Brekeke PBX

Version 3

Administrator's Guide

(Basic)

Brekeke Software, Inc.

Version

Brekeke PBX version 3, Administrator's Guide (Basic)

1. Introduction

This document explains the basic configuration of Brekeke PBX. For advanced configuration or product architecture information, please refer to the Brekeke PBX Administrator's Guide (Advanced).

Brekeke PBX comes in two editions:

- **Brekeke PBX** – IP-PBX software for single-tenant use (i.e., office PBX system)
- **Brekeke PBX MT Edition** – IP-PBX for multi-tenant use (i.e., hosted PBX service)

2. Installation

2.1. Installation Options

	Installer (exe format)	**pbx.war (zip format)**
OS	Windows OS	Linux OS, Windows OS
Install	New installation only	New installation Update installation
Instructions	Section 2.4 (Windows)	Section 2.5 (Linux) Section 2.6 (Update)

2.2. System Requirements

OS	• Microsoft Windows 2016 or later • Linux
Memory	2 GB minimum

2.3. Software Requirements

Java	Requirements about editions and versions can be found at our wiki: https://docs.brekeke.com/pbx/system-requirement ✓ *Brekeke products are confirmed to run on Java provided by Oracle.* ✓ *Tomcat installation is not required when Brekeke PBX is installed with the installer.*
Apache Tomcat	

2.3.1. Installing Java SE

Instructions for installing Java can be found at our wiki:

https://docs.brekeke.com/sip/installing-java-se

2.3.2. Installing Apache Tomcat (required for Linux or manual install only)

Instructions for installing Java can be found at our wiki:

https://docs.brekeke.com/sip/installing-apache-tomcat

2.4. Installing on Windows OS with the Installer

1) Download Brekeke PBX "Installer Executable (exe)" for Windows OS.
2) Install the software following the installer's instructions.
3) Open http://localhost:18080/pbx/ (default) in a web browser.

✓ *If you select a customized port number during the installation, you need to set the port number in the URL.*

4) Activate license following the instructions found in Section 3 of this document.
5) At the login page, enter the user ID and password.

✓ *Default administrator login: user ID = "sa", password = "sa".*

2.5. Installing on Linux OS

1) Download Brekeke PBX "Manual Install (zip)" for Linux OS.
2) Copy the pbx.war file directly into the "webapps" directory, which is under the Tomcat installation directory.
3) Start Tomcat.

✓ *To confirm if Tomcat is running, open http://localhost:8080 in a web browser. (If you select a customized port number during the installation, you need to set the port number in the URL.) When you see the Apache Jakarta Project page, Tomcat has been started successfully.*

4) Open http://localhost:8080/pbx/ in a web browser.

✓ If you *select* a customized port number during the installation, you need to set the port number in the URL.

5) Activate license following the instructions found in Section 3 of this document.
6) At the login page, enter the administrator user ID and password.

✓ *Default administrator login: user ID = "sa", password = "sa".*

2.6. Minor version updates

1) Download Brekeke PBX "Updates & Upgrades (zip)."
2) Shut down Brekeke PBX from Brekeke PBX Admintool > [PBX] > [Start / Shutdown].
3) Open [Maintenance] > [Update Software].
4) Click [browse] to select the "pbx.war" file that you downloaded in Step 1.
5) Click [upload] to upload the new file.

6) Restart your computer to complete updating Brekeke PBX.

3. Activation

3.1. Type of Activations

Brekeke offers two types of activation methods for our products: online and offline activation.

3.1.1. Online activation

- Requires the Brekeke software to be connected to the Internet at all times.
- Allows unlimited instant license transfers.

3.1.2. Offline activation

- Able to activate the software without internet access.
- Must contact Brekeke to transfer license. The maximum number of transfers per license is three and is subject to our License Transfer Policy.
  http://www.brekeke.com/purchasing/license-transfer-policy.php

3.2. Initial Activation

Instructions for how to activate Brekeke software can be found at our wiki:
https://docs.brekeke.com/lic/activating-a-license-for-the-first-time

3.3. Reactivating License

You will need to reactivate license when:

- Adding users
- Adding options
- Adding codec license
- Upgrading editions
- Activation failure (i.e., lost online access)
- Transfer license
- Major version upgrade

Instructions for how to reactivate Brekeke software can be found at our wiki:
https://docs.brekeke.com/lic/reactivate-license-to-update-license-changes-add-users-upgrade-editions-and-so-on

4. System and User Management

4.1. Administrators

There are three types of administrators for Brekeke PBX:

- System Administrator – The top-level, fully authorized administrator.
- Account Administrator – Administrator for single-tenant PBX. Privileges of the Account Administrator are assigned by any administrator.
- Tenant Administrator (available only for MT Edition) – Administrator for MT Edition. This type of administrator has privileges only for the tenant they administer. Privileges are assigned by any administrator.

✓ *For Brekeke PBX MT Edition, default system administrator "sa" is the only administrator for the whole system. The "sa" administrator needs to create a tenant administrator whenever a tenant is created.*

4.2. Users

4.2.1. Creating Users

1) Log in as system administrator "sa".
   Go to [PBX] > [Extensions] > [Users] page.
   For Brekeke PBX MT Edition, go to [PBX] > [Tenants]. Click [New Tenant] to create a new tenant. When the tenant page opens, go to [Extensions] > [Users] page.

✓ *Super admin "sa" cannot be deleted.*

2) Click [Create a new user]. At the user [Account] page, set up the user account information.
   Example:
   [Extension] 300
   [Login Password] your_password (password used to log in to Brekeke PBX Admintool).
   [User Type] (select Admin or User)

✓ *There is no default user login password. If no login password is set at the time a user is created, that user cannot log in to the account from Brekeke PBX Admintool.*

4.2.2. User Access Settings

The System Administrator (sa) can limit access to various Brekeke PBX menu settings in the [PBX] > [Options] > [User Access Settings] section. This selection will be applied to all administrators and users in the system. Note that the items in the [Admin Menu] sections are only available to administrators.

4.2.3. Default Values of User Extensions

The following table outlines the default values included for Brekeke PBX user extensions. If you change these settings, the functionality of the product may differ from the examples shown in this manual.

To change user settings, click a user’s extension number from the [PBX] > [Extensions] > [Users] list, or select a user from the [User] drop-down menu on the left-hand panel.
In Brekeke PBX MT Edition, go to [PBX] > [Tenants], click on the user’s tenant name and go to the [Extensions] > [Users] page.

<table>
<tr><th>Page name</th><th colspan="2">Setting item</th><th>Details of default values</th></tr>
<tr><td rowspan="3">Settings</td><td colspan="2">Call pickup group</td><td>Same group as the administrator</td></tr>
<tr><td colspan="2">Greeting message</td><td>Default system greeting</td></tr>
<tr><td colspan="2">Email notification</td><td>off</td></tr>
<tr><td rowspan="2">Phones</td><td colspan="2">Phone 1 > [Phone ID]</td><td>- Same as user extension
- <tenant>_<extension> (MT Edition)</td></tr>
<tr><td colspan="2">Type</td><td>Type 1</td></tr>
<tr><td rowspan="5">Inbound</td><td colspan="2">Plan</td><td>Plan 1 (Active)</td></tr>
<tr><td>Forward To ></td><td>[Phones]</td><td>All phones are selected as forwarding destinations. When an incoming call to this user is received, Brekeke PBX will forward the call to all phones.</td></tr>
<tr><td>Forward To ></td><td>[Ringer Time (sec)]</td><td>90 seconds</td></tr>
<tr><td>Forward To ></td><td>[Forwarding destinations (Busy)]</td><td>Forwards to user’s voicemail</td></tr>
<tr><td>Forward To ></td><td>[Forwarding destinations (No answer)]</td><td>Forwards to user’s voicemail</td></tr>
<tr><td>Account</td><td colspan="2">Language</td><td>Same language as the administrator</td></tr>
</table>

		who created this user

4.2.4. Assigning Phones to User Extensions

Up to four different phone IDs can be assigned to each Brekeke PBX user extension. The default phone ID is assigned in the [Phone 1] > [Phone ID] field and is the same as the Brekeke PBX user extension from when this user was created. For Brekeke PBX MT Edition, it is in the format of tenant name and user extension (as <tenant_name>_<extension>).

The phone ID field can be a SIP UA's user ID, which is registered at the bundled SIP server of Brekeke PBX, or a PSTN (analog or mobile) number that belongs to the user of this Brekeke user extension. Any devices with a phone ID number assigned to the user extension can use features of Brekeke PBX, such as call conference and call recording.

For Brekeke PBX MT Edition, we recommend using the format <tenant_name>_<extension> as the SIP UA's ID (phone number) used to register in the bundled SIP server of Brekeke PBX. This will make phone ID management much simpler and will also help to avoid ID conflicts among different tenants within the same system.

4.2.5. Phone Type

System administrators can define customized phone types at [PBX] > [Options] > [Phone Type].

In v3.4.X or later, Brekeke PBX supports WebRTC. If WebRTC clients connect to Brekeke PBX, system administrators need to make WebRTC available at "Phone Type." You can set it as follows:

1) Select phone type at [PBX] > [Options] > [Phone Type], then set WebRTC to "yes."
2) Enable changed phone types as follows:
   v3.3 or earlier -> Restart PBX
   v3.4 or later -> Push "Apply" button on [Phone Type] page (no need to restart PBX).

You also need WebSocket handling available to use WebRTC connection.
[SIP SERVER] > [Configuration] > [SIP] > [WS (WebSocket)] > [WS-handling] on

4.3. Tenants (MT Edition)

For Brekeke PBX MT Edition, the system administrator needs to create a tenant account for each tenant.

1) From [PBX] > [Tenants], display all current tenants.
2) In this example we will create a tenant named "test."
   Click on [New Tenant] to input "test" as the tenant name.
3) Once the tenant is created, go to the tenant setup menu by clicking on the tenant's name.
4) Click the [Options] menu under this tenant to set the tenant users and session limits from the [Capacity] section, then enable the required features for this tenant from the [Features] section.

✓ *Lower-case letters, digits, dots and underscores are valid characters for tenant names.*

5. PBX Features

5.1. Call Forwarding

With Call Forwarding, you can forward your calls to another phone number, such as an operator or your mobile phone. You can set up Call Forwarding to automatically forward all calls or just certain ones depending on how they were missed (busy, after ringing, etc.).

These settings are configured at [User] > [Inbound] > [Call Forwarding].

5.1.1. Forward Calls to Multiple Phones

Example:

In this example, calls to extension 301 will be also forwarded to extension 300:

1) In user extension 301's [Other Forwarding Destinations*] field, enter the user extension number to which you want to forward the call. In this case, enter user extension "300."
2) Make a call to extension 301. Both phones assigned under user extension 300 and 301 will ring.

5.1.2. Setting Up No Answer Call Forwarding

You can set a number to forward calls that are missed after the Ringer Time period has been reached. ("Ringer Time" is set under [Call Forwarding] > [Ringer Time (sec)].)

In this example, missed calls to extension 301 will be forwarded to extension 300.

Example:

1) In extension 301's [Forwarding destinations (No answer)] field, enter user extension "300."
2) Incoming calls will be forwarded to user extension 300 if extension 301 does not answer

before ringer timeout.

5.1.3. Setting Up Busy Call Forwarding

You can set a number to forward calls that are missed when the line is busy.

In this example, missed calls to extension 301 will be forwarded to extension 300.

1) In user extension 301's [Forwarding destinations (Busy)] field, enter user extension "300."
2) Calls will be forwarded to 300 if user 301's phones return a 486 Busy response or another error response.

5.1.4. Scheduled Call Forwarding

This feature is useful for creating scheduled rules to forward calls, such as calls received outside of business hours.

These settings are configured at [User] > [Inbound].

In this example, incoming calls during business hours to extension 5555 are directed to extension 1002 (Auto Attendant). After business hours, callers are scheduled to forward to extension 1003 and the caller will hear, "*To speak with a live operator, please call during regular business hours. Our regular business hours are Monday through Friday, 9 am to 6 pm*."

Example:

1) Create an IVR extension 1003 (Type: Auto Attendant) and upload the customized after-hour greeting from [Greeting messages].
2) From [Extensions] > [Schedule], click [Create a new schedule].
3) Enter below and save:
   [Extension] 5555
   [Default Forwarding Schedule] > [Call Forwarding] > [Destination] 1003
4) Click [Add Forwarding Schedule] on the right side of the screen on the [Schedule] page.
   Set [Forwarding Schedule 1] as:
   [Forward To] > [Call Forwarding] > [Forwarding Destination] field to 1002.
   Click [Conditions] tab and set the [Date/Time] section as:
   [Term] set schedule starting and ending Date and Year.
   [Days] Check 1^{st} - 5^{th} and Monday through Friday.
   [Time] select and check 9:00 - 18:00.

Save

5.2. Ring Group

In this example, a Ring Group (3000) is created for all user extensions 301-305. When calls are received at group extension 3000, all specified group extensions (301-305) will ring simultaneously. If no one answers the call within the ringer time (90 seconds), the call is set to be forwarded to user extension 300.

Example:

1) From [PBX] > [Extensions] > [Groups], click [Create a new group].
2) Set as follows:
   [Extension] 3000
   [Type] Simultaneous Ring
   [Group Extensions*] 301,302,303,304,305
   [Ringer time (sec)] 90
   [Forwarding destination (No answer)] 300
   [Tag] via 3000
3) Save

✓ *From Brekeke PBX v3.4.x or later, the Tag feature is available. The value of the field is displayed in the group extension's devices while ringing. If you use the Tag feature, you can set it from [PBX] > [Option] > [Phone Type] > [Edit Phone Type] > [Tag in Display Name].*

5.3. Call Pickup Group

Call Pickup is a function that allows users to answer incoming calls from any extension by dialing a preset number. When an extension rings, dial *<user extension> to answer the call. For example, when user extension 300 rings, dialing *300 will enable you to answer the call from any other extension within the system (or within the tenant for MT Edition).

✓ *If you are using a SIP phone that has a “Call Pickup” button, consult the manufacturer or your SIP phone’s manual on how to use the Call Pickup feature.*

Here are some other ways of using the Call Pickup feature:

Example:

Extension: 3000

Group Extensions*	301,302,303,304,305

♦ **Answer Calls That Are Directed to a Ring Group**

Calls received at a Ring Group extension can be answered from any extension by dialing *<ring group extension>. In the above example, by dialing *3000, you can pick up a call directed to Ring Group 3000.

- **Create a Call Pickup Group**

  By using the Ring Group setting, you can create a Call Pickup Group. In the above example, dialing *3000 enables a user to pick up calls for the Ring Group, or any calls directed to an extension in the Ring Group. In other words, by dialing *<ring group extension>, you can pick up a call that came in for a single extension (e.g., 303) within the Ring Group.

5.4. Call Hunting

With Call Hunting, you can forward calls to multiple lines or phone numbers until it finds one that answers.

These settings are configured at [PBX] > [Extensions] > [Groups].

Example:

✓ Click [Create a new group].

✓ Enter/Select the following values:

[Extension] 3001
[Type] Call Hunting
[Mode] Round-robin (or Top-down)
[Hunt group extensions*] 301,302,303,304,305
[Ringer time (sec)*] 5
[Waiting time in the queue (sec)] 120
[Max number of calls in the queue] 10
[Call interval (msec)] 3000
[Single attempt] no
[Forwarding destination (No Answer)] 300

✓ Save

In this example, the call will ring the extensions in the [Hunt group extensions*] field one-by-one with a five-second interval (Ringer time) between calling each user extension. If all of the extensions (301-305) in the group are busy or do not answer, the call will be queued. If any member becomes available within 120 seconds (the interval set in [Waiting time in the queue (sec)]), the call will be directed to the first available extension in the group. If the call is not answered within the ringer time length set for the extension, the call will hold for the time set at [Call interval (msec)](3000ms) before trying the next available extension in the group. If the specified interval set at [Waiting time in the queue (sec)] is reached, the call will be forwarded to the destination set in [Forwarding destination (No answer)]. In this example, the call will be forwarded to user 300.

✓ *The hold music can be customized in the [Sound files] > [Music on hold] field.*

5.5. Auto Attendant

These settings are configured at [PBX] > [Extensions] > [IVR].

Example:

1) Click [Create a new IVR].
2) Enter/Select the following values:
   [Extension] 3002
   [Type] Auto Attendant
   [Max input digits] 3
   [Max retry count] 5
   [Ring timeout (sec)] 10
   [Default operator] 300
   [DTMF timeout (sec)] 20
   [Transfer to unregistered users] disable
3) Save

In this example, the incoming call to Auto Attendant 3002 will hear the default greeting voice prompt to input an extension.

If there is no entry made for 20 seconds ([DTMF timeout (sec)]), the caller will be reminded to enter a selection. After five tries with no input ([Max retry count]), the call will be forwarded to the destination set at the [Default operator] field.

If the caller enters an extension number, the call will be transferred to the extension that matches the first three digits ([Max input digits]) of the entry. If the extension does not answer the call within ten seconds ([Ring timeout (sec)]), the caller will be asked to enter a different extension number.

When [Transfer to unregistered users] is enabled, the caller can apply Brekeke PBX features from Auto Attendant, such as accessing a voicemail box, call pickup and so on. When [Transfer to unregistered users] is enabled, a proper value must be set in the [IVR] > [Max input digits] field in order to allow a caller's input.

- ✓ *Optionally, a customized greeting for Auto Attendant can be uploaded from the [Sound files] section.*
- ✓ *In Brekeke PBX v3.4.5.1 or later, two separate timeout values for [DTMF timeout (sec)] can be set for "before initial entry" and for "before completion entries,"* ***separated by commas*** *as below:*
  [DTMF timeout (sec)] 20,30

5.6. Switch Plan

Switch Plan or Timer can be used to change a user's forwarding destination temporarily and quickly by making a different forwarding schedule plan active.

In this example, incoming calls during business hours go directly to user 300's assigned phone(s). However, during user 300's lunch break, you can send incoming calls directly to voicemail. User 300 can enter DND (Do Not Disturb) mode by switching between pre-defined inbound plans to route incoming calls to his/her voicemail temporarily during lunchtime.

The following samples show two different ways of implementation. More samples about changing active plan with switch plan are available at Brekeke Wiki:
https://docs.brekeke.com/pbx/switching-plan

5.6.1. Sample with Switch Plan IVR Extension

Step 1: Setting Up a Switch Plan Extension

1) From [Extensions] > [IVR], click the [Create a new IVR] button.
2) Set as follows:
   [Extension] 3004
   [Type] Switch Plan
   [Plan number] 2

[on/off] yes

3) Save the settings.

Step 2: Setting Up Plans in User Extension

1) From the user extension 300 > [Inbound] page, click the [Add new plan] button.
   "Plan 2" will show up in the plan dropdown list window.
2) Uncheck all phones listed in Plan 2 and set [Other Forwarding Destinations*] as "vm300."
3) Save the settings.

Step 3: Enter "Do Not Disturb" Mode

1) User 300 dials the Switch Plan extension 3004 from his/her phone to enter DND mode.
   At user 300 > [Inbound] page, Plan 2 will be set as active.
2) Any calls to user 300 at this time will be directed to user 300's voicemail inbox.

Step 4: Remove "Do Not Disturb" Mode

1) User 300 dials Switch Plan extension 3004 from his/her phone again to remove DND mode.
   At the user 300 > [Inbound] page, Plan 1 will be set as active again.
2) After exiting DND mode, any calls to user 300 will be directed to his/her assigned phone(s).

5.6.2. Sample with Timer

Step 1: Setting Up Plans in User Extensions

1) At the user extension 300 > [Inbound] page, click the [Add new plan] button.
   "Plan 2" will show up in the plan's dropdown list window.
2) Uncheck all phones listed in Plan 2 and set [Other Forwarding Destinations*] as "vm300."
3) Save the settings.

Step 2: Setting Up Timer

1) At the user extension 300 > [Inbound] page, click [Timer 1].
2) Set Timer 1 as below:
   [Term] Set Schedule starting and ending Date and Year.
   [Days] Check 1^{st} - 5^{th} and Monday through Friday.
   [Time] Select and check:
   - 12:00 Plan 2
   - 13:00 Plan 1
3) Save the settings.

Step 3: Enter "Do Not Disturb" Mode

Every workday at 12 pm, user 300 will enter DND mode.

According to the user 300 > [Inbound] page, Plan 2 will automatically be set as active. Any calls to user 300 between 12 pm and 1 pm will be directed to user 300's voicemail inbox.

Step 4: Remove "Do Not Disturb" Mode

Every workday at 1 pm, Plan 1 on the user 300 > [Inbound] page will be set as active again. After exiting DND mode, any calls to user 300 will be directed to his/her assigned phone(s).

5.7. Conference Call

5.7.1. Creating a Conference Room

The first step to using the Conference Call feature is to set up a Conference Room. In the following example, extension 2000 is set up as the conference number:

✓ *If you are using a SIP phone that has a "Conference Call" button, please consult the manufacturer of your SIP phone for information on how to set up the Conference Call feature.*

1) From [Extensions] > [Conference], click the [Create a new conference] button.
2) Set as follows:
   [Extension] 2000
   Leave all other settings as default.
3) Save the settings.

With the above settings, any user can enter the conference room by dialing 2000.

5.7.2. Limiting Members Who Can Enter the Conference Room

You can limit the members who can join the conference by specifying individual members (for example, "301, 102, 103") at the [Applies to (Caller numbers)*] field. With this setting, only users 301, 102 and 103 will be allowed to join the conference. No other users can join this conference room.

5.7.3. Simultaneous Calls to All of the Conference Members

A conference member can convene all members of the conference room at once. For example, set 301, 102 and 103 at [Auto Invite attendees*]. By dialing 2000, all conference members (301, 102 and 103) will be invited simultaneously.

5.7.4. Starting a Conference Call (Alternate Methods)

The Brekeke PBX Users Guide describes additional methods for starting a conference call.

5.8. Callback

The Callback feature is a method of making low-cost international calls via a third country. Brekeke PBX will ring a caller back when the caller dials the number of a Callback extension.

In the following example, extension 3000 is set up as a Callback extension:

1) From [Extensions] > [Callback], click the [Create a new callback] button.
2) Set as follows:
   [Extension] 3000
   [Ringer time (sec)] 90
   [Forwarding destination (No answer)] 300
   [Callback callee] 3002
3) Save the settings.

When a caller dials a Callback extension, the caller will hear a ring tone. If the caller hangs up before the ringer times out, Brekeke PBX will send an INVITE to the caller, who will then be connected to the number set in the [Callback callee] field. In this example, the caller will be connected to Auto Attendant 3002.

When a caller who dials a Callback extension does not hang up before ringer timeout, the call will be directed to the destination set in the [Forwarding destination (No answer)] field.

5.9. Confirm Call

1) Go to [PBX] > [Voice Prompts] and upload an audio file named "confirmcall," which will play a voice prompt to let the call recipient press a confirm key to establish the call.
2) Go to [PBX] > [ARS] and create a new ARS Route to use Confirm Call, or to add the Confirm Call setting to an existing ARS outbound route.
3) At "Patterns – OUT" in the ARS Route, set the voice prompt name uploaded in step 1 to [Confirm] field under [Deploy patterns]. You need to enclose the voice prompt name with curly brackets "{" and "}". In this example, the [Confirm] field is set as {confirmcall}.
4) Specify the confirm key in the [Key] field next to [Confirm] field. The default key is 5.
5) Make outbound calls by applying the ARS route with confirm call setting. When the call recipient answers the call, he/she will first hear the voice prompt "confirmcall." If the call recipient presses the confirm key (5) from his/her phone keypad before the voice prompt ends, the call will be established between the caller and callee. Otherwise, Brekeke PBX will disconnect the call.

- ✓ *If the [Confirm] field is set to {confirmcall}{name:&f1}, then the callee will hear the caller's name (if available) or the caller's phone number played after the voice prompt. In this case, you need to set [From] field in the Matching patterns, e.g., sip:(.*)@*

5.10. Paging

The phones that support paging will answer incoming calls automatically without taking the handset off-hook when these phones receive the paging information sent in the SIP header from Brekeke PBX.

A list of SIP phones that work with the Brekeke PBX paging function, as well as its sample configuration are available at the Brekeke Wiki:
https://docs.brekeke.com/interop/how-to-set-paging-function-on-the-phone-side

5.11. Busy Lamp Field (BLF)

When there is a call to a monitored phone, the corresponding key lamp on the monitoring phone will flash and the call can be picked up from the monitoring phone.

A list of SIP phones that work with these functions, as well as their sample configurations, are available at the Brekeke Wiki:
https://docs.brekeke.com/pbx/blf-sca-and-presence

5.12. Presence

Brekeke PBX can handle presence SUBSCRIBE requests from phones and return NOTIFY responses about the status of the monitored phones, such as "available" or "on the phone."

A list of SIP phones that work with these functions, as well as their sample configurations, are available at the Brekeke Wiki:
https://docs.brekeke.com/pbx/blf-sca-and-presence

5.13. Shared Call Appearance (SCA)

Users can monitor the external line status and select an available line to place an outbound call or to answer incoming calls.

A list of SIP phones that work with these functions, as well as their sample configurations are available at the Brekeke Wiki:
https://docs.brekeke.com/pbx/blf-sca-and-presence

5.14. Video

Video can be enabled from several locations in Brekeke PBX: Options, ARS Settings and User Settings. Video setting enabled in the phone type set under Users > [Phones] -> [Phone Type] has the highest priority. When the default setting is defined in [Phone Type] > [Video] field, the video setting under [ARS] route will be applied if the call can apply any route. When video setting in both User [Phone Type] and ARS route is set to the default, the [Video] setting at [Options] will be applied to the call.

The following steps show how to enable video for the phone assigned to a Brekeke PBX user extension.

1) From [Options] > [Phone Type] page, create a new phone type
2) At new phone type setting page, set [Video] field as on and set proper audio codecs in [Codec Priority] field.
3) Save the new phone type and restart Brekeke PBX from admintool.
4) From a PBX user extension > [Phones] page, select the phone type created above at the [Type] field under the phone which supports video for the call. And save the changes.
5) The call between PBX users with video phone type or from/to external user through ARS route with [Video] field set as "on" will have video feature.

5.15. Voicemail

5.15.1. Voicemail Settings

After creating user extensions, you can set up voicemail for each extension. If a call is missed (eitther did not answer or busy) after the specified length of time at [Ringer Time] under the Inbound setting, the call will be forwarded to voicemail by default. There is no default voicemail access PIN set at the time a user is created, so set the user’s voicemail box PIN from the [User] > [Settings] page.

By using preset prefixes, users can manage their voice messages using a dial keypad. The “vm” prefix is specified under the “mediaserver_prefix” route at the ARS settings.

- To go straight to voicemail, dial 07*<user extension>.
- To check voice messages from your phone (the phone assigned Phone ID), dial “8” to reach your voicemail inbox.
- To check your voice messages from another extension in the system, dial 08*<your user extension>.

5.15.2. Voicemail Notification by Email

Step 1: Setting Up an Email Sender

Brekeke PBX supports mail servers that provide “POP before SMTP” authentication, “SMTP” authentication or Encrypted Connection (SSL).

To set up an Email Sender:

1) Enter your mail server information and email user account information at [PBX] > [Options] > [Settings] > [Email settings].
2) Set [Encrypted connection (SSL)] on or off, depending on your mail server type.
3) Restart Brekeke PBX from Admintool to apply your changes.

Step 2: Setting Up an Email Recipient

1) Navigate to the [Voicemail settings] section on a user’s [Settings] page.
2) Set [Email address*] to the recipient’s email address(es).
3) Set [Email notification] to “on.”
4) Set [Attach WAV file to Email] to “on” or “off,” depending on whether you want to attach voice messages to email.

5.15.3. Message Waiting Indicator (MWI)

If your SIP UA supports MWI with a “SUBSCRIBE” message, it will be activated by default. If there is a special button on your SIP UA to retrieve messages, assign a number (default is “8”) to retrieve voicemail messages. Some types of SIP UAs can automatically call the SIP URI that is specified in the Brekeke PBX NOTIFY packet (for MWI) to retrieve the message without assigning the number manually.

For those SIP UAs that do not send “SUBSCRIBE” messages for MWI, you can create a new phone type and set the [MWI (NOTIFY without SUBSCRIBE)] field as “on” and select this phone type from the PBX user extension [Phones] page where these SIP UAs are assigned. Then Brekeke PBX will send a voicemail notification even if there is no “SUBSCRIBE” message for the MWI sent from these SIP UAs.

5.16. SRTP

From Brekeke PBX v3.2.x or later, Brekeke PBX supports SRTP.

SRTP can be enabled from several locations in Brekeke PBX: Options, ARS Settings and User Settings. SRTP setting enabled in the phone type set under Users > [Phones] -> [Phone Type] has the highest priority. When the default setting is defined in [Phone Type] > [SRTP] field, the SRTP setting under [ARS] route will be applied if the call can apply any route. When SRTP setting in both User [Phone Type] and ARS route is set to the default, the [SRTP] setting at [Options] will be applied to the call.

5.17. ARS

In Brekeke PBX v3.0.x, by clicking a route name on the [ARS] > [Running Status] page, the details of an active ARS route can be displayed. Also, from the [ARS] > [Settings] page, administrators can perform numerous actions on an ARS route, such as copy, delete and edit.

From Brekeke PBX v3.1.x, by clicking [View] in the ARS route list's [Status (Reg/Route)] column, the details of an active ARS route can be displayed. An ARS rule can also be edited by clicking the ARS route name. If the ARS rule is a template with the [Template] field checked, you can click [Edit Template] on the [Variable Setting] page to change the template setup and the variables' setups for each route under the template.

In Brekeke PBX MT Edition, the tenant name needs to be set for any ARS rule that will be used exclusively by a specific tenant. The tenant name is set in the ARS rule > [General] > [Tenant] field.

By assigning a tenant name in ARS rules for inbound calls, Brekeke PBX MT Edition will associate inbound calls with this tenant and will route inbound calls to the proper tenant extensions.

By assigning a tenant name in the ARS rules for outbound calls, Brekeke PBX MT Edition will look for the ARS route for this tenant, determine who the caller is and then route the call to the destination set for this tenant. If no tenant name is assigned in the ARS outbound route, the route will apply to calls from any tenant.

5.17.1. Route Settings / Route Template

- **General**

  Use the [General] section to enable or disable either an ARS template or any of the settings required by Brekeke PBX features, such as [Tenant] in MT Edition if you want to restrict route usage to a certain tenant, the [LineKey] and [Resource] fields for Shared Call Appearance, the [Group] field for ARS group, the [External] field for call logs database and for billing and so on.

- **Registration**

  Register your VoIP gateway port SIP number at Brekeke PBX bundled SIP Server. Brekeke PBX will accept calls from this gateway even if the gateway is not registered at Brekeke PBX bundled SIP Server.

  If a provider needs authentication information from a caller who makes outbound calls from UAs to an ITSP, Brekeke PBX will send the username and password that are set in this section to the provider in an IP address or in the domain set in the [Proxy Address] field.

- **Patterns**

  Define Patterns IN and OUT to receive and send calls from/to SIP devices and services.

- **Priority**

  It is useful to set priorities when there are multiple options for making calls, such as when you have multiple PSTN gateways for outbound calls or when you subscribe to multiple VoIP service providers. A lower number holds the higher priority.

- **Max Sessions**

  Set the [Max Sessions] field to define the maximum number of sessions that can be handled by each pattern, such as when there is a limited number of Gateway channels or subscribed lines for your SIP services. Setting this field to “-1” specifies an unlimited number of sessions.

  For ARS routes in the same group, there is only one session counter. The session counter for the group will increase by one when there is a call through any ARS route in this group, regardless of what is specified in Pattern IN or OUT. If this session counter is equal to any pattern [Max Sessions] value that is set in the same group’s ARS routes, the next matched session cannot apply to this pattern (IN or OUT) or to any ARS routes when all of the patterns' [Max Sessions] for the ARS routes in the same group are set to the same value.

5.17.2. Global Route Variables (Brekeke PBX v3.1.x or later)

Global variables can be accessed from any ARS route. The update of a global variable will affect the ARS routes in which the related variable name is set.

From [Field Settings], administrators can set a global variable's field type, name and description, and can also choose whether or not a global variable will be displayed on the ARS top page.

5.17.3. Variable Setting

Brekeke PBX v3.1.x or later

From the ARS template's [Variable Setting] page, administrators can create routes that use the same ARS patterns template and can also set the values of the variables to be used by each route.

- **Common Settings**

  Common variables can be accessed from any route created under the same ARS template. Updating a common variable will affect the ARS routes under the current ARS template. From [Field Settings], administrators can set a common variable's Field Type, Field Name, Input Rule and Descriptions, and can also choose whether a common variable will be displayed on the [Variable Settings] page.
  Configuring [Field Settings] on [Common Settings] is the same as on [Route Local Settings]. For more configuration details, please refer to "Route Local Settings" in the next section..

- **Route Local Settings**

  Create new routes and assign values to the route's local variables defined in the ARS template for each route, and also deploy multiple routes based on the same ARS template. From the [Field Settings] page, administrators can change the route's local variables'settings and also write JavaScript code to control the variable values set in each route.

  Sample script:

```
if( v1 == 3333 )
  v5= "192.168.200.20";
```

```
else
  v5= "192.168.200.30";
```

Set the above script in the [Route Local Settings] > [Field Settings] -> [Script] field. If a route has variable v1 with value 3333, this route variable v5 will be set as “192.168.200.20” automatically. The route variable v5 will be set as “192.168.200.30” when variable v1 is set to any value other than 3333.

The table below shows an example of other items on the [Field settings] page.

If you choose “Select” as the Field Type, you should set an input rule in “Input Rule.”

Field Type	Variable	Field Name	Initial Value	Input Rule	
Text	v1	Call Number		[0-9]	*1
Select	v2	ITSP Type	003	ITSP A:001;ITSP B:002;ITSP C:003;	*2

*1. When “Text” is selected as the Field Type, an Input Rule is not necessary. However, if you set an input rule using a regular expression, Brekeke PBX will check variables according to the rule on the [Variable Settings] page.

*2. When “Select” is selected as the Field Type, an Input Rule is needed. In the example, according to the input rule, “ITSP A,” ”ITSP B” and “ITSP C” are displayed as options for the select box on the [Variable Settings] page. “ITSP C” is displayed as the initial value of the select box.

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The variables defined in an ARS route can be edited from the [Edit Variables] page at the upper-right of the ARS route template.

Variables	Default value
v1	User ID/Number
v2	Password
v3 - v9	Customizable fields

5.17.4. ARS Default Plug-ins

Brekeke PBX offers the default plug-ins that use Notes. The default plug-ins are contains, lookup, and matches.

By using these plug-ins, you can search for a caller's number in a telephone directory (making notes under [Options] > [Notes] to save data list) and handle the call as your need, such as, decline the call, change the caller ID, or redirect the call.

For the details about the Brekeke PBX default plug-ins and how to configure ARS route to use the plug-ins, please refer to “Developer's Guide: ARS plug-in”.

5.18. PSTN Access Using a VoIP Gateway

Using a SIP-compliant VoIP Gateway, Brekeke PBX users can receive calls from a Public Switched Telephone Network (PSTN) and make calls to PSTN lines.

5.18.1. VoIP Gateway Setup

Set the following at your VoIP Gateway:

SIP proxy address	IP address of Brekeke PBX
Dialing number sent to Brekeke PBX	PSTN line number

5.18.2. ARS Route Setup

Set up “Patterns – IN” and “Patterns – OUT” in the default Gateway ARS Route to receive and make calls from/to the Gateway.

✓ *Default ARS rule “gw1” is the setup template for one-stage dialing.*

✓ *Default ARS rule “gw2” is the setup template for two-stage dialing.*

Use the following settings to register the gateway at the Brekeke PBX bundled SIP server. Many PSTN Gateways have a short interval between sessions, during which time the line is unavailable. You can modify the [Session interval (ms)] field setting to reflect this delay as needed.

✓ *In Brekeke PBX v3.1.x, check the [Template] field to enable the [Variable Setting] page.*

[Registration]

Register URI	sip:&v1@127.0.0.1	**Register expire (sec)**	3600
Proxy address	127.0.0.1	**Register update period (%)**	90
User		**Password**	

5.18.3. Receiving PSTN Calls

Create Gateway ARS Route “Patterns – IN” to receive calls from a gateway.

[Patterns – IN]

	Matching patterns	Deploy patterns
From		
To	sip:&v1@	&v3

5.18.4. Calling PSTN Numbers

One-Stage Dialing

If your VoIP Gateway supports One Stage Dialing, a Brekeke PBX user can make a PSTN direct call by setting an ARS Route as follows:

[Patterns – OUT]

	Matching patterns	Deploy patterns
From		
To	sip:([0-9]{7,25})@	sip:$1@gw_IPaddress

In this example, Regular Expressions were used to define the Matching and Deploy patterns. A Brekeke PBX user who dials a number that has between 7 and 25 digits will be considered as a PSTN call. Brekeke PBX will apply the above ARS Route and the call will be sent to the gateway.

If you use multiple VoIP Gateways for outbound calls, you can define more detailed dialing patterns in [Matching patterns] > [To], as well as change the [Priority] field as necessary to define the usage order of Gateways. Please note that lower numbers hold the higher priority. You can use the [Max Sessions] field to define the total number of sessions handled by each pattern.

Two-Stage Dialing

If your VoIP Gateway supports Two Stage Dialing, have the gateway's PSTN port register with Brekeke PBX bundled SIP server. Let's suppose the gateway's PSTN port has the SIP user name 111. To call a PSTN number, dial the gateway's registered PSTN port number (in this example, it's number 111) and then dial the destination PSTN number. Alternatively, you can configure the DTMF setting at the ARS Route OUT pattern so that the dialed numbers will be sent to the gateway as DTMF tones.

The OUT pattern to send a destination number by DTMF in Two-Stage Dialing is as shown below:

Patterns – OUT

Matching patterns		Deploy patterns	
From		**From**	
To	sip:111(.+)@	**To**	sip:111@gw_IPaddress
		DTMF	$1

✓ *If a delay is needed before sending a DTMF call, set the [DTMF] field as {file_name}$1. Default files are {120ms}, {240ms}, {500ms}, {1sec} and {2sec}. Customized files can be uploaded from [PBX] > [Voice prompts] and set [Language]: Common.*

5.18.5. Assign Values to Variables

Click the [Variables] link at the upper-right corner of the ARS Route template page. At the [Variable Setting] page, select [New Route] and assign values to the variables defined in the above Patterns – IN and Patterns – OUT.

Route Name	Gateway1 (Brekeke PBX v3.1.x or later)
Tenant	A tenant name (MT Edition)
v1	PSTN line number set in section “VoIP Gateway Setup”
v3	Specify a Brekeke PBX extension number, such as 300.

5.19. Connecting with Internet Telephony Service Providers (ITSPs)

5.19.1. Account Information for Third-Party SIP Server

Acquiring the information shown below is necessary to connect with a third-party SIP server.

Phone number	6504106636
SIP server IP address	sample_proxy.com
User ID	6504106636
Password	6636

✓ *Depending on the provider, there may be restrictions for connecting to services, such as available information or equipment used to connect. Please contact your VoIP service provider for more details.*

✓ *Please note that we do not guarantee connection with third-party products.*

5.19.2. Setting ARS for ITSP using multiple accounts

✓ *In Brekeke PBX v3.1.x, check the [Template] field to enable the [Variables] setting page.*

[Registration]

Field name	Sample settings	Explanation
Register URI	sip:&v1@sample_proxy.com	Enter SIP URI.
Proxy address	sample_proxy.com	Can be omitted when Proxy address is the same as the one in the [Register URI] field.
User	&v1	Set value at [Variables Settings] page.
Password (Brekeke PBX v3.0.x)	&v2	Set value at [Variables Settings] page. This field will be displayed in text format after saving.

[Patterns – IN]

In the Patterns – IN example below, Brekeke PBX user extension 300 (“&v3” value, set at the [Edit Variables] page) is set to ring when a call comes through a third-party SIP server. Leaving the “From” field blank carries over the Caller ID information.

	Matching patterns	Deploy patterns
To	sip:&v1@	&v3

✓ *Check [Apply to Request URI instead of To] when To header sent from ITSP is different from To defined in the ARS Route Patterns – IN.*

[Patterns – OUT]

Patterns – OUT defines patterns for converting the SIP URI to match your VoIP provider’s header format requirements. In the example below, dialed numbers with 7 to 25 digits will be directed through the VoIP service provider. To ensure that the recipient’s caller ID display will function, the “From header” will change according to the rules of the provider. Set [Priority] and [Max Sessions] as you need.

	Matching patterns	Deploy patterns
From		“&v1” <sip:&v1@sample_proxy.com>
To	sip:([0-9]{7,25})@	sip:$1@sample_proxy.com

✓ *Some VoIP service providers restrict the connection when the FROM or TO header information is different from their own header format.*

[Variables]

Click the [Variables] link at the upper-right corner of the ARS Route template page. On the [Variable Setting] page, select [New Route] and assign values to variables defined in the above Patterns – IN and Patterns – OUT.

Route Name	Route 6504106636 (Brekeke PBX v3.1.x or later)
Tenant	A tenant name (MT Edition)
Password / v2	6636
v1	6504106636
v3	Specify a Brekeke PBX extension number, such as 300.

✓ *For Brekeke PBX v3.0.x, set related variables from the [Edit Variables] page.*

5.20. ARS Outbound Route Failover

Utilizing the Automatic Route Selection (ARS) outbound route failover feature allows users to create redundant telecommunications systems. If an outbound route is not available or usable, Brekeke PBX will direct the session to an alternative route.

5.20.1. Usage Examples

- Brekeke PBX provides automatic failover to an alternative ITSP service in the event of a failure at your specified ITSP service.
- Brekeke PBX provides automatic failover to an analog telephone session via PSTN Gateway in the event of a failure at your specified ITSP service.
- Creating redundant analog telephone connections with multiple PSTN Gateways.

5.20.2. Settings Examples

The ITSP line is set for regular outbound sessions. When the ITSP line fails, the outbound sessions will be routed through the PSTN Gateway.

In the following example, two ARS routes are created: “ITSP_A” and “MyGateway.” The route with the highest priority, “ITSP_A,” will be used for outbound calls with a dialing numbers that have 7 to 25 digits. If there is no response within four seconds (Response timeout: 4000 ms) for INVITE messages or a “500-599” response is received, Brekeke PBX will continue searching for the next route that matches the outbound session request. In this case, the next-highest prioritized route, “MyGateway,” will be chosen as the alternative route for the session. Since the recovery time is set to one hour (3600000 ms) in “ITSP_A,” the matching sessions will be routed through route “MyGateway” for one hour after the failover. If “ITSP_A” is running again within an hour, the sessions will once again be routed through the highest priority route, “ITSP_A.”

Route name: ITSP_A

Patterns – OUT

OUT – 1		**Matching patterns**		**Deploy patterns**	
Priority	**1**	**From**		**From**	“xxx”<sip:xxx@itsp.com>
Max sessions	**4**	**To**	sip:([0-9]{7,25})@	**To**	sip:$1@itsp.com

Parameters			
Next route on failure	Yes	**Disable on registration failure**	Yes
Response timeout (ms)	4000	**Error codes**	500-599
Recovery time (ms)	3600000	**Disable on failure**	This route

- [Response Timeout (ms)] should be adjusted according to your environment. PSTN Gateways and SIP servers located in the local network may not require this long of a Response Timeout interval. For a route that requires an Internet connection or if some other kind of delay is expected, the Response Timeout intervals should be set to a longer time.

- [Disable on registration failure] is set to “yes” in Route “ITSP_A.” When registration is not working property at the “ITSP-A” route, it will be disabled and the “MyGateway” route will be used instead.

- [Disable on failure] is set as “This route,” which will disable the whole route. If there are other IN/OUT patterns defined in this route, they will be unusable when failover happens. If [Disable on failure] is set as “This pattern,” it will only disable the current pattern, so other patterns in this route will still be usable.

Route name: MyGateway

Patterns – OUT

OUT – 1		**Matching patterns**		**Deploy patterns**	
Priority	**100**	**From**		**From**	
Max sessions	**4**	**To**	sip:([0-9]{7,25})@	**To**	sip:$1@GW_IPaddress

5.21. ARS Route with Resource Setting (Brekeke PBX v3.1.x or later)

When multiple ARS routes for Shared Call Appearance (SCA) are set with the same name in the [Resource] field, Brekeke PBX SCA users will have more ARS route choices for making outbound calls than by using the ARS settings without this [Resource] setup.

Set up the ARS template and Brekeke PBX users as described below and assign phones with the Shared Call Appearance feature enabled to users.

Route Template:

[General]

Route name	outbound
Template	checked
Line key	checked
Resource	&v9

[Patterns – OUT]

OUT – 1		**Matching patterns**		**Deploy patterns**	
Priority	&v1	**From**		**From**	
Max sessions	&v2	**To**	sip:([0-9]{7,25})@	**To**	sip:$1@&v3

Variable Setting:

Route Name	v1	v2	v3	v9
1	90	1	192.168.200.10	gw
2	90	-1	192.168.200.20	itsp
3	100	1	192.168.200.30	gw

Brekeke PBX SCA Users

User: 300

[Settings] -> [Resource map]

line/1=gw/1

line/2=gw/2

User: 301

[Settings] -> [Resource map]

line/1=gw/1

line/2=itsp/1

When user 300 makes an outbound call from a phone with shared line 1, the call will apply route “outbound*1” with resource set as “gw” and the call will be sent to the destination IP 192.168.200.10.

While user 300 is talking, if user 301 also makes an outbound call from shared line 1, the call will apply route “outbound*3” with resource also set as “gw” and will be sent to the destination IP 192.168.200.30.

Because route “outbound*1” with resource “gw” has the [Max Sessions] variable v2 set to 1 and one session from user 300 is using this route, Brekeke PBX looks for the next available route with resource “gw” for the user 301 outbound call, which is route “outbound*3.”

If user 301 makes an outbound call from shared line 2, the call will apply route “outbound*2” with resource “itsp” and will be sent to IP 192.168.200.20.

5.22. ARS Route with Group Setting

Set up the ARS template and Brekeke PBX users as described below and then assign phones with the Shared Call Appearance feature enabled.

Route Template:

[General]

Route name	outbound
Template	checked

Group	&v8
Line key	checked
Resource	&v9

[Patterns – OUT]

OUT – 1		**Matching patterns**		**Deploy patterns**	
Priority	&v1	**From**		**From**	
Max sessions	&v2	**To**	sip:([0-9]{7,25})@	**To**	sip:$1@&v3

Variable Setting:

Route Name	v1	v2	v3	v8	v9
1	90	1	192.168.200.10	group1	gw
2	90	-1	192.168.200.20		itsp
3	100	1	192.168.200.30	group1	gw
4	110	1	192.168.200.40	group2	gw

Brekeke PBX SCA Users

User: 300

[Settings] -> [Resource map]

line/1=gw/1

line/2=gw/2

User: 301

[Settings] -> [Resource map]

line/1=gw/1

line/2=itsp/1

If user 300 makes an outbound call from shared line 1, the call will apply route “outbound*1” with resource set as “gw” and group as group1 and the call will be sent to the destination IP 192.168.200.10.

While user 300 is talking, if user 301 also makes an outbound call from shared line 1, the call will apply route “outbound*4” with the resource set as “gw” and the group as group2 and the call will be sent to the destination IP 192.168.200.40.

As explained in the above section about Max Sessions, routes in the same group use the same session counter. In this example, routes “outbound*1” and “outbound*3” with resource “gw” are both in “group1” and both routes have the [Max Sessions] variable v2 set to 1. When user 300 is making a call through route “outbound*1,” the number of sessions in group1 is 1, which will reach the [Max Sessions] in both routes “outbound*1” and “outbound*3.” Brekeke PBX will then look for the next available route with resource “gw” for user 301’s outbound call. This is route “outbound*4.”

The above example explains that the [Resource] setting is used for Brekeke PBX users to look for available routes to make outbound calls and that the [Group] setting is used to limit max sessions through the routes of both Patterns – IN and Patterns – OUT in the same group.

- ✓ *Multiple groups can be set in an ARS route [Group] field with group names separated by commas.*
- ✓ *When a session goes though a route set with multiple groups, the session counter of the related groups will increase by 1.*

5.23. DID

From the [DID] menu, administrators can access and modify the active routes setup.
For MT Edition, only the active DID route related to the tenant will appear in tenant administrator accounts. Tenant administrators can modify the route fields as necessary.

The following steps show how to enable DID access for a route and to set DID access privileges.

Step 1: Enable DID Menu

1) Log in to Brekeke PBX Admintool as system administrator (sa).
2) Go to the [Options] > [User Access Settings] page in the [Admin Menu] section.
3) Select on in [DID] menu.

Step 2: Enable a Route’s DID Access

1) Log in to Brekeke PBX Admintool as system administrator (sa).
2) Go to [ARS] and select an active ARS route template that can be accessed from a non-sa administrator’s [DID] menu.
3) Go to the selected route template’s [Variable Setting] page, click on the [Field Settings] link next to [Route Local Settings].
4) Select [Yes …] in the [DID] field.

[No] disables access to the routes from the DID menu in administrators' accounts.

[Yes (Modify only)] enables access to the active routes under the route template from administrators' accounts, which administrators can use to modify a route's variable settings.

[Yes (Modify/Add/Delete)] enables access to all routes under the route template from administrators' accounts, which administrators can use to modify a route's variable settings, to enable or disable a route and to add or delete a route.

- ✓ *The [Yes (Modify/Add/Delete)] option is not available for Brekeke PBX MT Edition.*

5) For Brekeke PBX MT Edition, set the tenant administrator's access privilege for each variable to one of the following options:

[Tenant Access (List)] enables or disables displaying of the variable in the route table of tenant administrator accounts.

[Tenant Access (Edit)], if checked, allows the variable settings to be modified from tenant administrator accounts. If unchecked, the variable settings will not be modifiable from tenant administrator accounts.

6. System Setup

6.1. Start / Shutdown

At the [Start / Shutdown] page, the system administrator can perform the following functions: Check Brekeke PBX and its bundled SIP Server running status; check current events, such as ARS route registration history; restart or shutdown Brekeke PBX and bundled SIP Server.

- ✓ Media Server running status has been added for the v3.4 or later.

6.2. Options

The following list displays the settings under the [Options] menu, which is only available to system administrators:

6.2.1. Settings

- ♦ **General Settings**

Name	Default value	Description
Start up	Auto	Auto: Brekeke PBX starts up automatically with the Tomcat (Brekeke PBX HTTP Service). Manual: Start up manually. Options: Auto / Manual

✓ *From v 3.6.x, The [Start up] item is moved under [Start/Shutdown] menu as an [Auto Start] check box.*

- ♦ **PBX System Settings**

Name	Default value	Description
Call pickup prefix	*	Prefix for picking up calls.
Port number	5052	Port number that Brekeke PBX will use. Modify as needed to avoid port conflicts.
Max concurrent sessions	Depends on license	Maximum number of concurrent sessions that Brekeke PBX can handle (cannot be modified).
Max number of UAs (User Agents)	Depends on license	Maximum number of SIP UAs that Brekeke PBX can handle (cannot be modified).
Min RTP port	30000	Minimum port number the RTP Protocol uses for sending voice data. Adjust setting as needed.
Max RTP port	49999	Maximum port number the RTP uses for sending voice data. Adjust setting as needed.

RTP relay	on	on – RTP is handled by Brekeke PBX. off – RTP is not handled by Brekeke PBX. (Applied unless there is a different RTP relay setting specified at the user phone type or the ARS Routes.)
SRTP	off	on – need to set [RTP relay] on, enable SRTP for all calls at Brekeke PBX system off – applied unless there is a different SRTP setting specified at the user extension phone type or the ARS route [SRTP] field
Video	off	on – need to set [RTP relay] on, enable video for all calls at Brekeke PBX system off – applied unless there is a different video setting specified at the user extension phone type or the ARS route [Video] field
Codec priority	0	G.711 u-law (PCMU) is used by default. When specifying multiple payloads, separate them with a comma (,). The following payload types can be used at Brekeke PBX: 0 - G.711 u-law 8 - G.711 A-law 18 - G.729 98 – iLBC If the codec priority is not set in ARS Routes or user phone type, this setting will be applied. From Brekeke PBX v3.3.x, Brekeke PBX supports dynamic payload type of iLBC even only payload 98 can be set for iLBC from Brekeke PBX Admintool [Codec priority] field.
Use remote preferred codec	no	Use the codec setting that is preferred at the remote SIP UA. If “default” is set in [Use Remote Preferred Codec] or in ARS or user phone type, this setting will be applied.
Max concurrent	10	Maximum number of concurrent sessions with call

recording sessions		recording.
Use user's name as Display Name	Yes	Use user account name as display name.
Ringing timeout (ms)	240000	Amount of time to wait for an answer from the dialed party after ringing starts.
Talking timeout (ms)	259200000	Maximum length of time a call can last in talking, measured by the amount of time since the last SIP packet was received. Value 0 signifies infinite.
Max hop number	20	Maximum number of SIP servers or Brekeke PBX that a call can go through (hop number).
Days to keep call logs	90	Number of days to keep call logs.
Session Timer (sec, 0=disable)	0	Interval to allow UAs and SIP server to determine whether the SIP session is still active.
Session keep alive (sec)	600	Interval to send keep-alive packets to UAs during a call when RTP relay is set to off and session timer has not been used.
RTP Session Timeout (ms)	600000	Timeout value for Brekeke PBX awaiting the next RTP packet.
100rel	off	Enable (on) / Disable (off) for using reliable provisional responses (1xx series).
RFC2833	on	Enable (on) / Disable (off) RFC2833 setting.
Valid client IP pattern		Web service security – used by Brekeke PAL and Brekeke Web Service; Set with regular expressions of the PAL and Web Service clients IP pattern.
Java VM arguments		Parameters passed to VM.

- **Media Server System Settings**

Name	Default value	Description
Port number	5056	Port number that the Media server system uses. Modify as needed to avoid port conflicts.
Max concurrent	Depends upon	Maximum number of concurrent sessions for the

session limit	the license	Media server (cannot be modified).
Codec priority	0	G.711 u-law (PCMU) is used by default. Separate with comma (,) when specifying multiple codecs. Also see the PBX [Codec priority] description.
Use remote preferred codec	no	Enable (no) / Disable (yes) on using remote codec by the endpoints.
Max stored messages	50	Maximum number of saved voicemail messages, plus any recorded files for each user's voicemail.
Message recording length (sec)	600	Maximum length of recording time for a voicemail message. If [Message recording length (sec)] in User setting is blank, this value will be applied.
Days to keep unsaved messages	30	Number of days before unsaved messages are deleted automatically from each user's voicemail inbox.
Conversation recording length (sec)	3600	Maximum recording length for each call.
Conversation recording file in voicemail inbox	Yes	Yes – save recording file in user's voicemail box. No – don't save recording file in user's voicemail box, but show recording files in call logs.
Days to keep conversation recording files	30	This field will be enabled when [Conversation recording file in voicemail inbox] is set to No.
Min RTP port	50000	Minimum port number the RTP uses for sending voice data.
Max RTP port	59999	Maximum port number the RTP uses for sending voice data.
Ringing timeout (ms)	240000	Timeout value for awaiting an answer from the dialed party once the ringing starts.
Talking timeout (ms)	259200000	Maximum length of time a call can last in talking, measured by the amount of time since the last SIP packet was received. Value 0 signifies infinite.
RTP Session Timeout (ms)	600000	Timeout value for awaiting the next RTP packet.

Java VM arguments		Parameters passed to VM.

✓ *ms = 0.001 second*

♦ **Email Settings**

Name	Default value	Description
SMTP Server		The SMTP server address for sending email notifications when the user receives a new voicemail message.
SMTP port	25	SMTP server's listening port.
SMTP authentication	on	Enable (on) / Disable (off) SMTP authentication setting.
Encrypted connection (SSL)	off	Enable (on) / Disable (off) Encrypted Connection (SSL), available in v2.3 or later.
POP3 server		Address of the POP3 server (for POP-before-SMTP authentication).
POP3 port	110	POP3 server's listening port.
User		Email account username for the above SMTP server.
Password		Email account password.
Password (confirm)		Input field for confirming the above password.
Email address (from)		Email notification sender's address.
Email subject	voicemail({to}) : from {from}	Email subject for the email notifications. The following variables can be configured: {from}: SIP URI who left the voicemail message {to}: voicemail box's SIP URI {from-number}: number in {from} {to-number}: number in {to} {time}: time when the messaged is recorded {recording-length}: length in time of the recorded message
Email body	from:{from} to:{to} time:{time} recording length(sec):{re	Email body for email notifications. Variables that can be used in this field are the same as in the [Email subject] above.

	cording-length	

✓ *From v3.6.x, the mail server setting parameters are moved under the [Email] menu -> [Settings].*

♦ **Multi-tenant Settings (MT Edition)**

Name	Default value	Description
G. 729 License Sharing	on	on – G.729 codec license is shared among tenants. off – The number of G.729 codec licenses is set on each tenant's [Options] page.

♦ **PAL Settings**

Name	Default value	Description
Notification for registration	yes	Sends notification to Brekeke PAL or Brekeke PAL WebSocket application (or not) when a user phone has registered.
PAL WebSocket	yes	This field has been available in v 3.4 or later. Yes – PAL WebSocket is enabled. No – PAL WebSocket is disabled.
Valid WebSocket client IP Pattern		Web service security – used by Brekeke PAL WebSocket; Set with regular expressions of the WebSocket clients IP pattern.

6.2.2. User Access Settings

Restricts the features and menus that are displayed and modifiable from a User or Admin account when the user logs into Brekeke PBX Admintool.

6.2.3. Phone Type

Name	Default value	Description
Type name	Type 1 Type 2 Type 3 Web Phone*	The default phone types that users can choose. User extension > [Phones] page [Type] field. Default setting is Type 1. *[Web Phone] is added for v3.6 or later. This phone type is used when the bundled web phone is used. Currently in v3.6, the web phone is provided as a beta release. It will be provided as a standard

		release in v3.7.
Description		Describe phone type.
WebRTC	no	When set to :yes, Brekeke PBX provides a WebRTC connection to WebRTC clients that have phones assigned to this Phone Type.
RTP relay	default	When set to default, user phones assigned with this phone type will apply the RTP relay setting in the [Options] menu. When set to off, Brekeke PBX will not relay RTP for the user phones assigned with this phone type.
SRTP	default	default – user phones assigned with this phone type will apply the SRTP setting in the [Options] menu. Optional – Both RTP and SRTP calls can be made with the user phones assigned with this phone type. Available with [RTP relay] on mandatory – only SRTP call will be made with the user phones assigned with this phone type. Available with [RTP relay] on off – Brekeke PBX will not enable SRTP for the calls with the user phones assigned with this phone type
Video	default	default – the user phones assigned with this phone type will apply the video setting in the [Options] menu. On – Brekeke PBX will enable video for the calls of the user phones assigned with this phone type off – Brekeke PBX will not enable video for the calls of the user phones assigned with this phone type
Codec priority		The default setting will apply the [Codec Priority] setting in the [Options] menu. Brekeke PBX will first apply the [Codec Priority] setting in the phone type assigned to user phones if the setting in the phone type is different from the one in the [Options] menu.
Use remote	default	When set to default, the setting in the [Options]

preferred codec		menu will be applied. Brekeke PBX will first apply the setting in the phone type assigned to user phones if the setting in the phone type is different from the one in the [Options] menu.
Keypad commands	on	When set to on, the phones assigned with this phone type can use Brekeke PBX keypad commands, such as #9, #8, etc. When set to off, the phones assigned with this phone type cannot use Brekeke PBX keypad commands.
MWI (NOTIFY without SUBSCRIBE)	off	When set to off, the phones assigned with this phone type can send SUBSCRIBE for MWI. When set to on, the phones assigned with this phone type cannot send SUBSCRIBE for MWI.
Tag in Display Name	off	This feature has been available in v 3.4 or later. Add as a prefix – Tag feature is enabled. Tag value is displayed as an added prefix of a "from" header on the devices assigned with this phone type. Replace – Tag feature is enabled. Tag value is displayed as a "from" header on the devices assigned with this phone type. Off – This feature is disabled.
Update remote party ID	off	When "re-INVITE" is set at this setting, remote-party ID will be updated with a re-INVITE request. This feature has been added in the v3.6 or later.
Stand-alone video client	off	Video client is available for use when this is set to "on", a video client is available for use. This feature has been added in the v3.6 or later.
Phonebook for Display Name	Yes	If caller's entry is in phone book, the first and last name registered in phonebook will be shown at callee's device.
Properties		Set properties that do not have corresponding fields in Brekeke PBX Admintool.

6.2.4. Auto Sync

The [Auto Sync] menu is used for Brekeke PBX redundancy setup. This feature requires a license upgrade.

6.2.5. SA (MT Edition)

SA is the only system administrator in Brekeke PBX MT Edition. From the [SA] page, you can change the SA login password and set the system language.

- ✓ From v3.6, The [SA] section is integrated with [System Administrators] menu. With this change, multiple administrators can be created under one system.

6.2.6. Advanced

The [Advanced] field allows you to set properties that do not have corresponding fields in Brekeke PBX Admintool. Please refer to other manuals and tutorials regarding the type of properties that may be edited here.

6.3. Voice Prompts

6.3.1. System Voice Prompts

Upload a customized sound file to overwrite the system default sound file, or to use as needed. A list of Name, Language and Description will be displayed.

Name	Description
Language	Choose folder in which to save an uploaded file.
Name	Name for the uploaded file in the folder. If the file name is the same as the system default sound file, the uploaded sound file will be played.
Description	A memo shown on the GUI to describe the file usage.
File name	To upload a file, click the [Browse] button. Select the file you want to upload and click [Upload]. The upload will then start.
Download	To download a recorded sound file, click Download ( ). The file will be downloaded to your PC as a WAV file.
Delete	To delete a recorded sound file, click Delete ( ). The selected file(s) will be deleted.

6.3.2. Notes for Sound Files

Uploaded sound files must be formatted as below:

Sample rate	8000 Hz
Bit-depth	16-bit
Channels	Mono

You may use sound-recording applications, such as Windows Microsoft Sound Recorder, to record sound files. We recommend that you adjust the pause and sound level to suit your needs.

6.4. Automatic Route Selection (ARS)

Brekeke PBX automatically selects the optimum call route from the preset routing options. This feature can be used for Least Cost Routing, traffic management and load balancing of VoIP Gateways or PBXs.

6.4.1. Adding a New Route

1) Choose [ARS] > [New Route].
2) Type the name of a route in the input field on the new popup window.
3) Click **[OK]** to add the route.

✓ *In Brekeke PBX v3.0.x, choose [ARS] > [Settings] > [New Route].*

6.4.2. Editing, Copying or Deleting a Route

1) Uncheck [Hide Disabled Rules] to show all ARS routes and details.
2) Click the ARS route name to edit a route.
3) Choose the copy or delete icon to perform the desired action.

✓ *In Brekeke PBX v3.0.x, select [ARS] > [Settings] from the submenu.*

✓ *In Brekeke PBX v3.8.1.x, import/export feature is available. A route can be exported by clicking an export icon in each route. Import can be conducted from the link at bottom of a route list.*

6.4.3. Viewing an Active Route

1) Check [Hide Disabled Rules] to display active rules only.
2) Click “View” in [Status(Reg/Route)] to show details about the active route.
   For the routes in an ARS template, click the ARS route template name and then “View” an active route under the route template.

✓ *In Brekeke PBX v3.0.x, choose [ARS] > [Running Status]. If no ARS route is enabled, there will be no* *route displayed under [Running Status]. Selecting [Settings] will display all ARS Routes.*

6.4.4. ARS > Route Template

- **General**

Name	Default value	Description
Route name		Name of the route.
Description		Description of the route.
Template		Define if this route is a template (Brekeke PBX v3.1.x or later).
Disabled	checked	Disable / Enable the ARS route.
Type	Type A	Used for special functions.
Group		ID for a group of ARS routes.
External	unchecked	When checked, Brekeke PBX will recognize this ARS route as an external line.
LineKey	unchecked	Check if you use Line keys (optional feature).
Session interval (ms)		Set the interval period between sessions for any VoIP FXO Gateways that require pausing between sessions.
Apply this route for incoming calls	Yes	This setting takes effect when there is a registration setup in the route. If set to yes, the incoming calls will apply this ARS route only when the INVITE request URL is the same as the one in the contact header of the REGISTER request sent by this route.
Resource		Set the resource name for users with the SCA feature enabled (optional feature, Brekeke PBX v3.1.x or later).
Tenant (MT Edition)		Set a tenant name.

- **Registration**

Name	Default value	Description

Register URI		SIP URI that is used to register Brekeke PBX at a remote registrar server. Leave this blank when there is no need to register Brekeke PBX to any remote registrar server.
Proxy address		IP address of the registrar server. This field is optional when the proxy address is the same as the address set in the Register URI field.
Register expire (sec)	3600	Set when REGISTER expires.
Register update period (%)	90	The percentage value of the interval until re-register occurs, calculated from the length specified in the register expire setting above.
User		User ID for authentication account required by remote registrar server. Entry is not necessary when authentication is not used.
Password		Password for authentication account required by remote registrar server. Entry is not necessary when authentication is not used.

♦ **Pattern – IN**

Name		**Default value**	**Description**
Priority		100	Lower numbers hold a higher priority.
Max sessions		-1	Specify the number of sessions (including RINGING and BYE sessions) that are allocated to the route.
Disabled		unchecked	Enable / Disable this pattern.
Matching patterns	From		Specify a matching rule for the From header using regular expressions. When the field is left blank, all calls will be considered as matched.

	To		Specify a matching rule for the To header using regular expressions. When the field is left blank, all calls will be considered as matched.
	Plugin		Java class name for the plug-in.
	Param		Parameters that will be used by the plug-in.
	Return		Pattern of the value returned by the plug-in.
	Apply to Request URI instead of To	unchecked	If checked, compares the Request URI instead of the To header; mostly designed for using ITSP accounts.
	Apply only to calls related to registration	unchecked	If checked, the route only applies to the calls related to registration.
Deploy patterns	From		Specify replace patterns for the From header using regular expressions.
	To		Specify replace patterns for the To header using regular expressions.
	Custom		Used for special functions.
Parameters	RTP relay	default	Select RTP relay ON / OFF. If default is selected, the setting will be the same as [Options] > [RTP relay] (unless specified at the User settings). on – RTP is handled by PBX. off – RTP is not handled by PBX.

	SRTP	default	If default is selected, the setting will be the same as [Options] > [SRTP] (unless specified at the User settings). off – SRTP is not handled by PBX. optional – Brekeke PBX will handle both RTP and SRTP call; available with [RTP relay] on mandatory – only SRTP call will be handled by Brekeke PBX; available with [RTP relay] on
	Codec priority		Specify codec to be used. Use a comma (,) when specifying multiple payloads. The following payload types may be used with Brekeke PBX: 0 - G.711 u-law 8 - G.711 A-law 18 - G.729 98 – iLBC From Brekeke PBX v3.3.x, Brekeke PBX supports dynamic payload type of iLBC even only payload 98 can be set for iLBC from Brekeke PBX Admintool [Codec priority] field.
	Use remote preferred codec	default	Enable (on) / Disable (off) for using the remote codec used by the endpoints. When set to default, the remote codec in the [Options] setting will be applied.
	Block SIP INFO (DTMF)	no	Block or pass-through the SIP INFO (DTMF) from a user to the other party.
	Send RTCP	off	off – PBX will not handle RTCP packets. on – PBX will handle RTCP packets.

	SDP 18x	default	Default: depends on the response from the callee. Block: remove SDP. Append: attach SDP. If SDP is not included in the packets, Brekeke PBX will play the Ring-Back Tone.
	Video	default	If default is selected, the setting will be the same as [Options] > [Video] (unless specified at the User settings). on – video call is handled by PBX, off – video is not handled by PBX.

♦ **Patterns – OUT**

Name		**Default value**	**Description**
Priority		100	Lower numbers hold a higher priority.
Max sessions		-1	Specify the number of sessions (including RINGING and BYE sessions) that are allocated to the priority.
Disabled		unchecked	Enable / Disable this pattern.
Matching patterns	From		Specify a matching rule for the From header using regular expressions. When the field is left blank, all calls will be considered as matched.
	To		Specify a matching rule for the To header using regular expressions. When the field is left blank, all calls will be considered as matched.
	User	^.+$	Mostly designed for multiple ITSP accounts, it specifies the users to which this ARS Route applies.

	Class		Specifies the users in the class to which this ARS Route applies.
	Plugin		Java class name for the plug-in.
	Param		The parameters which will be used by the plug-in.
	Return		The pattern of the value returned by the plug-in.
Deploy patterns	From		Specify replace patterns for the From header using regular expressions.
	To		Specify replace patterns for the To header using regular expressions.
	Target		Destination IP address. May omit entry when the destination IP address is specified in the To header domain.
	DTMF		When DTMF needs to be issued after calling a gateway (two-stage calling), you can specify the DTMF string using some part of the [To] matching pattern.
	Confirm		Define the voice prompt used with a confirm call.
	Key	5	Define the confirm key entry.
	Custom		Used for special functions.
Parameters	RTP relay	default	Select RTP relay ON / OFF. If default is selected, this value is the same as in [Option] menu > [RTP relay] (unless specified at the User settings). on – RTP is handled by Brekeke PBX. off – RTP is not handled by Brekeke PBX.

	SRTP	default	If default is selected, the setting will be the same as [Options] > [SRTP] (unless specified at the User settings). off – SRTP is not handled by PBX. optional – Brekeke PBX hande both RTP and SRTP call; available with [RTP relay] on mandatory – only SRTP call will be handled by Brekeke PBX; available with [RTP relay] on
	Codec priority		Specify codec to be used. Use a comma (,) when specifying multiple codecs.
	Block SIP INFO (DTMF)	no	Stop (or not) passing DTMF from a user to the other party when Brekeke PBX receives DTMF.
	Send RTCP	off	off – Brekeke PBX will not handle RTCP packets. on – Brekeke PBX will handle RTCP packets.
	Session timer(sec, 0=disable)	0	Interval to allow both UAs and SIP server to determine whether the SIP session is still active.
	100rel	off	Enable (on) / Disable (off) using reliable provisional responses (1xx series).
	Video	default	If default is selected, the setting will be the same as [Options] > [Video] (unless specified at the User settings). on – video call is handled by PBX off – video is not handled by PBX.
	Next route on failure	no	Set failover for outbound sessions (or not).

	Disable on registration failure	no	Enable (yes) / Disable (no) this Pattern when registration fails.
	Response timeout (ms)	-1	Period of time when a response has not been received before timeout is activated.
	Error codes	500	Failover will be activated when specified error codes are received for INVITE requests.
	Recovery time (ms)	0	Period of time until this pattern will be reactivated.
	Disable on failure	This route	Disable this route when using this OUT pattern fails. Can also be set to disable one pattern in the ARS route or multiple ARS routes with the same group ID. Options: This route, This pattern, This group

6.5. DID

Displays a list of DID-enabled routes. Administrators can modifiy enabled route fields, such as changing the destination extension where inbound calls to a related DID number will be sent. For more detailed setting instructions, please refer to the DID section above.

6.6. Call Status

The Call Status of ongoing calls is displayed under the [Call Status] menu. By specifying search criteria, the search result will be displayed on the screen. You can view detailed information for the selected search result.

Name	Description
Total	Total number of system active sessions
ID	Call ID
Status	Call status: In progress, Talking
UAs	Users' phone numbers in the current session

6.6.1. Status

Name	Description
ID	Call ID
Status	Call status
Call park	The number that has been parked
Conference	Conference number
Start	Time the call begins

6.6.2. UAs (User Agents)

Name	Description
User	Username
ARS	Used ARS route
URI	SIP URI
Connected	Time the call begins
Disconnect	Disconnect the call. (If the user does not have the right to disconnect, this option will not be displayed.)

6.7. Call Logs

Call log information is available through Brekeke PBX. By specifying a date, you can view the call log information for that date. By default, call log information is displayed in html in the browser. By clicking the [csv] button, you can download a log file to your local machine in .CSV format. Individual user call logs are also available under each user.

6.8. Notes

This menu item is used by Brekeke PBX plug-ins to access text data or to save script files for IVR script users. You can also use it for writing some memos.

Name	Description
Name	Name of the note
Description	Brief description of the note
User access level	Define if this note can be accessed by users, as well as what the access level is. Select from “No Access,” “Read only” or “Read/Write.”
Note	Text field where you can write your own notes

6.9. Extensions

On the left menu panel, click the [Extension] menu. Select different extension tabs and create extensions. Click Extension to edit this extension setting.

For user extension setting details, refer to the Brekeke PBX User Guide.

6.9.1. System Administrator

Name	Default value	Description
User type	Admin	Administrator user
User	sa	Default administrator username
Login password	sa	Default administrator login password

6.9.2. Group Extensions

♦ **Simultaneous Ring Group**

Name	Default value	Description
Extension		Extension
Type	Simultaneous ring	Extension type
Description		Extension description
Group extensions*		User extensions' IDs to which Brekeke PBX will forward a ring group call. Separate user extensions with commas.
Apply each extension's ring time	yes	The ring time settingof each extension is applied instead of the value of the [Ringer time (sec)] when this is set to "yes". This field has been added in the v3.6 or later.
Ringer time (sec)*	90	Amount of time phone will ring while waiting for the recipient to answer. After the length of time set here, the call will be transferred to the destination that is specified in the [Forwarding destination (No answer)] field. If no destination is set at [Forwarding destination (No answer)], the call will be terminated.

Forwarding destination (no answer)		Destination to which the call will be forwarded when ringer timeout has occurred.
Tag		Tag the call (Brekeke PBX v3.4. or later).

- **Call Hunting Group**

Name	Default value	Description
Extension		Extension
Type	Call Hunting	Extension type
Description		Extension description
Mode	Round-robin	There are two modes for call forwarding: Round-robin – Calls will be distributed starting from the top of the list. When a call is received, it is forwarded to the extension following the last extension that received a call. Top-down – Calls will always be distributed in the order listed in the field, beginning with the first number.
Hunt group extensions*		Enter extension number(s) to which all calls that are received at this extension number will be forwarded.
Apply each extension's ring time	yes	The ring time of each extension is applied instead of the value of the [Ringer time (sec)] when this is set to “yes”. This field has been added from v3.6 or later.

Ringer time (sec)*	20	Time interval Brekeke PBX waits before ringing each destination in the [Hunt group extensions*] field. Multiple time intervals can be set and separated by commas, such as, "5,15,0,10." With this setting, Brekeke PBX will ring the first destination when there is an incoming call, then wait for five seconds before ringing the second destination, then wait 15 seconds to ring third and fourth destinations together (because the third ring time is set to 0), and then wait ten seconds to ring the final destination.
Waiting time in the queue (sec)	0	Length of time queued calls will remain on hold before forwarding to the user destination set at [Forwarding destination (No answer)].
Max number of calls in the queue	10	Maximum number of calls that can wait in the queue at one time.
Call interval (msec)	3000	Amount of time to wait between dialing the available extensions in [Hung group extensions*] while callers are waiting in the queue.
Single attempt	no	Enable / Disable retrying calls when an initial call has not been answered. When this setting is enabled, the call will be transferred to the destination set at [Forwarding destination (No answer)] after the initial call is not answered.
Forwarding destination (No Answer)		Destination to which the call will be forwarded when timeout has occurred.

Sound Files

Name	Description
Music on hold	Audio file that contains the music/sound that will be played when the caller is on hold.

Tag	Tag the call (v3.4 or later).

Options in the queue (From Brekeke PBX v3.6 or later)

Name	Default value	Description
Transfer destinations* (0-9)		Destinations to which the call will be forwarded when DTMF is received (0-9).
DTMF Timeout	20	Length of time to wait for inputting destination number
Music on hold in the queue		Audio file that contains the music/sound that will be played when the caller is on hold in the queue.
Prompt for options		Audio file that contains the music/sound that will be played after the audio of [Music on hold in the queue] is ended.

6.9.3. Schedule Extensions

Name	Default value	Description
Extension		Extension
Description		Extension description
Forward To		Destination number to forward all calls that are received by this extension. Calls will be forwarded to the appropriate destination based on conditions defined in the schedule. The Tag field has been available since Brekeke PBX version 3.4.X. You can set an arbitrary string in the field.
Conditions		Specify schedule information for forwarding incoming calls. Tag can be added to calls as a part of the conditions in the v3.4 or later.

Name	Description
Default forwarding schedule*	Calls will be forwarded to the destination set here when no other schedules can be applied to the call.
Forwarding schedule [n]	Calls will be directed to the destination set here when the conditions are met.

Plan [n]	Multiple schedule plans can be created under a single extension. The active plan will be applied to calls.
Timer 1/Timer 2	Specify the schedule when a plan will be active.

6.9.4. IVR Extensions

For the IVR type [Flow] and [Script], refer to Brekeke PBX document – Developer's Guide: IVR

- **Auto Attendant**

Name	Default value	Description
Extension		Extension
Type	Auto Attendant	Extension type
Description		Extension description
Language	English	Select IVR language.
Max input digits	4	Maximum number of input digits
Max retry count	5	Maximum number of retries when an input error has occurred. After this number of retries has occurred, the call will be terminated.
Ring timeout (sec)	30	Length of time that the destination phone will ring when a call is received via Auto Attendant.
Default operator		Default destination (user extension number) for an incoming call that has not specified a call recipient.
DTMF timeout (sec)	20	Length of time to wait for inputting destination number

Speed dial		Set up “Speed dial” numbers for Auto Attendant. Example: • 1=100 In this case, instead of dialing 100 to reach user 100, a caller can dial 1. • 1==100 In this case, a caller can dial either 110# to reach user 110, or dial 1# to reach the user who is set as 100 in speed dial. • Regular expressions can be used for detecting input number pattern ^9.+$=100 In this case, all calls with inputting number starting with 9 will be transferred to user 100
Transfer to unregistered users	disable	Enables / Disables call transfers to an unregistered user. Options: disable / enable
Media before answer	off	This feature has been available since version 3.4.X. Enables (on) / Disables (off) – Allows the transfer of media packets from an external source before Brekeke PBX receives an answer.

Sound Files

Name	Description
Greeting message	Greeting message that is played for the Auto Attendant.
Retry message	A message to prompt the caller to re-enter the number when an input error has occurred.
Music on hold	Audio file that contains the music/sound that will be played when the caller is on hold.

♦ **Add / Remove Forwarding Destinations**

Name	Default value	Description
Extension		Extension

Type	Add / Remove forwarding destinations	Extension type
Description		Extension description
Language	English	Select IVR language.
Target groups*		By calling this extension, the caller's extension number will be added or deleted from the [Group Extensions] field for any group extensions that are set in this field.

♦ **Switch Plan**

Name	**Default value**	**Description**
Extension		Extension
Type	Switch Plan	Extension type
Description		Extension description
Language	English	Select IVR language.
Plan number	2	By calling this extension, the caller extension's [Inbound] page active plan value will change to the value set here.
On/Off	Yes	When set to yes, the plan number will be set as the active plan when a user calls this extension. The active plan will switch back to plan 1 when the user calls this extension again. When set to no, the plan number will be set as the active plan when a user calls this extension, but will not switch back to plan 1, regardless of how many times the user calls this extension.

6.9.5. Conference Extensions

Name	Default value	Description
Extension		Extension
Description		Extension description
Auto Invite attendees*		By specifying user extensions here, a user can dial this extension to invite multiple attendees to a conference.
Apply conf extension to From	no	Yes – Attendees can automatically see the conference extension number on their devices.
Applies to (caller numbers) *	*	Number patterns of callers who can join a conference. Wildcard characters can be used. A star (*) and a question mark (?) can be used for matching meta-characters. A star (*) means zero (0) or more characters; a question mark (?) means one character.
Host (1st attendee if left empty)		Available inv3.4 or later. By specifying a user extension here, that user is defined as the host of the conference. If no one is specified here, or if using v3.3.X or earlier, the user who enters the conference first is defined as the host of the conference.
Exit all when host leaves	no	Yes – when call host hangs up, conference call will end. No – when call host hangs up, conference call will continue if there are attendees in the conference room.
Broadcast	no	When set to yes, only the voice of the conference call host can be heard by other participants.

Hang up existing calls	no	When set to yes, and attendees PBX user already have another call in conversation at the time the conference call is initiated, Brekeke PBX will disconnect their current talking call and ring the attendees for conference call.

Sound Files

Name	Description
Music on hold while waiting	Audio file that contains the music/sound that will be played while waiting for other attendees to join the conference.

6.9.6. Callback Extensions

Name	Default value	Description
Extension		Extension
Description		Extension description
Ringer time (sec)	90	Amount of time phone will ring before directing the call to the forwarding destination.
Forwarding destination (no answer)		Destination where the caller will be directed when ringing times out.
Callback callee		Destination where Brekeke PBX will direct the caller when he/she disconnects a call before ringer time out.

6.9.7. Call Park (Available in v3.8.5.x or later)

To prevent leaving parked calls without anything, system provides the feature that defines the handling of parked calls that reach timed out.

Name	Default value	Description
New button		Create new call park settings.
Park Number(*1)	*	As default, all the numbers that are used for call parking will be targets of this setting.
Timeout(sec)	900	When call parking time reaches the timeout, the timeout action is executed.

Timeout action	No action	"No action" ->no action "Callback" ->PBX calls back to the extension that parked the call. "Call" ->PBX makes a call to the specified number.

6.9.8. Import/Export users (Available in v3.8.1.x or later)

Name	Default value	Description
Import users		
Template user		Select an existing user that is used as a template for creating new users by updating with imported data.
CSV File		Select an import file.
Separator	,	The separator that is used for separating values in an import file.
Map the first row as property keys	unchecked	If it is checked, the system maps the first row of the file as property keys of user.
Overwrite if users exist	unchecked	If it is checked, the system overwrites the existing users' properties with the imported data.
Export users		
Separator	,	The separator that is used for separating values in an export file.

7. Setup Items (MT Edition)

For tenant menu details that are not listed below, please refer to the system-level explanation above of the Tenant menu.

7.1. Tenants

From this menu, a system administrator can create a new tenant, delete a tenant and copy and modify existing tenant settings.

✓ *Shutting down Brekeke PBX MT Edition from Admintool is required when deleting a tenant.*

7.1.1. Copy and Set Up New Tenant

1) Click the copy icon of the tenant you wish to copy.
2) At the popup window, type in the new tenant name.
3) At the next screen, edit the setup for the new tenant and save the settings.

7.1.2. Settings Copied in New Tenant

In the new tenant, the following settings will be the same as the tenant that you copied:

- Users and user descriptions
- User settings, except [Phone ID]
- Voice prompts, including users' voice prompts and tenant system voice prompts
- Tenant options

Settings that will not be carried over:

- Tenant descriptions
- User extensions' Phone IDs
- Call logs
- Voicemail – all users' mailboxes will be empty.

7.2. Tenant Voice Prompts

Upload a customized sound file under a tenant to overwrite the system default sound file or to use as needed. The sound files uploaded at the tenant level will take effect only for the current tenant and will have higher priority than the sound files uploaded at system [Voice Prompts]. A list that includes Name, Language and Description will be displayed. For setup details, please refer to system level voice prompts.

7.3. Tenant Notes

This menu item is used by Brekeke PBX plug-ins to access text data or to save script files for IVR script users at the tenant level. You can also use this feature for writing memos. Notes created under a tenant have a higher priority than those created under system [Notes].
For setup details, please refer to system-level Notes.

The format for accessing tenant-level notes from Brekeke PBX plug-ins is as follows:
<tenant_name>.<note_name>

7.4. Tenant Options

Tenant settings can be modified from the Options menu under each tenant.

♦ **General**

Name	Description
Description	A brief description of the tenant

♦ **Capacity**

Name	Description
Maximum users	Maximum number of user extensions allowed under this tenant
Maximum sessions	Maximum number of concurrent sessions allowed for this tenant
Maximum recording sessions	Maximum number of concurrent recording sessions allowed for this tenant
Maximum G.729 license	Maximum number of concurrent G.729 sessions allowed for this tenant when G.729 license is not shared among system (license option required)
Maximum agents	Maximum number of call center agents (license option required) allowed for this tenant

♦ **Features**

Name	Description
Voicemail	Enable or disable this feature for the tenant
Auto-Attendant	Enable or disable this feature for the tenant
Conferencing	Enable or disable this feature for the tenant
Call Recording	Enable or disable this feature for the tenant
Round Robin – Topdown	Enable or disable this feature for the tenant
Call Queue	Enable or disable this feature for the tenant
PAL	Enable or disable this feature for the tenant

8. Maintenance

8.1. Back Up / Restore

You can back up current configurations and messages in users' voicemail boxes from [Maintenance] > [Back Up]. You can restore the backup data from the menu [Maintenance] > [Restore]. To best maintain your service, we recommend backing up Brekeke PBX on a regular basis.

8.2. Report (MT Edition)

Overview of all tenant setups that are hosted on the Brekeke PBX MT Edition system, as well as system tenant settings reports that can be exported in csv format.

9. Uninstallation

9.1. Uninstalling from Windows OS

1) Kill all Java processes (java.exe) used by Brekeke PBX from Task Manager.
2) Navigate to Start / All Programs / Brekeke PBX / Uninstall Brekeke PBX. The uninstaller will uninstall Brekeke PBX automatically.

✓ *If the uninstaller fails to delete the folder (C:\Program Files\Brekeke\pbx), you will need to restart the PC and delete the folder manually.*

9.2. Uninstalling from Linux OS

Delete the file “pbx.war” and the folder “pbx” in the directory “webapps,” which is located under the installation directory of Tomcat, then restart the machine.