# Brekeke PBX

**Version 3**

## Administrator's Guide

## (Basic)

**Brekeke Software, Inc.**

Version

Brekeke PBX Version 3 Administrator's Guide (Basic)

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# 1. Introduction

This document explains basic configuration of Brekeke PBX. For more advanced setting instructions and product information, refer to the Brekeke PBX Administrator's Guide (Advanced).

## 1.1. Editions

Brekeke PBX comes in several editions to meet the needs of different users.

| Edition | Explanation |
|---|---|
| Pro | Designed for businesses and services needing sophisticated call management and advanced features |
| Evaluation | Product trial prior to purchase with Pro Edition's feature set.<br>This license is free of charge. |
| Multi -Tenant | Designed as a platform for providing Hosted IP-PBX service for Service Providers. |

# 2. Installation

## 2.1. System Requirements

| | |
|---|---|
| **OS** | Microsoft Windows XP and later, Linux |
| **Java** | Java 6 or later (32bit / 64bit)<br>✓ *Brekeke products are confirmed to run on Java provided by Oracle Sun Microsystem* |
| **Apache Tomcat** | Version 6.x<br>✓ *Tomcat install is not required when Brekeke PBX installed with the executable installer.* |
| **Memory** | 512 MB minimum |

## 2.2. Select file to install Brekeke PBX

| | Executable Installer | pbx.war (zip format) |
|---|---|---|
| **OS** | Windows OS | Linux OS |
| **Install** | New Installation only | New Installation<br>Update Installation |
| **Instruction** | Section 2.3 (Windows) | Section 2.4 (Linux)<br>Section 2.5 (Update) |

## 2.3. Installation for Windows OS with the Executable Installer

**Step 1: Installing Java SE**

Install Java SE before installing the Brekeke PBX software.

1) Access the website http://www.oracle.com/technetwork/java/javase/downloads/index.html
2) Download and install the appropriate version of JRE or JDK for the type of Windows OS you are running.

**Step 2: Installing Brekeke PBX**

1) Obtain installer file from Brekeke's website.
2) Start installation following the installer instructions.

✓ *Brekeke PBX and bundled SIP server will be installed automatically.*

**Step 3: Starting Brekeke PBX HTTP Service**

If you did not check [Run Brekeke PBX 3.0] at the last stage of the installation, start Brekeke PBX HTTP Service by the following methods.

1) From taskbar, open Brekeke PBX Service Manager, and start Brekeke PBX service.
2) From Control Panel / Performance and Maintenance / Administrative Tools / Services, select and start "Brekeke PBX" service.
3) Restart computer, Brekeke PBX HTTP service will start automatically.

**Step 4: Starting Brekeke PBX Administration Tool (Admintool)**

1) Select Start / Program / Brekeke PBX / Brekeke PBX Admintool
2) Enter the 16 digit product ID in the given space.
   Entering the same product ID on multiple machines is not allowed.
3) At the login page, enter User ID and Password.
   The default setting for both administrator user ID and password is sa
4) Click [Start] button from the menu [Start / Shutdown]. If you see [Active] for status of both PBX and the bundled SIP Server, the Brekeke PBX server is started successfully. If you see [Inactive], Brekeke PBX server failed to start.

## 2.4. Installation for Linux OS

**Step 1: Installation of Java SE**

1) Access the website http://www.oracle.com/technetwork/java/javase/downloads/index.html
2) Download and install the appropriate version of JRE or JDK for the type of OS you are running.

**Step 2: Installation of Tomcat**

1) Access the website http://tomcat.apache.org/index.html and download the binary file of Tomcat version 6.x for the type of OS you are running.
2) Set JRE or JDK installation directory for the environment variable JAVA_HOME.

3) Install downloaded Tomcat file.

✓ *We recommend adding liveDeploy="false" to the server.xml file at Tomcat installation directory/conf/ as below .*

```
<Host name="localhost" appBase="webapps"
      unpackWARs="true" autoDeploy="true" liveDeploy="false"
      xmlValidation="false" xmlNamespaceAware="false">
```

**Step 3: Installation of Brekeke PBX**

1) Obtain the file pbx.war (zip format file) from Brekeke's website.
2) Copy war file directly into the "webapps" directory which is under the Tomcat installation directory

**Step 4: Starting Tomcat**

1) Start Tomcat.
2) Open a web browser and specify http://localhost:8080 as a URL (If you chose a port number other than the default "8080", specify the appropriate port number in the URL.)
3) Tomcat has started successfully if the Apache Jakarta Project page is displayed.

**Step 5: Starting Brekeke PBX Administration Tool (Admintool)**

1) At web browser, specify the URL http://localhost:8080/pbx/ (If you chose a port number other than default "8080", specify the appropriate port number in the URL.)
2) Enter the 16 digit product ID in the given space.
   Entering the same product ID on multiple machines is not allowed.
3) At the login page, enter User ID and Password.
   The default setting for both administrator user ID and password is sa.
5) Click [Start] button from the menu [Start / Shutdown]. If you see [Active] for status of both PBX and the bundled SIP Server, the Brekeke PBX server is started successfully. If you see [Inactive], Brekeke PBX server failed to start.

## 2.5. Updating Brekeke PBX

This section is for updating from an earlier version of Brekeke PBX v3.x to the current release. Please confirm that you have downloaded the update zip file (pbx.war) from Brekeke's website.

1) Open [Maintenance] > [Update Software].
2) If Brekeke PBX is active, [Shutdown] button is displayed. Click on [Shutdown].
3) Click the [browse] button to select the pbx.war file you have downloaded.
4) Click the [upload] button to upload the new file.
5) Restart your computer to complete updating Brekeke PBX.

# 3. Maintenance

## 3.1. Back Up / Restore

You can back up all of the current configurations and voicemail messages from the Brekeke PBX Admintool menu [Maintenance] > [Back Up]. We recommend backing up Brekeke PBX on a regular basis. You can restore the backup data from the menu [Maintenance] > [Restore]. To backup or restore, shutting down Brekeke PBX is required.

## 3.2. Activating License

1) Open [Maintenance] > [Activate License].
2) If Brekeke PBX is active, [Shutdown] button is displayed. Click on [Shutdown].
3) Read the End User License Agreement (EULA),
   Enter license ID then click on the [Accept terms and activate the license] button next to it.
   If reactivate current license, only click on the [Accept terms and activate the license] button at the bottom of [Activate License] page.
4) If your computer is connected to the Internet, license activation will start automatically.
   If there is not internet access, click on [Get Signature] and follow the instructions shown on the screen to activate license.
5) After completing the activation successfully, you will see the Login screen of Brekeke PBX Admintool.

# 4. Uninstall (Windows OS)

This topic will assist you with uninstalling the Brekeke PBX software from your computer with a Windows OS.

Navigate to **Start / Program / 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.*

# 5. Uninstall (Linux OS)

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

# 6. Setting Up Brekeke PBX

## 6.1. Setting Up Phone Type

There are three default phone types shown from Brekeke PBX Admintool > [Options] > [Phone Type] page when this is the first time Brekeke PBX installation or there is no other phone types created when upgraded from previous version. System administrators can also define new phone types with various combined settings for RTP relay, codecs and keypad commands.

Depending on system UAs location, codecs and Brekeke PBX feature requirements, user clients or system administrators can choose suitable phone type for each Brekeke PBX users assigned phones.

## 6.2. Setting Up User Extensions

After installing Brekeke PBX, you will need to create users extensions, and assign phone IDs to each user extension so that the UAs with assigned phone ID number can call each other and apply Brekeke PBX features.

### 6.2.1. Creating Users

In this example, user extensions 100 - 105 will be created.

1) Click Brekeke PBX Admintool > [PBX Admin] > [Extensions] > [Users].
   Default administrator “sa” has already been created. (Default password is “sa”.)
2) Click on the **[Create a new user]** button**,** user **[Account]** page will be shown and set as following:
   [Extension] 100
   [Login Password] 100
   [Login Password (confirm)] 100
   There is no default user login password. If no login password is set at the time user is created, user cannot login its account from Brekeke PBX Admintool.
3) Change the user account settings as needed
4) Create other user extensions 101 – 105 as above.

### 6.2.2. Default Values of Users

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

To change user settings, click User ID from [PBX Admin] > [Extensions] > [Users] list, or select user from [User] menu on the left panel, and navigate each page under the selected user.

| Page Name | Setting item | Details of default values |
|---|---|---|
| **Account** | Language | Same language as the administrator who created this user |
| **Settings** | Call Pickup group | Same group as the administrator |
| | Greeting message | Default system greeting |
| | Email notification | off |
| **Phones** | Phone 1 > [Phone ID] | The same as user ID |
| | Type | Type 1 |
| **Inbound** | Plan | Plan 1 (Active) |
| | Forward To > [Phones] | All phones are checked by default. When there is an incoming call to this user, Brekeke PBX will forward call to all assigned phones under this user. |
| | Forward To > [Call Forwarding] > [Ringer Time (sec)] | 90 seconds |
| | Forward To > [Call Forwarding] > [Forwarding destinations (No answer)] | Forwarded to user’s Voice Mail |
| | Forward To > [Call Forwarding] > [Forwarding destinations (Busy)] | Forwarded to user’s Voice Mail |

### 6.2.3. Assigning Phones to User Extensions

For each Brekeke PBX user, up to 4 different phone IDs can be assigned. The default phone ID is the same as Brekeke PBX user ID when this user is created and is assigned to [Phone 1] > [Phone ID] field

The phone ID can be a SIP UA’s user ID which is registered at Brekeke PBX bundled SIP Server, or a PSTN number which belongs to the owner of this user. The devices with the phone ID number assigned under a Brekeke PBX user can apply Brekeke PBX features to the call..

### 6.2.4. Voicemail Settings

After creating the user extensions, you can set up voicemail for each of the users. As described in the section above, user 100's [Forwarding destination (No answer)] and [Forwarding destination (Busy)] are set to "Voice Mail" by default. If devices assigned under user 100 do not answer after ringing for 90 seconds (or when user 100's line is busy), the call will be forwarded to voicemail. The "vm" prefix is specified under the "mediaserver_prefix" route at the ARS settings.

- To leave a voice message directly, dial 07*< user extension ID>.
- To check voice messages from your own phone, dial "8" to reach your voicemail inbox.
- To check voice messages from other phones, dial 08*<your user extension ID> to directly access your voicemail inbox.

### 6.2.5. Voicemail Notification by Email

**Step 1: Setting Up Email Sender**

Brekeke PBX supports mail servers which provide "POP before SMTP" authentication or "SMTP" authentication or Encrypted Connection (SSL).

To set up Email Sender:

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

**Step 2: Setting Up Email Recipient**

To set up an email recipient:

1) Navigate to a user [Settings] page by selecting a user from Brekeke PBX Admintool > [PBX Admin] > [Users]
2) Set [Voicemail settings] > [Email address*] to the recipient email address(es)
3) Set [Voicemail settings] > [Email notification] to "on".
4) Set [Voicemail settings] > [Attach WAV file to Email] to "on" or "off" (depending on whether you want to attach the voice message to the email or not)

### 6.2.6. Setting Up Call Forwarding

Call Forwarding is used when users are not available and you want those incoming calls to be redirected to another extension or to voicemail. In this example, calls to extension 101 will also be forwarded to extension 100.

1) Enter user extension number to which you want to forward the call. In this case, enter user extension “100” in user extension 101’s [Other Forwarding destinations*] field.
2) Make a call to extension 101. Both phones assigned under user extension 100 and 101 will ring.

### 6.2.7. Setting Up No Answer Call Forwarding

To forward incoming calls to another extension instead of using Voice Mail to answer calls after ringer time, set up the forwarding extension in the [Inbound] > [Forwarding destination (No answer)] field.

1) Enter user extension “100” in extension 101’s [Forwarding destinations (No answer)] field
2) Incoming calls will be forwarded to user extension 100 if user 101 phones do not answer in ringer time.

### 6.2.8. Setting Up Busy Call Forwarding

To forward incoming calls to another user extension while all phones under a user is “busy”, instead of using Voice Mail, set up the forwarding extension in the [Inbound] > [Forwarding destination (Busy)] field.

1) Enter user extension “100” in user extension 101’s [Forwarding destinations (Busy)] field
2) Calls will be forwarded to 100 if user 101 phones return a 486 Busy response or another error response.

## 6.3. Message Waiting Indicator (MWI)

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

For those SIP UAs that do not send “SUBSCRIBE” message for MWI, you can set Brekeke PBX to send voicemail notification to all PBX users.
To Enable MWI, set as below and restart Brekeke PBX from admintool.
[PBX Admin] > [Options] > [Settings] > [Message Waiting Indicator] = on
Default value is off.

## 6.4. Setting Up Ring Groups

In this example, a Ring Group (1000) is created for all user extensions 101 through 105. When calls are received at Group extension 1000, all specified group extensions (101-105) will ring simultaneously. If no one answers the call within ringer time (10 sec), the call will be forwarded to user extension 100 voicemail box.

1) From [PBX Admin] > [Extensions] > [Groups], click [Create a new group]
2) Set as following:
   [Extension] 1000
   [Type] Simultaneous Ring
   [Group Extensions*] 101,102,103,104,105
   [Ringer time (sec)] 10
   [Forwarding destination (No answer)] vm100
3) Save the setting

## 6.5. Setting Up Call Pickup Group

Call Pickup is a function that allows users to answer incoming calls to any other user extensions by dialing a pre-set number. When a user extension rings, dial *** <user extension ID>** to answer the call. For example, when user extension 100 rings, dialing * 100 will enable you to answer the call from any other Brekeke PBX user extension.

✓ *If you are using a SIP phone that supports a "Call Pickup" button, please consult with the manufacturer of your SIP phone on how to set up Call Pickup feature.*

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

♦ **Answer calls that are directed to a Ring Group**

Calls directed toward a Simultaneous Ring Group extension can be answered from any user extension using Call Pickup. (For details on how to set up a Ring Group, please refer to section "Setting Up Ring Groups")

Extension: 1000 (Simultaneous Ring Group)

| | |
|---|---|
| **Group Extensions*** | 101,102,103,104,105 |

Dialing *1000 (Simultaneous Ring Group extension number) enables one to pick up the calls and the user extensions in the Ring Group will stop ringing. Incoming calls can still be answered by dialing *<user extension ID>, such as *101, *102… However, using the group extension does not require you to remember each user extension ID in the group. Dialing

*<ring group extension number> feature works even when a call is directly to a single extension (e.g. 103), or comes through the Auto Attendant.

- **One touch Call Pickup for specified Call Pickup Group**

  By setting up a Call Pickup Group number (generally a ring group number) in a user extension, you only need to dial *(Star) to pickup the incoming calls to any of user extensions in the group. You may specify a Call Pickup Group at your user extension's [Settings] > [Call Pickup group], such as, 1000.

  When there is an incoming call to any user extensions, 101 – 105, in Simultaneous Ring Group extension 1000 > [Group Extensions] field, you can answer the call by dialing * (Star) only.

## 6.6. Setting Up Call Hunting

Here is how to set up Call Hunting:

1) From [PBX Admin] > [Extensions] > [Groups], click [Create a new group]
2) Set as following:

   [Extension] 1001

   [Type] Call Hunting

   [Mode] Round robin / Top-down

   [Hunt group extensions*] 101,102,103,104,105

   [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)] 100
3) Save the setting

In this example, the call will ring the user extensions in [Hunt group extensions*] field one by one with waiting time [Ringer time(sec)](5 sec) interval between calling each user extension. When all of the group extensions (101-105) are busy or do not answer, the call will be queued. If any member becomes available within the interval set in [Waiting time in the queue (sec)], Brekeke PBX will ring the available members one by one with waiting time [Call interval (msec)](3000 ms) to ring next group extension after previous user ringing times out. If all members continue to be busy after the specified interval at [Waiting time in the queue (sec)], the call will be forwarded to the destination set in [Forwarding destination (No answer)]. In this example, the call will be forwarded to user 100.

The music on hold, which caller is hearing when waiting in queue, can be changed by uploading new sound file from [Sound files] > [Music on hold] field.

## 6.7. Setting Up Auto Attendant

The example below shows Auto Attendant for extension 1002.

1) From [PBX Admin] > [Extensions] > [IVR], click [Create a new IVR] button
2) Set as following:
   [Extension] 1002
   [Type] Auto Attendant
   [Max input digits] 3
   [Max retry count] 5
   [Ring timeout (sec)] 10
   [Default operator] 100
   [DTMF timeout (sec)] 20
   [Transfer to unregistered users] disable
3) Save the settings

In this example, the incoming call to Auto Attendant 1002 will hear default greeting voice prompt for user extension input.

If there is no input in 20 sec ([DTMF timeout (sec)]), Brekeke PBX will ask for re-inputting user extension. After 5 times of no extension input ([Max retry count]), the call will be forwarded to destination set at [Default operator] field.

If caller input extension numbers, the call will be transferred to the user extension whose ID is the same as the first 3 digits ([Max input digits]). If transferred recipient does not answer call within 10 sec ([Ring timeout (sec)]), Brekeke PBX will stop ringing input user extension and Auto Attendant will ask for inputting another extension number.

When [Transfer to unregistered users] is set to enable, caller can apply Brekeke PBX features from Auto Attendant, such as accessing voicemail box, call pickup, call monitoring and so on. Proper value needs to set at Auto Attendant > [Max input digits] field to allow caller’s input.

Optionally, an audio file for Auto Attendant greeting can be uploaded from Auto Attendant > [Sound files] section.

## 6.8. Setting Up Call Forwarding Schedule

This feature is useful for creating rules for call forwarding during specified times. In this example, incoming calls to Schedule extension 5555 during business hours are directed to the Auto Attendant at extension 1002. After business hours, callers are scheduled to hear “To speak with a live operator, please call during regular business hours. Our regular business hours are Monday through Friday, 9 a.m. to 6 p.m.”

The following settings can be used to set up such a scenario.

1) Create an Auto Attendant 1003 and upload after business hour greeting to [Greeting messages] under Auto Attendant 1003
2) From [PBX Admin] > [Extensions] > [Schedule], click [Create a new schedule] button
3) Set as following:
   [Extension] 5555
   [Default Forwarding Schedule] > [Call forwarding] > [Destination] 1003
4) Click on [Add Forwarding Schedule] button
   Set under [Forwarding Schedule 1] as following:
   At [Forward To] > [Call forwarding] > [Forwarding Destination] field, set 1002
   Click on [Conditions] tab, set [Date/Time] section as following
   [Term] set Schedule starting and ending Date and Year
   [Days] Check 1st- 5th and Monday through Friday
   [Time] select and check 9:00 - 18:00
5) Save the settings

## 6.9. Setting Up Switch Plan

Switch Patterns can be used to temporarily and quickly change a user’s forwarding destination by setting different forwarding schedule plan as active.

In this example, incoming calls during business hours go directly to user 100 assigned phones. However, during user 100’s lunch break, user prefers incoming calls to go directly to voicemail. User 100 can enter DND (Do Not Disturb) mode by switching between pre-defined inbound plans to route incoming calls to his voicemail temporarily during lunch time.

We can achieve this by using a Switch Pattern as described below.

**Step 1: Setting Up Switch Plan extension**

1) From [PBX Admin] > [Extensions] > [IVR], click [Create a new IVR] button
2) Set as following:
   [Extension] 1004
   [Type] Switch Plan
   [Plan number] 2
   [on/off] yes
3) Save the settings

**Step 2: Setting Up Plans in user extension**

1) From User 100 > [Inbound] page, click [Add new plan] button
   "Plan 2" will show in the plan drop down list window
2) Uncheck all phones list in Plan 2 and set [Other Forwarding Destinaitons*] as vm100
3) Save the settings

**Step 3: Enter Do Not Disturb Mode**

1) User 100 dial Switch Plan extension 1004 from his phone to enter DND mode
   At user 100 > [Inbound] page, Plan 2 will be set as active
2) Any calls to user 100 at this time will be directed to user 100 voicemail inbox.

**Step 4: Remove Do Not Disturb Mode**

1) User 100 dial Switch Plan extension 1004 from his phone again to remove DND mode.
   At user 100 > [Inbound] page, the plan before entering DND mode will be set as active again.
2) After resuming from DND mode, any calls to user 100 will be directed to his/her assigned phones.

## 6.10. Setting Up Conference Call

### 6.10.1. Creating a Conference Room

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

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

1) From [PBX Admin] > [Extensions] > [Conference], click [Create a new conference] button
2) Set as following:
   [Extension] 2000
   Leave all other settings as default

3) Save the settings

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

### 6.10.2. Limiting Members Who Can Enter the Conference Room

You can limit members that join the conference by specifying members (for example “101,102,103”) at **[Applies to (Caller numbers)*] field**. With this setting only user 101, 102, and 103 will be allowed to join the conference. Any other users,cannot join this conference room.

### 6.10.3. Simultaneous Calls to All of the Conference Members

A conference member can convene all members of the conference room at once. For example, set 101, 102, 103 at **[Forwarding destinations*]**. By dialing 2000, all conference members (101, 102, and 103) will be invited simultaneously.

### 6.10.4. Starting a Conference Call (Alternate Methods)

Additional methods for starting a conference call are described in the Brekeke PBX Users Guide.

## 6.11. Setting Up Callback

With callback feature, Brekeke PBX will ring caller back when caller dial to Callback user. In the following example, user 3000 is set up as Callback extension.

1) From [PBX Admin] > [Extensions] > [Callback], click [Create a new callback] button
2) Set as following:
   [Extension] 3000
   [Ringer time (sec)] 90
   [Forwarding destination (No answer)] 100
   [Callback callee] 1002
3) Save the settings

When caller dial to Callback extension, caller will hear ring tone. If caller hangs up call before ringer timeout, Brekeke PBX will send INVITE to caller back and caller will be directed to destination set in [Callback callee] field when original call answer the call. In this example, caller will be connected to Auto Attendant 1002.

If caller who dial to Callback extension does not hang up call and ringer timeout, the call will be directed to the destination set in [No answer forwarding destination] field

## 6.12. Setting Up Confirm Call

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

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

## 6.13. Setting Up Paging

The phones which 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 Brekeke PBX paging function and its sample configuration are available at Brekeke Wiki http://wiki.brekeke.com/wiki/paging-function_phone-list

## 6.14. Setting Up Busy Lamp Field, Presence, and Shared Call Appearance

♦ BLF (Busy Lamp Field)

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

♦ SCA (Shared Call Appearance)

With SCA, users can monitor the external line status and select an available line to place an outbound call or answer the incoming calls.

♦ Presence

With 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 function and its sample configuration are available at Brekeke Wiki http://wiki.brekeke.com/wiki/BLF-SCA-and-Presence.

## 6.15. ARS Settings

For more information about the ARS feature, please refer to Brekeke PBX Administrator's Guide (Advanced).

- **General**

  Use [General] section to enable or disable ARS template and settings required by some Brekeke PBX features, such as, Shared Call Appearance, ARS group, call logs database for billing and so on.

- **Registration**

  - Register 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.

  - When making outbound calls from UAs to an ITSP through Brekeke PBX and provider needs authentication information from caller, Brekeke PBX will send user and password set in this section to provider IP or domain set in [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 priority 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. Lower numbers hold the higher priorities

- **Max Sessions**

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

  For the ARS routes in the same group, there is only one session counter. The session counter will increase 1 when there is a call through ARS route no matter Pattern IN or OUT. If this session counter is equal to any [Max Sessions] value set in the ARS routes in the same group, the next matched session cannot apply to this pattern (IN or OUT), or any ARS routes when all patterns' [Max Sessions] of the ARS routes in the same group are set the same value.

- **Edit Variables**

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

## 6.16. PSTN Access Using a VoIP Gateway

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

### 6.16.1. VoIP Gateway Setup

Set the following at your VoIP Gateway:

| | |
|---|---|
| **SIP proxy address** | IP address of Brekeke bundled SIP server |
| **Dialing number sent to Brekeke PBX** | PSTN line number |

### 6.16.2. ARS Route Setup

Setup “Patterns - IN” and ”Patterns – OUT” in Gateway ARS Route to receive and make calls from/to the Gateway.

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

**[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** | |

### 6.16.3. 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` |

Click the [Edit Variables] link at upper-right corner of ARS Route page

| v1 | v2(password) | v3 |
|---|---|---|
| PSTN line number set at Section “VoIP Gateway Setup” | (leave blank) | Specify a Brekeke PBX extension number |

### 6.16.4. 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 dials a number, whose digits are between 7 and 25, will be considered as a PSTN call. Brekeke PBX will apply the above ARS Route and the call will be sent to gateway.

If you have multiple VoIP Gateways used for outbound calls, define more detailed dialing pattern in [Matching patterns] > [To], and change [Priority] field as your need to define the usage order of Gateways. Please note lower numbers hold higher priorities. And use [Max Sessions] field to define the total 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 us suppose the gateway’s PSTN port has the SIP user name, 111. To call a PSTN number, dial the gateway’s PSTN registered port number (in this example, it is number 111) and then dial the destination PSTN number, or configure DTMF setting at ARS Route OUT pattern so that the dialed numbers will be sent to gateway as DTMF tones.

The OUT pattern to send destination number by DTMF in two stage dialing is shown as below:

**Patterns - OUT**

| Matching patterns | | Deploy patterns | |
|---|---|---|---|
| **From** | | **From** | |

| | | | |
|---|---|---|---|
| **To** | `sip:111(.+)@` | **To** | `sip:111@gw_IPaddress` |
| | | **DTMF** | `$1` |

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

## 6.17. Connecting with Internet Telephony Service Providers (ITSPs)

### 6.17.1. 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 upon 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.*

### 6.17.2. Setting ARS for ITSP using multiple accounts

**[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 the Proxy address is the same as the one in [Register URI] field |
| **User** | `&v1` | Set value at [Edit Variables] page |
| **Password** | `&v2` | Set value at [Edit Variables] page<br>This field will be displayed in text format after saving. |

**[Patterns – IN]**

In the pattern-IN example below, Brekeke PBX user extension 100 (&v3 value, set at [Edit Variables] page) is set to ring when a call comes through the 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 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 be changed 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 FROM or TO header information is different from their own header format.*

**[Edit Variables]**

Click the [Edit Variables] link at upper-right corner of ARS Route page

| v1 | v2(password) | v3 |
|---|---|---|
| 6504106636 | 6636 | 100 |

## 6.18. 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 fail over the session to an alternative route.

### 6.18.1. Usage Examples

- Brekeke PBX provides automatic failover to alternative ITSP service in the event of failure at your specified ITSP service.
- Brekeke PBX provides automatic failover to analog telephone session via PSTN Gateway in the event of failure at your specified ITSP service.

♦ Creating redundant analog telephone connections with multiple PSTN Gateways.

### 6.18.2. Setting Examples

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

In the following example, two ARS routes are created, “ITSP_A” and “MyGateway”. The route with highest priority “ITSP_A” will be used for outbound calls with 7 to 25 digit dialing number. If there is no response within 4 seconds (Response timeout: 4000 ms) for INVITE messages or “500-599” response was received, Brekeke PBX will continue searching next route matching outbound session request. And the next highest prioritized route “MyGateway” will be chosen for the alternative route for the session. Since the recovery time is set for 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 back on running in an hour, the sessions will be routed through the highest priority route, “ITSP_A” again.

**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. For PSTN Gateways and SIP servers located in the local network may not require setting long Response Timeout intervals. For the route that requires an internet connection or if a delay is expected, the Response Timeout intervals should be set longer.

- [Disable on registration failure] is set as "yes" in Route "ITSP_A". When registration is not working property at "ITSP-A" route, it will be disabled and "MyGateway" route will be used instead.
- [Disable on failure] is set as "This route". It will disable whole route. If there are other IN/OUT patterns defined in this route, they will be unusable when failover happens. Set as "This pattern" will only disable the current pattern and 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` |

# 7. Setup Items

## 7.1. Start/Shutdown

At start/shutdown page, system administrator can check Brekeke PBX and its bundled SIP Server running status, current events like ARS route registration history, and restart or shutdown Brekeke PBX and bundled SIP Server.

## 7.2. Options

The following list displays the settings under the **[Options]** menu. This menu is only available to system administrators.

### 7.2.1. Settings

♦ **General Settings**

| Name | Default value | Description |
|---|---|---|
| **Start up** | Auto | Auto: Brekeke PBX starts up automatically with Tomcat (Brekeke PBX HTTP Service).<br>Manual – Start up manually.<br>Options: Auto/Manual |

♦ **PBX System Settings**

| Name | Default value | Description |
|---|---|---|
| **Call pickup prefix** | * | Prefix for picking up calls |
| **Port number** | 5052 | The port number that Brekeke PBX will use.<br>Modify as needed to avoid port confliction |
| **Max concurrent sessions** | Depends on license | The maximum number of concurrent sessions that Brekeke PBX can handle. (Cannot be modified) |
| **Max number of UAs (User Agents)** | Depends on license | The maximum number of SIP UAs that Brekeke PBX can handle. (Cannot be modified) |
| **Min Port** | 30000 | Minimum port number the RTP Protocol uses for sending voice data. Adjust setting as needed |
| **Max 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.<br>off – RTP is not handled by Brekeke PBX. |

| | | |
|---|---|---|
| | | (Applied unless there is different RTP relay setting specified at the User phone type or ARS Routes) |
| **Codec priority** | 0 | G.711 u-law (PCMU) is used by default. Separate with comma (,) when specifying multiple payloads. The following payload type can be used at the Brekeke PBX:<br>0 - G.711 u-law<br>8 - G.711 A-law<br>18 - G.729<br>98 - iLBC<br>If Codec priority is not set in ARS Routes or user phone type setting, this setting will be applied. |
| **Use Remote Preferred Codec** | no | Use codec setting that is preferred at the remote SIP UA. If “default” is set in [Use Remote Preferred Codec] in ARS or user phone type, this setting will be applied. |
| **Max concurrent recording sessions** | 10 | Maximum concurrent sessions with call recording |
| **Ringing Timeout (ms)** | 240000 | Timeout value for awaiting an answer from the dialed party after ringing starts. |
| **Talking Timeout (ms)** | 259200000 | The maximum length of time a call can last in talking. When no other SIP packets are received for a period of time. 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 both 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) on using reliable provisional responses (1xx series) |
| **RFC2833** | on | Enable (on) / Disable (off) RFC2833 setting Available |
| **Valid client IP Pattern** | | web service security -- used by Brekeke PAL and Brekeke Web Service; set with regular expression |
| **Java VM arguments** | | Parameters passed to VM |

♦ **Media Server System Settings**

| Name | Default value | Description |
|---|---|---|
| **Port number** | 5056 | The port number that Media server system uses. Modify as needed to avoid port confliction |
| **Max concurrent session limit** | Depends upon the license | Maximum number of concurrent sessions for Media server. ( cannot be modified.) |
| **Codec priority** | 0 | G.711 u-law (PCMU) is used by default. Separate with comma (,) when specifying multiple codecs. See also the description for [Codec priority] |
| **Use Remote Preferred Codec** | no | Enable (no) / Disable (yes) on using remote codec used by the endpoints. |
| **Max stored messages** | 50 | Maximum number of saved voicemail messages and any recorded file 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 | The number of days before unsaved messages is deleted automatically from each user's voicemail inbox. |
| **Message Waiting Indicator** | off | Enable(on) / Disable(off) Message Waiting Indicator (Voice mail notification to phones) |
| **Conversation recording length (sec)** | 3600 | Maximum recording length for each call. |
| **Min Port** | 50000 | Minimum port number the RTP uses for sending voice data. |

| | | |
|---|---|---|
| **Max 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 after ringing starts. |
| **Talking Timeout (ms)** | 259200000 | Timeout value for canceling a session. The timeout value is calculated after the last SIP session received while session is in talk. |
| **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 since version 2.3 |
| **POP3 server** | | The address of the POP3 server. (for POP-before-SMTP authentication) |
| **POP3 Port** | 110 | POP3 server’s listening port |
| **User** | | email account user name for the above SMTP server. |
| **Password** | | email account password |
| **Password (confirm)** | | Input field for confirming the above password. |
| **Email address (from)** | | Email notifications sender’s address. |
| **Email subject** | voicemail({to}) : from {from} | E-mail Subject for the e-mail notifications.<br>The following variables can be configured. |

| | | |
|---|---|---|
| | | {from}: SIP URI who left the voicemail message<br>{to}: voicemail box's SIP URI<br>{from-number}: number in {from}<br>{to-number}: number in {to}<br>{time}: time when the messaged is recorded.<br>{recording-length}: time length of the recorded message |
| **Email body** | from:{from}<br>to:{to}<br>time:{time}<br>recording length(sec):{recording-length | Email body for the e-mail notifications.<br>Variables can be used in this field are the same as [Email subject] (See above). |

- **PAL Settings**

| Name | Default value | Description |
|---|---|---|
| **Notification for Registration** | Yes | Send notification to PAL application or not when a user phone has registered. |

### 7.2.2. User Access Settings

Restrict the features those are displayed and can be modified from User account when User log into Brekeke PBX Admintool

### 7.2.3. Phone Type

| Name | Default value | Description |
|---|---|---|
| **Type Name** | Type 1<br>Type 2<br>Type 3 | The default phone types which can be chosen from User extension > [Phones] page [Type] field.<br>Default setting Type 1 |
| **Description** | | Describe phone type |
| **RTP relay** | Default/off | When set as default, user phones assigned with this phone type will apply the RTP relay setting the same as the RTP relay setting in [Options] menu.<br>When set as off, Brekeke PBX will not relay RTP for the user phones assigned with this phone type. |

| | | |
|---|---|---|
| **Codec Priority** | | The default setting will apply [Codec Priority] setting in [Options] menu. Brekeke PBX will first apply the [Codec Priority] setting in phone type assigned to user phones if the setting in phone type is different from the settings in [Options] menu. |
| **Use Remote Preferred Codec** | Default | Set as default, the setting in [Options] menu will be applied. Brekeke PBX will first apply the setting in phone type assigned to user phones if the setting in phone type is different from the settings in [Options] menu. |
| **Keypad Commands** | On/off | Set as on, the phones assigned under user extension can use Brekeke PBX keypad command, such as #9, #8 Set as off, the phones assigned under user extension cannot use Brekeke PBX keypad command |
| **Properties** | | Set properties which do not have corresponding fields in the Brekeke PBX Admintool |

### 7.2.4. Auto Sync

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

### 7.2.5. Advanced

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

## 7.3. Voice Prompts

### 7.3.1. System Voice Prompts

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

| Name | Description |
|---|---|
| **Language** | Choose folder to save uploaded files |
| **Name** | The name for the uploaded files in the folder<br>If file name is the same as system default sound files, the uploaded sound file will be played. |
| **Description** | A memo shown on the GUI for the file usage |
| **File Name** | To upload a file, click the [Browse] button.<br>Select the file you want to upload and click [Upload].<br>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 the recorded sound files, click **Delete ( )**. The selected files will be deleted. |

### 7.3.2. Notes for Sound Files

Uploaded sound files must be formatted as below.

| | |
|---|---|
| **Sample rate** | 8 kHz |
| **Bit-Depth** | 16 bit |
| **Channels** | Mono |

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

## 7.4. Automatic Route Selection (ARS)

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

### 7.4.1. Adding a New Route

To add a new route:

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

### 7.4.2. Editing, Copying, or Deleting a Route

To edit/copy/delete a route:

1) Select **[ARS] > [Settings]** from the submenu.
2) Uncheck [Hide Disabled Rules] and [Hide Details] to show all ARS Routes and details
3) Click on ARS route name to edit a route
4) Choose copy, or delete icon to complete related actions.

### 7.4.3. Viewing an Active Route

To view an active ARS route:

1) Choose **[ARS]** > **[Running Status]**.
2) Uncheck [Hide Details] to show details about each Route
3) Click on route name to show the details about the active route

If no ARS route is enabled, there will be no route displayed under [Running Status]. Select [Settings] will display all ARS Routes.

### 7.4.4. ARS > Route Template

- **General**

| Name | Default value | Description |
|---|---|---|
| **Route name** | | The name for the route |
| **Description** | | The description for the route |
| **Disabled** | checked | Disable / enable the ARS route |
| **Type** | Type A | Used for special occasions |
| **Group** | | ID for a group of ARS Routes |
| **External** | Unchecked | When checked, Brekeke PBX will recognize this ARS route as external line. |
| **LineKey** | unchecked | Check if you use Line keys. (Optional feature) |
| **Session interval (ms)** | | Set interval period between sessions for some VoIP FXO Gateways that require pausing between sessions. |
| **Apply this route for incoming calls** | Yes | This setting takes effect when there is registration setup in the same route.<br>If set as yes, the incoming calls will apply this ARS route only when INVITE request URL is the same as that in the contact header of REGISTER request sent by this route. |

♦ **Registration**

| Name | Default value | Description |
|---|---|---|
| **Register URI** | | SIP URI that is used to register Brekeke PBX at 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 REGISTER expires. |
| **Register Update Period (%)** | 90 | The percentage value of the interval until re-register occurs is calculated from the length specified in the Register Expire setting. |
| **User** | | User ID for authentication account. Entry is not necessary when authentication is not used. |
| **Password** | | Password for authentication account. 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 From header using regular expressions. When the field is left blank, all calls will be considered as matched. |

| | | | |
|---|---|---|---|
| | To | | Specify a matching rule for To header using regular expressions. When the field is left blank, all calls will be considered as matched. |
| | Plugin | | The 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 |
| | Apply to Request URI instead of To | unchecked | If checked, compare the Request URI instead of To header Mostly design for using ITSP accounts. |
| | Apply only to calls related to registration | unchecked | If checked, the route only apply to the calls related to registration. |
| **Deploy patterns** | From | | Specify replace patterns for From header using regular expressions. |
| | To | | Specify replace patterns for To header using regular expressions |
| | Custom | | Used for special occasions |
| **Parameters** | RTP relay | default | Select RTP relay ON/OFF.<br>If “Default” is selected, the setting is the same as [Options] > [RTP relay]. (Unless specified at the User settings)<br>on – RTP is handled by PBX.<br>off – RTP is not handled by PBX. |
| | Codec Priority | | Specify codec to be used. Use a comma (,) when specifying multiple payload. The following payload type can be used at the Brekeke PBX:<br>0 - G.711 u-law<br>8 - G.711 A-law<br>18 – G.729<br>98 - iLBC |

| | | | |
|---|---|---|---|
| | Use Remote Preferred Codec | default | Enable (on) / Disable (off) on using remote codec used by the endpoints. "default": remote codec setting in [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<br>on – PBX will handle RTCP packets |
| | SDP 18x | default | "default": depend upon the situation of how sessions are established<br>"block": remove SDP<br>"append": attach SDP.<br>If SDP is not included in packets, Ring-Back Tone will be played by Brekeke 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 From header using regular expressions. When the field is left blank, all calls will be considered as matched. |
| | To | | Specify a matching rule for To header using regular expressions. When the field is left blank, all calls will be considered as matched calls. |

| | | | |
|---|---|---|---|
| | User | ^.+$ | Mostly designed for multiple ITSP accounts. It is used for specifying which account this ARS Route is applying. |
| | Plugin | | The 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 From header using regular expressions. |
| | To | | Specify replace patterns for To header using regular expressions. |
| | Target | | Destination IP address. May omit entry when the destination IP address is specified in To header domain. |
| **Deploy patterns** | DTMF | | For when DTMF needs to be issued after calling gateway (2-stage calling), you can specify the DTMF string using some part of [To] Matching Pattern. |
| | Confirm | | Define voice prompt used with confirm call |
| | Key | 5 | Define confirm key entry |
| | Custom | | Used for special occasions |
| **Parameters** | RTP relay | default | Select RTP relay ON/OFF.<br>If “Default” is selected, [Option] menu > [RTP relay] (Unless specified at the User settings).<br>on – RTP is handled by Brekeke PBX.<br>off – RTP is not handled by Brekeke PBX. |
| | Codec Priority | | Specify codec to be use. Use a comma (,) when specifying multiple codec. |

| | | | |
|---|---|---|---|
| | Block SIP INFO (DTMF) | no | Stop or not for passing the DTMF from a user to the other party when Brekeke PBX received DTMF |
| | Send RTCP | off | off - Brekeke PBX will not handle RTCP packets<br>on – Brekeke PBX will handle RTCP packets |
| | Session Timer(sec, 0=disable) | 0 | Interval to allow both user agents and SIP server to determine whether the SIP session is still active. |
| | 100rel | off | Enable (on) / Disable (off) on using reliable provisional responses (1xx series) |
| | Next route on failure | no | Set failover for outbound sessions or not |
| **Parameters** | Disable on registration failure | no | Enable (yes)/Disable (no) this Pattern when registration failed |
| | Response timeout (ms) | -1 | The period of time before timeout is activated when response has not been received |
| | Error codes | 500 | Failover will be activated when specified error codes are received for INVITE requests |
| | Recovery time (ms) | 0 | The period of time till this pattern will be reactivated |
| | Disable on failure | This route | Disable this route when using this OUT pattern failed. Also can be set to disable one pattern in the ARS route or multiple ARS routes with the same group ID.<br><br>Options: This route, This pattern, This group |

## 7.5. Call Status

The Call Status of ongoing calls is displayed under the **[Call Status]** menu. By specifying search criteria, the search result is 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** | The users' phone number in current session |

### 7.5.1. Status

| Name | Description |
|---|---|
| **ID** | Call ID |
| **Status** | Call Status |
| **Call Park** | The number that has been parked |
| **Conference** | Conference number |
| **Start** | Time the call begin |

### 7.5.2. UAs (User Agents)

| Name | Description |
|---|---|
| **User** | User Name |
| **ARS** | Used ARS route |
| **URI** | SIP URI |
| **Connected** | Time the call begin |
| **Disconnect** | Disconnect the call. (If the user does not have rights to disconnect, this option will not be displayed. |

## 7.6. Call Logs

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

## 7.7. Notes

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

| Name | Description |
|---|---|
| Name | Name of the note. |
| Description | A brief description of the note |
| User access level | Define if this note can be accessed by users and access level<br>Select from “No Access”, “Read only”, “Read/Write” |
| Note | Text field where you can write your own notes. |

## 7.8. Extensions

Click the menu **[Extension]** on the left menu panel. Select different extension tabs and create extensions. Click on extension ID to edit this extension setting.

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

### 7.8.1. System Administrator

| Name | Default value | Description |
|---|---|---|
| **User Type** | Admin | Administrator user |
| **User** | sa | Default administrator user name |
| **Login Password** | sa | Default administrator login password |

### 7.8.2. Group Extensions

- **Simultaneous Ring Group**

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Simultaneous Ring | Extension type |
| **Description** | | Extension description |
| **Group Extensions*** | | User extensions’ ID to which Brekeke PBX will forward ring group call<br>Separate user extension ID by comma |

| | | |
|---|---|---|
| **Ringer time (sec)*** | 90 | Ringer timeout for 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. |

♦ **Call Hunting group**

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Call Hunting | Extension type |
| **Description** | | Extension description |
| **Mode** | Round-robin | There are two modes for call forwarding.<br>Round Robin: Calls will be distributed from the top of the list. When a call is received, it is forwarded to the extension following the last extension to have received a call.<br>Top-down: Calls will always be distributed in the order listed in the field. |
| **Hunt group extensions*** | | Enter phone number(s) to forward all calls that are received at this extension number. |
| **Ringer time (sec)*** | 20 | The awaiting time interval between Brekeke PBX rings each destination in [Hunt group extensions*] field<br>Multiple time interval can be set and separated by comma, such as, 5.15.0.10<br>With the setting, Brekeke PBX will ring first destination when there is an incoming call, and wait for 5 sec to ring the second destination, then wait 15 sec to ring third and forth destination together because the third ring time is set as 0, then wait 10 sec to ring next destination |

| | | |
|---|---|---|
| **Waiting time in the queue (sec)** | 0 | The length of time for queued calls will be held on hold till forwarded to the user destination set at [Forwarding destination (No answer)]. |
| **Max number of calls in the queue** | 10 | The maximum number of calls can wait in the queue. |
| **Call interval (msec)** | 3000 | The interval period for calls in queue to ring a client that end the call session. |
| **Single attempt** | no | Enable/disable to retry calls when an initial try has not been answered. When this setting is enabled, the call will be transferred to the destination set at [Forwarding destination (No answer)] after initial try is not answer. |
| **Forwarding destination (No Answer)** | | Destination to which the call will be forwarded when timeout has occurred. |

### 7.8.3. Schedule Extensions

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Call Hunting | Extension type |
| **Description** | | Extension description |
| **Forwarding destinations*** | | Destination phone 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. |
| **Conditions** | | Specify schedule information by which to forward incoming calls. |

| Name | Description |
|---|---|
| **Default Forwarding Schedule*** | The call will be forwarded to the destination set here when all other schedules can not be applied to the call |
| **Forwarding Schedule [n]** | The call will be directed to the destination set in forwarding destination when conditions is fulfilled |

| | |
|---|---|
| **Plan [n]** | Multiple schedule plans can be created under an extension<br>The Plan showing active will be applied. |

### 7.8.4. IVR Extensions

- **Auto Attendant**

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Auto Attendant | Extension type |
| **Description** | | Extension description |
| **Max input digits** | 4 | Maximum number of input digits accepted by the Auto Attendant. |
| **Max retry count** | 5 | Maximum number of retries when an input error has occurred. After retries of the number of times set here, the call will be terminated. |
| **Ring timeout (sec)** | 30 | The length of time that a 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. |
| **Speed dial*** | | Set up “Speed dial” for Auto Attendant.<br>Example: 0=0001. In this case, instead of dialing 0001 to reach the user 0001, a caller can dial 0. |
| **Transfer to unregistered users** | disable | Enables/disables call transfers to an unregistered user.<br>Options: disable/enable |

**Sound Files**

| Name | Description |
|---|---|
| **Greeting message** | Greeting message that is used for the Auto Attendant. |
| **Retry message** | A message to prompt the caller to reenter when an input error occurred. |
| **Music on hold** | An audio file that contains music/sound that will be used for music on hold. |

- **Add/Remove Forwarding Destinations**

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Add/Remove forwarding destinations | Extension type |
| **Description** | | Extension description |
| **Target groups*** | | By calling this extension, caller's extension number will be added/deleted from [Group Extensions] field in group extension set in this field. |

- **Switch Plan**

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Type** | Switch Plan | Extension type |
| **Description** | | Extension description |
| **Plan Number** | 2 | By calling this extension, caller user extension's [Inbound] page active plan value will be changed to the value set here |
| **On/Off** | Yes | When set as yes, plan number will be set as active plan when user call to this extension and active plan will switch back to plan 1 when call this extension again.,<br>When set as no, user active plan can only set to this plan number but cannot switch back to plan 1 no matter how many times call to this extension |

### 7.8.5. Conference Extensions

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Description** | | Extension description |
| **Forwarding destinations*** | | By specifying user extensions ID, a user can invite multiple users to this conference by dialing this extension. |

| | | |
|---|---|---|
| **Applies to (Caller numbers) *** | * | Restrict on caller that can dial in and join this conference.<br>A star (*) and a questions mark (?) can be used for matching meta-characters. A star (*) means zero (0) or more characters and a question mark (?) means one character. |

### 7.8.6. Callback Extensions

| Name | Default value | Description |
|---|---|---|
| **Extension** | | Extension ID |
| **Description** | | Extension description |
| **Ringer time (sec)** | 90 | Ringing timeout before direct call to the forwarding destination |
| **Forwarding destination (No answer)** | | The destination caller will be directed to when ringing time out |
| **Callback callee** | * | The destination Brekeke PBX will connect caller to when caller disconnect call before ringing timeout |