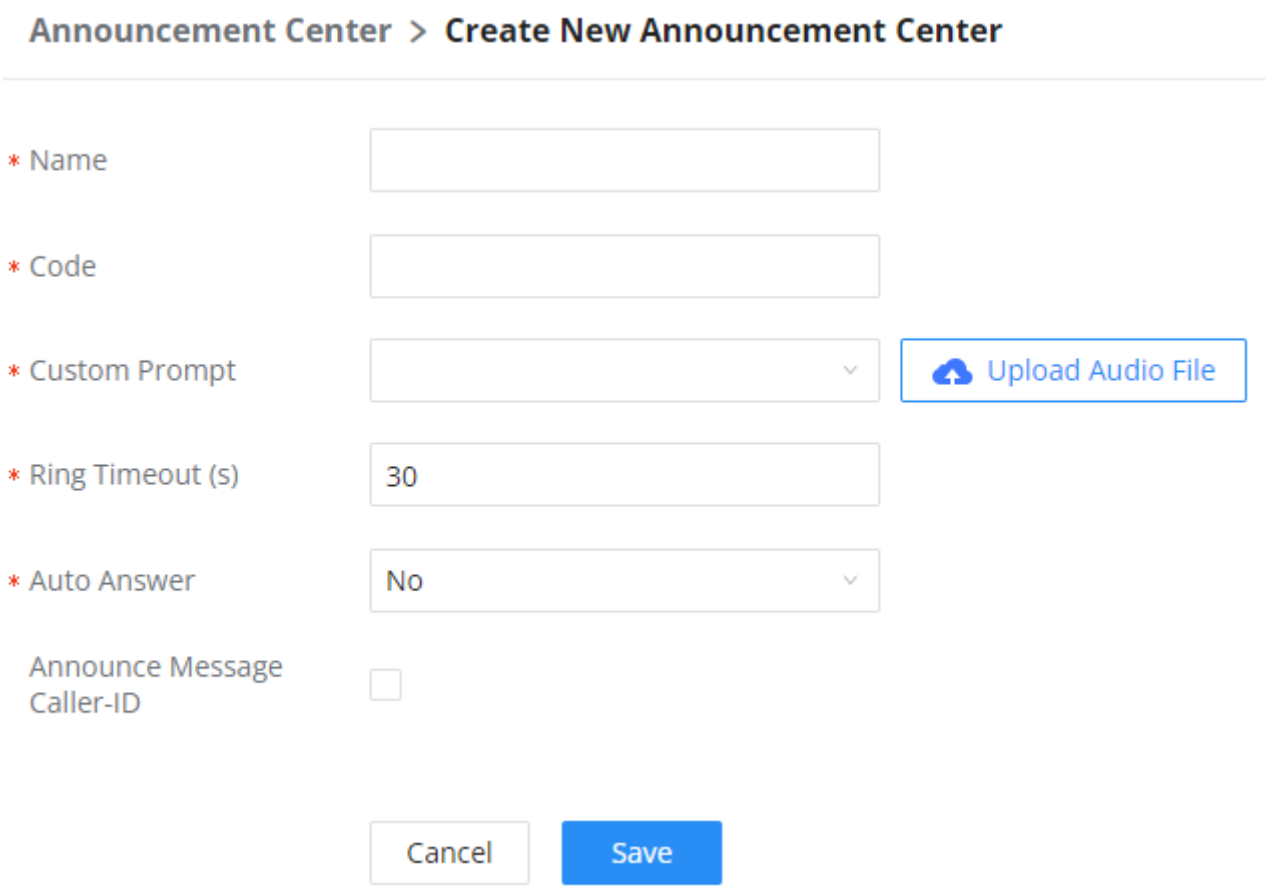
1. Click to add new group.
2. Give a name to the newly created group.
3. Create a group number which is used with code to send voice message.
4. Select the extensions to be included in the group, who will receive the voice message.



Announcement Center Group Configuration

In this example, group "Test" has number 666. Extension 1000, 1001 and 1002 are in this group.

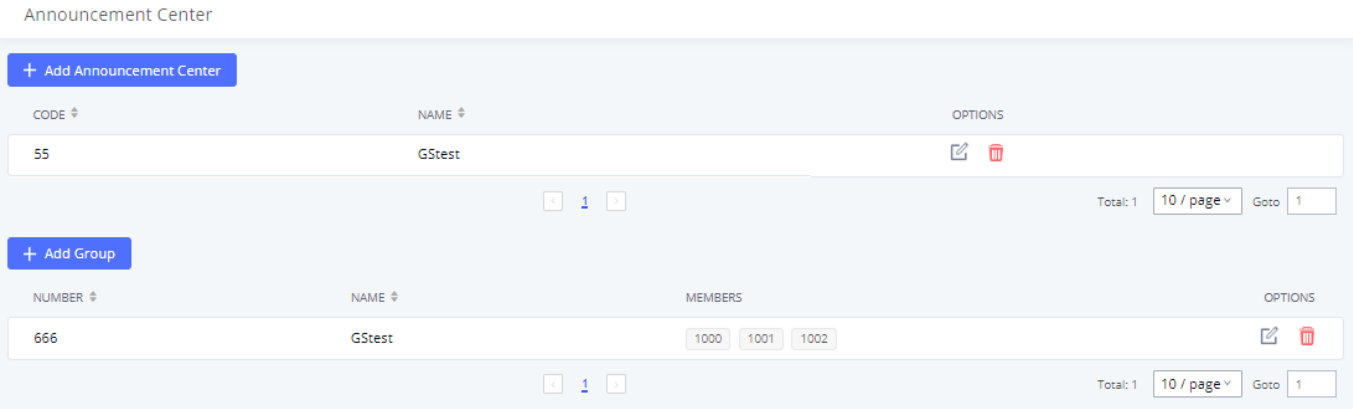
5. Click to create a new Announcement Center.
6. Give a name to the newly created Announcement Center.
7. Specify the code which will be used with group number to send the voice message to.
8. Select the message that will be used by the code from the Custom Prompt drop down menu. To create a new Prompt, please click "Prompt" link and follow the instructions in that page.



Announcement Center Code Configuration

Name	Configure a name for the newly created group to identify the group.Note: Name cannot exceed 64 characters.
Number	Configure the group number. The group number is used in combination with the code. For example, if group number is 666, and code is 55. The user can dial 55666 to send prompt 55 to all members in group 666.Note: The combination number must not conflict with any number in the system such as extension number or conference number and cannot exceed 64 characters.
Internal Members	Choose the local extensions to add to the group.
LDAP Members	Choose the LDAP contacts to add to the group.
Custom Members	Enter the custom phone numbers to add to the group. Note: The maximum number of custom numbers which can be added are 50 custom number.

Code and Group number are used together to direct specified message to the target group. All extensions in the group will receive the message. For example, we can send code 55 to group 666 by dialing 55666 from any extension registered to the PBX. All the members in group 666 which are extension 1000, 1001 and 1002 will receive this voice message after they pick up the call.



Announcement Center Example

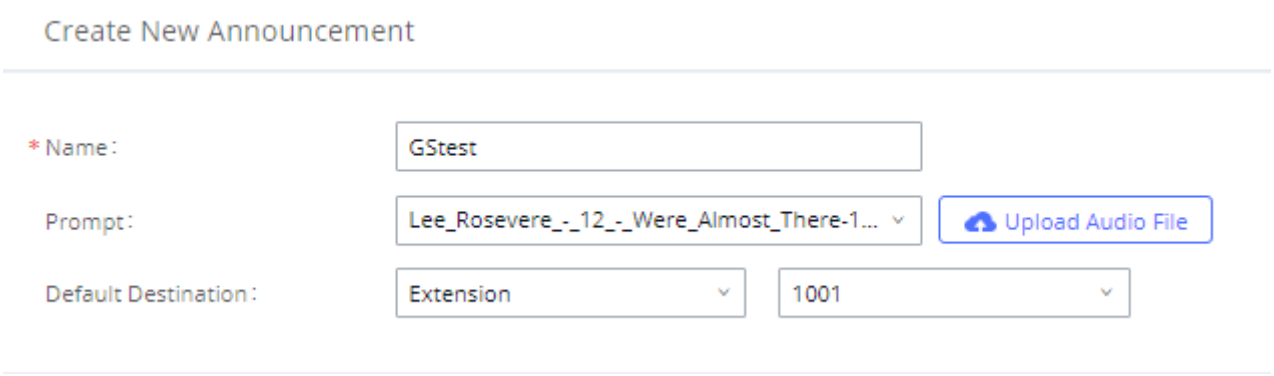
Announcement

The Announcement feature (not to be confused with Announcement Paging and Announcement Center) is a feature that allows users to set an unskippable audio file to play to callers before routing them to a configured destination. Announcements can be configured as a destination in the Inbound Routes page.

To configure Announcement, users need to follow below steps:

1. Navigate on the web GUI under **Basic Call Features → Announcement**
2. Click on A close up of a sign Description generated with high confidence to add a new Announcement.
3. Configure the required fields Name, Prompt, Default Destination to be used for the announcement.

Save and apply the configuration.



Announcement settings

The table below gives more description of the configuration parameters when creating Announcement.

Name	Configure the name of the Announcement.
Prompt	Audio file that needs to be uploaded in order to be played for a specific destination.
Default Destination	Select the destination where to play the audio file.

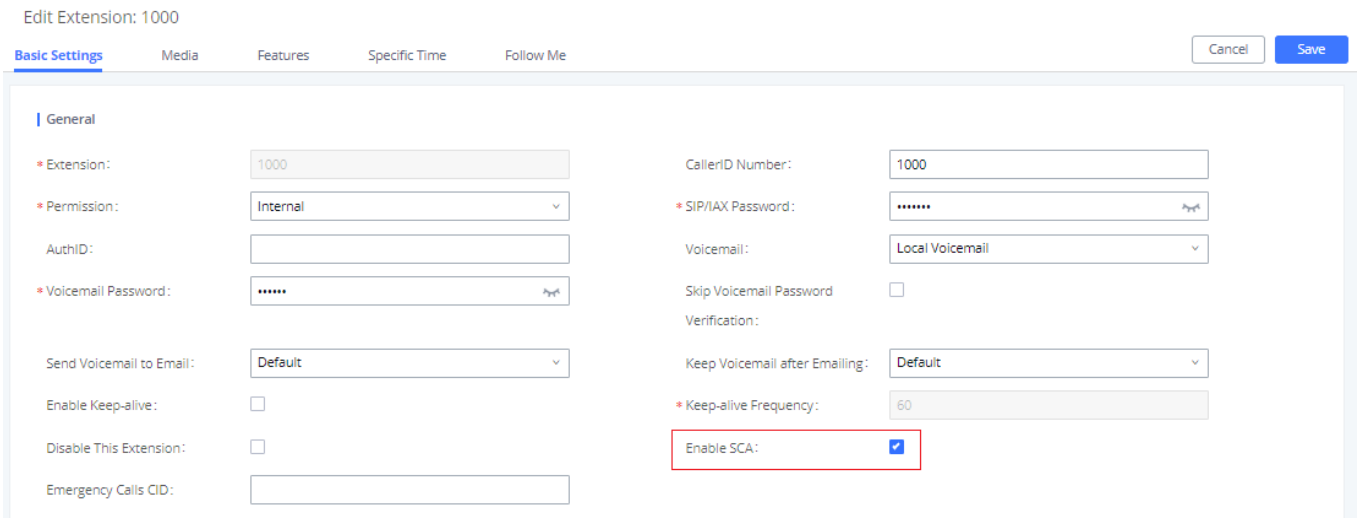
Announcement Parameters

Shared Call Appearance (SCA)

Shared Call Appearance (SCA) functionality has been added to the PBX. With SCA, users can assign multiple devices to one extension, configure endpoints to monitor that extension, make actions on behalf of that extension such as viewing call status and placing and receiving calls, and even barging into existing calls. To configure the SCA functionality, please follow the steps below:

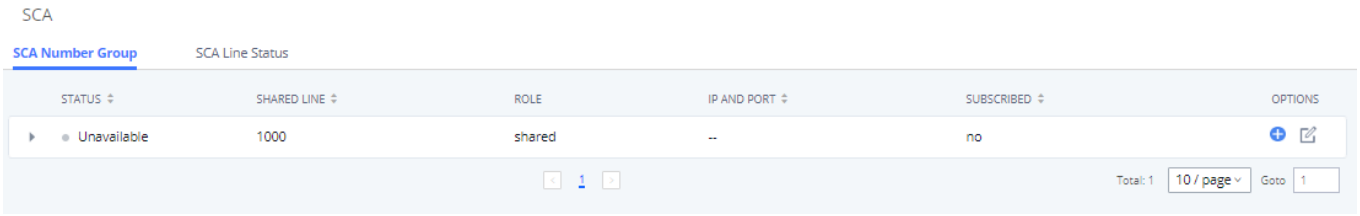
1. Users can enable SCA by navigating to the Extensions page, editing the desired extension, and enabling the option SCA.

With SCA enabled, the Concurrent Registrations field can only have a value of 1.



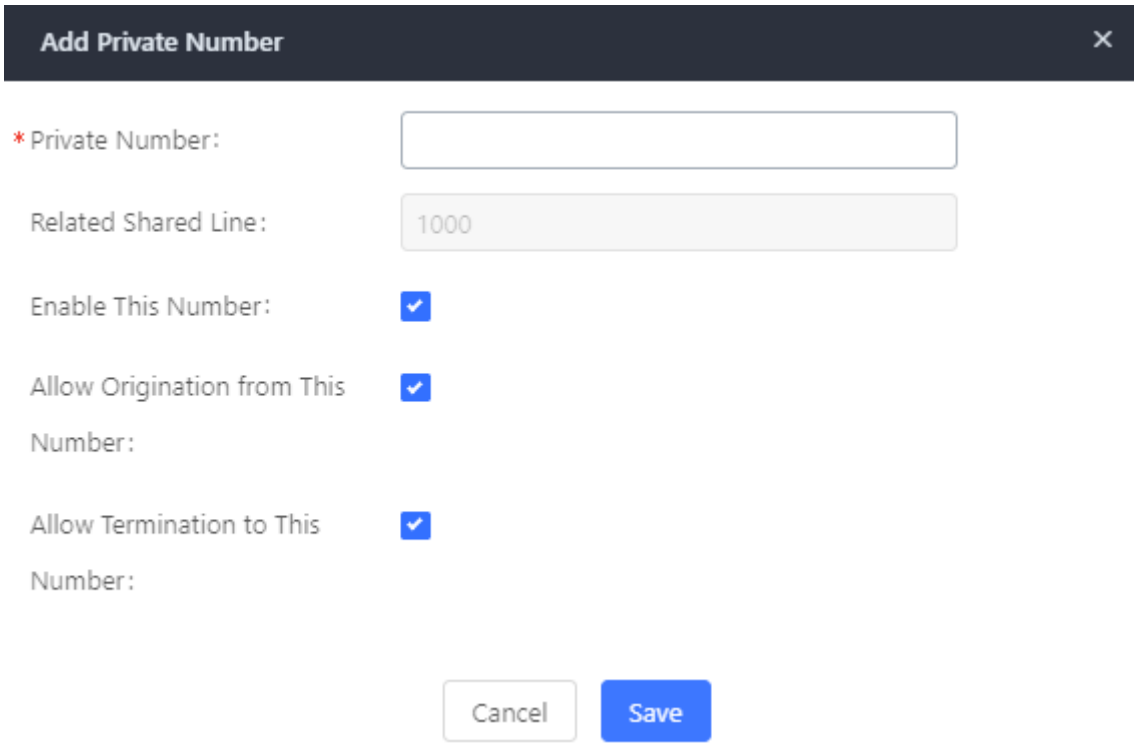
Enabling SCA option under Extensions Settings

2. After enabling the option, navigate to **Advanced Call Features→SCA**. The newly enabled SCA extension will be listed. Click the "+" button under the Options column to add a number that will share the main extension's call appearance, which will be called private numbers.



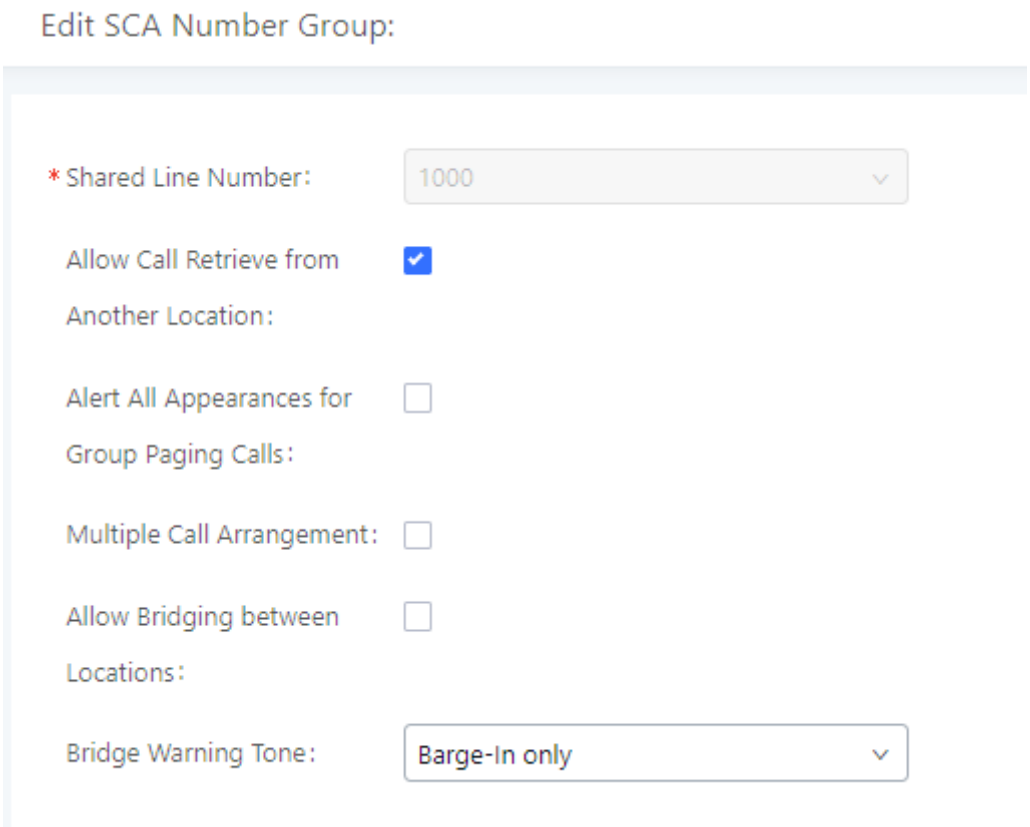
SCA Number Configuration

3. Configure the private number as desired.



SCA Private Number Configuration

4. Once the private number has been created, users must now register a device to it. To properly register a device to the private number, use the configured private number as the SIP User ID. Auth ID and Password will be the same as the main extensions. Once registration is complete, SCA is now configured.



SCA Options

5. Next, configure the VPK or MPK to Shared for both the main extension and the private number. SCA is now configured for both endpoint devices.

The following table describe the SCA Number configuration setting:

Private Number	Configures the private number for the SCA.
Related Shared Line	Display the related shared line.
Enable This Number	Whether enable this private number. If not enabled, this private number is only record in DB, it will not affect other system feature.
Allow Origination from This Number	Enable this option will allow calling from this private number. By default, it is enabled.
Allow Termination to This Number	Enable this option will allows calls to this private number. By default, it is enabled.

Add SCA Private Number

The following table describes the options available when editing the SCA number:

Shared Line Number	While SCA is enabled, this number will be the same as the extension number.
Allow Call Retrieve from Another Location	Allows remote call retrieval. Must be enabled in public hold. By default, it is enabled.
Alert All Appearances for Group Paging Calls	Allows all SCA group members to ring when the SCA shared number is paged. If disabled, only the SCA shared number will ring when paged. By default, it is disabled.
Multiple Call Arrangement	Allows simultaneous calls in an SCA group. By default, it is disabled.
Allow Bridging between Locations	Allows location bridging for SCA group. Must be enabled when using the Barge-In feature. By default, it is disabled.
Bridge Warning Tone	Configures the notification in the bridge when another party join. None: No notification sound. Barge-In only: Notification sound will play when another party join. Barge-In and Repeat: Notification sound will play when another party joins and repeat every 30 seconds. By default, it is set to "Barge-In Only".

Editing the SCA Number

Emergency Calls

Emergency Calls

IPPBX supports configuration and management of numbers to be called in emergency situation, thus bypassing the regular outbound call routing process and allowing users in critical situation to dial out for emergency help with the possibility to have redundant trunks as point of exit in case one of the lines is down.

The PBX is fully compliant with Kari's Law and Ray Baum's Act, for more information, please refer to the following links:

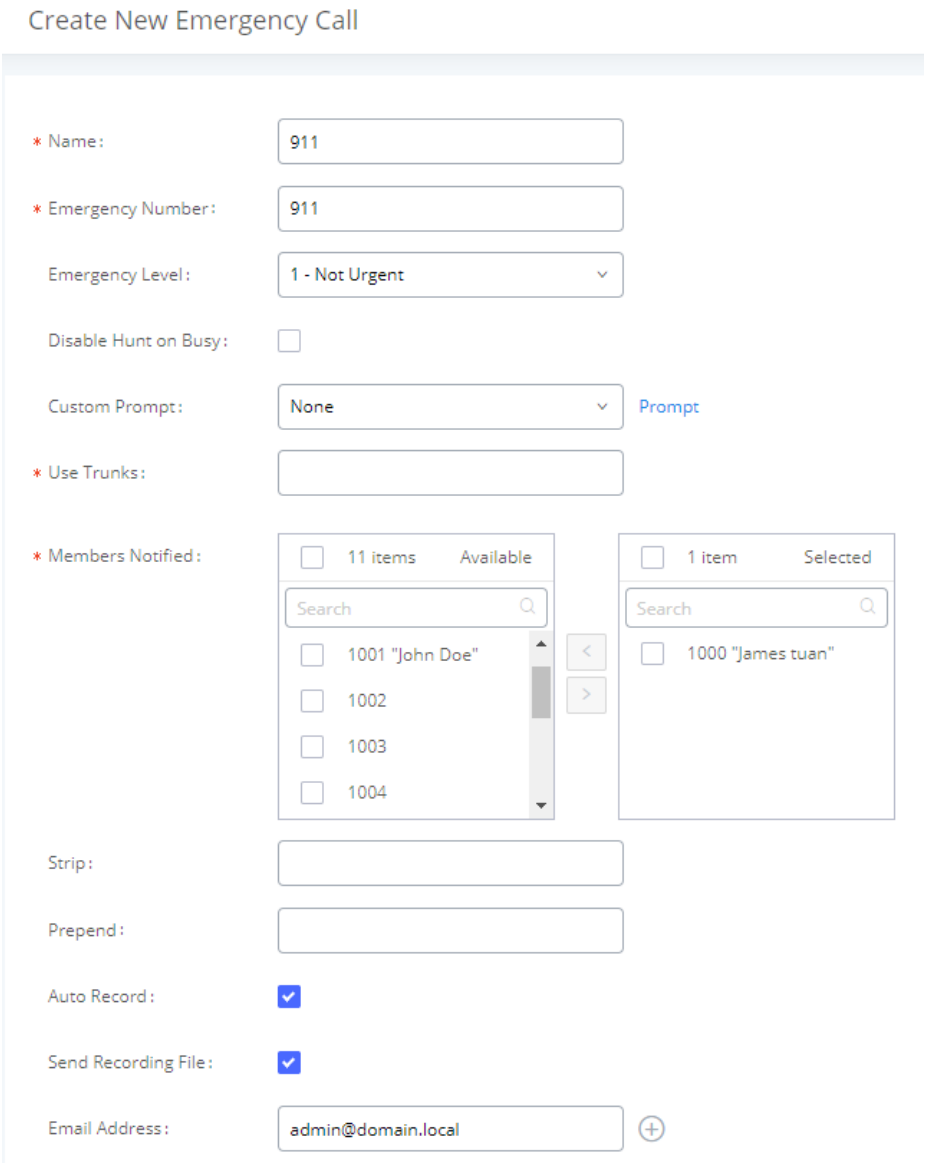
https://www.fcc.gov/general/9-1-1-and-e9-1-1-services

In addition, Emergency calls can be automatically recorded by toggling on the new Auto Record and recordings can be viewed in the new Emergency Recordings tab on the same page. Additionally, users can have these recordings be sent to the configured email address(es).

Email alerts are also supported after enabling the notification for the event under "**Maintenance → System Events**"

To configure emergency numbers, users need to follow below steps:

1. Navigate on the web GUI under "**Advanced Call Features → Emergency Calls**"
2. Click on A close up of a sign Description generated with high confidence to add a new emergency number.
3. Configure the required fields "Name, Emergency Number and Trunk(s) to be used to reach the number".
4. Save and apply the configuration.

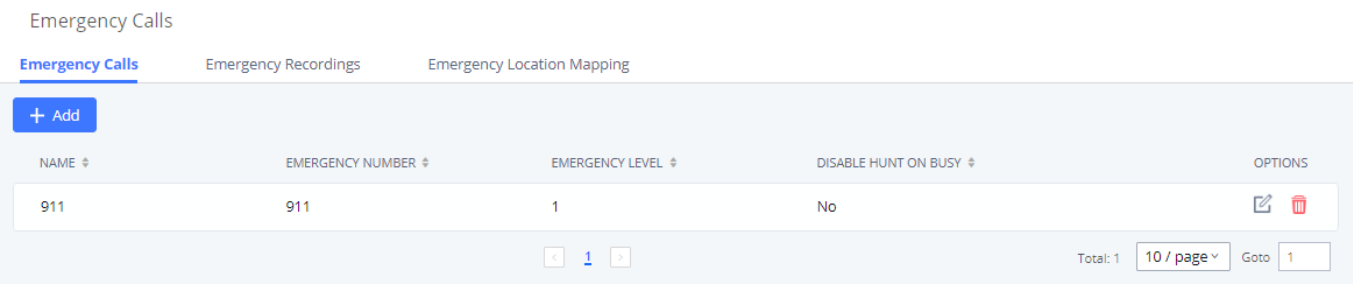


Emergency Number Configuration

The table below gives more description of the configuration Parameters when creating emergency numbers.

Name	Configure the name of the emergency call.For example, "emergency911","emergency211" and etc.
Emergency Number	Config the emergency service number. For example,"911","211" and etc.
Emergency Level	Select the emergency level of the number. Level "3" means the most urgent.
Disable Hunt on Busy	If this option is not enabled, when the lines of trunks which the coming emergency call routes by are completely occupied, the line-grabbing function will automatically cut off a line from all busy lines so that the coming emergency call can seize it for dialing out.Note: This option is not enabled by default.
Custom Prompt	This option sets a custom prompt to be used as an announcement to the person receiving an emergency call. The file can be uploaded from the page "Custom Prompt". Click "Prompt" to add an additional record.

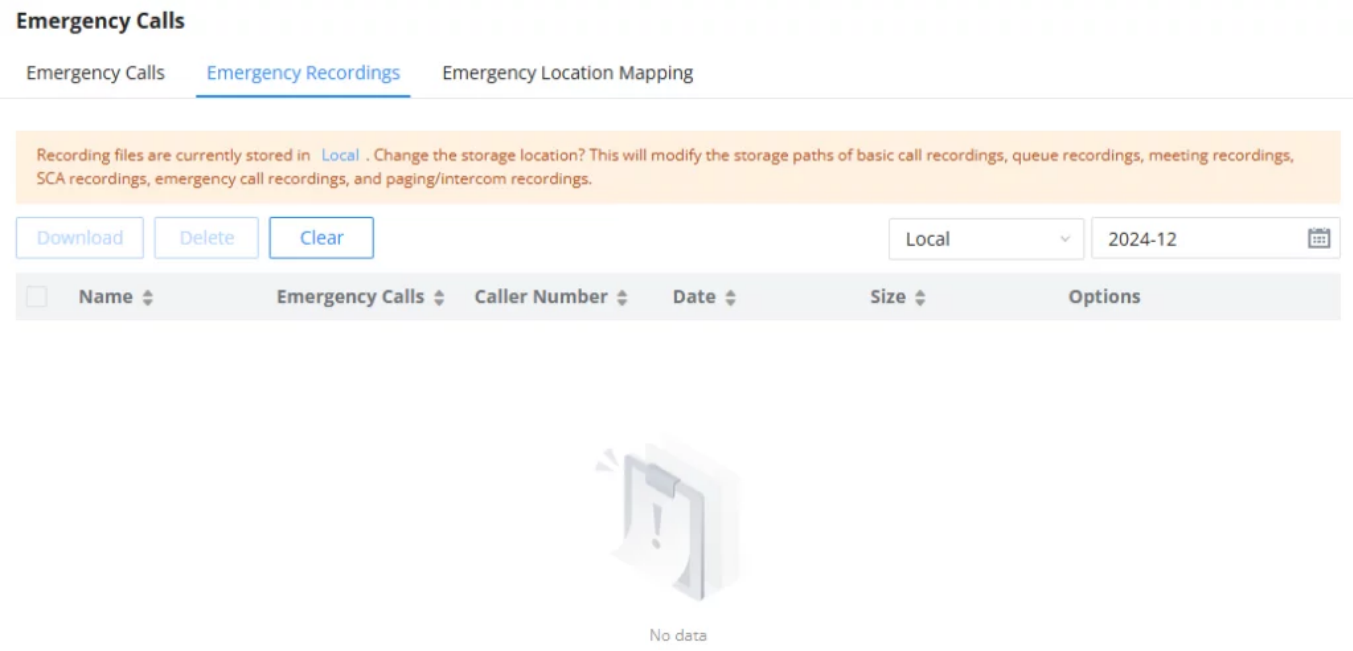
Use Trunks	Select the trunks for the emergency call. Select one trunk at least and select five trunks at most.
Members Notified	Select the members who will be notified when an emergency call occurs.
Strip	Specify the number of digits that will be stripped from the beginning of the dialed number before the call is placed via the selected trunk.
Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Auto Record	If enabled, the calls using this extension or trunk will be automatically recorded, default is disabled Note: Emergency recording will be also available under Emergency Recording Tab.
Send Recording File	If Enabled the recordings will be sent to the configured Email address(es)
Email Address	Used to specify the Email address(es) for Emergency calls recordings.



911 Emergency Sample

Emergency Recordings

The IPPBX allows recording emergency calls and they can be found under WebUI -> **Advanced Call Feature -> Emergency Calls -> Emergency Recordings**

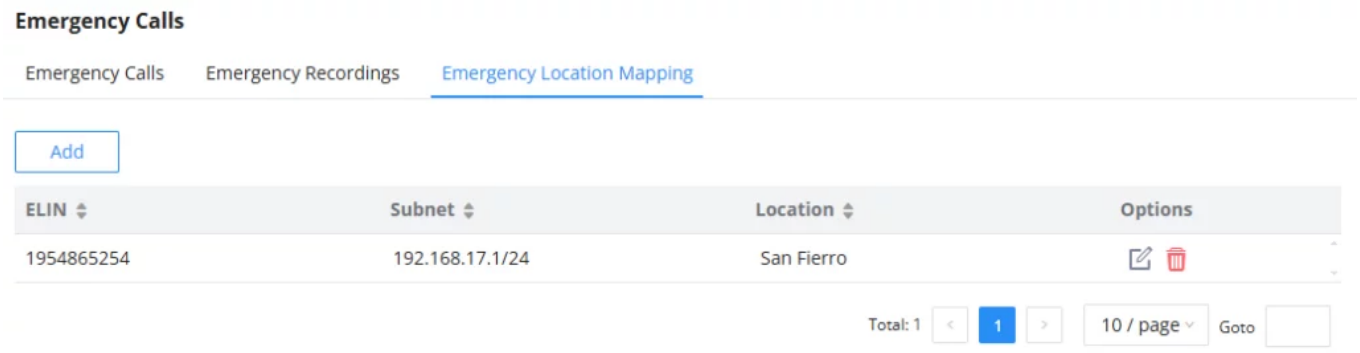


Emergency Recordings

Emergency Location Mapping

In compliance with Kari's Law and the Ray Baum's Act, IPPBX's Emergency Calls feature supports emergency location mapping. This will allow users to associate subnets with emergency location

identification numbers (ELINs), which can then be used by E911 service providers for example to determine the exact location of callers. The new options can be found under **Advanced Call Features->Emergency Calls->Emergency Location Mapping**.



Emergency Location Mapping

- **ELIN:** The emergency location identification number registered with the E911 provider. This number will be sent out as the emergency call's CID number.
- **Subnet:** The network subnet that the ELIN will be associated with. The ELIN that is sent to E911 providers is based on the subnet that a calling endpoint is registered from. Example: "xxx.xxx.xxx.xxx/24" or "xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx:xxxx/64".
- **Location:** Location associated with the configured subnet. This is used for the IPPBX administrator's reference.
- **Geolocation Routing:** Toggles whether to include the Geolocation header in the emergency call SIP INVITE message. The Location field value will be used as the Geolocation header value.

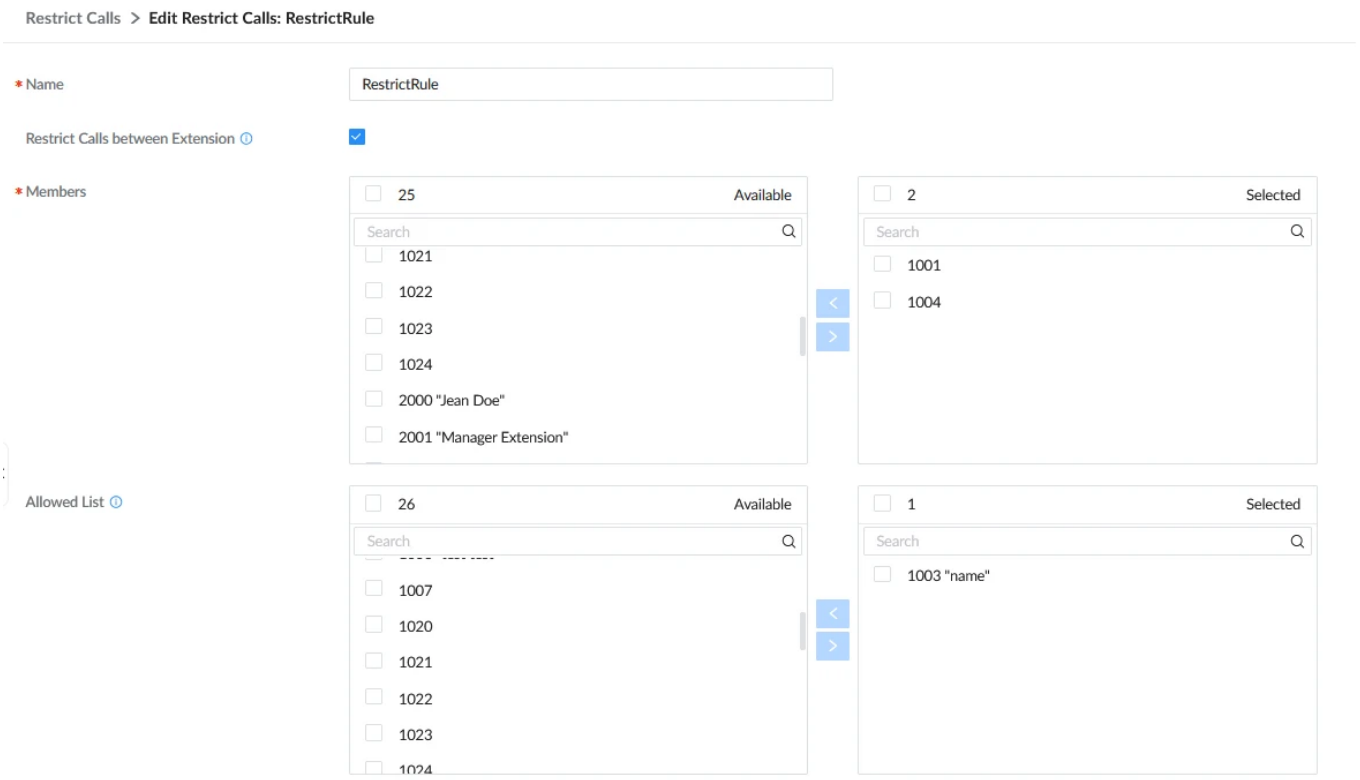
Please note that ELIN Mapping is supported only on peer trunks. It would not apply on register trunks.

Restrict Calls

The **Restrict Calls** feature allows administrators to control outgoing call permissions for specific internal extensions. When a rule is enabled, users designated as **Members** are restricted to placing outbound calls only to the extensions listed in the **Allowed List**.

This section describes the configuration of this feature in the **Advanced Call Features -> Restrict Calls** page.

This feature only restricts outbound calls placed by assigned Members. Non-allowed extensions can still place incoming calls to these Members. For strict two-way restriction between two groups, you must create two separate Restrict Calls rules.



Restrict Calls

Configure Restrict Calls

- Click **Add** to create a new Restrict Calls rule.
- Click the **Edit** icon to modify an existing rule.
- Click the **Delete** icon to remove a rule.

When creating or editing a rule, users can configure the following:

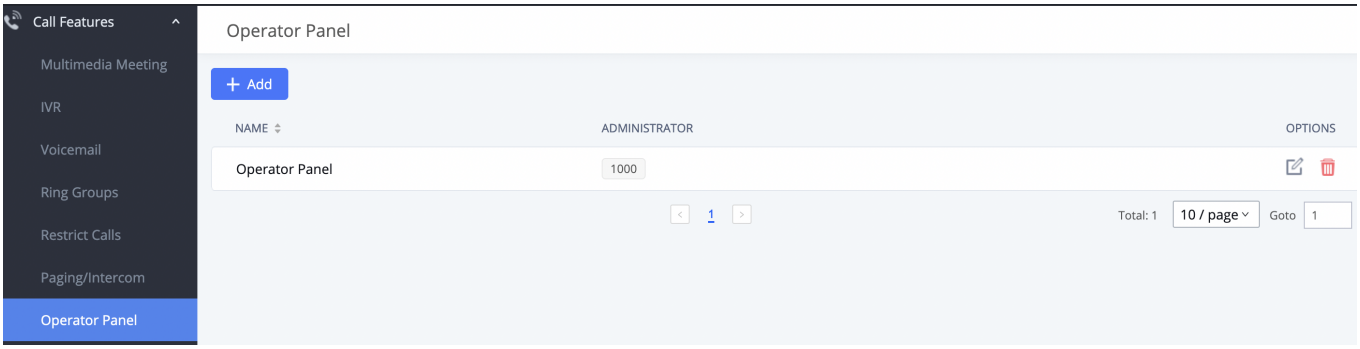
Name	Enter a descriptive name for the Restrict Calls rule.
Restrict Calls Between Extension	Toggles the restriction rule on or off.Enabled: Applies the rule. Members can only place outbound calls to numbers in the Allowed List.Disabled: Bypasses the rule entirely. Members will have no outbound call restrictions. (Enabled by default)
Members	Select the internal extensions whose outbound calling ability will be restricted when the rule is enabled.
Allowed List	Select the specific internal extensions that the Members are permitted to call.

Operator Panel

Configure Operator Panel

Operator Panel settings can be accessed via **Advanced Call Features→Operator Panel**.

The PBX supports the operator panel so that IPPBX extension can be used as admin to manage calls and activities such as extension status, call queue status, transfer, barge-in, hangup, etc. On Grandstream Wave client, it can display the extensions, ring group, voicemail, call queue, call park status under the management of the extension. This section describes how to configure the operator panel.

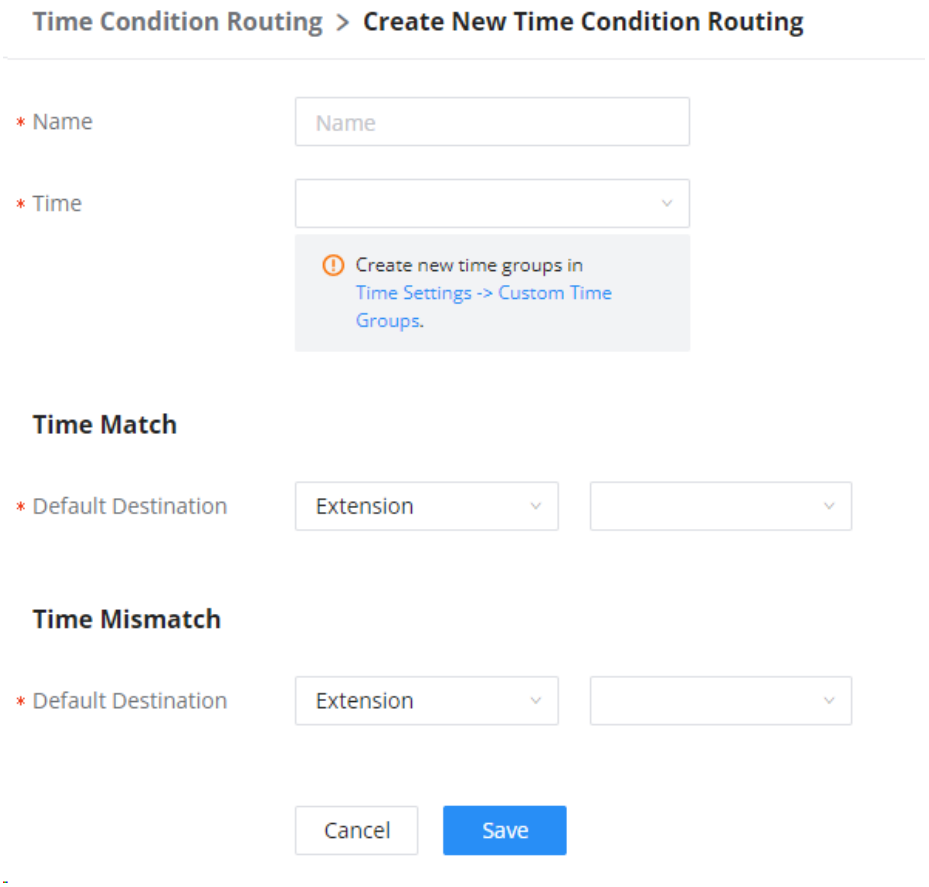


Operator Panel Configuration Page

- Click on "Add" to create the operator panel.
- Click on to edit the operator panel.
- Click on to delete the operator panel.

Time Condition Routing

Time Condition Routing allows the user to create default destinations for call queues and ring groups which are based on time conditions under **Advanced Call Features->Time Condition Routing.**



Time Condition Routing

Parameter	Description
Name	Enter a unique and easily identifiable name for this routing.
Time	Configures the time period to use for this time-based routing.
Time Match	
Default Destination	Calls received during the selected time period will be routed to this destination.

Time Mismatch	
Default Destination	Calls received outside the selected time period will be routed to this destination.

MESSAGING

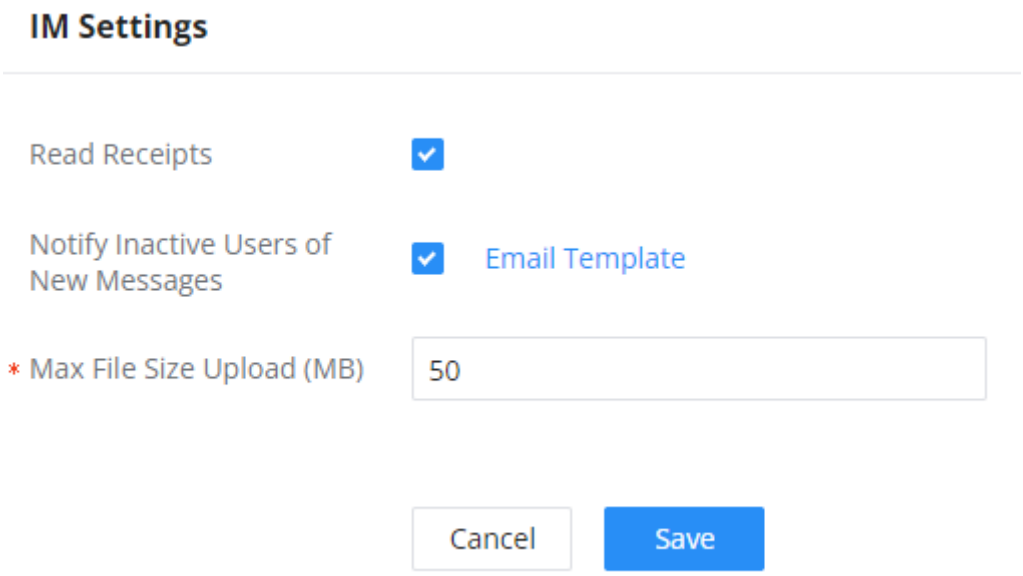
In this category, the user can configure settings related to the messaging features offered by the PBX. This includes instant messaging feature offered by the PBX, as well as Live Chat, and Message Broadcast.

IM Settings

In this section, the user can configure settings related to instant messaging feature. The user can either options related to the local instant messaging service or the cloud instant messaging service.

IM Settings

In IM Settings tab, the user can choose to enable or disable read receipts when exchanging messaging using Wave.

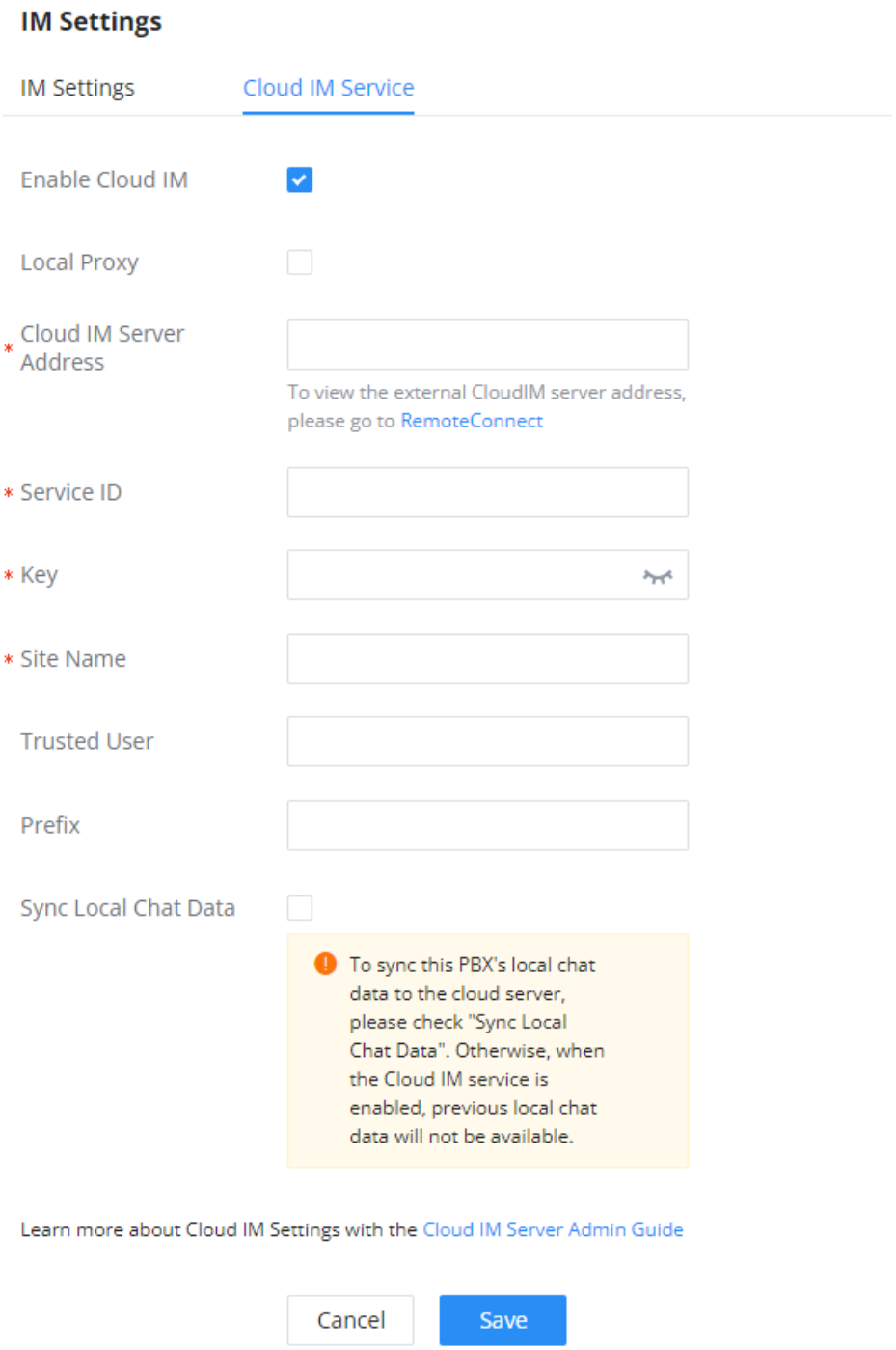


IM Settings

Parameter	Description
Read Receipts	Toggle on/off r the read receipts in the instant messages exchanged in Wave. When read receipts are enabled, Wave will show when a message is delivered and read by the recipient.
Notify Inactive Users of New Messages	If an extension under this server has not logged into Wave for more than 7 days, and a new message is received, an email notification will be sent to the extension notifying them of the amount of direct messages and @mentions received in the last 24 hours. Applies to both local IM and Cloud IM.
Max File Size Upload (MB)	Configures the maximum size of individual files users can upload to Wave chat. Only applicable when using local IM. If using Cloud IM or a custom IM server, the maximum file size upload must be configured on GDMS or the IM server instead. On GDMS, this can be found by navigating to the top right of the page -> Plans & Services -> My Plans -> Options column -> Edit Cloud IM page.

Cloud IM Service

After enabling **Cloud IM**, it means that all IM data in Wave is stored in the external server Cloud IM, and is no longer stored locally in the PBX. GDMS can configure the External Cloud IM service for PBX devices. At this time, the PBX device synchronizes the configuration item information.



Cloud IM Service

Cloud IM Service	
Enable Cloud IM	If you have purchased the Cloud IM package or purchased the Grandstream IM server, you can configure it. If you have not purchased it, the configuration will not take effect, but PBX local IM service is allowed. Please note that after enabling this feature, local chat data will not be visible.
Local Proxy	If enabled, the local proxy will be used to forward files and text messages if the IM server cannot be connected to upon Wave login due to certificate issues.

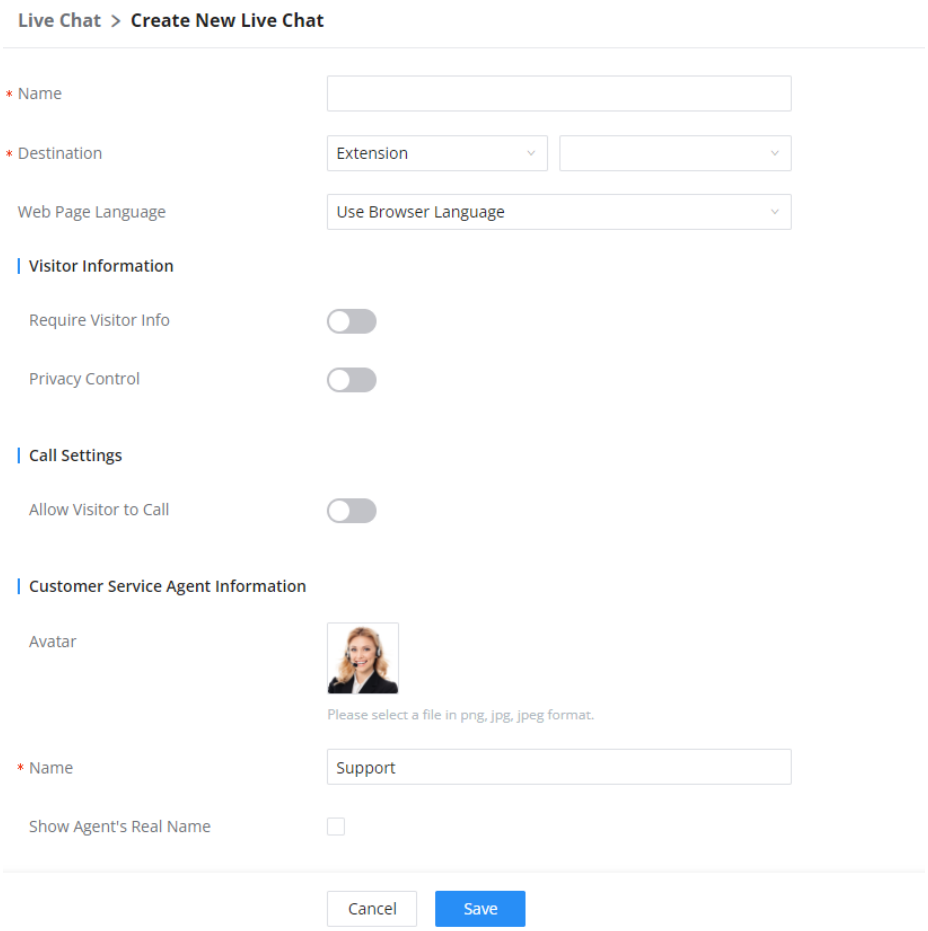
Cloud IM Server Address	The address of the server that provides IM service, you can fill in the address of the Cloud IM server provided by the RemoteConnect package or the IM server address of the GDMS.
Service ID	The service ID of the Cloud IM server.
Key	The Key to the Cloud IM server.
Site Name	Enter the name of the site.
Trusted User	The trusted user of the cloud IM. Only letters, numbers, and special characters are allowed.
Prefix	As the extension prefix, it is added before the extension number.
Sync Local Chat Data	Syncing existing local chat data to Cloud IM server. The Wave chat feature will not be available during the syncing process. It is recommended to avoid syncing during active working hours.- Time Range All Last 12 Months Last 6 Months Last 3 Months Last Month - Data Type IM Data Images Files

Please note that synchronization of the local chat can only occur in the initial connection to a Cloud IM Server . If the PBX is already connected to a Cloud IM server, or the Cloud IM server has already been synced to other PBXs, local chat data will not be able to be synced.

Only account details and department information will be synced on local IM and cloud IM. Other configurations such as profile picture, work status, and favorite contacts will not be synced and these are stored in local IM or cloud IM respectively. Therefore, please be aware that when switching between local IM and cloud IM, part of the data cannot be synced and the previously stored data on local IM or cloud IM (depending on which one is switched to) will be retrieved.

Live Chat

Live Chat feature allows to create chat channels that can be embedded on your website to enable your client to reach your customer service more easily. The client can contact your agents through text chat, and the agent can initiate a call with the client.



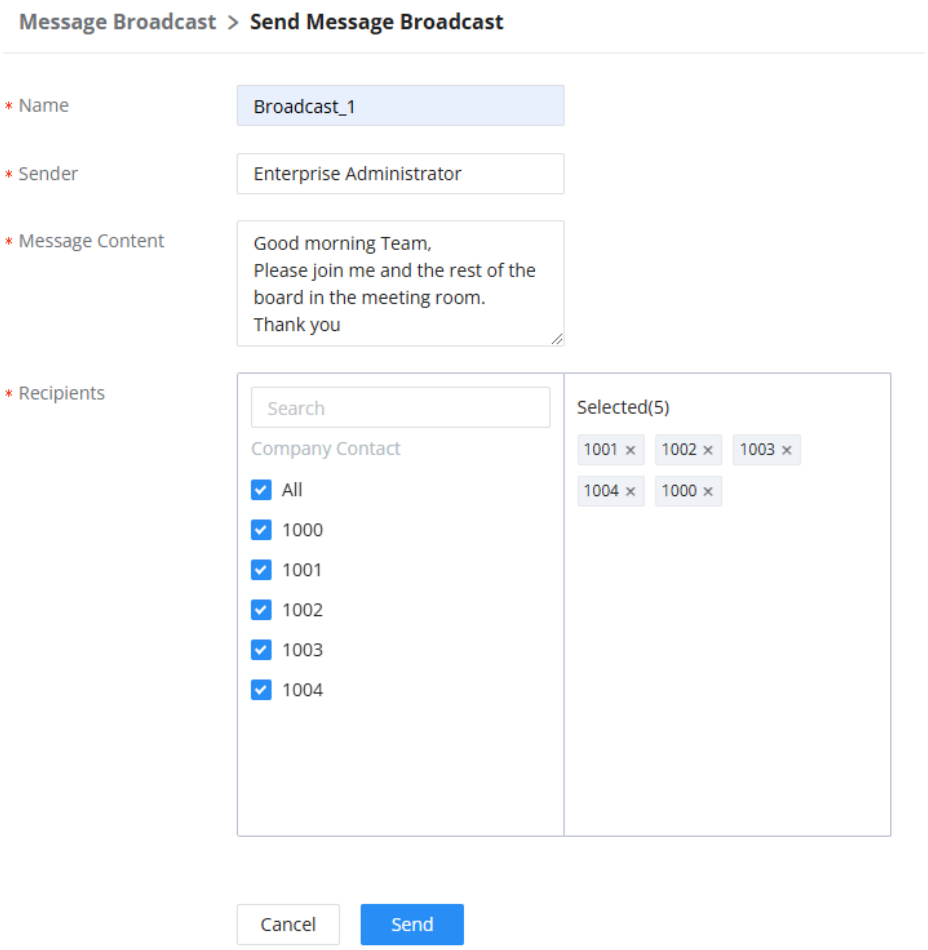
Create New Live Chat

Name	Enter the name of the Live Chat
Destination	Configure the destination of this Live Chat.
Web Page Language	Configure the language of visitor page.
Visitor Information	
Require Visitor Info	Configure the visitor information required to start a live chat session.
Privacy Control	If enabled, visitors must consent to allow the processing of personal data and cookies before entering Live Chat.
Call Settings	
Allow Visitor To Call	If enabled, visitors will have the option to call the configured destination from this Live Chat.
Customer Service Agent Information	
Avatar	Upload the avatar of the agent. Please select a png, jpg, or jpeg format file.
Name	Enter the name of the agent.
Show Agent's Real Name	Enable this option to show the agent's real name in the live chat.
Chat Settings	
Welcome Message	Configure the initial message to display to visitors when they first enter the live chat session.
Reply to First Message	Configure the message to send to visitors in response to their first message.
Visitor Chat Log Retention Time (days)	This value determines how long a visitor's chat history will be kept before it's deleted automatically. Note:

Live Chat Link Address	This link can be embedded onto web pages. Clicking it will connect visitors to the configured Live Chat destination. This link can also be entered directly into the browser address bar for testing purposes.

Message Broadcast

Message broadcast feature allows the administrator to broadcast a text message to all the endpoints selected by the administrator. The administrator can select departments or individual extensions to boardcast a text message.

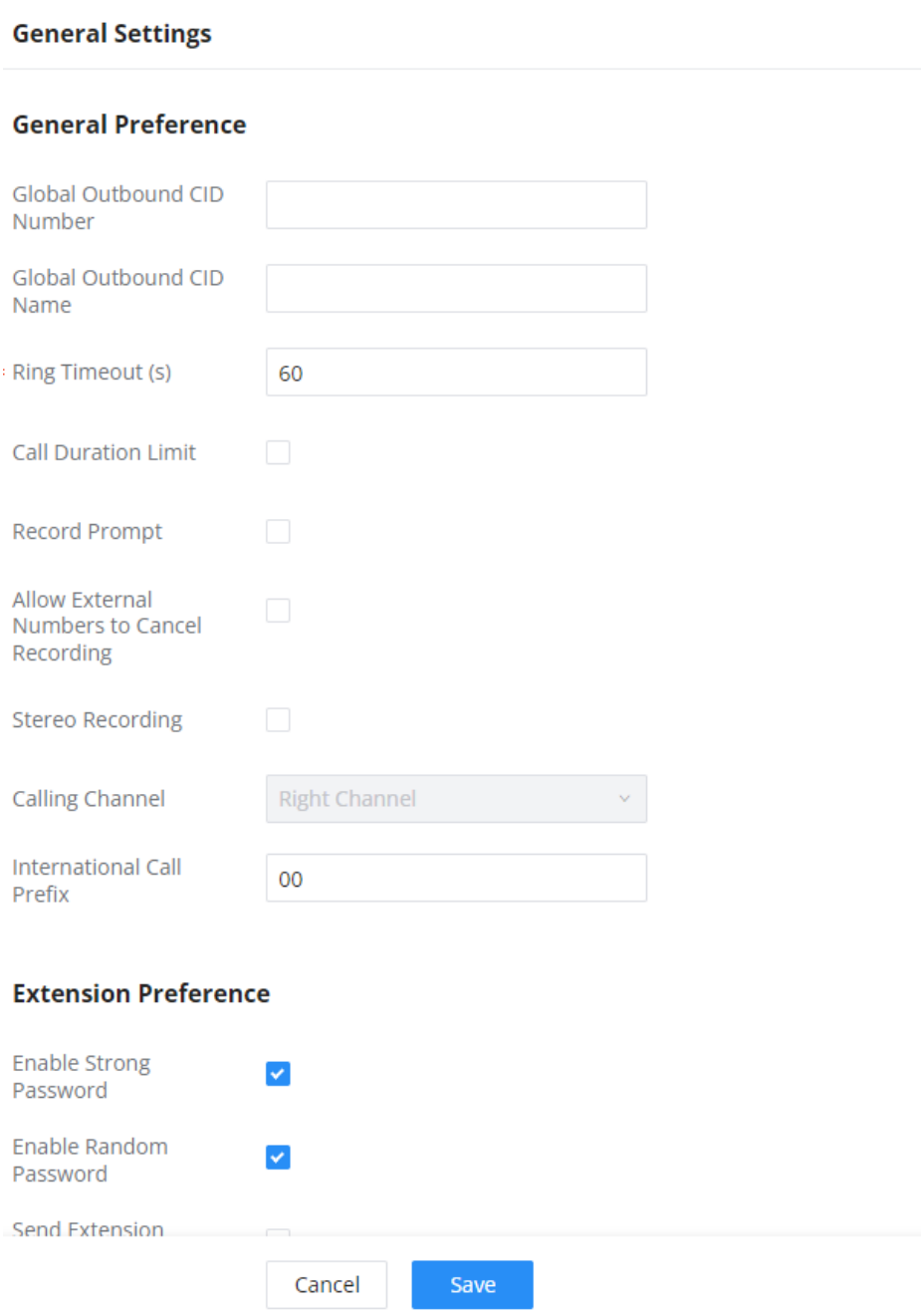


Message Broadcast

PBX SETTINGS

In this category, the user can find the settings related to the PBX operations, such as, extension range settings, SIP settings, RTP settings, Music on Hold settings

General Settings



General PBX Settings

General Preferences	
Global Outbound CID	Global-level CID Number used for all outgoing calls if no other CID numbers have been configured for the calling party.
Global Outbound CID Name	Global-level CID Name used for all outgoing calls if no other CID names have been configured for the calling party.
Ring Timeout (s)	Number of seconds to ring the extension before forwarding the call to voicemail. If Extensions > Features > Ring Timeout is not configured, the ring timeout value configured in General Settings > Ring Timeout will be used.
Call Duration Limit	Block calls for the configured duration. If Extensions > Features > Call Duration Limit and Outbound Routes > Call Duration Limit are not configured, General Settings > Call Duration Limit will be used.Note: This setting is disabled by default
Recording Settings	
Record Prompt	If enabled, the system will play voice prompt "This call will be recorded" before the recording is started.Note: This setting is disabled by default

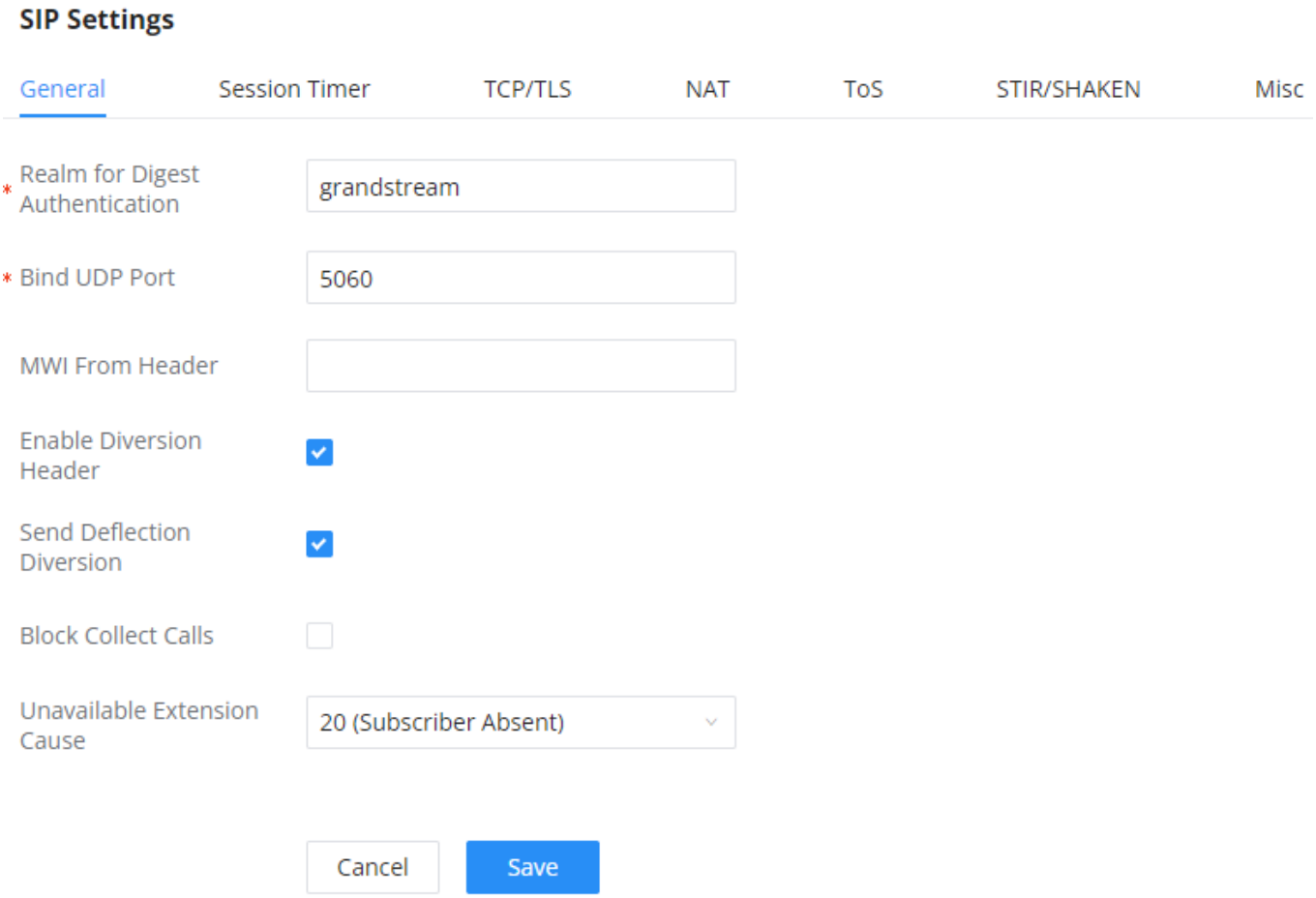
Allow External Numbers to Cancel Recording	If enabled, external call parties will be given the option to decline the recording of calls.
Merge Same Call Recordings	If enabled, the system will merge all recordings created during a call regardless of how many times a user starts and stops recording during a call.
Stereo Recording	If enabled, the caller and callee's audio will be split into two channels during call recording. Not applicable to calls with more than 2 parties.
Calling Channel	Configure the audio channels for the calling party and the called party. If the caller is selected as the right channel, the callee will be used for the left channel, and vice-versa. Note: This option is available when "Stereo Recording" is enabled.
International Call Prefix	When this configuration is empty, International Call Prefix can be empty or +.
Extension Preferences	
Enforce Strong Password	If enabled, a strong password policy will be enforced. This does not affect user login passwords, which must be strong.
Enable Random Password	If enabled, the extension will created with a randomly generated password.
Send Extension Update Emails	If enabled, an email will be sent to an extension's configured email address after creating it or modifying that extension's settings.
Disable Extension Range	If set to "Yes", users could disable the extension range pre-configured/configured on the IPPBX. The default setting is "No". Note: It is recommended to keep the system assignment to avoid inappropriate usage and unnecessary issues.
User Extensions	1000-6299 User Extensions is referring to the extensions created under Web GUI > Extension/Trunk > Extensions page.
Meeting Extensions	6300-6399 This extension range is used for creating meeting rooms
Ring Group Extensions	6400-6499 This extension range is used for ring groups
Queue Extensions	6500-6599 This range of extensions is used for queueing
Voicemail Group Extension	6600-6699 This extension range is used for voicemail groups.
IVR Extensions	7000-7100 This extension range is used for
Dial by Name Extensions	7101-7199 This extension range is used for Dial by Name feature
Fax Extensions	7200-8200 This extension range is used for T.38 Fax
Zero Config Extension	
Pick Extensions	4000-4999 This refers to the extensions that can be manually picked from end device when being provisioned by the IPPBX. There are two related options in zero config page?Zero Config Settings, "Pick Extension Segment" and "Enable Pick Extension". If "Enable Pick Extension" under zero config settings is selected, the extension list defined in "Pick Extension Segment" will be sent out to the device after receiving the device's request. This "Pick Extension Segment" should be a subset of the "Pick Extensions" range here. This feature is for the GXP series phones that support selecting extension to be provisioned via phone's LCD.
Auto Provision Extensions	5000-6299 This sets the range for "Zero Config Extension Segment" which is the extensions can be assigned on the IPPBX to provision the end device.
Default Extension Segment	Clicking this button will reset the extension range to their default values.

SIP Settings

SIP Settings page contains all the options which are related to call signaling.

General

In General tab, the user can find settings like the bind port for SIP protocol, the realm of digest etc...

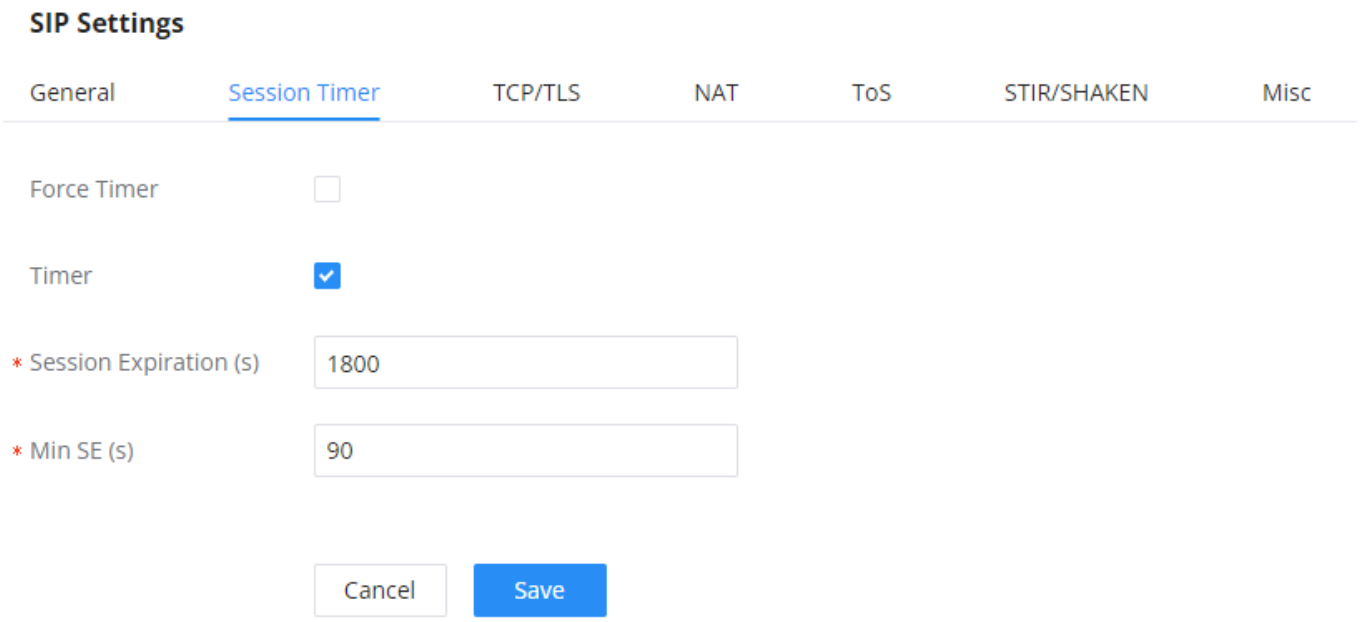


General SIP Settings

Realm For Digest Authentication	Configure this as the host name or domain name of the PBX. Realms MUST be globally unique according to RFC3261.
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IPv4 Address	Configure the IPv4 address to bind to. "0.0.0.0" means to bind to all IP addresses.
Bind IPv6 Address	Configure the IPv6 address to bind to. "[::]" means to bind to all IP addresses.
MWI From Header	If disabled, the server will not transfer the Diversion Header
Enable Diversion Header	If set to "No", all transfers initiated by the endpoint in the IPPBX will be disabled (unless enabled in peers or users). The default setting is "Yes".
Send Deflection Diversion	If 'Enable Diversion Header' and this option enabled, the INVITE request will contain Diversion with reason 'deflection' while the inbound call been routed to an external number.
Block Collect Calls	If enabled, collect calls will be blocked.Note: Collect calls are indicated by the header "P-Asserted-Service-Info: service-code=Backward Collect Call, P-Asserted-Service-Info: service-code=Collect Call".

Session Timer

In this tab, the user can define settings related to the duration of each SIP session. SIP protocol uses the message re-INVITE and UPDATE as a method to keep the session alive.



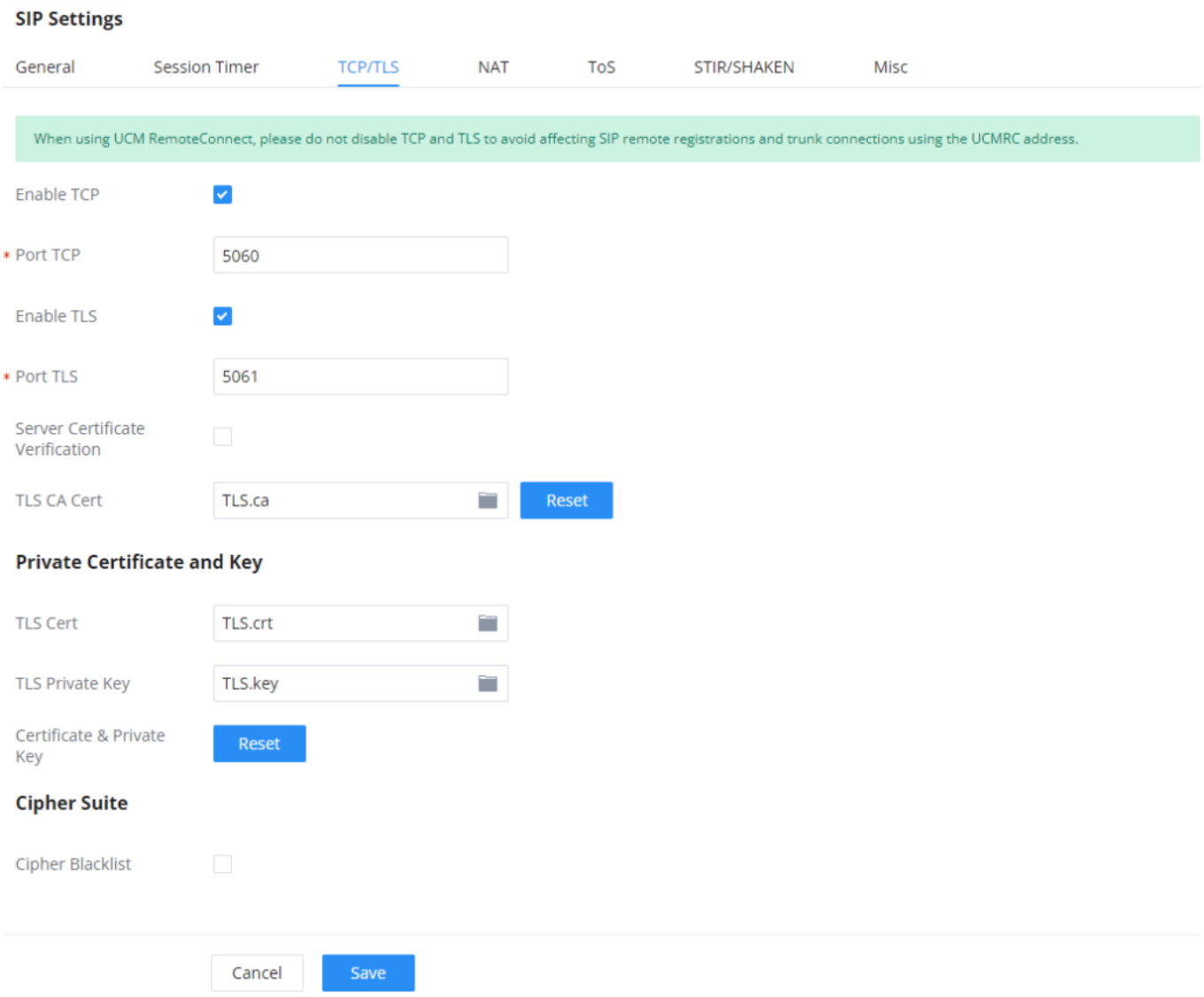
SIP Session Timer

Force Timer	If checked, always request, and run session timer.
Timer	If checked, run session timer only when requested by other UA.
Session Expire	Configure the maximum session refresh interval (in seconds). Default is 1800.
Min SE	Configure the minimum session refresh interval (in seconds). The default setting is 90.

SIP Settings/Session Timer

TCP/TLS

In this page the administrator can configure the options related to SIP registering for both SIP endpoints and SIP trunks when TCP or TLS is used as the transport protocol. In this page, the user can also upload TLS certificate and TLS private key to encrypt the SIP traffic.



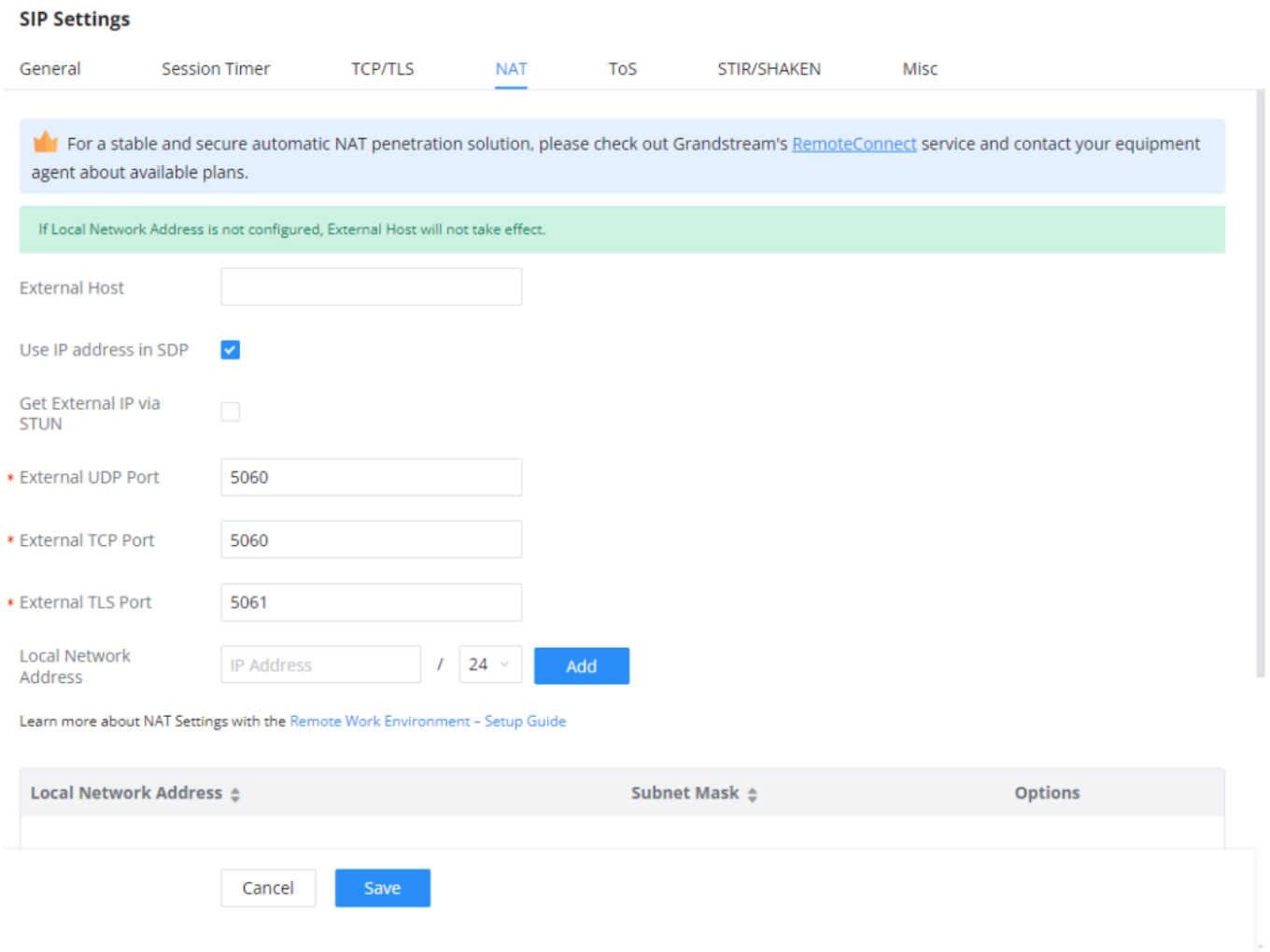
TCPTLS Settings

TCP Enable	Configure to allow incoming TCP connections with the PBX. The default setting is "No".
TCP Bind IPv4 Address	Configure the IP address for the TCP server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 5060. For example, 192.168.1.1:5062.
TCP Bind IPv6 Address	Configure the IPv6 address for the TCP server to bind to. "[::]" means bind to all interfaces. The port number is optional with the default being 5060. For example, [2001:0DB8:0000:0000:0000:0000:1428:0000]:5060.
TLS Enable	Configure to allow incoming TLS connections with the PBX. The default setting is "Yes".
TLS Bind IPv4 Address	Configure the IPv4 address for TLS server to bind to. "0.0.0.0" means binding to all interfaces. The port number is optional, and the default port number is 5061. For example, 192.168.1.1:5063. Note: The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses.
TLS Bind IPv6 Address	Configure the IPv6 address for TLS server to bind to. "[::]" means bind to all interfaces. The port number is optional with default being 5061. For example, [2001:0DB8:0000:0000:0000:0000:1428:0000]:5061. Note: The IP address must match the common name (host name) in the certificate so that the TLS socket will not bind to multiple IP addresses.
TLS Do Not Verify	If enabled, the TLS server's certificate will not be verified when acting as a client. The default setting is "Yes".

TLS Self-Signed CA	This is the CA certificate if the TLS server being connected to requires self-signed certificate, including server's public key. This file will be renamed as "TLS.ca" automatically.Notes:Supported file extensions include .crt, .cer, .der, .pem, .key, and .ca.DER-formatted certificates are supported and will automatically be converted to PEM format upon upload.The size of the uploaded certificate file must be under 2MB.
Reset Certificates	Clicking on this button will reset the certificates.
Private Certificate and Key	
TLS Cert	Upload the certificate file used for TLS connections. It must contain the client's private key and the server's signed certificate. Uploaded files will automatically be saved as "TLS.crt".Notes:Supported file extensions include .crt, .cer, .der, .pem, .key, and .ca.DER-formatted certificates are supported and will automatically be converted to PEM format upon upload.The size of the uploaded certificate file must be under 2MB.
TLS Key	This file must be named with the CA subject name hash value. It contains CA's (Certificate Authority) public key, which is used to verify the accessed servers. This file will be renamed as "TLS.key" automatically.Notes:Supported file extensions include .crt, .cer, .der, .pem, .key, and .ca.DER-formatted certificates are supported and will automatically be converted to PEM format upon upload.The size of the uploaded certificate file must be under 2MB.
Reset Certificates	Clicking on this button will reset the certificates.
Cipher Suite	
Restrict Cipher List	By default, all SIP TLS encryption suites are in effect on the system, and when turned on, you can configure the encryption suites allowed to be used.
Cipher Suite	Select the encryption suites that are allowed to be used for SIP TLS connections, in the order of priority as configured.

NAT

In this page the user can find settings related to NAT which are taken into consideration when emitting a call from the PBX.



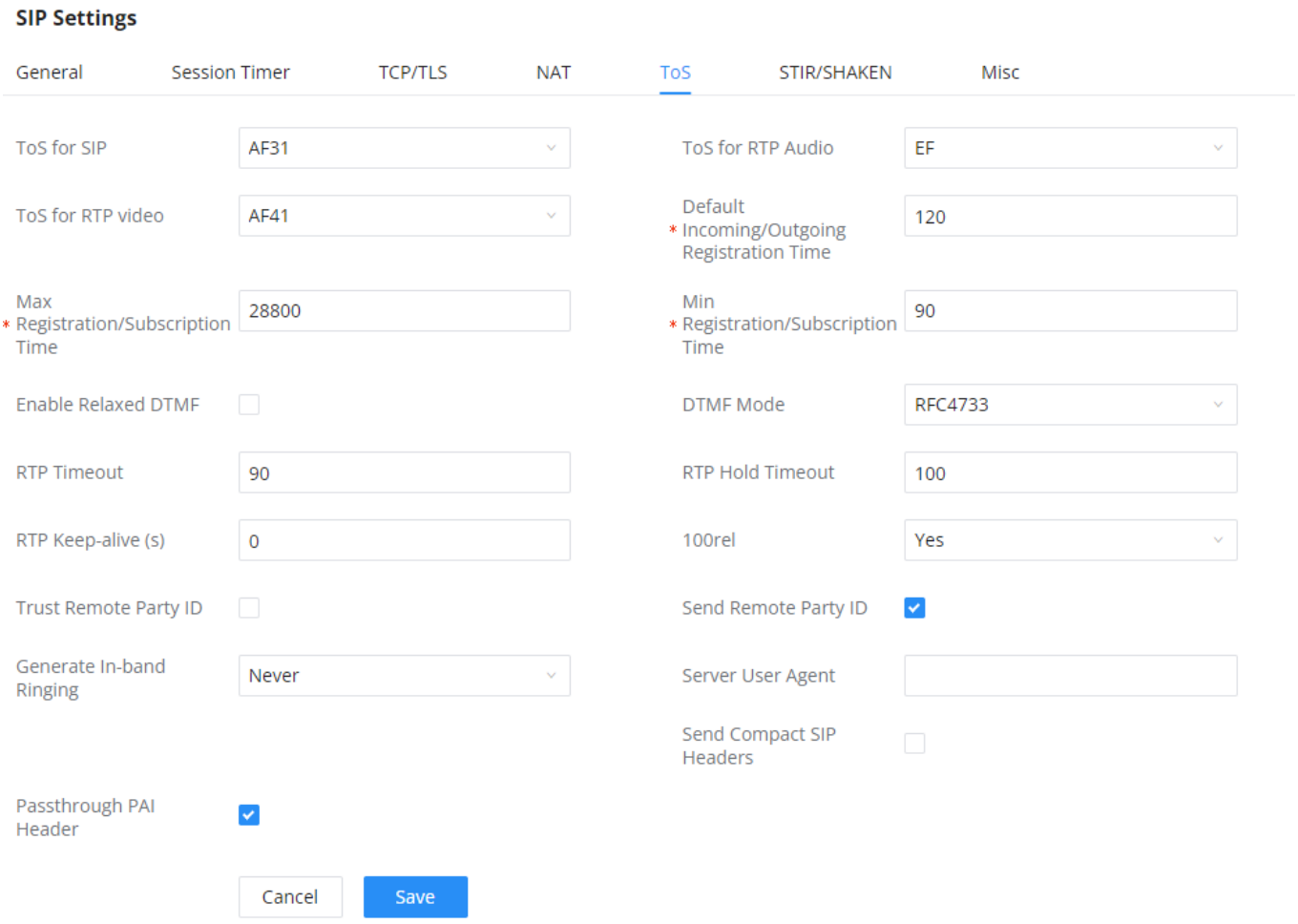
NAT Settings

External Host	Configure a static IP address and port (optional) used in outbound SIP messages if the IPPBX is behind NAT. If it is a host name, it will only be looked up once.
Use IP address in SDP	If enabled, the SDP connection will use the IP address resolved from the external host.
External UDP Port	Configure externally mapped UDP port when the PBX is behind a static NAT or PAT.
External TCP Port	Configure the externally mapped TCP port when the IPPBX is behind a static NAT or PAT.
External TLS Port	Configures the externally mapped TLS port when IPPBX is behind a static NAT or PAT.
Local Network Address	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/16

SIP Settings/NAT

ToS

ToS (Type of Service) define the attributes related to SIP settings when emitting a call from the PBX. The ToS values define the priority of the packets transferred, the PBX attributes the values to packet trasferred to other switching/routing equipment. Thus, indicating the importance of the packet. Based on the importance and the priority the packet, the switching/routing equipment decides whether or not to drop the packet and the queuing policy of the packet.



ToS Settings

ToS for SIP	Configure the Type of Service for SIP packets. The default setting is None.
ToS for RTP Audio	Configure the Type of Service for RTP audio packets. The default setting is None.
ToS for RTP Video	Configure the Type of Service for RTP video packets. The default setting is None.
Default Incoming/Outgoing Registration Time	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.
Max Registration/Subscription Time	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the IPPBX. The default setting is 3600.
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the IPPBX. The default setting is 60.
Enable Relaxed DTMF	Select to enable relaxed DTMF handling. The default setting is "No".
DTMF Mode	Select DTMF mode to send DTMF. The default setting is RFC4733. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, a-law or u-law are required. When "Auto" is selected, "RFC4733" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC4733".
RTP Timeout	During an active call, if there is no RTP activity within the timeout (in seconds), the call will be terminated. The default setting is no timeout. Note: This setting does not apply to calls on hold.
RTP Hold Timeout	When the call is on hold, if there is no RTP activity within the timeout (in seconds), the call will be terminated. This value of RTP Hold Timeout should be larger than RTP Timeout. The default setting is no timeout.

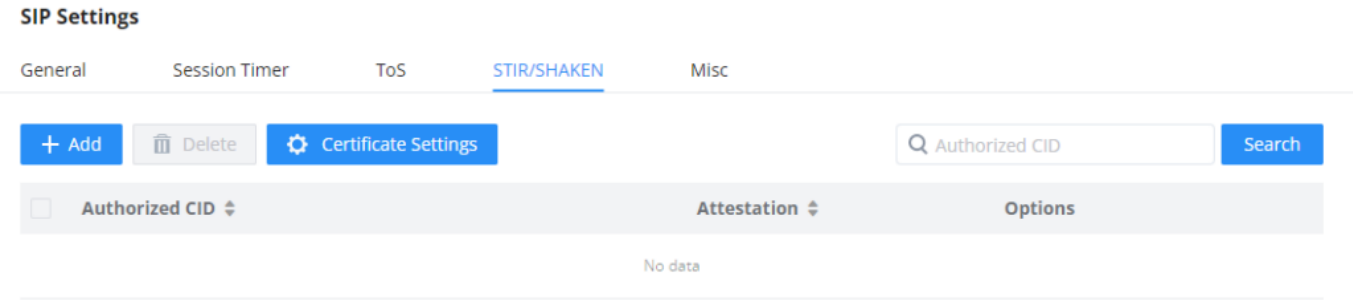
RTP Keep-alive	This feature can be used to avoid abnormal call drop when the remote provider requires RTP traffic during proceeding. For example, when the call goes into voicemail and there is no RTP traffic sent out from IPPBX, configuring this option can avoid voicemail drop. When configured, RTP keep-alive packet will be sent to remote party at the configured interval. If set to 0, RTP keep-alive is disabled.
100rel	Configure the 100rel setting on IPPBX. The default setting is "Yes".
Trust Remote Party ID	Configure whether the Remote-Party-ID should be trusted. The default setting is "No".
Send Remote Party ID	Configure whether the Remote-Party-ID should be sent or not. The default setting is "No".
Generate In-Band Ringing	Configure whether the PBX should generate Inband ringing or not. The default setting is "Never". Yes: The PBX will send 180 Ringing followed by 183 Session Progress and in-band audio. No: The PBX will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing. Never: Whenever ringing occurs, the IPPBX will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly.
Server User Agent	Configure the user agent string for the PBX.
Send Compact SIP Headers	If enabled, compact SIP headers will be sent. The default setting is "No".
Passthrough PAI Header	Passthrough PAI Header

SIP Settings/ToS

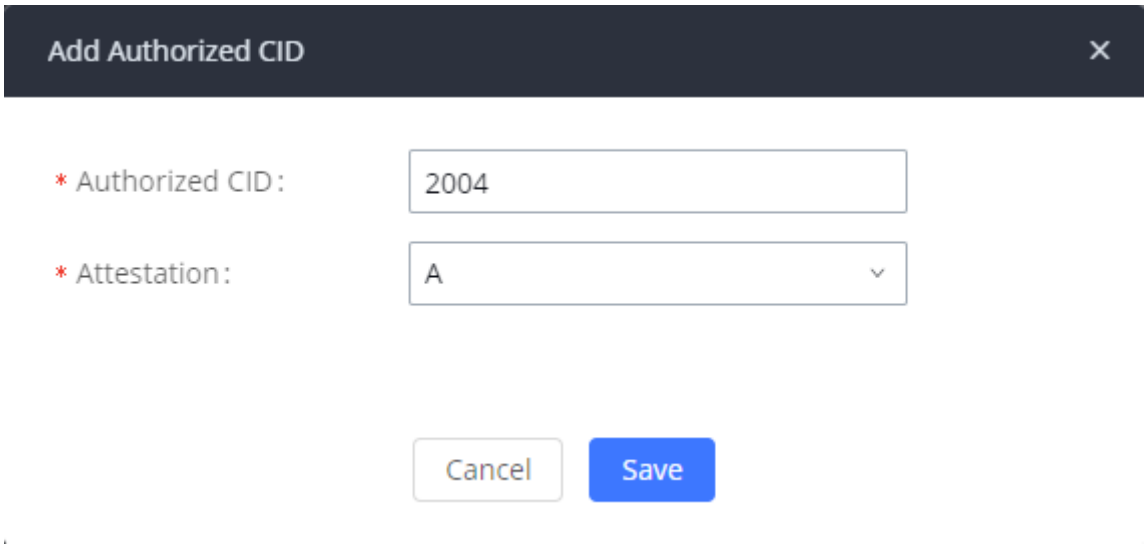
STIR/SHAKEN

To prevent robocalls, IPPBX now supports STIR/SHAKE protocols. Related options have been added as a new tab in the **SIP Settings** page.

Clicking on the **Add** button will show the following window:



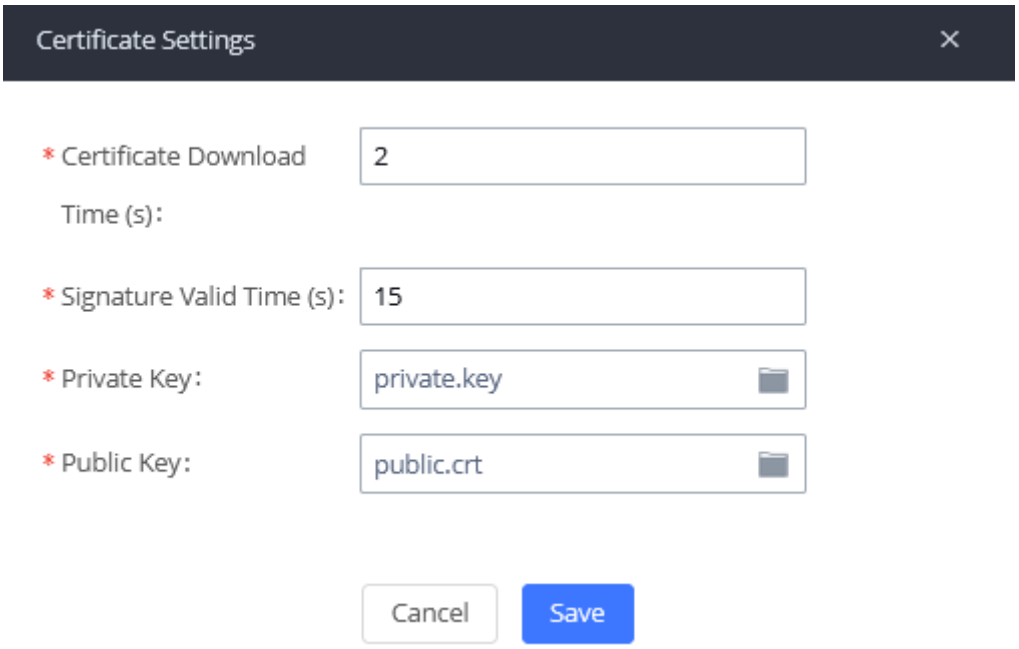
STIRSHAKEN



STIRSHAKEN Add Authorized CID and Attestation level

Authorized CID	Configure the authorized CID.
Attestation	Configure the attestation level, which is the level of confidence of the carrier that the CID has not been spoofed. The following options are available: A (Full attestation): The carrier is associated with the caller and the number. There is high confidence that the CID has not been spoofed. B (Partial attestation): The carrier is associated with the caller but not the number. There is uncertainty about whether the CID has been spoofed or not. C (Gateway attestation): The carrier is not associated with the caller and has no confidence at all about the number. Generally used for traceback.

Clicking on the **Certificate Settings** button will bring up the following window:



STIRSHAKEN Certificate Settings

Certificate Download Time (s)	Configure the public key download timeout period, the default value is 2 seconds.
Signature Valid Time (s)	Configure the validity period of the digital signature, the default value is 15 seconds.
Private Key	Configure the Private key. Note: The uploaded file must be less than 2MB in file size, only supports the .key format and must be ECC type. This file will automatically be renamed to "private.key".
Public Key	Configure the Public Key. Note: The uploaded file must be less than 2MB in file size, only supports the .crt format and must be ECC type. This file will automatically be renamed to "public.crt".

SIP Settings/STIR/SHAKEN - Certificate Settings

Misc

Outbound SIP Registrations	
Register Timeout	Configure the register retry timeout (in seconds). The default setting is 20.
Register Attempts	Configure the number of registration attempts before the IPPBX gives up.The default setting is 0, which means the IPPBX will keep trying until the server side accepts the registration request.
Video	
Max Bit Rate (kb/s)	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
Support SIP Video	Select to enable video support in SIP calls. The default setting is "Yes".
Reject Non-Matching INVITE	If enabled, when rejecting an incoming INVITE or REGISTER request, the IPPBX will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP usernames. The default setting is "No".
SDP Attribute Passthrough	
Enable Attribute Passthrough	If enabled, and IPPBX receives a call that contains unknown FEC/FECC/FBCP attributes, they will be passed through the IPPBX unmodified.
Early Media	
Enable Use Final SDP	If enabled, call negotiation will use final response SDP.
Ignore 180 Response	If enabled, ignore the ringing indication if has sent 183 response.
Blind Transfer	
Allow callback when blind transfer fails	If enabled, the IPPBX will call back to the transferrer when blind transfer fails (due to the destination being busy or not answering).Note: This feature applies only to internal calls.
Blind transfer timeout	Configure the amount of time in seconds that the transferred party will wait for the destination to answer before being redirected back to the transferrer. Default is 60 seconds.
DNS	
DNS mode	This option affects the DNS query only during Calls. When you choose A&AAAA, IPPBX will do both A and AAAA type DNS query; when you chose A, IPPBX will only do A type DNS query; and when you chose AAAA, IPPBX will only do AAAA type DNS query.
Hold	
Forward HOLD Requests	Configure the IPPBX to forward HOLD requests instead of processing holds internally. This serves to meet the standards set by some providers that require HOLD requests to be passed along from endpoint to endpoint. This option is disabled by default.Note: Enabling this option may cause hold retrieval issues and MOH to not be heard.

RTP Settings

RTP protocol is used to transmit the media data between the IPPBX and the SIP endpoints. In this section, the user can define settings for RTP and BFCP protocols, as well as the payload type indicators for the media data exchanged.

RTP Settings

Here, the user can set the ports for RTP and BFCP as well as the NAT-traversal-relatedsettings for the media packets.

RTP Start	Configure the RTP port starting number. The default setting is 10000.
RTP End	Configure the RTP port ending address. The default setting is 20000.
Strict RTP	Configure to enable or disable strict RTP protection. If enabled, RTP packets that do not come from the source of the RTP stream will be dropped. The default setting is "Disable".
RTP Checksums	Configure to enable or disable RTP Checksums on RTP traffic. The default setting is "Disable".
ICE Support	Configure whether to support ICE. The default setting is enabled.ICE is the integrated use of STUN and TURN structure to provide reliable VoIP or video calls and media transmission, via a SIP request/ response model or multiple candidate endpoints exchanging IP addresses and ports, such as private addresses and TURN server address.
STUN Server	Configure STUN server address. STUN protocol is a Client/Server and also a Request/Response protocol. It is used to check the connectivity between the two terminals, such as maintaining a NAT binding entries keep-alive agreement. The default STUN Server is stun.ipvideotalk.com.Valid format:[(hostname | IP-address) [':' port]The default port number is 3478 if not specified.
BFCP UDP Start	Configure BFCP UDP port starting number. The default setting is 50000.
BFCP UDP End	Configure BFCP UDP port ending number. The default setting is 52999.
BFCP TCP Start	Configure BFCP TCP port starting number. The default setting is 53000.
BFCP TCP End	Configure BFCP TCP port ending number. The default setting is 55999.
TURN Server	Configure TURN server address. TURN is an enhanced version of the STUN protocol and is dedicated to the processing of symmetric NAT problems.
TURN Server Name	Configure turn server account name
TURN Server Password	Configure turn server account password.
Connection Protocol	Protocol used to connect to the TURN server.
Number of ICE Candidates	This configures the number of pre-collected ICE candidates to gather and send to remote peers. The higher the number, the greater the network traffic consumption.

Payload Type Settings

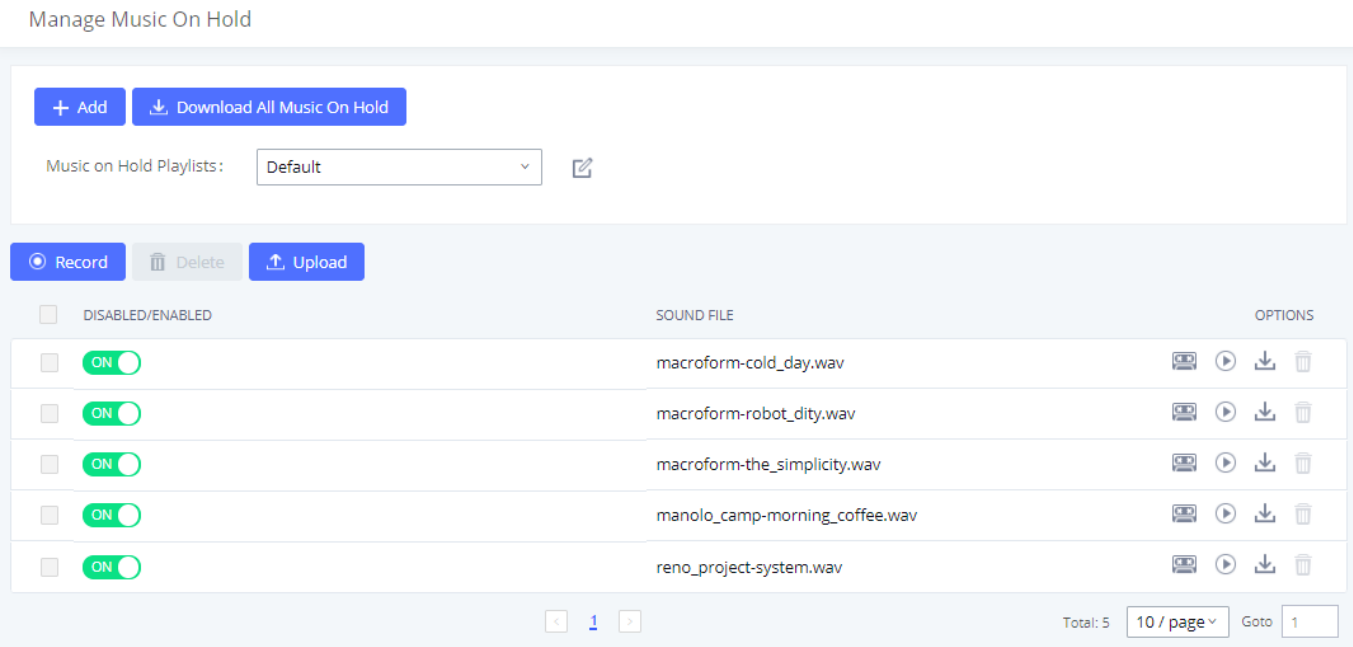
The PBX payload type for audio codecs and video codes can be configured here.

Audio Codecs	
AAL2-G.726	ADPCM (G.726, 32kbps, AAL2 codeword packing).
DTMF	Dual-tone Multi-frequency.
G.721 Compatible	G.721 Compatible

G.726	ADPCM (G.726, 32kbps, RFC3551 codeword packing).
ILBC	ILBC Free Compression.
Opus	Opus
G.722.1	G.722.1: Low-complexity coding, 24kbps
G.722.1C	G.722.1C: Low-complexity coding, 48kbps
Audio FEC Payload Type	Audio FEC Payload Type
Audio RED Payload Type	Audio RED Payload Type
Video Coding	
H.264	H.264 Video.
H.265	H.265 Video.
H.263P	H.263+ Video.
VP8	VP8 Video.
Other Settings	
Main Video FEC	Main Video FEC.
RTP FECC	RTP FECC
RTX	Used for packet retransmission. PBX supports only video RTP retransmission.

Music On Hold

Music On Hold settings can be accessed via Web GUI→**PBX Settings→Music On Hold**. In this page, users could configure music on hold class and upload music files. The "default" Music On Hold class already has 5 audio files defined for users to use.



Music On Hold Default Class

- Click on "Create New MOH Class" to add a new Music On Hold class.
- Click on to configure the MOH class sort method to be "Alpha" or "Random" for the sound files.
- Click on next to the selected Music On Hold class to delete this Music On Hold class.
- Click on to start uploading. Users can upload:

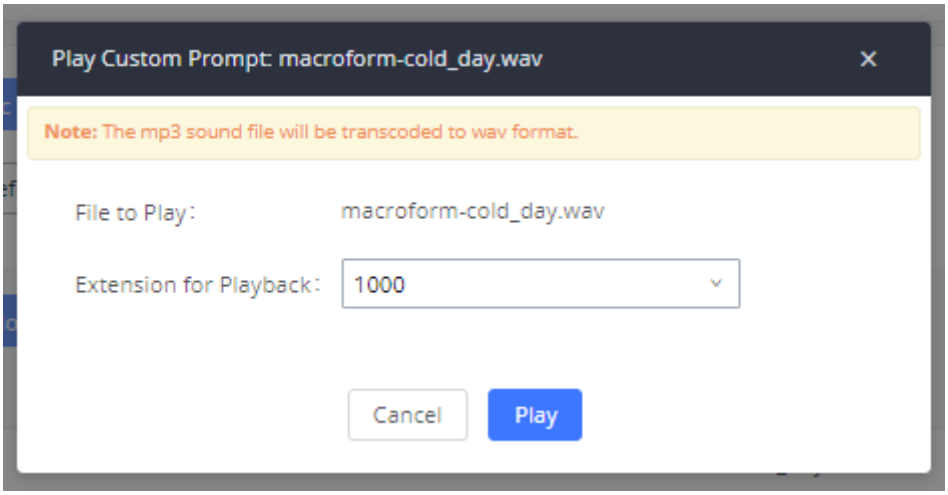
- Single files with 8KHz Mono Music file, or
- Music on hold files in a compressed package with .tar, .tar.gz and .tgz as the suffix. The file name can only be letters, digits, or special characters -_
- the size for the uploaded file should be less than 30M, the compressed file will be applied to the entire MoH.

- Users could also download all the music on hold files from IPPBX. In the Music On Hold page, click on and the file will be downloaded to your local PC.
- Click on to disable it from the selected Music On Hold Class.
- Click on to enable it from the selected Music On Hold Class.
- Select the sound files and click on to delete all selected Music On Hold files.

The PBX allows Users to select the Music On Hold file from WebGUI to play it. The PBX will initiate a call to the selected extension and play this Music On Hold file once the call is answered.

Steps to play the Music On Hold file:

1. Click on the button for the Music On Hold file.
2. In the prompted window, select the extension to playback and click .

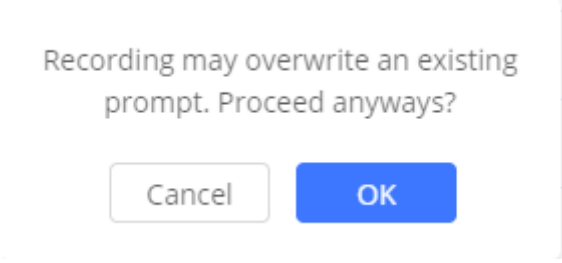


Play Custom Prompt

3. The selected extension will ring.
4. Answer the call to listen to the music playback.

Users could also record their own Music On Hold to override an existing custom prompt, this can be done by following those steps:

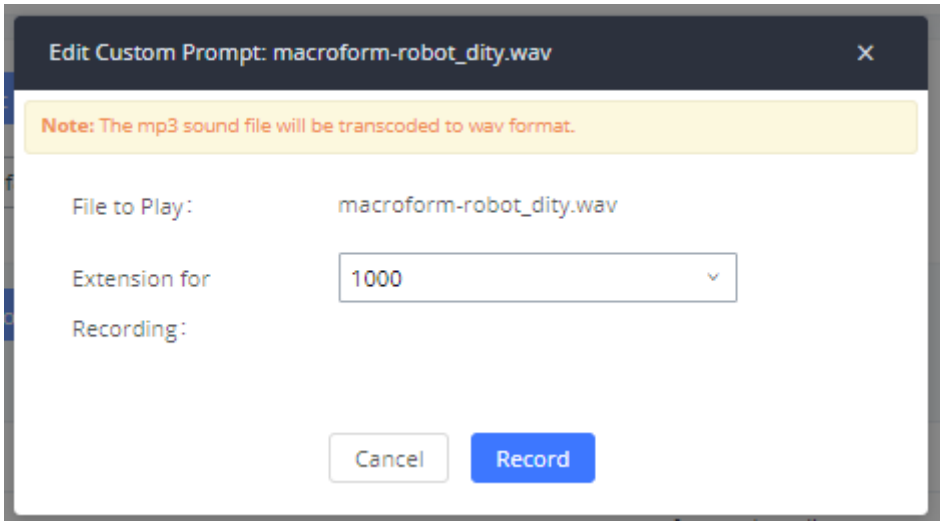
1. Click on .
2. A message of confirmation will pop up, as shown below.



Information Prompt

3. Click .

4. In the prompted window, select the extension to playback and click .



Record Custom Prompt

5. Answer the call and start to record your new music on hold.
6. Hangup the call and refresh Music On Hold page then you can listen to the new recorded file.

Once the MOH file is deleted, there are two ways to recover the music files.

- Users could download the MOH file from this link: http://downloads.asterisk.org/pub/telephony/sounds/releases/asterisk-moh-opsound-wav-2.03.tar.gz

After downloading and unzip the pack, users could then upload the music files to IPPBX.

- Factory reset could also recover the MOH file on the IPPBX.

Voice Prompt

Language Settings

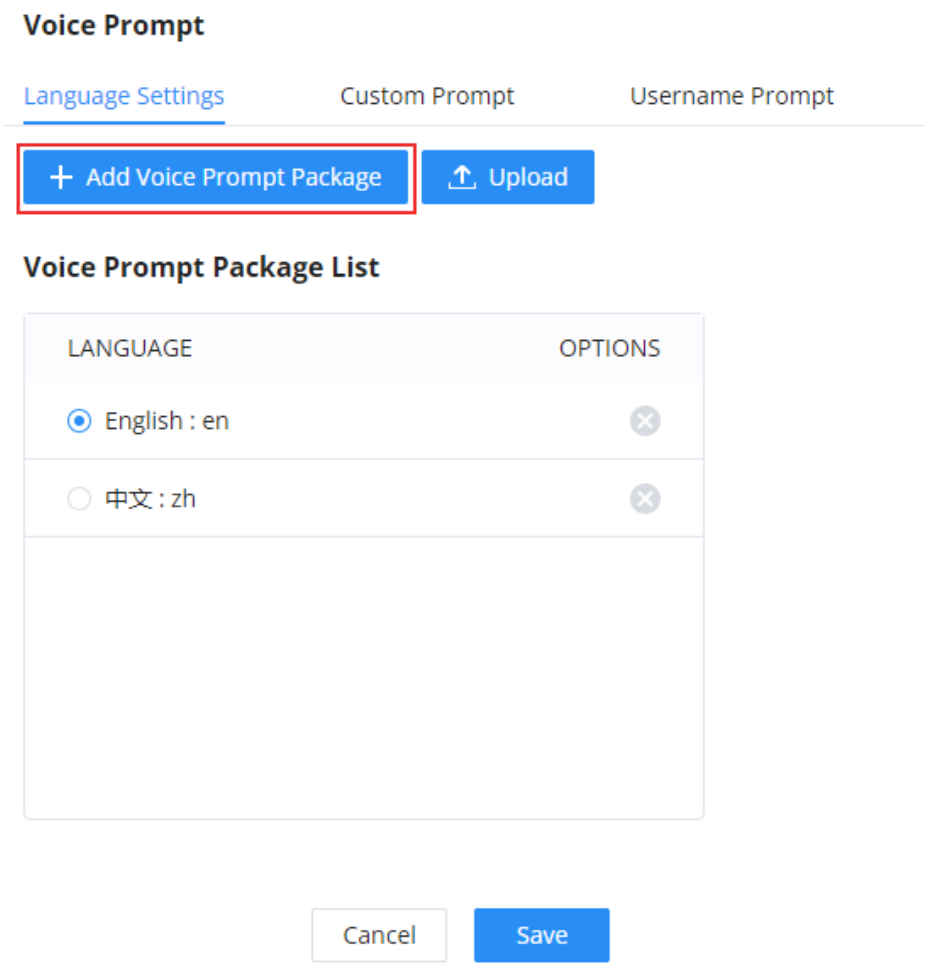
The IPPBX supports multiple languages in Web GUI as well as system voice prompt. Currently, there are 16 languages supported in system voice prompt: **English (United States), Arabic, Chinese, Dutch, English (United Kingdom), French, German, Greek, Hebrew, Italian, Polish, Portuguese, Russian, Spanish, Catalan, Swedish and Turkish.**

English (United States) and Chinese voice prompts are preloaded in with the IPPBX already. The other languages provided by Grandstream can be downloaded and installed from the IPPBX Web GUI directly. Additionally, users could customize their own voice prompts, package them and upload to the IPPBX.

Language settings for voice prompt can be accessed under Web GUI→**PBX Settings→Voice Prompt→ Language Settings**.

- **Download and Install Voice Prompt Package**

To download and install voice prompt package in different languages from IPPBX Web GUI, click on "Add Voice Prompt Package" button.

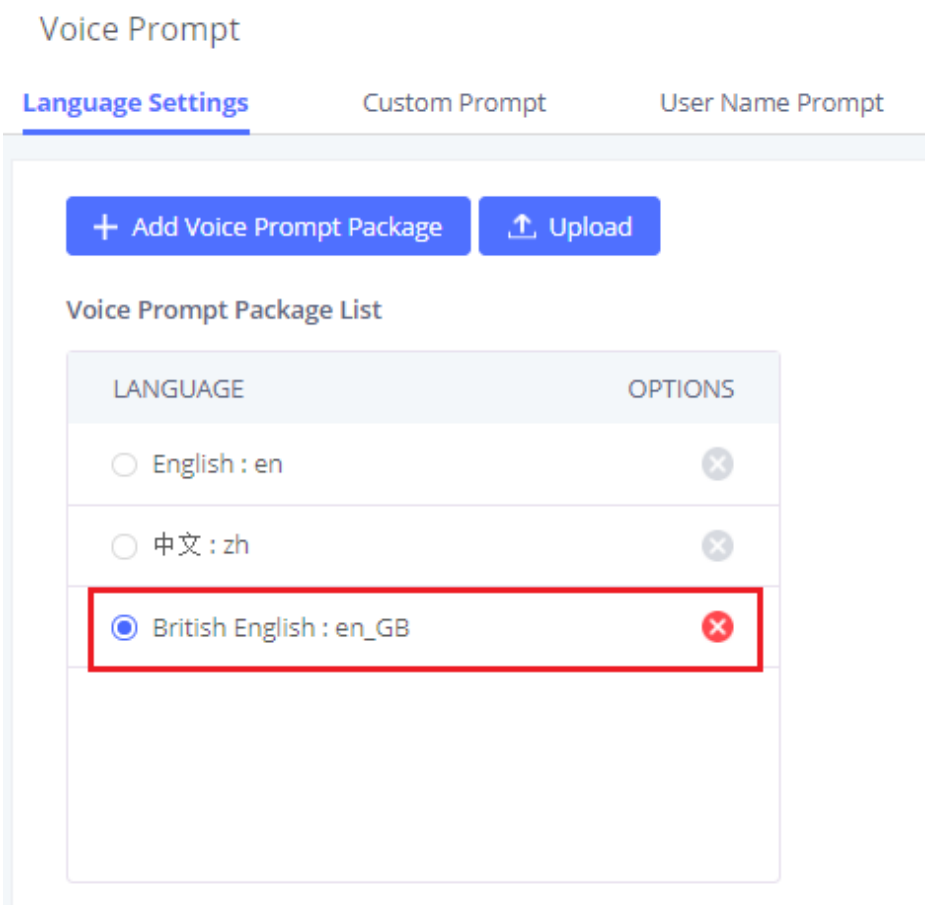


Language Settings for Voice Prompt

A new dialog window of voice prompt package list will be displayed. Users can see the version number (latest version available V.S. current installed version), package size and options to upgrade or download the language.

Voice Prompt Package List

Click on to download the language to the IPPBX. The installation will be automatically started once the downloading is finished.

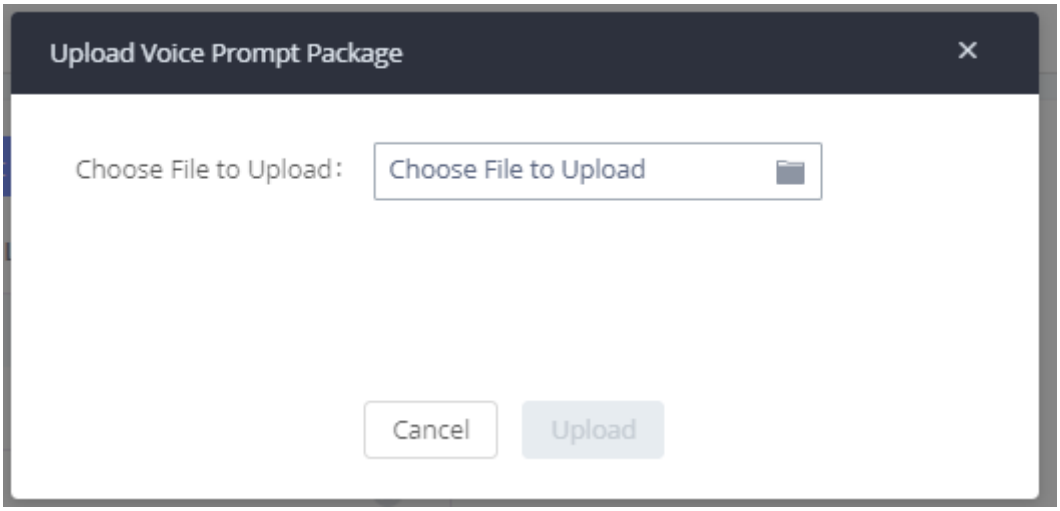


New Voice Prompt Language Added

A new language option will be displayed after successfully installed. Users then could select it to apply in the IPPBX system voice prompt or delete it from the IPPBX.

Custom Prompt

On the IPPBX, if the user needs to replace some specific customized prompt, the user can upload a single specific customized prompt from Web GUI→**PBX Settings→Voice Prompt→Language Settings** and click on **"Upload"** instead of the entire language pack.



Upload Single Voice Prompt for Entire Language Pack

Username Prompt

There are two ways to customize/set new username prompt:

Upload Username Prompt File from Web GUI

1. First, Users should have a pre-recorded file respecting the following format:

- PCM encoded / 16 bits / 8000Hz mono.
- In .tar/.tar.gz/.tgz format
- File size under 30M.
- Filename must be set as the extension number with 18 characters max. For example, the recorded file name 1000.wav will be used for extension 1000.

2. Go under web GUI **PBX Settings → Voice Prompt → Username Prompt** and click on "**Upload**" button.

3. Select the recorded file to upload it and press Save and Apply Settings.

- Click on to record again the username prompt.
- Click on to play recorded username prompt.
- Select username prompts and press to delete specific file or select multiple files for deletion using the button "**Delete**" .

Record Username via Voicemail Menu

The second option to record username is using voicemail menu, please follow below steps:

- Dial *98 to access the voicemail
- After entering the desired extension and voicemail password, dial "0" to enter the recordings menu and then "3" to record a name.

Another option is that each user can record their own name by following below steps:

- The user dials *97 to access his/her voicemail
- After entering the voicemail password, the user can press "0" to enter the recordings menu and then "3" to record his name.

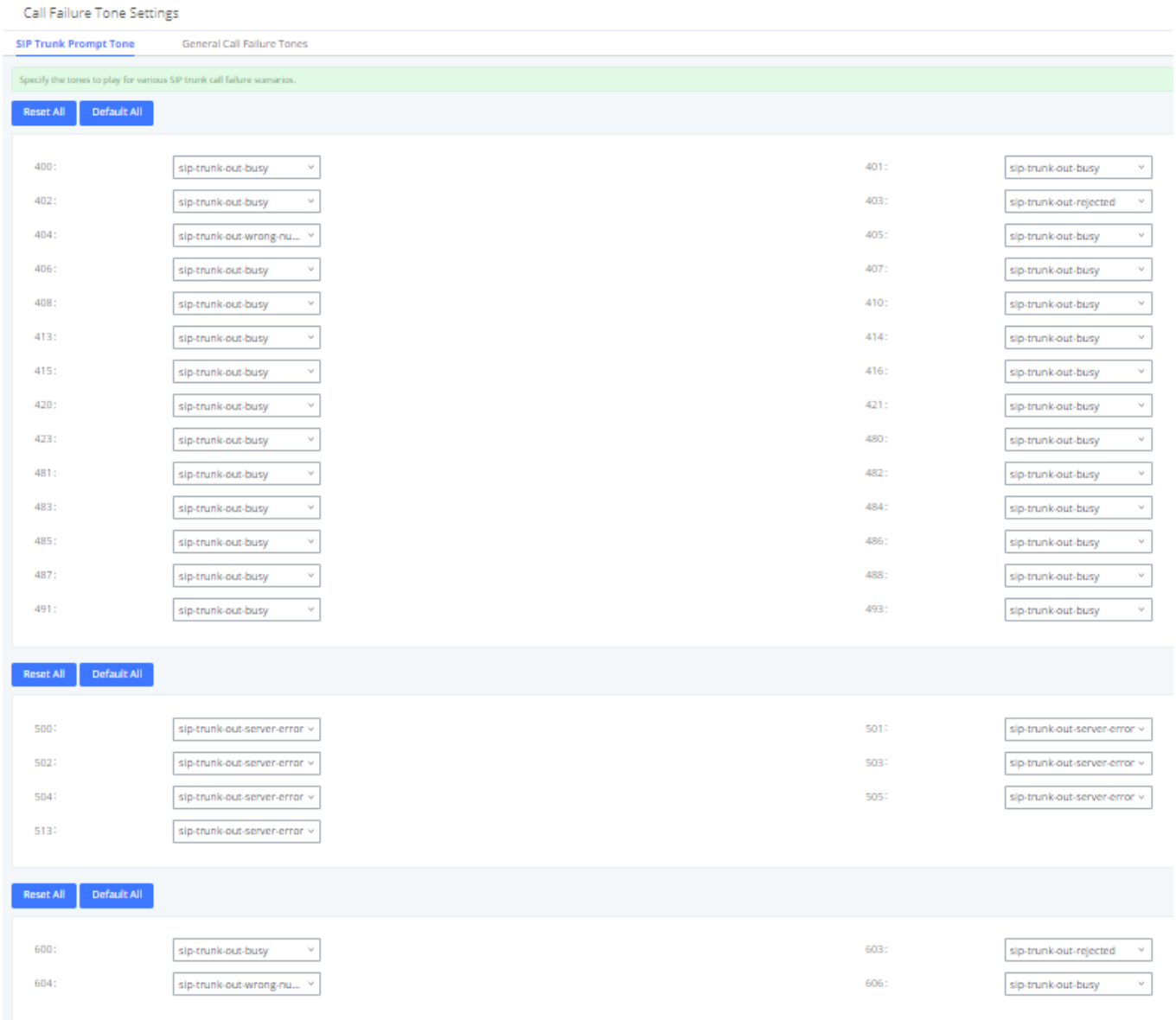
Call Prompt Tones

SIP Trunk Prompt Tone

Prompt Tone Settings tab has been added to the IPPBX to help users choose which prompt will be played by the IPPBX during call failure, the following voice message responses have been added and can be set to be played for 4XX, 5XX, and 6XX call failures:

- Default for 404 and 604 status codes: **"**Your call can't be completed as dialed. Please check the number and dial again.**"**
- Default for 5xx status codes: **"**Server error. Please check your device.**"**
- Default for 403 and 603 status codes: **"**The call was rejected by the server. Please try again later.**"**
- Default for all other status codes: **"**All circuits are busy now. Please try again later.**"**

Additionally, custom voice messages recorded and uploaded in **PBX Settings→Voice Prompt→Custom Prompt** can be used for these failure responses instead of the default messages.

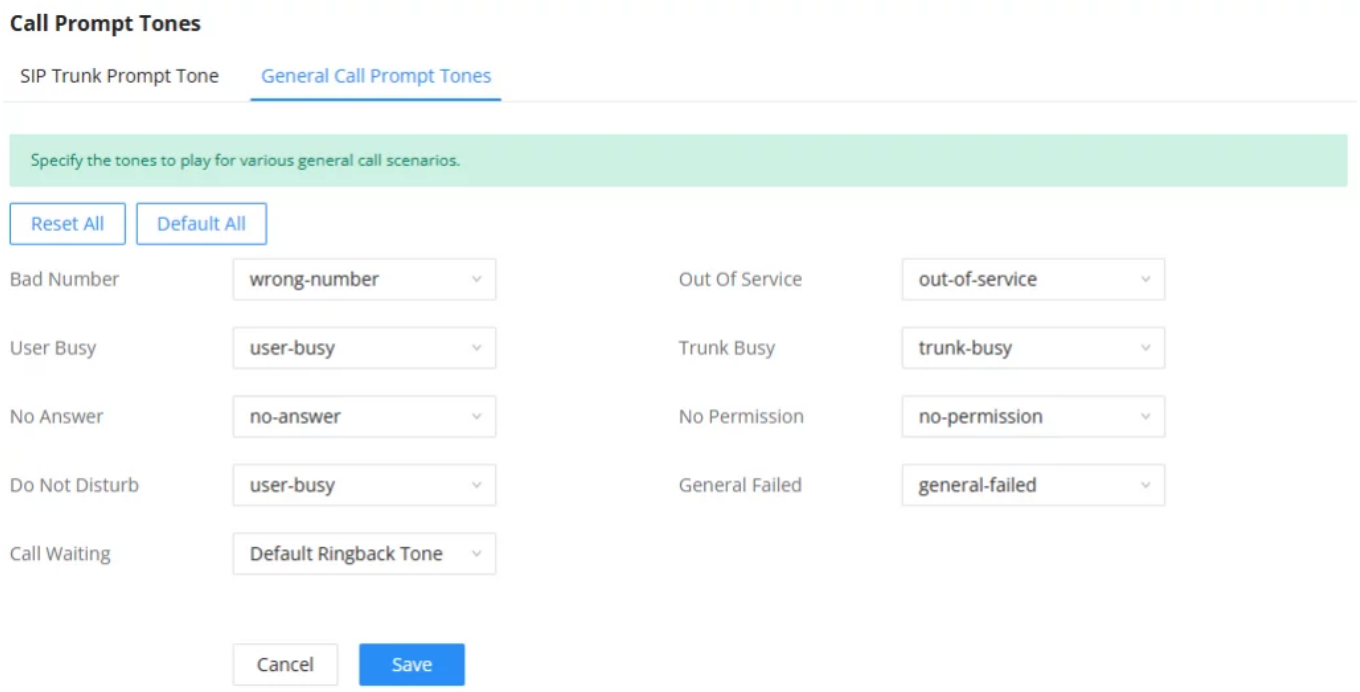


SIP Trunk Prompt Tone

General Call Prompt Tones

Moreover, users also have the possibility to customize the prompt for typical call failure reasons like (no permission to allow outbound calls, busy lines, incorrect number dialed ...Etc.).

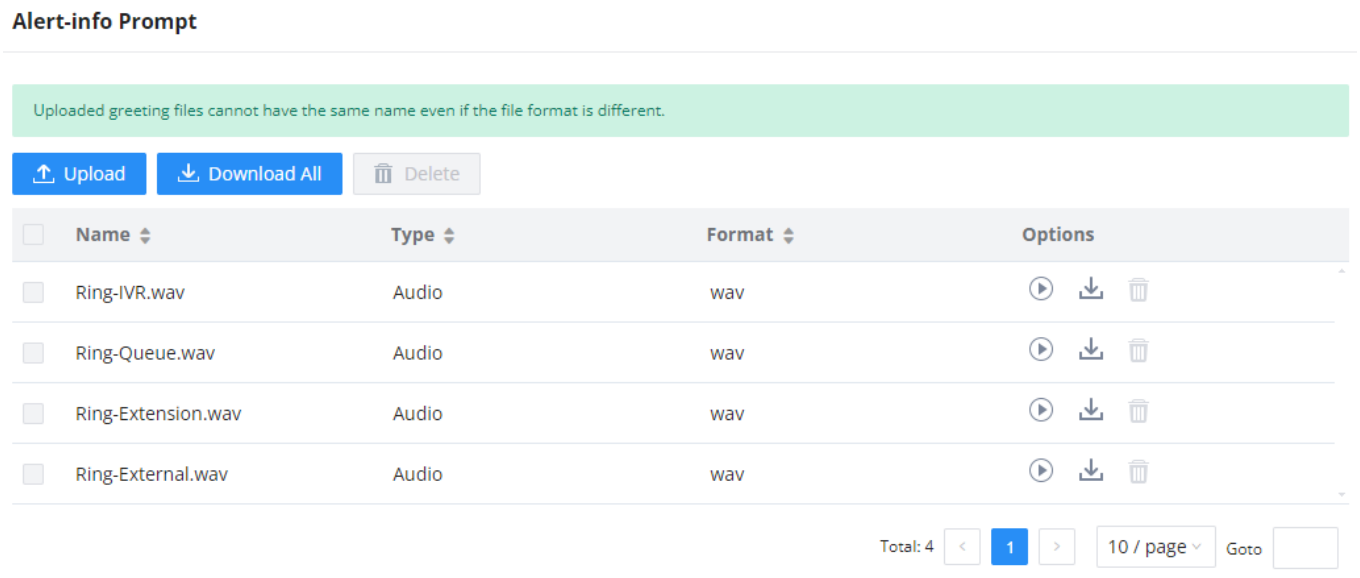
To customize these prompts user could record and upload their own files under "**"PBX Settings → Voice Prompt → Custom Prompts**" then select each one for specific call failure case under "**PBX Settings** -> **Call Prompt Tones → General Call Prompt Tones**" page as shown on the following figure:



General Call Prompt Tones

Alert-Info Prompt

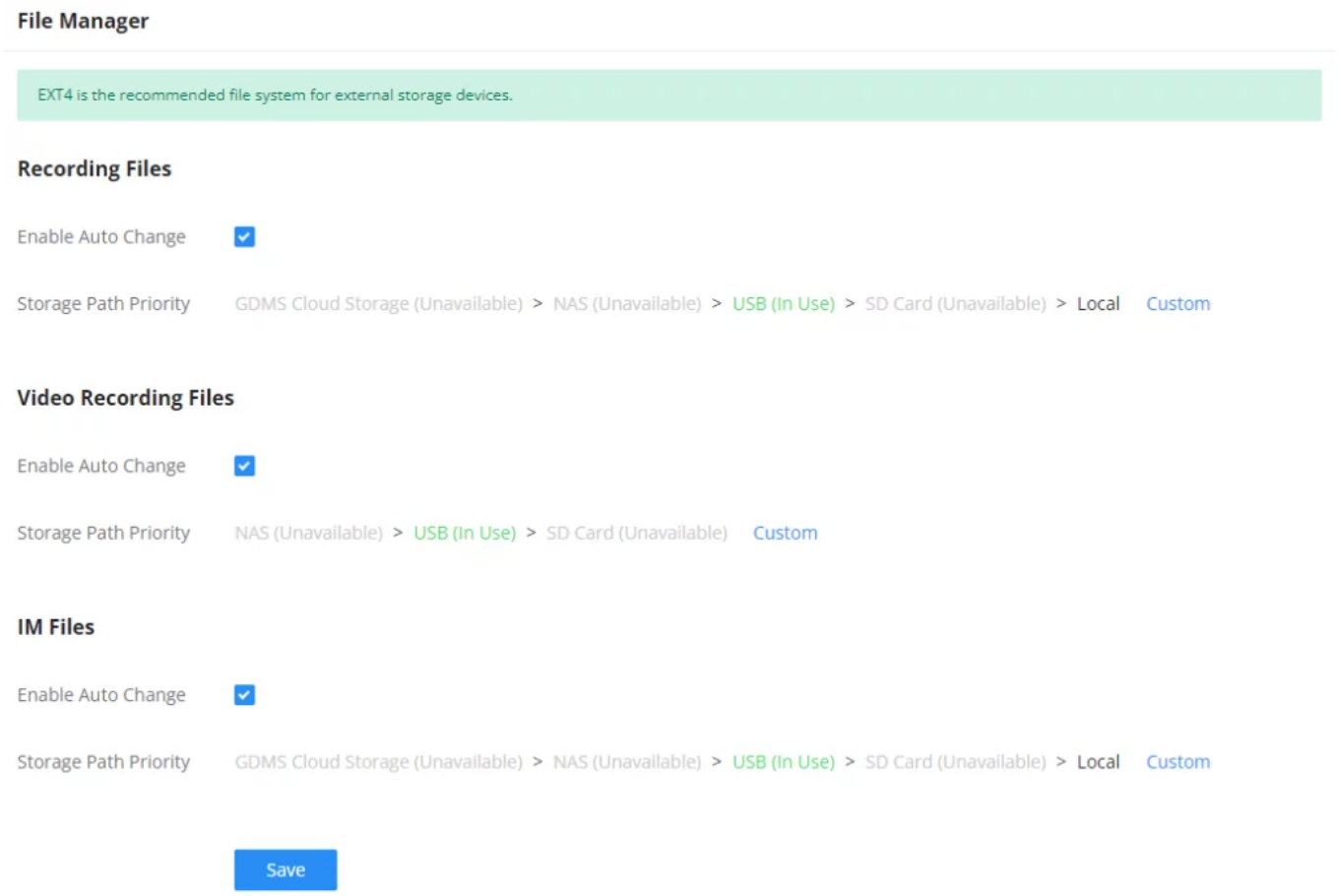
The user can change upload the ringtone files which can be used with Alert-info to play various ringtones depending on the configured behaviour. In this setting, the user can view all the ringtones uploaded, play them, download them, or delete them. If the user wishes to upload new ringtones, he/she can click on "Upload" and choose the ringtone files to upload.



Alert info Prompt

File Manager

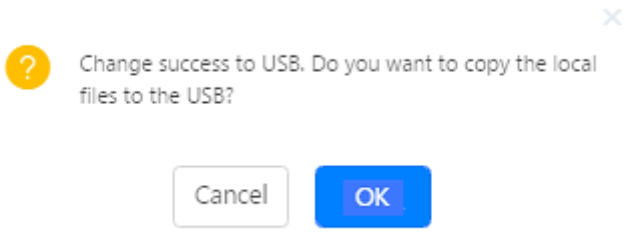
IPPBX supports automatic or manual recording of calls and storage of IM chat files. Only recording files and IM files can be stored locally or on the GDMS, meanwhile, video recording files can only be stored on NAS. Local storage means that the files will be stored in the internal memory of the IPPBX. For extra storage capacity the user can plug a USB flash drive or an SD card, the IPPBX will always store in the USB drive first, then the SD card. In case no storage drive is attached, it will automatically start storing the files internally.



File Manager

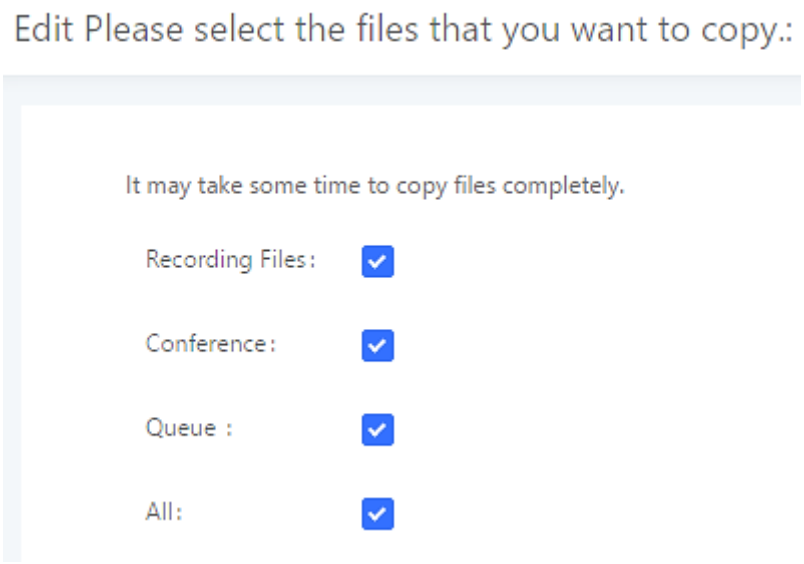
- If "**Enable Auto Change**" is enabled, the files will be automatically saved in the available USB Disk or SD card plugged into the IPPBX. If both USB Disk and SD card are plugged in, the files will be always saved in the USB Disk.
- When "Enable Auto Change" is enabled, the option "**Storage Path Priority**"will appear. It allows the user to configure the priority of each storage unit in the priority list (The storage on top of the list has the highest priority). The default priority list is GDMS Cloud Storage > NAS > USB 1 > USB > SD Card > Local
- If "**Local**" is selected, the files will be stored in the internal storage. If a storage drive is inserted, the IPPBX will store the files into the storage drive instead of internal storage. Priority list is USB drive, then SD Card.
- If "**GDMS Cloud Storage**" is selected, data will no longer be stored locally and if you need to listen to the recording, download the file to the computer side and play it offline.

Once "USB Disk" or "SD Card" is selected, click on "OK". The user will be prompted to confirm to copy the local files to the external storage device.



Recordings Storage Prompt Information

Click on "OK" to continue. The users will be prompted a new dialog to select the categories for the files to be copied over.



Recording Storage Category

Once a storage device has filled up, the IPPBX will choose the next available storage device based on the Storage Path Priority.

On the IPPBX, users have the following options when select the categories to copy the files to the external device:

- **Recording Files**: Copy the normal recording files to the external device.
- **Conference**: Copy the conference recording files to the external device.
- **Queue**: Copy the call queue recording files to the external device.
- **All**: Copy all recording files to the external device.

Storage Device Management

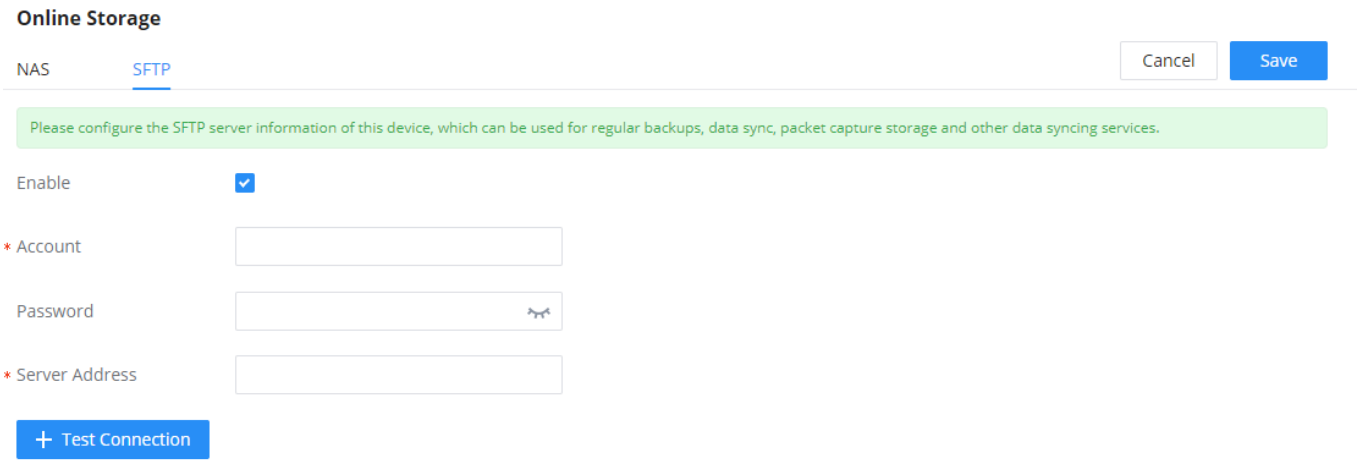
NAS

The IPPBX supports adding and backing up recordings to a network-attached storage (NAS) server. Following table describes NAS settings:

Enable	Enabled / Disable the NAS recording functionality.
Host	Configure the Domain or IP address of the NAS server.Note: Currently, only IP addresses are supported in the Host/IP field.
Share Name	Specify the name of the shared folder.
Username	Specify the account username to access the NAS server.
Password	Configure the account password to access the NAS server. The password can include letters, number, and special characters.
Security Mode	Select a security mode based on the server settings to ensure proper connection establishment. The default value is ntlmssp.
Status	If configured correctly, the Status field will show "Mounted", and the newly added NAS server will be shown on the Mounted Netdisk List. Additionally, the NAS will appear as a selectable storage option in the PBX Settings→Recording Storage page and CDR→Recording Files page.

SFTP

SFTP integration offers sychronizing files between from the IPPBX to the SFTP server. SFTP is used to synchronize the following data: CDR Records, Recording Files, Voicemail, and Fax.



SFTP

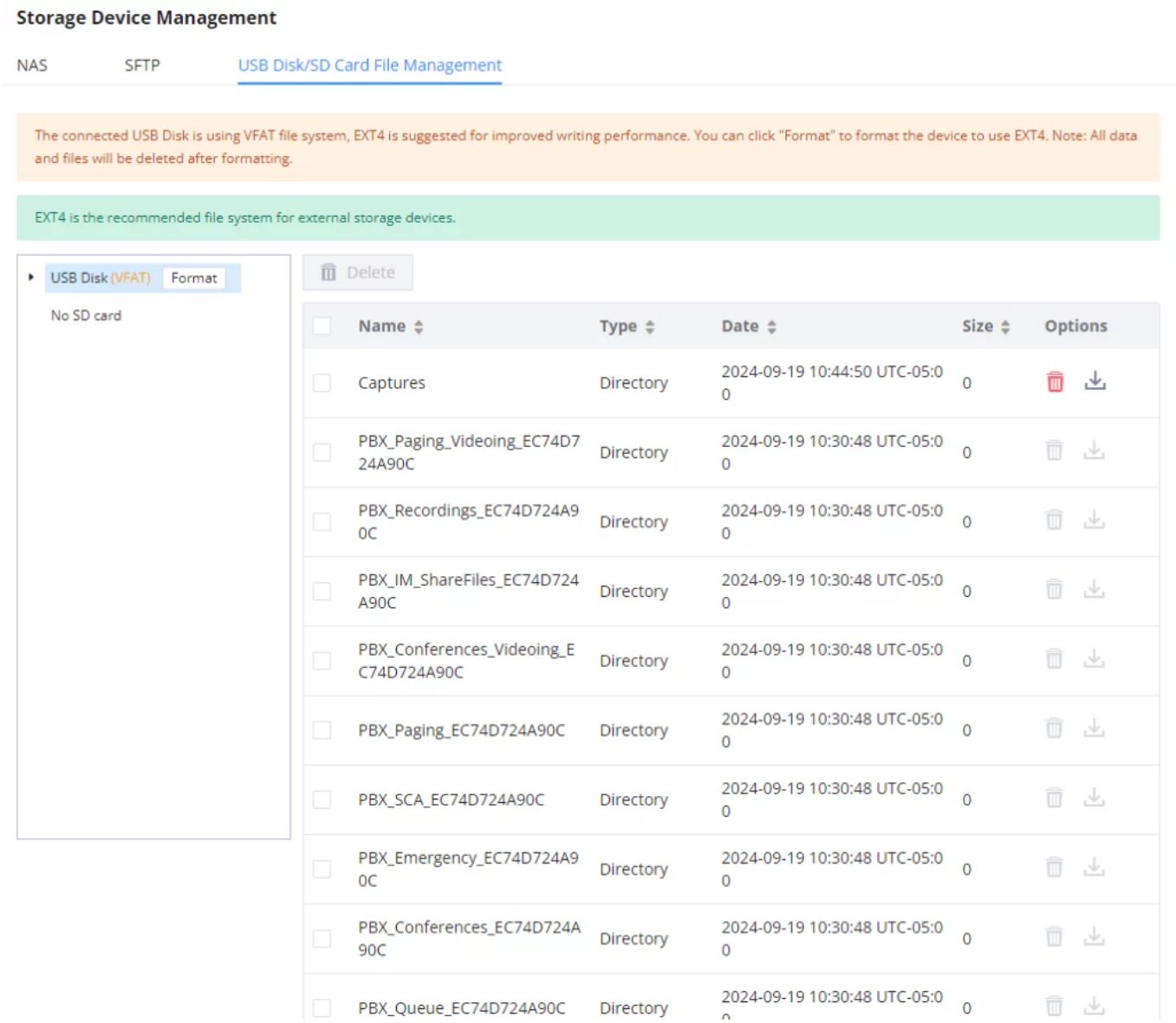
Enable	Tick this box to enable SFTP integration.
Account	Enter the account of the SFTP server.
Password	Enter the password of the account of the SFTP server.
Server Address	Enter the address of the SFTP server.

USB Disk/SD Card File Management

This feature allows to browse the SD cards and USB devices which are connected to the GCC device. The user can view all the files in the storage unit. The user can use the button to delete the file or the button to download the file. Please note that file directories cannot be deteled or downloaded.

To format an attached storage unit, you can click on the "Format" button. Please note that this will wipe all the

data stored and it cannot be recovered



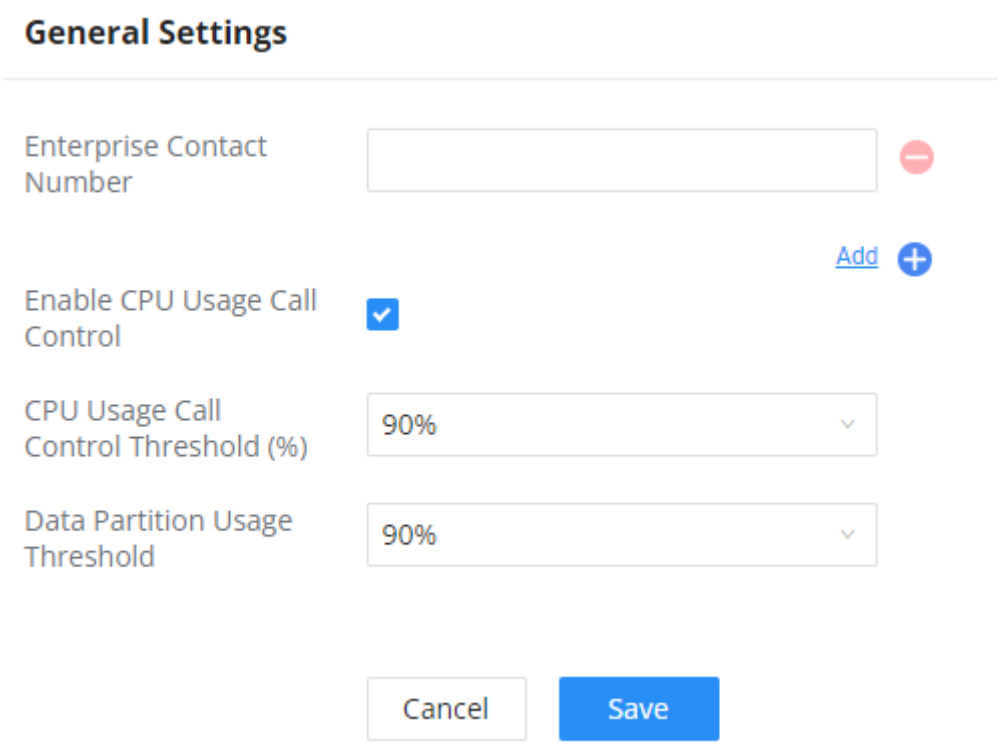
USB DiskSD Card File Management

SYSTEM SETTINGS

This section will explain the available system-wide parameters and configuration options on the IPPBX. This includes settings for the following items: General Settings, HTTP server, Security Settings, LDAP Server, Time Settings, and Email Settings.

General Settings

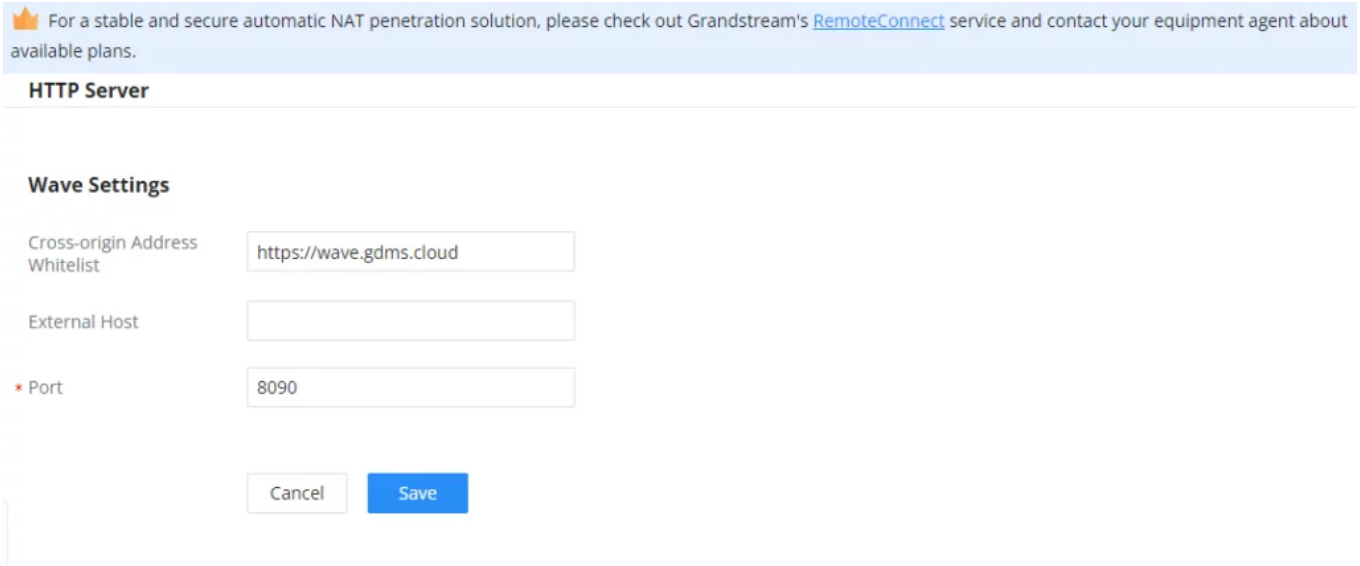
On general settings, the user can configure the **Device Name** and **Enterprise Contact Number**.



IPBBX General Settings

HTTP Server

In this page the user can set setting which are related to Wave Desktop and Web access.



HTTP Server Settings

Parameter	Description
Cross-origin Address Allowlist	The PBX will accept cross-server requests from addresses in the allowlist, which should be formatted as https://domain, https://ip:port or *. Entering * will allow cross-server requests from all addresses.
External Host	Configure a URL and port (optional) used to access the PBX web portal or a public link to the video meeting room if the PBX is behind NAT.
Port	The port to access Wave Web and Wave Mobile. If behind NAT, please make sure to map the external port to this port.

DDNS Settings

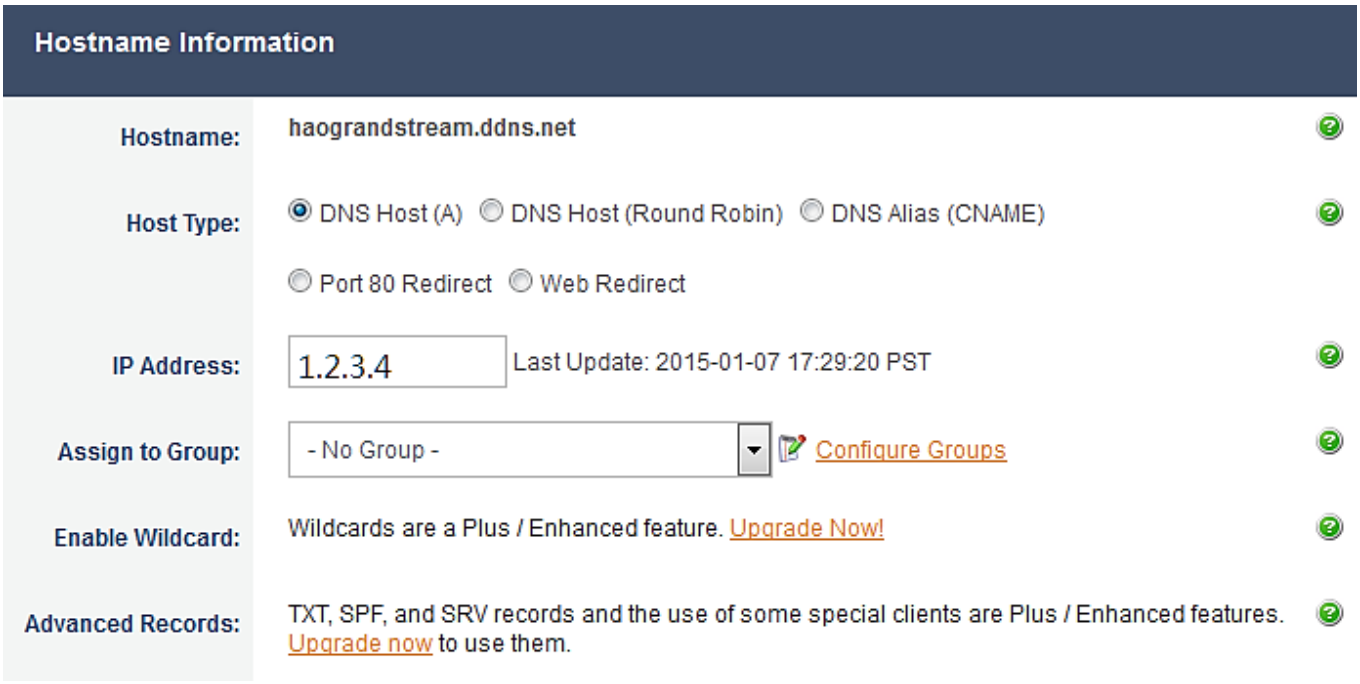
The DDNS (Dynamic DNS) setting allows users to access the GCC IPPBX system using a domain name instead of an IP address. This is particularly useful when the IPPBX is deployed in environments with dynamic IPs, enabling consistent remote access without needing to manually update IP address changes.

The GCC IPPBX supports Dynamic DNS services from the following providers:

- **dyndns.org**
- **no-ip.com**
- **freedns.afraid.org**
- **zoneedit.com**
- **oray.net**

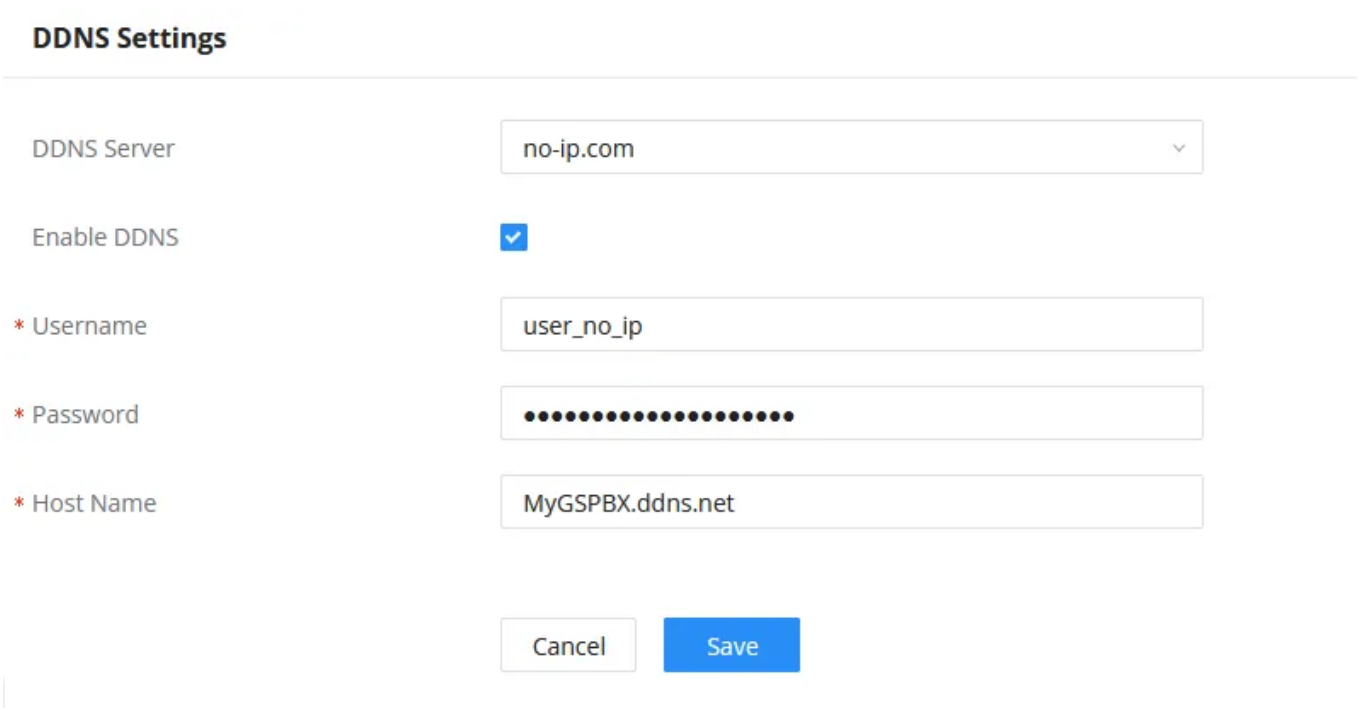
Below is an example of using noip.com for DDNS.

1. Register domain in DDNS service provider. Please note the GCC IP-PBX needs to have public IP access.



Register Domain Name on noipcom

2. On Web GUI→**System Settings→Network Settings→DDNS Settings,** enable DDNS service and configure username, password, and host name.



DDNS Settings

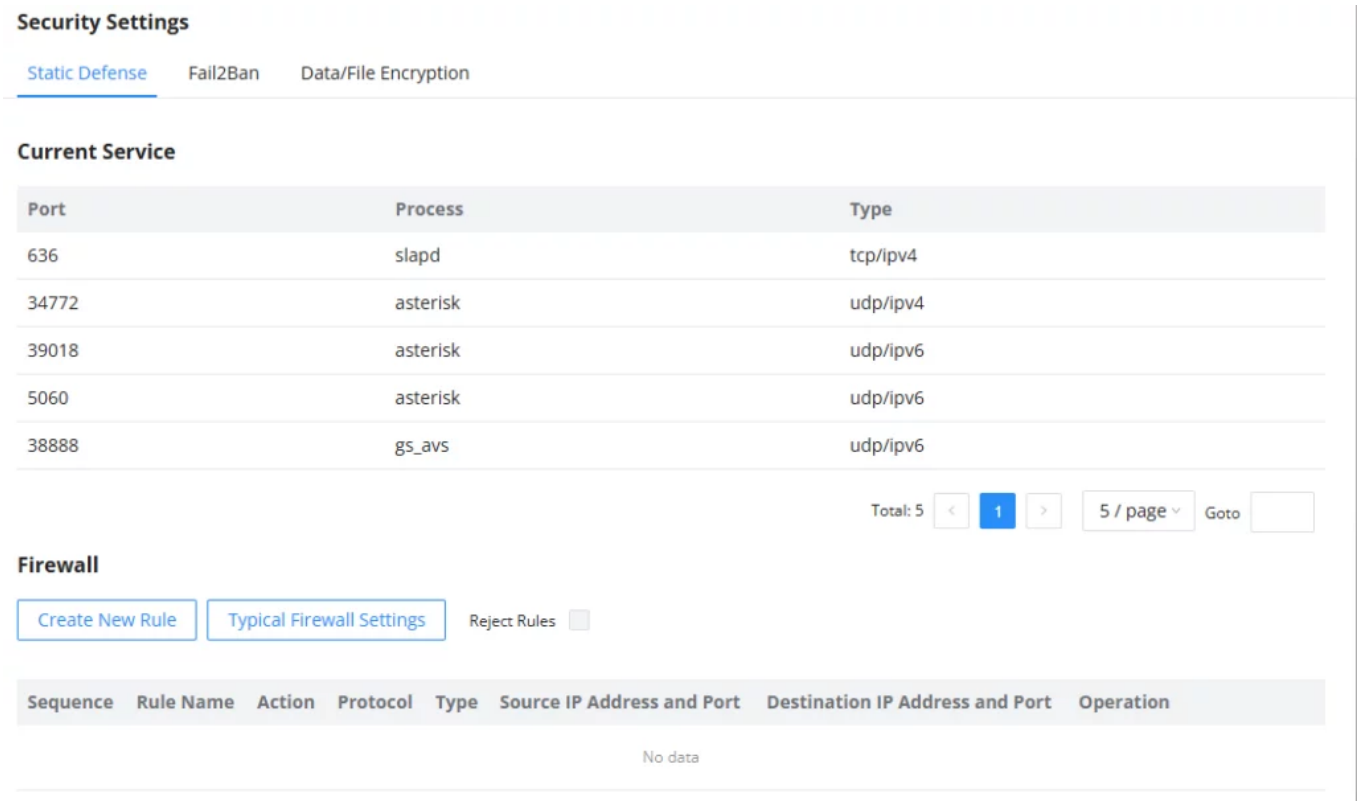
Security Setting

The GCC IPPBX provides users firewall security configurations to prevent certain malicious attacks to the GCC IPPBX system. Users could configure to allow, restrict, or reject specific traffic through the device for security and bandwidth purpose. The GCC IPPBX also provides the Fail2ban feature for authentication errors in SIP REGISTER, INVITE and SUBSCRIBE. To configure firewall settings in the GCC IPPBX, go to Web GUI→**System Settings**.

In this section, the user can configure many mechanisms to increase the security of the PBX.

Static Defense

In this page, the user can set static firewall rules which prevent unauthorized access to the services running on the PBX.



Static Defense

Under **System Settings** > **Security Settings** > **Static Defense** page, users will see the following information:

- Current service information with port, process, and type.
- Typical firewall settings.
- Custom firewall settings.

The following table shows a sample current service status running on the IPPBX.

Port	Process	Type	Protocol or Service
7777	Asterisk	TCP/IPv4	SIP
389	Slapd	TCP/IPv4	LDAP
6060	zero_config	UDP/IPv4	IPPBX zero_config service
5060	Asterisk	UDP/IPv4	SIP
4569	Asterisk	UDP/IPv4	SIP
38563	Asterisk	udp/ipv4	SIP
10000	gs_avs	udp/ipv4	gs_avs
10001	gs_avs	udp/ipv4	gs_avs
10002	gs_avs	udp/ipv4	gs_avs
10003	gs_avs	udp/ipv4	gs_avs
10004	gs_avs	udp/ipv4	gs_avs
10005	gs_avs	udp/ipv4	gs_avs
10006	gs_avs	udp/ipv4	gs_avs
10007	gs_avs	udp/ipv4	gs_avs
10010	gs_avs	udp/ipv4	gs_avs

10012	gs_avs	udp/ipv4	gs_avs
10013	gs_avs	udp/ipv4	gs_avs
10014	gs_avs	udp/ipv4	gs_avs
10015	gs_avs	udp/ipv4	gs_avs
10018	gs_avs	udp/ipv4	gs_avs
10019	gs_avs	udp/ipv4	gs_avs
10020	gs_avs	udp/ipv4	gs_avs
6066	Python	udp/ipv4	python
3306	Mysqld	tcp/ipv4	mysqld
45678	Python	udp/ipv4	python
8439	Lighttpd	tcp/ipv4	HTTP
8088	asterisk	tcp/ipv4	SIP
8888	Pbxmid	tcp/ipv4	pbxmid
25	Master	tcp/ipv4	master
636	Slapd	tcp/ipv4	SLDAP
4569	asterisk	udp/ipv6	SIP
42050	asterisk	udp/ipv6	SIP
7681	Pbxmid	tcp/ipv4	pbxmid

IPPBX Firewall→Static Defense→Current Service

For typical firewall settings, users could configure the following options on the IPPBX.

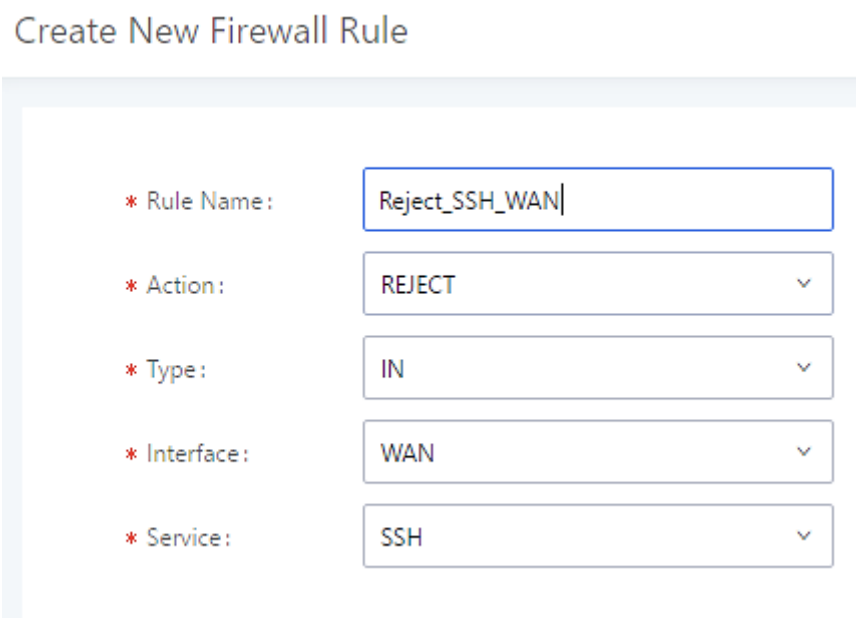
Ping Defense Enable	If enabled, ICMP response will not be allowed for Ping requests. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (IPPBX) interface.
SYN-Flood Defense Enable	Allows the IPPBX to handle excessive amounts of SYN packets from one source and keep the web portal accessible. There are two options available and only one of these options may be enabled at one time. eth(0)LAN defends against attacks directed to the LAN IP address of the IPPBX. eth(1)WAN defends against attacks directed to the WAN IP address of the IPPBX. SYN Flood Defense will limit the amount of SYN packets accepted by the IPPBX from one source to 10 packets per second. Any excess packets from that source will be discarded.
Ping-of-Death Defense Enable	Enable to prevent Ping-of-Death attack to the device. The default setting is disabled. To enable or disable it, click on the check box for the LAN or WAN (IPPBX) interface.

Typical Firewall Settings

Under "Custom Firewall Settings", users could create new rules to accept, reject or drop certain traffic going through the IPPBX. To create a new rule, click on the "Create New Rule" button and a new window will pop up for users to specify rule options.

Right next to the "Create New Rule" button, there is a checkbox for the option "Reject Rules". If it is checked, all the rules will be rejected except the firewall rules listed below. In the firewall rules, only when there is a rule that meets all the following requirements, the option "Reject Rules" will be allowed to check:

- Action: "Accept"
- Type "In"
- The destination port is set to the system login port (e.g., by default 8089)
- The protocol is not UDP



Create New Firewall Rule

Rule Name	Specify the Firewall rule name to identify the firewall rule.
Action	Select the action for the Firewall to perform. ACCEPT REJECT DROP
Type	Select the traffic type. IN If selected, users will need to specify the network interface "LAN" or "WAN" (for IPPBX) for the incoming traffic. OUT
Interface	Select the interface to use the Firewall rule.
Service	Select the service type. FTP SSH Telnet HTTP LDAP Custom If "Custom" is selected, users will need to specify Source (IP and port), Destination (IP and port), and Protocol (TCP, UDP, or Both) for the service. Please note if the source or the destination field is left blank, it will be used as "Anywhere".
Source IP Address and Port	Configure a source subnet and port. If set to "Anywhere" or left empty, traffic from all addresses and ports will be accepted. A single port or a range of ports can be specified (e.g., 10000, 10000-20000).
Destination IP Address and Port	Configure a destination subnet and port. If set to "Anywhere" or left empty, traffic can be sent to all addresses and ports. A single port or a range of ports can be specified (e.g., 10000, 10000-20000).
Protocol	Select the protocol for the rule to be used.

Save the change and click on the "Apply" button. Then submit the configuration by clicking on "Apply Changes" on the upper right of the web page. The new rule will be listed at the bottom of the page with sequence number, rule name, action, protocol, type, source, destination, and operation. More operations are below:

- Click on

  to edit the rule.

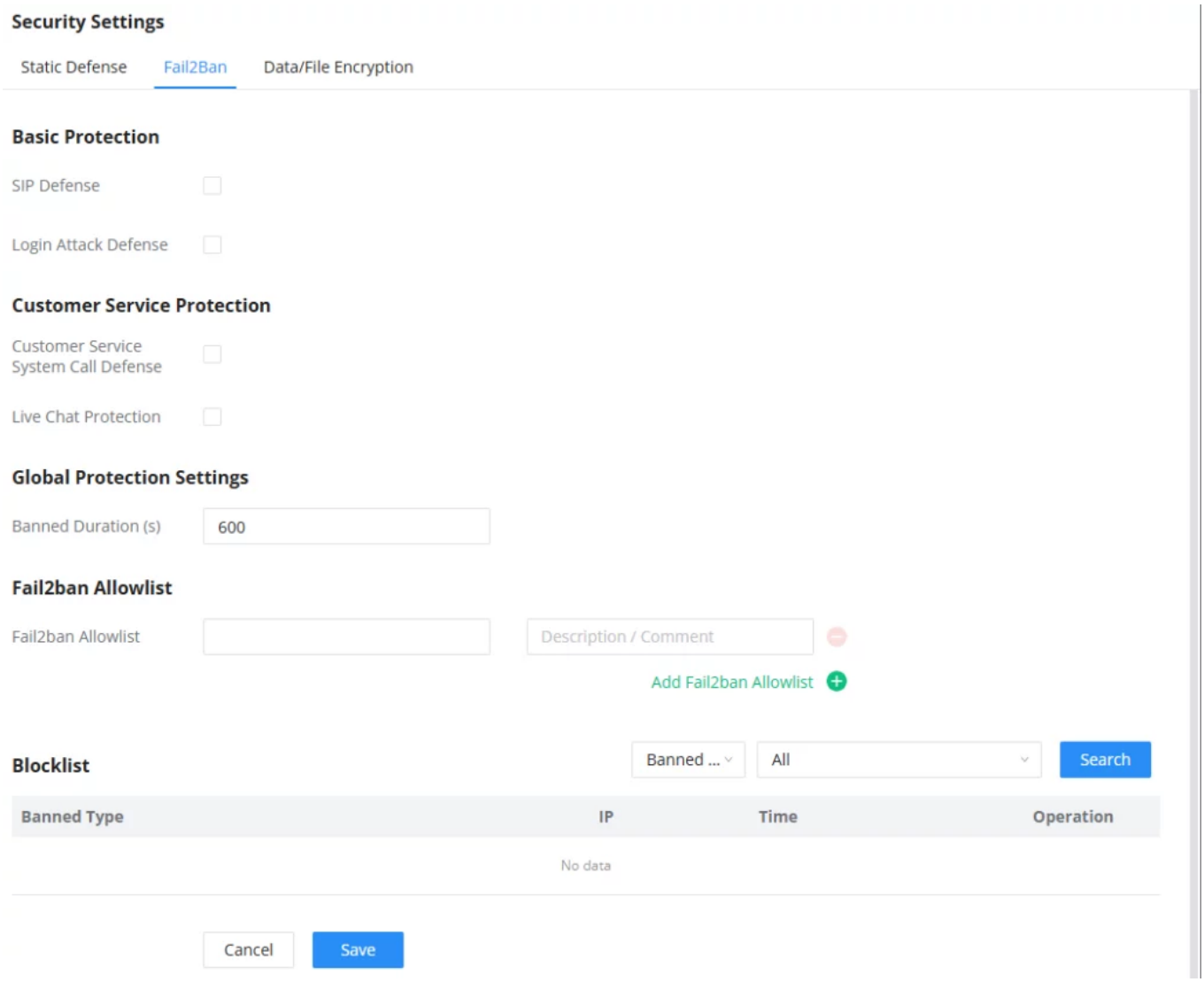
- Click on

  to delete the rule.

Fail2Ban

Fail2Ban feature allows to control the attempts to register to the IPPBX using a SIP endpoint, it allows a set number of attempts. If the client fails to provide the correct authentication ID and the authentication password, the client's IP address will be blacklist for a time duration which can be configured on the same page.

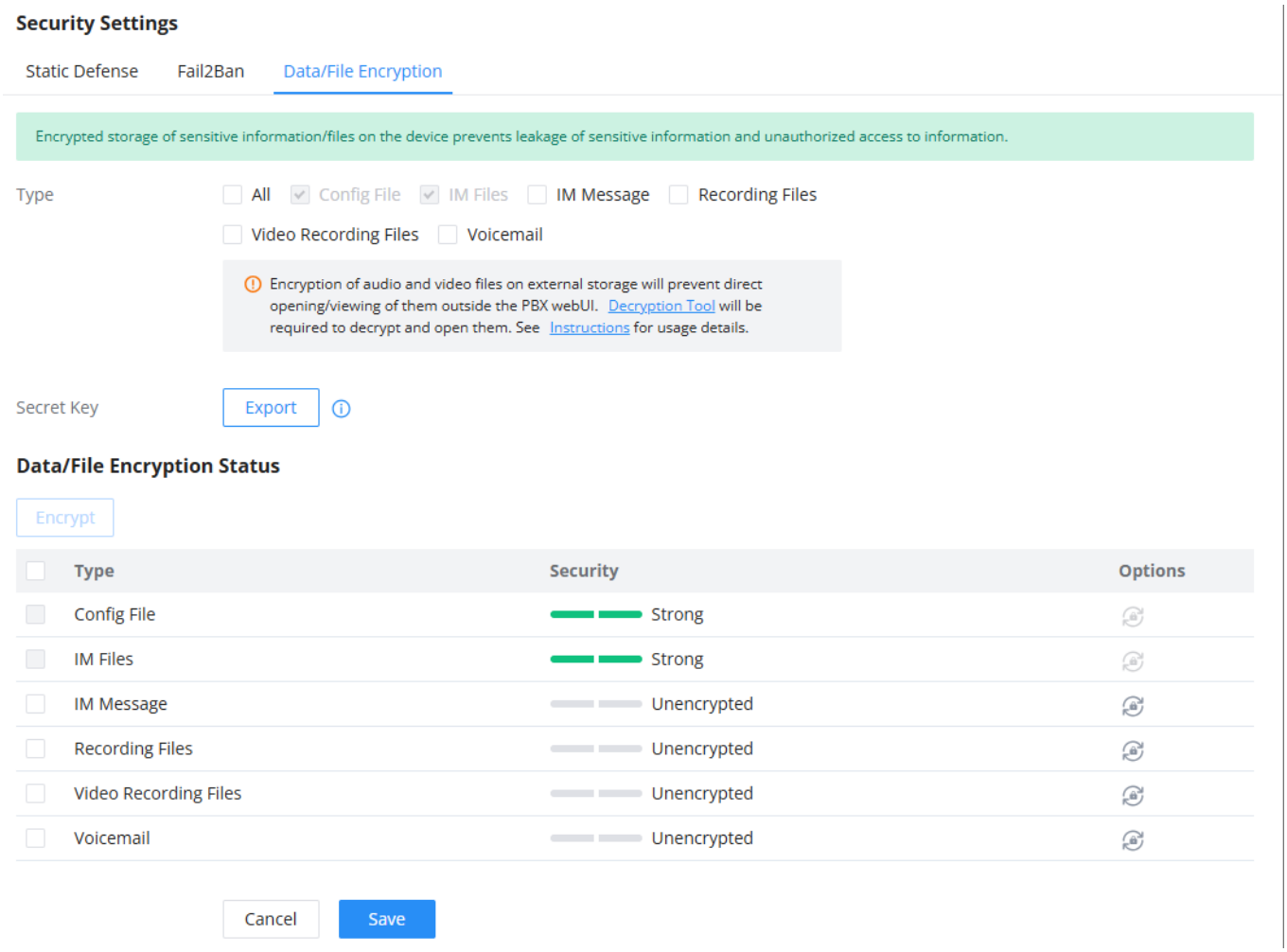
Fail2ban also protects the repetitive login attempts which target the web server of the device.



Fail2Ban

Data/File Encryption

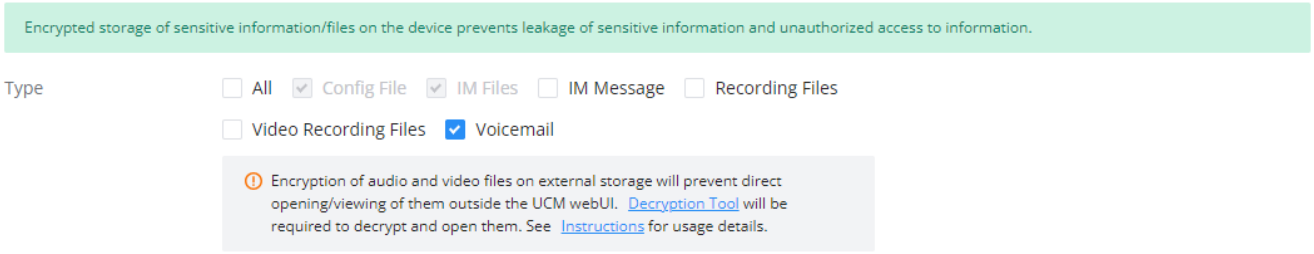
The GCC PBX module offers encryption for your data stored on the internal storage space of the PBX or on the attached storage units (e.g., SD Card, USB HDD/SSD, USB Flash Drive). This feature will harden the security and confidentiality of your data, which renders it impossible to read data in case they end up acquired by unauthorized parties. To access this feature, please navigate to **PBX Settings > Security Settings > Data/File Encryption**.



DataFile Encryption

Enabling Data/File Encryption

To choose which files are going to encrypted, use the following settings to enable the encryption for specific types of data.



File/Data Type

Once the user chooses which data to encrypt, the user can save the setting. This means that any data created after this configuration will be encrypted, however, all the data which has been created prior to this configuration is not encrypted. To encrypt it please see the sections below.

Encrypting Files

When the user click on "Export" button, a CSV file which contains the secret key will be downloaded. This key can be stored safely and used to decrypt the data later on.

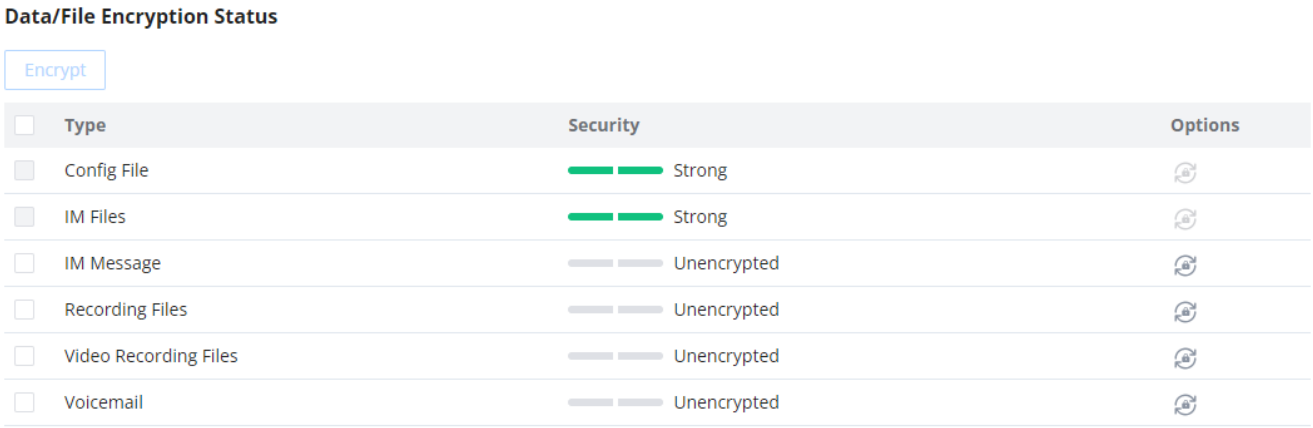
Secret Key Export/Update

You can update the secret key using the "Update" button, this would require entering the administrator's password to allow the user to change the secret key.

Please note that only the super administrator can change the secret key.

If the key has been updated after it has been exported, this will render the old key obsolete and the user will not be able to decrypt the data. We highly recommend that you do not change the encryption key after it has been backed up. In case you need to change the encryption key, please make sure to export the new one and store it safely.

In the following section, the user can view the status of the files, whether they are encrypted or not. The user can also select the different data types to encrypt. This will encrypt the existing data.



Data/File Encryption Status

The user can select the types of stored files to encrypt, then use the button to start the encryption. Or you can encrypt the files using the following encrypting button .

To open and play encrypted audio/video files outside of the PBX system, users need a decryption tool (found on the Grandstream Tools page), a key file and the password for it (set when exporting the key file).

For more information on how to use the decryption tool, please refer to the following guide :

IP PBX - CDR Tool Guide

- In an HA environment, the Primary PBX's key will be used as the Master Key, and this will be synced with the Secondary PBX so both PBXs will use the same key to encrypt and decrypt files.
- Files on the standby machine are not automatically encrypted so users will need to manually re-encrypt all new files that were created before the failover.

- Downgrading from a firmware version with data encrypted to a firmware version which does not support

data encryption will require the user to perform a factory reset.

- Backing up the data of the PBX before upgrading the firmware is highly recommended to avoid the possibility of data corruption/loss.

LDAP Server

The IPPBX has an embedded LDAP/LDAPS server for users to manage the corporate phonebook in a centralized manner.

- By default, the LDAP server has generated the first phonebook with **PBX DN** "ou=pbx,dc=pbx,dc=com" based on the IPPBX user extensions already.
- Users could add new phonebook with a different **Phonebook DN** for other external contacts. For example, "ou=people,dc=pbx,dc=com".
- All the phonebooks in the IPPBX LDAP server have the same **Base DN** "dc=pbx,dc=com".

Term Explanation:

cn= Common Name

ou= Organization Unit

dc= Domain Component

These are all parts of the LDAP Data Interchange Format, according to RFC 2849, which is how the LDAP tree is filtered.

If users have the Grandstream phone provisioned by the IPPBX, the LDAP directory will be set up on the phone and can be used right away for users to access all phonebooks.

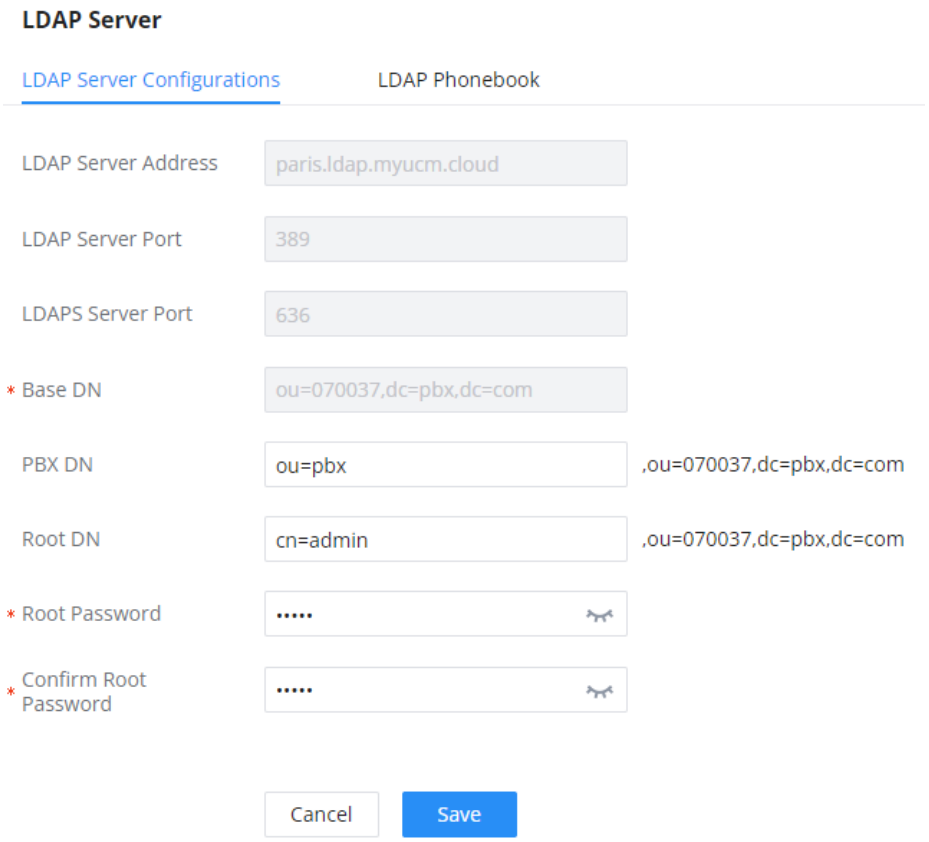
Additionally, users could manually configure the LDAP client settings to manipulate the built-in LDAP server on the IPPBX. If the IPPBX has multiple LDAP phonebooks created, in the LDAP client configuration, users could use "dc=pbx,dc=com" as Base DN to have access to all phonebooks on the IPPBX LDAP server, or use a specific phonebook DN, for example "ou=people,dc=pbx,dc=com", to access to phonebook with Phonebook DN "ou=people,dc=pbx,dc=com " only.

IPPBX can also act as an LDAP client to download phonebook entries from another LDAP server.

To access the LDAP server and client settings, go to Web GUI→**Settings→LDAP Server**.

LDAP Server Configurations

The following figure shows the default LDAP server configurations on the IPPBX.

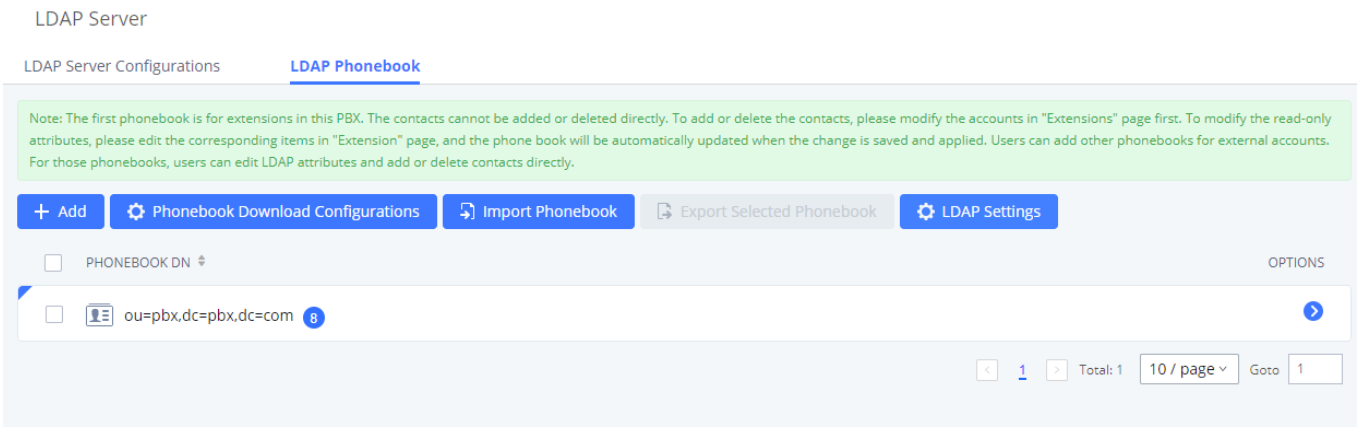


LDAP Server Configurations

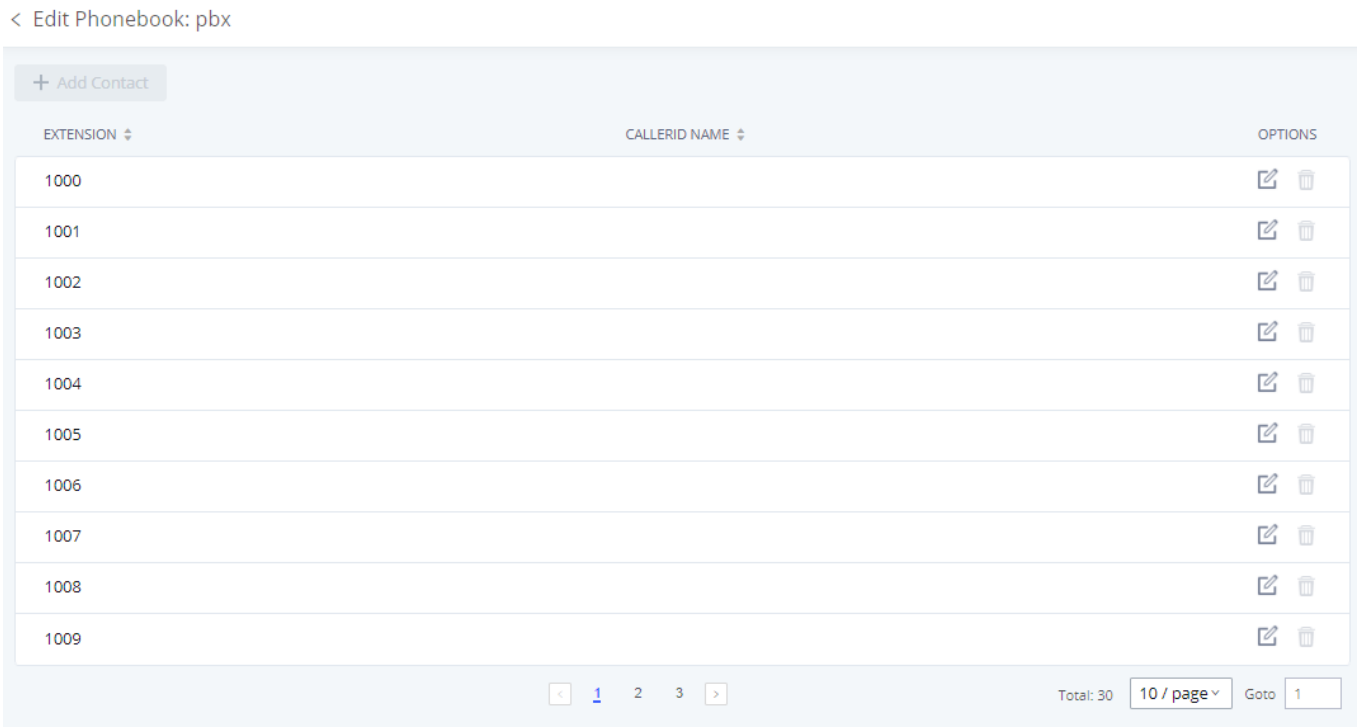
The IPPBX LDAP server supports anonymous access (read-only) by default. Therefore, the LDAP client does not have to configure a username and password to access the phonebook directory. The "Root DN" and "Root Password" here are for LDAP management and configuration where users will need to provide for authentication purposes before modifying the LDAP information.

The default phonebook list in this LDAP server can be viewed and edited by clicking on/for the first phonebook under LDAP Phonebook.

The IPPBX supports secure LDAP (LDAPS) where the communication is encrypted and secure.



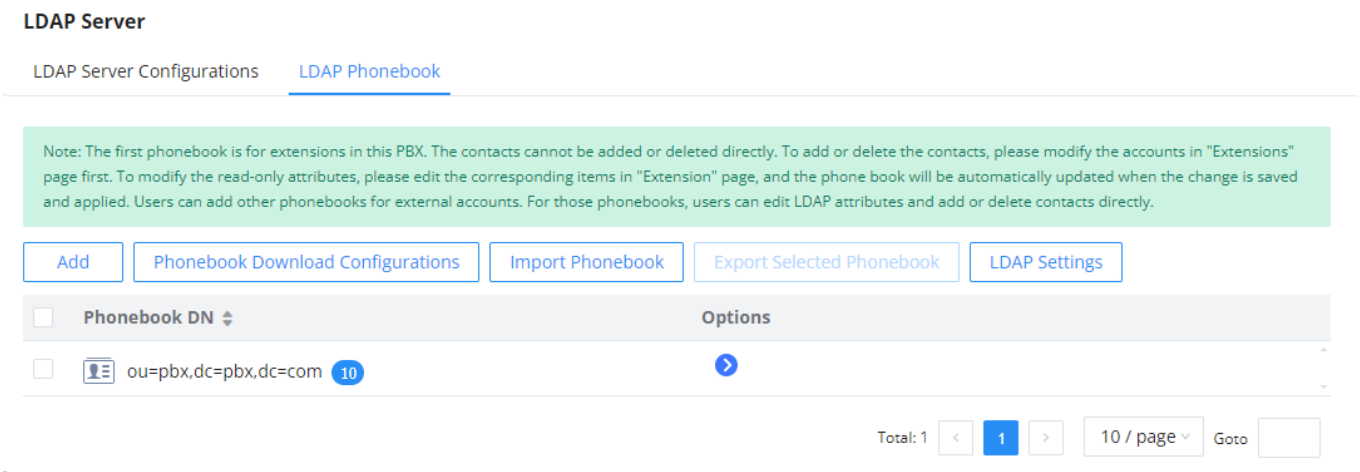
Default LDAP Phonebook DN



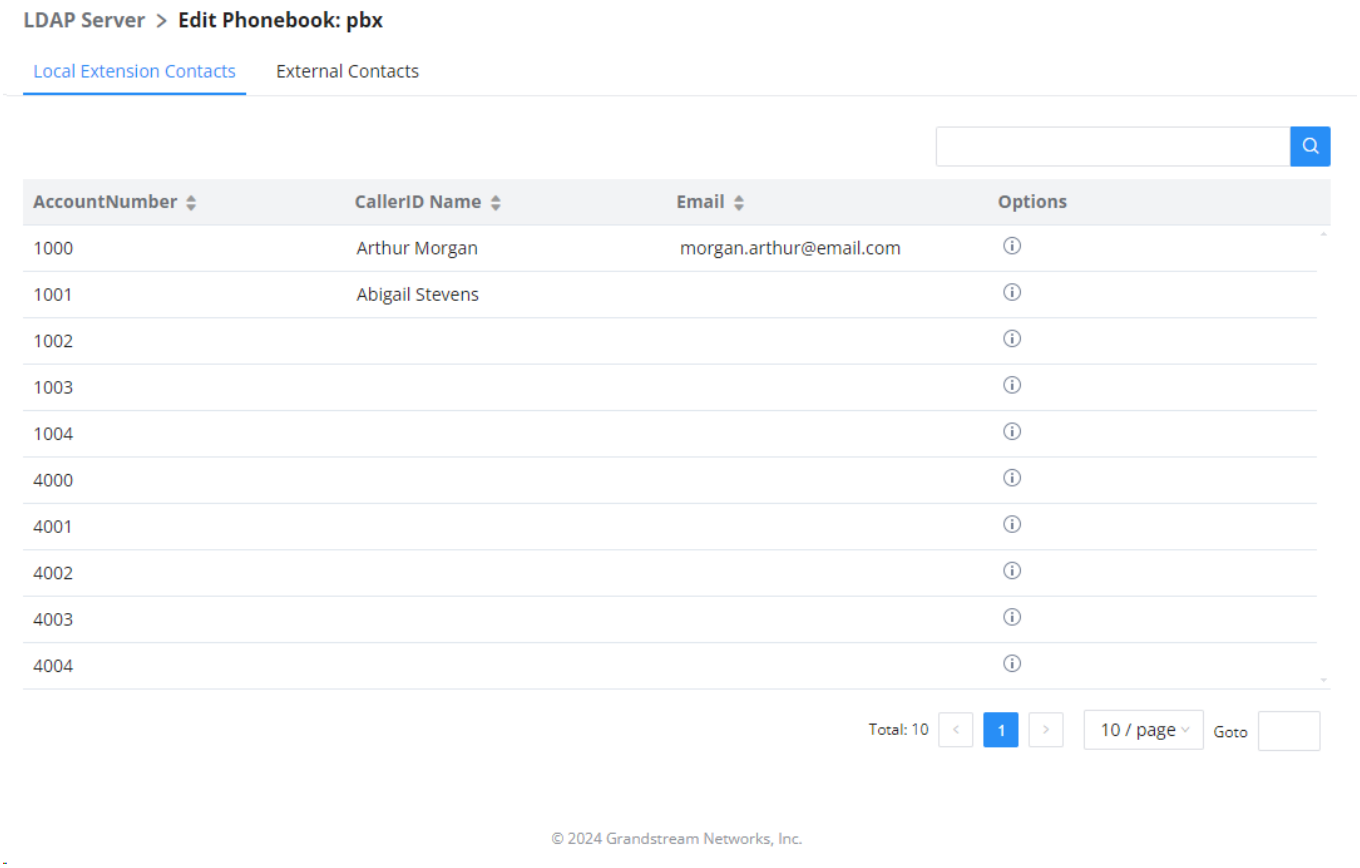
Default LDAP Phonebook Attributes

LDAP Phonebook

Users could use the default phonebook, edit the default phonebook, add new phonebook, import phonebook on the LDAP server as well as export phonebook from the LDAP server. The first phonebook with default phonebook dn "ou=pbx,dc=pbx,dc=com" displayed on the LDAP server page is for extensions in this PBX. Users cannot add or delete contacts directly. The contacts information will need to be modified via Web GUI→**Extension/Trunk→Extensions** first. The default LDAP phonebook will then be updated automatically.

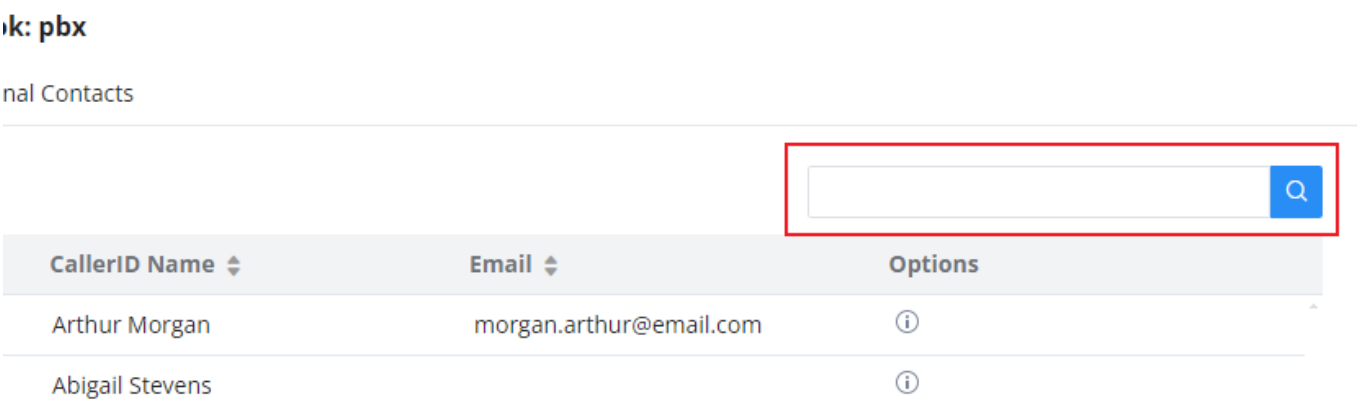


LDAP Server→LDAP Phonebook



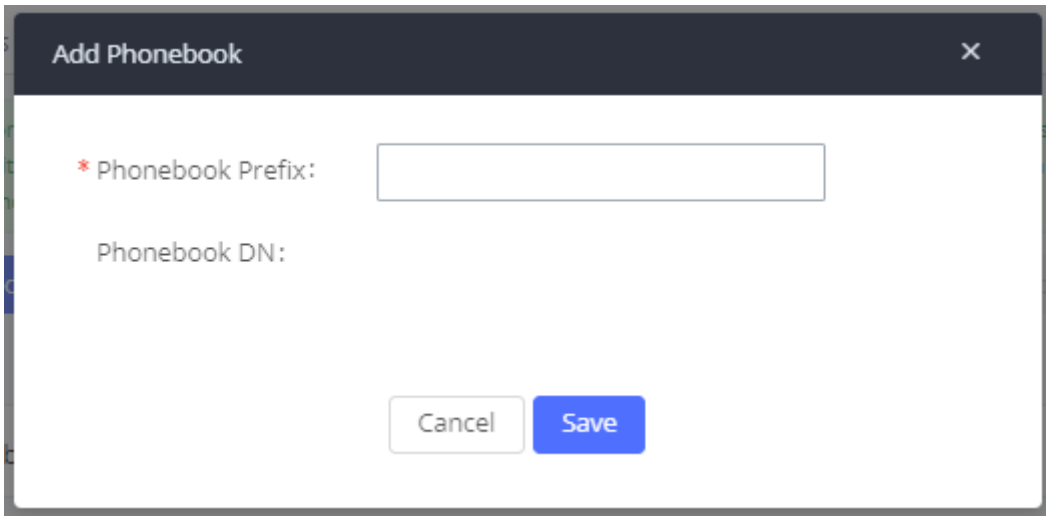
The user can use the search bar on the top right corner of page, please see the screenshot below.

You can enter the extension number or the name of the contact; if you are unsure about the name or the extension, the search feature supports fuzzy matching.



- **Add new phonebook**

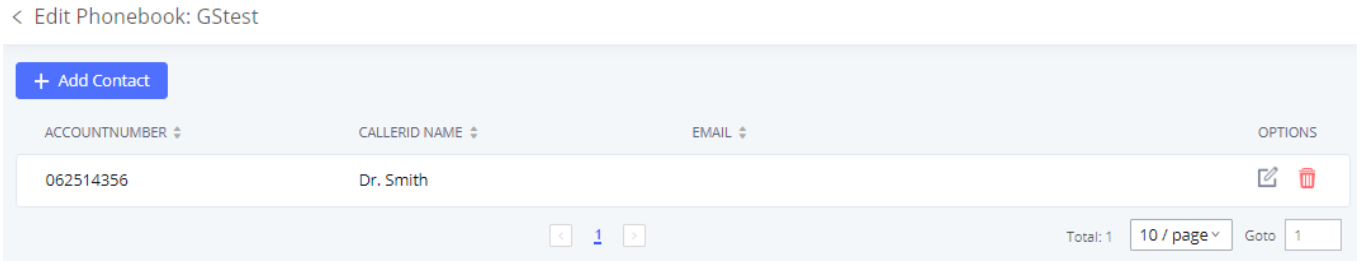
A new sibling phonebook of the default PBX phonebook can be added by clicking on "Add" under "LDAP Phonebook" section.



Add LDAP Phonebook

Configure the "Phonebook Prefix" first. The "Phonebook DN" will be automatically filled in. For example, if configuring "Phonebook Prefix" as "people", the "Phonebook DN" will be filled with "ou=people,dc=pbx,dc=com".

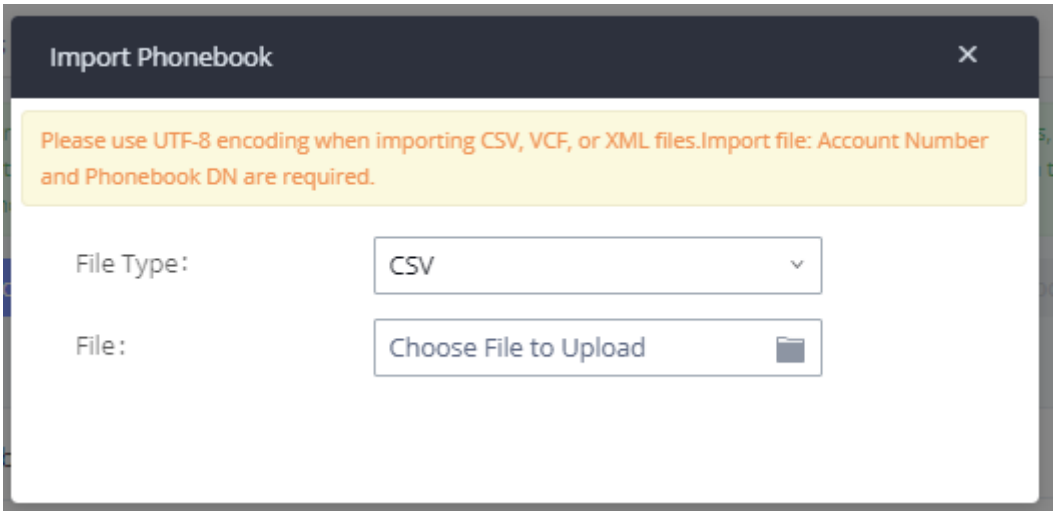
Once added, users can select to edit the phonebook attributes and contact list (see figure below) or select to delete the phonebook.



Edit LDAP Phonebook

- **Import phonebook from your computer to LDAP server**

Click on "Import Phonebook" and a dialog will prompt as shown in the figure below.



Import Phonebook

The file to be imported must be a CSV, VCF or XML file with UTF-8 encoding. Users can open the file with Notepad and save it with UTF-8 encoding.

Here is how a sample file looks like. Please note "Account Number" and "Phonebook DN" fields are required. Users could export a phonebook file from the IPPBX LDAP phonebook section first and use it as a sample to start with.

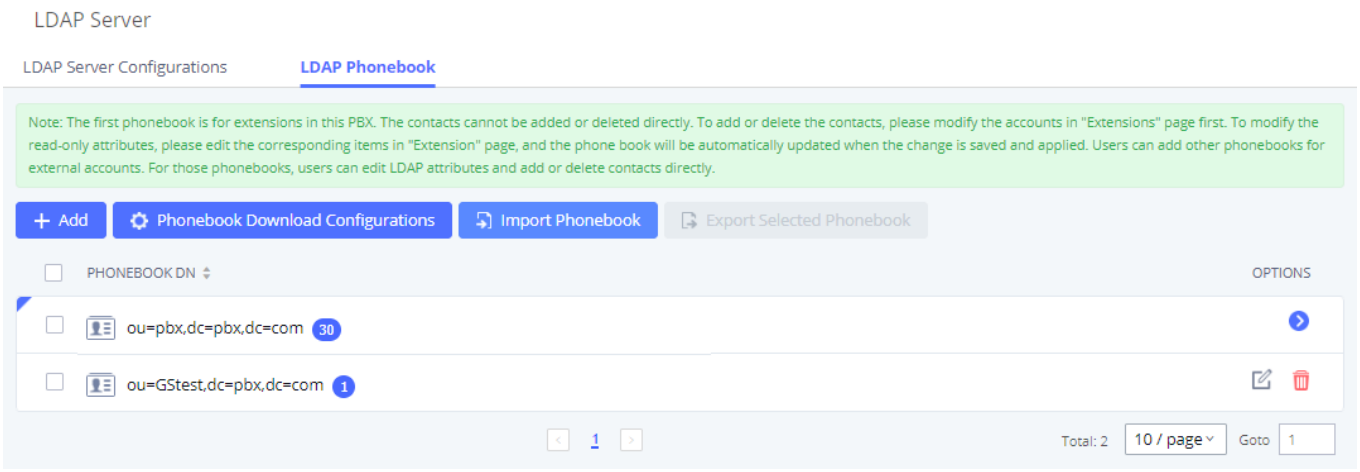
	A	B	C	D	E	F	G	H	I	J
1	First Nam	Last Name	Account Number	CallerID Name	Email	Department	Mobile Number	Home Number	Fax	Phonebook DN
2	John	Doe	1001	1001		IT	1001000000			phonebook
3	Jane	Doe	1002	1002		Sales	1002000000			phonebook
4	William	Chung	1003	1003		Marketing	1003000000			phonebook
5	Linda	Kuo	1004	1004		Accounting	1004000000			phonebook
6	Steve	Chang	1005	1005		Support	1005000000			others

Phonebook CSV File Format

The Phonebook DN field is the same "Phonebook Prefix" entry as when the user clicks on "Add" to create a new phonebook. Therefore, if the user enters "phonebook" in "Phonebook DN" field in the CSV file, the actual phonebook DN "ou=phonebook,dc=pbx,dc=com" will be automatically created by the IPPBX once the CSV file is imported.

In the CSV file, users can specify different phonebook DN fields for different contacts. If the phonebook DN already exists on the IPPBX LDAP Phonebook, the contacts in the CSV file will be added into the existing phonebook. If the phonebook DN does not exist on the IPPBX LDAP Phonebook, a new phonebook with this phonebook DN will be created.

The sample phonebook CSV file in above picture will result in the following LDAP phonebook in the IPPBX.

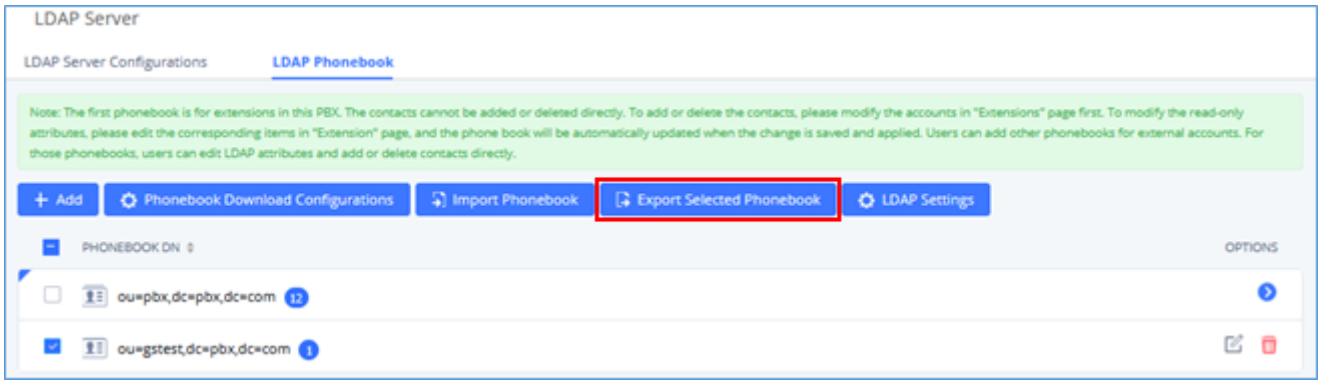


LDAP Phonebook After Import

As the default LDAP phonebook with DN "ou=pbx,dc=pbx,dc=com" cannot be edited or deleted in LDAP phonebook section, users cannot import contacts with Phonebook DN field "pbx" if existed in the CSV file.

- **Export phonebook to your computer from the IPPBX LDAP server**

Select the checkbox for the LDAP phonebook and then click on "Export Selected Phonebook" to export the selected phonebook. The exported phonebook can be used as a record or a sample CSV, VFC or XML file for the users to add more contacts in it and import to the IPPBX again.



Export Selected LDAP Phonebook

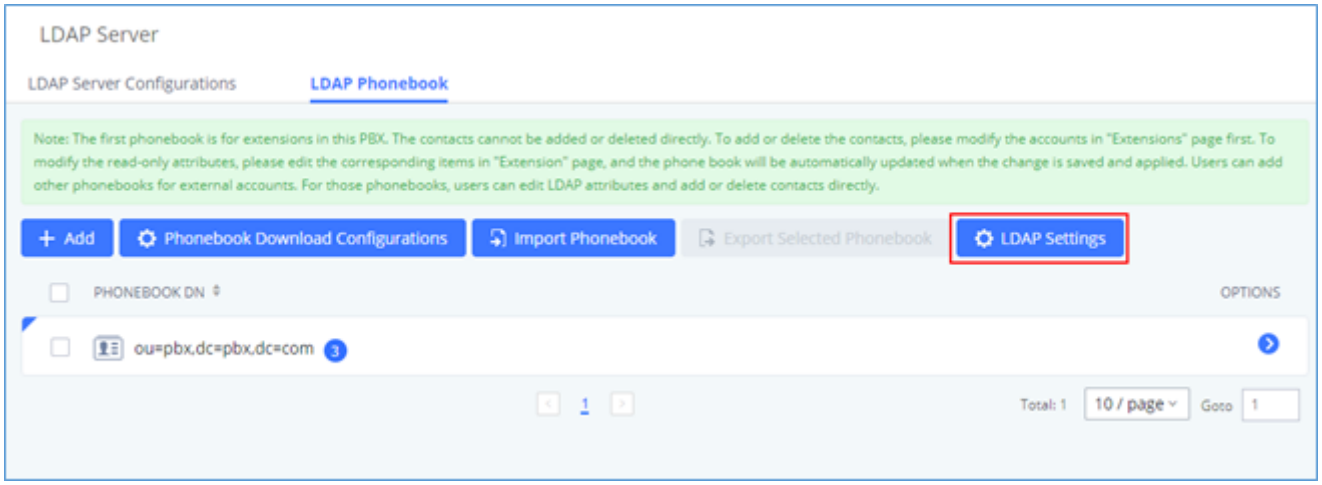
LDAP Settings

Prerequisites to support contacts sync-up to IP Phones, IPPBX needs to support the following:

1. If Cloud IM is enabled, IPPBX can send remote IPPBX's contacts to each end device.
2. Contacts from remote IPPBX can be synced by Cloud IM or LDAP sync via trunk. The contacts data must be complete and consistent.
3. If Cloud IM is enabled, the contacts sent from IPPBX to end device should integrate Cloud IM contacts.
4. If Cloud IM is disabled, the contacts sent from IPPBX to end device should only contain contacts on the IPPBX.

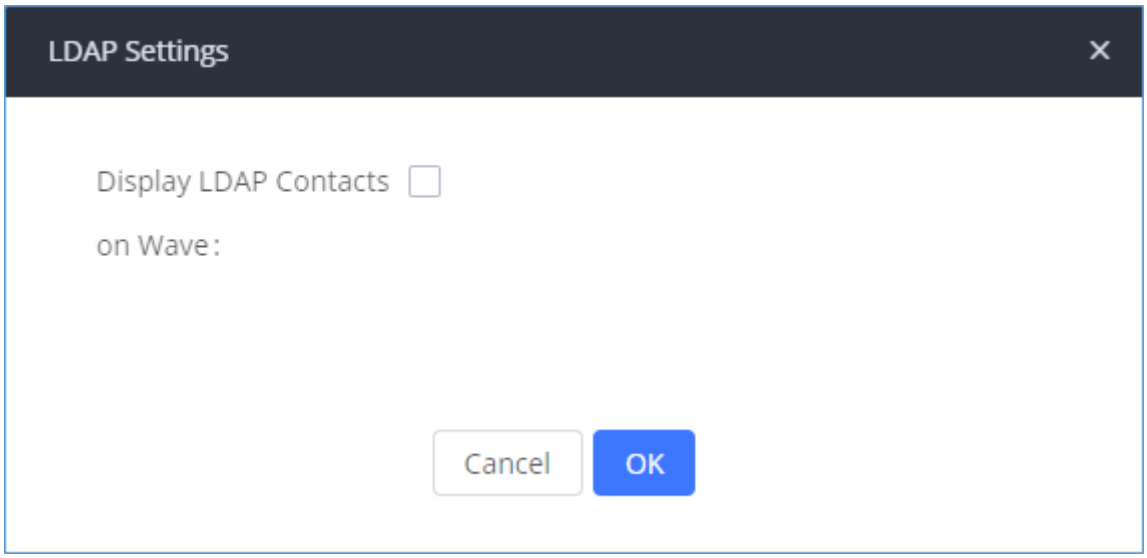
To support contacts sync-up to Wave, it allows Wave to obtain enterprise contacts from Cloud IM or LDAP. On IPPBX SIP peer trunk, if LDAP sync is enabled, end point can obtain remote IPPBX extensions' info via LDAP. Also, it will allow configuring whether to sync up LDAP contacts on Wave so that Wave doesn't receive duplicate contacts info.

Under IPPBX **webUI→ System Settings→ LDAP Server**, click on "LDAP Settings", option "Wave enable LDAP phonebook" is available for configuration. If enabled, all Wave users on this IPPBX will display LDAP contacts. Otherwise, it will not display.



LDAP Settings

Please note the LDAP contacts displayed on Wave will exclude the duplicate contacts from Cloud IM.



Display LDAP Contacts on Wave

LDAP Client Configuration

The configuration on LDAP client is useful when you use other LDAP servers. Here we provide an example on how to configure the LDAP client on the IPPBX.

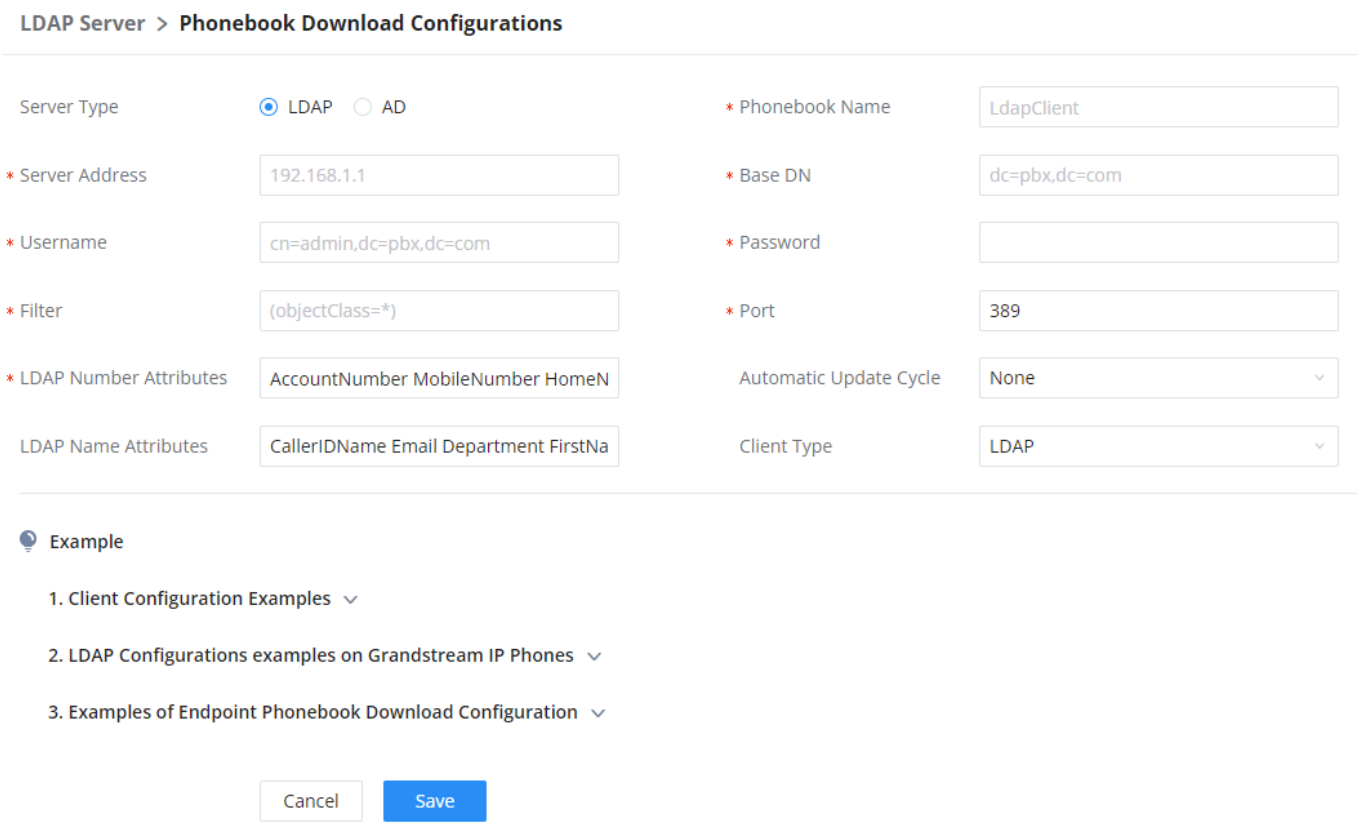
Assuming the remote server base dn is "**dc=pbx,dc=com**", configure the LDAP client as follows:

LDAP Client Type

Parameter	Description
Base DN	Specifies the location in the directory where the search is requested to begin. By default it's "dc=pbx,dc=com".
PBX DN	Specifies the location in the directory where the search for PBX entry is requested to begin. It narrows the search scope and decreases directory lookup time. By default it's "ou=pbx,dc=pbx,dc=com".
Root DN	Specifies the location in the directory where the search for the admin user entry is requested to begin. It narrows the search scope and decreases directory lookup time. By default it's "cn=admin,dc=pbx,dc=com"
Root Password	Defines the root password for authentication. By default, is "admin".
Confirm Root Password	Confirms the root password for authentication.
LDAP Cert	Certificate for LDAPS connections. Uploaded files must be less than 2MB in file size and will be automatically renamed to "server.crt".
LDAP Private Key	Private key for LDAPS connections. Uploaded files must be less than 2MB in file size and will automatically be renamed to "private.key".
LDAP CA Cert	Root certificate for LDAPS connections. Uploaded files will be automatically renamed to "server.ca".

The IPPBX can automatically update the phonebook, by configuring the 'LDAP Automatic Update Cycle'. Available options are: 1 day/2days/7 days. It is set to 'None' by default.

The following figure gives a sample configuration for IPPBX acting as a LDAP client.



LDAP Client Configurations

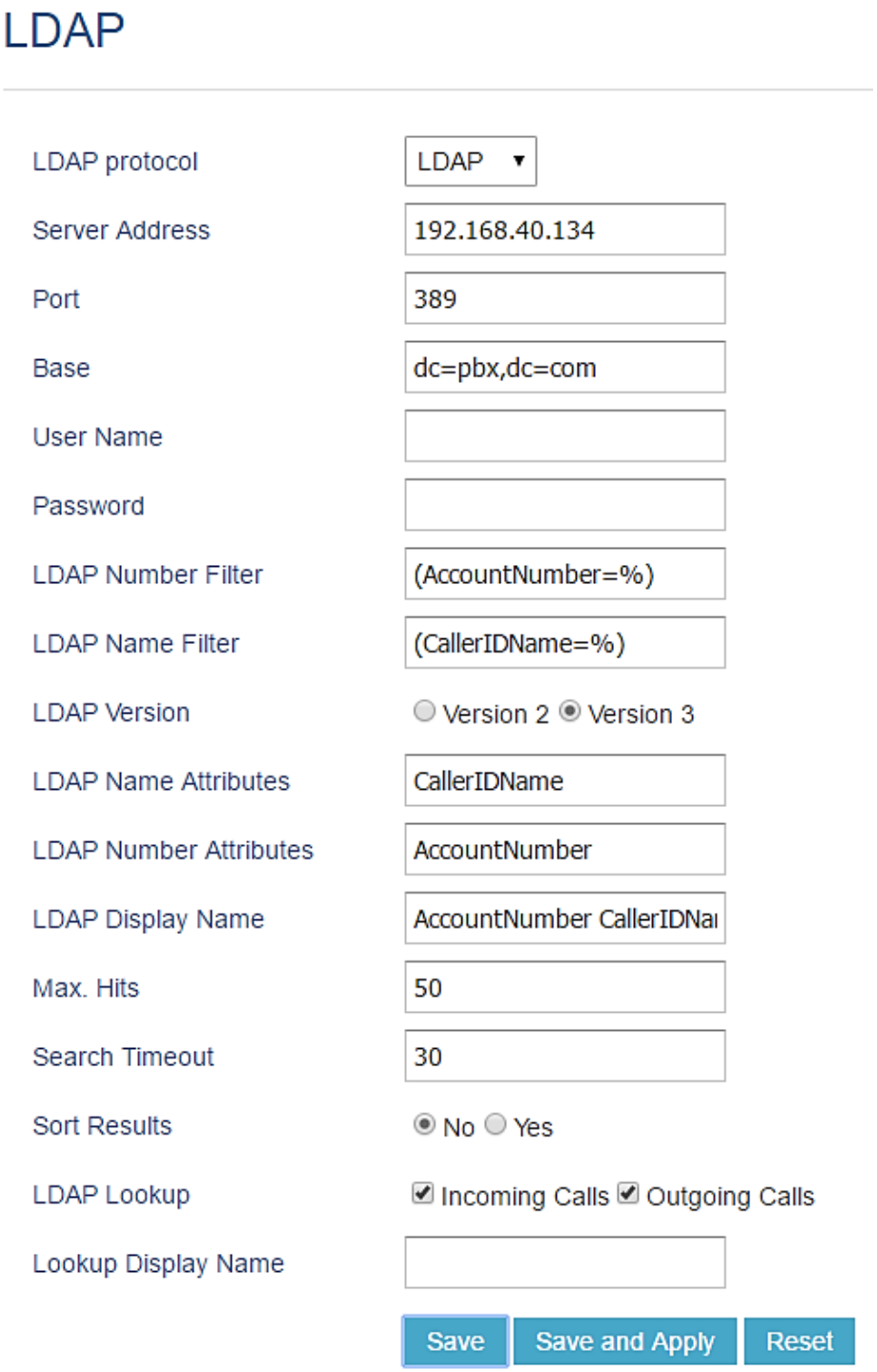
Parameter	Description
Base DN	Specifies the location in the directory where the search is requested to begin. By default it's "dc=pbx,dc=com".
PBX DN	Specifies the location in the directory where the search for PBX entry is requested to begin. It narrows the search scope and decreases directory lookup time. By default it's "ou=pbx,dc=pbx,dc=com".
Root DN	Specifies the location in the directory where the search for the admin user entry is requested to begin. It narrows the search scope and decreases directory lookup time. By default it's "cn=admin,dc=pbx,dc=com"
Root Password	Defines the root password for authentication. By default, is "admin".
Confirm Root Password	Confirms the root password for authentication.
LDAP Cert	Certificate for LDAPS connections. Uploaded files must be less than 2MB in file size and will be automatically renamed to "server.crt".
LDAP Private Key	Private key for LDAPS connections. Uploaded files must be less than 2MB in file size and will automatically be renamed to "private.key".
LDAP CA Cert	Root certificate for LDAPS connections. Uploaded files will be automatically renamed to "server.ca".

To configure Grandstream IP phones as the LDAP clients for IPPBX, please refer to the following example:

- **Server Address:** The IP address or domain name of the IPPBX
- **Base DN:** dc=pbx,dc=com
- **Username:** cn=admin,dc=pbx,dc=com
- **Password:** admin (by default)
- **LDAP Name Attribute:** CallerIDName Email Department FirstName LastName
- **LDAP Number Attribute:** AccountNumber MobileNumber HomeNumber Fax
- **LDAP Number Filter:** (AccountNumber=%)

- **LDAP Name Filter:** (CallerIDName=%)
- **LDAP Display Name:** AccountNumber CallerIDName
- **LDAP Version:** If existed, please select LDAP Version 3
- **Port:** 389

The following figure shows the configuration information on a Grandstream GXP2170 to successfully use the LDAP server as configured in **[LDAP Server Configurations]**.



GXP2170 LDAP Phonebook Configuration

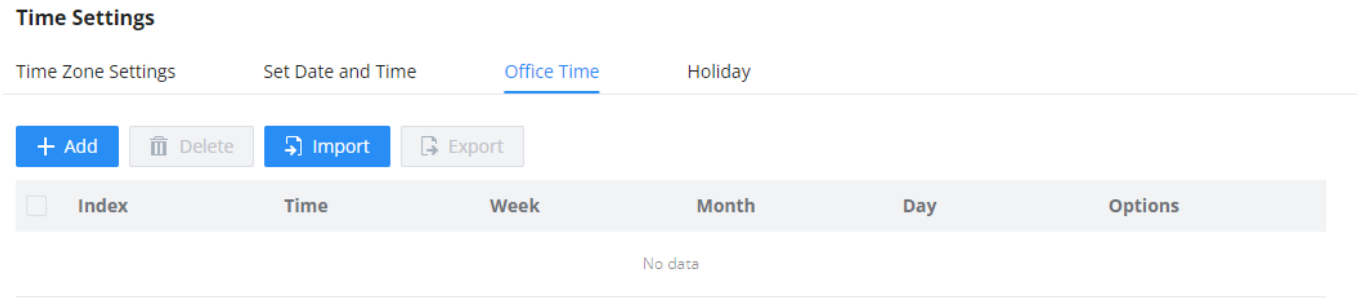
The IPPBX LDAP server is no longer supporting the anonymous binding of the LDAP client.

Time Settings

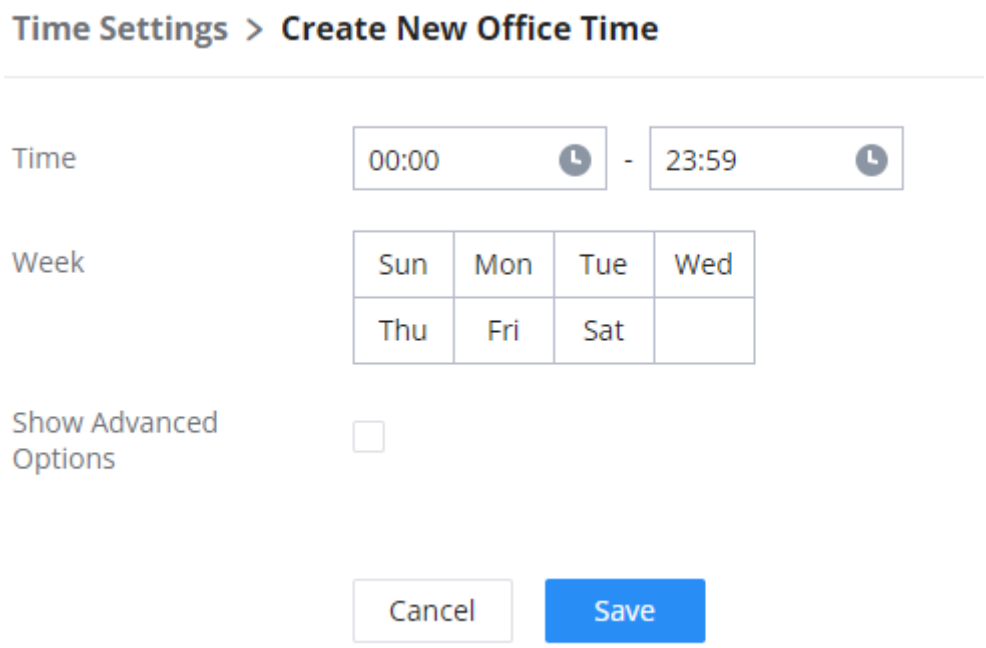
Time settings allow the user to configure the Office Time, Holiday, and Custom Time Groups.

Office Time

On the IPPBX, the system administrator can define "office time" which can be used to configure time condition for extension call forwarding and inbound rules. To configure office time, log in to the Web GUI, enter the **System Settings→Time Settings→Office Time**, and click the "Add" button to see the following configuration page.



Office Time



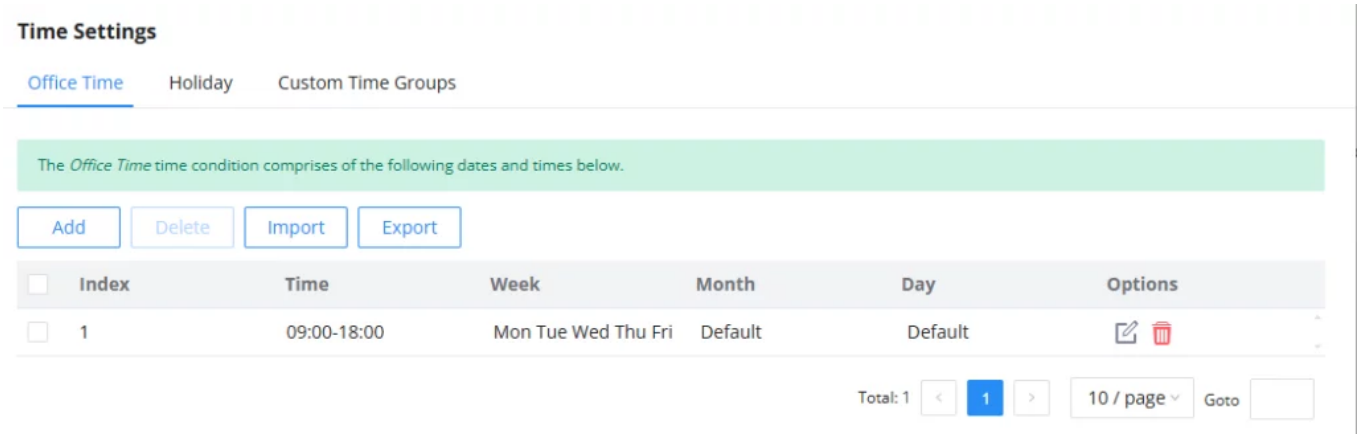
Create New Office Time

Start Time	Configure the start time for office hour.
End Time	Configure the end time for office hour
Week	Select the workdays in one week.
Show Advanced Options	Check this option to show advanced options. Once selected, please specify "Month" and "Day" below.
Month	Select the months for office time.
Day	Select the workdays in one month.

Create New Office Time

Select "Start Time", "End Time" and the day for the "Week" for the office time. The system administrator can also define month and day of the month as advanced options. Once done, click on "Save" and then "Apply Change" for the office time to take effect. The office time will be listed in the web page as the figure shows below.

The office time feature support import/export CSV files.



Office Time

- Click on

  to edit the office time.

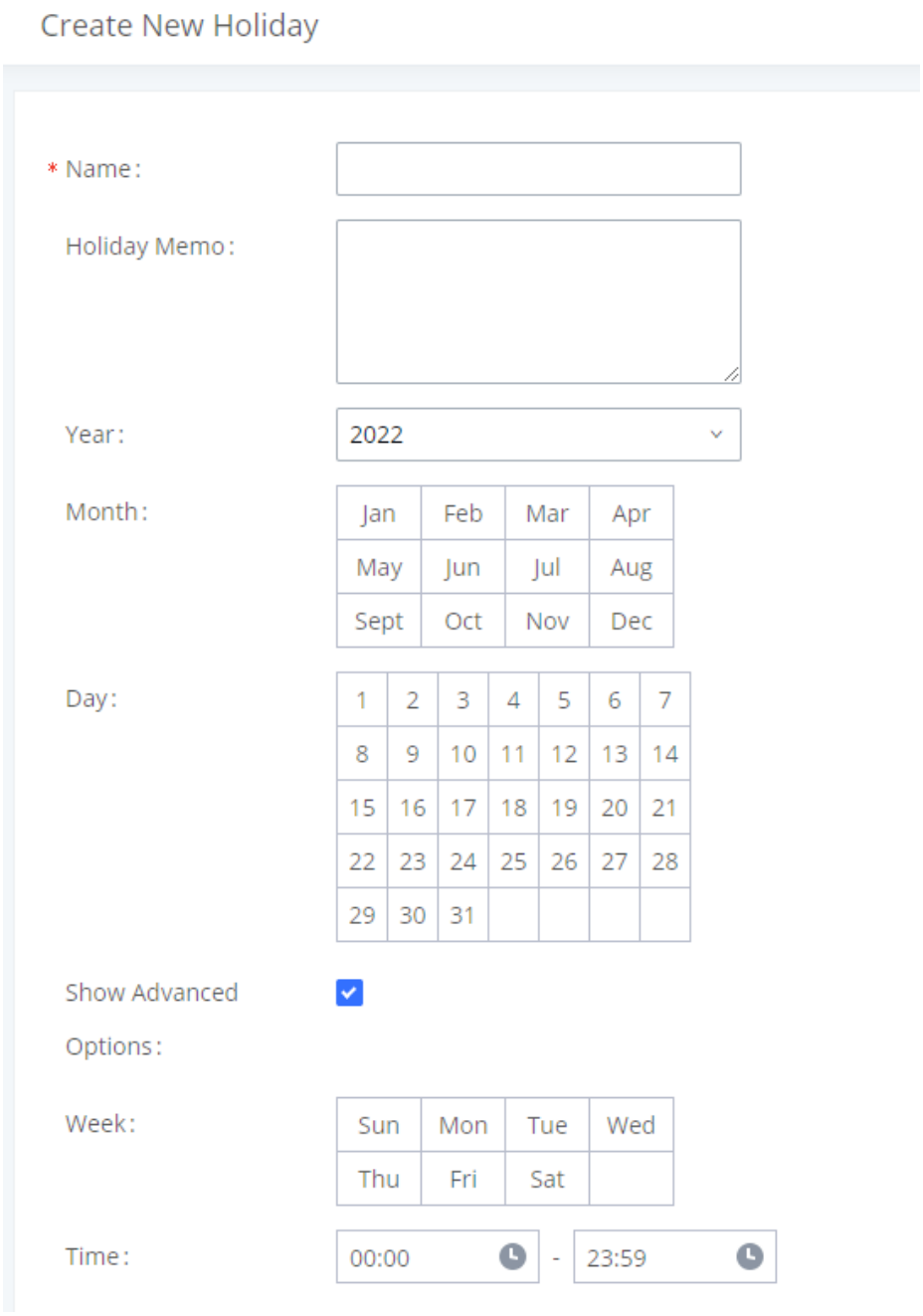
- Click on

  to delete the office time.

- Click on **"Delete"** to delete multiple selected office times at once.

Holiday

On the IPPBX, the system administrator can define "holidays" which can be used to configure time condition for extension call forwarding and inbound rules. To configure office time, log in to the Web GUI, enter the **System Settings→Time Settings→Holiday**, and click the "Add" button to see the following configuration page.



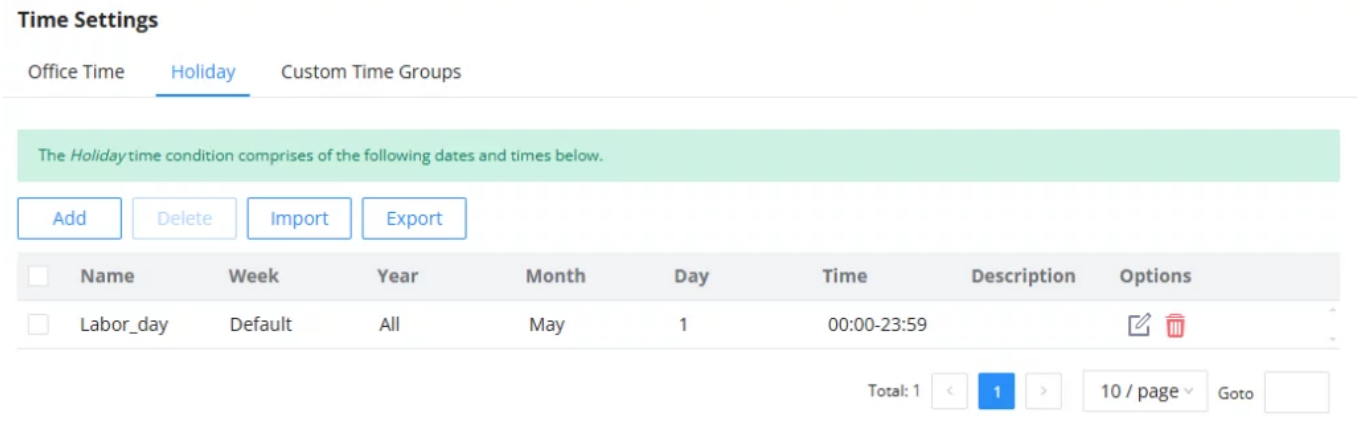
Create New Holiday

Name	Specify the holiday name to identify this holiday.
Holiday Memo	Create a note for the holiday.
Month	Select the month for the holiday.
Year	Select the Year for the holiday.Note: In the "Year" option, select "All" to set annual fixed holiday information.
Day	Select the day for the holiday.
Show Advanced Options	Check this option to show advanced options. If selected, please specify the days as holiday in one week below.
Week	Select the days as holiday in one week.
Time	Select the time on which the holiday starts.

Enter holiday "Name" and "Holiday Memo" for the new holiday. Then select "Month", "Day" and the exact

"Hour". The system administrator can also define days in one week as advanced options. Once done, click on "Save" and then "Apply Change" for the holiday to take effect. The holiday will be listed in the web page as the figure shows.

The Holiday feature support import/export CSV files.



Settings→Time Settings→Holiday

- Click on

  to edit the holiday.

- Click on

  to delete the holiday.

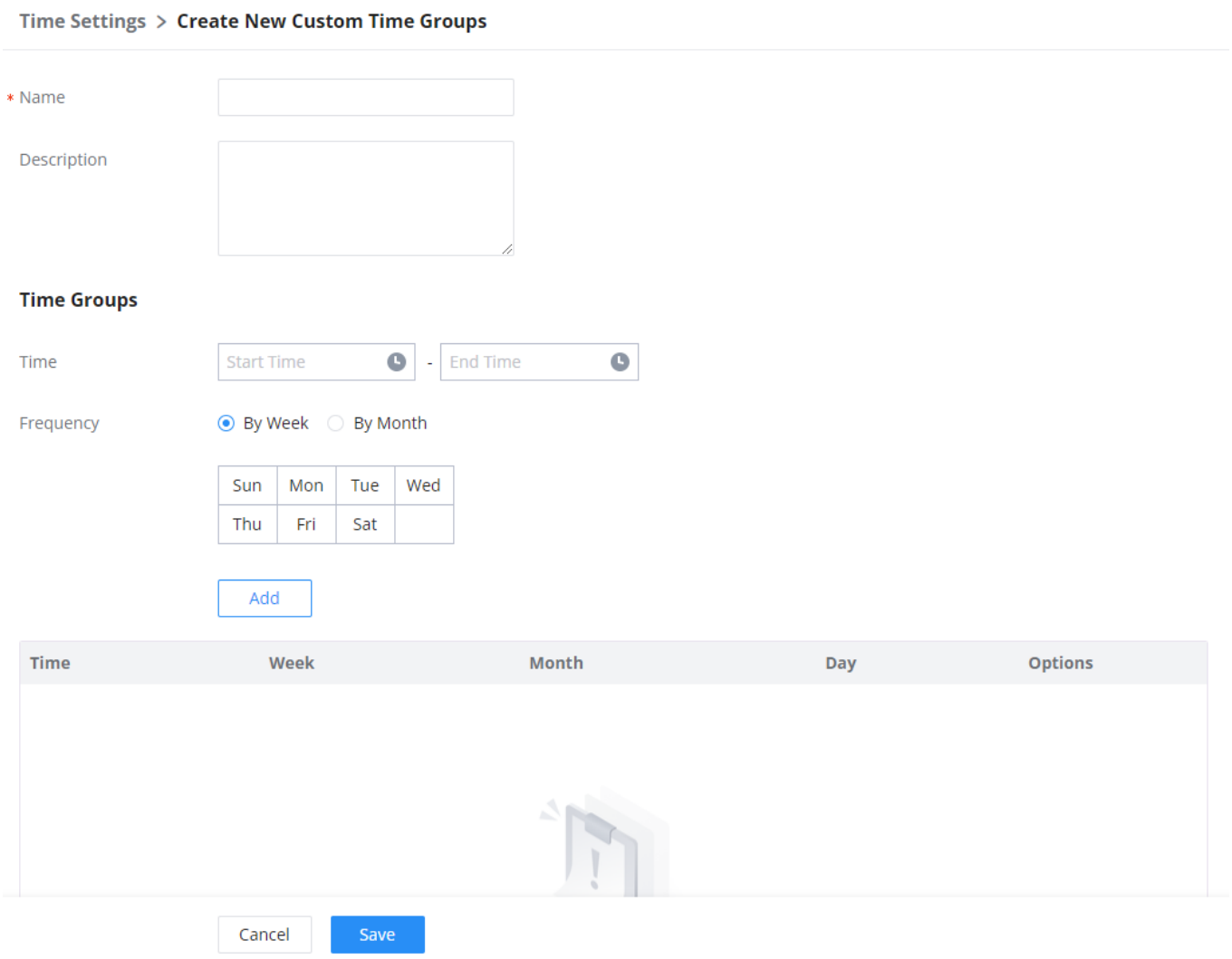
- Click on **"Delete"** to delete multiple selected holidays at once.

Custom Time Groups

Users can create custom time frames which can be used as a routing condition in the inbound and outbound routes. Multiple time ranges can be added and the frequency can be customized to be every specific weekday or every specific day/week of the selected months.

Users can also export and import these custom time groups in CSV format for easier management.

To access Custom Time Groups, please navigate to **System Settings > Time Settings > Custom Time Groups**



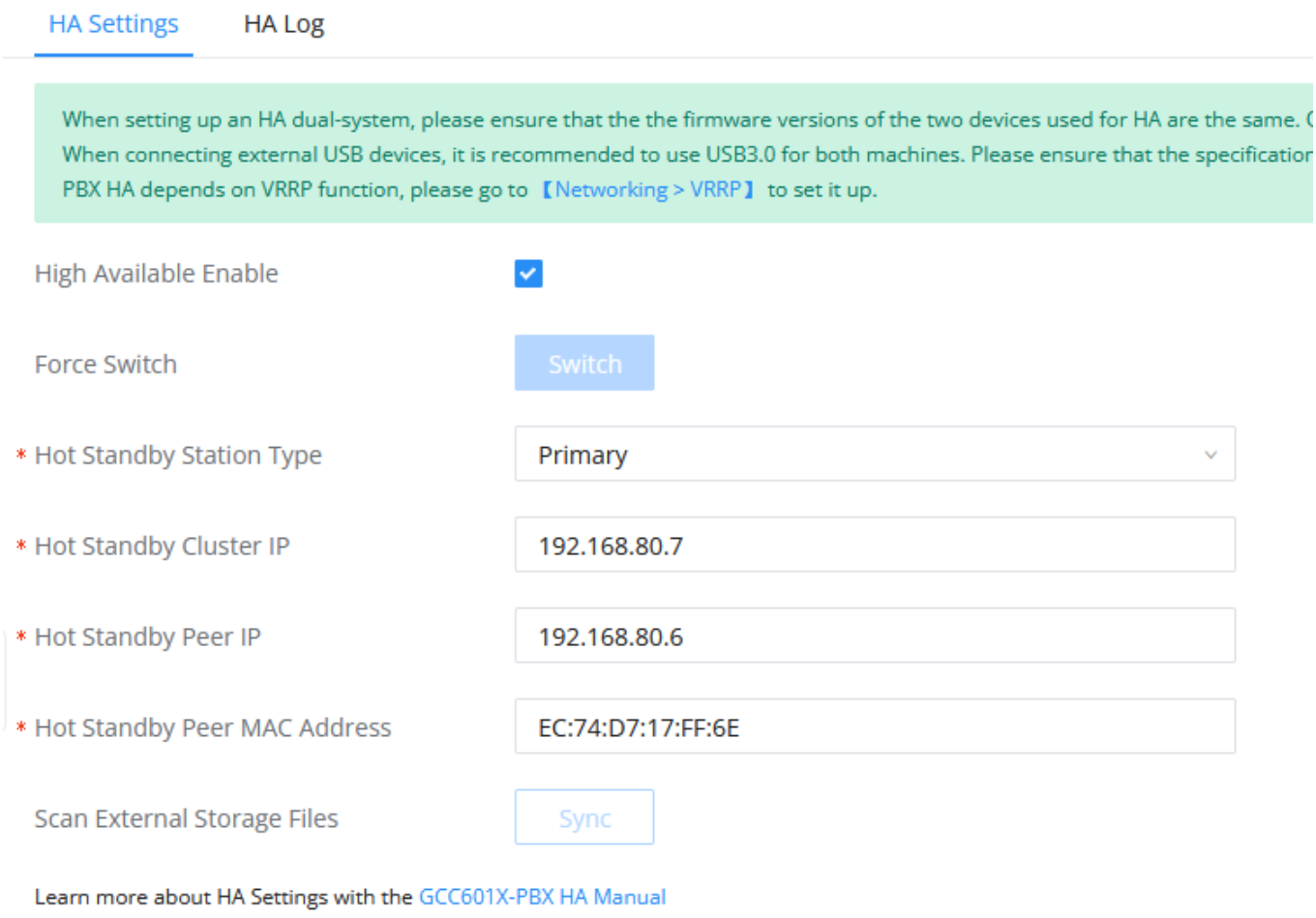
Create New Custom Time

Parameter	Description
Name	Enter the name of the time group.
Description	Enter the description of the time group.
Time Groups	
Time	Select the time period for this group.
Frequency	Select the frequency of this group per week/month.

HA (High Availability)

Dual-system hot standby provides a highly reliable and fault-tolerant solution for enterprises using the GCC6020 Series integrated PBX solution. Based on two GCC devices of the same product model and software version, one of them is in the "Active" working state in real-time, and the other is in the "Standby" working state. The daily data on the host server will be synchronized to the standby machine in real-time, and the standby machine always monitors the running status of the host. When the host fails, including hardware failures and severe software failures, the standby machine will immediately take over the business and enter the "Active" working state to ensure that the business is not interrupted.

PBX HA depends on VRRP function to be deployed correctly. To setup VRRP for GCC, please refer to the following guide: https://documentation.grandstream.com/knowledge-base/gcc6000-series-vrrp-user-guide/.

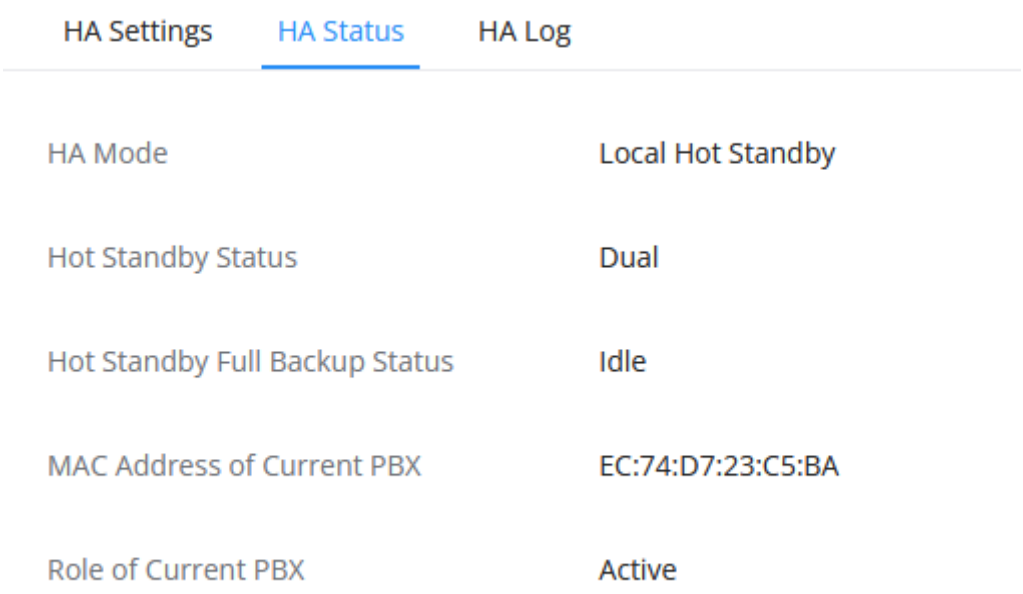


HA Settings

Parameter	Description
High Available Enable	Enable High Availability feature.
Hot Standby Station Type	Used for the initial assignment of the HA active/standby role of the PBX system. If set to Primary, the current PBX system will be assigned as the intial active device. If set to Secondary, the peer PBX system will be assigned as the initial standby device. The roles of the PBX systems may change as HA switchovers occur.
Hot Standby Cluster IP	To use this service, the active and standby PBX systems need to use the same static IP address.
Hot Standby Peer IP	Enter IP address of the peer Hot Standby device.
Hot Standby Peer MAC Address	The MAC address of the peer device can be obtained from its webUI under the Home > Overview page.
Scan External Storage Files	Only applicable if there are more than 5000 PBX files in external storage such as SD card, USB, or NAS. Users can click this button to scan those paths in order to display all available files on the PBX web UI. Configured file storage paths can be viewed on the File Manager page.It is recommended to configure external storage data sync after setting up HA for the first time. Files created after HA has been set up will be automatically displayed on the PBX web UI and do not need to be scanned for.

HA Status

When High Availability is established, the HA Status page will show information such as the HA mode, the status of the HA, the backup status, the MAC address and the role of the PBX.



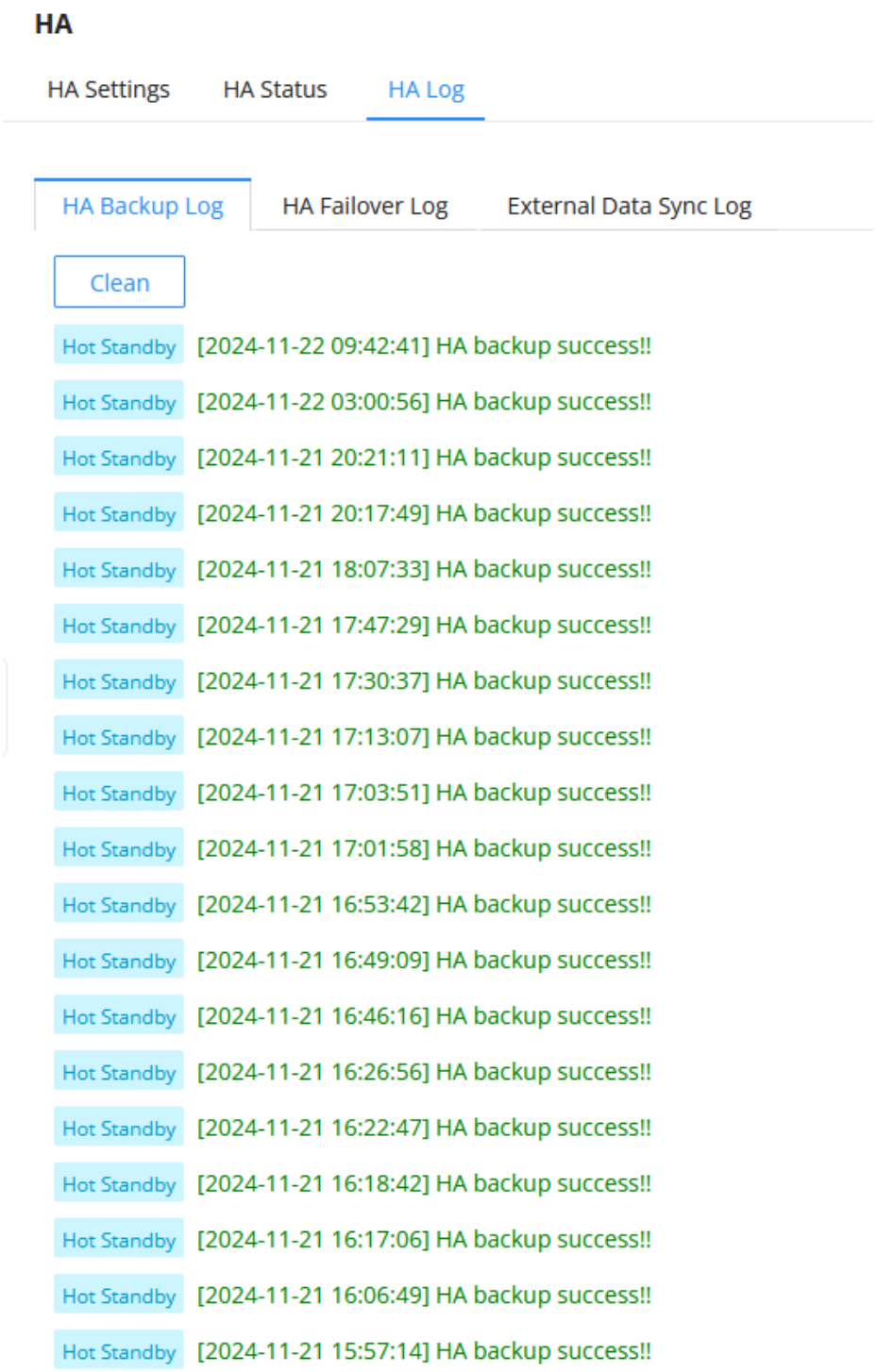
HA Status

HA Log

In the HA Log page, the user can view the backup events between the primary and the secondary GCC devices. The user can also view the switchover events which occur in case the active device has failed. In addition to that, the user can view the events of data synchronization with the attached storage unit attached to the GCC device.

- **HA Backup Log**

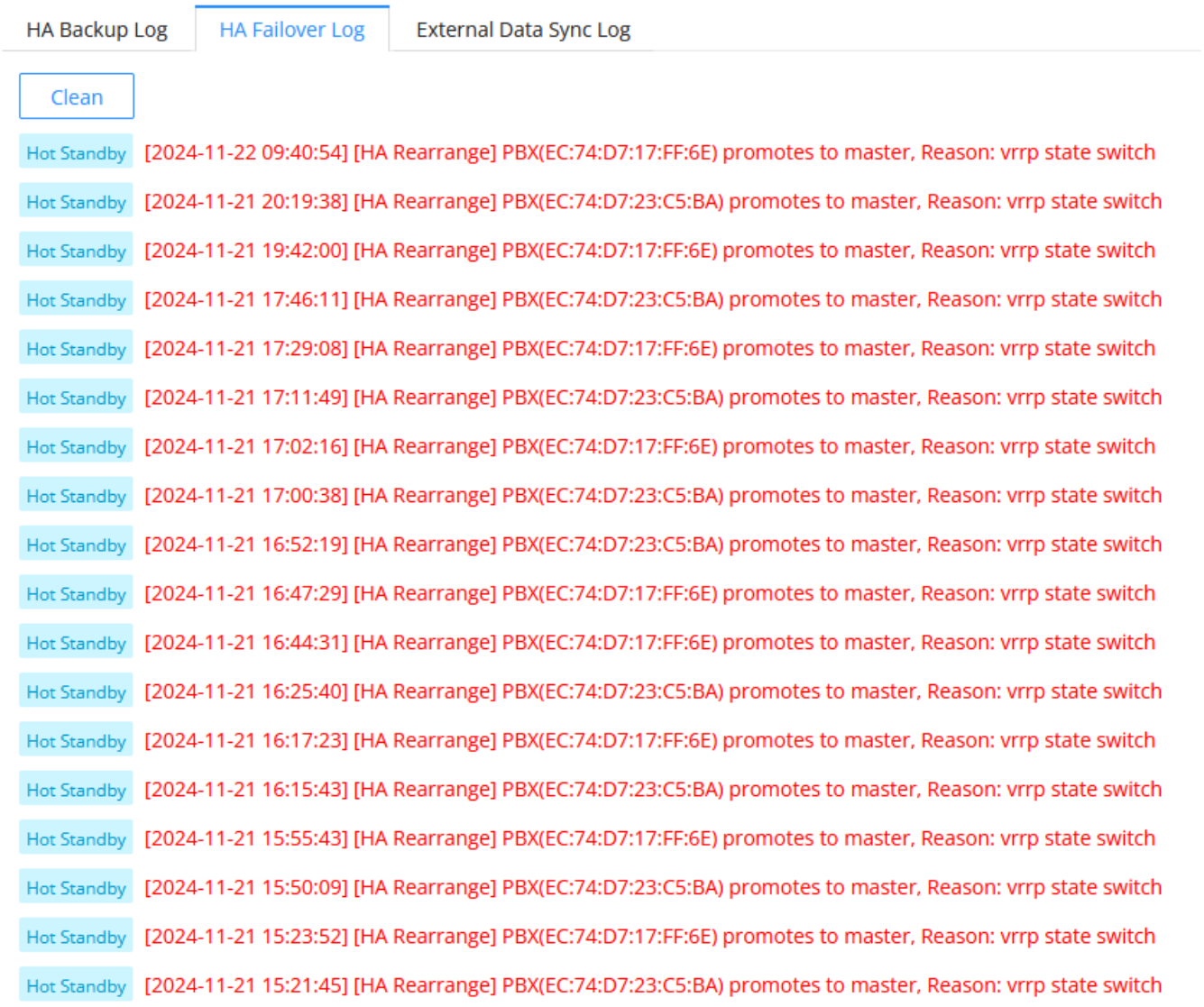
HA Backup Log shows the events when when the data and configuration has occurred.



HA Backup Log

- **HA Failover Log**

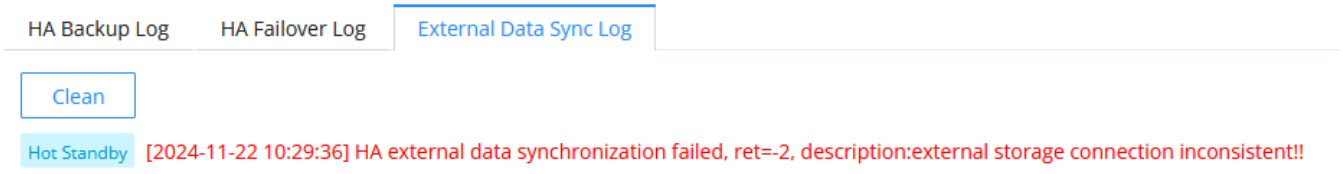
The HA Failover Log shows the events of when the HA switchover has occurred. It shows also the reason of the switchover.



HA Failover Log

- **External Data Sync Log**

External Data Sync Log shows the events when data sync to external storage has been triggered, it shows also whether it has been performed successfully or if has failed.



External Data Sync Log

Cluster

Clustering allows multiple units to be deployed in the same environment to work together on telephony features. This setup provides higher processing capacity, supporting the bandwidth required for handling numerous calls simultaneously. By clustering multiple GCC devices, the system can accommodate more concurrent calls and meeting participants.

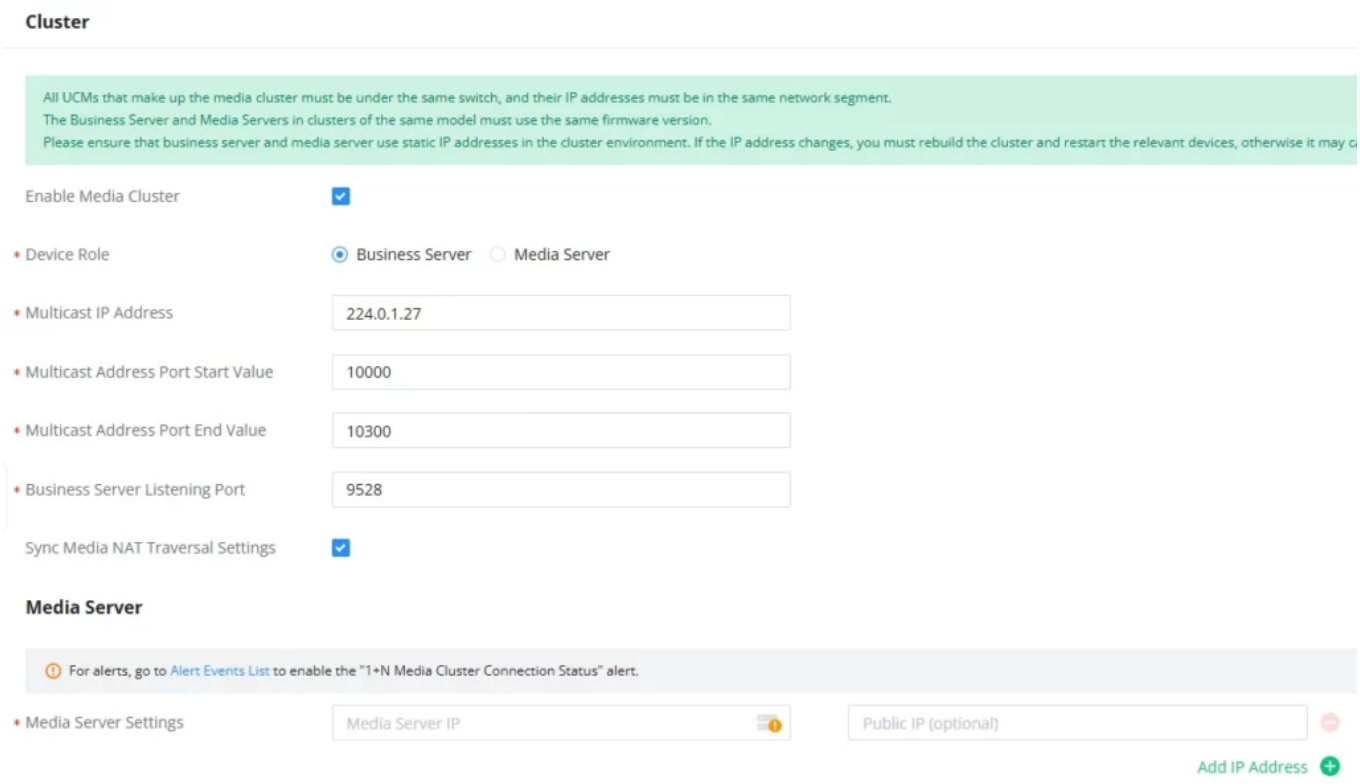
- The user can deploy a maximum of 5 media servers.
- Each cluster deployment requires at least one business server and one media server.

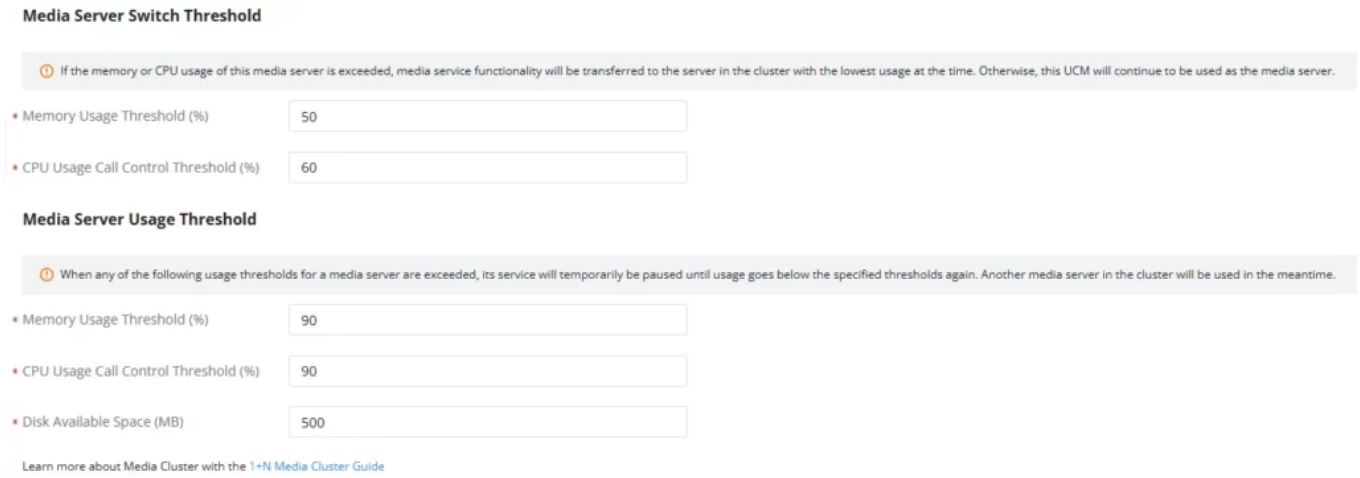
To configure the cluster, we will need two types of servers; a Business server, which is a one and only server that will be used for the administration of the operations. And a media server, which can either be one singular server, or for deployments that require more resource, the user can deploy up to 5 GCC units as media server. Media server will handle media related traffic.

- All the GCC units that form the media cluster must be under the same switch, and their IP addresses must be in the same network segment.
- The Business Server and Media Servers in clusters of the same model must use the same firmware version.
- Please ensure that IP addresses reservation is created for the business server and the media server(s) in the DHCP server to ensure that the servers will use the same IP addresses to communicate. In the case the IP addresses have been changed, the cluster needs to be rebuilt.

Business Server

To configure Business server, please access the GCC web UI then navigate to **System Settings -> Cluster**, then tick the option "Enable Media Cluster", and choose "Business Server" as the **Device Role.**

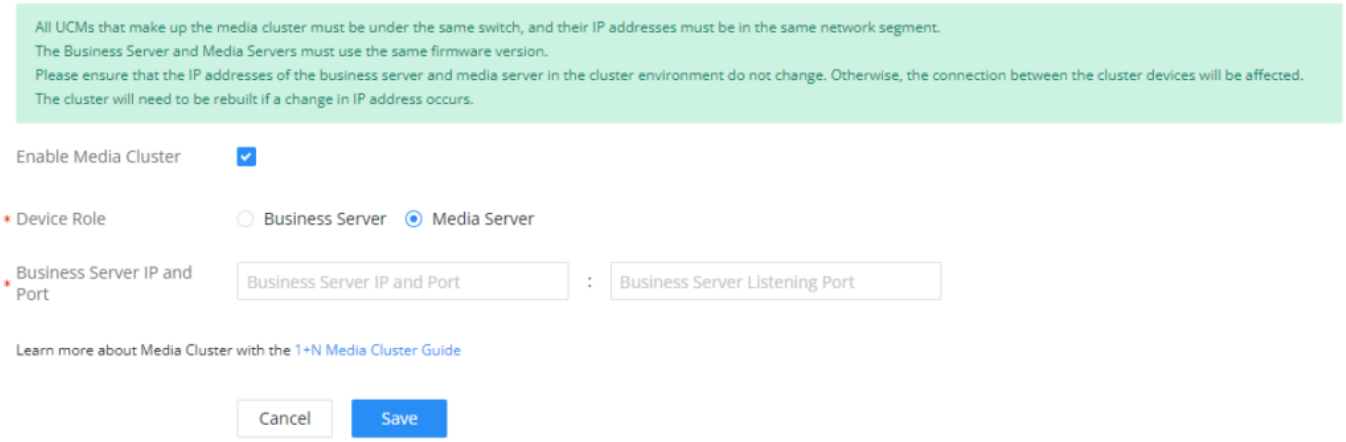




Parameter	Description
Enable Media Cluster	Enable Media Cluster feature.
Device Role	Configure the device role. Business Server: Choosing this role will make the GCC as the main server for operations.Media Server: Choosing this role will make the GCC as a media server only. All the PBX related features will be disabled.
Multicast IP Address	The allowed multicast IP address range is 224.0.1.0 - 238.255.255.255
Multicast Address Port Start Value	Enter the multicast address port start value. The port number value can be within the range 1024 - 65535. Note: The multicast address port start value must be lower or equal to the multicast address port end value.
Multicast Address Port End Value	Enter the multicast address port end value. The port number value can be within the range 1024 - 65535.Note: The multicast address port end value must be greater or equal to the multicast address port start value.
Business Server Listening Port	The port on which the business server listens and receives information reported by the media server. The value of the listening port value can be within 1024 - 65535.
Sync Media NAT Traversal Settings	If enabled, all media servers in the cluster will use the same Media NAT Traversal settings as the business server. If disabled, each server will need to be configured individually.
Media Server	
Media Server Settings	Enter the IP address of the media server along with the public IP (optional). The user can pair up to 5 media servers with one Business server.The statuses of the media servers are indicated using three colors:? Green: The server is available. Yellow: The server is in use. Gray: The server is offline.
Media Server Switch Threshold	
Memory Usage Threshold (%)	When the memory usage of this media server is exceeded, media services will be assumed by another media server in the cluster with the lowest usage at the time.
CPU Usage Call Control Threshold (%)	When the CPU usage of this media server is exceeded, media services will be asssumed by another media server in the cluster with the lowest usage at the time.
Media Server Usage Threshold	
Memory Usage Threshold (%)	When this server's memory usage exceeds the specified threshold, service from this server will be temporarily paused.
CPU Usage Call Control Threshold (%)	When this server's CPU usage exceeds the specified threshold, service from this server will be temporarily paused.
Disk Available Space (MB)	When the percentage of available space in the device's disk data partition falls below the set value, this server will not be used until the amount of available space is above the threshold again.

Media Server

To configure Media server, please access the web UI then navigate to **System Settings > Cluster,** then tick "Enable Media Cluster".



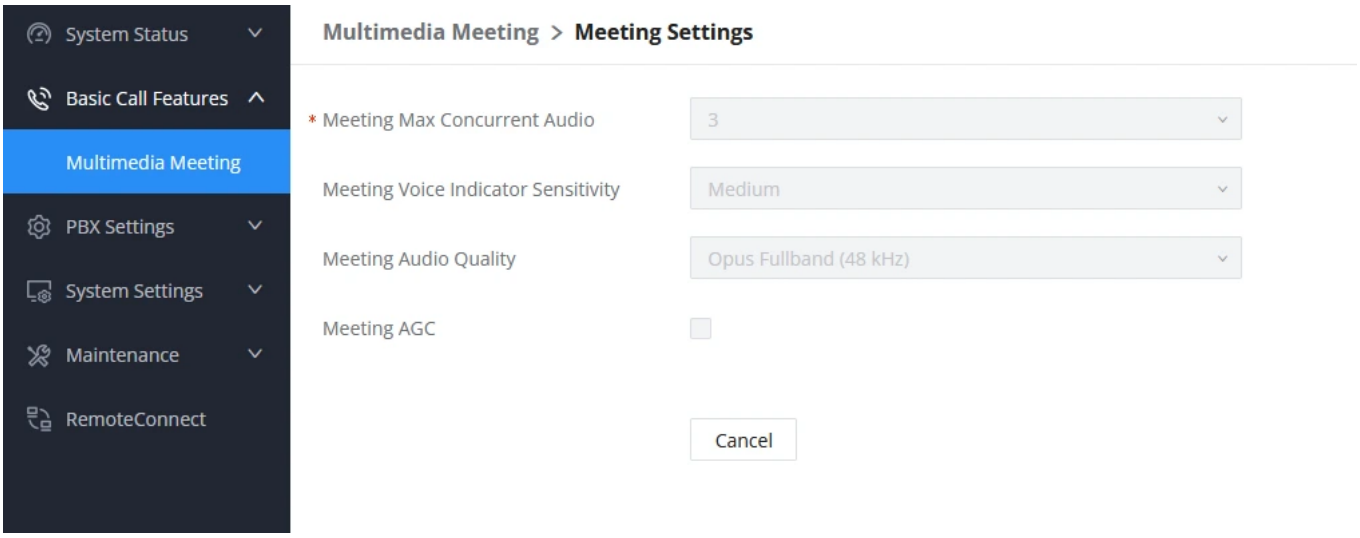
Enable Media Cluster	Enable Media Cluster feature.
Device Role	Configure the device role.
Business Server IP and Port	Enter the IP address of the business server with the listening port. If the business server is using HA Local Hot Standby, the Hot Standby Cluster IP address must be used. If the business server is using HA Remote Disaster Recovery, the primary GCC and secondary GCC must have their own respective cluster environments.

1+N Media Cluster Synchronization

In order to avoid inconsistencies between the configurations of the media server and the business server in a cluster, the following configuration parameters can be synchronized:

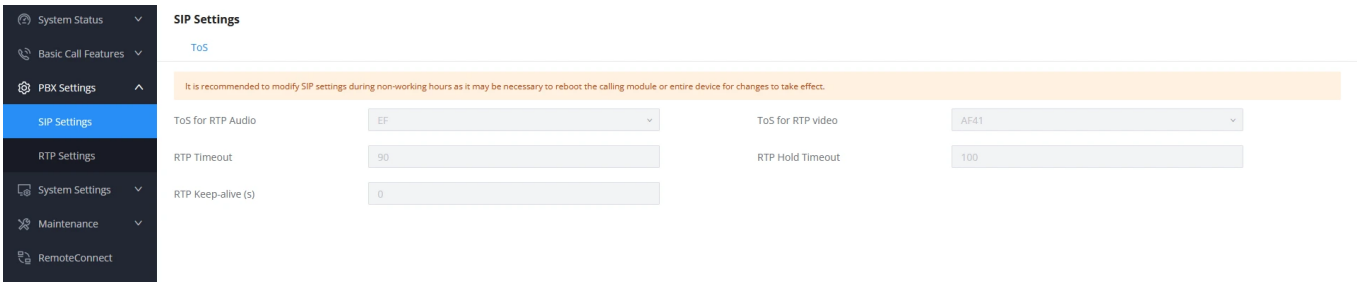
- **Basic Call Features -> Multimedia Meeting**

For media servers, all the configuration options on this page are greyed out to display the information configured on the business server.



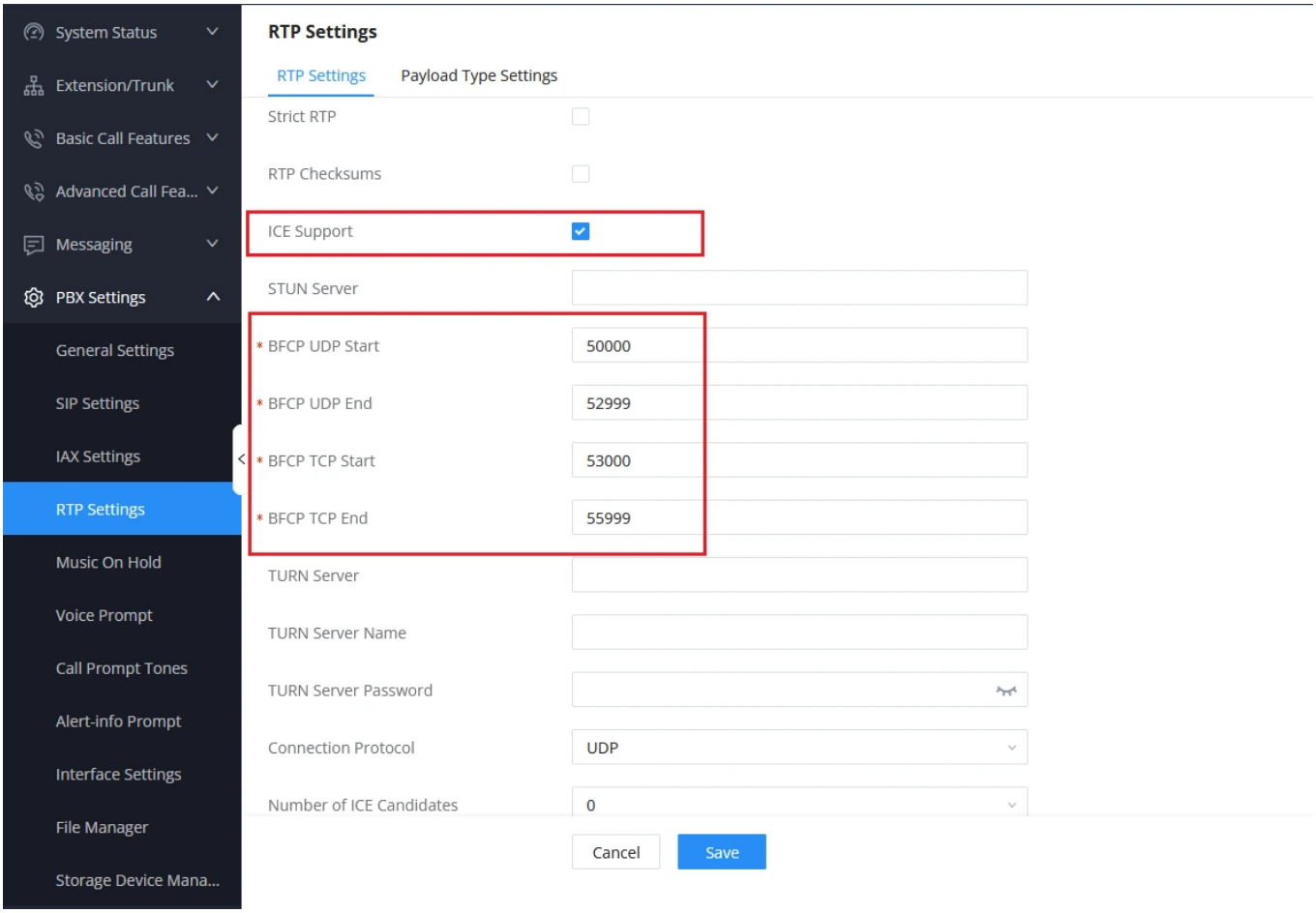
- **PBX Settings -> SIP Settings -> ToS**

This page for media servers shows the RTP related configurations of the business server.

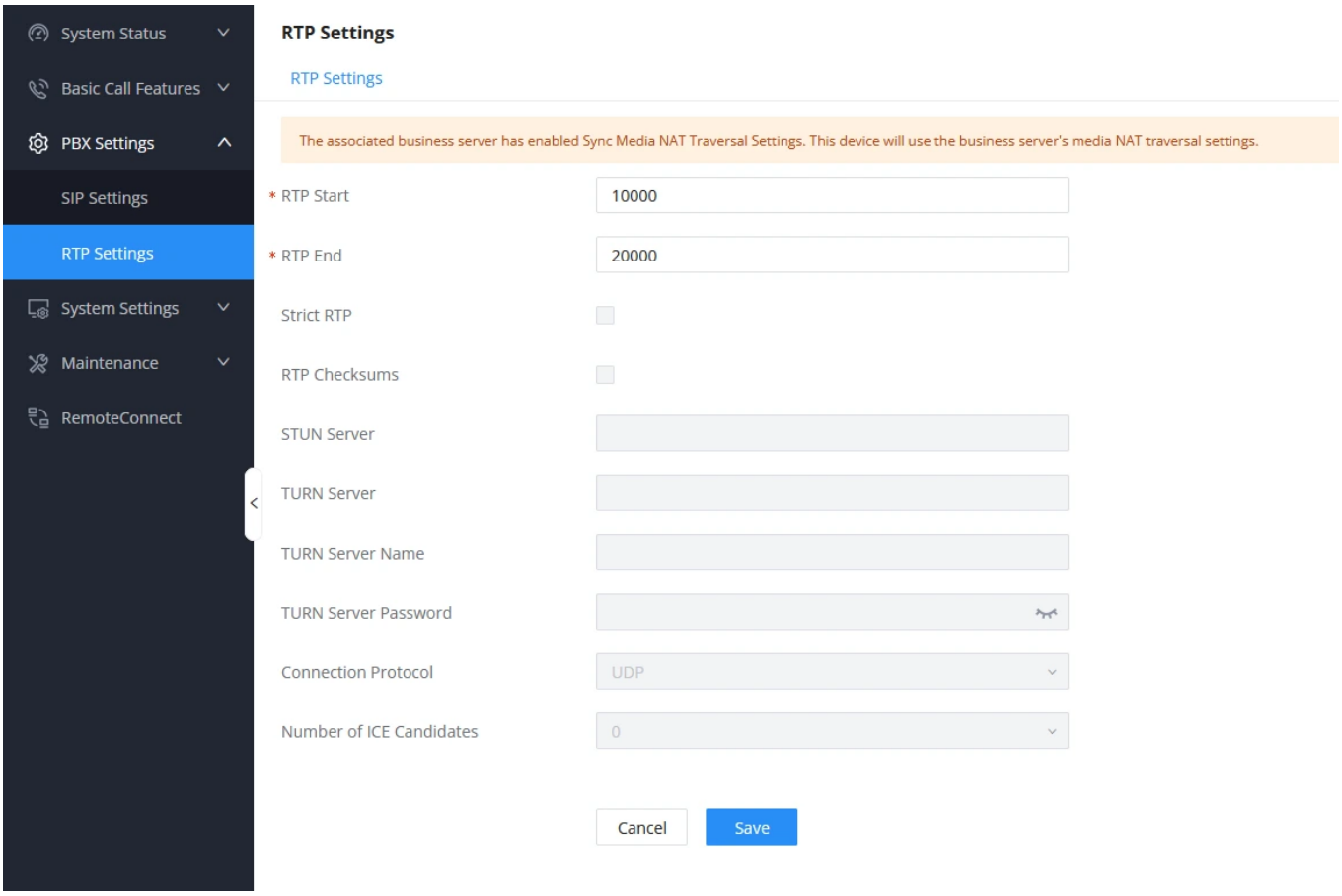


- **PBX Settings -> RTP Settings**

The following media NAT Traversal parameters will only be visible on the business server:



The remaining settings will be greyed out on the media server's side. Only the configuration on the business server will be taken into consideration.

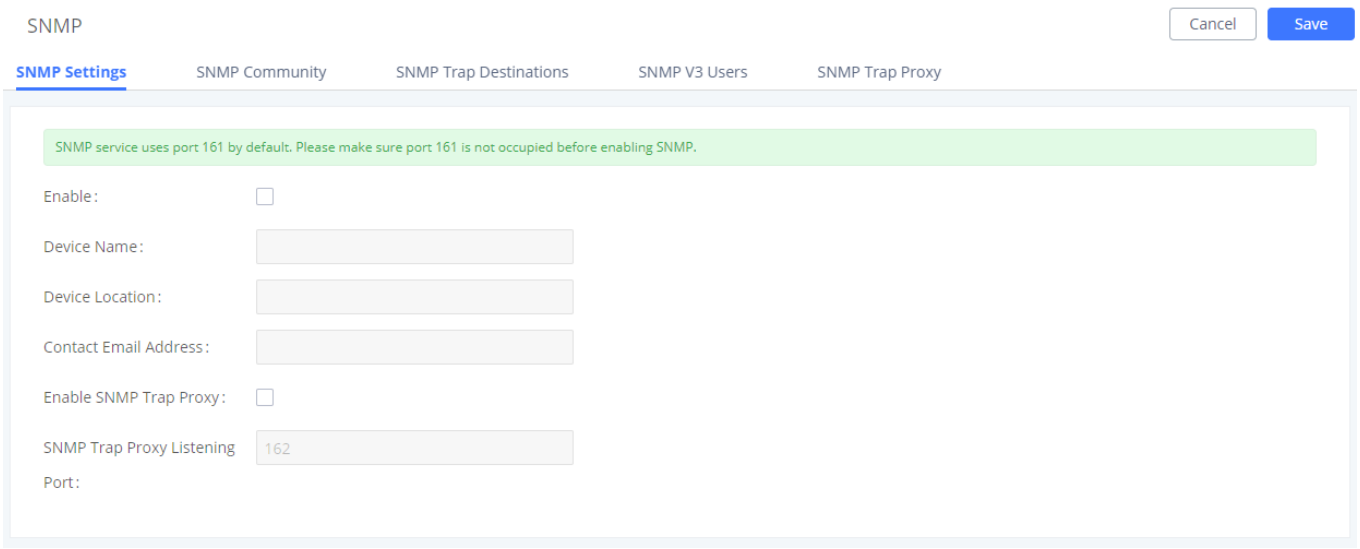


The SRTP Debugging option to allow decryption of packet captures will not be available on the media server under **Maintenance -> Network Troubleshooting -> Ethernet Capture**.

SNMP

SNMP integration in the IPPBX allows the administrator to monitor the components of the IPPBX remotely. SNMP is a useful integration to have a centralized monitoring dashboard that shows all the devices on your network, with that status of internal component. To configure the SNMP integration, please navigate to the web GUI of the IPPBX, then go to **System Settings -> SNMP -> SNMP Settings**

SNMP Settings

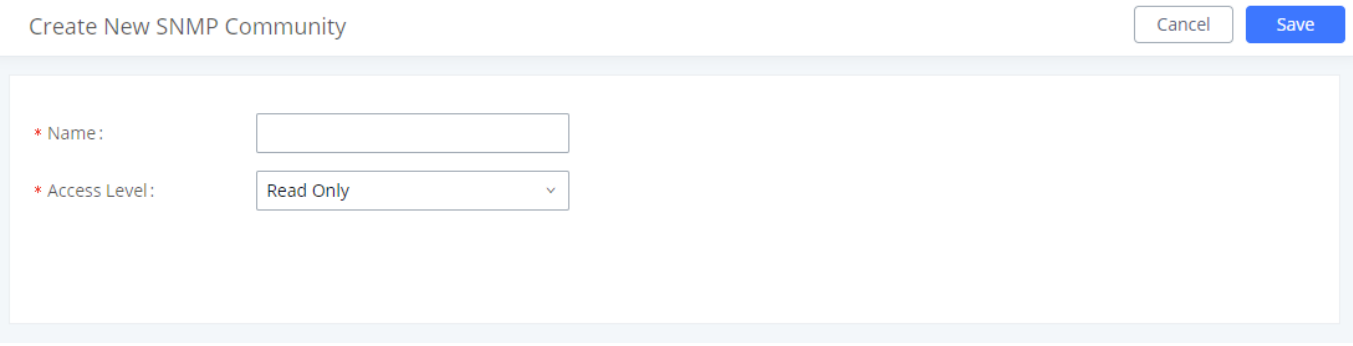


SNMP Settings

Enable	Tick this box to enable SNMP.
Device Name	Enter the device name.
Location	Enter the location.
Contact Email Address	Enter the email address used to send the SNMP alerts to.
Enable SNMP Trap Proxy	Tick this box to enable a proxy for SNMP Trap.
SNMP Trap Proxy Listening Port	The port number on which the SNMP Trap Proxy is listening on.

SNMP Community

You can also create SNMP communities and affect a certain level of access. An SNMP community is a group created to aggregate many management stations. The community name is used to authenticate and identify these machines in the NMS (Network Management System).

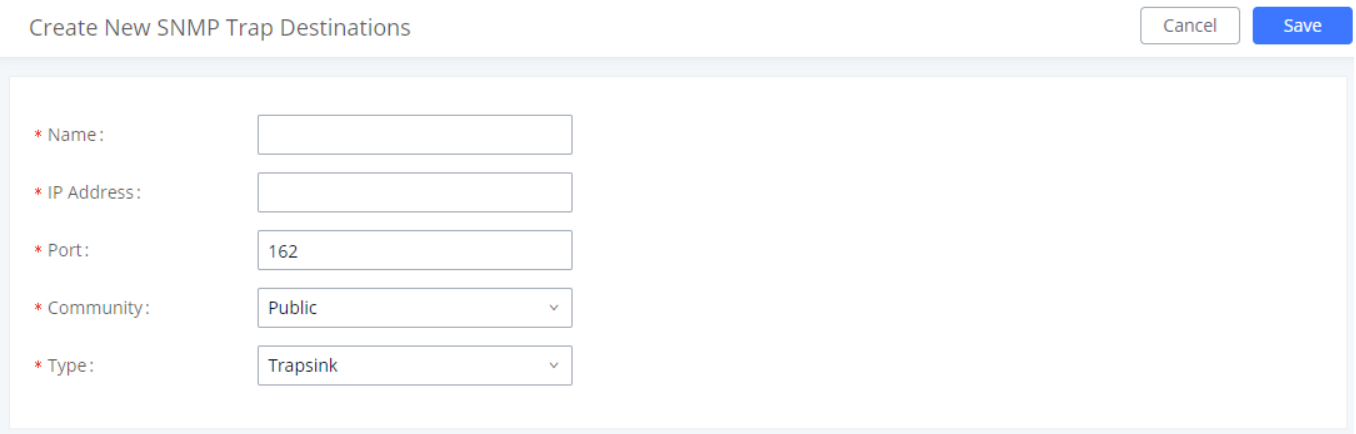


Create SNMP Community

Name	Enter a name for the community
Access Level	Select an access level: Read Only: The SNMP community will be able only to read SNMP messages.

SNMP Traps Destination

SNMP Traps is a very useful feature when there are many network components to manage. Instead of sending requests to all the machines in the network in order to view their SNMP logs risking slowing down or bringing the network to a complete halt, SNMP Traps can be configured so these machines can send unrequested messages to the manager to notify it about critical events and general failures.



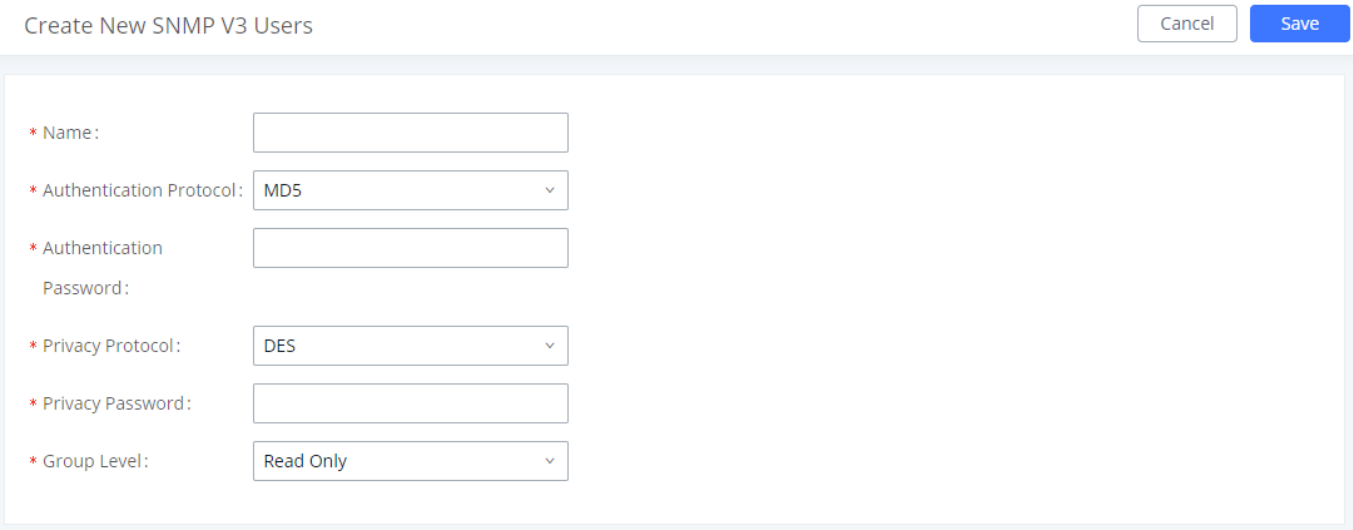
Create SNMP Trap Destination

Name	Enter a name of your SNMP Trap destination.
IP Address	Enter the SNMP Trap destination's IP address.

Port	Enter the port of the SNMP Trap destination.
Community	Select the community that you want
Type	Select the type of SNMP: Trapsink: Select this option if you want to send SNMP v1 traps. Trap2sink: Select this option if you want to send SNMP v2 traps. Informsink: Select this option if you want to send "Inform" notifications only.

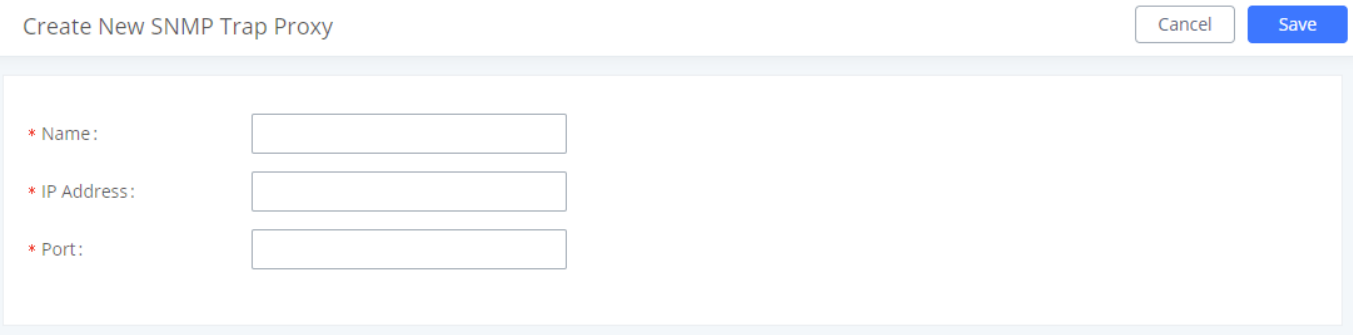
SNMP Version 3

IPPBX also supports SNMP v3 in case the system administrator decides to add more security to the monitoring process. SNMP v3 is a very good solution to monitor devices that interface directly with Internet. SNMP v3 offers more security than its predecessors by hashing the authentication information, encrypting the SNMP messages exchanged between the managed devices and the network management system which prevent eavesdropping. Also, it prevents any data tampering which protects the integrity of the data exchanged.



SNMP V3 Configuration

SNMP Trap Proxy



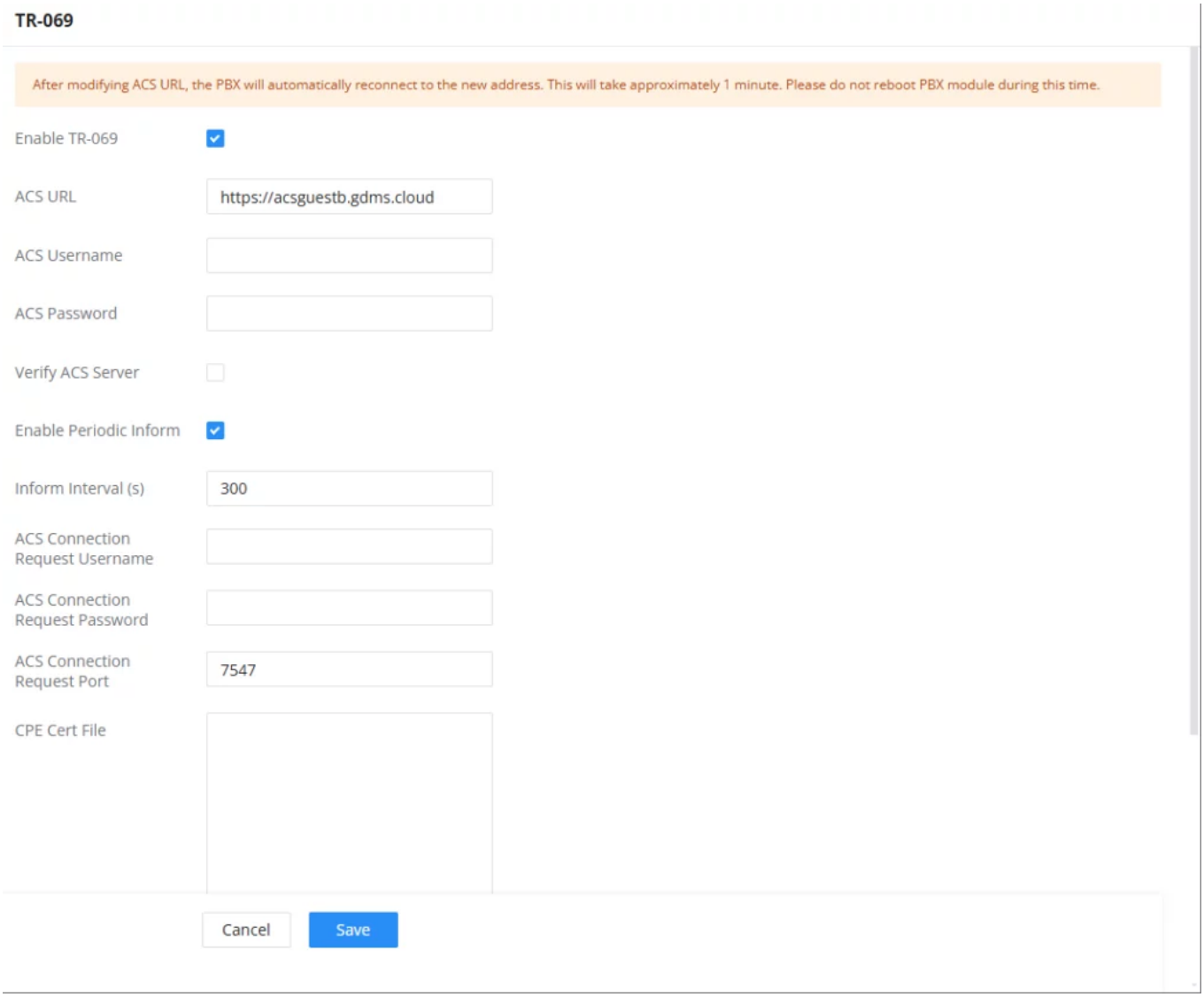
SNMP Trap Proxy

Name	Enter a name for the proxy server.
IP Address	Enter the proxy server's IP address.
Port	Enter the proxy server's port.

TR-069

TR-069 allows remote configuration of the IPPBX. The administrator can use TR-069 to manage multiple

IPPBXs at the same time which reduces the time spent on configuring each IPPBX.



TR 069

To configure TR-069 on Grandstream devices, set the following parameters:

Parameter	Description
Enable TR-069	Toggle it on to enable TR-069. It is enabled by default
ACS URL	URL for TR-069 Auto Configuration Servers (ACS), e.g., http://myacs.grandstream.com
TR-069 Username	ACS username for TR-069 must be the same as in the ACS configuration.
TR-069 Password	ACS password for TR-069 must be the same as in the ACS configuration.
Verify ACS Server	Enables verification of the server certificate when interacting with GDMS. For security purposes, it is recommended to enable this option. However, verification is not needed for self-signed certifications.
Enable Periodic Inform	If enabled, Inform messages will be sent periodically based on the Inform Interval value.
Enable Periodic Inform	Enables periodic inform. If set to Yes, the device will send inform packets to the ACS.
Inform Interval (s)	A periodic time when PBX will send inform packets to TR-069 ACS serverThis option is specified in seconds.The default value is 86400.
ACS Connection Request Username	The username for the ACS to connect to PBX.

ACS Connection Request Password	The password for the ACS to connect to the PBX.
ACS Connection Request Port	Port for incoming connection requests.The default value is 7547.
CPE Cert File	The Cert file for PBX to connect to the ACS via SSL.
CPE Cert Key	The Cert key for PBX to connect to the ACS via SSL.

CONTACTS

Address book management is under IPPBX web UI->Maintenance, and it has two sections "Contact Management" and "Department management".

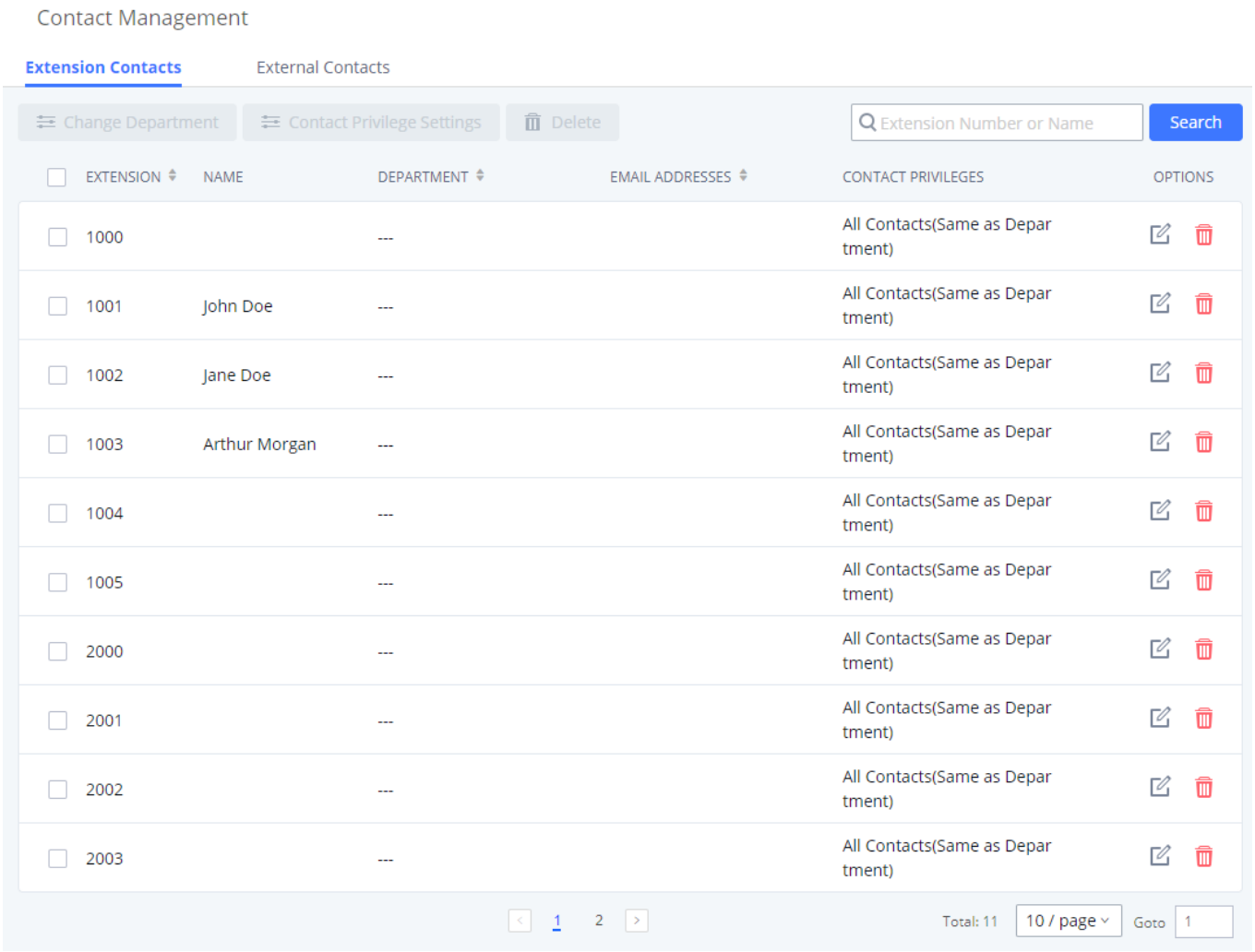
Contact Management

Contact management page displays extension contacts and external contacts information.

- Extension contacts

Extension contacts page shows all the extensions that has "Sync Contact" option enabled in extension settings page. The extension contacts here can be edited or deleted individually or in batch. No new extension contact can be added directly from this page. If an extension contact is deleted from this page, "Sync Contact" option is disabled from this extension. This will not delete the extension from IPPBX.

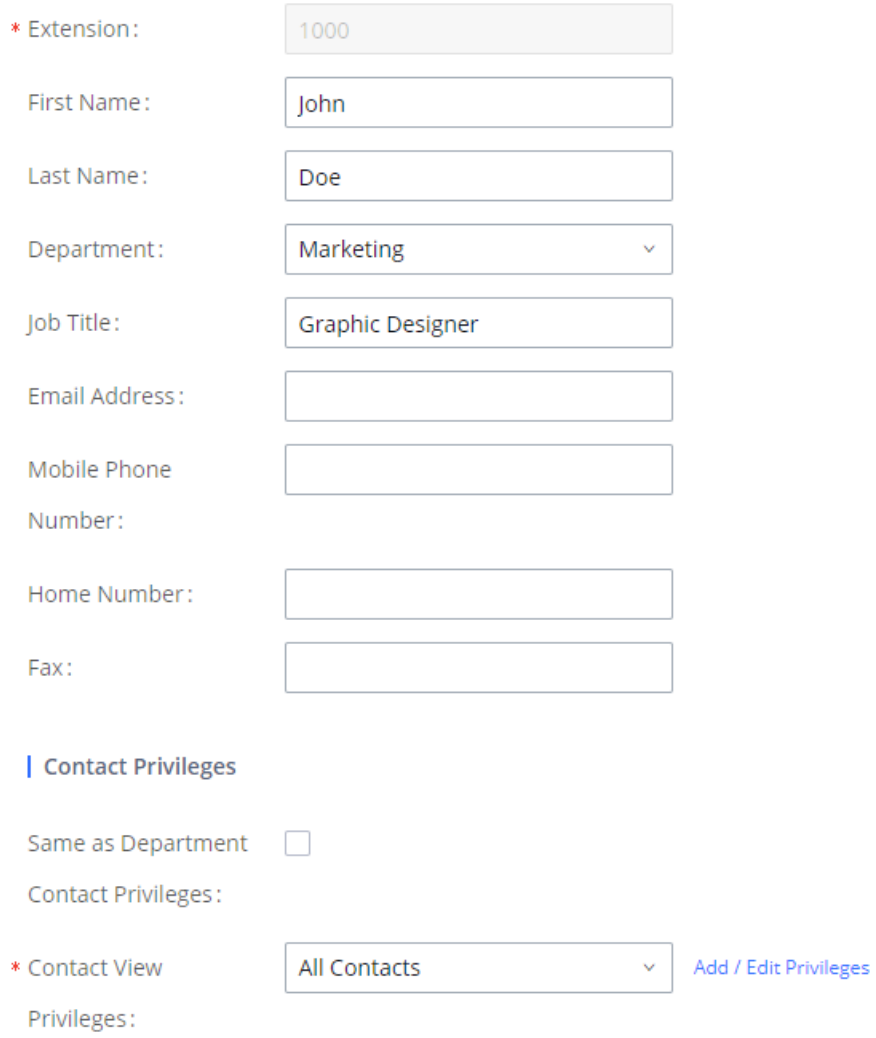
"Delete" extension contact will only remove this extension from extension contact page and it will not sync to contacts on IPPBX. The extension itself still exists on IPPBX.



Extension Contacts

"Delete" extension contact will only remove this extension from extension contact page and it will not sync to contacts on IPPBX. The extension itself still exists on IPPBX.

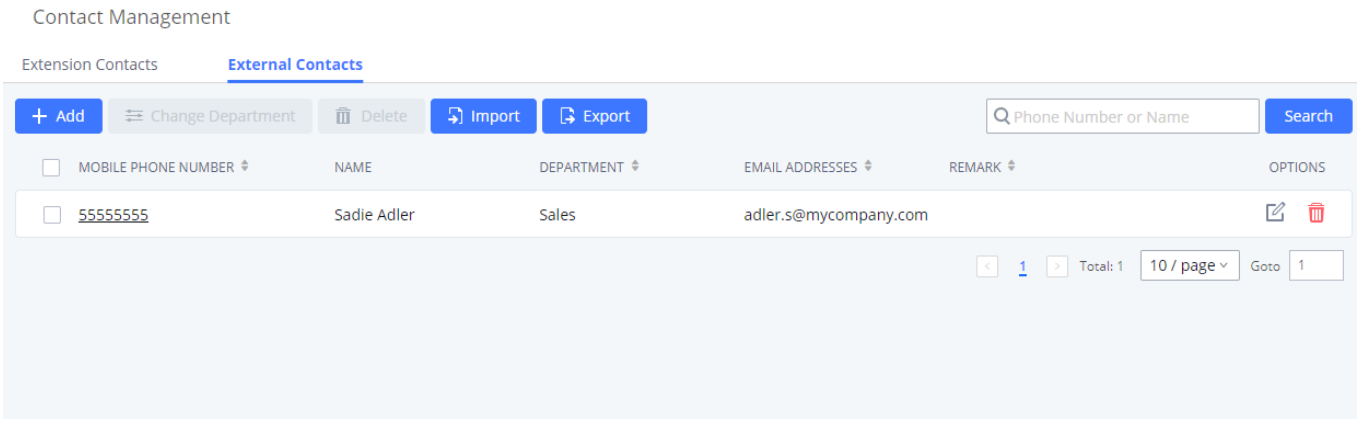
Click Edit icon to configure name, department, email address and etc for each extension contact.



Edit Extension Contact

- External contacts

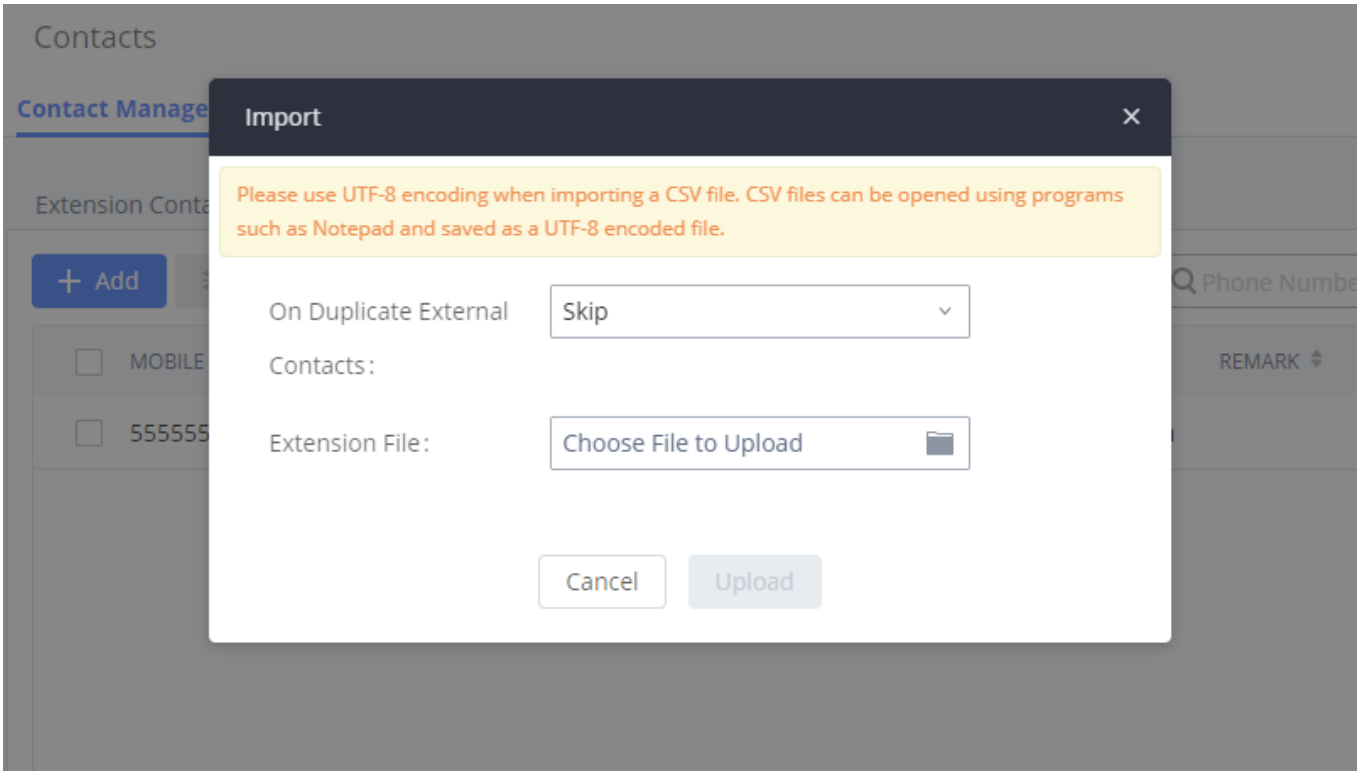
On external contacts page, the admin can create single external contact, import contacts in batch, edit contacts, delete contacts and export contacts.



Click on "Export" icon, a CSV format file will be generated with the current external contacts.

Click on "import" icon, then follow the steps below to add external contacts in batch:

- **Step 1**: For option "On Duplicate External Contacts", select whether to skip duplicate contact on the imported CSV file or update the duplicate IPPBX contact with the information in the CSV.
- **Step 2**: Choose file from local PC to upload.
- **Step 3**: Click on "Upload".
- **Step 4**: Click on "Apply" to complete importing external contacts.



Import External Contacts

Department Management

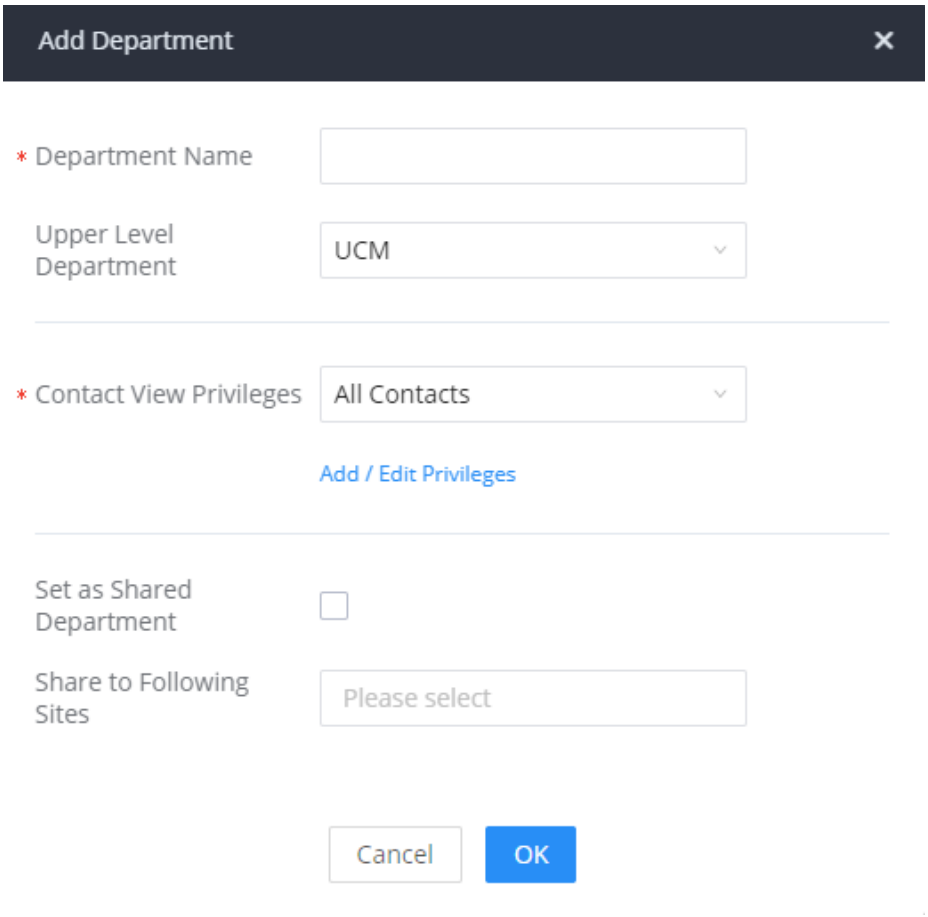
Departments are organizational units that allows organizing extensions within groups that specify the specialty of a the extension owners within a company. This makes finding contacts easier within the IPPBX contact books.

Department Management

Click on "Add" to create a new department. Configure the department name and select the superior department. By default the superior department is the root directory. If the IPPBX has cloud IM configured, the root directory will be the department in cloud IM.

On the department list:

- Click on to create sub department.
- Click on to add member to the department.
- Click on to edit the department.



Edit Department

If the IPPBX has Cloud IM enabled and configured, the following options will be available:

- **Set as Shared Department:** Toggles the shared status of the department. If there are existing cross-server members in this department, this option will be grayed out.
- **Share to Following Sites:** Select the sites (servers) to that will be allowed to move their extensions in and out of the shared deparmtent.
- **Apply Settings to Sub-Level Departments:** Only available when editing an exisiting deparmtent. Apply the same sharing settings to its sub-department. If a department has sharing enabled and is configured to be shared with specific sites, its sub-department will also have sharing enabled and be shared with the same sites.

Shared departments can be distinguished by an icon next to their name.

Shared Department Symbol

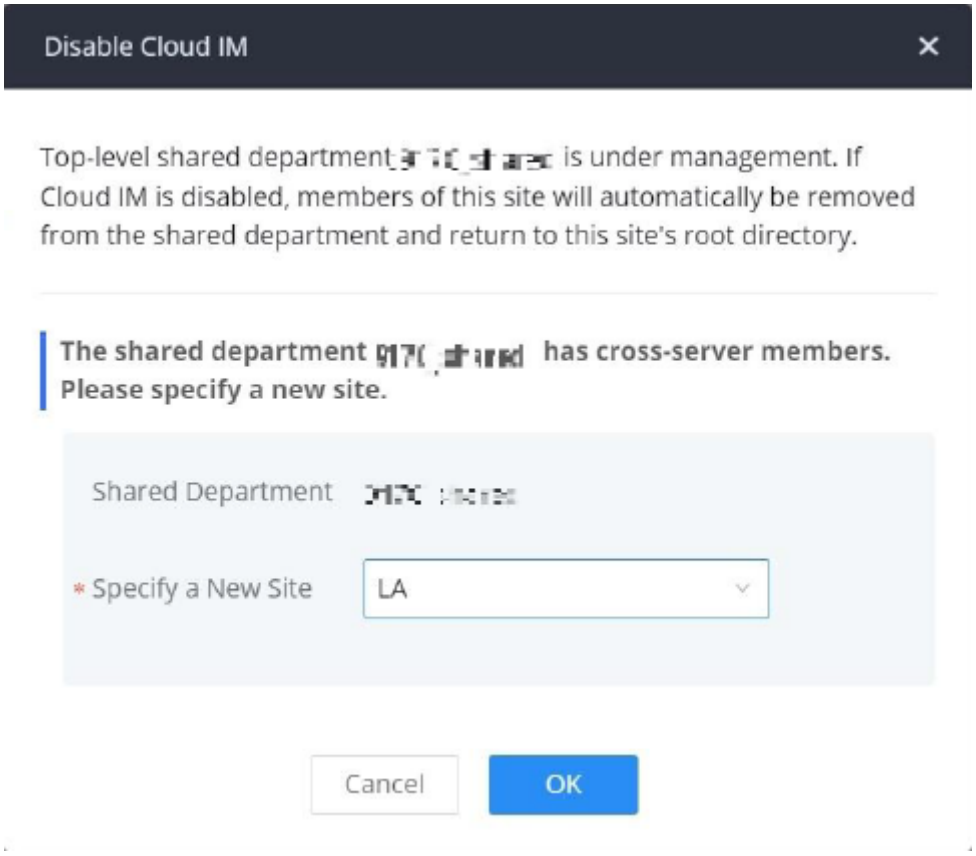
There are 3 potential marks in the center of that icon that represent the permission level that the current IPPBX has for that site

- : The + symbol indicates that the current IPPBX is the creator of the shared department. It has full management priviliges over it, which include the ability to edit it, delete it, add sub-departments under it, add its own extensions to it and remove **any** extensions from it, including extensions from other servers.
- : The checkmark symbol indicates that the department was shared to the current IPPBX has been authorized to move its own extensions in and out of department.
- : No symbol indicates that the department is from a IPPBX that is under the same Cloud IM account as the current IPPBX, but it has not been shared to the current IPPBX for management. The current IPPBX can only view this shared department and its members but cannot interact with it unless the shared depratment has the current IPPBX's extensions as members. In this case, the current IPBX can still remove members from the shared department.

- Removing a member from a shared department will move it to the root directory of its original site.
- A shared department cannot be deleted unless all cross-server memebers are removed from it

If a IPPBX with a shared depratment no longer wants to use CloudIM, it must first do the following:

- If there are no cross-server memebers in its shared departments, it must set them as local departments and assign new parent departments to them.
- If there are existing cross-server members in its shared departments, it must assign the shared departments to another site for management.



Assign Shared Department

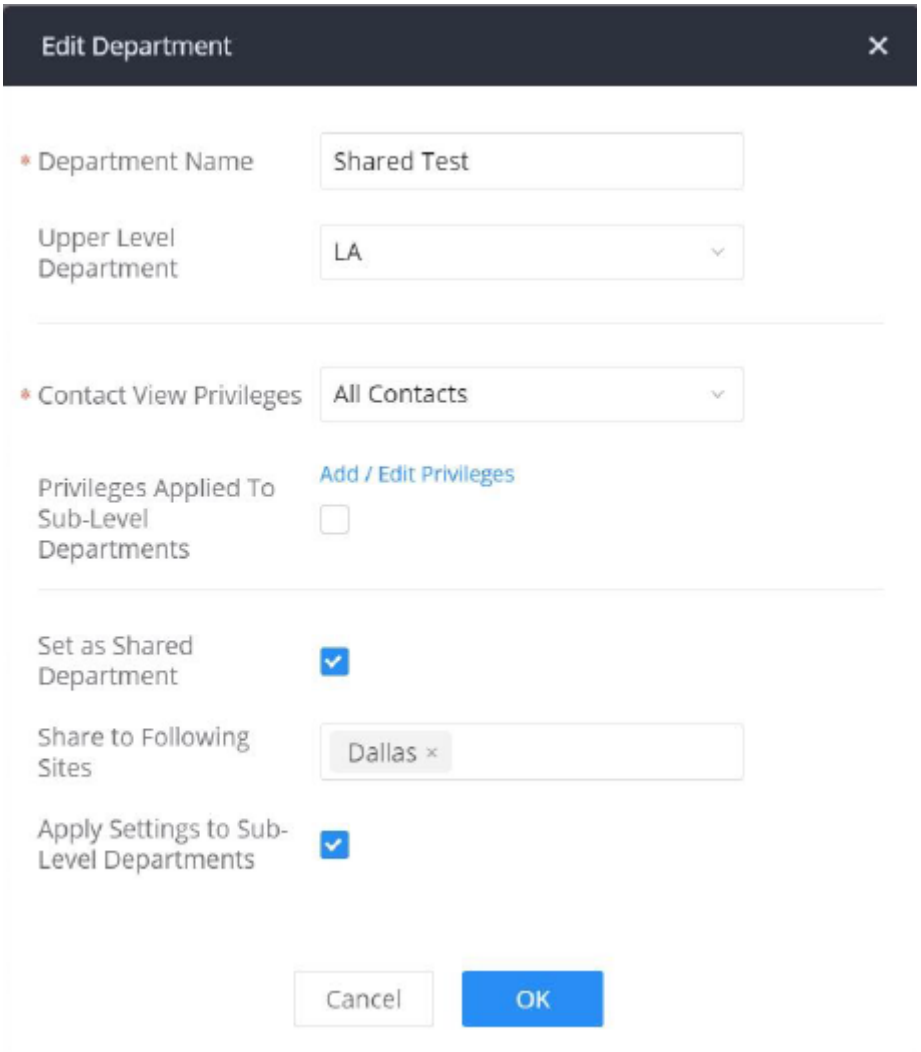
If the IPPBX re-enables Cloud IM, it will not reclaim ownership of its original shared departments.

The user can create up to 100 departments with up to 4 levels of nested departments.

Shared Departments

Share departmens offer a centralized way for administrators to manage departments and members across multiple servers using the same Cloud IM account. To get started, make sure that the IPPBX has Cloud IM enabled and configured. Then, navigate to **Contacts -> Department Management**, then add/edit a specific department.

Tick the option **Set as Shared Department**, then select the sites to which the department will be shared. If the user wishes to apply the same sharing settings to the departments nested under the parent department, please tick the option **Apply Settings to Sub-Level Departments.**



Edit Department

Privilege Management

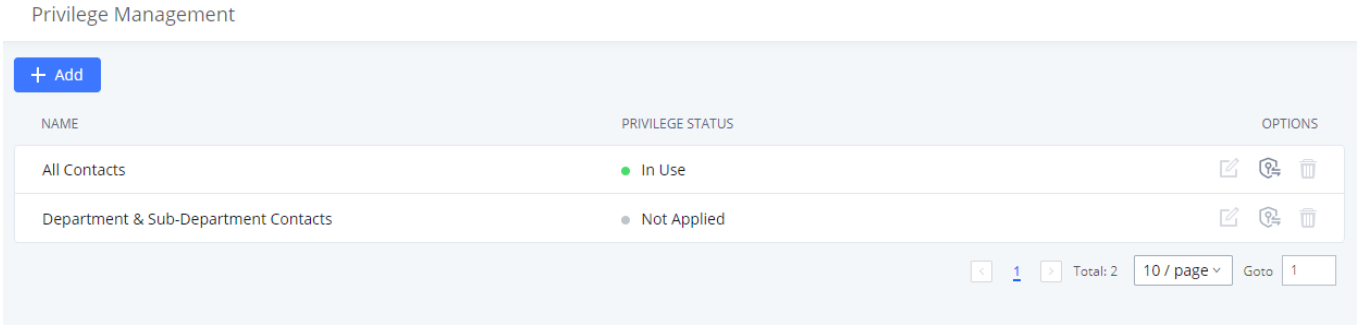
The user can configure custom privileges other than the default ones (All contacts, Departments and sub-departments contacts). These custom privileges allow more flexible ways of allowing contacts to view all or specific contacts from other departments.

IPPBX admin can add or edit Privilege Management; under IPPBX web UI→**Contacts Privilege Management**, there are 2 default privileges:

- Visible to all contacts.
- Only the contact person's department and sub-department contacts are visible.

When Cloud IM is enabled on the IPPBX, a third privilege is created:

- Local Contacts: Restricts the contacts shown to the contacts of the local IPPBX.

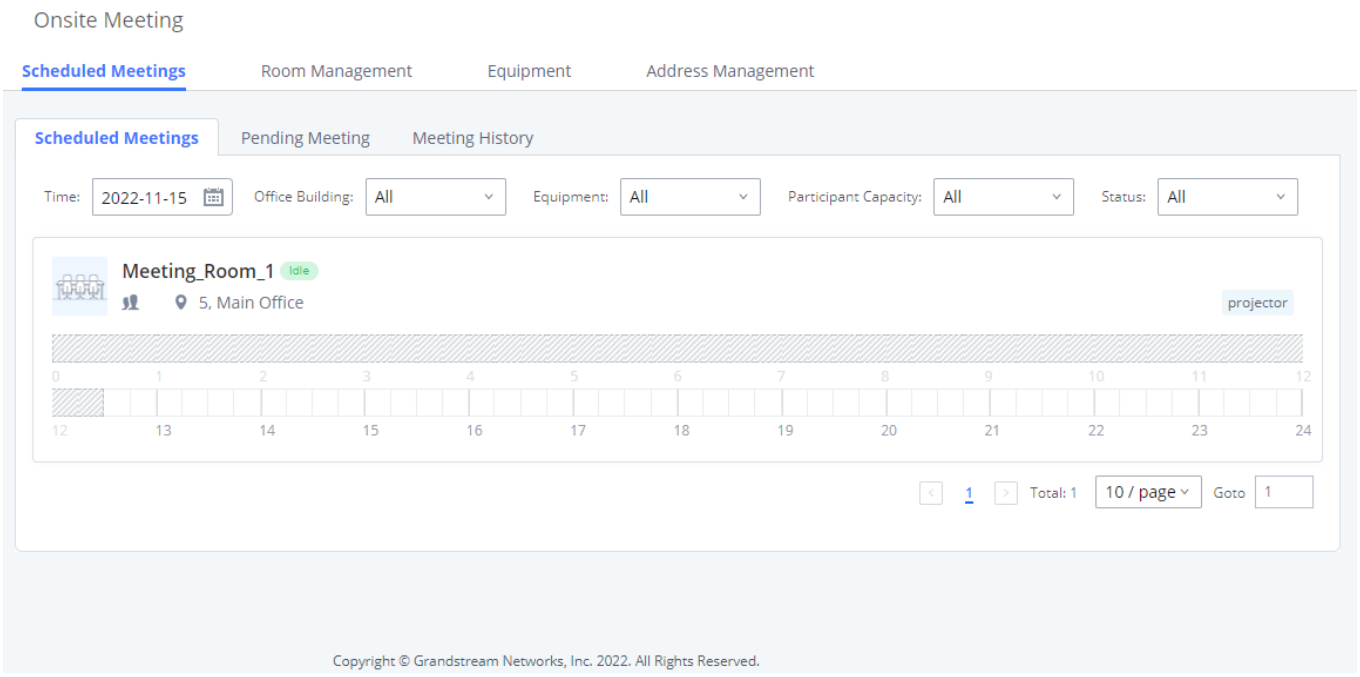


Privilege Management Cloud IM Disabled

DEVICE MANAGEMENT

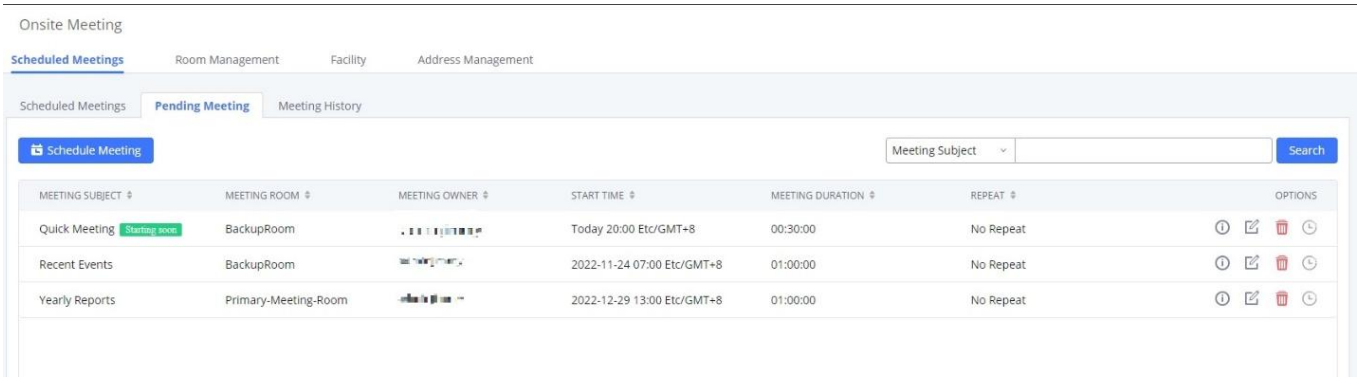
Onsite Meeting

For workplaces that require employees to return to physical offices for work, Grandstream IPPBX offers the Onsite Meetings feature, a new way to stay organized and keep up-to-date with in-person meetings. This feature allows administrators to create and manage onsite meeting rooms, specify meeting room locations, schedule meetings, and add conferencing equipment. The new feature can be found under the **Device Management > Onsite Meeting** page. The first page that appears is the **Scheduled Meetings** page and tab page, which provide an overview of all created meeting rooms. It provides information about the rooms' meeting schedules for the day, their locations, their member capacity, and their equipment.



Schedule Onsite Meetings

The **Pending Meeting** tab and **Meeting History** tab show detailed information about upcoming meetings and previous meetings respectively. From the **Pending Meeting** tab, users can delete upcoming meetings and extend the duration of ongoing meetings. **The Meeting History** tab will display the last 6 months of onsite meetings.



Pending Onsite Meetings

MAINTENANCE

Maintenance allows the user to manage the PBX device acces privileges, logs, data and software, as well as diagnose any issues that the network or the device itself might encounter.

User Management

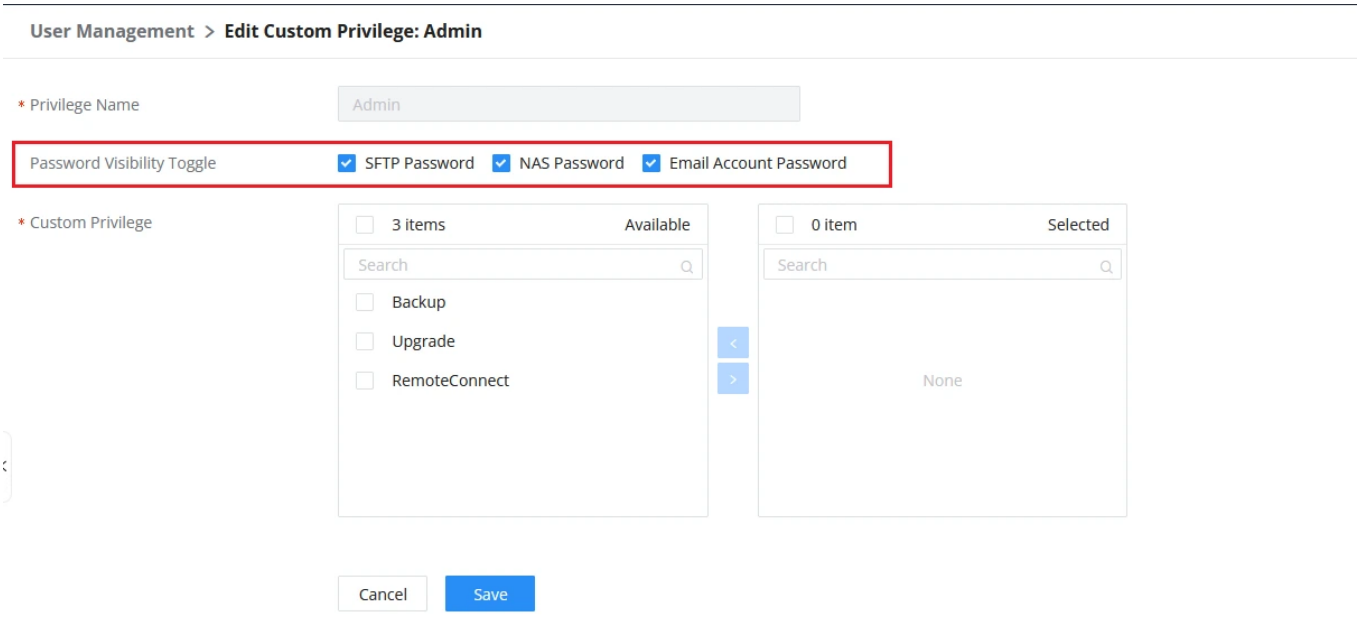
The configuration in this page is related to general management of the users who are accessing the device directly.

Custom Privilege

By default, three levels are supported: **Super Administrator**, **Administrator**, **Wave Administrator**:

Super Administrator

- This is the highest privilege. Super Admin can access all pages on IPPBX Web GUI, change configuration for all options and execute all the operations.
- Super Admin can create, edit, and delete one or more users with "Admin" privilege
- Super Admin can edit and delete one or more users with "Consumer" privilege
- Super Admin can view operation logs generated by all users.
- By default, the user account "admin" is configured with "Super Admin" privilege and it is the only user with "Super Admin" privilege. The Username and Privilege level cannot be changed or deleted.
- Super Admin could change its own login password on Web GUI→**Maintenance→Login Settings** page.
- Super Admin could view operations done by all the users in Web GUI→**Maintenance→User Management →Operation Log**
- The Super Admin can administrators to view SFTP, NAS, and Email Account passwords.



Password Visibility Toggle

Administrator

Users with "Admin" privilege can only be created by "Super Admin" user.

"Admin" privilege users are not allowed to access the following pages:

Maintenance→Upgrade

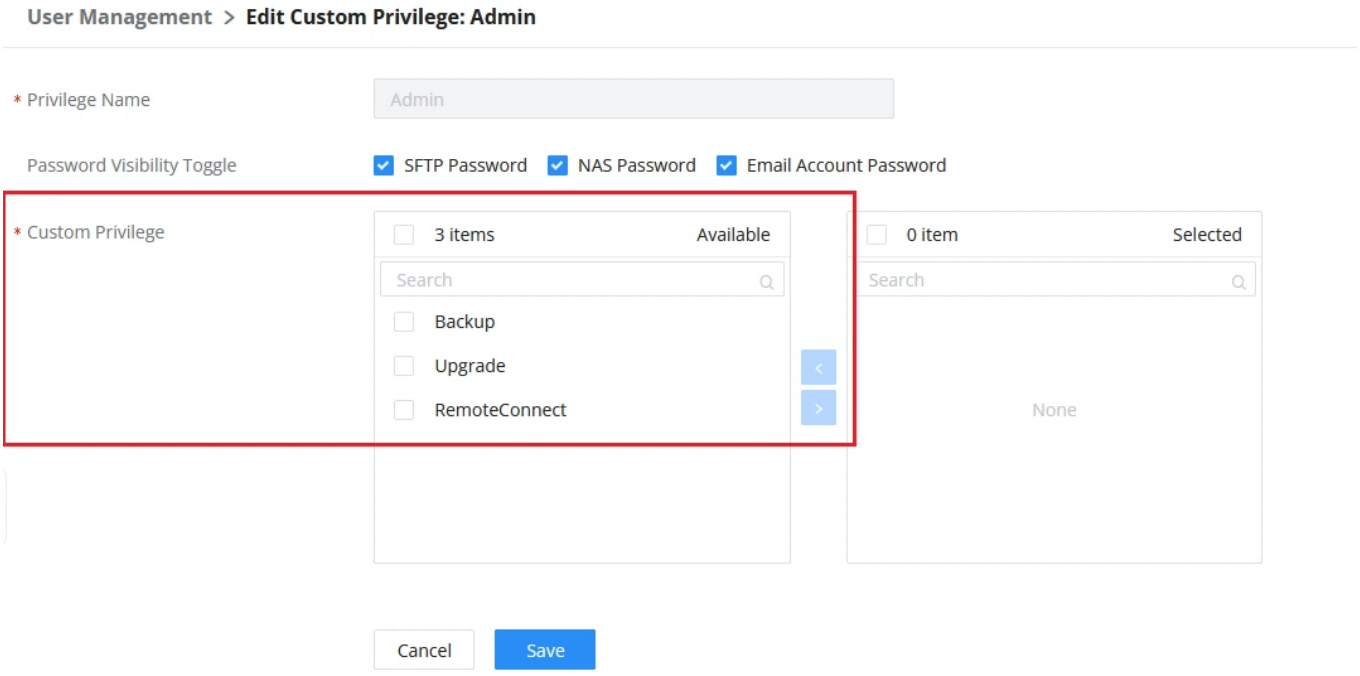
Maintenance→Cleaner

Maintenance→Reset/Reboot

Settings→User Management→Operation Log

- "Admin" privilege users cannot create new users for login.

Note: By default, administrator accounts are not allowed to access backup menu, but this can be assigned to them by editing the option "**Maintenance → User Management → Custom Privilege**" then press to edit the "Admin" account and include backup operation permission for these types of users.

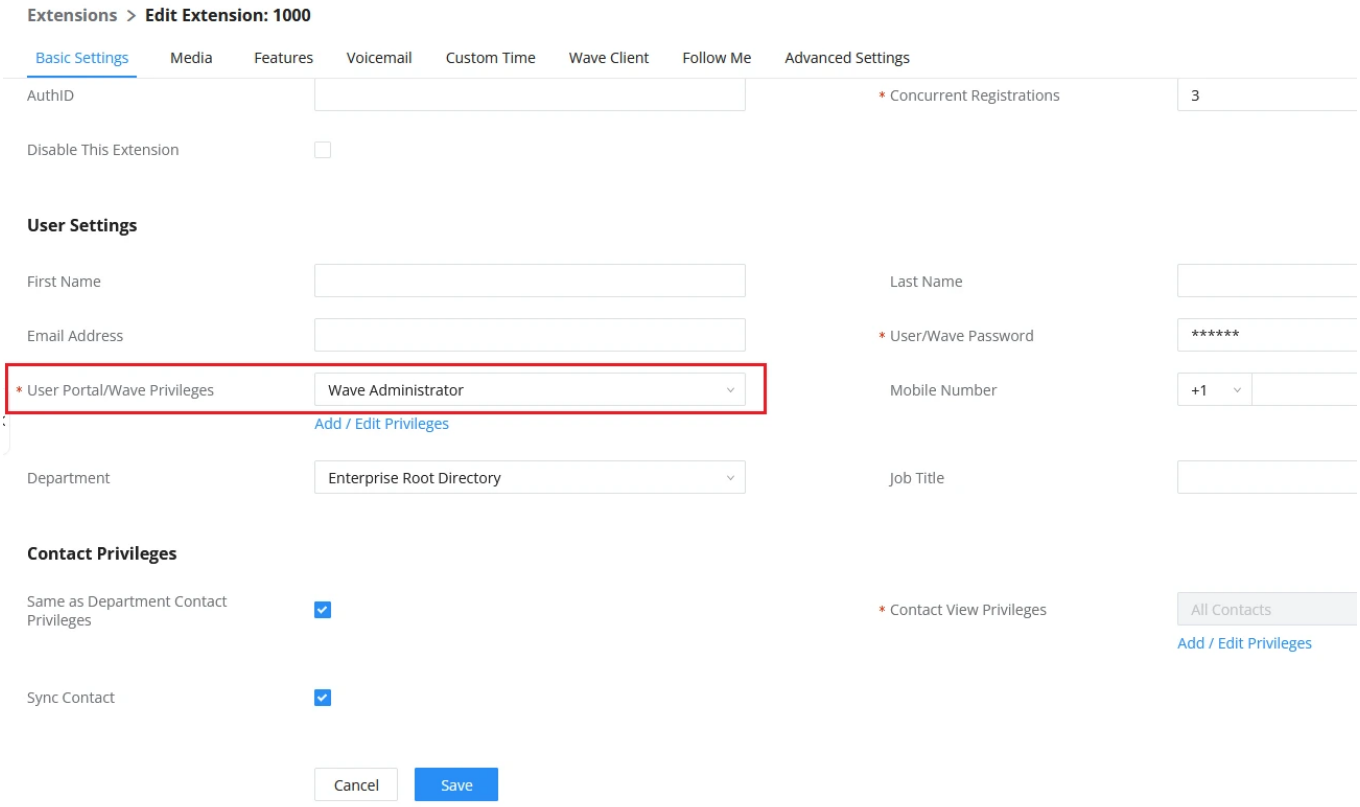


Assign Backup permission to "Admin" users

Wave Administrator

- This permission type does not support editing or deletion.
- This includes management of Wave-related function settings only and does not involve access to the PBX module.
- Users can set the Wave Admin privilege for specific extension under **Extension/Trunk → Extensions → Edit Extension**.

For more information, please refer to the Wave Administrator Guide guide.



User Portal/Wave Privileges

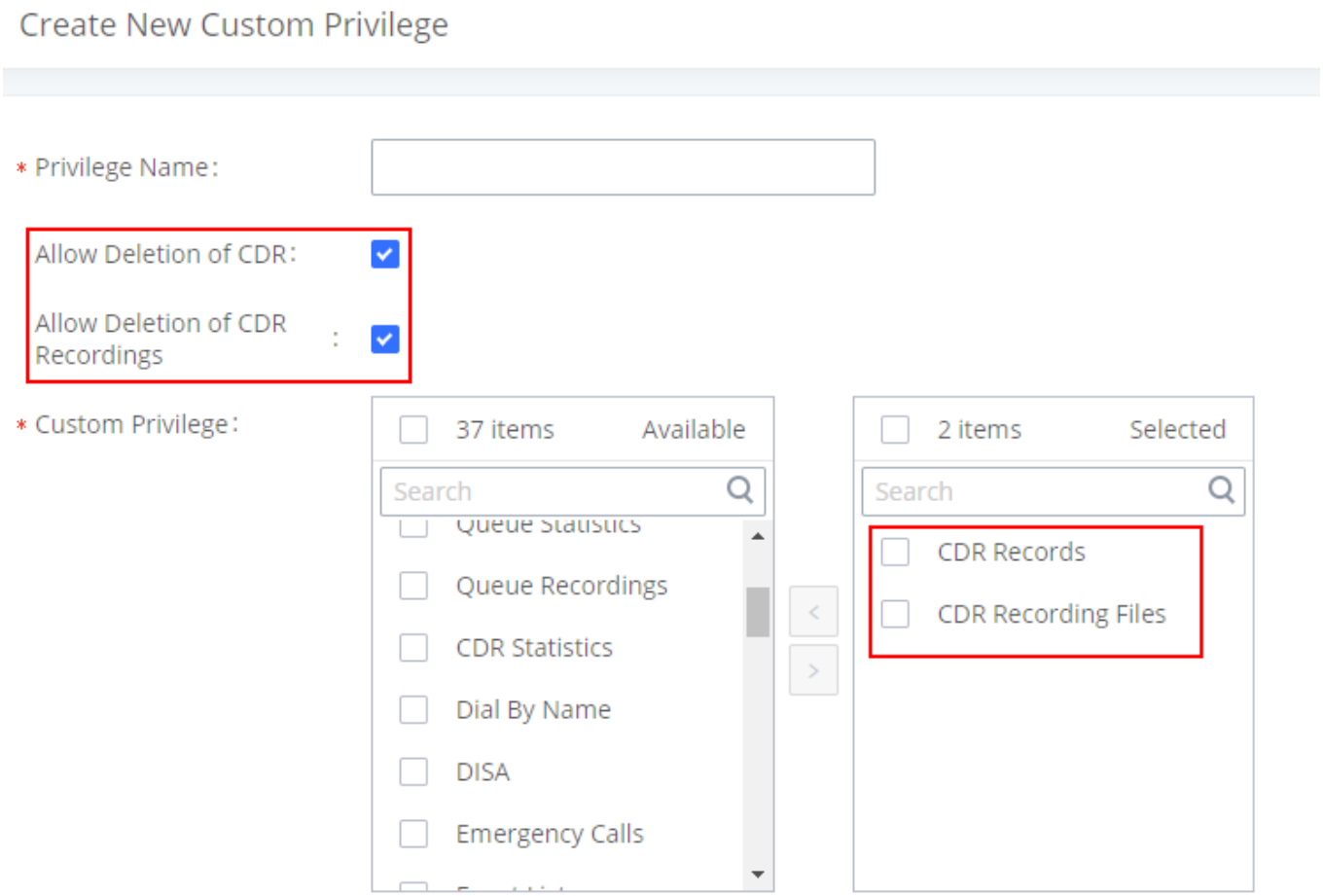
Custom Privilege

API Configuration	Backup	Callback	Call Queue	Queue Statistics
Queue Recordings	CDR Recordings	CDR Records	CDR Statistics	Dial By Name
DISA	Emergency Calls	Event List	Extensions	Extension Groups
Outbound Routes	Inbound Routes	Fax/T.38	Fax Sending	Feature Codes
IVR	Paging/Intercom	Parking Lot	Pickup Groups	PMS - Wakeup Service
Ring Groups	Restrict Calls	SCA	Speed Dial	System Status
System Events	LDAP Server	Time Settings	Multimedia Meeting	Voicemail
Voice Prompt	Schedule Call	Contacts	Zero Config	Announcement
RemoteConnect				

The Super Admin user can create users with different privileges.

Log into the GCC device as super admin and go to **Maintenance→User Management→Custom Privilege**, create privilege with customized available modules.

When you add CDR Records and CDR Recording Files custom privileges, additional privileges will appear (All Deletion of CDR and Allow Deletion of DCR Recordings , respectively). This offers more flexibility on the privileges that the admin assigns to the user.

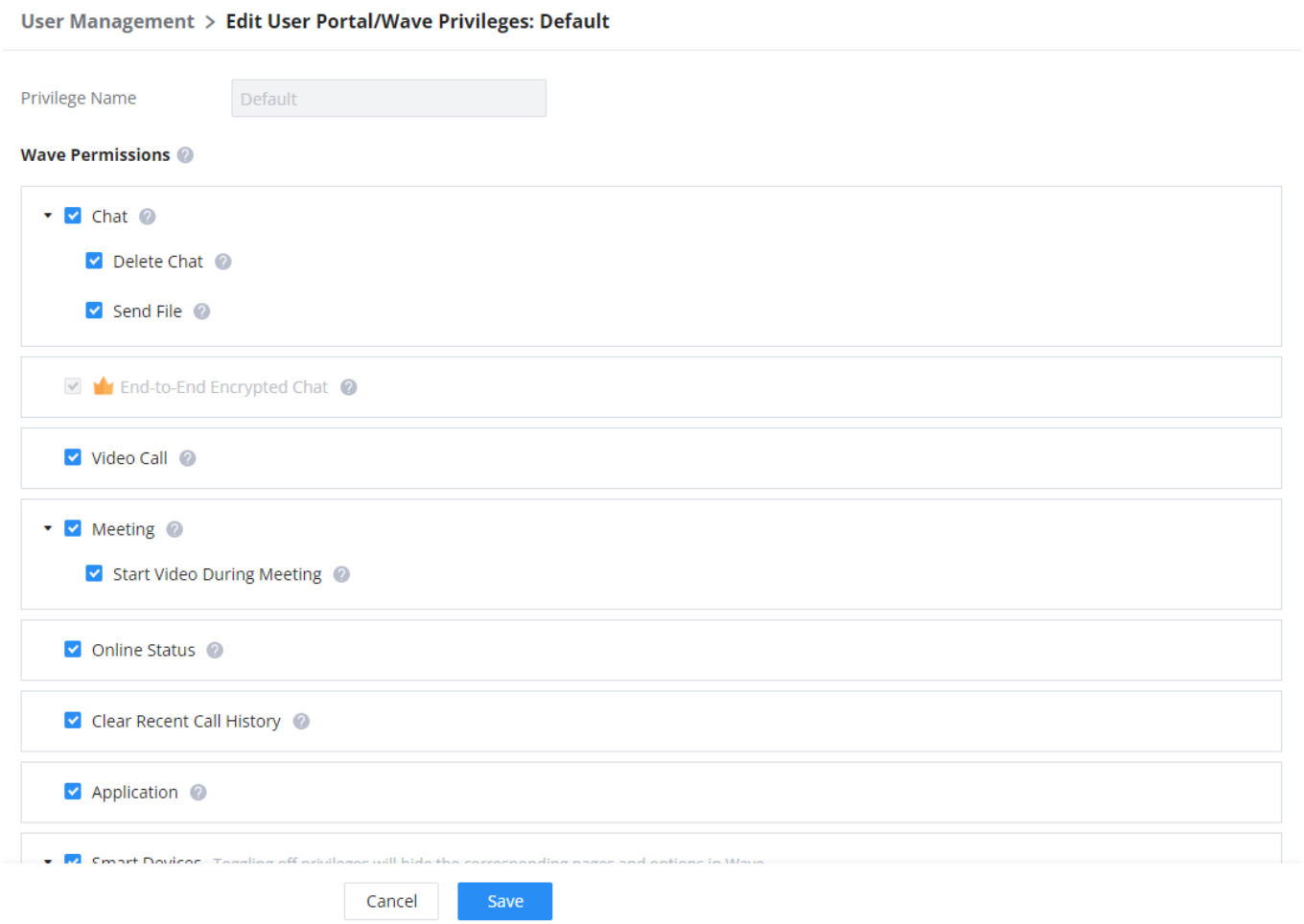


Create New Custom Privilege

To assign custom privilege to a sub-admin, navigate to Web GUI→**Maintenance→User Management→ User Information→**Create New User/Edit Users, select the custom privilege from "Privilege" option.

User Portal/Wave Privileges

The user can create customize privileges related to an extension's User Portal and Wave. The created privilege can be affected to the extensions to limit or allow them to use certain functionalities related to Wave and the User Portal.



User Portal/Wave Privileges

Wave Permissions

- **Chat:** Toggles ability to use the Wave Chat feature.
    - **Delete Chat:** Toggles support for Wave to delete chats and chat history. This data will only be deleted on the Wave client side.
    - **Send File:** Toggles file/image sending support in Wave chat. If disabled, users will still be able to download, view and forward chat files.
    - **Download Chat Logs:** If enabled, chat logs will be downloadable from the Wave client, including chat logs from Wave/WhatsApp/Telegram/LiveChatsessions.
- **End-to-End Encrypted Chat*:** Toggles ability to use the Wave End-to-End Encrypted Chat feature.
- **Video Call:** Toggles ability to use the Wave Video Call feature.
- **Meeting:** Toggles ability to use the Wave Meeting feature.
    - **Start Video During Meeting:** Toggles ability to use the Wave Start During Meeting feature.
- **Online Status:** Toggles ability to set Wave online status such as "Busy", "Appear Away", "Do Not Disturb", "Appear Offline", etc. If unchecked, the status will be displayed as only either "Online" or "Idle".
- **Remote Logout:** If enabled, Wave users will be able to log out of their accounts from other logged-in devices.
- **Clear Recent Call History:** Toggles ability to delete recent call history entries and entire recent call history on Wave.

- **Application:** Toggles ability to access the "Applications" page under Wave Desktop and Wave Web.
- **Smart Devices:** Toggling off privileges will hide the corresponding pages and options in Wave.

    - **Door System**
    - **Monitor**
    - **Call Device (CTI)**
- **3rd Party Applications**

    - **App Store**: Toggles ability to access the Wave App Store. If unchecked, the App Store will be hidden, but installed apps can still be used.
    - **Pre-installed Apps***: Configure Wave pre-installed add-ins and related settings.

User Portal/Wave Privileges

- **Account Settings:** If unchecked, the User Portal -> Basic Information -> Account Settings page and the Wave -> Sidebar -> User -> Account Settings option will be hidden.
- **Extension Settings:** If unchecked, the extension's User Portal->Basic Information->Extensions page and the Wave->Sidebar->User->Call Settings option will be hidden.

    - **Do Not Disturb:** Toggles ability to set DND through the User Portal.
    - **SIP/IAX Password & AuthID**: Toggles ability to access the **SIP/IAX Password** and **AuthID** settings under the User Portal->Basic Information->Extensions->Basic Settings page.
    - **Configuration Voicemail**
- **Deleting Recordings:** Toggles ability to delete recordings through the User Portal and Wave. For Wave, this includes the ability to delete call logs, meeting details,and recordings.
- **Personal Data:** Toggling off privileges will hide the corresponding pages and options in the User Portal and Wave.

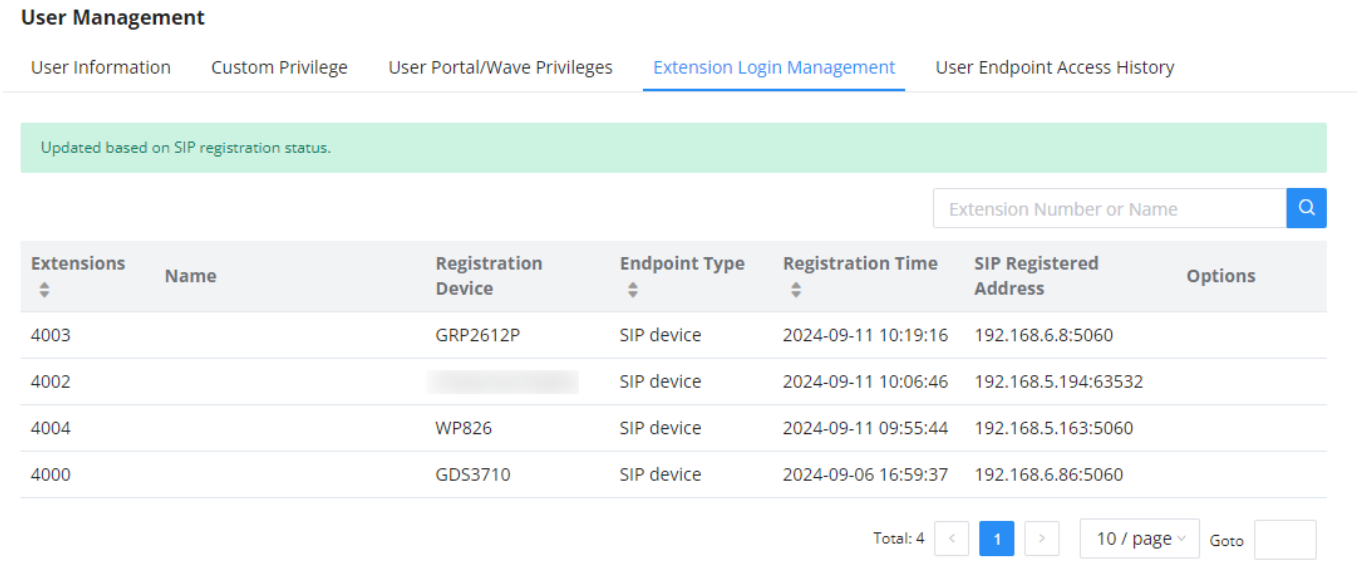
    - CDR
    - Follow Me
    - Voicemail
    - Recordings files
    - Fax Files
    - SCA
- **Other Features:** Toggling off privileges will hide the corresponding pages and options in the User Portal and Wave.

    - Fax Sending
    - Call Queue
    - Schedule Call

*: Features which are marked by an asterisk are a part of RemoteConnect plans.

Extension Login Management

Extension Login Management allows the administrator to review the logged-in sessions of SIP devices and Wave.

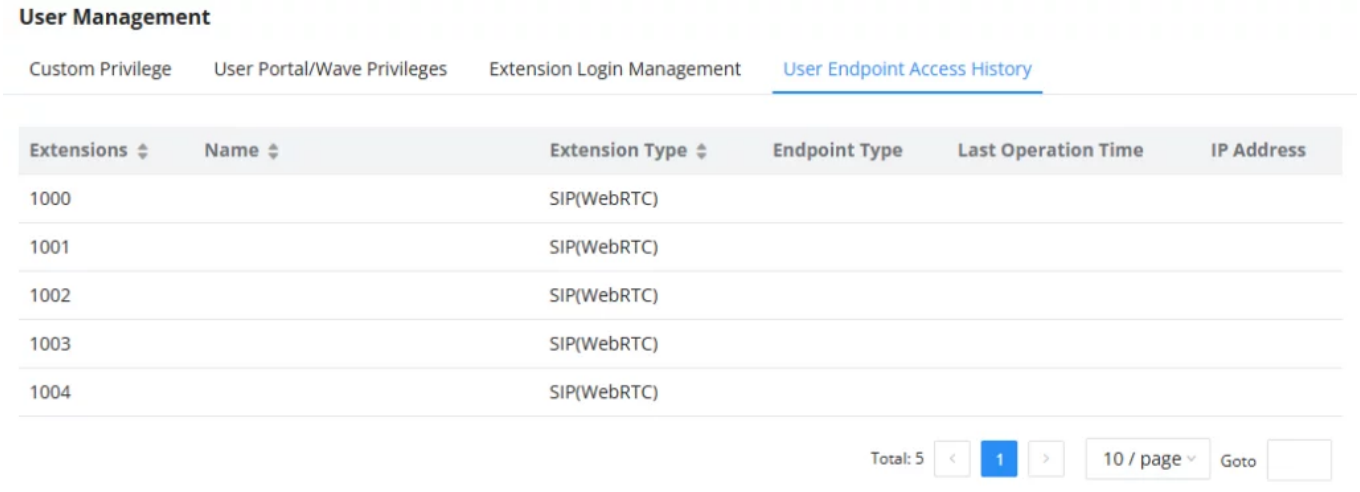


Extension Login Management

For Wave sessions, the administrator can click on to terminate a Wave session. SIP sessions cannot be logged out.

User Endpoint Access History

The User Endpoint Access History tab allows the administrator to view the access history of all extensions, the time on which the access has occurred, the IP addresses from which the extensions were accessed, and whether they were accessed from the User Portal, Wave Web/Desktop, or mobile. Extension access from the SIP endpoints won't be logged in this page.

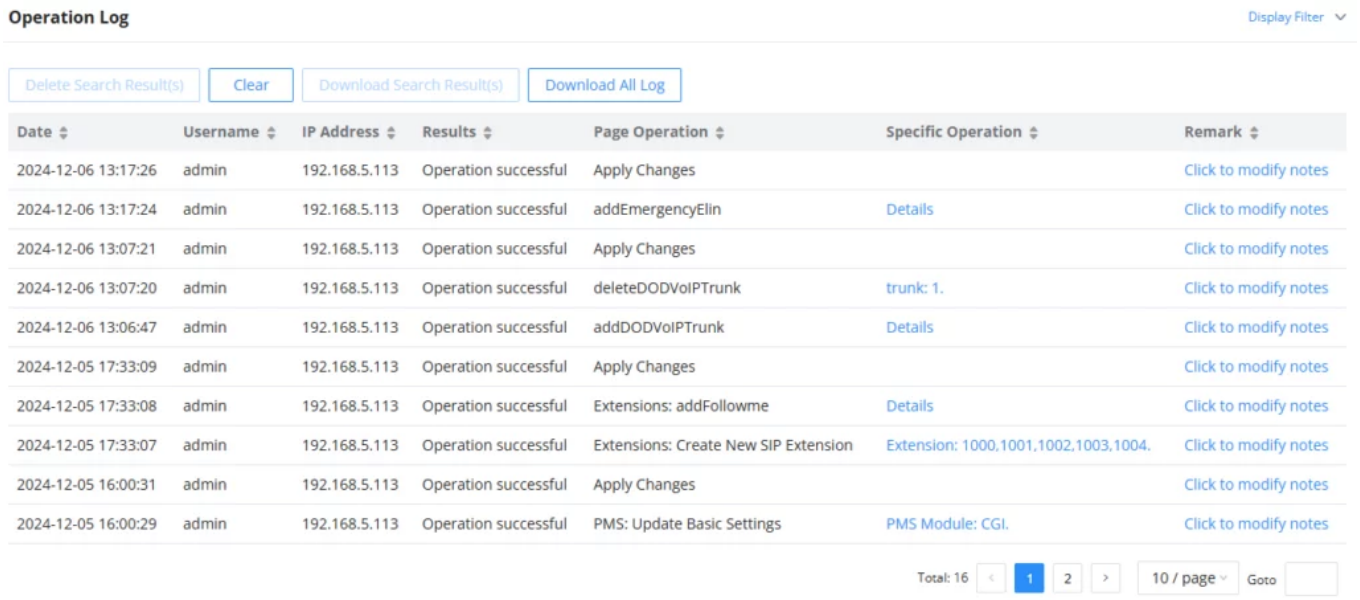


User Endpoins Access History

Operation Log

Super Admin has the authority to view operation logs on GCC device (PBX Module) Web GUI→ **Maintenance→User Management→Operation Log** page. Operation logs list operations done by all the Web GUI users, for example, Web GUI login, creating trunk, creating outbound rule etc. There are 7 columns to record the operation details "Date", "Username", "IP Address", "Results", "Page Operation", "Specific

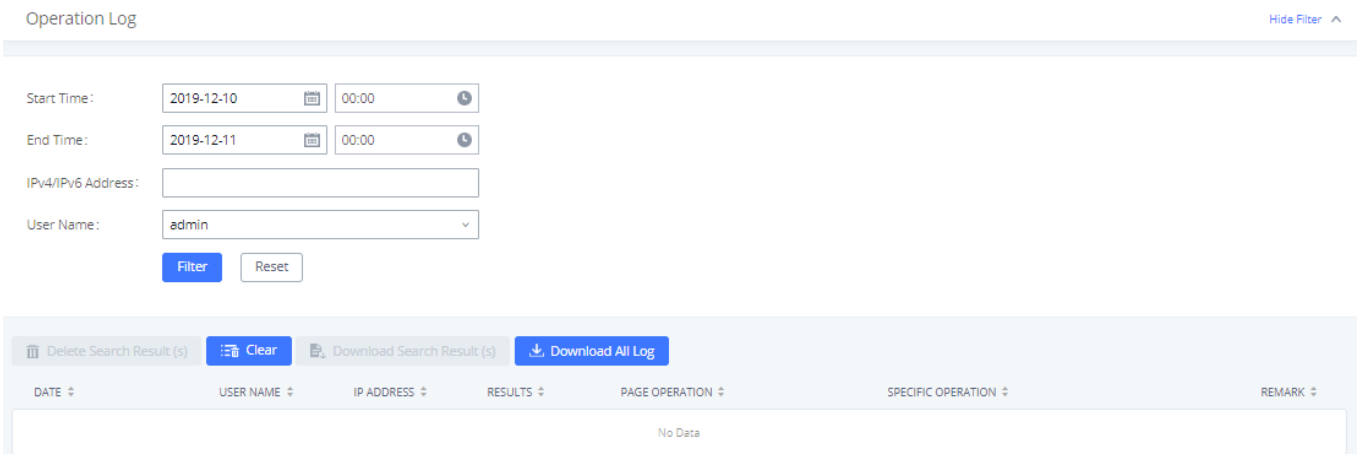
Operation" and "Remark".



Operation Logs

The operation log can be sorted and filtered for easy access. Click on or at the top of each column to sort. For example, clicking on for "Date" will sort the logs according to newer operation date and time. Clicking on for "Date" will reverse the order.

User could also filter the operation logs by time condition, IP address and/or username. Configure these conditions and then click on "Display Filter".



Operation Logs Filter

The above figure shows an example that operations made by user "support" on device with IP 192.168.40.173 from 2014-11-01 00:00 to 2014-11-06 15:38 are filtered out and displayed.

To delete operation logs, users can perform filtering first and then click on to delete the filtered result of operation logs. Or users can click on to delete all operation logs at once.

Syslog

On the IPPBX, users could dump the syslog information to a remote server under Web GUI→**Maintenance →Syslog**. Enter the syslog server hostname or IP address and select the module/level for the syslog information as well as Process Log Level.

The default syslog level for all modules is "error", which is recommended in your IPPBX settings because it

can be helpful to locate the issues when errors happen.

Some typical modules for IPPBX functions are as follows and users can turn on "NOTICE" and "VERBOSE" levels besides "error" level.

- **pbx**: This module is related to general PBX functions.
- **pjsip**: This module is related to SIP calls.

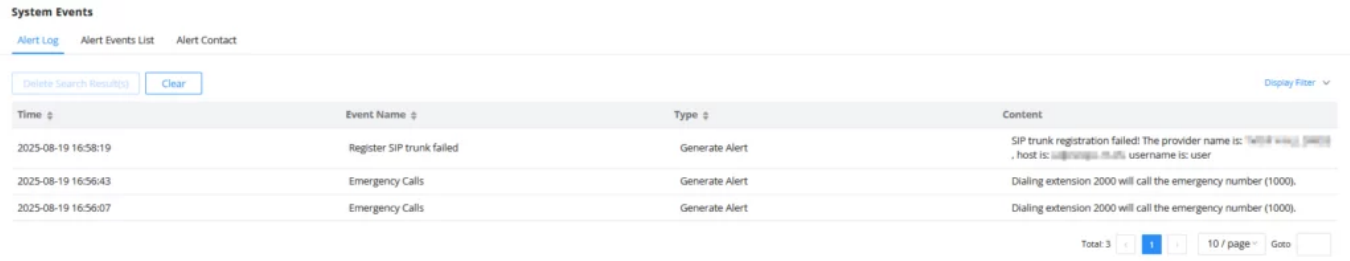
- Syslog is usually for debugging and troubleshooting purpose. Turning on all levels for all syslog modules is not recommended for daily usage. Too many syslog prints might cause traffic and affect system performance.
- The reserved size for Syslog entries on the cache memory of the IPPBX is 50M, once this sized is reached the IPPBX will clean up 2M of the oldest Syslog entries to allow to save new logs.

System Events

The GCC6xxx can monitor important system events, log the alerts, and send Email notifications to the system administrator.

Alert Log

Under **Maintenance** -> **System Events** -> **Alert Log**, system messages from triggered system events are listed as alert logs.



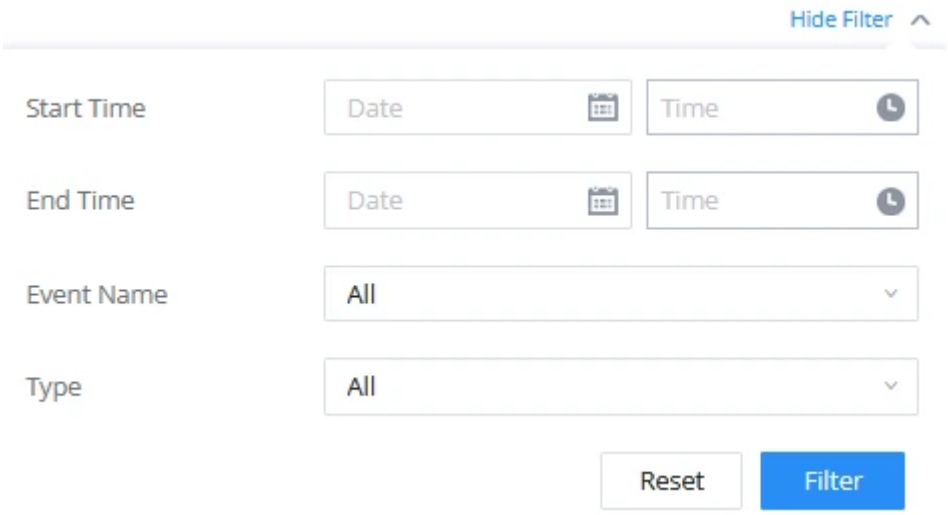
System Events -> Alert Log

User could also filter alert logs by selecting a certain event category, type of alert log, and/or specifying a certain time period. The matching results will be displayed after clicking on "Display Filter" . Alert logs are classified into two types by the system:

1. **Generate Alert:** Created when an alert condition is triggered (e.g., unauthorized access attempt, system threshold reached).

2. **Restore to Normal:** Created when the alert condition is cleared (e.g., system status returns to normal).

Users can filter logs by type to display only **Generate Alert** or only **Restore to Normal**, depending on their needs.



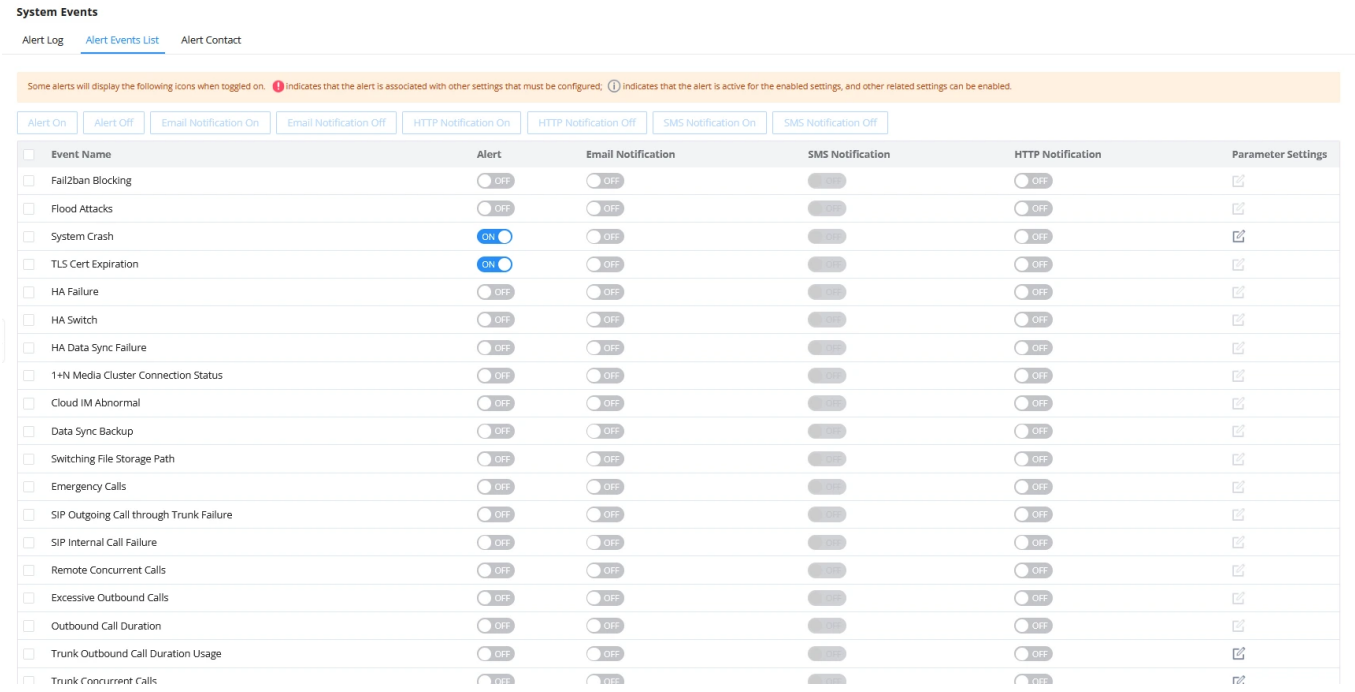
Filter for Alert Log

Alert Events List

The system alert events list is located under **Maintenance -> System Events -> Alert Events List**. The events described in this section, along with their corresponding actions, are currently supported on the GCC6xxx.

When these events occur, the system can generate alerts, email notifications, SMS notifications, and HTTP notifications.

Click on to adjust the parameters of events that support configuration.



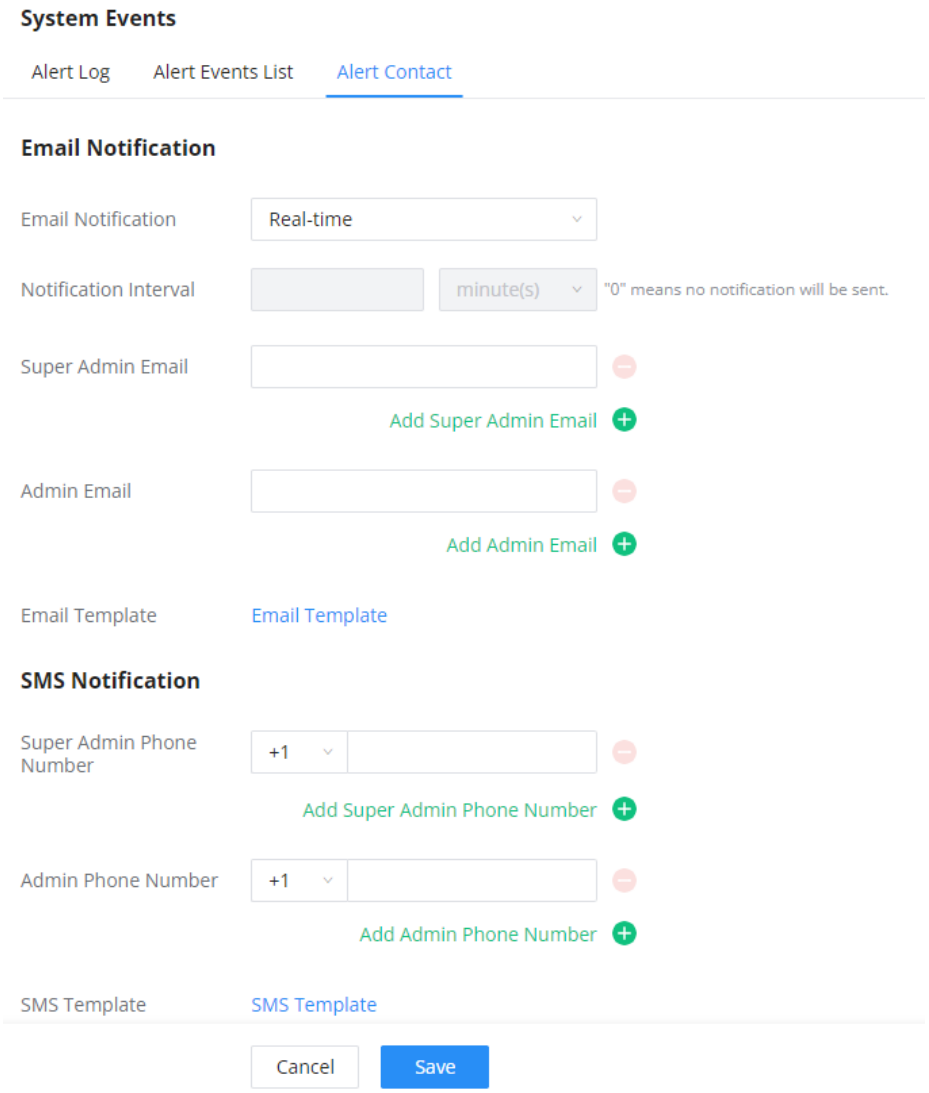
Alert Event List

Event Name	Description
Fail2ban Blocking	If the system Fail2ban is blocking, the event will be recorded in the alert log.
Flood Attacks	An alert will be generated in case a DDoS attack attempt is detected by the GCC. The event will be registered in the alert log and it will be pushed to the GDMS.

System Crash	The GCC will detect the event at each cycle based on the specified time. Users can enter the number and then select second(s)/minute(s)/hour(s)/day(s) to configure the cycle.
TLS Cert Expiration	Starting 7 days before the HTTP Server TLS certificate in the GCC device expires, an expiration countdown notification is sent every day; the certificate has expired, an expiration notification is sent; after the alarm notification is generated, a valid new certificate is uploaded, and a notification to restore the TLS certificate is generated.
HA Failure	After the HA dual-system hot backup disaster recovery function is enabled on the GCC device, the HA fault alarm is automatically turned on. When the device has a software and hardware related fault, an HA fault alarm is generated.
HA Switch	Once a switch between primary GCC device and secondary GCC has been detected, an alert event will be sent.
HA Data Sync Failure	In case of Data Sync failure in HA deployment, the GCC device will generate an alert event.
1+N Media Cluster Connection Status	Users can enable a connection status alarm (1+N Media Cluster Connection Status) to be notified when a media resource server goes online/offline.
Cloud IM Abnormal	An alert message will be generated f the Cloud IM encounter any issue or exhibit any abnormal behavior.
Data Sync Backup	If the system performs data synchronization and backup abnormalities, the event will be recorded in the alert log.
Switching File Storage Path	Once the file storage path on the GCC device is changed, an alert will be sent.
Emergency Calls	If the system generates an emergency call, the event will be recorded in the alert log.
SIP Outgoing Call through Trunk Failure	If the system SIP trunk outgoing call fails, the event will be recorded in the alert log.
SIP Internal Call Failure	If the system SIP extension call fails within the office, the event will be recorded in the alert log.
Remote Concurrent Calls	If the remote concurrent call fails, the event will be recorded in the alert log.
Excessive Outbound Calls	When an extension initiates calls frequently, an alert will be logged in the alert log and a notification will be pushed.
Outbound Call Duration	This alert is triggered if a call reaches the duration limit set on the outbound rule.
Trunk Outbound Call Duration Usage	This alert is triggered if a call reaches the duration limit set on the trunk.
Trunk Concurrent Calls	This alert is triggered if the number of concurrent calls is reached on a specific trunk.
Register SIP trunk failed	The GCC will detect the failure of SIP trunk registration at a set interval.
SIP Peer Trunk Status	If the SIP peer trunks status is abnormal, the event will be recorded in the alert log.
Register SIP failed	Configure the sending period of the SIP registration failure alert. The first registration failure alert of the same IP to the same SIP account will be sent immediately, and then no alerts will be sent for similar failure warnings in the cycle time. After the cycle time expires, an alert will be sent again to count the number of occurrences of similar SIP registration failure alerts during the cycle. When set to 0, alerts are always sent immediately.
SIP Lost Registration	If System SIP extension registration is lost, the event will be recorded in the alert log.Clicking on the edit button, the user can access to enable or disable Ignore Wave App Registration Loss.

Alert Contact

This feature allows the administrator to be notified when one of the Alert events mentioned above happens. Users could add administrator's Email address under **Maintenance** -> **System Events** -> **Alert Contact** to send the alert notification to an email (Up to 10 Email addresses can be added) or also specify an HTTP server where to send this alert.



Alert Contact

Email Notification	
Email Notification	Alert email notification delivery method: Real-time: Notifications will be sent out as soon as the alerts are generated. Periodic: Alerts generated within the configured "Notification Interval" time window will be queued up and sent all at once in a single notification.
Notification Interval	When the notification delivery method chosen is Periodic, this option will be available.? The sending frequency of alert email notifications. All alert events that have occurred within a send cycle will be sent.
Super Admin Email	Configure the email addresses to send alert notifications to.Up to 10 email addresses can be added.
Admin Email	Configure the email addresses to send alert notifications to.Up to 10 email addresses can be added.
Email Template	Please refer to section [Email Templates]
SMS Notification	
Super Admin Phone Number	Send alert notifications via SMS. Multiple phone numbers are supported.

Admin Phone Number	Send alert notifications via SMS. Multiple phone numbers are supported.
SMS Template	Click on the link to access the SMS templates page.
HTTP Notification	
Protocol	Protocol used to communicate with the server. HTTP or HTTPS.Default one is HTTP.
HTTP Server	The IP address or FQDN of the HTTP/HTTPS server.
HTTP Server Port	HTTP/HTTPS port
Warning Template	Customize the template used for system warnings.By default: {"action":"${ACTION}","mac":"${MAC}","content":"${WARNING_MSG}" }
Alarm Notification	Alert notification delivery method: Real-time: Notifications will be sent out as soon as the alerts are generated. Periodic: Alerts generated within the configured "Notification Interval" time window will be queued up and sent all at once in a single notification.
Notification Interval	When Alarm Notification method is set to Periodic, this option will be available. The frequency at which HTTP alert notifications are sent. All alert events that have occurred within a send cycle will be sent.
Resource Usage Notification Template	Customize the notification template to receive relevant alert information.By default: {"action":"${ACTION}","cpu":"${CPU_USED}","memery":"${MEM_USED}","disk":"${DISK_USED}","external_disk":"${EXTERNAL_DISK_USED}"}Note: The notification message with "action:0" will be sent periodically if Notification Interval is set.
Notification Interval	Modifies the frequency at which notifications are sent in seconds.No notifications will be sent if the value is "0". Default value: 20
Template Variables	${MAC} : MAC Address${WARNING_MSG} : Warning message${STATUS} : Alarm Status${TIME} : Current System Time${CPU_USED} : CPU Usage${MEM_USED} : Memory Usage${ACTION} : Message Type${DISK_USED} : Local Disk Usage${EXTERNAL_DISK_USED} : Disk Usage

Task Management

Task Management feature allows the user to schedule certain tasks to be performed at a configured date and time. The user can choose whether to perform this task once or repeatedly

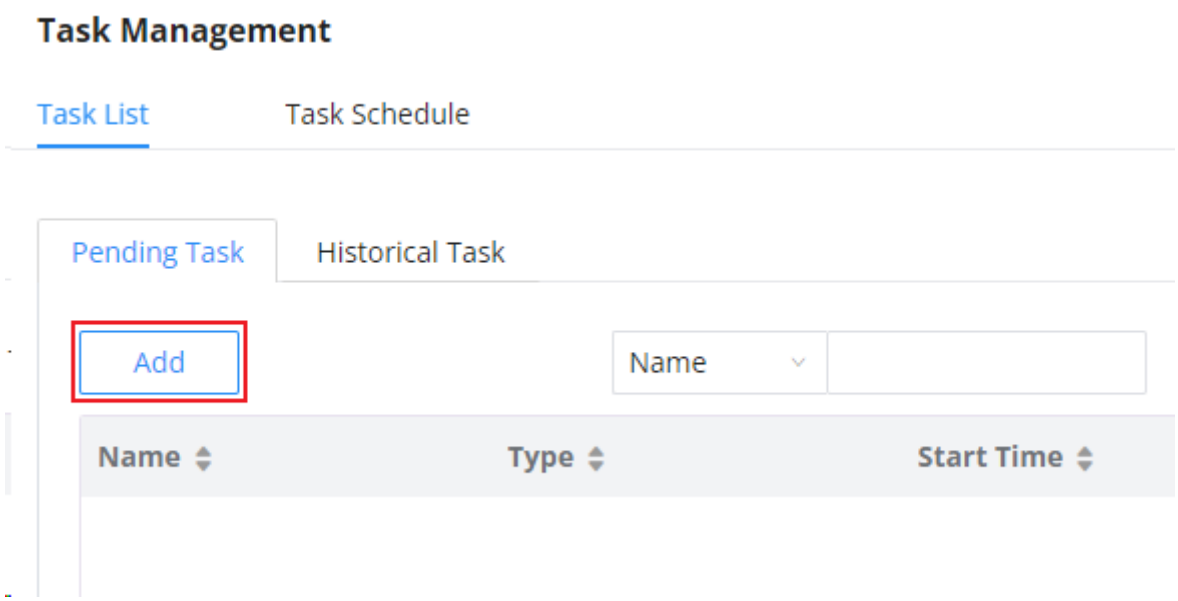
The task which can be scheduled are the following:

- Paging/Intercom
- Backup
- Data Sync
- Cleaner (CDR, Reports, IM Data, Files)

Task List

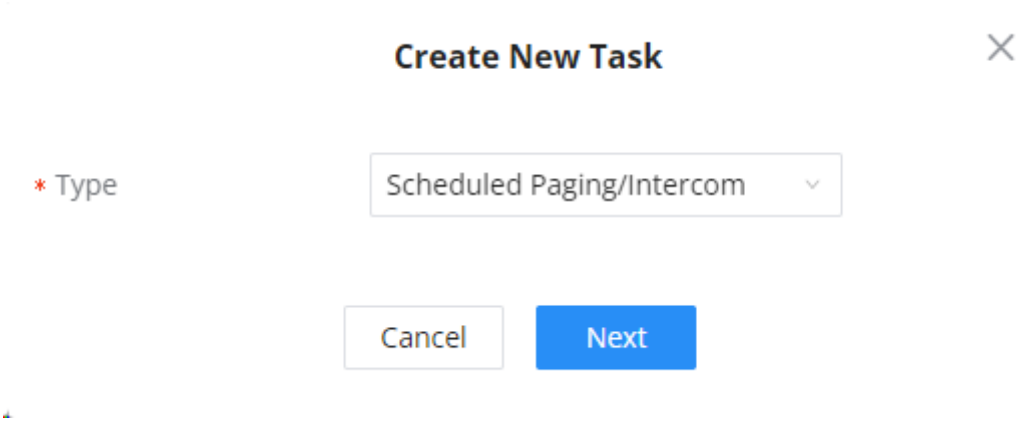
The user can schedule a task in this section to be performed once. The user can schedule 4 possible tasks, Scheduled Paging/Intercom, Scheduled Backup, Scheduled Data Sync, and Scheduled Cleaner.

To schedule a task, please navigate to **Maintenance > Task Management > Task List > Pending Task**, then click on button.



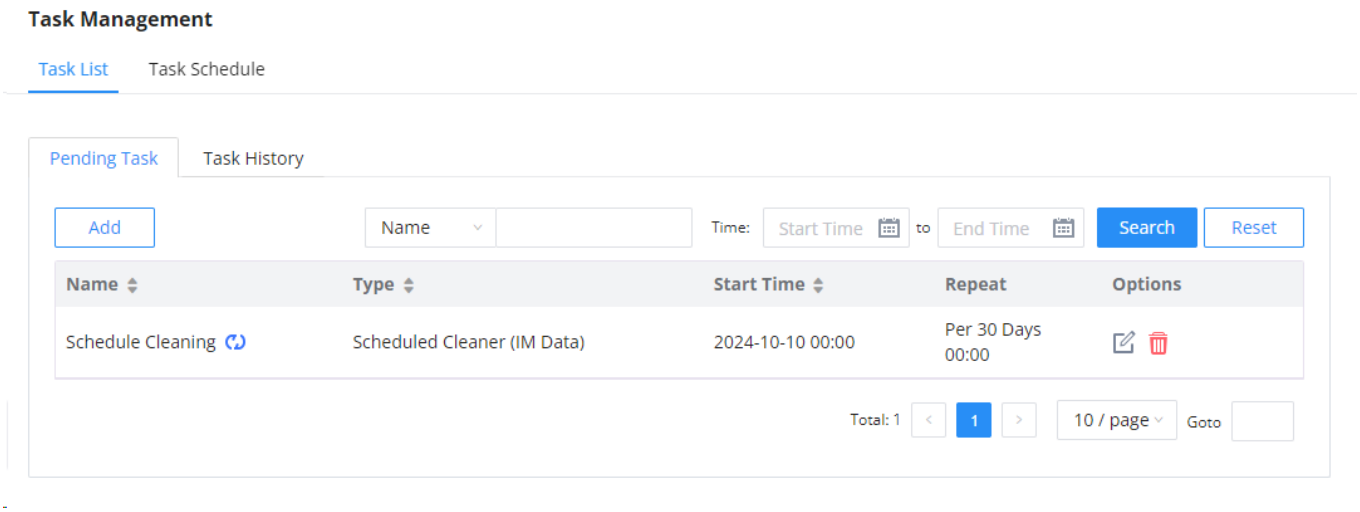
Add Task

Then select the type of the task from the drop-down menu.



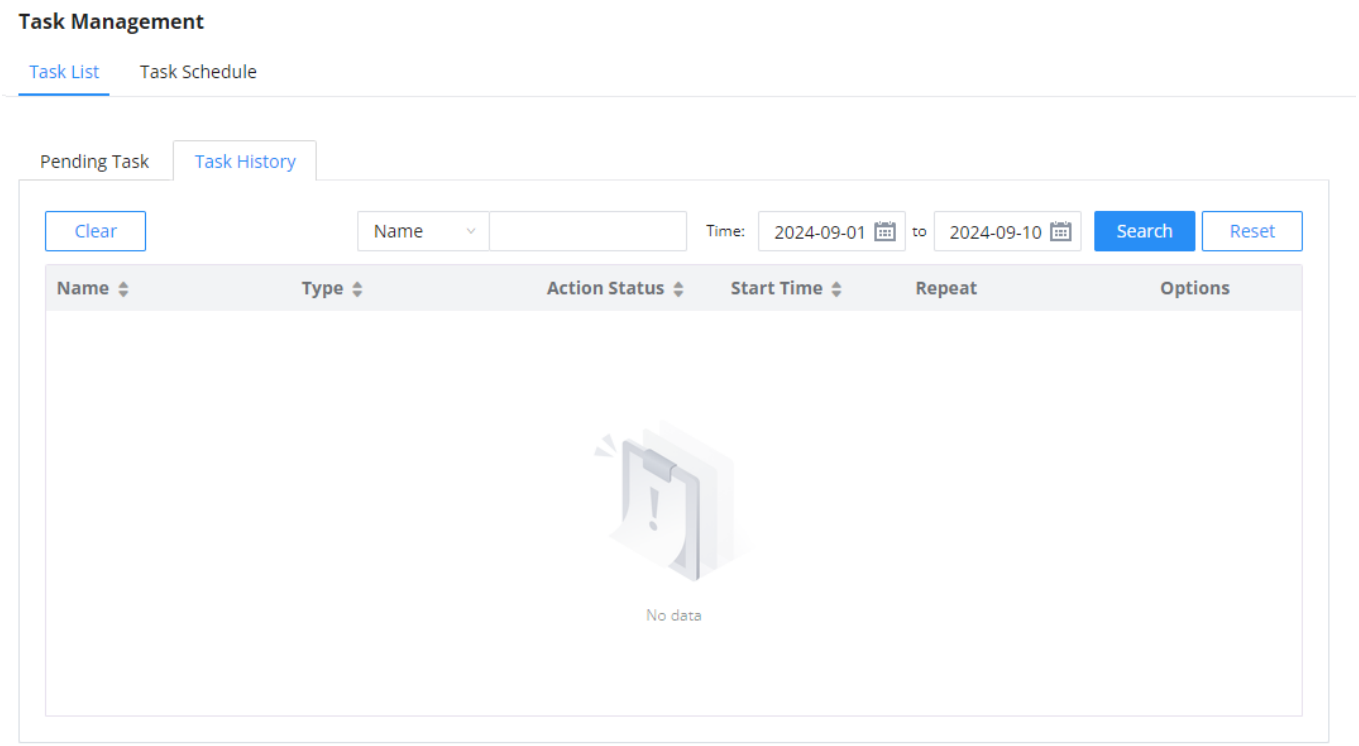
Create New Task

Click "Next" and the set the needed parameters accordingly.



Pending Task

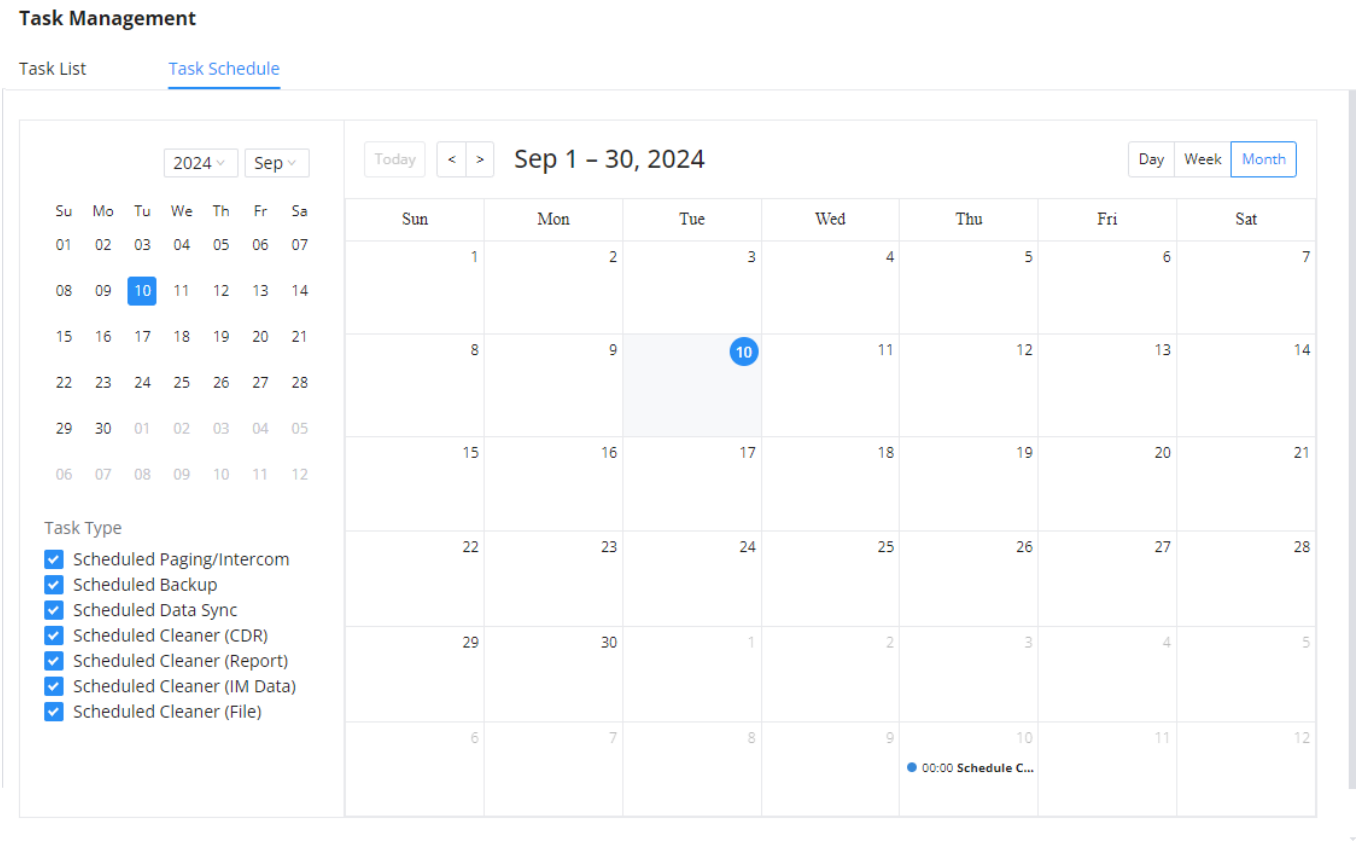
The user can check the log of the tasks which have been performed in "Task History".



Task History

Task Schedule

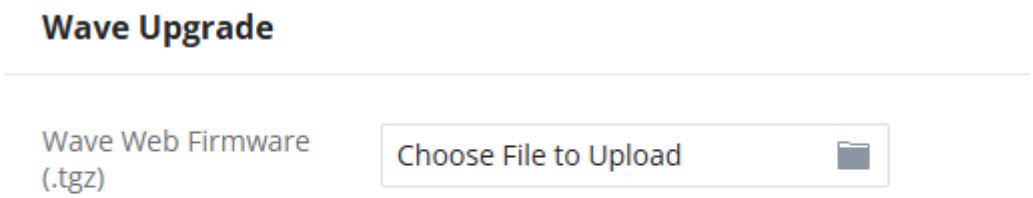
To get an overview about all the tasks which have been scheduled, the user can click on **Task Schedule** tab to view the full schedule.



Task Schedule

Wave Upgrade

To upgrade Wave Web version, please use the following option. You can access this page by accessing the PBX web UI and navigating to **Maintenance > Wave Upgrade.**



Wave Upgrade

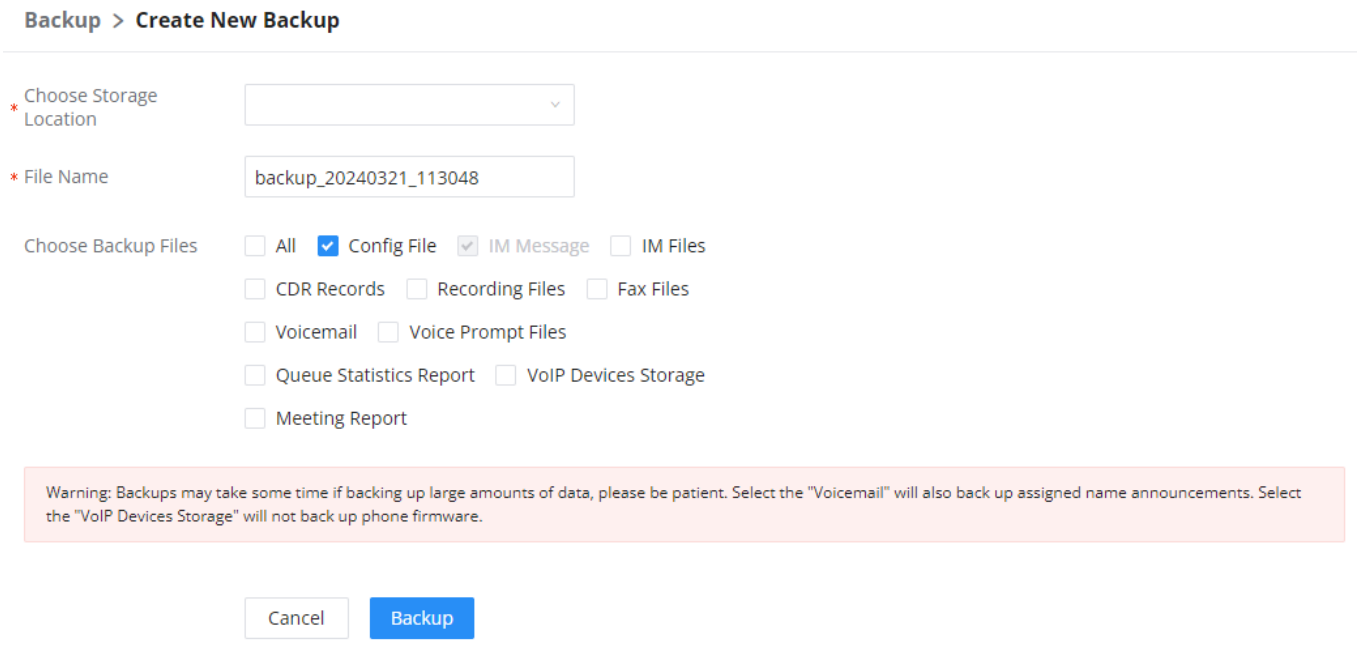
Backup

Backup/Restore

Users could backup the IPPBX configurations for restore purpose under Web GUI→**Maintenance→Backup →Backup/Restore**.

- **Regular Backup**

Click on "Backup" to create a new backup file. Then the following dialog will show.

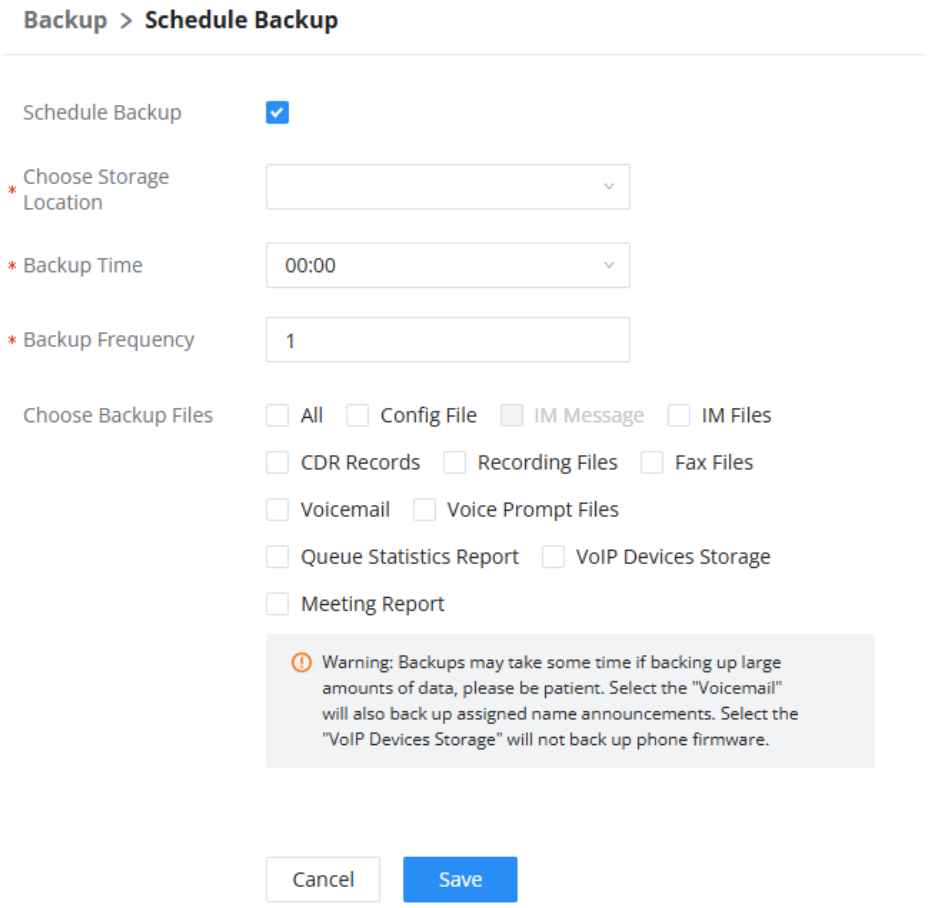


Create New Backup

1. Choose the location of where the backup will be stored. The user can store the backup in GDMS Cloud Storage.
2. Choose the type(s) of files to be included in the backup.
3. Name the backup file.
4. Click on "Backup" to start backup.

- **Scheduled Backup**

The user can configure to have the backup performed following a specific schedule.



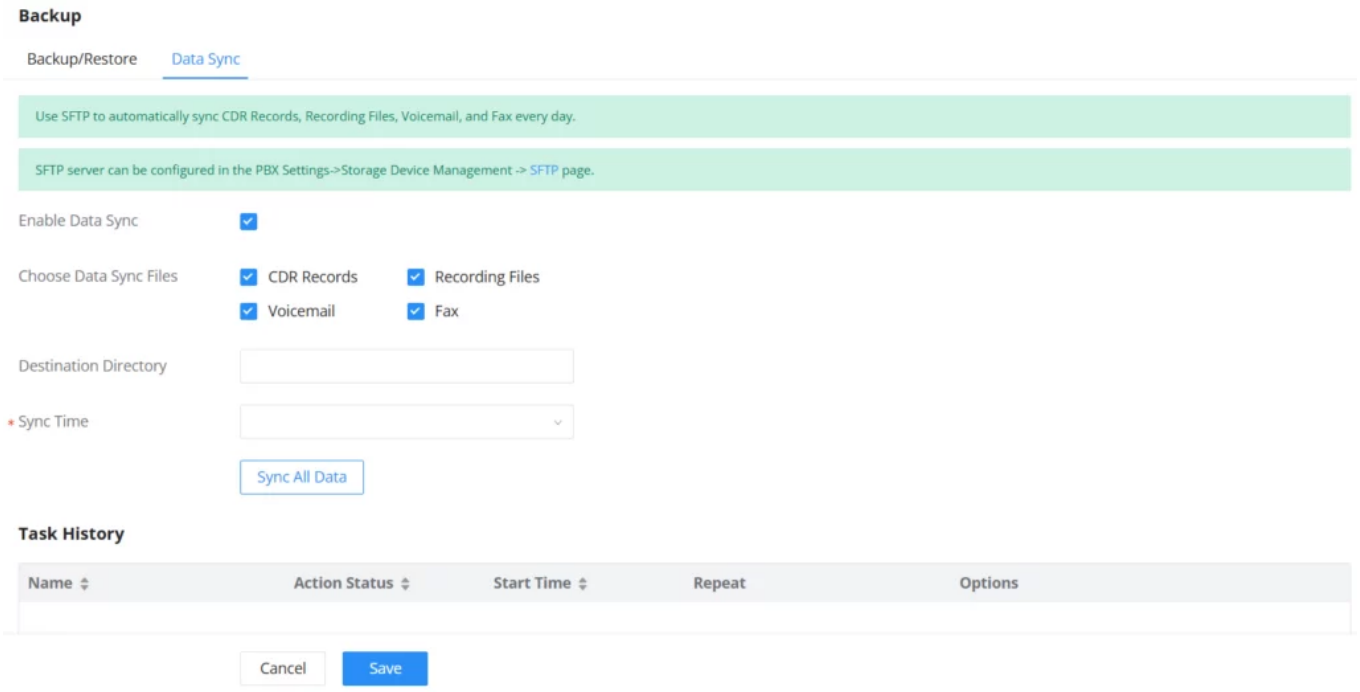
Schedule Backup

1. Choose the location of where the backup will be stored. The user can store the backup in GDMS Cloud Storage.
2. Choose the time when the backup will be triggered.
3. Choose the backup frequency. The integer entered indicates the interval of the backup in days.
4. Choose the type(s) of files to be included in the backup.

Data Sync

Besides local backup, users could backup the voice records/voice mails/CDR in a daily basis to a remote server via SFTP protocol automatically under Web GUI→**Maintenance→Backup→Data Sync**.

The client account supports special characters such as @ or "." Allowing the use email address as SFTP accounts. It allows users as well to specify the destination directory on SFTP server for backup file. If the directory does not exist on the destination, IPPBX will create the directory automatically.



Data Sync

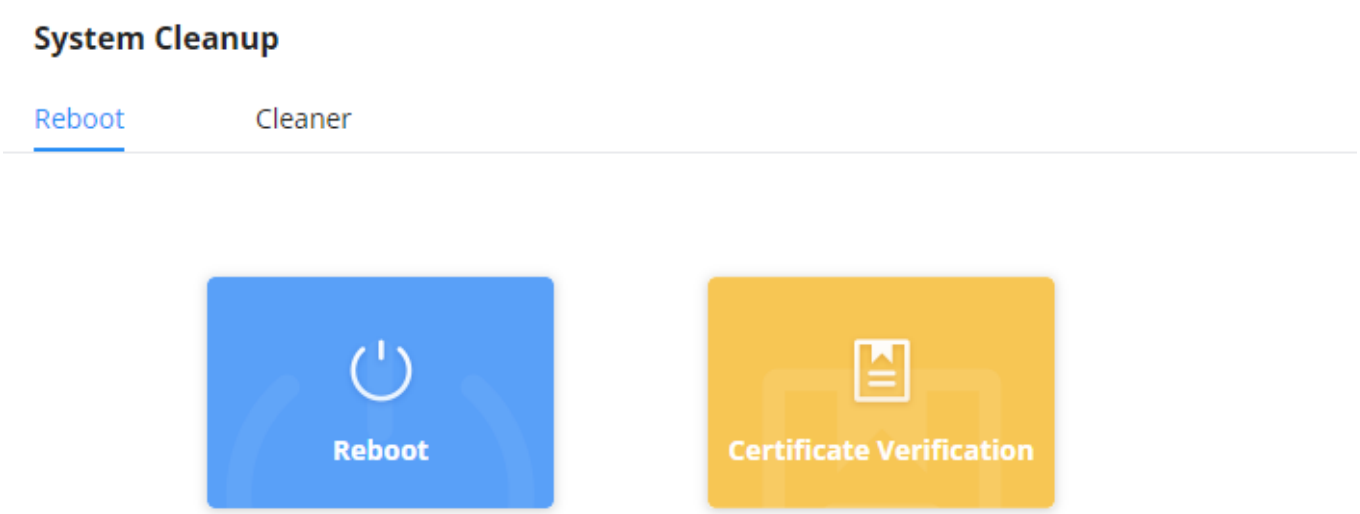
Enable Data Sync	Enable Data Sync by ticking the box.Note: The default setting is disabled.
Choose Data Sync Files	Choose the type of data to sync. CDR Records Recording Files Voicemail Fax
Destination Directory	Enter ther destination filepath with the folder name. Format: 'xxx/yyy/zzz'. If the directory does not exist, the PBX will create it automatically.
Sync Time	Enter 0-23 to specify the backup hour of the day.

The IPPBX will then try connecting the server to make sure the server is up and accessible for the IPPBX. Save the changes and all the backup logs will be listed on the web page. After data sync is configured, users could also manually synchronize all data by clicking on instead of waiting for the backup time interval to come.

System Cleanup

Reboot

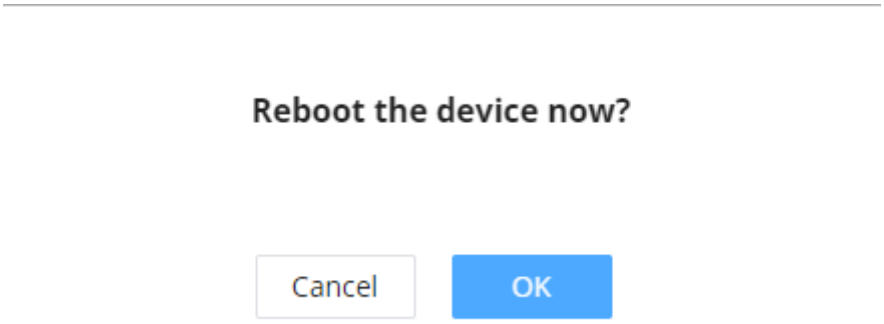
Users could perform reset and reboot under Web GUI**>Maintenance>System Cleanup>Reboot**.



Reboot

Reboot

When the user clicks on reboot, a confirmation prompt will appear. To proceed with rebooting the device, please click "OK".



Reboot Confirmation Prompt

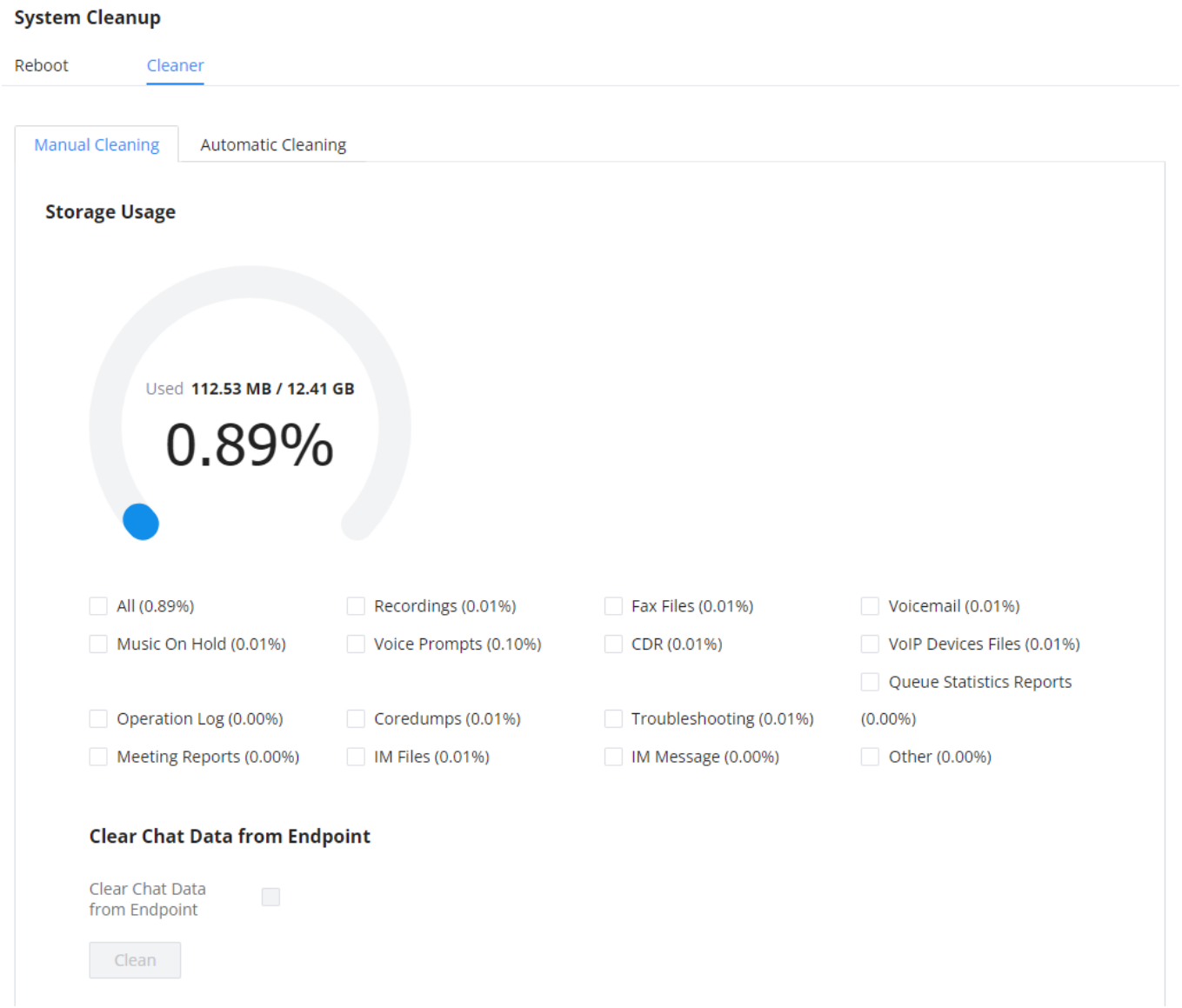
Certificate Verification

This option is used to verify the validity of the TLS certificate used to connect to the PBX module over HTTPS.

Cleaner

Users could configure to clean the Call Detail Report/Voice Records/Voice Mails etc... manually and automatically under Web GUI→**Maintenance→System Cleanup/Reset→Cleaner**.

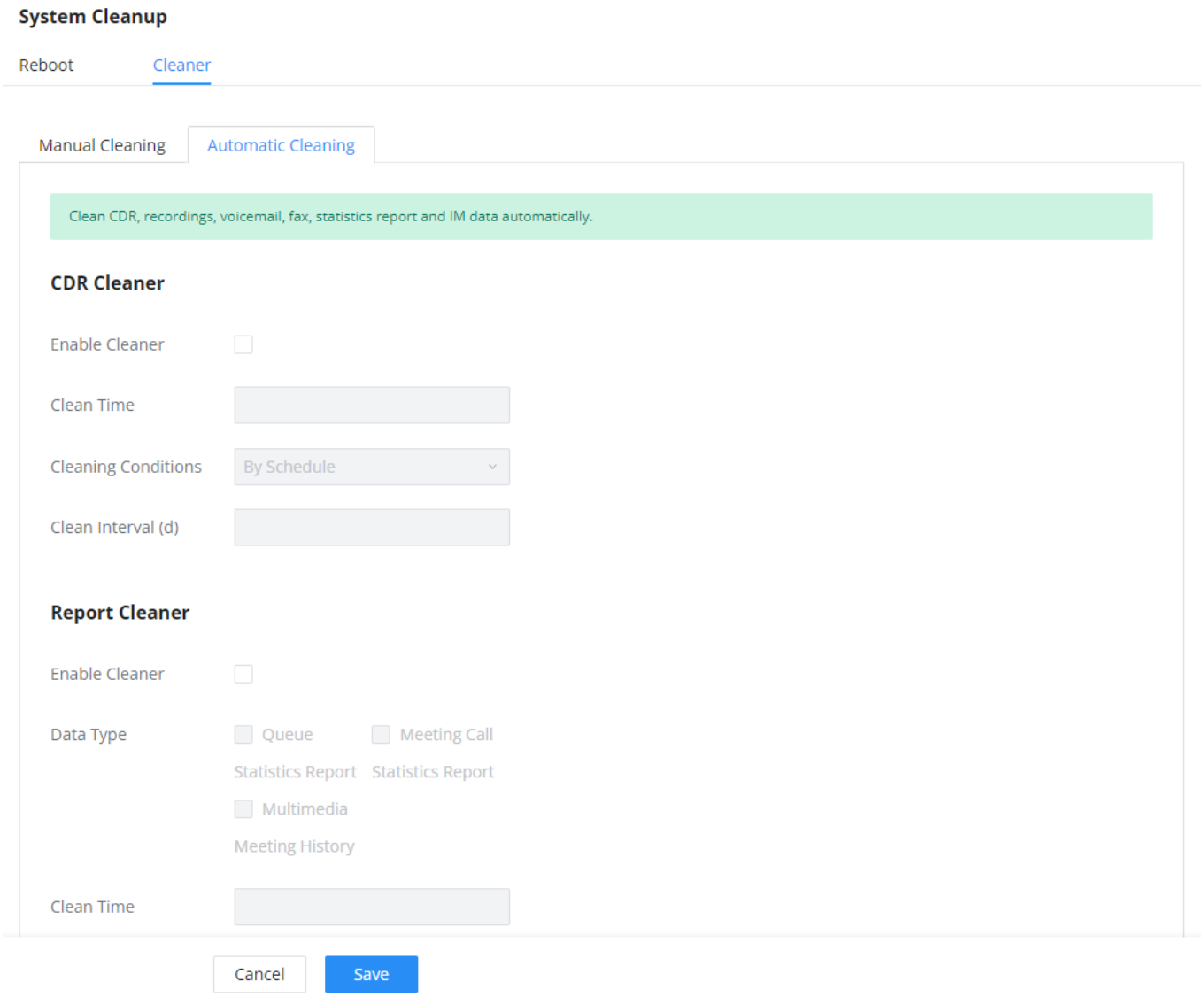
The following screenshot show the settings and parameters to configure the manual cleaner feature on IPPBX.



Manual Cleaning

IPPBX regularly cleans up CDRs, report data, chat data, recording files, historical appointment meeting records, voice mail, backup files, and fax files. The report data includes queue statistics report and conference room call statistics report; chat data includes chat messages and chat shared files; historical appointment conferences include audio and video conference appointment records. Automatic cleanup is not enabled by default and supports regular cleanup of database data based on dimensions such as cleanup time, cleanup conditions, and cleanup interval.

User can also set an automatic cleaning under **Cleaner→Automatic Cleaning**. The following screenshot show the settings and parameters to configure the cleaner feature on IPPBX.



Automatic Cleaning

CDR Cleaner	
Enable Cleaner	Enable the CDR Cleaner function.
Clean Time	Enter 0-23 to specify the hour of the day to clean up CDR.
Cleaning Conditions	By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated in the past 3 days. Keep Last X Records: If the max number of CDR has been reached, CDR will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.) Keep Last X Days: Delete all entries older than X days.
Clean Interval	Enter 1-30 to specify the day of the month to clean up CDR when By Schedule is selected as Cleaning Conditions.
Max Entries	Set the maximum number of CDR entries to keep when Keep Last X Records is selected as Cleaning Conditions.Default is 50000. Valid range: 10000 - 100000.
Keep Last X Days	Enter the number of days of call log entries to keep when Keep Last X days is selected as Cleaning Conditions.Default is 30. Valid range: 1 - 100.
Report Cleaner	
Enable Cleaner	Enable scheduled queue log cleaning.

Data Type	Select a type of data to clean. Queue Statistics Report Meeting Call Statistics Report Multimedia Meeting History
Clean Time	Enter the hour of the day to start the cleaning. The valid range is 0-23.
Cleaning Conditions	By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago. Keep Last X Records: If the max number of Queue Statistics Report entries has been reached, Queue Statistics Report entries will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.) Keep Last X Days: Delete all entries older than X days.
Clean Interval	Enter how often (in days) to clean queue logs when By Schedule is selected as Cleaning Conditions. The valid range is 1-30.
Max Entries	Set the maximum number of Queue Statistics Report entries to keep when Keep Last X Records is selected as Cleaning Conditions.Default is 50000. Valid range: 10000 - 100000.
Keep Last X Days	Enter the number of days of Queue Statistics Report entries to keep when Keep Last X days is selected as Cleaning Conditions.Default is 30. Valid range: 1 - 730.
IM Data Cleaner	
Enable Cleaner	Enable IM data cleaner
Data Type	Select a type of file to clean. IM Files IM Messages
Clean time	Enter the hour of the day to start the cleaning. The valid range is 0-23.
Cleaning Conditions	By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to remove all records that were generated 3 days ago. Keep Last X Records: If the max number of Conference Call Statistics Report has been reached, Conference Call Statistics Report will be deleted starting with the oldest entry at the configured cleaning time. (Note: The amount of records displayed on the page of call queue statistics is not one-to-one with the actual amount of records in the database.) Keep Last X Days: Delete all entries older than X days.
Clean Interval	Enter how often (in days) to clean queue logs when By Schedule is selected as Cleaning Conditions. The valid range is 1-30.
Max Entries	Set the maximum number of CDR Conference Call Statistics Report entries to keep when Keep Last X Records is selected as Cleaning Conditions.Default is 50000. Valid range: 10000 - 100000.
Keep Last X Days	Enter the number of days of Conference Call Statistics Report entries to keep when Keep Last X days is selected as Cleaning Conditions.Default is 30. Valid range: 1 - 100.
File Cleaner	
Enable Cleaner	Enabling files cleaning.
Clean Files in External Device	If enabled the files in external device (USB/SD card) will be automatically cleaned up as configured.
Choose Cleaner File	Select the files for system automatic clean. Basic Call Recording Files Meeting Recording Files Meeting Video Recordings Paging/Intercom Recordings Paging/Intercom Video Recordings Call Queue Recording Files Voicemail Files Emergency Calls Recording Files Fax SCA Recording Files
Clean Time	Enter the hour of the day to start the cleaning. The valid range is 0-23.
Cleaning Conditions	By Schedule: If the clean interval is 3, cleaning will be performed every 3 days to delete all files. By Threshold: Check at the configured cleaning time every day to see if the storage threshold has been exceeded and perform cleaning of all files if it has. Keep Last X Days: Delete all files older than X days.
File Clean Interval	Enter 1-30 to specify the day of the month to clean up the files.

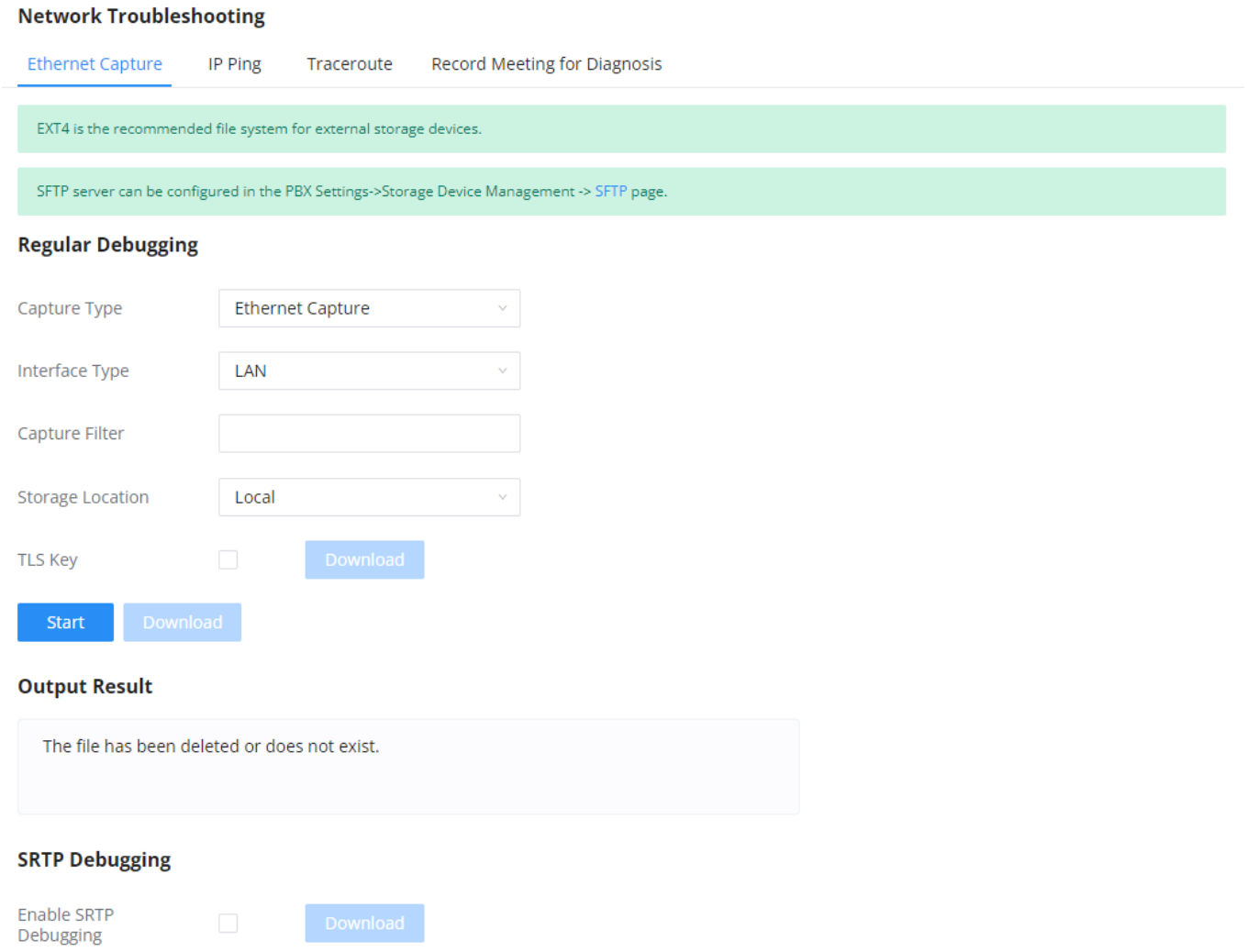
File Clean Threshold	Enter the internal storage disk usage threshold (in percent). Once this threshold is exceeded, the file cleanup will proceed as scheduled. Valid range is 0-99.
Keep Last X Days	Automatically delete all recordings older than this x days when the threshold is reached. If not set, all data is cleared. Valid range: 1 - 100.
Cleaner Log	
Cleaner Log	Clicking on the Clean button will clear the cleaner log.

All the cleaner logs will be listed on the bottom of the page.

Network Troubleshooting

Ethernet Capture

Ethernet Capture allows capturing the traffic of the IPPBX for troubleshooting purposes. To access Ethernet Capture feature, please navigate to **Maintenance -> Network Troubleshooting -> Ethernet Capture**



Ethernet Capture

The capture packets can be stored locally and downloaded for analysis. However, if the user is diagnosing a randomly-occurring issue, he/she can run a continuous packet capture which can be limited by the size of the packet capture and the number of packet capture instances.

When the maximum packet capture file size is reached, a new packet capture file will be created. When the maximum number of capture files number is reached, then the IPPBX will delete the oldest file created file and replace it with the new one.

Parameter	Description
Capture Type	Ethernet Capture: Gets a packet capture of all network traffic going through the device.WebSocket Capture: Gets a packet capture of WebSocket protocol. Mainly used for troubleshooting Wave Web calling and conferencing issues.
Interface Type	Select the network interface to monitor.
Capture Filter	Enter the filter to obtain the specific types of traffic, such as (host, src, dst, net, proto...).
Storage Location	Local: Store the captured packets in the local storage. SFTP Server: Save the capture trace to a SFTP server. Please make sure that SFTP is correctly configured under PBX Settings -> Online Storage -> SFTP Server External Storage: Save the capture trace in a usb flash drive or an SD card. This requires that a USB flash drive or SD card to be plugged into the PBX. File formats supported are FAT32 and ExFat.
Save to External Storage	When or more external storage units are connected to the PBX, the user will be able to pick which one to use.? Note: This option is available only when you choose "External Storage" as the storage destination of the capture trace.
Destination Directory	When SFTP is selected, this option will appear. Please enter the directory path in which you would like to store the captured packets.
Packet Capture Size	This option appears only when "External Storage" or "SFTP" options are selected. Define the packet capture size, the option available are: 50MB, 100MB, and 200MB.
Number of Packet Capture	Define the maximum number of the packets captured. The available options are 5, 10, and 20 packets.
TLS Key	SSL/TLS packets can be decrypted through Wireshark. The packet capture must contain the TLS handshake process.
Start	Start capturing network traffic.
Stop	Stop capturing network traffic.
Download	Download the captured packets. This option can only be used when the captured packets are stored locally.
Enable SRTP Debugging	Check this box to troubleshoot calls encrypted with TLS/SRTP.

The output result is in .pcap format. Therefore, users could specify the capture filter as used in general network traffic capture tool (host, src, dst, net, protocol, port, port range) before starting to capture the trace.

Capture files saved on external devices will now have "capture" prepended to file names.

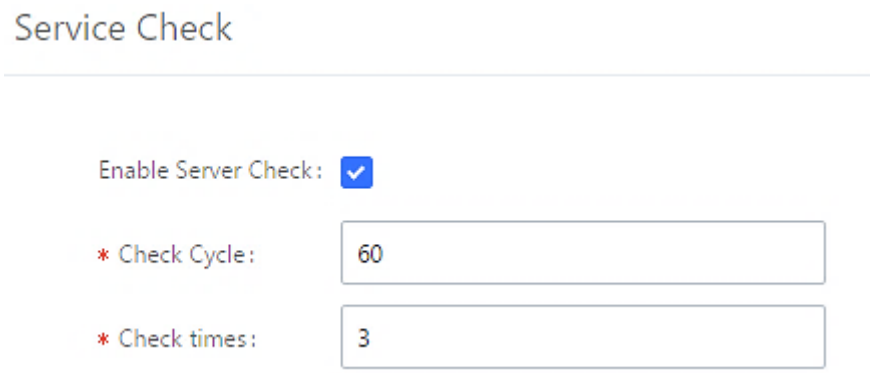
Record Meeting For Diagnosis

Recording the meeting on the IPPBX allows the user to take a recording sample of a meeting to diagnose the issues that might be encountered. This allows the user to pinpoint the connections which are encountering issues when using the meeting feature.

Service Check

Enable Service Check to periodically check IPPBX service status. Check Cycle is configurable in seconds and the default setting is 60 sec. Check Times is the maximum number of failed checks before restart the

IPPBX. The default setting is 3. If there is no response from IPPBX after 3 attempts (default) to check, current status will be stored and the internal service in IPPBX will be restarted.



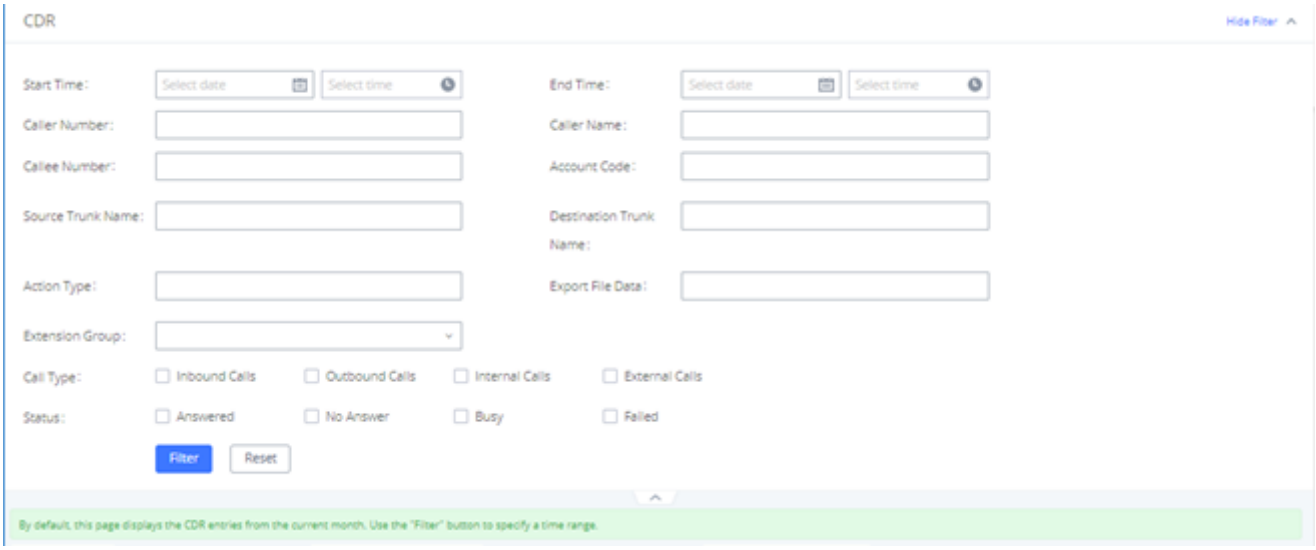
Service Check

CDR

CDR

CDR (Call Detail Record) is a data record generated by the PBX that contains attributes specific to a single instance of phone call handled by the PBX. It has several data fields to provide detailed description for the call, such as phone number of the calling party, phone number of the receiving party, start time, call duration, etc.

On the IPPBX, the CDR can be accessed under Web GUI→**CDR**→**CDR**. Users could filter the call report by specifying the date range and criteria, depending on how the users would like to include the logs to the report. Click on "Filter" button to display the generated report.



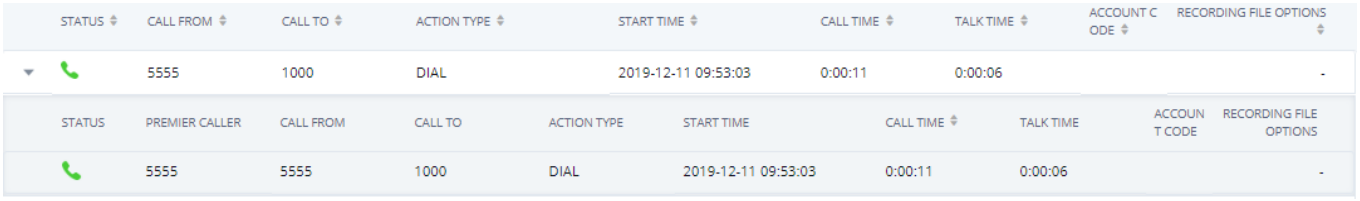
CDR Filter

Call Type	Groups the following: Inbound calls: Inbound calls are calls originated from a non-internal source (like a VoIP trunk) and sent to an internal extension. Outbound calls: Outbound calls are calls sent to a non-internal source (like a VoIP trunk) from an internal extension. Internal calls: Internal calls are calls from one internal extension to another extension, which are not sent over a trunk. External calls: External calls are calls sent from one trunk to another trunk, which are not sent to any internal extension.
Status	Filter with the call status, the available statuses are the following: AnsweredNo AnswerBusyFailed

Source Trunk Name	Select source trunk(s) and the CDR of calls going through inbound the trunk(s) will be filtered out.
Destination Trunk Name	Select destination trunk(s) and the CDR of calls going outbound through the trunk(s) will be filtered out.
Action Type	Filter calls using the Action Type, the following actions are available: Announce Announcement page Dial Announcements Callback Call Forward Conference DISA Follow Me IVR Page Parked Call Queue Ring Group Transfer VM VMG Video Conference VQ_Callback Wakeup Emergency Call Emergency Notify SCA
Extension Group	Specify the Extension Group name to filter with.
Export File Data	Select the fields that will be exported, the following fields are available: Account Code Session Premier caller Action type Source trunk name Destination trunk name Caller number Caller ID Caller name Callee number Answer by Context Start time Answer time End time Call time Talk time Source channel Dest channel Call status Dest channel extension Last app Last data AMAFLAGS UIQUEID Call type NAT
Account Code	Select the account Code to filter with. If pin group CDR is enabled, the call with pin group information will be displayed as part of the CDR under Account Code Field.
Start Time	Specify the start time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
End Time	Specify the end time to filter the CDR report. Click on the calendar icon on the right and the calendar will show for users to select the exact date and time.
Caller Number	Enter the caller number to filter the CDR report. CDR with the matching caller number will be filtered out. User could specify a particular caller number or enter a pattern. '.' matches zero or more characters, only appears in the end. 'X' matches any digit from 0 to 9, case-insensitive, repeatable, only appears in the end. For example: 3XXX: It will filter out CDR that having caller number with leading digit 3 and of 4 digits' length. 3.: It will filter out CDR that having caller number with leading digit 3 and of any length.
Caller Name	Enter the caller name to filter the CDR report. CDR with the matching caller name will be filtered out.
Callee Number	Enter the callee number to filter the CDR report. CDR with the matching callee number will be filtered out. Note: The "Callee Number" filter field supports specifying Pattern (example: 3XXX) or using Leading digits (example: 3.) as filtering options.

CDR Filter Criteria

The call report will display as the following figure shows.



STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-

STATUS	PREMIER CALLER	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-

Call Report

The CDR report has the following data fields:

- **Start Time**

Format: 2019-12-11 09:53:03

- **Action Type**

Example:

IVR

DIAL

WAKEUP

- **Call From**

Example format: 5555

- **Call To**

Example format: 1000

- **Call Time**

Format: 0:00:11

- **Talk Time**

Format: 0:00:06

- **Account Code**

Example format:

Grandstream/Test

- **Status**

Answered, Busy, No answer or Failed.

Users could perform the following operations on the call report.

- **Sort by "Start Time"**

Click on the header of the column to sort the report by "Start Time". Clicking on "Start Time" again will reverse the order.

- **Download Searched Results**

Click on "Download Search Result(s)" to export the records filtered out to a .csv file.

- **Download All Records**

Click on "Download All Records" to export all the records to a .csv file.

- **Delete All**

Click on button to remove all the call report information.

- **Delete Search Result**

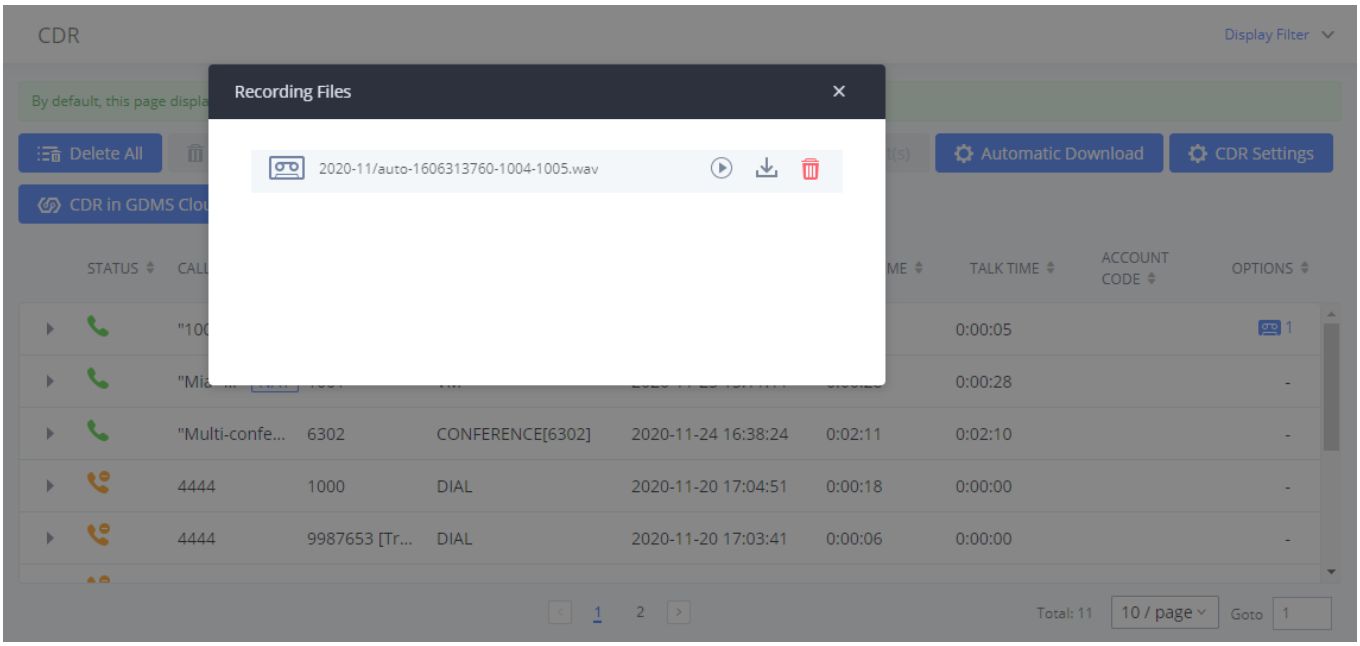
On the bottom of the page, click on button to remove CDR records that appear on search results.

When deleting CDR, a prompt will now appear asking whether to delete all recording files or not.

- **Play/Download/Delete Recording File (per entry)**

If the entry has audio recording file for the call, the three icons on the rightest column will be activated for users to select. In the following picture, the second entry has audio recording file for the call.

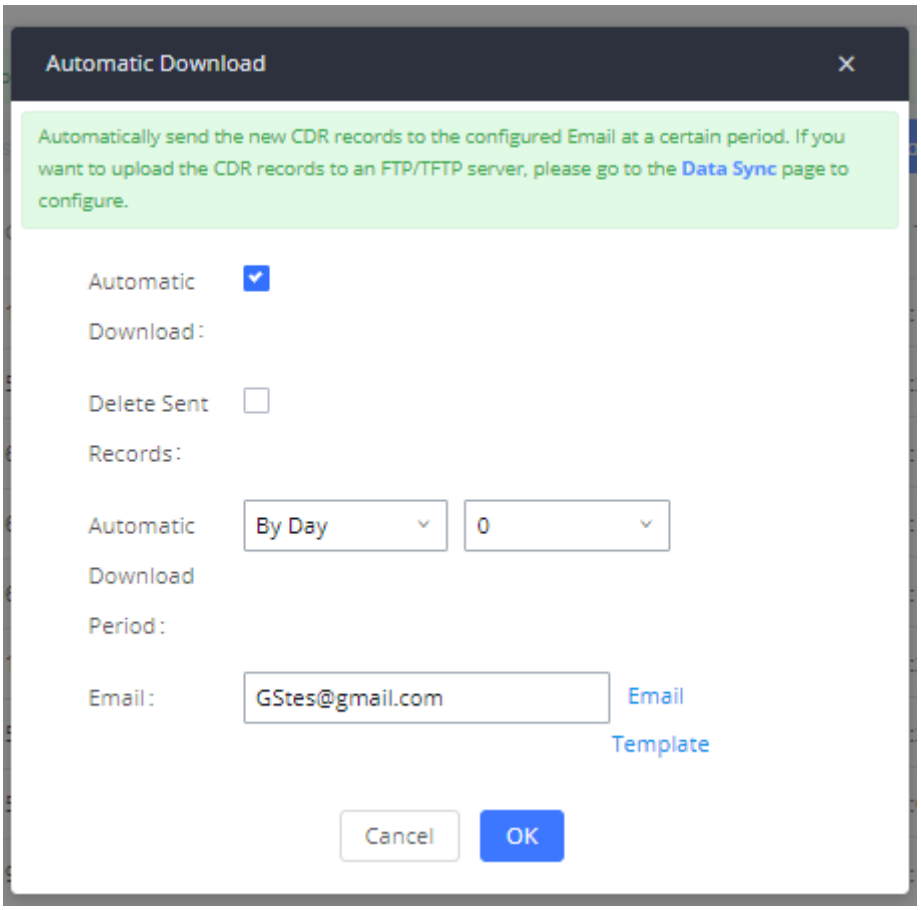
Click on to play the recording file; click on to download the recording file in .wav format; click on to delete the recording file (the call record entry will not be deleted).



Call Report Entry with Audio Recording File

- **Automatic Download CDR Records**

User could configure the IPPBX to automatically download the CDR records and send the records to multiple Email recipients in a specific hour. Click on "Automatic Download Settings" and configure the parameters in the dialog below.



Automatic Download Settings

To receive CDR record automatically from Email, check "Enable" and select a time period "By Day" "By Week" or "By Month", select Hour of the day as well for the automatic download period. Make sure you have entered an Email or multiple email addresses where to receive the CDR records.

users have the option to delete the sent records "Delete Sent Records"

The user can click on the option icon for a specific call log entry to view details about this entry, such as premier caller and transferred call information.

STATUS	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	5555	1000	DIAL	2019-12-11 09:53:03	0:00:11	0:00:06		-

CDR Report

	"ablili lolo" 1000...	9985632	DIAL	2019-12-10 03:23:14	0:00:13	0:00:07	1

STATUS	PREMIER CALLER	CALL FROM	CALL TO	ACTION TYPE	START TIME	CALL TIME	TALK TIME	ACCOUNT CODE	RECORDING FILE OPTIONS
	1000	"ablili lolo" 1000...	9985632	DIAL	2019-12-10 03:23:14	0:00:00	0:00:00		-
	1000	"ablili lolo" 1000...	6500	QUEUE[6500]	2019-12-10 03:23:14	0:00:00	0:00:00		1
	1000	"ablili lolo" 1000...	5555	QUEUE[6500]	2019-12-10 03:23:14	0:00:13	0:00:07		-

Detailed CDR Information

Downloaded CDR File

The downloaded CDR (.csv file) has different format from the Web GUI CDR. Here are some descriptions.

- **Caller number, Callee number**

"Caller number": the caller ID.

"Callee number": the callee ID.

caller number	callee number	context	calerid	source channel	dest channel	lastapp
	2009	from-internal	"Wake Up Call" <WakeUp>	Local/2009@from-internal-00000001;2	PJSIP/2009-00000013	Dial
2007	31100	from-internal	"" <2007>	PJSIP/2007-00000014	DAHDI/1-1	Dial
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial
1100	2014	from-did-direct	"1100" <1100>	DAHDI/1-1	PJSIP/2014-00000017	Dial

Downloaded CDR File Sample

- **Context**

There are different context values that might show up in the downloaded CDR file. The actual value can vary case by case. Here are some sample values and their descriptions.

from-internal: internal extension makes outbound calls.

ext-did-XXXXX: inbound calls. It starts with "ext-did", and "XXXXX" content varies case by case, which also relate to the order when the trunk is created.

ext-local: internal calls between local extensions.

- **Source Channel, Dest Channel**

Example:

caller number	callee number	context	calerid	source channel	dest channel	lastapp
2009	1100	from-internal	"John Doe" <2009>	PJSIP/2009-00000015	PJSIP/trunk_1-00000016	Dial

Downloaded CDR File Sample Source Channel and Dest Channel 2

"SIP" means it is a SIP call. There are three format:

(a) **PJSIP/NUM-XXXXXX,** where NUM is the local SIP extension number. The last XXXXX is a random string and can be ignored.

(c) **PJSIP/trunk_X/NUM,** where trunk_X is the internal trunk name, and NUM is the number to dial out through the trunk.

(c) **PJSIP/trunk_X-XXXXXX,** where trunk_X is the internal trunk name and it is an inbound call from this trunk. The last XXXXX is a random string and can be ignored.

There are some other values, but these values are the application name which are used by the dialplan.

Local/@from-internal-XXXXX: it is used internally to do some special feature procedure. We can simply ignore it.

Hangup: the call is hung up from the dialplan. This indicates there are some errors or it has run into abnormal cases.

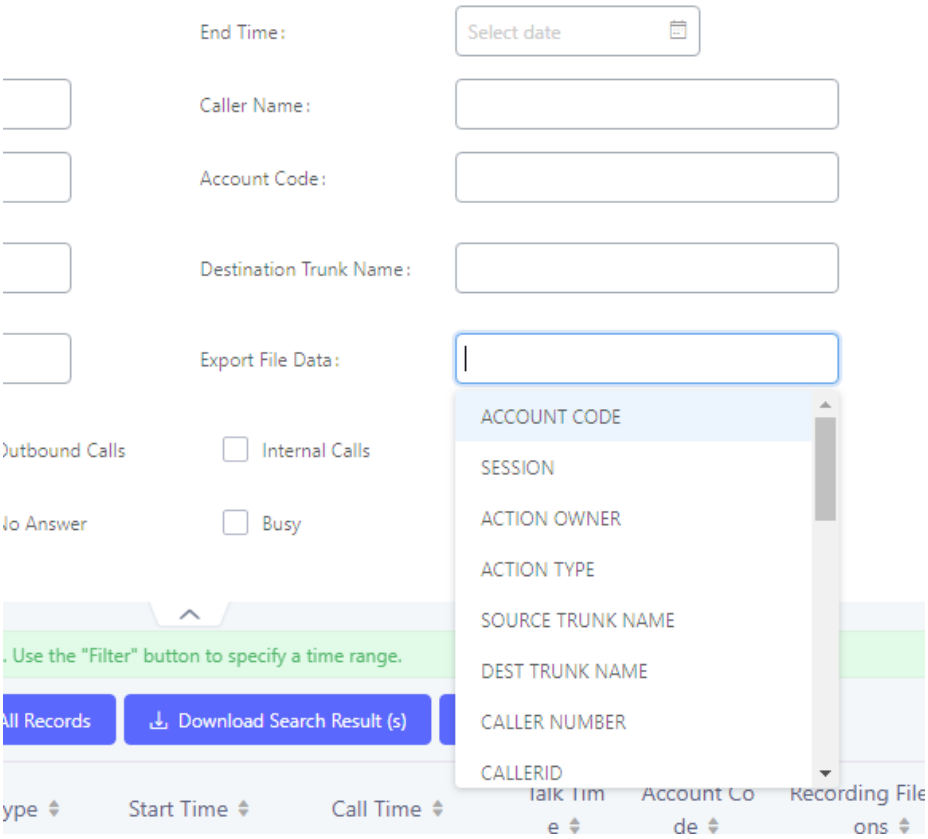
Playback: play some prompts to you, such as 183 response or run into an IVR.

ReadExten: collect numbers from user. It may occur when you input PIN codes or run into DISA

The language of column titles in exported CDR reports and statistics reports will be based on the IPPBX's display language.

CDR Export Customization

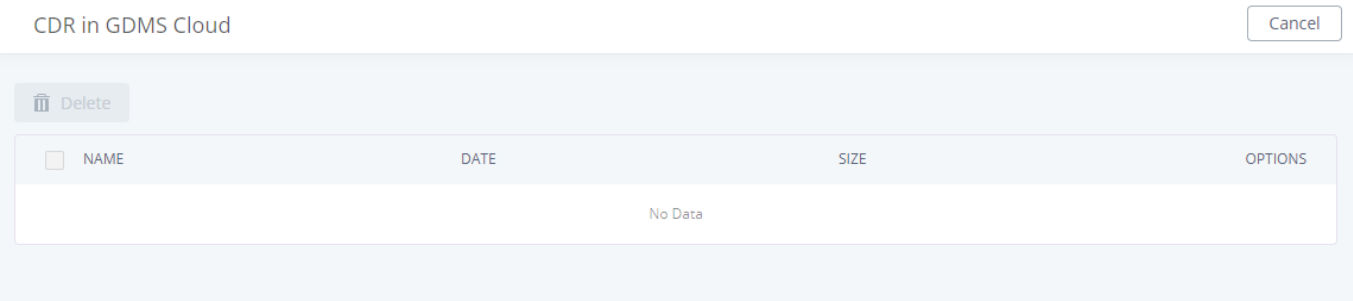
Users can select the data they want to see in exported CDR reports by first clicking on the Filter button on the CDR page under **CDR→CDR** and selecting the desired information in the Export File Data field.



CDR Export File data

CDR in GDMS Cloud

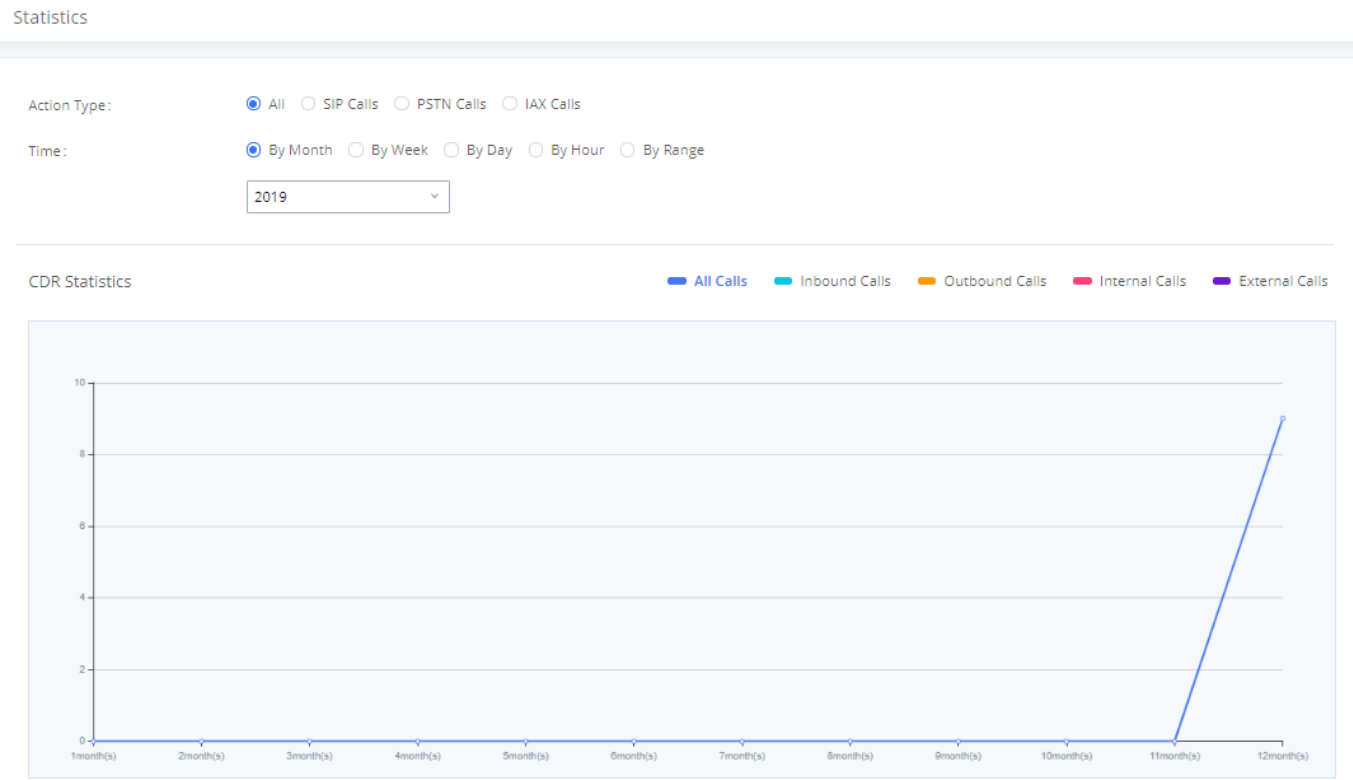
Cloud Storage for CDR Record which can be displayed under **CDR → CDR in GDMS Cloud**.



CDR in GDMS Cloud

Statistics

The IPPBX supports the function of concurrent call statistics. This function provides users with statistics on the number of concurrent calls of all VOIP trunks (SIP trunks). Users can set search criteria to generate custom charts. Select the trunk and time to view the chart of the maximum number of concurrent calls corresponding to the trunk in a certain day or month.



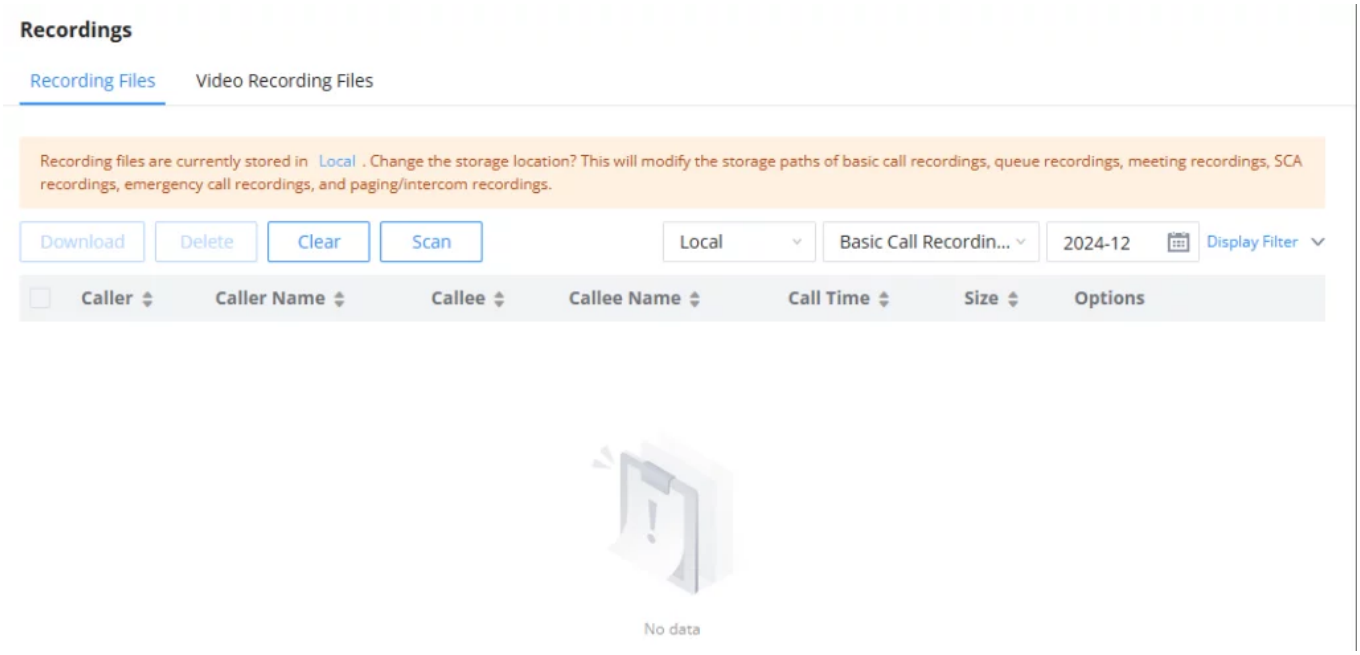
CDR Statistics

Trunk Type	Select one of the following trunk type. All SIP Calls
Call Type	Select one or more in the following checkboxes. Inbound calls Outbound calls Internal calls External calls All calls
Time Range	By month (of the selected year). By week (of the selected year). By day (of the specified month for the year). By hour (of the specified date). By range. For example, 2016-01 To 2016-03.

CDR Statistics Filter Criteria

Recordings

This page lists all the recording files recorded by "Auto Record" per extension/ring group/call queue/trunk, or via feature code "Audio Mix Record". If external storage device is plugged in, for example, SD card or USB drive, the files are stored on the external storage. Otherwise, internal storage of the IPPBX will be used.



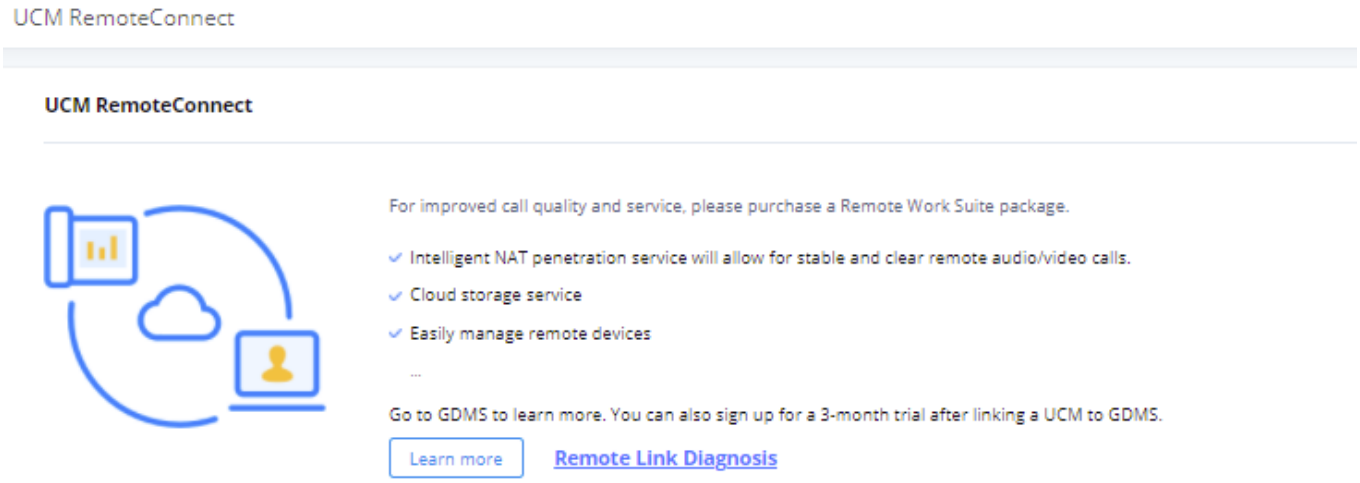
CDR→Recording Files

- Click on "**Download**" to batch-download the selected recording files.
- Click on "**Download All**" to download all the recording files.
- Click on "**Delete**" to batch-delete the selected recording files.
- Click on "**Clear**" to delete all the recording files.
- Click on "**Scan**" to retrieve the file information and display all the recording files on external storage. The IPPBX automatically retrieves the info of the first 5000 files from external storage already. This button can be used when the number of files stored on the external storage exceeds 5000 files and it requires manual file scanning.
- Select either "**USB Disk**" or "**Local**" to show recording files stored on external or internal storage, depending on selected storage space.
- Select whether to show call recordings, queue recordings or conference recordings.
- Click on to download the recording file in .wav format.
- Click on to delete the recording file.
- To sort the recording file, click on the title "Caller", "Callee" or "Call Time" for the corresponding column. Click on the title again can switch the sorting mode between ascending order or descending order.

REMOTECONNECT

An integrated & important part of Grandstream's GDMS cloud-based device management service which runs on Amazon Web Services (AWS) with 99.999% reliability. RemoteConnect cloud service supports hassle-free Work-From-Home audio/video communications & collaborations using WebRTC-based license-free "Grandstream Wave" softphones for desktop/Web/mobile devices (plus GUV series of USB headsets/Webcams), zero-touch out-of-box automated NAT firewall traversal for remote users & devices, IT-friendly remote management of the IPPBX and attached endpoint devices, and more.

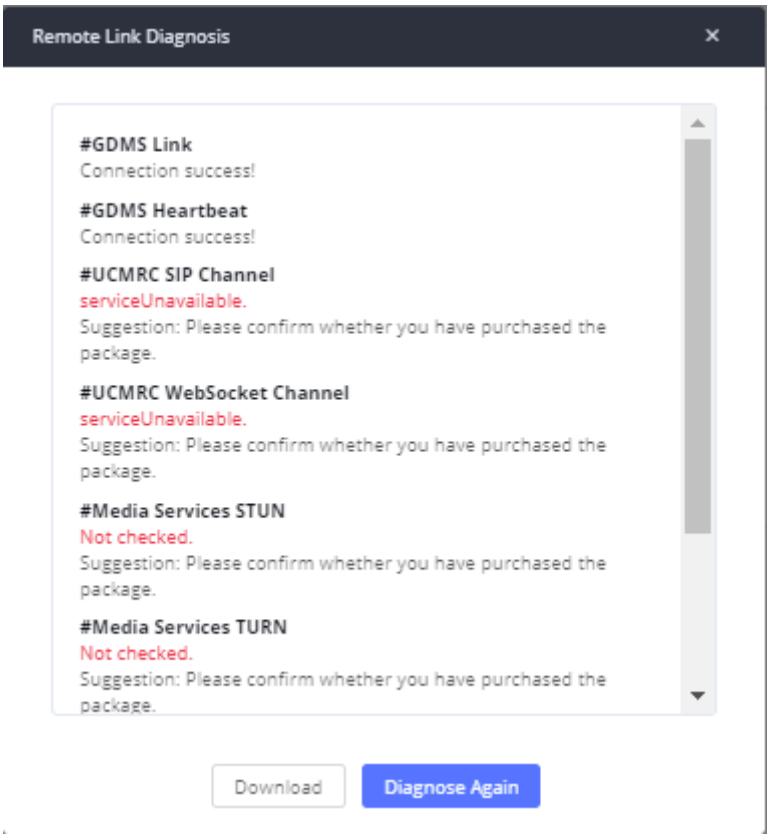
The RemoteConnect can be configured under **RemoteConnect** tab on the web UI of the IPPBX.



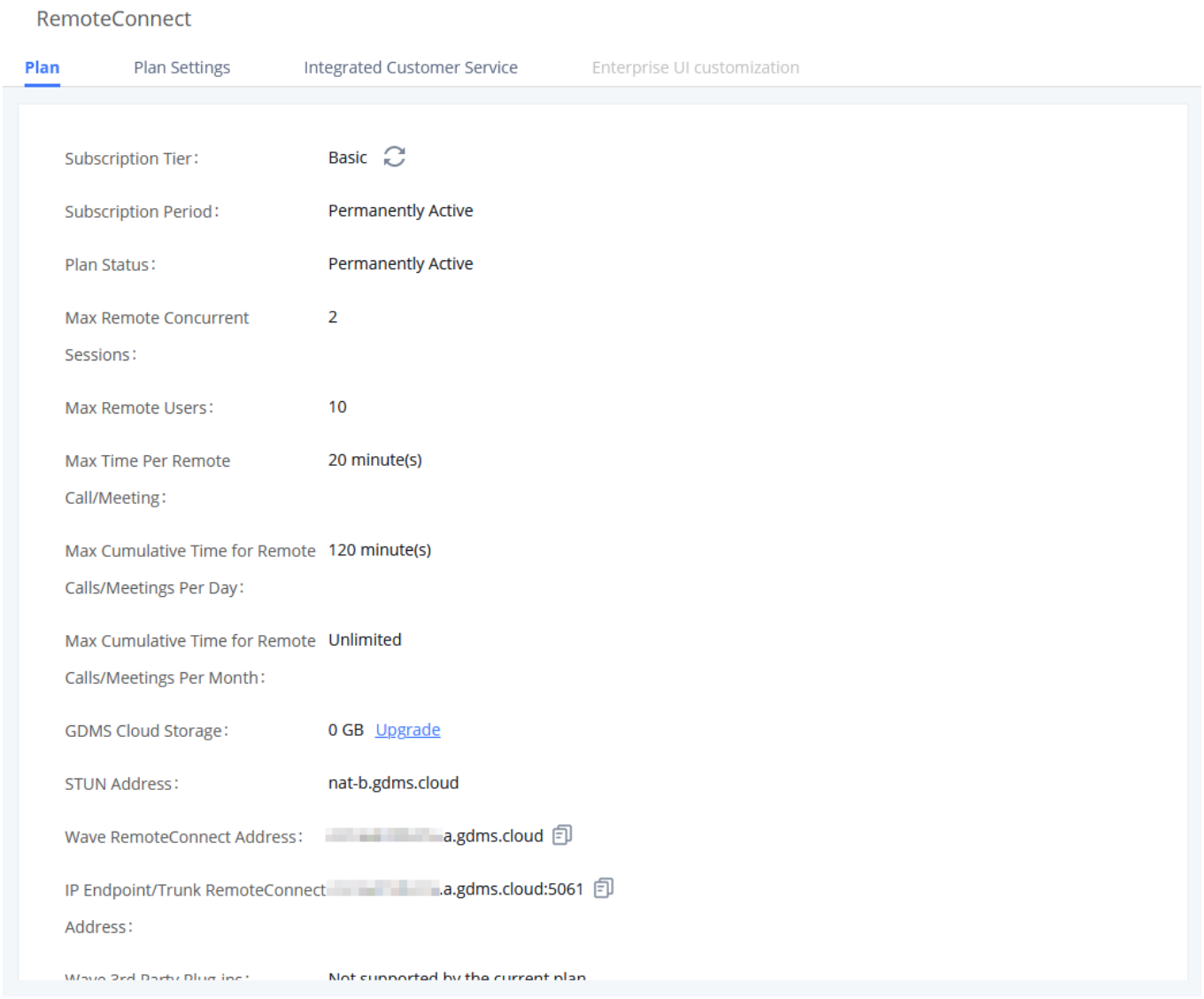
RemoteConnect

On the GDMS platform, sign in and go to **Device→PBX Device** page, click on "Add Device" to add your IPPBX device to the GDMS system, once done an open beta plan will be assigned to the IPPBX.

In daily operation, the user can click the "Diagnosis" button to diagnose the remote service system. The specific diagnosis content includes media service (STUN/TURN), GDMS link and heartbeat detection, tunnel service (SIP/Web Socket), Cloud IM, IPPBX bandwidth speed measurement.



Remote Diagnosis



RemoteConnect Plan in Effect

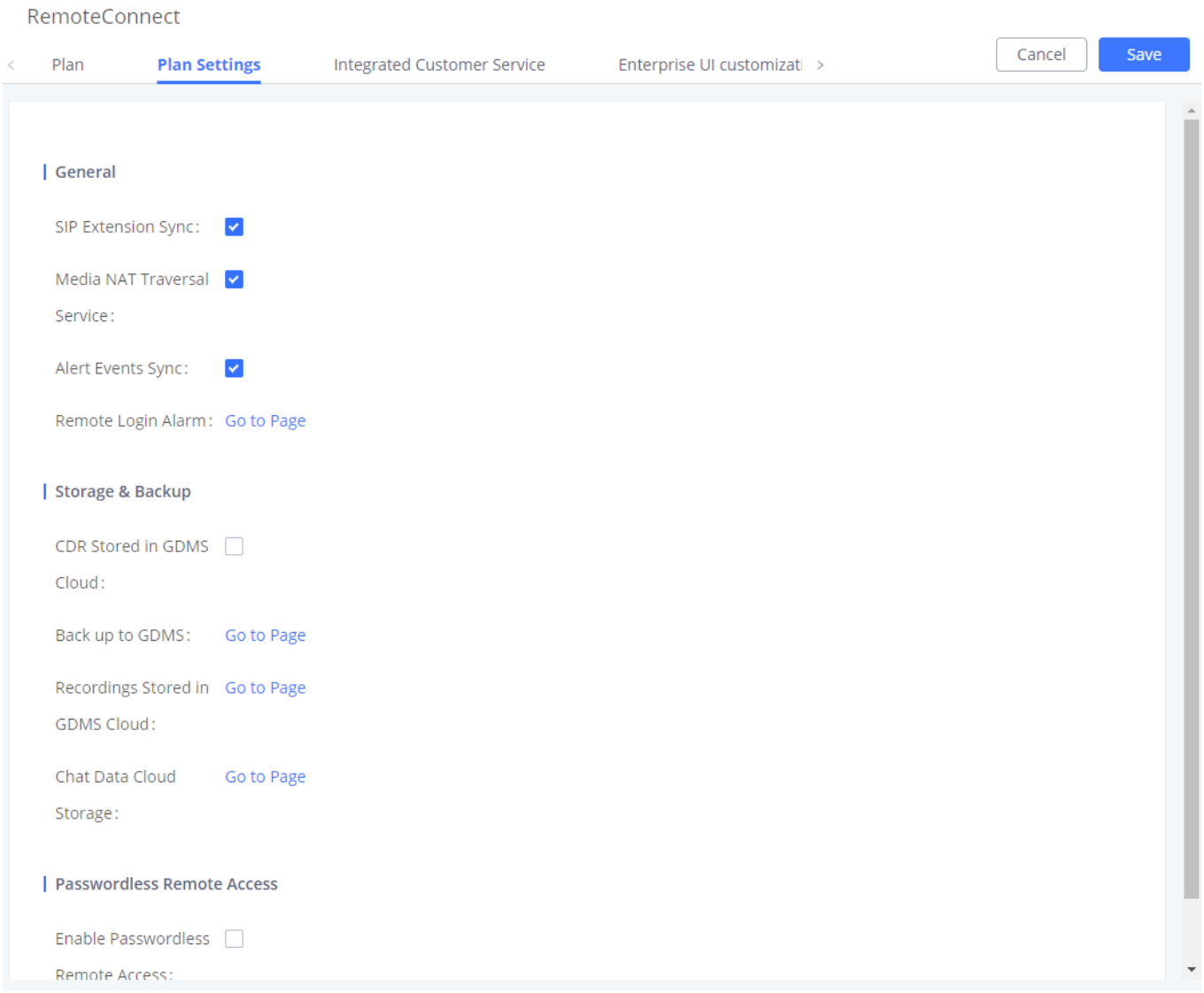
After the IPPBX is added on GDMS, automated NAT traversal, SIP extension sync-up and basic statistics features are available without manual configuration required.

Plan Settings

After IPPBX is added into GDMS, all SIP extensions on the IPPBX will be synced up to GDMS automatically for users to allocate and manage SIP extension for their end devices. Also, the media NAT Traversal service, alert event sync configuration items are checked by default, the CDR data cloud storage in GDMS should be manually checked according to user needs.

The IPPBX supports to allow authorized GDMS user to access IPPBX without entering the password once the super admin or admin checks "Enable Passwordless Remote Access". If super admin enables this option, then the IPPBX will be accessed using the super admin account. If admin enables this option, then the IPPBX will be accessed using the admin account. Super admin and admin can see whether this option is enabled. Additionally, super admin can disable all accounts who enabled this option while admin can only disable access for the account itself.

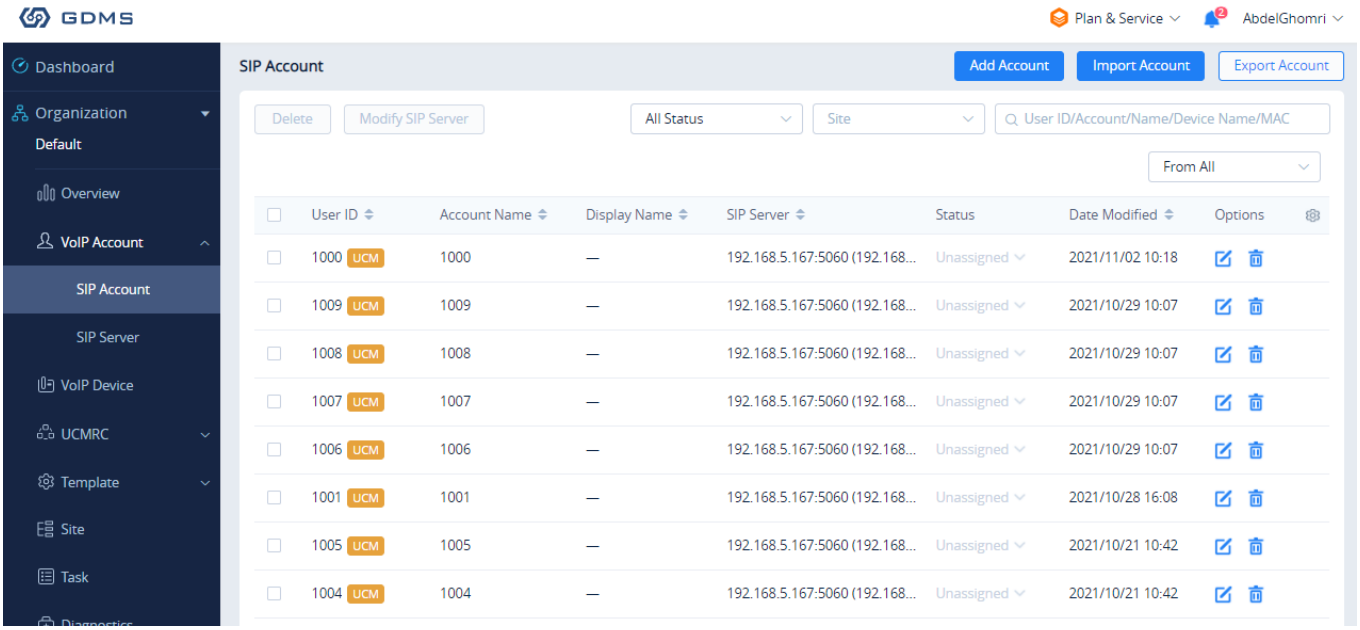
The settings are under IPPBX web GUI: **RemoteConnect→Plan Settings**.



RemoteConnect Plan Settings

After adding IPPBX to the GDMS platform, IPPBX will synchronize all SIP extensions to the GDMS platform, this allows to use the GDMS platform for account allocation and terminal management.

The accounts synchronized to GDMS platform can be viewed on the GDMS-> VoIP Account->SIP Account page. As shown in the figure below:



IPPBX SIP Extensions synchronized to GDMS

The Media NAT Traversal provides a fully automatic intelligent external network penetration service to ensure that you can make normal calls/conferences on the external network.

CDR data cloud storage provides a service of dumping to GDMS to prevent CDR from continuously increasing occupying IPPBX storage space.

Alarm event synchronization is to synchronize the alarm information generated on IPPBX to the GDMS server.

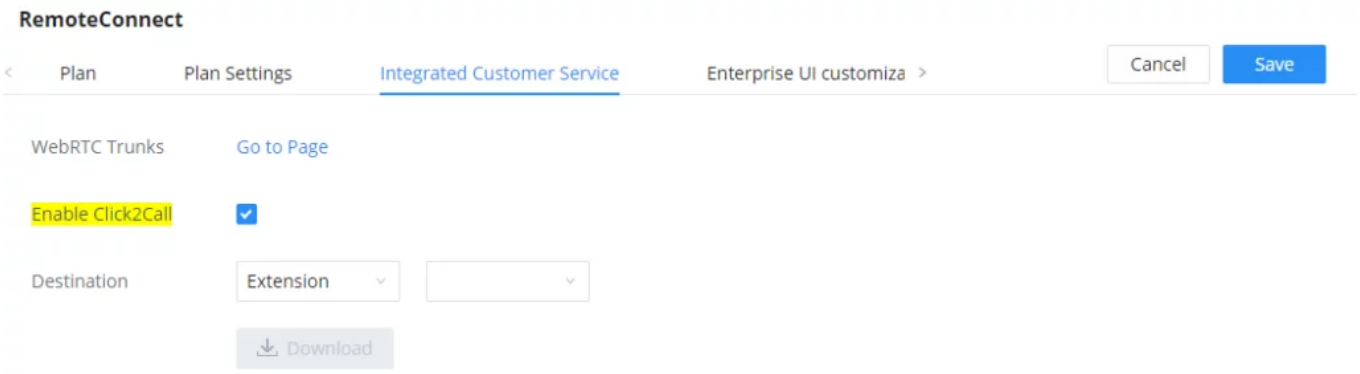
Allow the administrator/super administrator to open it. After clicking on open with an account, the subsequent password-free login will use the account. All administrators and super administrators can see whether this IPPBX is enabled.

Super administrators can check and uncheck all the exemption lists; administrators can check and uncheck the exemption status of this account, and the corresponding account exemption access function will be closed after cancellation.

- Deleting an account on GDMS only removes the association between the account and the device, and does not delete the SIP account information on IPPBX.
- Any creation, deletion or modification of the SIP account on IPPBX will be automatically synchronized to the GDMS cloud platform.
- After checking the "Media NAT traversal service", the TURN service and other related traversal settings set by the user will not take effect.

Integrated Customer Service

To configure the Integrated Customer Service, go to the **Web GUI → RemoteConnect → Integrated Customer Service** page that allows users to download the SDK provided by the customer service system and integrate it on the website, so that the website can contact customer service for call operations. The call queue is used as the customer service number.



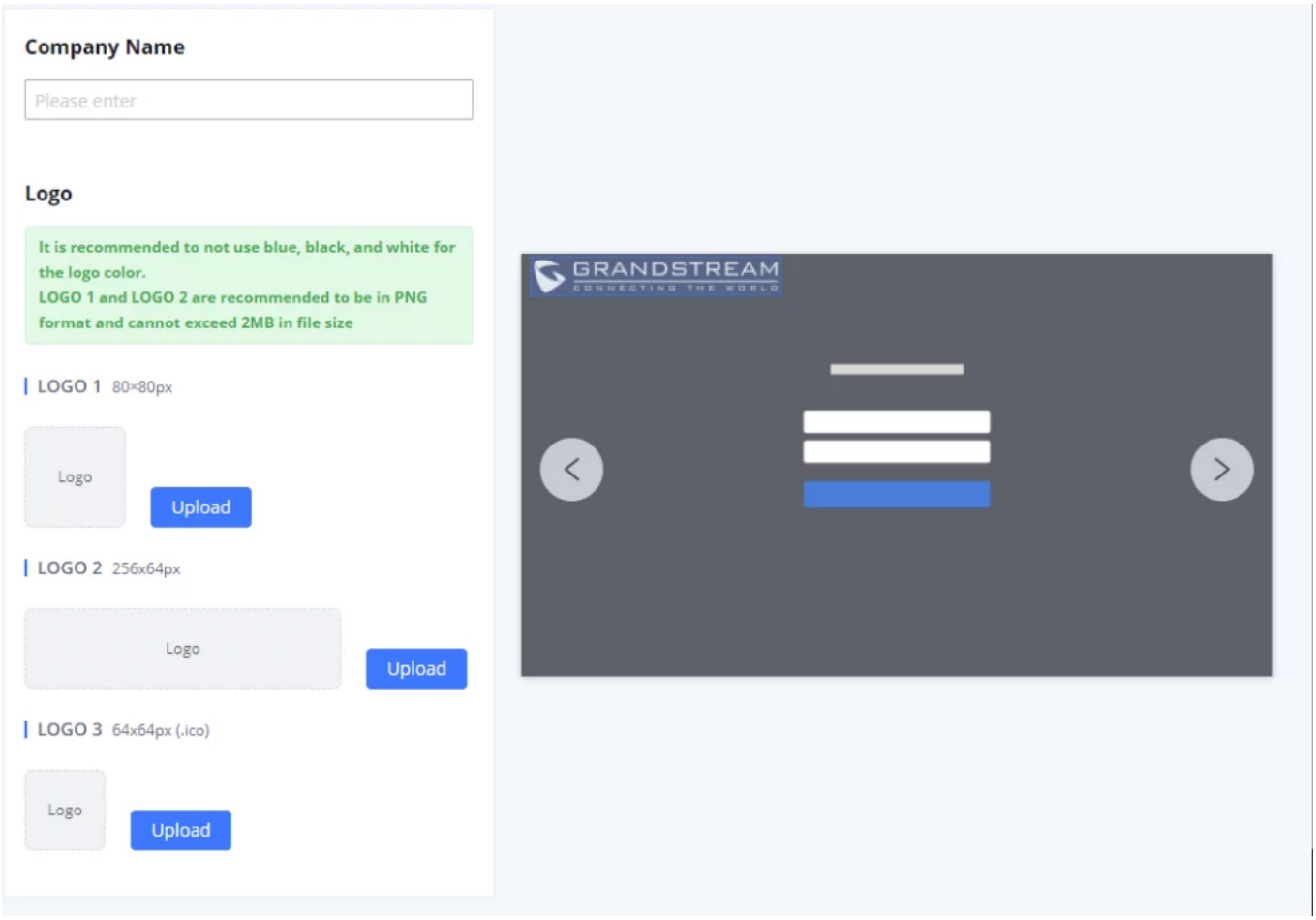
Integrated Customer Service interface

Enabling Click2Call will allow users to initiate a direct call from the web browser by clicking on the call button embedded on the website graphical interface. The calls initiated can be directed to call queues or a specific extension.

Enterprise UI Customization

With a RemoteConnect plan, on the **Web GUI → RemoteConnect → Enterprise UI Customization** page, users can edit the company name and select a local image file as the new logo. The company name acts on the text part with the logo, and the pictures are in different formats and sizes according to the logo position, which are 64*64px (only ico format is supported), 256*256px, 80*80px, which supports users in the "IPPBX management platform/login" "", "Reset Password", "Email Template", "Wave_PC", "Wave Login", "Browser Label", "Guide Page" interface preview.

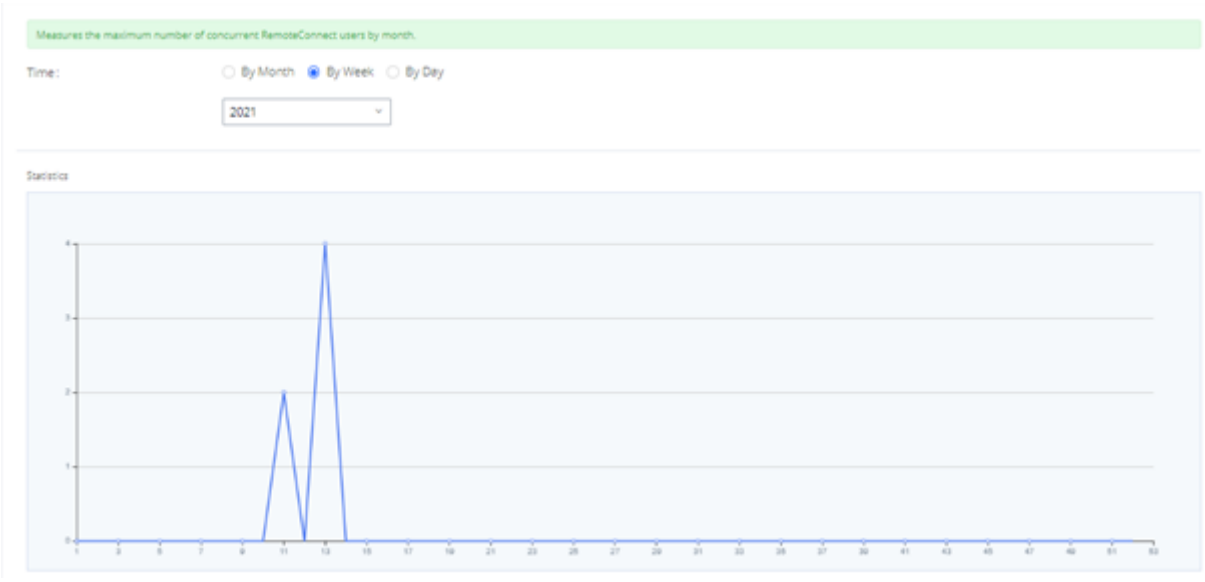
- LOGO 1: Replaces Browser tab icon
- LOGO 2: Replaces the Grandstream banner on the top left corner of the management login page and emails.
- LOGO 3: Replaces the Grandstream logo on the top left corner of the Wave Web interface and IPPBX management interface.



Enterprise UI Customization

Statistics

After using RemoteConnect, all remote calls will be logged and concurrent remote calls will be displayed on the IPPBX. The concurrent remote calls can be viewed under IPPBX web GUI →**RemoteConnect→Statistics** page.



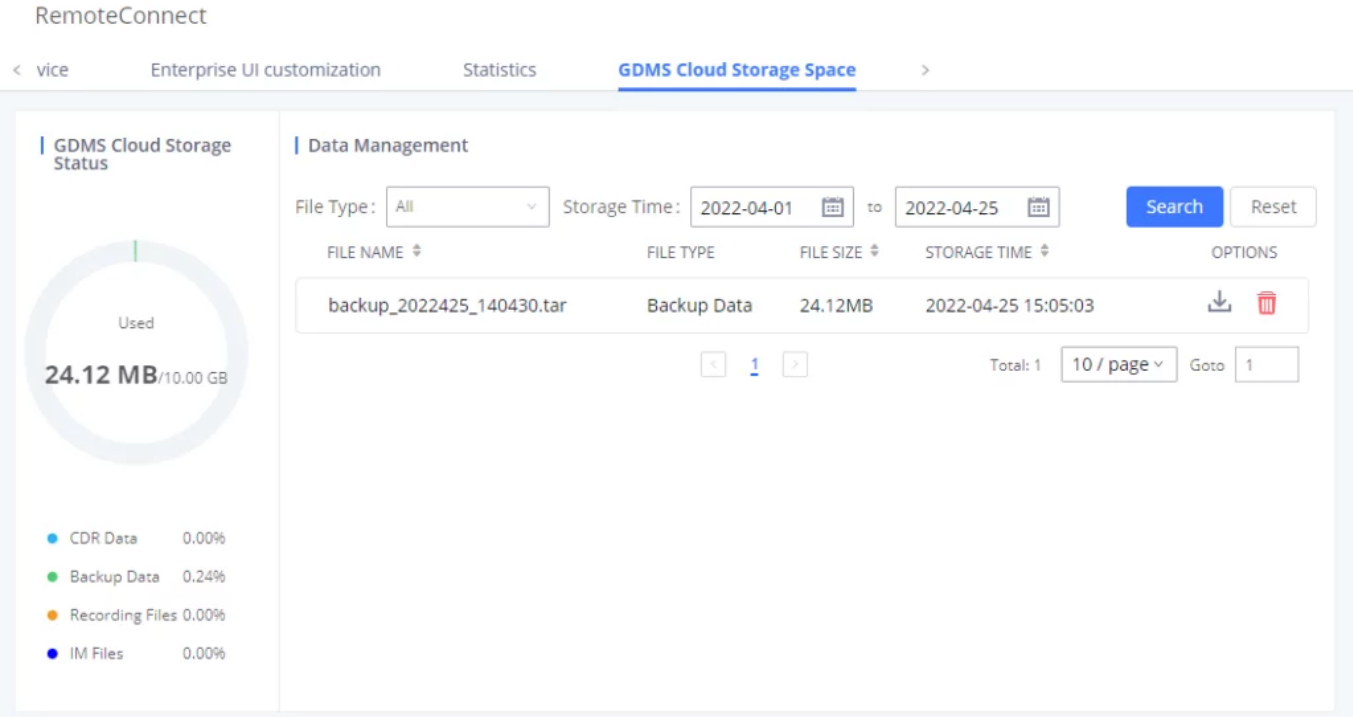
Concurrent Remote Calls

For more information, please visit http://ucmrc.gdms.cloud/intro.html

GDMS Cloud Storage Space

When the correspondent RemoteConnect plan is active, the user can access to a GDMS Cloud Storage Space to get an overview about how the storage space is being used. The data is represented in four categories of file types: CDR Data, Backup Data, Recording Files, and IM Files.

The user is also able to see the names of all the files stored in the GDMS Cloud Storage Space.



GDMS Cloud Storage Space

INTEGRATIONS

API Configuration

PBX API allows creating an interface between the IPPBX and an application to send HTTPS requests to the database of the IPPBX.

API Settings

Before accessing the API, the administrators need to enable API and configure the access/authentication information on the PBX first under **Integrations → API Configuration**. The API configuration parameters are listed in the tables below.

You can create multiple users with different login credentials to access to the HTTPS API.

For more details on CDR API (Access to Call Detail Records), REC API (Access to Call Recording Files) and PMS API, please refer the document in the link here:

- https://documentation.grandstream.com/knowledge-base/cdr-rec-api/
- https://documentation.grandstream.com/knowledge-base/cdr-rec-api/
- PMS API

API Queries Supported

The new API supports now new queries listed below which will accomplish certain requests and get data about different modules on PBX.

getSystemStatus	addInboundRoute	listPaginggroup
getSystemGeneralStatus	getInboundRoute	addPaginggroup
listAccount	updateInboundRoute	getPaginggroup
getSIPAccount	deleteInboundRoute	updatePaginggroup
updateSIPAccount	playPromptByOrg	deletePaginggroup
listVoIPTrunk	listBridgedChannels	MulticastPaging
addSIPTrunk	listUnBridgedChannels	MulticastPagingHangup
getSIPTrunk	Hangup	listIVR
updateSIPTrunk	Callbarge	addIVR
deleteSIPTrunk	listQueue	getIVR
listOutboundRoute	getQueue	updateIVR
addOutboundRoute	updateQueue	deleteIVR
getOutboundRoute	addQueue	cdrapi
updateOutboundRoute	deleteQueue	recapi
deleteOutboundRoute	loginLogoffQueueAgent	pmsapi
listInboundRoute	pauseUnpauseQueueAgent	queueapi
mute	Unmute	hold
unhold	dialExtension	dialOutbound
callTransfer	transferNumberInbound	transferNumberOutbound
dialIVR	dialIVROutbound	dialQueue
dialRinggroup	dialOutboundTwo	listUser
getUser	updateUser	listExtensionGroup
listPinSets	refuseCall	acceptCall
applyChanges	addSLATrunk	updateSLATrunk
deleteDigitalTrunk	addDigitalTrunk	getDigitalTrunk
listDigitalTrunk	updateDigitalTrunk	listDepartment
getRecordInfosByCall	addMessageBroadcast	

CDR Real-time Output Settings	
Enable	Enables real-time CDR output module. This module connects to selected IP addresses and ports and posts CDR strings as soon as it is available.
Server Address	CDR server IP address
Port	CDR server IP port
Upload Prompts User Configuration	
Username	Username used to upload prompts.
Password	Password used to upload prompts.

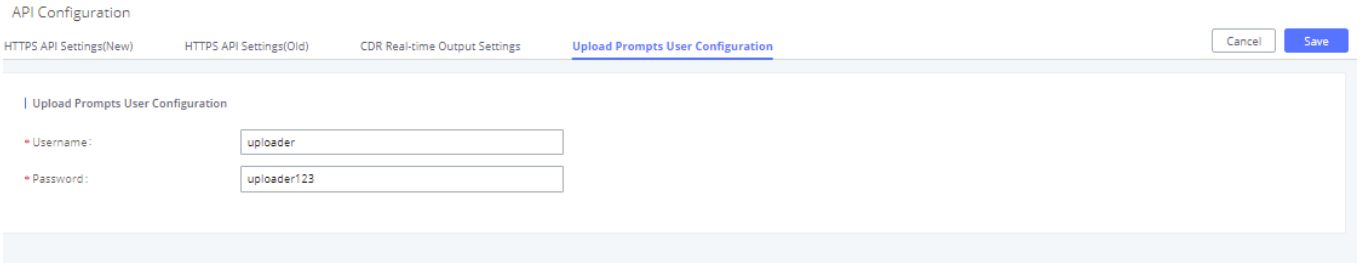
Upload Voice Prompt via API

Customers now can use the "Upload Prompts User Configuration" to upload/replace voice prompt files as an alternative method to the manual upload method on IPPBX **PBX Settings→ Voice Prompt→ Custom Prompt.**

The workflow of the prompt file upload goes as:

An HTTP/HTTPS request is sent to the IPPBX to upload/replace a voice prompt file, the request should include authentication details to the IPPBX and the name of the file to be uploaded. Then the IPPBX will contact an FTP server that should be hosted on the same IP address of the HTTP/HTTPS requester and download the prompt file from the FTP server.
The steps and conditions to upload the voice prompt via API are listed below:

1. Configure the prompt User under **Integrations→ API Configuration →Upload Prompts User Configuration.** By default, the username and password for voice prompt user are "Username: uploader; Password: uploader123".



Upload Prompt User Configuration

2. Hash the password of the user configured to an MD5 Encryption format.
3. Set the permission on the FTP server to Anonymous on the local computer hosting the FTP server and make sure that the default FTP port 21 is used.
4. Send an HTTP/HTTPS command to trigger the Prompt file upload on the IPPBX. If IPPBX's HTTP server is set to HTTPS, the example of the request sent to the IPPBX is:

```
https://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3
```

If IPPBX's HTTP server is set to HTTP, the example of the request sent to the IPPBX is:

```
https://192.168.124.89:8089/cgi?action=uploadprompt&username=uploader&password=9191a6394c21b3aabd779213c7179462&filename=test.mp3
```

If the File name on the HTTP/HTTPS request exists already on the IPPBX's Custom voice prompts list the existing file will be overwritten by the new file downloaded from the FTP server.

For more details on CDR API (Access to Call Detail Records) and REC API (Access to Call Recording Files), please refer the document in the link here:

AMI

The IPPBX supports Asterisk Manager Interface (AMI) with restricted access. AMI allows a client program to connect to an Asterisk instance commands or read events over a TCP/IP stream. It is particularly useful when the system admin tries to track the state of a telephony client inside Asterisk.

User could configure AMI parameters on IPPBX Web GUI→**Integrations→AMI**. For details on how to use AMI on IPPBX, please refer to the following AMI guide:

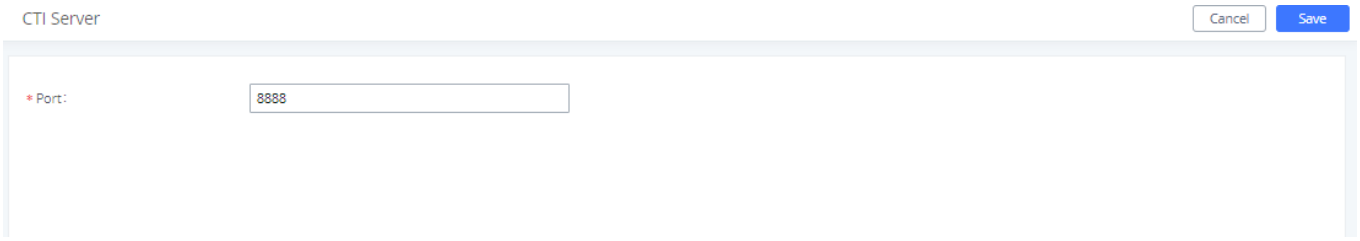
Please do not enable AMI on the IPPBX if it is placed on a public or untrusted network unless you have taken steps to protect the device from unauthorized access. It is crucial to understand that AMI access can allow AMI user to originate calls and the data exchanged via AMI is often very sensitive and private for your IPPBX system. Please be cautious when enabling AMI access on the IPPBX and restrict the permission granted to the AMI user. By using AMI on IPPBX you agree you understand and acknowledge the risks associated with this.

CTI Server

IPPBX does support CTI server capabilities which are designed to be a part of the CTI solution suite provided by Grandstream, including GXP21XX and GXP17XX enterprise IP phones along with GS Affinity app.

Mainly the IPPBX will by default listening on port TCP 8888 for the connections from GS affinity application in order to interact, modify and serve data requests by the application which includes setting call features for the connected extension as call forward and DND.

Users can change the listening port under the menu page, Web GUI→**Integrations→CTI Server** as shown on below screenshot:



CTI Server Listening port

More information about GS affinity and CTI Support on Grandstream products series please refer to the following link:

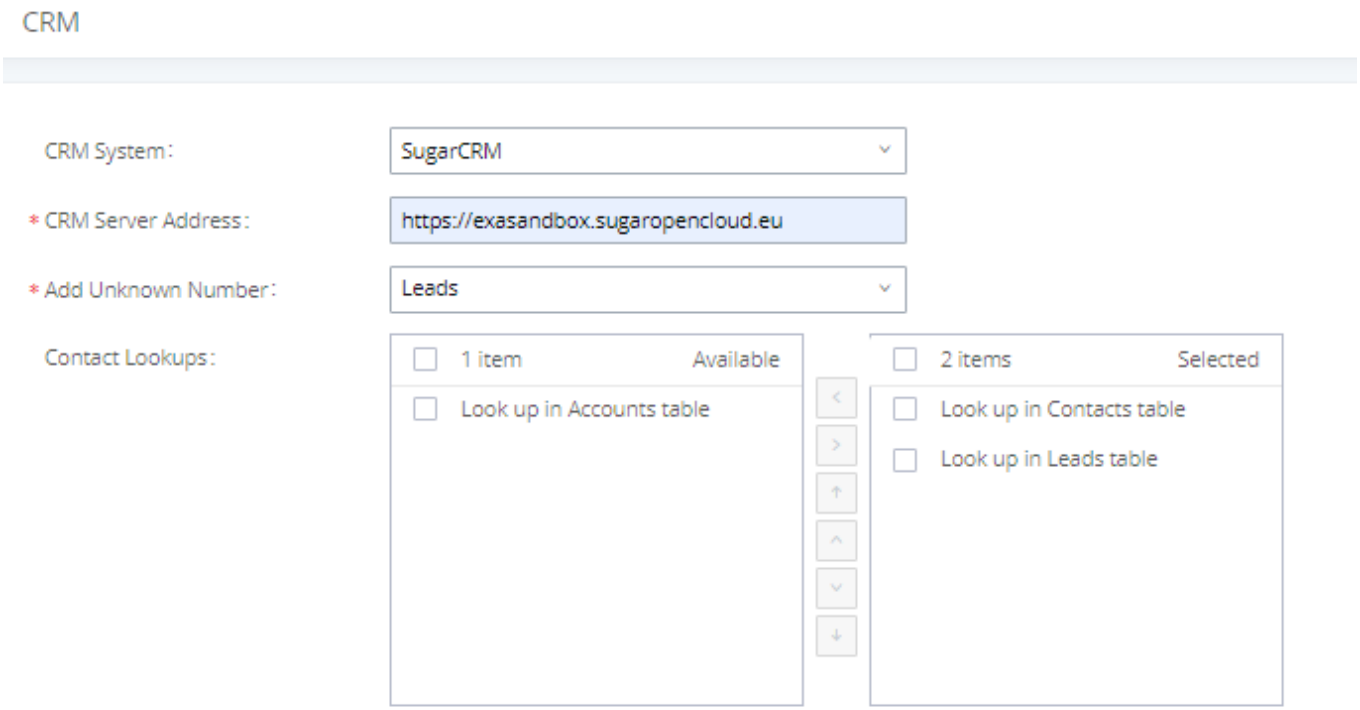
CRM

Customer relationship management (**CRM**) is a term that refers to practices, strategies and technologies that companies use to manage and analyze customer interactions and data throughout the customer lifecycle, with the goal of improving business relationships with customers.

The PBX support the following CRMs: SugarCRM, VtigerCRM, Salesforce CRM and ACT! CRM, which allows users to look for contact information in the Contacts, Leads and / or Accounts tables, shows the contact record in CRM page, and saves the call information in the contact's history.

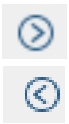
Sugar CRM

Configuration page of the SugarCRM can be accessed via admin login, on the IPPBX WebGUI→ **Integrations→CRM**.



SugarCRM Basic Settings

1. Select "SugarCRM" from the CRM System Dropdown in order to use SugarCRM.

CRM System	Select a CRM system from the dropdown menu.
CRM Server Address	Enter the IP address of the CRM server.
Add Unknown Number	Add the new number to this module if it cannot be found in the selected module.
Contact Lookups	 Select from the "Available" list of lookups and press to select where the IPPBX can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

SugarCRM Settings

Once settings on admin access are configured:

2. Click on and .

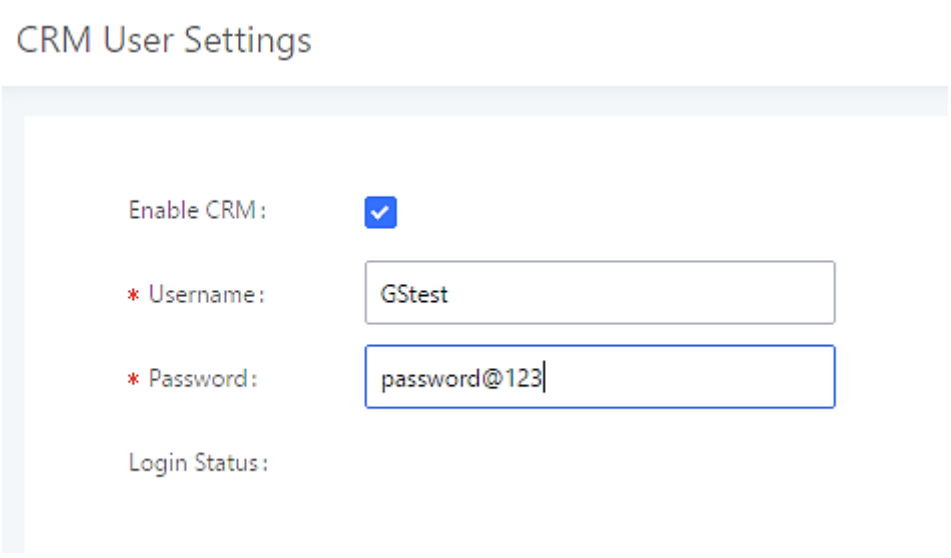
3. Logout from admin access.

4. Login to the IPPBX as user and navigate under "**User Portal→Other Feature→CRM User Settings**".

Click on "**Enable CRM**" and enter the username/password associated with the CRM account then click on

and

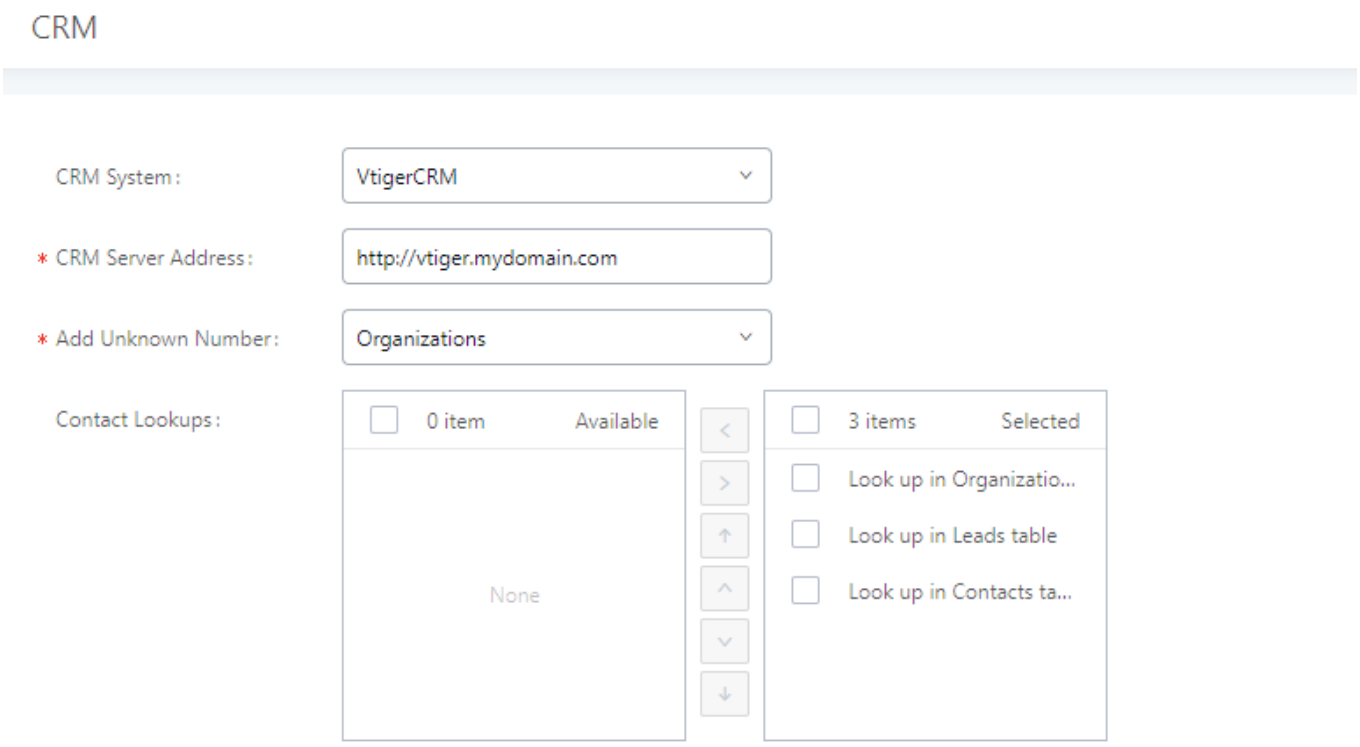
. The status will change from "Logged Out" to "Logged In". User can start then using SugarCRM features.



CRM User Settings

Vtiger CRM

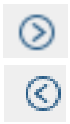
Configuration page of the VtigerCRM can be accessed via admin login, on the IPPBX WebGUI > **Integrations** > **CRM**.



VtigerCRM Basic Settings

1. Select "Vtiger CRM" from the CRM System Dropdown in order to use Vtiger CRM.

CRM System	Select a CRM system from the dropdown menu.
CRM Server Address	Enter the IP address of the CRM server.
Add Unknown Number	Add the new number to this module if it cannot be found in the selected module.

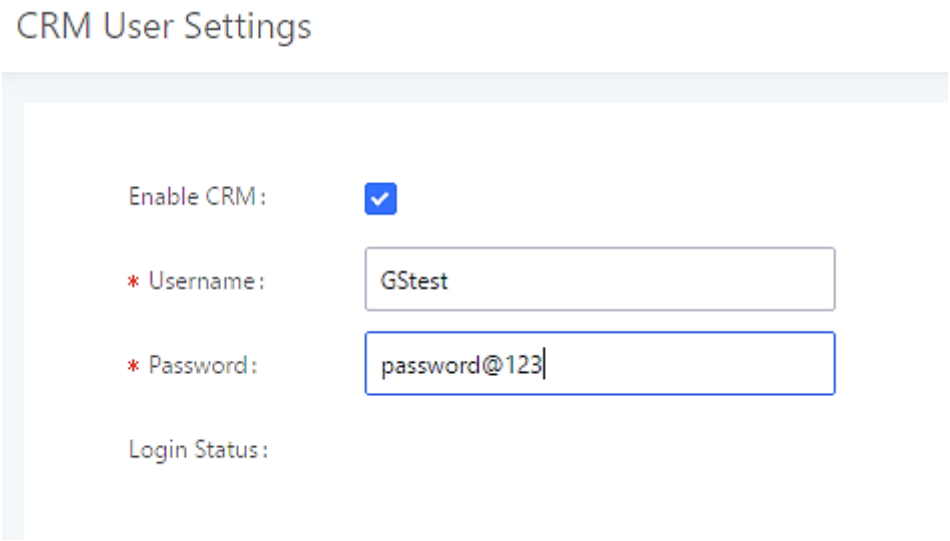
Contact Lookups	 Select from the "Available" list of lookups and press to select where the IPPBX can perform the lookups on the CRM tables, Leads, Organizations, and Contacts.

VtigerCRM Settings

Once settings on admin access are configured:

2. Click on and .
3. Logout from admin access.
4. Login to the IPPBX as user and navigate under "**User Portal→ Integrations →CRM User Settings**".

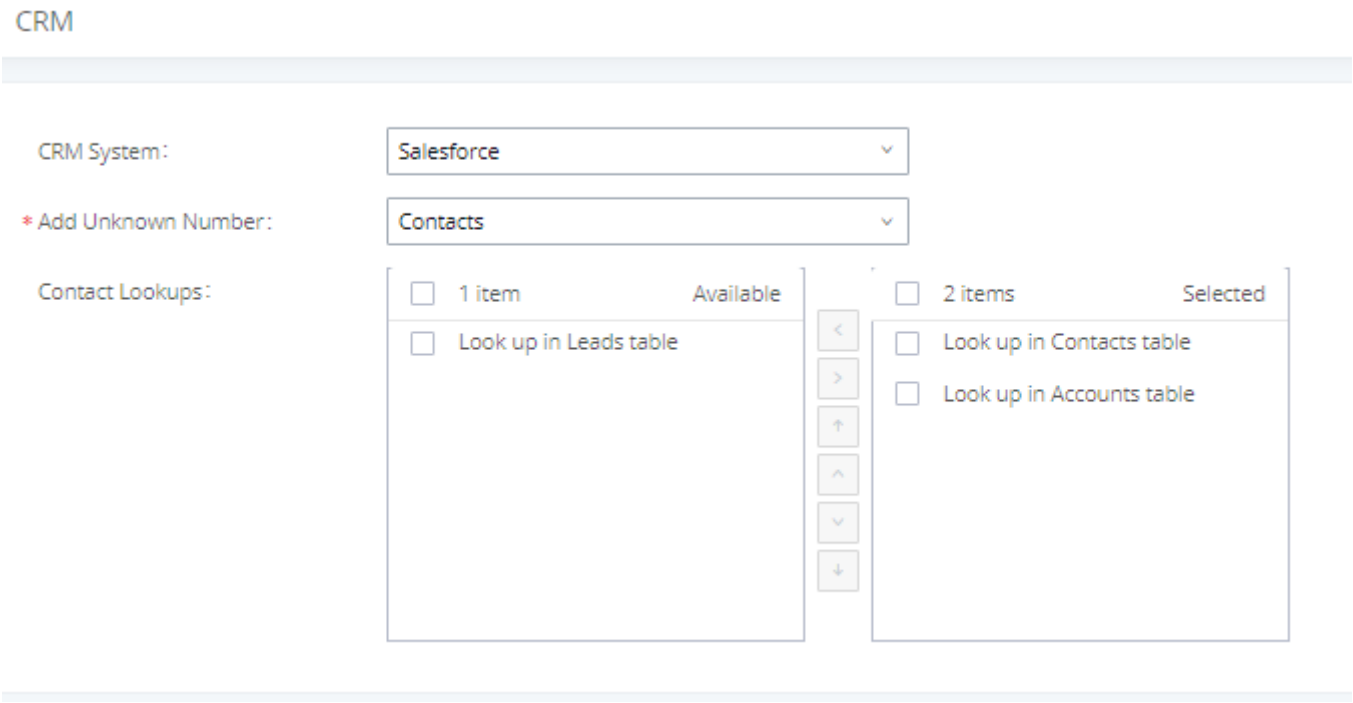
Click on "**Enable CRM**" and enter the username/password associated with the CRM account then click on and . The status will change from "Logged Out" to "Logged In". User can start then using SugarCRM features.



CRM User Settings

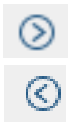
Salesforce CRM

Configuration page of the Salesforce CRM can be accessed via admin login, on the IPPBX Web GUI→ **Integrations →CRM**".



Salesforce Basic Settings

1. Select "Salesforce" from the CRM System Dropdown in order to use Salesforce CRM.

CRM System	Select a CRM system from the dropdown menu, four CRM systems are available: SugarCRM, VtigerCRM, Salesforce and ACT! CRM.
Add Unknown Number	Add the new number to this module if it cannot be found in the selected module.
Contact Lookups	 Select from the "Available" list of lookups and press to select where the IPPBX can perform the lookups on the CRM tables, Leads, Accounts, and Contacts.

Salesforce Settings

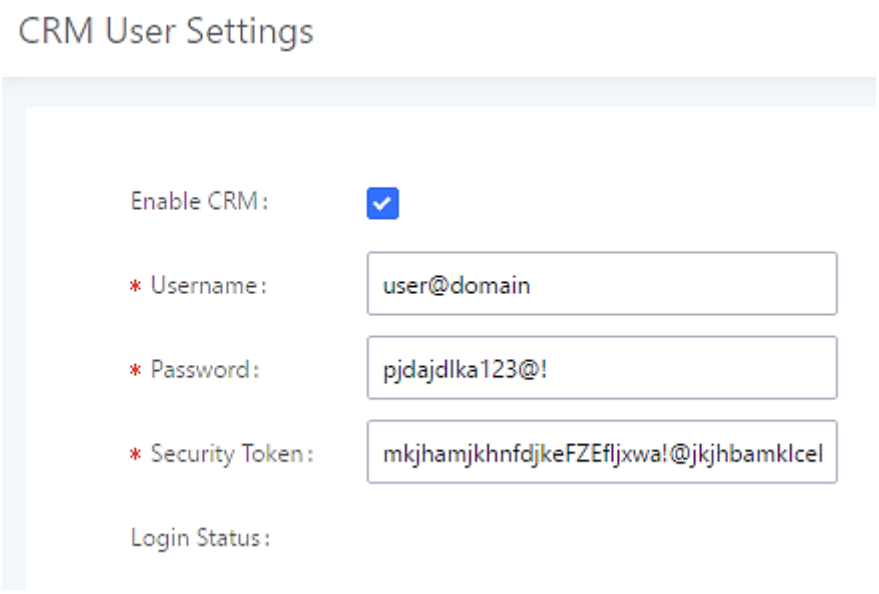
Once settings on admin access are configured:

2. Click on and .

3. Logout from admin access.

4. Login to the IPPBX as user and navigate under "**User Portal→ Integrations →CRM User Settings**".

Click on "**Enable CRM**" and enter the **username**, **password** and **Security Token** associated with the CRM account then click on and . The status will change from "Logged Out" to "Logged In". User can start then using Salesforce CRM features.



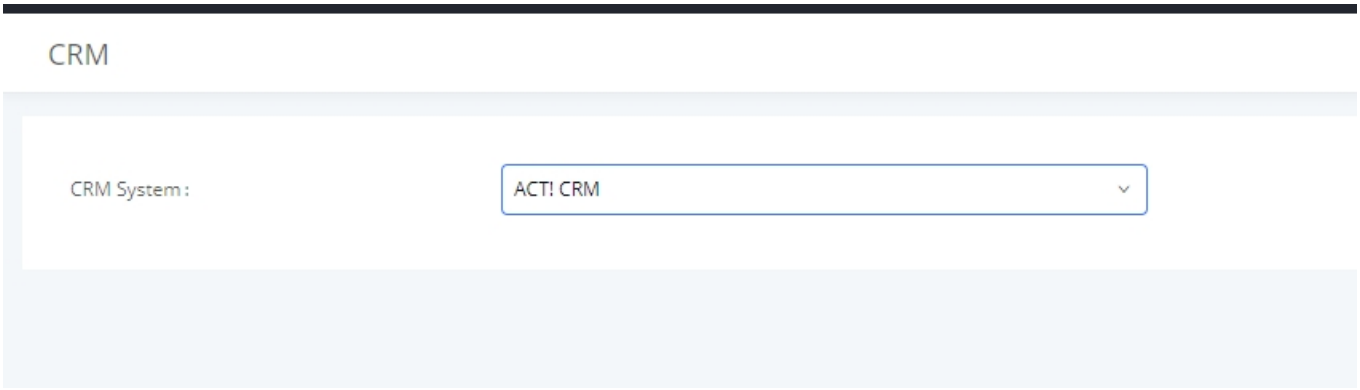
Salesforce User Settings

ACT! CRM

Configuration page of the ACT! CRM can be accessed via admin login, on the IPPBX Web GUI→ **Integrations→CRM**".

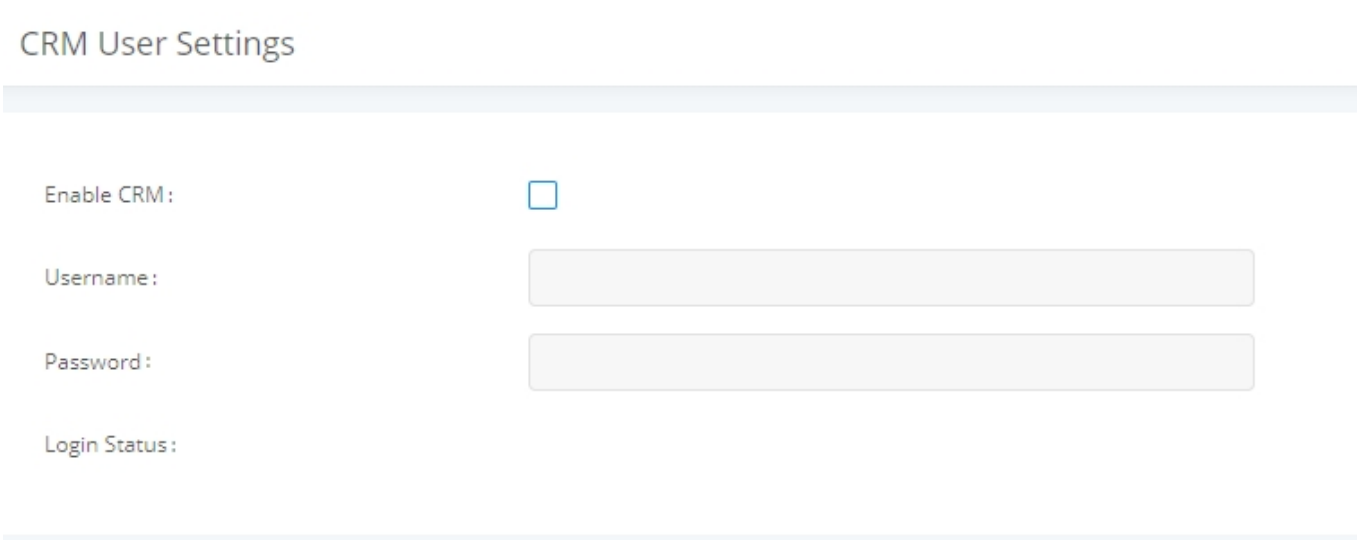
The configuration steps of the ACT! CRM are as follows:

1. Navigate to **Integrations→CRM** and select the "ACT! CRM" option.



Enabling ACT CRM

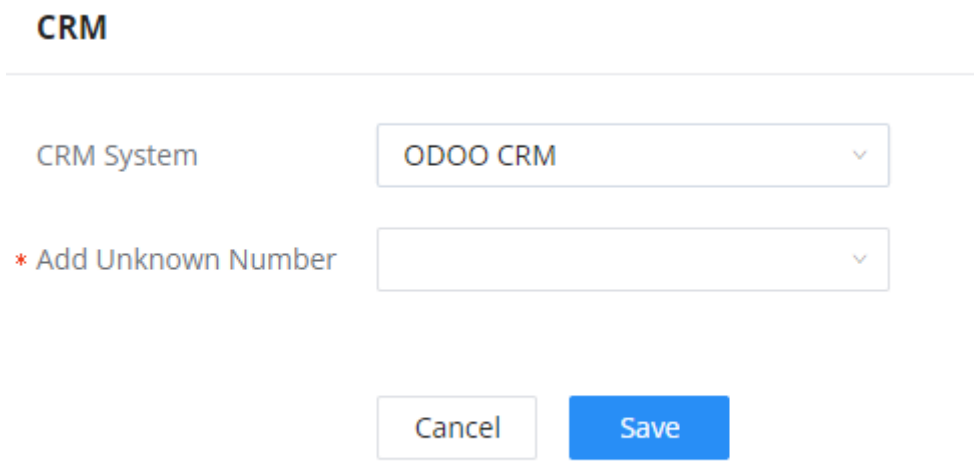
2. Log into the IPPBX as a regular user and navigate to **Integrations→CRM User Settings** and check "Enable CRM" option and enter the username and password, which will be the ACT! CRM account's **API Key** and **Developer Key,** respectively. To obtain these, please refer to the ACT! CRM API developer's guide here: https://www.act.com/



Enabling CRM on the User Portal

Odoo CRM

The GCC PBX supports integration with Odoo CRM. To enable Odoo CRM, please access the PBX's web UI then navigate to **Integration > CRM**.

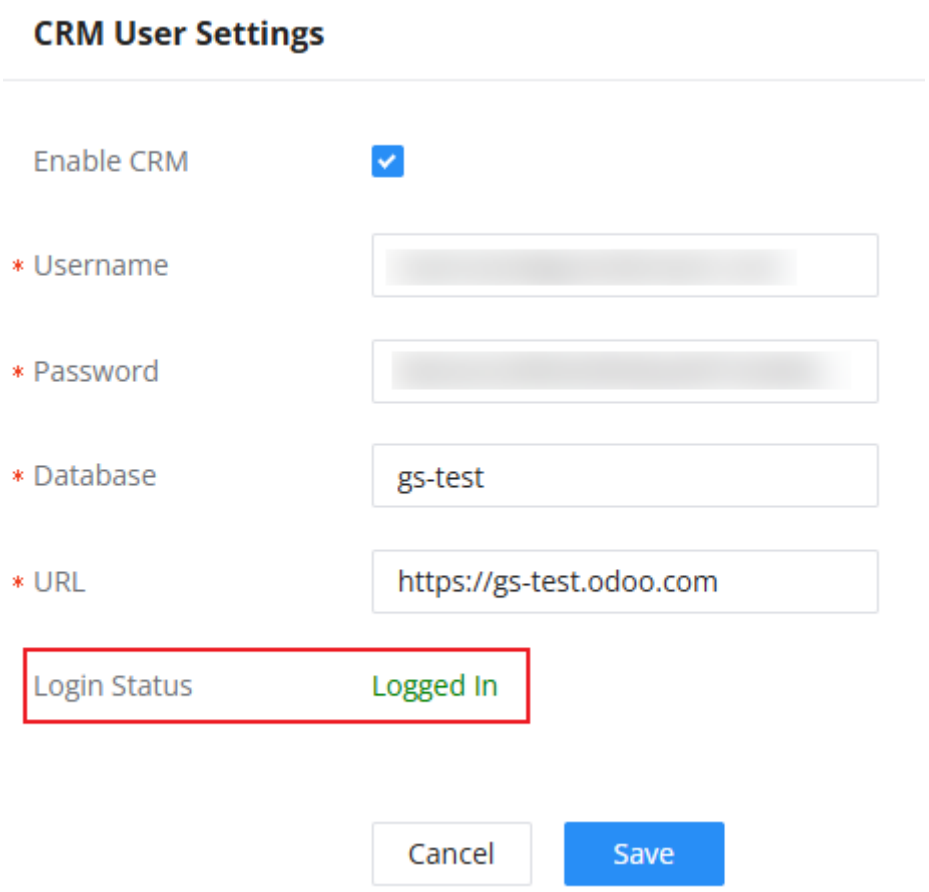


Odoo CRM Configuration

Click "Save" then "Apply Changes".

Once Odoo CRM is enabled, the user can log into the user portal by access the PBX web UI and entering their username and password, then they can navigate to **Integrations > CRM User Settings**, then they configure their account using username and password created on Odoo platform.

When the integration has been successfully established, the Login Status will indicate that the user has successfully logged in, as the figure below indicates.



CRM User Settings

PMS

PBX supports Hotel Property Management System PMS, including check-in/check-out services, wakeup calls, room status, Do Not Disturb which provide an ease of management for hotel applications. This feature can be found on Web GUI→**Integrations→PMS.**

PMS API

The PMS API allows users to use their own middleware to work with PMS systems instead of currently supported integrations.

Additionally, this API allows access to read and modify certain IPPBX parameters that current supported PMS integrations cannot. To use this, users must first enable and configure the HTTPS API settings.

For more details, please refer to online https://documentation.grandstream.com/knowledge-base/https-api/, Pmsapi section.

Connecting to PMS

On the IPPBX WebGUI→**Integrations→PMS→Basic Settings**" set the connection information for the PMS platform.

Field	Description
PMS Module	Users can select the desired PMS module from the drop-down list. char pms Mitel HSC IDS PMS API
Wakeup Prompt	A customized prompts that can be played when the wakeup call is answered. To customize it please navigate to PBX Settings ? Voice Prompt ? Custom Prompt
Username	This username is used to authenticate into the PMS API.

Password	This password is used to autheticate into the PMS API.
Back Up Voicemail Recordings	Back up voicemail recordings to external storage after check-out.
Automatically Clear Wakeup Calls	Scheduled wakeup calls for rooms can be cleared upong checking in or checking out. None: The wakeup calls won't be automatically cleared. Check out: The wake up calls assigned to the guest will be cleared when they check out. Check In: The wake up calls assigned to a guest will be cleared when a new client checks in.
Automatically Clear Wave Chat History	If enabled, room Wave chat history will be automatically cleared upon check-in or check-out.
Automatically Reset User/Wave Password	If enabled, the User/Wave password of the room extension will be automatically reset to a random password upon check-out.

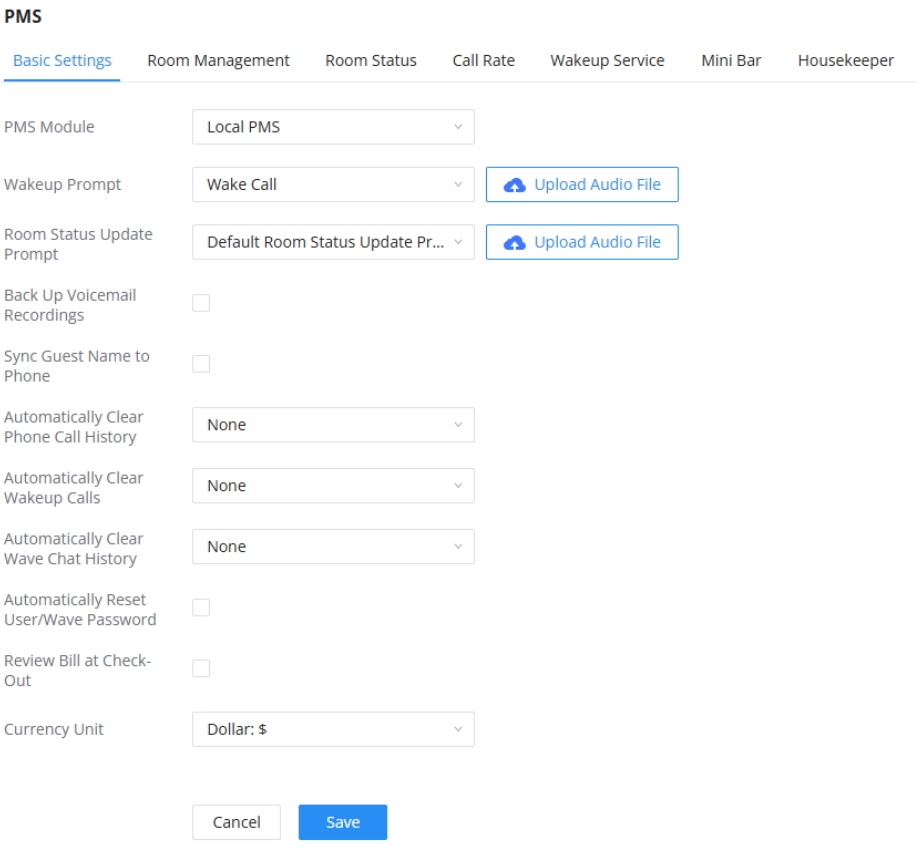
In order to use some PMS features please activate the feature code associated under "**Basic Call Features →Feature Codes**"

- Update PMS Room Status
- PMS Wake Up Service

Local PMS

PBX offers a local Property Management System to give the user basic management features without having to purchase a PMS for the most basic property management actions. In addition to Room Management, Rooms Status for checking-in and checking-out, Wakeup Service, and Housekeeper functions, the PBX allows a number of additional functions upon checking-out, like backing up voicemail recordings, clearing wakeup calls and Wave history automatically, in addition to resetting Wave's password. The user can use the Local PMS feature to check-in and check-out clients from the web user interface.

To access Local PMS, please navigate to **Integrations > PMS > Basic Settings**, and choose Local PMS as the PMS Module.



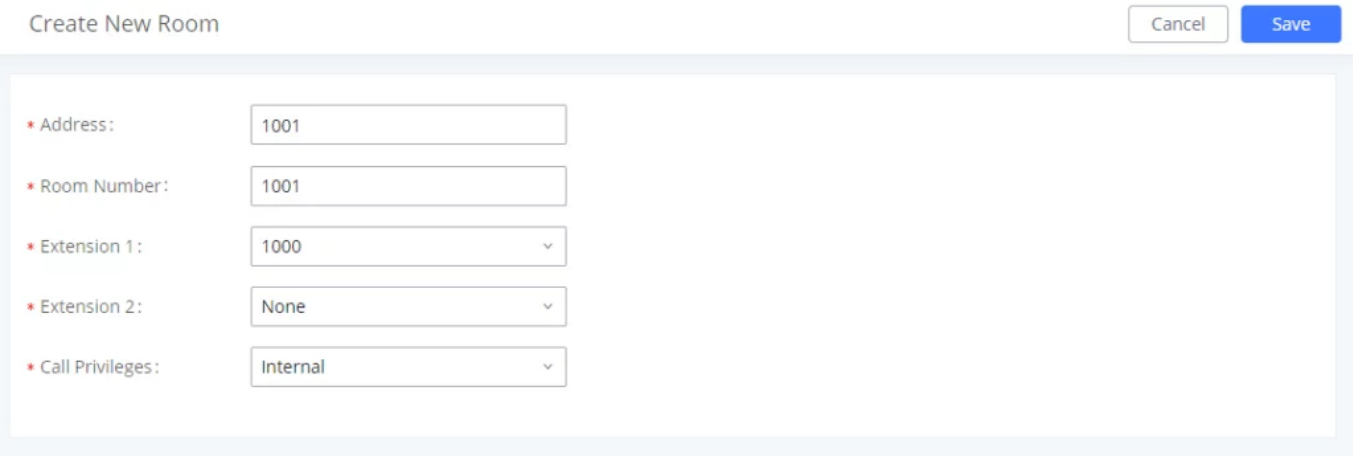
Local PMS

Parameter	Description
Wakeup Prompt	A customized prompts that can be played when the wakeup call is answered. To customize it please navigate to PBX Settings ? Voice Prompt ? Custom Prompt
Room Status Update Prompt?	Choose a previously uploaded prompt or upload the prompt which will be played when the room status is changed. If the room status codes have been change, please update the Room Status accordingly.
Back Up Voicemail Recordings	Back up voicemail recordings to external storage after check-out, When Enabled, The user can set the SFTP server for storage purposes by defining the following attributes : Email Address: Configure the email address to send the backup to.Account: Configures the account on the SFTP server.Password: Defines the account passwordServer Address: Defines the SFTP server address (e.g., xxx.xxx.xxx.xxx:22).Destination Directory: Specify the directory in SFTP server to save the voicemail recordings to. Format: "xxx/xxx/xxx". If this directory does not exist, UCM will create this directory automatically.Test the Connection: This option tests the connection to the SFTP server defined.
Sync Guest Name to Phone	Provisions the name of the checked-in guests to endpoints via Zero Config. Requires endpoints to be discoverable and provisinable by Zero Config.
Automatically Clear Phone Call History	Configures whether or not the call history of phones will be automatically cleared upon check-in or check-out. Currently only supported on Grandstream phones.None: Call history will not be deleted after checking-in or checking-out. Check Out: Call history will be delete when the guest checks-out. Check In: Call history will be delete when a new guest checks-in.
Automatically Clear Wakeup Calls	Scheduled wakeup calls for rooms can be cleared upong checking in or checking out.None: The wakeup calls won't be automatically cleared.Check out: The wake up calls assigned to the guest will be cleared when they check out.Check In: The wake up calls assigned to a guest will be cleared when a new client checks in.
Automatically Clear Wave Chat History	If enabled, room Wave chat history will be automatically cleared upon check-in or check-out.
Automatically Reset User/Wave Password	If enabled, the User/Wave password of the room extension will be automatically reset to a random password upon check-out.
Review Bill at Check-Out	If enabled, a pop-up window with all the charges will appear during guest check-out for reviewing purposes.
Currency Unit	The currency unit for the call rate. These are the supported options which you can choose from: American Dollar Euro Sterling Pound Yen Won Hong Kong Dollar Australian Dollar Canadian Dollar BahtSingapore Dollar Swiss Franc Swedish Krona Danish Krone Norwegian Krone New Zealand Dollar South African Rand Brazilian Rial Indian Rupee Russian Ruble Vietnamese Dong Polish Zolty Czech Koruna Turkish Lira Custom: Enter the currency unit.
Periodically Email Recordings	If enabled, new recordings will be automatically sent to the specified email inbox periodically at the specified intervals.
Type	Configures the type of recordings that should be sent periodicallyWakeup ServiceCheck-in/Check-out HistoryCDR Records
Period	Specifies the automatic download period. The default option is "By Day". At the specific time of each day, week or month, the last cycle of new records will be sent to the user.By DayBy WeekBy Month
Email	Specifies the automatic download period. The default option is "By Day". At the specific time of each day, week or month, the last cycle of new records will be sent to the user.

Please note that the following features appear only when choosing "Local PMS" in the PMS Module.

Room Management

In Room Management tab, the user can create a room and affect up to two extensions to it. This will appear in Room Status tab, and from there the user can change the Check-in/Check-out.

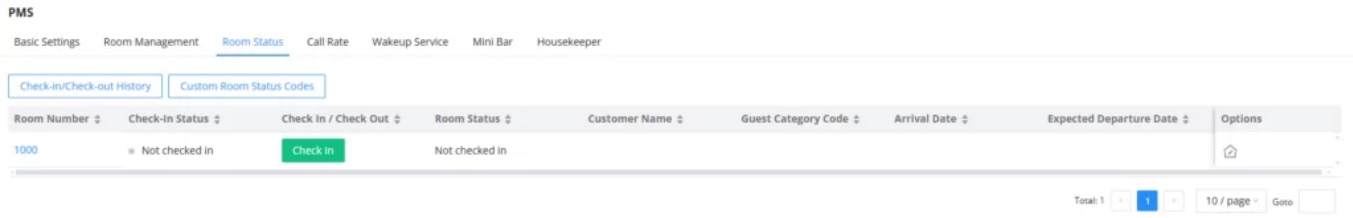


Create New Room

Call Privileges allows the administrator to set the level of call privilege of the room.

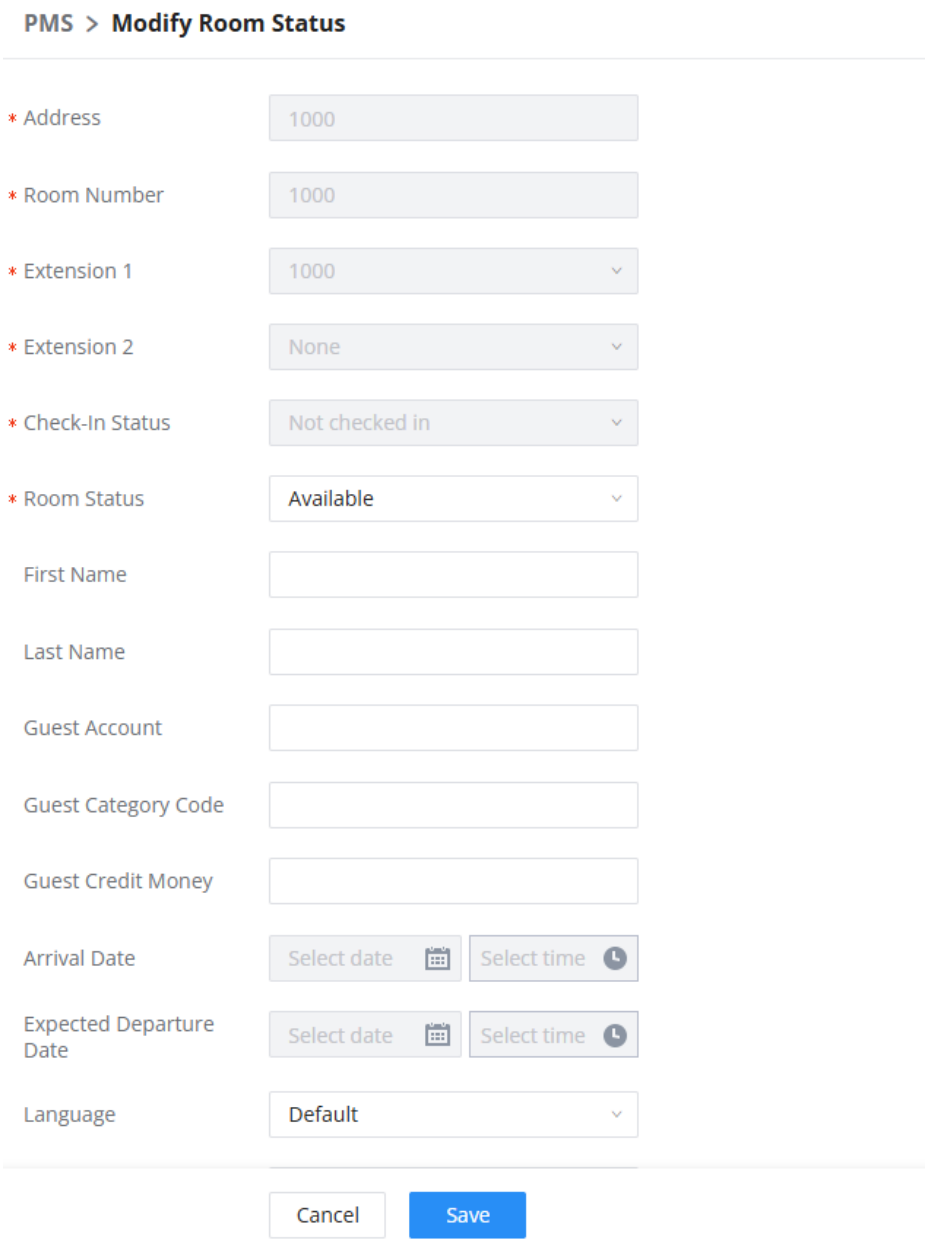
Room Status

In this page, the user can view the rooms created with their related information. The user can also check guests in and out from this page.

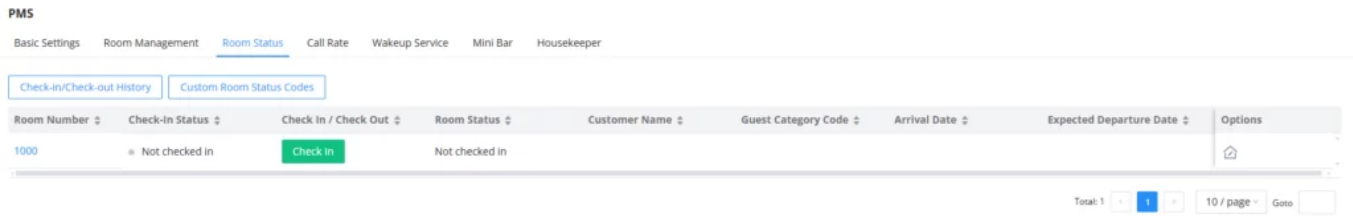


Room Status

To change the status of a room, please click on the button, and the page in the figure below will appear.



Change Room Status



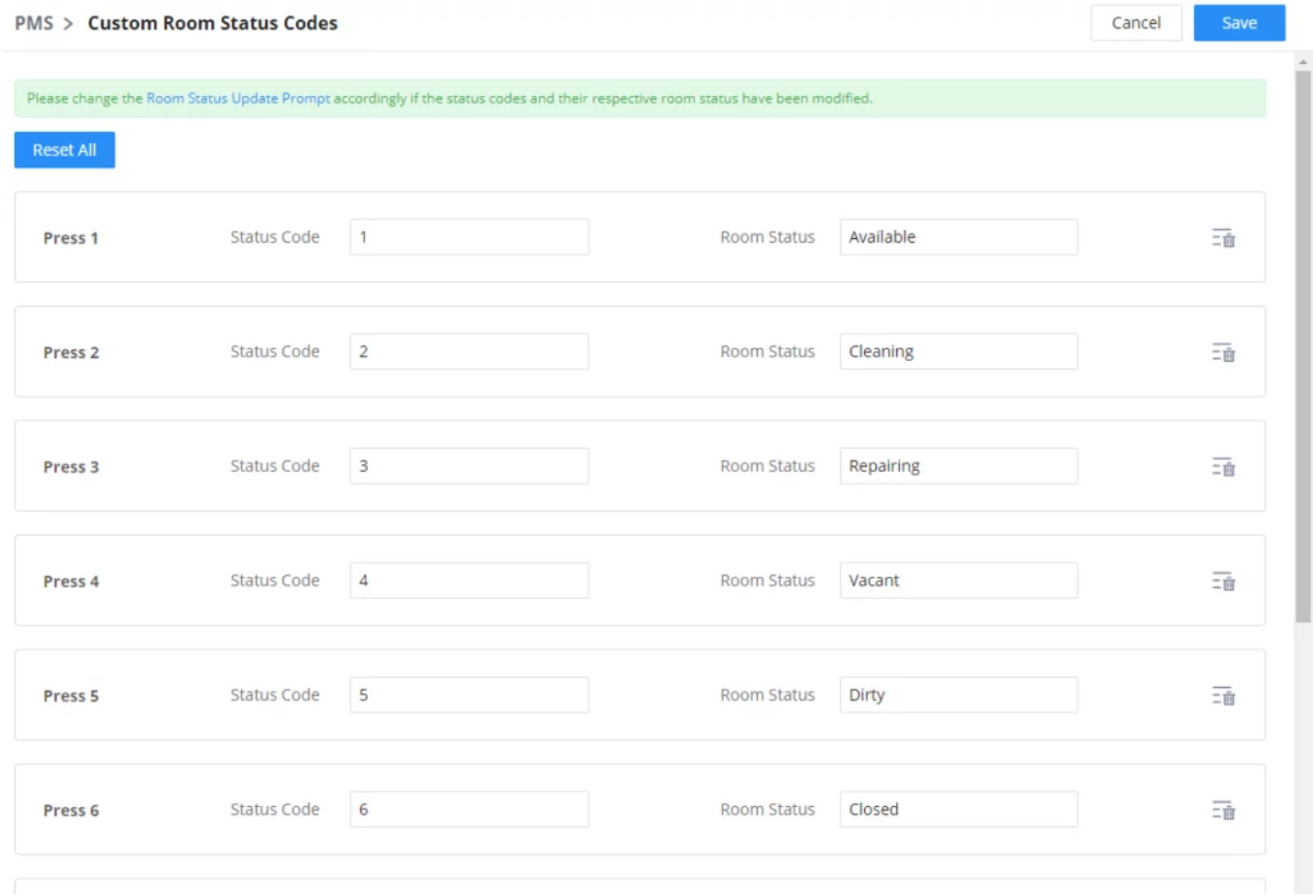
Room Check in

The user can click on **Check-in/Check-out Records** to view the history of the checked-in and checked-out guests.

The Call Privilege configured during a guest's check-in will be reset to the room's default call privilege upon guest check-out.

Custom Room Status Codes

The user can customize the existing room statuses or add more statuses along with the corresponding name. The user can customize the status code to up to 16-digit code. To customize room status, please click on .



Custom Room Status Codes

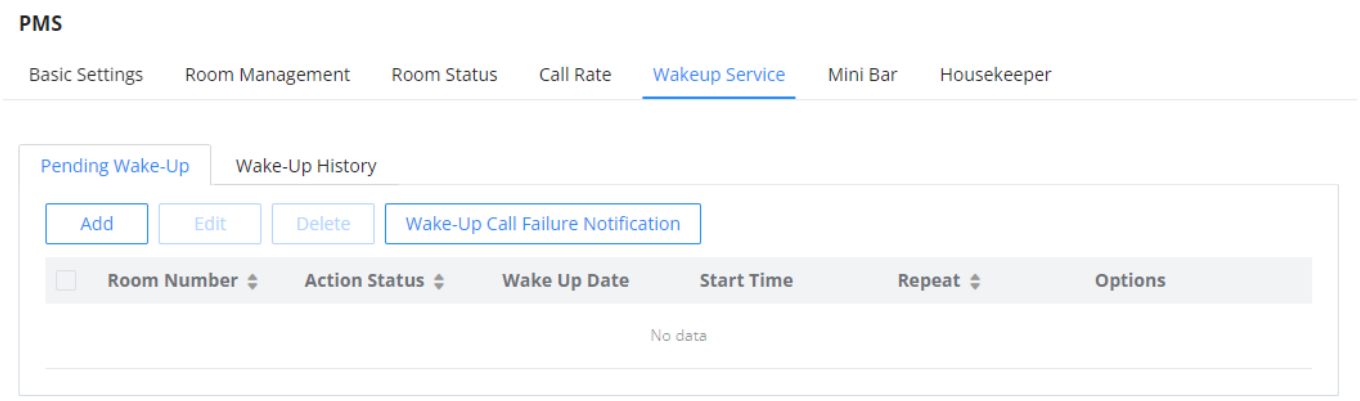
Wakeup Service

In some cases, guests will request the hotel staff to wake them up at a specific time, you can do that by configuring a wake-up time related to the room number of the guest, where the specific IP phone on that room will ring the extension related to the room number at that specific time, this option is supported on the integrated local PMS on the GCC PBX.

Pending Wake-up

In this page, the user can schedule wakeup calls, please follow the steps below.

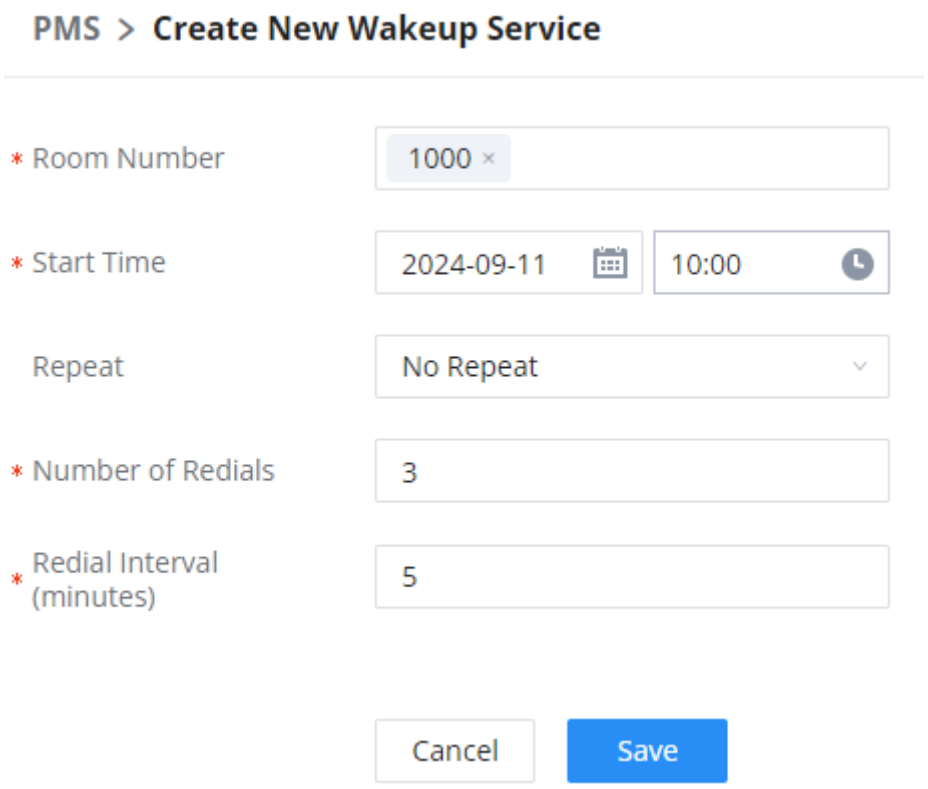
1. Select the wake-up service tab, click to create a new wakeup schedule.



Wakeup Service

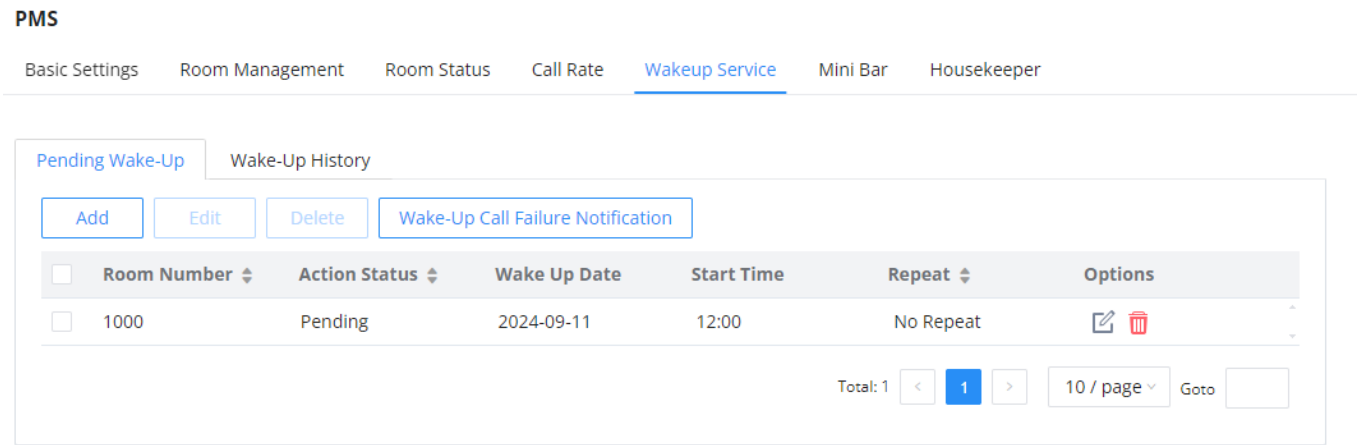
2. The configuration consists of defining some attributes such as :

- **Room number:** The room number on which the phone extension will ring at a specific time.
- **Start Time:** Define the time and the date of the wake up call
- **Repeat:** Select the frequency of the call: Daily, Weekly, Monthly.
- **Number of Redials:** Configures the number of times the system will repeat the call attempt after the task has started, but the call is not connected.
- **Redial Interval (minute):** The time interval between the end of the last call and the next initiated call when the Wake Up Call is not answered.



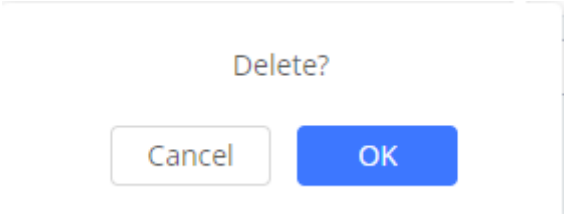
Create a New Wakeup Service

3. The newly displayed entry will be:



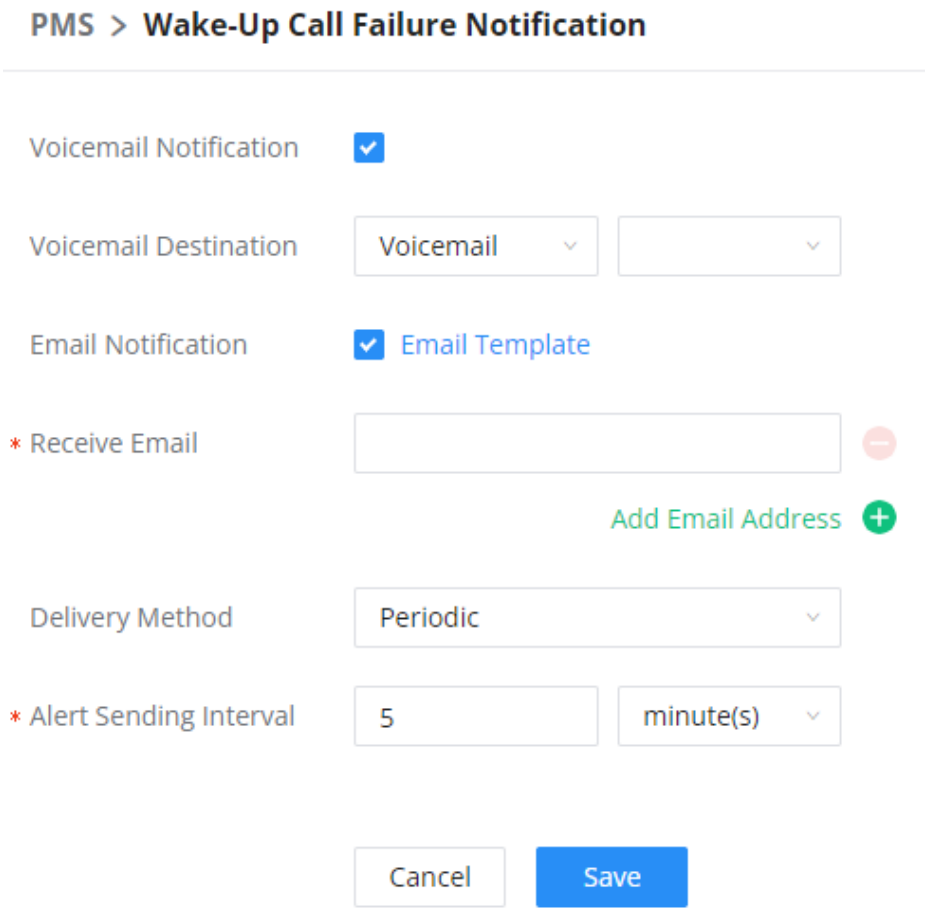
Wakeup Service

To delete a specific wake-up service, Click the icon and confirm the deletion by clicking "OK"



Delete Action Confirmation

The administrator can configure to send a voicemail or an email notification in case the Wake Up call is not answered.



Wakeup Call Failure Notification

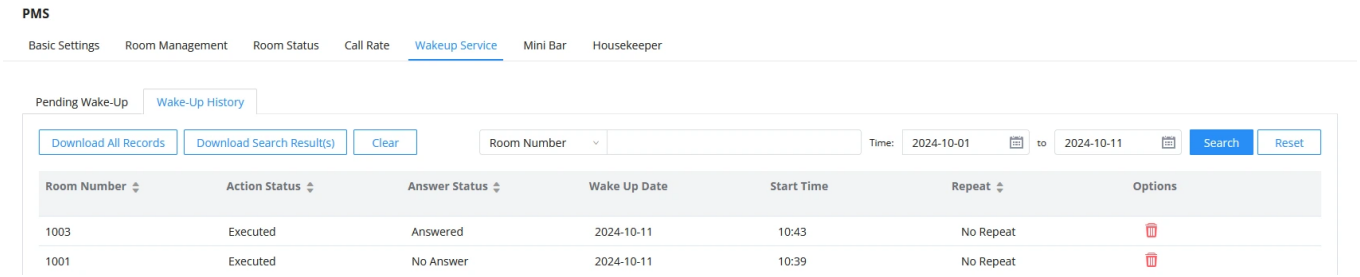
The parameters of the "Wake-Up Call Failure Notification" page are:

Voicemail Notification	Enable to send a notification to the configured voicemail extension/group when a wake-up call has failed, meaning it has non-answered status.Default settings is "disabled".Note: No notification will be sent if no failed wake-up call has occurred.
Voicemail Destination	Choose the voicemail/group to receive the failed wake-up call notifications.
Email Notification	Enable to send notifications of failed wake-up calls to the email addresses configured, based on the interval chosen. Any wake-up call that is not marked as "Answered" will be included in these notifications. If there are no failed wake-up calls, no alerts will be sent. Default settings is "disabled".
Receive Email	Configure the e-mail address to receive failed wake-up call notifications. Maximum number of allowed email addresses is 5.

Delivery Method	Choose the delivery method of the email notifications.Real-time: Notifications will be sent out immediately after alerts are generated.Periodic: Notifications will be queued up and sent out all at once every send cycle. The interval between each send cycle can be configured via the "Alert Sending Interval" option.Default setting is "Periodic".
Alert Sending Interval	The frequency of notification emails for all failed wakeup calls that happen during that cycle in minutes, hours or days. Default setting is 5 minutes.
Key Settings	Set the settings for the key. You can select predefined statuses for a specific key, and you can also enter a custom status which will be sent on the notificatiom email. The user can set a preloaded voice prompt or upload a custom one.

Wakeup History

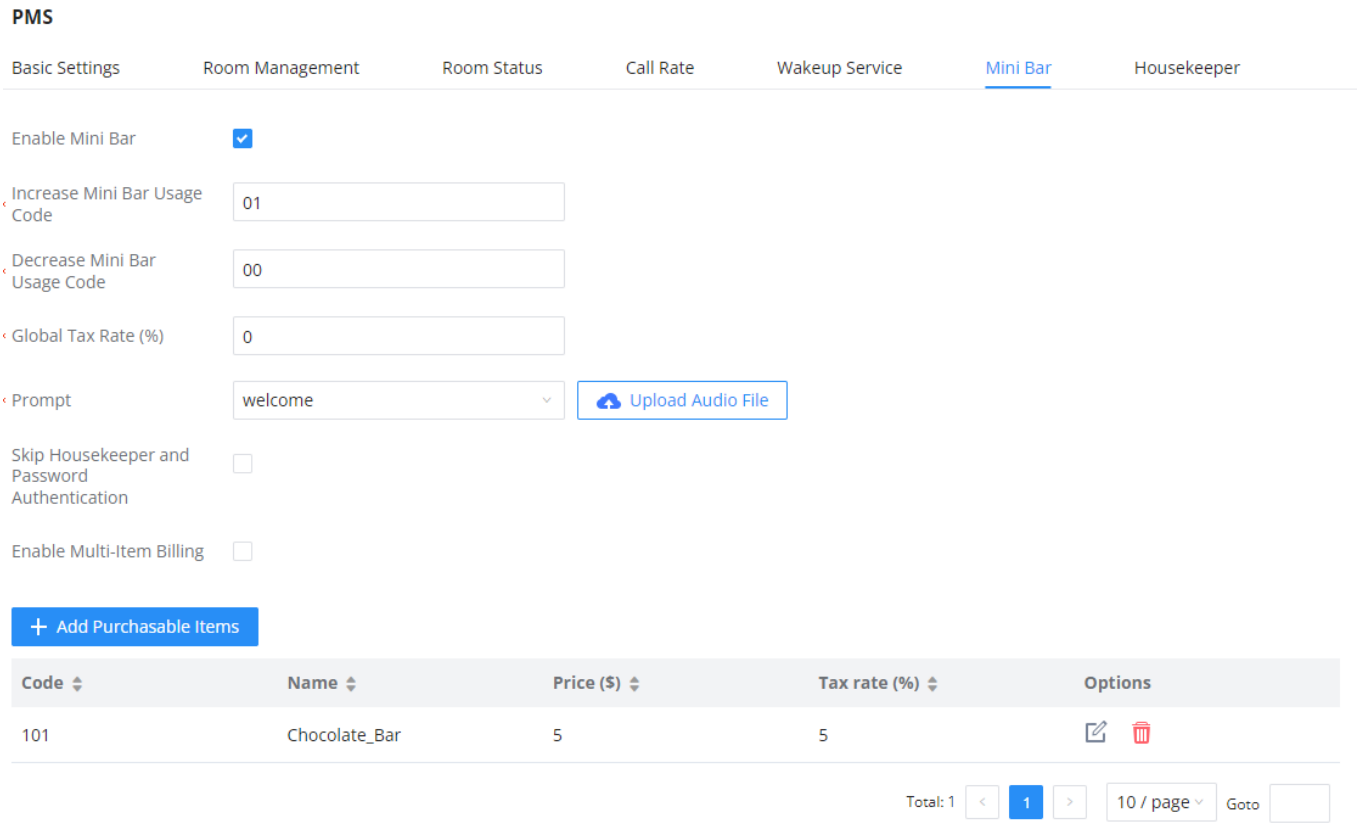
The Wake-Up History tab allows users to easily search for past wakeup calls by room number or answer status under a specified period. It is possibe to download a CSV file containing the specific search results or a file with all historical wakeup call data.



Wakeup History

Mini Bar

The mini bar feature allows billing for the goods consumed by the guest during his/her stay. The housekeeper can dial the feature code configured for "Increase Mini Bar Usage Code" or "Decrease Mini Bar Usage Code" to bill the goods.

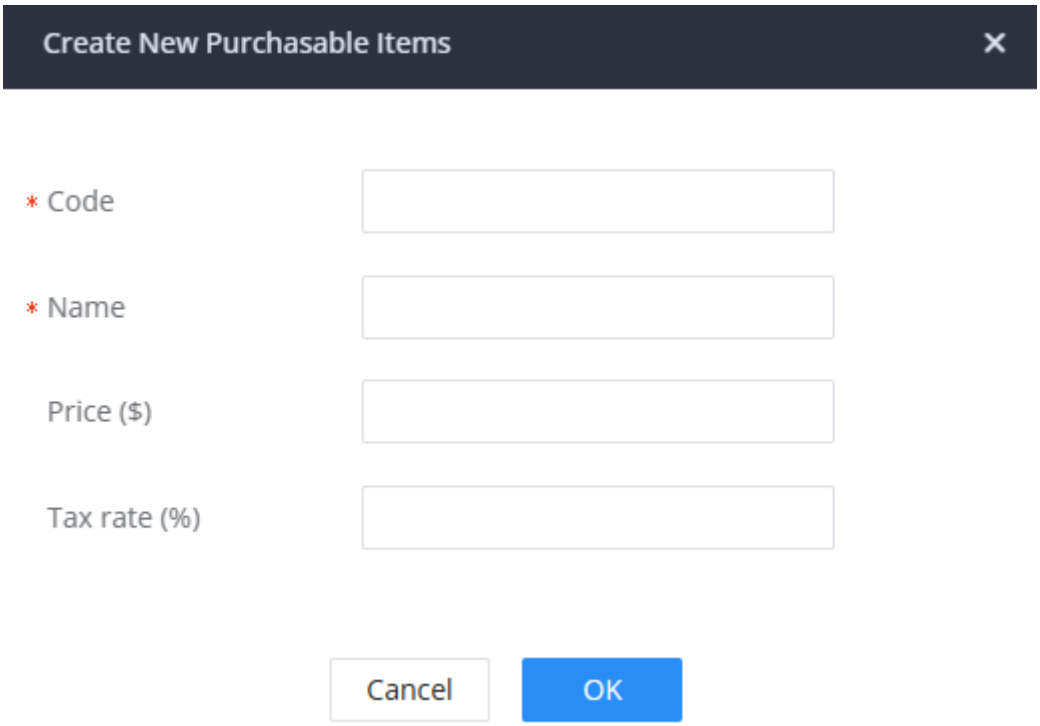


Mini Bar

Enable Mini Bar	Enables or disables the Mini Bar feature. When enabled, feature codes can be dialed to increase or decrease the usage of Mini Bar items.
Increase Mini Bar Usage Code	Specifies the feature code used to access Mini Bar billing. The housekeeper dials this code followed by the housekeeper code, password, and item code to add the item to the guest's bill.
Decrease Mini Bar Usage Code	Specifies the feature code used to access Mini Bar billing. The housekeeper dials this code followed by the housekeeper code, password, and item code to remove the item from the guest's bill.
Global Tax Rate (%)	Specifies the global tax rate applied to Mini Bar items. If no individual tax rate is configured for an item, the global Mini Bar tax rate is applied.
Prompt	Specifies the tone played when a housekeeper dials the Mini Bar feature code. The tone indicates the corresponding item code. Press Add Audio and Upload Audio File to upload an audio file, or create an AI Generated Speech prompt by configuring the following settings and pressing AI Generate: Name: Enter the name of the prompt. Supports 2-64 characters, including letters and special characters. Text: Enter the text to be converted to speech. Supports 1-5000 characters and must match the selected language. Language: Select the language of the prompt. Voice: Select he voice used for the prompt. Speech Speed and Volume can be adjusted.
Skip Housekeeper and Password Authentication	Disables housekeeper and password authentication when enabled. The housekeeper code is 0000 by default when this option is enabled.
Enable Multi-Item Billing	Enables or disables multiple Mini Bar items to be entered during a single call. Separate each item code with an asterisk (*).

Add Purchasable Items

The administrator can add new purchasable items to the Mini bar by clicking button, the following window will appear:

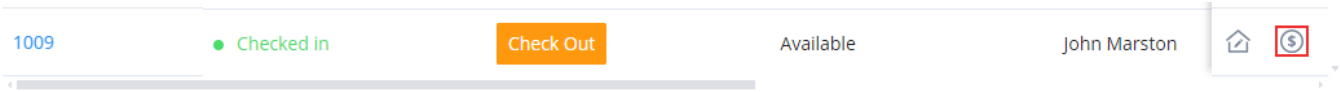


Create New Purchasable Item

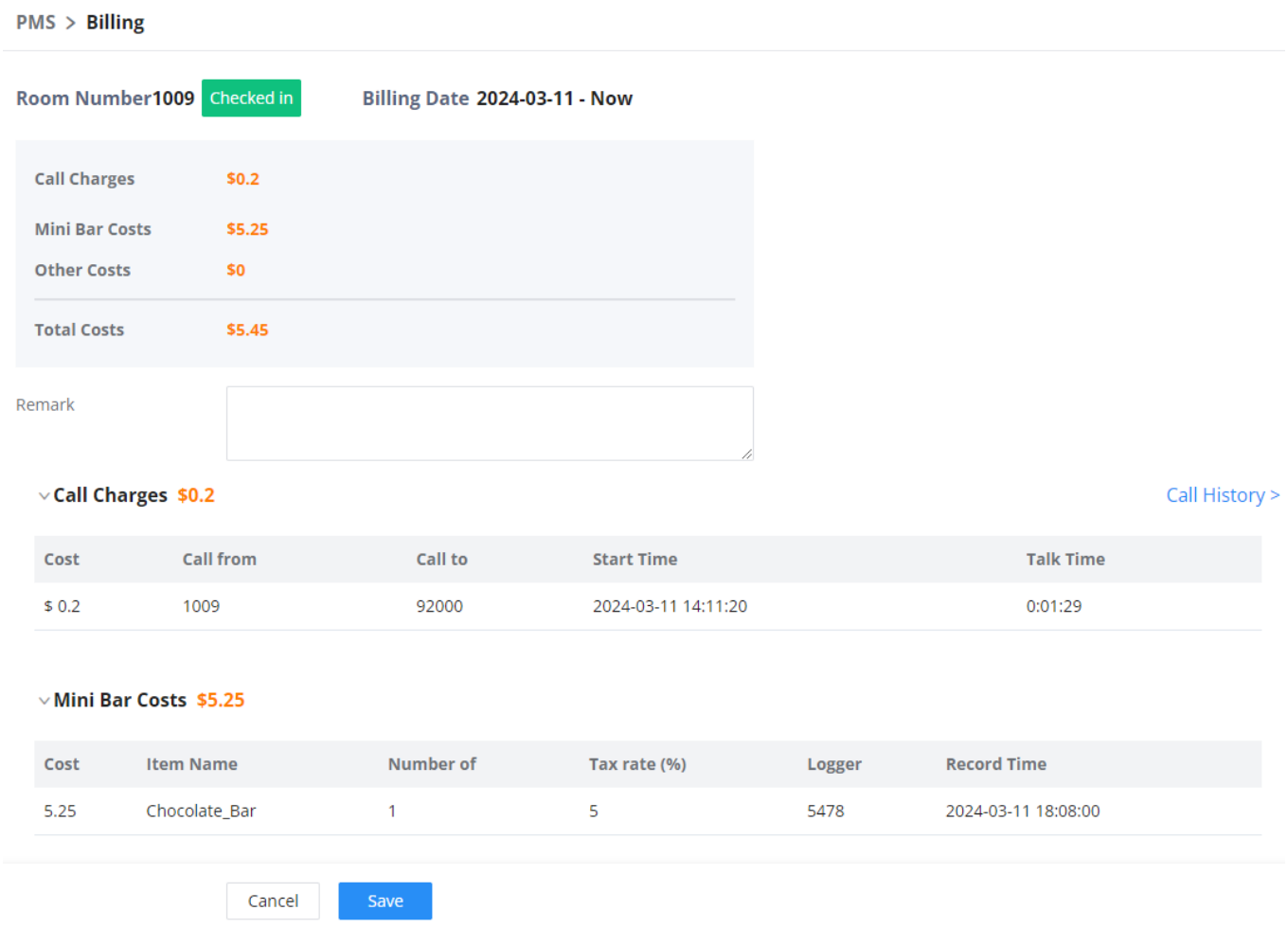
Code	Specifies the item code. Supports 2-18 digits only.
Name	Specifies the item name. Supports up to 64 characters.
Price ($)	Specifies the item price. Supports up to 12 characters.
Tax rate (%)	Specifies the tax rate for the item. Supports percentage values from 0 to 100 with "." as the decimal separator. If left blank, the Global Tax Rate (%) value is applied.

In the example given above, the housekeeper should dial the code 01 to bill an item or 00 to remove an item from the bill. Once that number is dialed, the housekeeper will be prompted to enter his/her code along with the password, then the housekeeper can enter the code of the item to either add or retract from the bill.

When an item is added to the bill, the user can view it by navigating to **Integration > Room Status**, and then clicking on



Check Billing Information

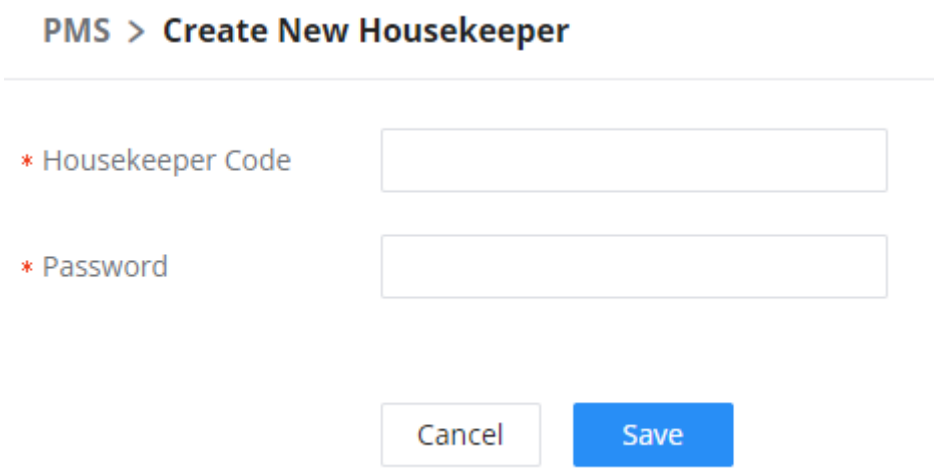


PMS - Billing

Housekeeper

The cleaning services in hotels and properties can be managed using the Housekeeper service configuration option, this can be done by assigning a specific code and password to each available housekeeper, these credentials will be used by the housekeepers to change the status of the room to "Cleaning" or "Cleaned" by entering the specific PMS feature code followed by their credentials on the IP phone related to the room number, this will help the property manager to control the current state of the rooms, the specific person that was responsible for the cleaning task and also to enhance security protocols.

In order to create a new housekeeper, click on under IPPBX WebGUI→**Integrations→PMS→ Housekeeper**.



Create New Housekeeper

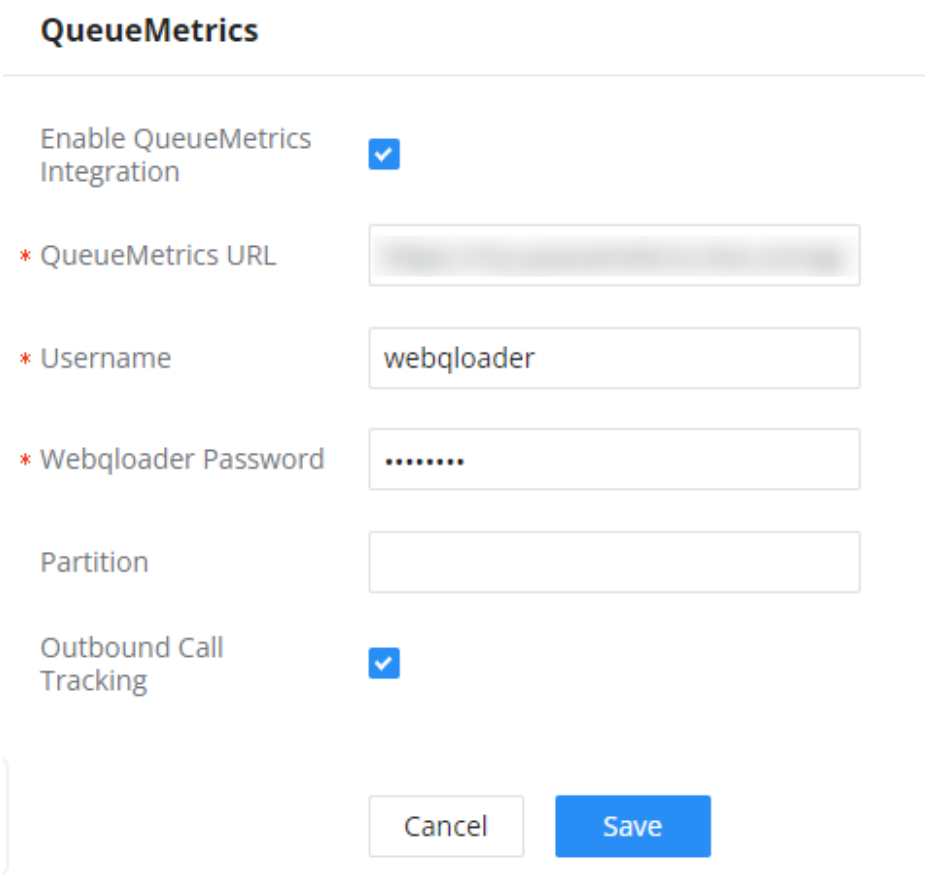
Housekeeper Code	Enter the Code to use when the houskeeper wants to change the status of the room.
Password	Enter the password associated with the housekeeper.

Create New Housekeeper

Please note that you can dial the "**Update PMS Room Status**" feature code, the "**Housekeeper**" feature code, and the "**Room Status**" feature code all at once to change the room status.

QueueMetrics

The QueueMetrics docking tool provides an interface for IPPBX system and QM docking. Pass the IPPBX call queue report to QueueMetrics in a richer form. QueueMetrics is a call center control platform that supports login and logout of frequently used agents in the call center, provides call reports, real-time queue monitoring and other functions.



QueueMetrics

Parameter	Description?
Enable QueueMetrics Integration	Tick this box to enable QueueMetrics integration module.
QueueMetrics URL	Enter the URL of the QueueMetrics on-premise server you have installed. (i.e. http://xxx.xxx.xxx.xxx:8080/queuemetrics.).
Username	Please enter the username used to interface with QueueMetrics. This is typically the QueueMetrics webqloader user. Please confirm that the user is enabled to avoid connection failure.
Webqloader Password	Please enter the webqloader password.
Partition	Enter the data storage partition identifier
Outbound Call Tracking	If enabled, QueueMetrics will track the outgoing calls of all extensions. Note: Outbound Call Tracking is available only on the PBX.

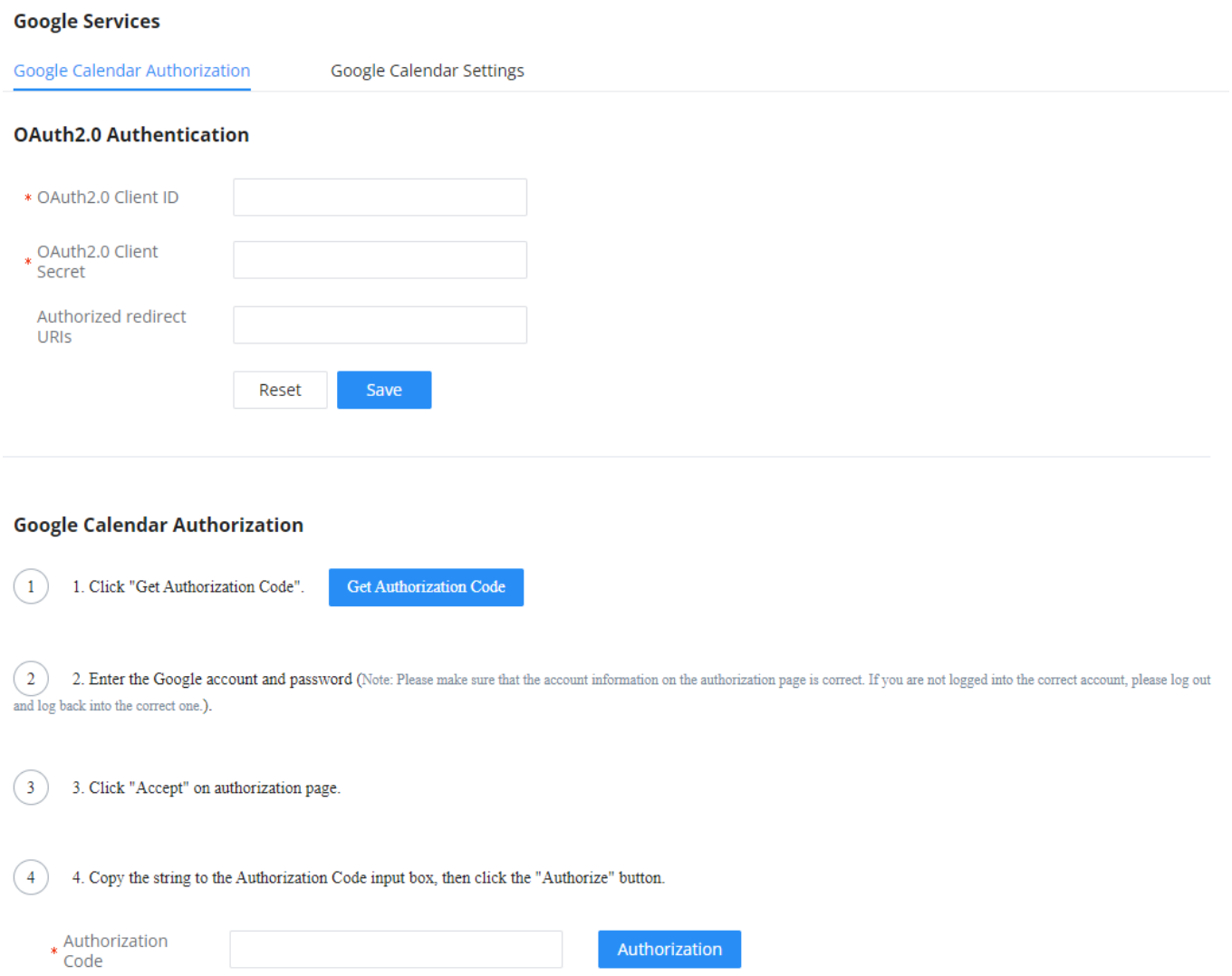
For more information, please refer to the dedicated how-to guide:
https://documentation.grandstream.com/knowledge-base/queuemetrics-integration/

Google Services

Google Services integration allows integrating the PBX with Google Calendar to automatically synchronize the created Multimedia and Onsite Meetings schedule with Google Calendar of the host and the participants. In order to use this integration the user needs to enable API on Google Cloud Console and obtain the Client ID and the Client Secret, then enter the authorized redirect URIs.

Google Calendar Authorization

In Google Calendar Authorization configuration page, please enter the generated OAuth2.0 Client ID, OAuth2.0 Client Secret, and Authorized redirect URIs to integrate the PBX with Google Calendar.



Google Calendar Authorization

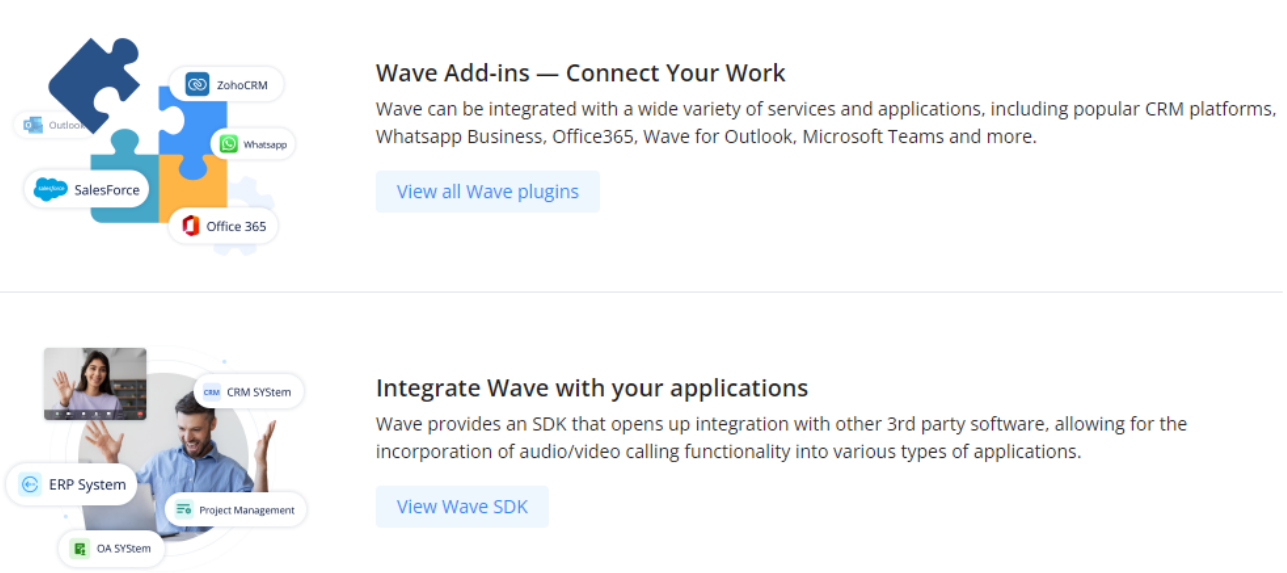
Google Calendar Settings

On Google Calendar Settings configuration page the user can enable "Google Calendar Auto Refresh Synchronization" to synchronize the meetings created automatically with Google Calendar. The user can also create a calendar label to mark different events on the calendar with customized labels.

To learn more about labels, please refer to the following link:
https://support.google.com/calendar/answer/12377581

For more information, please refer to the dedicated how-to guide in the following link:
https://documentation.grandstream.com/knowledge-base/google-calendar-api-integration/

Wave Integration



Wave Integration

CHANGELOG

Firmware version 1.0.7.48 (PBX version 1.0.27.90)

- No major changes.

Firmware version 1.0.7.46 (PBX version 1.0.27.90)

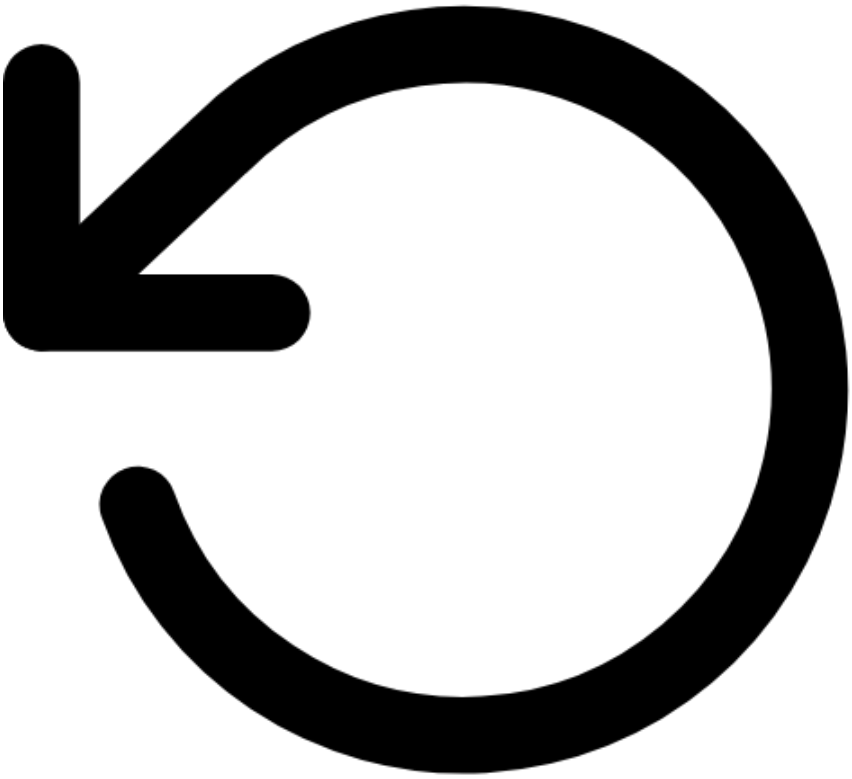
- No major changes.

Firmware version 1.0.7.44 (PBX version 1.0.27.89)

- Added support for media resource clustering. [Cluster]
- Added 1+N Media Cluster Connection Status Alarm [1+N Media Cluster Connection Status]

PBX Version 1.0.27.70

- This the initial release



Return to the main page