Brekeke PBX

Version 2

Administrator's Guide

(Basic)

Brekeke Software, Inc.

Version

Brekeke PBX Version 2 Administrator’s Guide (Basic)

Revised February 2010

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
Basic	Supports the essential IP-PBX features for typical business offices
Evaluation	Product trial prior to purchase with Pro Edition's feature set. 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 2000/XP/2003/Vista, Red Hat Linux
Java	Version 1.6 or later (32bit) ✓ *Brekeke products are confirmed to run on Java provided by Sun Microsystems.*
Apache Tomcat	Version 5.5.12 or later ✓ *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	Red Hat Linux OS
Install	New Installation only	New Installation Update Installation
Instruction	Section 2.3 (Windows)	Section 2.4 (Red Hat Linux) 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://java.sun.com/javase/downloads/index.jsp
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 [Start Brekeke PBX] at the last stage of the installation, start Brekeke PBX HTTP Service by the following method.

1) Open [Control Panel]> [Performance and Maintenance]> [Administrative Tools]> [Services], then select and start [Brekeke PBX].
2) After the restart, Brekeke PBX HTTP service will start automatically.

Step 4: Starting Brekeke PBX Administration Tool (Admintool)

1) Select [Start]> [Program]> [Brekeke]> [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. [i]
4) Click the menu [RESTART/SHUTDOWN]. If you see [Active] for Brekeke PBX Status, Media server Status and the bundled SIP server Status, the Brekeke PBX server was started successfully. If you see [Inactive], the Brekeke PBX server failed to start.

2.4. Installation for Red Hat Linux OS

Step 1: Installation of Java SE

1) Access the website http://java.sun.com/javase/downloads/index.jsp
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 5.5.12 or later for the type of OS you are running.
2) Set JRE or JDK Install directory for the environment variable JAVA_HOME.
3) Install using the download 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.

Copy 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.

[i] Default user ID and password are sa/sa (case sensitive).Recommend to change the default password for "sa" user

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. [ii]
4) Click the menu [RESTART/SHUTDOWN]. If you see [Active] for Brekeke PBX Status, Media server Status and the bundled SIP server Status, the Brekeke PBX server was started successfully. If you see [Inactive], the Brekeke PBX server failed to start.

2.5. Updating Brekeke PBX

This section is for updating from an earlier version of Brekeke PBX v2.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) If your license is not activated, the license activation window will be displayed. (Refer to the section 3.2 “Activating License”.)
6) Restart your computer to apply the changes.

[ii] Default user ID and password are sa/sa (case sensitive).Recommend to change the default password for “sa” user

3. Maintenance

3.1. Back Up / Restore

You can back up all of the current configurations and voicemail messages from the 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 necessary.

3.2. Activating License

1) Open [Maintenance] > [Activate License].
2) If Brekeke PBX is active, the [Shutdown] button is displayed. Click on [Shutdown].
3) Read the End User License Agreement (EULA) then click on the [I agree] button.
4) If your computer is connected to the Internet, click on [Activate] button. If not, click on [Get Signature] and follow the instructions shown on the screen.
5) After completing the activation successfully, you will see the Login screen of Brekeke PBX Admintool.

4. Setting Up Brekeke PBX

4.1. Setting Up Brekeke PBX Users

4.1.1. Creating Users

After installing Brekeke PBX, you will need to create users (extensions). In our example, extensions 1001 through 1005 will be created.

1) Click Brekeke PBX Admintool > [Users]
   Default administrator “sa” has already been created. (Default password is “sa”.)
2) Click on the **[New User]** submenu and enter a user ID (such as 1001) in the popup window. A new user will be added to Brekeke PBX. User edit page will be shown when a new user is created.
3) Change the user settings as needed.
4) Continue adding other users as extensions 1002 to 1005.

4.1.2. 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.

Setting item	Details of default values
Enabled	Yes
Language	Same language as the administrator who created this user
Password	Same password as the administrator who created this user
Ringer time (sec)	90 seconds
Forwarding destinations (No answer)	Forwarded to user’s Voicemail [iii]
Forwarding destination (Busy)	Forwarded to user’s Voicemail [iv]
Call Pickup group	Same group as the administrator
Greeting message	Default system greeting
Email notification	Off

✓ *To change user default settings, click the Edit button or User ID (Brekeke PBX v2.3 and later).*

[iii] Prefix number for users’ voicemail is set in the **[Voicemail prefix]** menu under the **[Options]** >[Settings] menu. The default prefix for voicemail is set at vm. (e.g., vm1001 for extension 1001 user)
[iv] Prefix number for users’ voicemail is set in the **[Voicemail prefix]** menu under the **[Options]** >[Settings] menu. The default prefix for voicemail is set at vm. (e.g., vm1001 for extension 1001 user)

4.2. Voicemail Settings

After creating the user extensions, you can set up voicemail for each of the users. As described in the section above, user 1001's [Forwarding destination (No answer)] and [Forwarding destination (Busy)] are set to "vm1001" by default. The prefix "vm" is the prefix for reaching the voicemail inbox directly. If 1001 does not answer after ringing for 90 seconds (or when 1001'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*<extension number>.
- 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 extension number> to directly access your voicemail inbox.

4.3. Voicemail Notification by Email

4.3.1. Setting Email Sender

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

To set up the Email Sender:

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

4.3.2. Setting Up Email Recipient

To set up an email recipient:

1) Navigate to Brekeke PBX Admintool > [Users > Edit] > [Voicemail settings]
2) Set Email address to the recipient 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 the voice message to the email or not)

4.4. Message Waiting Indicator (MWI)

For those SIP UAs that support MWI, you can set Brekeke PBX to support voicemail notification. The following section will help you set up your SIP UAs:

4.4.1. Settings at Brekeke PBX

To Enable MWI, set:

Brekeke PBX Admintool > [Options] > [Settings] > [Message Waiting Indicator] = on

4.4.2. Settings at SIP UA

If your SIP UA supports MWI with “Subscribe” message, the MWI feature will be activated even when the Brekeke PBX [Message Waiting Indicator] is set to “off”. If there is a special button 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 NOTIFY packet (for MWI) to retrieve the message without assigning the number manually.

4.5. Setting Up Call Forwarding

Call Forwarding is used when users are not available at their extensions and you want those incoming calls to be redirected to another extension or to voicemail. In this example, calls to extension 1002 will be forwarded to extension 1001.

1) Enter the extension number to which you want to forward the call. In this case, enter extension “1001”, in extension user 1002’s [Forwarding destinations*] field.
2) Make a call to extension 1002. Extension 1001 will ring.

4.6. Setting Up Ring Groups

In this example, a Ring Group (1000) is created for all extensions 1001 through 1005. When calls are received at extension 1000, all specified extensions (1001-1005) ring simultaneously.

1) Create extension 1000.
2) Enter 1001, 1002, 1003, 1004, 1005 (separated users by comma) in extension 1000’s **[Forwarding destinations*]** field.

4.7. Setting Up No Answer Call Forwarding

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

1) Enter the extension “1000” in extension 1001’s [Forwarding destinations (No answer)] field
2) Calls are forwarded to 1000 (Ring group) if 1001 does not answer and its ringer time expires.

4.8. Setting Up Busy Call Forwarding

To forward incoming calls to another extension while a user is “busy”, instead of using Voicemail, set up the forwarding extension in the field, **[Forwarding destination (Busy)]**.

1) Enter the extension “1000” in extension 1001’s [Forwarding destinations (Busy)] field
2) Calls are forwarded to 1000 (Ring group) if 1001 returns a 486 Busy response or another error response.

4.9. Setting Up Call Pickup Group

Call Pickup is a function that allows users to answer incoming calls to any Brekeke PBX extension by dialing a pre-set number. When a Brekeke PBX extension rings, dial *** <extension>** to answer the call. For example, when extension 1001 rings, dialing *** 1001** 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 Ring Group extension can be answered from any extension using Call Pickup. (For details on how to set up a Ring Group, please refer to section 4.6 “Setting Up Ring Groups”)

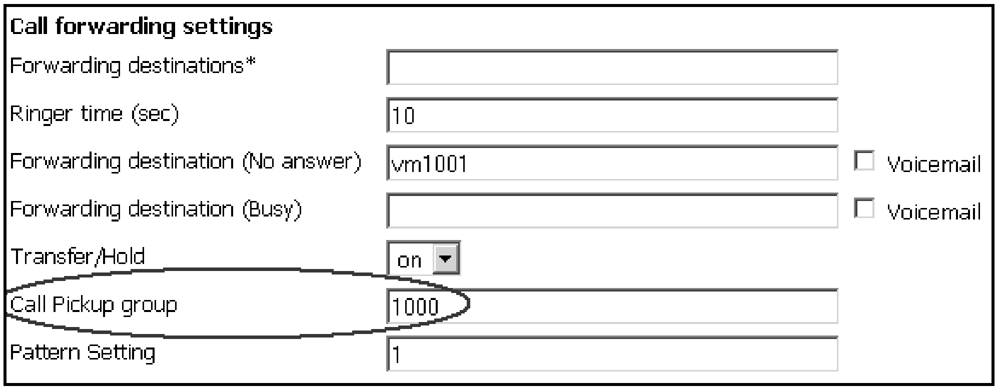
Extension: 1000 (Ring Group)

Forwarding destinations*	1001,1002,1003,1004,1005

Dialing *1000 (Ring Group extension number) enables one to pick up the calls to any of the extensions in the Ring Group. Incoming calls can still be answered by dialing *<extension> for each number; however, using the group extension number does not require you to remember each extension in the group. This feature works even when a call comes directly to a single extension (e.g. 1003), or comes through the Auto Attendant.

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

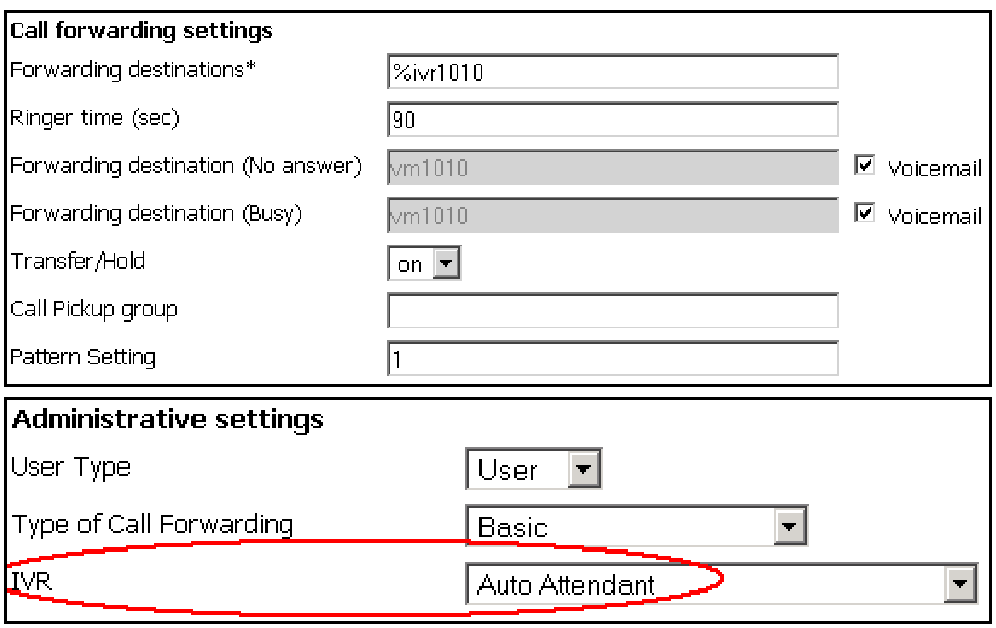
By setting up a Call Pickup Group number (generally a ring group number) in your extension, you only need to dial *(Star) to pickup the incoming calls to any of the extensions in the group. You may specify a Call Pickup Group at your extension’s [User > Edit] > [Forwarding destinations settings] >[Call Pickup group]



4.10. Setting Up Auto Attendant

The example below shows Auto Attendant for extension 1010.

1) Create an extension "1010"
2) Select [Auto Attendant] from [Administrative setting] > [IVR]. A message box, "Do you want to change the forwarding destination to IVR?" will appear.
3) Click "OK". Auto Attendant is automatically set for [Forwarding destinations*] field (ex., %ivr1010).
4) Click the [Save] button. The [Auto Attendant settings] fields will appear at the bottom of page.
5) Optionally, change the fields under [Auto Attendant settings] and an audio file for Auto Attendant greeting can be uploaded. For more detailed information, refer to section 5.3 "Users > Edit".



4.11. Setting Up Call Queuing (Pro Edition Only)

Here is how to set up Call Queue:

1) Select [Round robin/Top-down] from extension 1000's [User > Edit] > [Administrative settings] > [Type of Call Forwarding] field
2) Click the [Save] button. Queuing setup fields will be shown at [Call forwarding settings]
3) Set Call forwarding settings.

In this example, the call will be queued when all of the group members 1001-1005 are busy. If any member becomes available within the interval set in [Waiting time in the queue (sec)], Brekeke PBX will ring the available member. 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 1000's voicemail, vm1000.

Call forwarding settings		
Forwarding destinations*	1001, 1002, 1003, 1004, 1005	
Ringer time (sec)*	10	
Waiting time in the queue (sec)	300	
Max number of calls in the queue	10	
Call interval (msec)	3000	
Single attempt	no	
Busy/No Answer Forwarding	vm1000	☐ Voicemail
Mode	Round robin	
Transfer/Hold	on	
Pattern Setting	1	
Call Pickup group		

4.12. Setting Up Call Forwarding Schedule

This feature is useful for creating rules for call forwarding during specified times. In this example, incoming calls to extension 5555 during business hours are directed to the Auto Attendant at extension 1010. 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) To create an extension "5555",
2) Customize a wav file for "Call during business hours" prompt message. From the **[Voice Prompts]** menu, select "Voicemail personal greeting" and upload the customized wav file to extension 5555.
3) Select **[Schedule]** from extension 5555 [Users > Edit] >**[Administrative settings]** > **[Type of Call Forwarding]**.
4) Change [Voicemail setting] > [Greeting message] to Personal Greeting and save the settings, the Schedule setting fields will show.
5) Click "…" at **[Schedule settings 1]** > **[Schedule]**. A window will pop up. Select Monday through Friday and 18:00 - 8:59 there.
6) Click "…" at **[Schedule setting 2]** > **[Schedule]**. A window will pop up. Select Sunday, Saturday, 24 hours (00:00 – 00:00).
7) Enter vm5555 (Voicemail for 5555) at both **[Schedule setting 1]**> **[Forwarding destinations*]** and **[Schedule setting 2]** > **[Forwarding destinations*]**
8) In **[Default settings]**, enter 1010 (Auto attendant) at **[Forwarding destinations*]**

4.13. Setting Up Conference Call

4.13.1. 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) Create an extension "2000".
2) Select **[Conference]** at **[Administrative settings]** > **[Type of Call Forwarding].**
3) Leave all other settings as default

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

4.13.2. Limiting Members Who Can Enter the Conference Room

You can limit members that join the conference by specifying members (for example "1001,1002,1003") at **[Call forwarding settings]** > **[Applies to (Caller numbers)*]**. Doing so, only 1001, 1002, 1003 will be allowed to join the conference. With these settings, neither users 1004 and 1005, nor any other users, will be allowed to join the conference.

4.13.3. Simultaneous Calls to All of the Conference Members

A conference member can convene all members of the conference room at once. For example, set 1001, 1002, 1003 at **[Call forwarding settings]** > **[Forwarding destinations*]**. By dialing 2000, all conference members (1001, 1002, and 1003) will be invited simultaneously.

4.13.4. Starting a Conference Call (Alternate Methods)

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

4.14. Setting Up Switch Patterns (Pro Edition Only)

Switch Patterns can be used to temporarily and quickly change a user's forwarding destination by creating a switch pattern. In this example, incoming calls during business hours go directly to user 8000, however during user 8000's lunch break they prefer incoming calls to go directly to voicemail. During the lunch break, user 8000 sets his phone to DND (Do Not Disturb) mode. All calls during the lunch time will be forwarded to user 8000's voicemail. We can achieve this using a Switch Pattern as described below.

1) Create extension 8000
   At the extension 8000 [Users > Edit] page, set:
   [Forwarding destinations*] = 8000/vm8000
   [Pattern Setting] = 1
2) Create Switch Patterns extension (for example, extension 1).
3) Select "Switch patterns" under [Administrative settings] > [IVR].
4) Click "OK" at pop up window to confirm change the forwarding destination to IVR; %ivr1.will be added to field [Forwarding destinations*].
5) Click the [save] button, then [Switch patterns] field will appear at the bottom of the page
6) [Switch patterns] > [Pattern number]: 2 (this field is located under Switch patterns at the bottom of the screen)

4.14.1. Enter Do Not Disturb Mode

1) User 8000 can enter their DND (Do Not Disturb) mode by dialing "1" from his own phone.
2) Any calls to user 8000 during this time will be directed to voicemail inbox.
3) User 8000 can dial "1" again to remove DND mode.
4) After resuming from DND mode, any calls to user 8000 will be directed to his/her extension line.

✓ *In [user edit] page, [Pattern Setting] is set as 1 by default.*

4.15. ARS Settings

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

- **General**

  Use [General] fields to input the third party account information which is needed by the third party SIP services to accepting calls from Brekeke PBX.

  ✓ *From Brekeke PBX v2.2.7.7 and later, calls from non-registered users will be rejected by Brekeke PBX. You may find that calls from SIP devices and services cannot go through Brekeke PBX. You need to use [General] fields to register SIP devices and services at the bundled SIP server of Brekeke PBX, and with the dial plan named “To PBX From ITSP” for accepting the incoming calls from these SIP devices and services. Please check the following section for setup details.*

- **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.

- **Edit Variables**

Variables	Default value
v1	User ID/Number
v2	Password
v3	Customizable field
v4	Customizable field
v5	Customizable field
v6	Customizable field
v7	Customizable field
v8	Customizable field
v9	Customizable field

4.16. 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.

4.16.1. 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

4.16.2. ARS Rule Setup

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

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

Default Dial plan “To PBX From ITSP” is needed for the setups below:

Register URI	`sip:&v1@127.0.0.1`	**Realm**	`127.0.0.1`
Proxy Address	`127.0.0.1`	**Register Expire (sec)**	`3600`
User		**Register Update Period(%)**	`90`
Password		**Session interval (ms)**	
LineKey	`unchecked`		

4.16.3. Receiving PSTN Calls

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

Patterns - IN

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

Click [Edit Variables] link at the upper right corner of ARS rule page to set v1 and v3 value.

v1	v2(password)	v3
PSTN line number set at Section 4.16.1	(leave blank)	Specify a Brekeke PBX extension number

4.16.4. 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 rule as follows:

Patterns - OUT

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

In this example, we assume the VoIP Gateway's IP address is gw1_address. 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 rule 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 rule 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 > [Voice prompt] > [System].*

4.17. Connecting with Internet Telephony Service Providers (ITSPs)

4.17.1. 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.*

4.17.2. Setting ARS for ITSP using multiple accounts

General

Input third party account(s) information in the [General] fields of [ARS] rule.

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
Realm		Leave this field blank unless it is necessary
User	`&v1`	Set value at [Edit Variables] page
Password	`&v2`	Set value at [Edit Variables] page This field will be displayed in text format after saving.

Patterns

Patterns – IN settings define patterns for when a call is initiated and received through a third party SIP server.

In the pattern-IN example below, Brekeke PBX extension 100 (value of &v3, 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.

Patterns - IN

	Matching patterns	Deploy patterns
From		
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 rule Patterns – IN.*

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.

Patterns - OUT

	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 rule page

v1	v2(password)	v3
6504106636	6636	100 (extension)

4.18. ARS Outbound Route Failover (Pro Edition Only)

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 failover the session to an alternative route.

4.18.1. 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.

4.18.2. 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, 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: 4,000 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 (36,000,000 ms) in “ITSP_A”, the matching sessions will be routed through “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`

4.19. Setting Up Confirm Call (Version 2.2 or later)

1) Go to [Brekeke PBX]>[Voice Prompts]>[System] and upload an audio file for the "Confirm Call".

✓ *In this example, we’ll use prompt file named "confirmcall".*

2) Go to [Brekeke PBX]>[ARS] and create a new ARS rule for the "Confirm Call" or add "Confirm Call" to any existing ARS.

3) At "Patterns – OUT" in the ARS rule, set value to "Confirm" parameter under [Deploy patterns]. You need to enclose the value of the "Confirm" parameter with curly brackets, "{" and "}".

4) Specify the confirm key for "Confirm Call". The parameter "Key" next to the parameter "Confirm” specifies which key is used to confirm calls. The default key is 5.

✓ *If you wish to use the audio file "confirmcall" which you just uploaded in step 1, you should write {confirmcall} in the "Confirm" parameter.*

✓ *If set {confirmcall}{name:&f1} in [Confirm] parameter, 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:(.*)@*

4.20. Setting Up Paging (Version 2.2 or later)

4.20.1. Dial Plan

Add the following dial plan if your Brekeke PBX meets one of the following conditions:

- ♦ Your Brekeke PBX was upgraded to version 2.2 or later from an earlier version
- ♦ Your Brekeke PBX version is earlier than version 2.2.

Matching Patterns	Deploy Patterns
`$port = ^15062` `$localhost = ^true` `$request = ^INVITE` `X-PBX-Param = page` `$page.header = (.+),(.+)`	`$auth = false` `&net.sip.hide.loopback = true` `$b2bua = false` `%1 = %2` `X-PBX-Param =` `&net.sip.fixed.addrport.uas = true`

4.20.2. Setting Up Phones for Paging

A list of SIP phones that work with Brekeke PBX paging function and its sample configuration are available at Brekeke Wiki > [Interoperability] > [SIP Phones] > [How to set paging function on the phone side]

4.21. Setting Up Busy Lamp Field (BLF), Presence, and Shared Call Appearance (SCA) (Version 2.3 and later later)

Brekeke PBX v2.3 contains the following new features:

- ♦ 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, everyone can see the status of incoming lines and can select an available line to place a call or answer the incoming call.

- ♦ 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".

4.21.1. Dial Plan

The dial plan named “Subscribe” must be updated as shown below, if your Brekeke PBX was upgraded to version 2.3 or later from a previous version.

Matching Patterns	Deploy Patterns
`$request = ^SUBSCRIBE` `Event =` `^x-.*\|^message-summary$|^dialog$\|` `^line-seize$|^presence$\|^call-info$`	`$target = 127.0.0.1:15062` `$transport = udp` `$auth = false`

4.21.2. Setting Up Phones

A list of SIP phones that work with these function and its sample configuration are available at Brekeke Wiki > [Brekeke PBX FAQ] > [Software Detail] > [Functions] > [BLF, SCA, and Presence]

4.22. Setting UP IVR Script (Version 2.4 and later)

Feature IVR script is added since Brekeke PBX v2.4 and later. This feature requires Java 1.6 and upgraded Brekeke PBX license with Script option.

Check “Brekeke PBX IVR Script Developer’s Guide” for the detailed setup and methods of IVR Script.

5. Setup Items

5.1. Options Menu

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

5.1.1. Settings

♦ **General Settings**

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

♦ **SIP Settings**

Name	Default value	Description
SIP Proxy address [v]	127.0.0.1	Defines the IP Address or Hostname of the bundled SIP server that Brekeke PBX uses as a SIP Proxy.

♦ **Phone Number Settings**

Name	Default value	Description
IVR prefix [vi]	ivr	By using this prefix before an extension, a caller can reach that user's IVR. This prefix is defined at mediaserver_prefix (ARS). By default, dial "**06***" to access the user's IVR.
Voicemail prefix [vii]	vm	Using this prefix before an extension allows a caller to reach that user's voicemail inbox directly to leave a message. This prefix is defined at mediaserver_prefix (ARS). To directly access the user's voicemail inbox, dial "**07***".

[v] Do not change this value unless such setting is necessary.
[vi] Do not change this value unless such setting is necessary.
[vii] Do not change this value unless such setting is necessary.

Name	Default value	Description
Voicemail review/ Setting prefix viii	msg	Using this prefix before an extension allows access to that extension's voicemail inbox. To check messages, dial "08*"; the caller will be asked for a password before access is granted. This prefix is defined at mediaserver_prefix (ARS).
Max extension length ix	-1	The maximum digits that can be used as extension ID. Default = -1 means any length.
Call Pickup prefix	*	Dialing this prefix allows users to answer incoming calls directed to other call pickup group users. Prefix + ringing extension number Default = * + ringing extension number
Park number (min)	60	The minimum code number for retrieving parked calls. (The number is generated randomly by Brekeke PBX.) This setting is applicable only when keypad command is used to park the calls. Used with Park number (max) to define the range.
Park number (max)	89	The maximum code number for retrieving parked calls. (The number is generated randomly by Brekeke PBX.) This setting is applicable only when keypad command is used to park the calls. Used with Park number (min) to define the range.

♦ **PBX System Settings**

Name	Default value	Description
Port number	15060	The port number that Brekeke PBX will use. (Cannot be modified)
Max concurrent sessions	Depends on the edition	The maximum number of concurrent sessions that Brekeke PBX can handle. (Cannot be modified)
Max number of UAs (User Agents)	Depends on the edition	The maximum number of SIP UAs that Brekeke PBX can handle. (Cannot be modified)
Min Port	11000	Minimum port number the RTP Protocol uses for sending voice data.

[viii] Do not change this value unless such setting is necessary.
[ix] Do not change this value unless such setting is necessary.

Name	Default value	Description
Max Port	11999	Maximum port number the RTP uses for sending voice data.
RTP relay	on	on – RTP is handled by Brekeke PBX. off – RTP is not handled by Brekeke PBX. (Unless specified at the User settings or on the ARS.)
Codec priority	0	G.711 u-law (PCMU) is used by default. Use a comma (,) when specifying multiple payload. The following payload type can be used at the Brekeke PBX: 0 - G.711 u-law 8 - G.711 A-law 18 – G.729 98 – iLBC If Codec priority is not set in ARS or user setting, this setting will be used.
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 setting, this setting will be used.
Max concurrent recording sessions	10	Maximum concurrent sessions with call recording (Pro edition only)
Ringing Timeout (ms)	240000(ms)	Timeout value for awaiting an answer from the dialed party after ringing starts.
Talking Timeout (ms)	259200000 (ms)	The maximum length of time a call can last when SIP packets are not 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.

Name	Default value	Description
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 awaiting the next RTP packet after the system received the last one.
100rel	off	Enable (on) / Disable (off) on using reliable provisional responses (1xx series)
RFC2833	on	Enable (on) / Disable (off) RFC2833 setting Available since version 2.3
Valid client IP Pattern		web service security -- used by Brekeke PAL and Brekeke Web Service
Java VM arguments		Parameters to pass to VM

- **Media Server System Settings**

Name	Default value	Description
Port number	25060	The port number that Media server system uses. (This port number cannot be modified.)
Max concurrent session limit	Depends upon the license/edition	Maximum number of concurrent sessions for Media server. (The limit cannot be modified.)
Codec priority	0	G.711 u-law (PCMU) is use by default. Use a comma (,) when specifying multiple codecs. See also the description for [Codec priority] in PBX System Settings for the details.
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 inbox.
Message recording length (sec)	600(sec)	Maximum length of recording time for a voicemail message. If [Message recording length (sec)] in User setting is blank, this value will be used.

Name	Default value	Description
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 (sec)	Maximum recording length for each call. (Pro Edition only)
Min Port	12000	Minimum port number the RTP uses for sending voice data.
Max Port	12999	Maximum port number the RTP uses for sending voice data.
Ringing Timeout (ms)	240000(ms)	Timeout value for awaiting an answer from the dialed party after ringing starts.
Talking Timeout (ms)	259200000(ms)	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(ms)	Timeout value for awaiting the next RTP packet after the system received the last one.
Java VM Arguments		Parameters to pass 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

Name	Default value	Description
POP3 server		The address of the POP3 server. (for POP-before-SMTP authentication)
POP3 Port	110	POP3 server's listening port
User		Account user name for the above SMTP server.
Password		Password corresponding to the account user name
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. 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}: time length of the recorded message
Email body	from:{from} to:{to} time:{time} recording length(sec):{recording-length	Email body for the e-mail notifications. Same variables can be used as [Email subject] (See above).

5.1.2. Advanced

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

5.1.3. Notes

The menu item [Options] > [Notes] is used by Brekeke PBX plug-ins to access text data. 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	Access level Select from “No Access”, “Read only”, “Read/Write”
Note	Text field where you can write your own notes.

5.1.4. Auto Sync

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

5.2. 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.

✓ *This feature is available for both Basic and Pro Editions of Brekeke PBX.*

5.2.1. Adding a New Route

To add a new route:

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

5.2.2. Editing, Copying, or Deleting a Route

To edit/copy/delete a new route:

1) Select **[Settings]** from the submenu.
2) Choose edit , copy, or delete.

5.2.3. Viewing an Active Route

To view active ARS route:

1) Choose **[ARS]** menu.
2) Select **[Running Status]**.

If no ARS route is enabled, a list of ARS route will not be displayed under [Running Status]. Select [Settings] will display all ARS routes.

5.2.4. 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

Name	Default value	Description
Type	Type A	Used for special occasions
Group		ID for a group of ARS rules
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.
Realm [x]		Realm that is used for authentication.
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.
User		User ID for authentication account. Entry is not necessary when authentication is not used.
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.
Password		Password for authentication account. Entry is not necessary when authentication is not used.
Session interval (ms)		Set interval period between sessions for some VoIP FXO Gateways that require pausing between sessions.
LineKey	unchecked	Check if you use Line keys. (Optional feature)

[x] Do not change this value unless such setting is necessary.

- **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	Check/Uncheck to Request URI instead of To. Mostly design for using ITSP accounts.
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

Name		**Default value**	**Description**
Parameters	RTP relay	default	Select RTP relay ON/OFF. If “Default” is selected, [Option] menu > [RTP relay]. (Unless specified at the User settings) on – RTP is handled by PBX. 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: 0 - G.711 u-law 8 - G.711 A-law 18 – G.729 98 - iLBC
	Use Remote Preferred Codec	default	Enable (on) / Disable (off) on using remote codec used by the endpoints. “default”: remote codec setting in Option 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
	Remove SDP (18x) SDP 18x	default	Remove SDP or not when 180 Ringing or 183 Session Progress (18x response) from called party contains SDP. “default”: depended upon the situation of how sessions are established “block”: remove SDP “append”: attach SDP. If SDP is not included in packets, Ring-Back Tone will be played in RTP.

♦ **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 rule 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.

Name		Default value	Description
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 the confirm key entry
	Custom		Used for special occasions
Parameters	RTP relay	default	Select RTP relay ON/OFF. If “Default” is selected, [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.
	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 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 (Pro Edition only)

Name		Default value	Description
Parameters	Disable on registration failure	no	Enable (yes)/Disable (no) this Pattern when registration failed (Pro Edition only)
	Response timeout (ms)	-1	The period of time before timeout is activated when response has not been received (Pro Edition only)
	Error codes	500-599	Failover will be activated when specified error codes are received for INVITE requests (Pro Edition only)
	Recovery time (ms)	0	The period of time till this pattern will be reactivated (Pro Edition only)
	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. (Pro Edition only) Options: This route, This pattern, This group

5.3. Users > Edit

Click the menu **[Users]** on the menu bar, select a user, and click the edit button for that user to set the following items:

5.3.1. General Settings

Name	Default value	Description
Enabled	Yes	Enable/Disable the user Available since version 2.3
Descriptions		The description of this user. This field is optional.
Language	Ask your SA (System Administrator)[xi]	Selected Language will be used by the Brekeke PBX Admintool and as the default voicemail guidance prompt. Options: English/Japanese
Password	Ask your SA[xii]	Password for the Brekeke PBX Admintool login and accessing the user's voicemail inbox.
Password (confirm)	Ask your SA[xiii]	Reenter password for confirmation.

5.3.2. Call Forwarding Settings

[Basic] is selected under Type of Call Forwarding (Administrator)

Name	Default value	Description
Forwarding destinations*		Enter phone number(s) or SIP-URI to directly forward all calls that are received at this extension number. Multiple forwarding destinations can be specified by separating entries with a comma (,) delimiter.
Ringer time(sec)	90	The length of time that the user's phone will ring. After the length of time set here, the call will be transferred to the destination that is specified in [Forwarding destination (No answer/Busy).] If no destination is set at [Forwarding destination (No answer/Busy)], the call will be terminated.

[xi] These options will be set as the Administrator who created this user.
[xii] These options will be set as the Administrator who created this user.
[xiii] These options will be set as the Administrator who created this user.

Name	Default value	Description
Forwarding destination (No answer)	vm<extension>	Phone number to which the call will be forwarded when Ringer timeout has occurred.
Voicemail	Voicemail Prefix[xiv]	If this box is unchecked, then user can forward the call to another extension to which the field specify If this box is checked, then the call will be automatically forwarded to the user's voicemail box. Options: check/uncheck
Forwarding destination (Busy)	vm<extension>	Phone number to which the call will be forwarded when the called Phone number or SIP-URI is busy.
Voicemail	Voicemail Prefix[xv]	If this box is uncheck, then user can forward the call to another extension to which the field specify If this box is check, then the call will be automatically forwarded to the user's voicemail box. Options: check/uncheck
Keypad Commands	on	Enable/disable call features with DTMF commands Options: on/off
Call Pickup group		Enable one touch Call Pickup for the preset group extensions by assigning the group number.
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations*], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.

✓ ** This category allows multiple entries (divided with commas).*

[xiv] This value is set in the [Option] menu.
[xv] This value is set in the [Option] menu.

5.3.3. Call Forwarding Settings

[Round Robin/Top-down] is selected under Type of Call Forwarding

Name	Default value	Description
Forwarding destinations*		Enter phone number(s) or SIP-URI to directly forward all calls that are received at this extension number.
Ringer time (sec)*	20	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/Busy)] field. If no destination is set at [Forwarding destination (No answer/Busy)], the call will be terminated.
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/Busy)].
Max number of calls in the queue	10	The maximum number of calls 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/Busy)].
Busy/No Answer Forwarding	vm<extension>	Phone number to which the call will be forwarded when Ringer timeout has occurred.
Mode	Round-robin	There are two modes for call forwarding. 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. Top-down: Calls will always be distributed in the order listed in the field.
Keypad Commands	on	Enable/disable call features with DTMF commands Options: on/off

Name	Default value	Description
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations*], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.
Call Pickup group		Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.

✓ ** This category allows multiple entries (divided with commas).*

5.3.4. Call Forwarding Settings

[Conference] is selected under Type of Call Forwarding by the administrator

Name	Default value	Description
Forwarding destinations*		By specifying phone number(s) or SIP URI(s), a user can invite multiple users to this conference by dialing this extension.
Applies to (Caller numbers) *	*	Phone number(s) that can join this conference. 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.
Keypad Commands	on	Enable/disable call features with DTMF commands Options: on/off
Call Pickup group		Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.
Pattern Setting	1	Multiple calling patterns can be set by dividing [Forwarding destinations*], [Ringer time], [Forwarding destination (No answer/Busy)] settings by slash(es) "/". The pattern you would like to use will be set by specifying the numerical order of pattern from the left.

✓ ** This category allows multiple entries (divided with commas.)*

5.3.5. Call Forwarding Settings

[Schedule] is selected under Type of Call Forwarding by the administrator

Name	Default value	Description
Forwarding destinations*		Destination phone number(s) or SIP URI(s) to forward all calls that are received by this extension. Multiple entries can be specified by using a comma (,) delimiter. Calls will be forwarded to the appropriate destination based on conditions defined in the schedule.
Ringer time (sec)	90	The length of time that the callee's phone will ring. The appropriate Ringer time will be applied as the condition specified in **[Schedule]** or [**Applies to (Caller numbers)]** or **[Not Applies to (Caller numbers)]**.
Forwarding destination (No answer)	vm<extension>	Phone number to which the call will be forwarded when Ringer timeout has occurred.
Forwarding destination (Busy)	vm<extension>	Phone number to which the call will be forwarded when called Phone number or SIP-URI is busy..
Schedule		Specify schedule information by which to forward incoming calls.
Applies to (Caller numbers)*		Enter applicable Caller ID information for the schedule settings. Multiple Caller IDs can be specified by separating entries with a comma (,) delimiter. By adding a * (wildcard) after a number, you can specify all numbers that begin with that number. When the field is left blank, all numbers will be applied to the schedule.
Not Applies to (Caller numbers)*		Enter Caller ID information to be exempted from this setting. Multiple Caller ID numbers can be specified by separating entries with a comma (,) delimiter. By adding a * (wildcard) after a number, you can specify all numbers that begin with that number.

✓ ** This category allows multiple entries. (divided with commas)*

5.3.6. Voicemail Settings

Name	Default value	Description
Greeting message	Default system greeting	Select the greeting message for this user's voicemail. Options: Default system greeting /Personal greeting (user created)/Alternative greeting (user created).
Message recording length (sec) Greeting Only		Length of the time for recording a voice message. When the box is checked, caller cannot leave a voicemail message after greeting voice prompt, Options: check/uncheck
Message forwarding*		The extension number(s) to which received voicemail messages will be forwarded. Multiple numbers can be specified using a comma (,) delimiter.
Email address*		The e-mail address to which the notification of the arrival of voicemail messages is sent. Multiple addresses can be specified using a comma (,) delimiter.
Email notification	off	Enables/Disables email notification. Options: on/off
Attach WAV file to email	off	Enables attachment of voicemail messages in wav format to email notifications.

✓ ** This category allows multiple entries (divided with commas).*

5.3.7. Administrative Settings (SA)

Name	Default value	Description
User Type	User	Specifies the type of user. Options: User/Administrator
Type of Call Forwarding	Basic	Specifies the type of Call Forwarding. Options: Basic, Round robin/Top-down/, Schedule, Conference
IVR	None	Setting for the usage of the Interactive Voice Response (IVR) system. The following options are available: **Auto Attendant**: Calls will be answered by the Auto Attendant **Setup**: Enable mailbox management over IVR system. Note: For more details please refer to [set up menu] in voicemail navigation map **Add/Remove Forwarding Destinations**: By calling this extension, caller will be added/removed to/from the Forwarding Destinations of the users set in [Add/Remove Forwarding destinations*] > [Target users]. **(Pro Edition only)** **Switch Patterns**: Caller's [Pattern Setting] will be changed. **(Pro Edition only)** **None**: No IVR service

✓ *(SA)This menu is only available to the system administrator*

5.3.8. PBX Settings (SA)

Name	Default value	Description
RTP relay	default	Select RTP relay ON/OFF. If "Default" is selected, [Option] menu > [RTP relay], or [ARS] setting will be used. on – RTP is handled by Brekeke PBX. off – RTP is not handled by Brekeke PBX. (Unless specify at the User settings or on the ARS.)
Codec priority		Specify codec to be used. Use a comma (,) when specifying multiple codec.

Name	Default value	Description
Use Remote Preferred Codec	default	Enable (on) / Disable (off) on using remote codec used by the endpoints. “default”: RTP relay setting in ARS will be valid.
Call Recording	off	“on”: records all calls of this user. “off”: doesn’t record all calls of this user. You can stop or start recording when initiating a call or during a call. Please refer to Brekeke PBX User’s guide.**(Pro Edition only)**
Join other’s conversation	yes	Allow (yes) or not (no) this user to join other user’s conversation.
Allow others to join my conversation	yes	Allow other users join this user’s conversation.
Automatic Monitoring		Allow user to “monitor” another user’s conversation. For example, user 1000 wants to monitor user 1002 and “Automatic Monitoring*” is set on user 1002 Users > New/Edit by user 1000. If any other user extension calls user 1002 or receiving a call from user 1002, user 1000 will also ring. User 1000 can speaks and listens to the conversation that is between user 1002 and another user. **(Pro Edition only)** If a tilde (~) is placed before the extension, user 1000 cannot speaks during the conversation. If a caret (^) is placed before the extension, user 1000 cannot listens to the conversation. If both (~) and (^) are placed before the extension, user 1000 cannot speaks nor listens to the conversation.
Max sessions	unlimited	Specify the maximum received session numbers for the user.
Resource map		Map clients’ parameters to Brekeke PBX parameters

✓ *(SA)This menu is only available to the system administrators*

5.3.9. Auto Attendant Settings (SA) – [Administrative settings]-[IVR]

Name	Default value	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(sec)	The length of time that a user's phone will ring when a call is received via Auto Attendant.
Default operator		Default destination (phone number or SIP-URI) for an incoming call that has not specified a call recipient.
Speed dial*		Set up "Speed dial" for Auto Attendant. Example: 0=0001,5=sip:sales@brekeke.com. 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. Options: disable/enable

- ✓ ** This category allows multiple entries (divided with commas).*
- ✓ *(SA)This menu is only available to the system administrators.*

5.3.10. Add/Remove Forwarding Destinations (SA) - [Administrative settings]-[IVR] (Pro Edition only)

Name	Default value	Description
Target users*		By calling this extension, caller's extension number will be added/deleted from [Forwarding destinations*] of the users set in this [Target users] field.

- ✓ ** This category allows multiple entries (divided with commas).*
- ✓ *(SA)This menu is only available to the system administrators.*

5.3.11. Switch Patterns - [Administrative settings]-[IVR] (Pro Edition only)

Name	Default value	Description
Pattern Number	2	By calling this extension, caller's [Pattern Setting] value will be changed to the value set here

5.3.12. Script - [Administrative settings]-[IVR]

Name	Default value	Description
Note		Note name, which is created at [Options]>[Notes] with customized IVR script code
Function		Function name in the above Note
Parameter		Parameter value sent to above Note function (Optional depend on function definition)
Auto Answer	Yes	Enable auto answer or not

5.4. 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.

5.4.1. 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

5.4.2. 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.

5.5. Call Log

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, but 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 by logging in the admintool with the user.

5.6. Voice Prompts

5.6.1. User Voice Prompts

- Choose the [Voice Prompts] menu.
- A list of User, Description, and Type will display.
- Select () and a list of User Details for that user will display.

5.6.2. Types of Message Files

File Name	Description
Voicemail personal greeting	Personal voicemail greeting message the user has created.
Voicemail alternative greeting	Another voicemail greeting message the user has created.
Name	A message file that contains the user’s name. (For example, when you record a message for another user’s voicemail, the recipient will hear “There is a message from ‘name’”.)
Music on hold	An audio file that contains music/sound that will be used for music on hold.
Auto Attendant greeting message	Greeting message that is used for the Auto Attendant.
Auto Attendant retry message	A message to prompt the caller to reenter when an input error occurred.
Upload	To upload a file, select a file type from the pull-down list, and click the **[Browse]** button. Select the file you want to upload and click **[Upload]**. The upload will then start.

✓ *The number in the parentheses after the name of above field shows the file size. When no file exists for the type of message file, (none) is displayed.*

5.6.3. System Voice Prompts (SA)

Choose the [Voice Prompts] > [System] menu. 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
Description	A memo shown on the GUI for the file usage
Name	**Description**
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 the recorded sound files, click **Delete ( )**. The selected files will be deleted.

✓ *(SA)This menu is only available to the system administrators.*

5.6.4. 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.

5.7. Voice Mail

The following section describes how to set up voice mail for the extensions created by the System Administrator.

Follow these steps:

- Choose the [Voice Mail] menu.
- A list of User, Description, Type, and Messages will display.
- Select ( ) and a list of voice mail messages will display.
- If no voice mail message is displayed, that means no voice mail is available for that user.
- Select ( ) to play or download; From Brekeke PBX version 2.3, click on the [Date and Time of Call]

♦ Select ( ) to delete; From Brekeke PBX version 2.3, select the check box before each message or check the top box to select all recorded messages, and then click the [Delete] to delete the selected messages,

5.8. User Message

Name	Default value	Description
Date and Time of Call **Status** **Caller** **Size (Bytes)**	(Not visible)	If there are voicemail messages for this user, this item will appear on the screen. The type of message (new/saved), date and time the message was received, and file size will be displayed.
Download		To download the voicemail message as an audio file (WAV format), select the desired message from the pull-down list to download. The file will be downloaded to your PC.
Delete		To delete a voicemail message, select the desired message and click **[Delete]**. The message will be deleted from the voicemail inbox.

6. 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]>[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.*

7. Uninstall (Red Hat Linux OS)

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