

Brekeke PBX

Version 2.1

Administrator's Guide (Basic)

Brekeke Software, Inc.

Version

Brekeke PBX v2.1 Administrator's Guide (Basic)

Revised December 2007

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- 1. INTRODUCTION..... 6**
- 2. SYSTEM REQUIREMENTS 7**
- 3. INSTALLATION FOR WINDOWS OS..... 8**
 - 3.1. Step 1: Installing Java SE 8
 - 3.2. Step 2: Installing Brekeke PBX..... 8
 - 3.3. Step 3: Starting Brekeke PBX HTTP Service..... 8
 - 3.4. Step 4: Starting Brekeke PBX Administration Tool 8
- 4. INSTALLATION FOR RED HAT LINUX OS..... 10**
 - 4.1. Step 1: Installation of Java SE JDK 10
 - 4.2. Step 2: Installation of Tomcat..... 10
 - 4.3. Step 3: Installation of Brekeke PBX..... 10
 - 4.4. Step 4: Starting Tomcat..... 10
 - 4.5. Step 5: Starting Brekeke PBX Administration Tool 10
- 5. MAINTENANCE 12**
 - 5.1. Back Up / Restore..... 12
 - 5.2. Updating Brekeke PBX..... 12
 - 5.3. Activating your License 12
- 6. SETTING UP BREKEKE PBX 13**
 - 6.1. Example of Brekeke PBX Set Up..... 13
 - 6.1.1. Creating Users 13
 - 6.1.2. Default Values of Users..... 13

- 6.2. Voicemail Settings 14**
- 6.3. Voicemail Notification by Email 15**
 - 6.3.1. Setting Email Sender 15
 - 6.3.2. Setting up Email Recipient..... 16
- 6.4. Message Waiting Indicator (MWI)..... 16**
 - 6.4.1. PBX Settings for Message Waiting Indicator 16
 - 6.4.2. Phone Settings for Message Waiting Indicator 16
- 6.5. Setting Up Call Forwarding 17**
- 6.6. Setting up Ring Groups 17**
- 6.7. Setting Up No Answer Call Forwarding..... 18**
- 6.8. Setting Up Busy Call Forwarding..... 18**
- 6.9. Setting Up Call Pickup Group 19**
- 6.10. Setting Up Auto Attendant..... 20**
- 6.11. Setting Up Call Queuing (Pro Edition Only) 21**
- 6.12. Setting Up Call Forwarding Schedule 22**
- 6.13. Setting Up Conference Call 25**
 - 6.13.1. Creating a Conference Room 25
 - 6.13.2. Limiting Members Who Can Enter the Conference Room 25
 - 6.13.3. Simultaneous Calls to All of the Conference Members..... 25
 - 6.13.4. Starting a Conference Call (Alternate Method) 25
- 6.14. Setting Up Switch Patterns (Pro Edition Only) 26**
- 6.15. How to Enter Do Not Disturb Mode..... 27**
- 6.16. PSTN Access Using a VoIP Gateway 28**
 - 6.16.1. Receiving PSTN Calls..... 28
 - 6.16.2. Calling PSTN Numbers 28
- 6.17. Connecting with Internet Telephony Service Providers (ITSPs) 30**

- 6.17.1. Account Information for Third Party SIP Server..... 30
- 6.17.2. Setting ARS for ITSP using multiple accounts 30

- 6.18. ARS Outbound Route Failover (Pro Edition Only) 34**
- 6.18.1. Usage Examples 34
- 6.18.2. Setting Examples 34

- 7. SETUP ITEMS..... 36**

- 7.1. Option Menu..... 36**
- 7.1.1. General Settings 36
- 7.1.2. SIP Settings 36
- 7.1.3. Phone Number Settings..... 36
- 7.1.4. PBX System Settings 37
- 7.1.5. Media Server System Settings..... 38
- 7.1.6. Email Settings 40
- 7.1.7. Advanced 41
- 7.1.8. Notes..... 41

- 7.2. Call Status 42**
- 7.2.1. Status 42
- 7.2.2. User Agents..... 42

- 7.3. Automatic Route Selection (ARS)..... 43**
- 7.3.1. Adding a New Route 43
- 7.3.2. Editing, Copying, or Deleting a Route..... 43
- 7.3.3. Viewing an Active Route 43
- 7.3.4. General 44
- 7.3.5. Pattern - IN..... 45
- 7.3.6. Patterns - OUT 46

- 7.4. Call Log..... 49**

- 7.5. Users > New Edit 49**
- 7.5.1. General Settings 49
- 7.5.2. Call Forwarding Settings..... 50
- 7.5.3. Call Forwarding Settings..... 52
- 7.5.4. Call Forwarding Settings..... 54

- 7.5.5. Call Forwarding Settings..... 55
- 7.5.6. Voicemail Settings..... 55
- 7.5.7. Administrative Settings (SA) 56
- 7.5.8. PBX Settings (SA)..... 57
- 7.5.9. Auto Attendant Settings (SA) – [Administrative settings]-[IVR]..... 59
- 7.5.10. Add/Remove Forwarding Destinations (SA) - [Administrative settings]-[IVR] (Pro Edition only)..... 59
- 7.5.11. Switch Patterns - [Administrative settings]-[IVR] (Pro Edition only)..... 59

- 7.6. Voice Mail 60**
- 7.6.1. User Messages 60

- 7.7. Voice Prompts 61**
- 7.7.1. Types of Message Files 61

- 7.8. Notes for Sound Files..... 62**

- 8. UNINSTALL (WINDOWS)..... 63**

- 9. UNINSTALL (RED HAT LINUX)..... 63**

1. Introduction

An award-winning product, Brekeke PBX is a full-featured IP-PBX system easily managed through a web-based administrative interface. Brekeke PBX is feature rich and fully scalable to meet the needs of any size organization. Brekeke PBX solutions are flexible and affordable, from Basic with traditional telephony features, to Pro which includes a full set of Call Center features. Brekeke's SIP-based telephony products are highly compatible with most IP phones, gateways, and service providers.

The following is a brief list of some of the features included with Brekeke PBX:

- Auto Attendant
- ARS (Automatic Route Selection)
- ARS Outbound Route Failover (Pro Edition only)
- Call Forwarding
- Call Conference
- Call Monitoring
- Automatic Call Monitoring (Pro Edition only)
- Call Hunting
- Call Pickup
- Call Recording (Pro Edition only)
- Call Hold
- Call Transfer
- Call Queue (Pro Edition only)
- Ring Groups
- Voicemail, with Voicemail forwarding, and notification by email

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

2. System Requirements

Brekeke PBX runs on the following environments:

OS	Microsoft Windows, Red Hat Linux
Java	JDK 1.4 or newer. We recommend JDK 1.5.
Memory	512 MB minimum

✓ *You must install Apache Tomcat v4.1.2 or later if you use an OS other than Windows.*

3. Installation for Windows OS

This section is meant for those installing Brekeke PBX for the first time or for those who have uninstalled and are re-installing Brekeke PBX and using a Windows operating system. For those who are updating software to a newer version, please go to the section called Updating Brekeke PBX for instructions on how to update.

3.1. Step 1: Installing Java SE

You must install Java SE before installing the Brekeke PBX software.

- Access the website <http://java.sun.com/javase/downloads/index.jsp>,
- Download and install the appropriate version of JRE or JDK for the type of OS you are running.

3.2. Step 2: Installing Brekeke PBX

- Obtain installer file from Brekeke's website.
- Start the installer by double-clicking the file.
- Continue the installation by following the installer instructions.

Brekeke PBX and Brekeke SIP Server for PBX will be installed automatically. If you check [Run Brekeke PBX] at the last stage of the installation and click the [Finish] button, Brekeke PBX HTTP service will start automatically.

3.3. Step 3: Starting Brekeke PBX HTTP Service

If you did not check [Start Brekeke PBX] at the last stage of the installation, please start Brekeke PBX HTTP Service by the following method.

- Open [Control Panel]> [Performance and Maintenance]> [Administrative Tools]> [Services], then select and start [Brekeke PBX].
- Restart the machine. Brekeke PBX HTTP service will start automatically.

3.4. Step 4: Starting Brekeke PBX Administration Tool

- 1) Select [Start]> [Program]> [Brekeke]> [Brekeke PBX]> [Brekeke PBX Admintool].
- 2) You will be asked to enter Brekeke PBX Product ID. Enter the 16 digit product ID in the given space. Entering the same product ID on multiple machines is not allowed.
- 3) You will see the Login screen of Brekeke PBX Admintool.
- 4) Enter User ID and Password. Click the [Login] button.

5) Important Information**[Default] User ID & Password (Case sensitive)**

User	sa
Password	sa

- 6) To change the display language of the Admintool, please select [Users] > Click the Edit button of Admin user > Select desired language from [General setting]-[Language].
- 7) Click the menu [Start/Shutdown]. If you see [Active] for PBX Status, Media Server Status and Brekeke SIP Server Status, the Brekeke PBX server was started successfully. If you see [Inactive], the Brekeke PBX server failed to start.

✓ *You can install Brekeke PBX on Windows Operation Systems using the same method explained in next section.*

4. Installation for Red Hat Linux OS

This section is meant for those installing Brekeke PBX for the first time or for those who have uninstalled and are re-installing Brekeke PBX and using Red Hat Linux. For those who are updating software to a newer version, please go to the section called Updating Brekeke PBX for instructions on how to update.

4.1. Step 1: Installation of Java SE JDK

- Access the website <http://java.sun.com/javase/downloads/index.jsp>
- Download and install the appropriate version of JRE or JDK for the type of OS you are running.

4.2. Step 2: Installation of Tomcat

- Access the website <http://tomcat.apache.org/index.html> and download the binary file of Tomcat version 4.1.2 or later for the type of OS you are running.
- Set JRE or JDK Install directory for the environment variable JAVA_HOME.
- Install using the download file.

4.3. Step 3: Installation of Brekeke PBX

- Obtain the file pbx.war from Brekeke Software.
- Copy file directly into the webapps directory which is under the Tomcat install directory.

4.4. Step 4: Starting Tomcat

- Start Tomcat.
- 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.)
- Tomcat has started successfully if the Apache Jakarta Project page is displayed.

4.5. Step 5: Starting Brekeke PBX Administration Tool

- Open a web browser and 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.)
- You will be asked to enter Brekeke PBX Product ID. Enter the 16 digit product ID in the given space. Entering the same product ID on multiple machines is not allowed.
- You will see the Login screen of Brekeke PBX Admintool.
- Enter values for User ID and Password. Then click the [Login] button.

➤ **Important Information**

[Default] User ID & Password (Case sensitive)

User	sa
Password	sa

- If you wish to change the display language of the Admintool, please do the following: Select [Users] > Click the Edit button of Admin user> Select desired language from [General setting]-[Language]
- Click the menu [Start/Shutdown]. If you see [Active] for PBX Status, Media Server Status and Brekeke SIP Server Status, the Brekeke PBX server was started successfully. If you see [Inactive], the Brekeke PBX server failed to start.

5. Maintenance

5.1. 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].

Brekeke PBX should be inactive to do backup/ restore. The [Shutdown] button will be displayed while Brekeke PBX is active.

5.2. Updating Brekeke PBX

This section is for updating from a previous version of Brekeke PBX 2.x to a more recent release.

You should confirm that you have downloaded the update zip file from Brekeke's website before beginning these steps.

- 1) Choose the menu [Maintenance] > [Update Software].
- 2) If Brekeke PBX is active, [Shutdown] button is displayed. To continue with the update, click the [Shutdown] button.
- 3) Click the [browse] button to select the appropriate file (for example, pbx.war).
- 4) Click the [upload] button to upload the new file.
- 5) If your license is not activated, the licence activation window will be displayed.(Refer to the section "Activate License".
- 6) Restart your computer to reflect the changes.

5.3. Activating your License

- 1) Choose the menu [Maintenance] > [Activate License].
- 2) If Brekeke PBX is active, [Shutdown] button is displayed. To continue, click [Shutdown].
- 3) License Agreement will be shown. To continue, click [I agree] button.
- 4) If your computer is connected to the Internet, click [Activate] button. If not, click [Get Signature] and follow the instructions shown there.
- 5) After completing the activation successfully, you will see the Login screen of Brekeke PBX Admintool.

6. Setting Up Brekeke PBX

Because Brekeke PBX includes many defaults and is very flexible, your configuration should be based on the preferences of your business. However, there are some basic functions that many users will want to set up before proceeding with using Brekeke PBX.

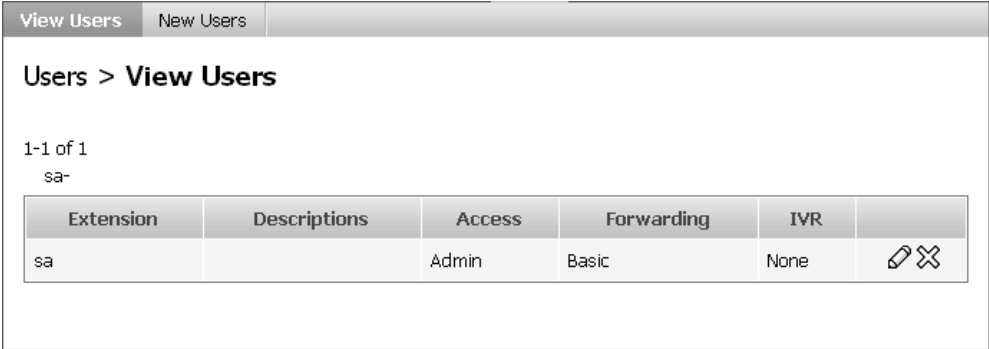
6.1. Example of Brekeke PBX Set Up

6.1.1. Creating Users

After installing Brekeke PBX, you will need to create users. This section explains how to set up extensions for your users. In our example, extensions 1001 through 1005 will be created.

Brekeke PBX Admintool > Users

- 1) Choose the **[Users]** menu and view on the **[View Users]** window.



- 2) Click **[New User]** submenu and enter "1001" (User ID) in a new window. A new user will be added to Brekeke PBX.
- 3) Continue adding other users for extensions 1002 to 1005.

6.1.2. Default Values of Users

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

Setting item	Details of default values
Language	Same language as the administrator
Password	Same password as the administrator
Ringer time (sec)	90 seconds
Forwarding destinations (No answer)	Forwarded to user's Voicemail ⁱ

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

Forwarding destination (Busy)	Forwarded to user's Voicemail ⁱⁱ
Call Pickup group	Same group as the administrator
Greeting message	Default system greeting
Email notification	No

✓ To change the default setting, click the Edit button to change the default settings.

6.2. Voicemail Settings

After creating user extensions, you can set up voicemail for each of the users. As described in section 6.1.2, 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 for 90 seconds after calling 1001 (or when 1001's line is busy), the call will be forwarded to voicemail.

The "vm" prefix which allows reaching the voicemail inbox directly is specified under the "mediaserver_prefix" route at the ARS setting. If your server is upgraded from the previous version v1.x, the setting at the "mediaserver_prefix" route is not necessary.

- To leave a voice message directly, dial "07*" (number where you want to reach voicemail)".
- 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 number)" to access directly to voicemail inbox.

ⁱⁱ Access number for users' voicemail is set in the **[Phone number setting]** menu under the **[Option]** menu. The default prefix for voicemail is set at vm. (e.g., vm1001 for extension 1001 user)

6.3. Voicemail Notification by Email

Users can receive notification of existing voicemails by receiving an email with an attached file.

6.3.1. Setting Email Sender

The current version of Brekeke PBX supports a mail server which does “POP before SMTP” authentication or “SMTP” authentication. To set up an email sender, enter the fields as shown below:

Brekeke PBX Admintool > Option > Settings

Email settings	
SMTP Server	smtp.example.com
SMTP Port	25
SMTP authentication	on <input type="checkbox"/>
POP3 Server	smtp.example.com
POP3 Port	110
User	johndoe
Password
Password (confirm)
Email address (from)	john@example.com
Email subject	voicemail({to}) : from {from}
Email body	to: {to} time: {time}

6.3.2. Setting up Email Recipient

To set up an email recipient for each user set the following fields at each user setting:

Brekeke PBX Admintool > Users > New/Edit

Voicemail settings	
Greeting message	Default system greeting ▾
Message forwarding*	<input type="text"/>
Email address*	<input type="text"/>
Email notification	off ▾
Attach WAV file to Email	off ▾

6.4. Message Waiting Indicator (MWI)

For those user agents that support a Message Waiting Indicator, you can set Brekeke PBX to support voicemail notification. The following section will help you set up your user agents to use MWI (Message Waiting Indicator).

6.4.1. PBX Settings for Message Waiting Indicator

To Enable Message Waiting Indicator, set:

Brekeke PBX Admintool > Options > Message Waiting Indicator = on

- ✓ *If your IP phone supports MWI features by "Subscribe" message, you do not need to set this option to "on".*

6.4.2. Phone Settings for Message Waiting Indicator

If your IP phone supports MWI with "Subscribe" message, the MWI feature will be activated even when the setting at section 5.1 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 phones can automatically call the SIP URI which is specified in NOTIFY packet (for MWI) to retrieve the message without assigning the number manually.

6.5. Setting Up Call Forwarding

Call Forwarding can be used for Brekeke PBX to support your office situations 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 to be forwarded to extension 1001.

Brekeke PBX Admintool > Users > New/Edit

- 1) Click the menu **[Users]** to open the **[View Users]** window.
- 2) Click the **Edit** button.
- 3) Enter the extension number you want to forward the call to, in this case extension "1001", in the entry field **[Forwarding destinations*]**.
- 4) Click the **[Save]** button to save changes.

6.6. Setting up Ring Groups

In this example, a Ring Group called Group 1000 is created for all extensions 1001 through 1005. When calls are received by any extension in the group, all extensions (1001-1005) ring simultaneously.

Brekeke PBX Admintool > Users

- 1) Choose the **[Users]** menu.
- 2) To create a Ring Group extension, click the **[New User]** submenu.
- 3) Enter *1000* in the entry field, User 1000 will be added.
- 4) Enter 1001, 1002, 1003, 1004, 1005 in the **[Forwarding destinations*]** field.
- 5) Click the **[Save]** button to save changes.

6.7. Setting Up No Answer Call Forwarding

To forward incoming calls to another extension instead of using Voicemail to answer calls that are not answered, set up the extension to forward to in the field **[Forwarding destination (No answer)]**. In the following example, calls are forwarded to 1000 (Ring group) if 1001 does not answer for 10 seconds.

The screenshot shows the 'Users > New/Edit' configuration page for user 1001. It is divided into 'General settings' and 'Call forwarding settings'. Under 'General settings', there are fields for 'Descriptions', 'Language' (set to English), 'Password', and 'Password (confirm)'. Under 'Call forwarding settings', there are fields for 'Forwarding destinations*', 'Ringer time (sec)' (set to 10), and 'Forwarding destination (No answer)' (set to 1000). The 'Forwarding destination (No answer)' field and its value '1000' are circled in red.

6.8. Setting Up Busy Call Forwarding

To forward incoming calls while a user is “busy”, instead of using Voicemail, set up the extension to forward to in the field, **[Forwarding destination (Busy)]**. In the following example, calls are forwarded to 1000 (Ring Group) if the phone of 1001 returns a 486 Busy response or other error response.

The screenshot shows the 'Users > New/Edit' configuration page for user 1001, identical in layout to the previous one. Under 'Call forwarding settings', the 'Forwarding destination (Busy)' field is set to 1000 and is circled in red.

6.9. 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 phone rings, dial * + extension to answer the call. For example, when extension 1001 rings, dialing * + 1001 will enable you to answer the call from any Brekeke PBX user's extension.

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

- ◆ *Using Call Pickup feature to answer calls that are directed to a Ring Group extension*
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 6.3 Ring Group Setup)

Extension: 1000

Ring Group	1001,1002,1003,1004,1005
-------------------	--------------------------

Dialing *1000 (Ring Group extension number) enables one to pick up any call that is directed 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*
Brekeke PBX lets you set up your Call Pickup Group number, so that you only need to dial *(Star) to pickup the incoming calls directed to any of the extensions that belong to your custom group number. You may specify a Call Pickup Group at **[Forwarding destinations settings]** in Brekeke PBX Admintool (shown below).

The screenshot shows the 'Call forwarding settings' interface. The 'Call Pickup group' field is highlighted with a red oval and contains the value '1000'. Other fields include 'Forwarding destinations*', 'Ringer time (sec)' (10), 'Forwarding destination (No answer)' (vm1001), 'Forwarding destination (Busy)', 'Transfer/Hold' (on), and 'Pattern Setting' (1). There are also checkboxes for 'Voicemail'.

6.10. Setting Up Auto Attendant

The example below shows Auto Attendant answering calls that come into extension 1010.

Brekeke PBX Admintool > Users

- 1) Choose the **[Users]** menu.
- 2) To create an Auto Attendant extension, click **[New User]** submenu and enter 1010 in the entry field User 1010 is added, and User Edit screen is displayed.
- 3) Select **[Auto Attendant]** from the list of **[IVR]** in the **[Administrative setting]**. After selecting IVR, a message box, "Do you want to change the forwarding destination to IVR?" will appear. When you click "OK", Auto Attendant is set for [Forwarding destinations*] field (ex., %ivr1010).
- 4) Click the **[Save]** button to save changes.
- 5) Depending on your needs, the items inside **[Auto Attendant setting]** can be updated and an audio file for Auto Attendant greeting can be uploaded. For more detailed information, refer to 7.4 User Setting.


Call forwarding settings	
Forwarding destinations*	<input type="text" value="%ivr1010"/>
Ringer time (sec)	<input type="text" value="90"/>
Forwarding destination (No answer)	<input type="text" value="vm1010"/> <input checked="" type="checkbox"/> Voicemail
Forwarding destination (Busy)	<input type="text" value="vm1010"/> <input checked="" type="checkbox"/> Voicemail
Transfer/Hold	<input type="text" value="on"/>
Call Pickup group	<input type="text"/>
Pattern Setting	<input type="text" value="1"/>

Administrative settings	
User Type	<input type="text" value="User"/>
Type of Call Forwarding	<input type="text" value="Basic"/>
IVR	<input type="text" value="Auto Attendant"/>

6.11. Setting Up Call Queuing (Pro Edition Only)

Call Queuing is a useful feature to help avoid missing calls if all lines are busy. For example, if an incoming call comes to the Ring Group 1000, and all extensions 1001-1005 are busy, the call can be put into a Call Queue where the caller will hold until someone is available to answer.

Brekeke PBX Admintool > Users > Edit

- 1) Click the menu **[Users]** to open the **[Users]** window.
- 2) Click the Edit button () for user 1000 and open the edit window for the extension.
- 3) Select **[Round robin/Top-down]** from the list of **[Type of Call Forwarding]** in the **[Administrative settings]**.
- 4) Click the **[Save]** button
- 5) Set Call forwarding settings.

In this example, the call is put in a queue while all of the group members 1001-1005 are busy. If any member becomes available within the time 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 period at **[Waiting time in the queue (sec)]**, the call will be forwarded to the destination set in **[Forwarding destination (No answer)]** (user 1000's voicemail vm1000 in this example).

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

6.12. Setting Up Call Forwarding Schedule

Call Forward Scheduling is a tool that is useful for creating rules for call forwarding during specified times. The example being used is during business hours that incoming calls are directed to extension 5555 first then 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.” In this scenario, the gateway setting needs to be changed to direct incoming calls to extension 5555.

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

Brekeke PBX Admintool > Users

- 1) Choose the **[Users]** menu.
- 2) To create an extension 5555, click on the **[New User]** and enter 5555 in the new entry field. User 5555 will be added and the edit page for user 5555 will open.
- 3) Select **[Schedule]** from the list of **[Type of Call Forwarding]** in the **[Administrative settings]**
- 4) Click the **[Save]** button
- 5) Customize a wav file for the message “Please call during business hours”. Click the **[Voice Prompts]** menu and select “Voicemail personal greeting” to upload the wav file.
- 6) Underneath **[Upload]**, select “Voicemail personal greeting” at the drop-down list, click on the **[Browse]** button to find the customized file, and click on the **[Upload]** button.
- 7) Click “...” at **[Schedule settings 1] > [Schedule]**. A window will pop up. Select Monday through Friday and 18:00 - 8:59 there.

Term -

Days of week
 First Second Third Fourth Fifth
 Sun Mon Tue Wed Thu Fri Sat

Days
 DD | DD-DD | MM/DD | MM/DD-MM/DD

 This field allows multiple entries. (separated by commas)

Time
 : - :
 : - :
 : - :

- 8) Click “...” at **[Schedule setting 2] > [Schedule]**. A window will pop up. Select Sunday, Saturday, 24 hours (00:00 – 00:00).

Term -

Days of week
 First Second Third Fourth Fifth
 Sun Mon Tue Wed Thu Fri Sat

Days
 DD | DD-DD | MM/DD | MM/DD-MM/DD

 This field allows multiple entries. (separated by commas)

Time
 : - :
 : - :
 : - :

- 9) Enter vm5555 (Voicemail for 5555) at both **[Schedule setting 1] > [Forwarding destinations*]** and **[Schedule setting 2] > [Forwarding destinations*]**
- 10) In **[Default settings]**, enter 1010 (Auto attendant) at **[Forwarding destinations*]** and click **[Save]**

Default settings	
Forwarding destinations*	<input type="text" value="1010"/>
Ringer time (sec)	<input type="text" value="90"/>
Forwarding destination (No answer)	<input type="text"/>
Forwarding destination (Busy)	<input type="text"/>
Transfer/Hold	<input type="text" value="on"/>
Call Pickup group	<input type="text"/>
Pattern Setting	<input type="text" value="1"/>

6.13. Setting Up Conference Call

The Conference Call feature of Brekeke PBX can increase efficiency and communication for your business. Using Brekeke PBX you can create conference rooms and manage the calls with multiple options. In the following example, user 2000 is set up as the conference number.

6.13.1. Creating a Conference Room

The first step to using the Conference Call feature is to set up a Conference Room.

Brekeke PBX Admintool > Users

- 1) Choose the **[Users]** menu.
- 2) To create extension 2000, click **[New User]** submenu and enter 2000 in the new text entry field. User 2000 is created and the edit page opens.
- 3) Select **[Conference]** at **[Administrative settings] > [Type of Call Forwarding]**.
- 4) Click the **[Save]** button.

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

6.13.2. Limiting Members Who Can Enter the Conference Room

You can limit members that join the conference by specifying members (for example 1001, 1002, and 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.

6.13.3. 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.

6.13.4. Starting a Conference Call (Alternate Method)

Additional methods for starting a conference call are included in the Brekeke PBX User Guide.

6.14. 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 set their 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.

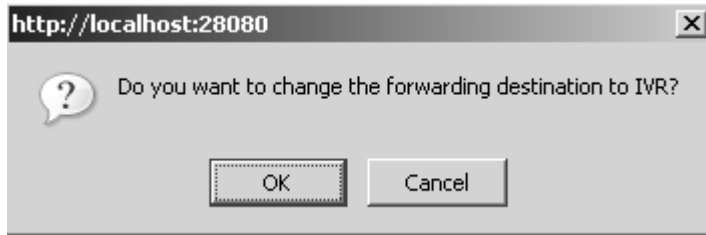
Call forwarding settings	
Forwarding destinations*	8000/vm8000
Ringer time (sec)	90
Forwarding destination (No answer)	vm8000 <input checked="" type="checkbox"/> Voicemail
Forwarding destination (Busy)	vm8000 <input checked="" type="checkbox"/> Voicemail
Transfer/Hold	on ▼
Call Pickup group	
Pattern Setting	1

On the Users > New/Edit for User 8000, create the following:

- 1) **[Forwarding destinations*]** = 8000/vm8000.
- 2) **[Pattern Setting]** = 1.
- 3) Click the [save] to save current settings.

Administrative settings	
User Type	User ▼
Type of Call Forwarding	Basic ▼
IVR	Switch patterns ▼

- 4) Click on the **[New User]** submenu and create another new extension user (for example, user 1).
- 5) Select "Switch patterns" under Administrative settings > IVR.
A window prompt will appear.



- 6) Click "OK".
- 7) [Forwarding destinations*] : %ivr1.
- 8) Click the [save] button.

The below field will appear after saving is entered.

- 9) [Pattern number]: 2 (this field is located under **Switch patterns at the bottom of the screen**)

6.15. How to Enter Do Not Disturb Mode

- 1) User 8000 can enter their DND (Do Not Disturb) mode by dialing "1" from his/her 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.

Important Note: Now that Pattern Setting 1, a switch pattern, is created, any user can use it.

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. Receiving PSTN Calls

To receive a PSTN call at a Brekeke PBX extension, you do not need any special settings on Brekeke PBX. Set the following at your VoIP Gateway:

- SIP proxy address
Specify the IP address of Brekeke SIP Server
- An SIP URI which PSTN calls are directed to
Specify a Brekeke PBX user name (For the example in the section 6.9, 5555).

6.16.2. 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:

Navigate to Brekeke PBX Admintool > ARS > Settings

- *General*
Changing these settings is not required here unless you use authorization for connecting to a PSTN Gateway. Many PSTN Gateways have a short interval between sessions when the line becomes unavailable. With this example we have added 2000 milliseconds for session intervals.

ARS > Route Template			
General Edit Variables			
Route name	gw1		
Description	Sample settings for 4-port FXO gateway		
Disabled	<input checked="" type="checkbox"/>		
Register URI		Realm	
Proxy Address		Register Expire (sec)	3600
User		Register Update Period (%)	90
Password	XXXXXXXXXX	Session interval (ms)	2000

- *Patterns*

Create a pattern OUT as follows:

OUT - 1	Matching patterns	Deploy patterns
Priorities <input type="text" value="100"/>	From <input type="text"/>	From <input type="text"/>
Max Sessions <input type="text" value="4"/>	To <input type="text" value="sip:9(.{6,})@"/>	To <input type="text" value="sip:\$1@gw1_address"/>
<input type="checkbox"/> Disabled	User <input type="text"/>	DTMF <input type="text"/>
<input type="button" value="Copy"/> <input type="button" value="Delete"/>		Target <input type="text"/>
Parameters		
RTP relay	<input type="text" value="default"/>	Codec Priority <input type="text"/>
Block SIP INFO (DTMF)	<input type="text" value="no"/>	Send RTCP <input type="text" value="off"/>
Session Timer(sec, 0=disable)	<input type="text" value="1800"/>	100rel <input type="text" value="off"/>
Next route on failure	<input type="text" value="no"/>	Disable on registration failure <input type="text" value="no"/>
Response timeout (ms)	<input type="text" value="50000"/>	Error codes <input type="text"/>
Recovery time (ms)	<input type="text" value="0"/>	Disable all OUT patterns on failure <input type="text" value="yes"/>

In this example, we assume the VoIP Gateway's IP address is gw1_address. A Brekeke PBX user dials 9-prefix and a PSTN number for calling out to the PSTN. For example, to make a call to 401-6636, dial "94016636".

It is useful to set priorities when there are multiple options for making calls, such as when you have multiple VoIP Gateways or when you subscribe to multiple VoIP service providers. The maximum session number can be set in the **[Max Sessions]** field. Priority can be defined in the **[Priorities]** field. Please note, lower numbers hold higher priorities.

- Two Stage Dialing

If your VoIP Gateway supports Two Stage Dialing, have the gateway's PSTN port register with Brekeke SIP Server. Let us suppose the gateway's PSTN port has the SIP user name, 111. To call a PSTN number, configure setting at OUT pattern at ARS settings so that the dialed numbers will be sent to gateway as DTMF tones.

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
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

Brekeke PBX Admintool > ARS > Settings

Input third party account(s) information in the **[General]** section of **[ARS]**. "Register URI" must be filled in using the SIP URI format (sip:user@domain.com)."Proxy Address" can be omitted if the address is the same as the one in "Register URI". "Realm" can be left blank if the realm setting is not necessary. "User" can be filled on the Edit Variables page. "Password" can be change to clear text when registering multiple accounts under one provider.

General		Edit Variables	
Route name	sample_proxy		
Description			
Disabled	<input type="checkbox"/>		
Register URI	sip:&v1@sample.proxy.com	Realm	
Proxy Address	sample.proxy.com	Register Expire (sec)	3600
User	&v1	Register Update Period (%)	90
Password	*****	Session interval (ms)	

- **[Password]**: enter "&v2".
- Click the [save] button.

General Edit Variables			
Route name	sample_proxy		
Description			
Disabled	<input type="checkbox"/>		
Register URI	sip:&v1@sample.proxy.com	Realm	
Proxy Address	sample.proxy.com	Register Expire (sec)	3600
User	&v1	Register Update Period (%)	90
Password	&v2	Session interval (ms)	
LineKey	<input type="checkbox"/>		

Once the setting is saved, "&v2" in text will appear.

General Edit Variables			
Route name	sample_proxy		
Description			
Disabled	<input type="checkbox"/>		
Register URI	sip:&v1@sample.proxy.com	Realm	
Proxy Address	sample.proxy.com	Register Expire (sec)	3600
User	&v1	Register Update Period (%)	90
Password	&v2	Session interval (ms)	

Click the "Edit Variables" link as indicates.

ARS > Route Variables Edit T								
Route name : sample_proxy								
v1=user ID/Number v2=password v3=Incoming call number v4=PBX user								
v1	v2(password)	v3	v4	v5	v6	v7	v8	v9
6504106636	*****	100						

Once the "Edit Variable" link is selected, this page opens.

- [v1]: Enter the user ID/Number that you obtain from ITSP (for example, 6504106636).
- [v2(password)]: Enter the password that you obtain from ITSP (for example, 6636).
- [v3]: Incoming call number (for example, 100).

- Click the [Save and Update] button.

Variable Names

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

- *Patterns*
Use these settings to define patterns for when a call is initiated and received through a third party SIP Server. In the example pattern [IN] below, extension 100 (&v3) 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. Settings for the “OUT” pattern define patterns for converting SIP URI to match your VoIP provider’s header format requirements. In the example below, dialed numbers beginning with 9 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.

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

Patterns - IN [New]

IN - 1	Matching patterns	Deploy patterns
Priorities <input type="text" value="100"/>	From <input type="text"/>	From <input type="text"/>
Max Sessions <input type="text" value="-1"/>	To <input type="text" value="sip:&v1@"/>	To <input type="text" value="&v3"/>
<input type="checkbox"/> Disabled	<input type="checkbox"/> Apply to Request URI instead of To	
<input type="button" value="Copy"/> <input type="button" value="Delete"/>	Parameters	
	RTP relay <input type="text" value="default"/>	Codec Priority <input type="text"/>
	Use Remote Preferred Codec <input type="text" value="default"/>	Block SIP INFO (DTMF) <input type="text" value="no"/>
	Send RTCP <input type="text" value="off"/>	SDP 18x <input type="text" value="default"/>

Patterns - OUT New

OUT - 1	Matching patterns	Deploy patterns
Priorities <input type="text" value="100"/>	From <input type="text"/>	From <input <sip:&v1@sample.proxy.com>"="" type="text" value="&v1"/>
Max Sessions <input type="text" value="-1"/>	To <input type="text" value="sip:9(+.)@"/>	To <input type="text" value="sip:\$1@sample.proxy.com "/>
<input type="checkbox"/> Disabled	User <input type="text" value="&v4"/>	DTMF <input type="text"/>
<input type="button" value="Copy"/> <input type="button" value="Delete"/>		Target <input type="text"/>
Parameters		
RTP relay	<input type="text" value="default"/>	Codec Priority <input type="text"/>
Block SIP INFO (DTMF)	<input type="text" value="no"/>	Send RTCP <input type="text" value="off"/>
Session Timer(sec, 0=disable)	<input type="text" value="1800"/>	100rel <input type="text" value="off"/>
Next route on failure	<input type="text" value="no"/>	Disable on registration failure <input type="text" value="no"/>
Response timeout (ms)	<input type="text" value="-1"/>	Error codes <input type="text" value="500-599"/>
Recovery time (ms)	<input type="text" value="0"/>	Disable all OUT patterns on failure <input type="text" value="yes"/>

- **Priorities**
It is useful to set priorities when there are multiple options for making calls, such as when you have multiple PSTN Gateways or when you subscribe to multiple VoIP service providers. Maximum session number (such as port numbers of Gateways or subscribed line numbers of VoIP services) can be set in the **[Max Sessions]** field. Setting “-1” specifies an unlimited number of sessions. Priority can be defined in the **[Priorities]** field. Lower numbers hold the higher priorities

6.18. 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.

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.

Route name: ITSP_A Patterns (OUT)

Patterns - OUT		New	
OUT - 1	Matching patterns	Deploy patterns	
Priorities <input type="text" value="1"/>	From <input type="text"/>	From <input sip:xxx@itsp_a.domain>"="" type="text" value="\" xxx\"<=""/>	
Max Sessions <input type="text" value="4"/>	To <input type="text" value="sip:9{[0-9]{7,10}}@"/>	To <input type="text" value="sip:\$1@itsp_a.domain"/>	
<input type="checkbox"/> Disabled	User <input type="text"/>	DTMF <input type="text"/>	
<input type="button" value="Copy"/> <input type="button" value="Delete"/>		Target <input type="text"/>	
Parameters			
RTP relay	<input type="text" value="on"/>	Codec Priority	<input type="text"/>
Block SIP INFO (DTMF)	<input type="text" value="yes"/>	Send RTCP	<input type="text" value="off"/>
Session Timer(sec, 0=disable)	<input type="text" value="1800"/>	100rel	<input type="text" value="off"/>
Next route on failure	<input type="text" value="yes"/>	Disable on registration failure	<input type="text" value="yes"/>
Response timeout (ms)	<input type="text" value="4000"/>	Error codes	<input type="text" value="500-599"/>
Recovery time (ms)	<input type="text" value="3600000"/>	Disable all OUT patterns on failure	<input type="text" value="no"/>

Route name: MyGateway Patterns (OUT)

OUT - 1		Matching patterns	Deploy patterns
Priorities	100	From	From
Max Sessions	4	To	To sip:\$1@192.168.0.111
<input type="checkbox"/> Disabled		User	DTMF
<input type="button" value="Copy"/>	<input type="button" value="Delete"/>		Target
Parameters			
RTP relay	default	Codec Priority	
Block SIP INFO (DTMF)	no	Send RTCP	off
Session Timer(sec, 0=disable)	1800	100rel	off
Next route on failure	no	Disable on registration failure	no
Response timeout (ms)	-1	Error codes	500-599
Recovery time (ms)	0	Disable all OUT patterns on failure	yes

- ◆ Under regular operation, the route with highest priority “ITSP_A” will be used for outbound sessions. If there is no response (longer than 4 seconds) for INVITE messages or “500-599” response was received, Brekeke PBX will continue route searching to meet 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), the session will be routed through “MyGateway” for one hour after the failover. If “ITSP_A” is back on running, the session will be routed through the highest priority route, “ITSP_A”.
- ◆ 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.
- ◆ Setting under Parameters, Disable on registration failure is set for “yes” as default setting. When registration is not working property at “ITSP-A” route, it will be disabled and “MyGateway” route will be used instead.

7. Setup Items

7.1. Option Menu

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

7.1.1. 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

7.1.2. SIP Settings

Name	Default value	Description
SIP Proxy address	127.0.0.1	Defines the IP Address or Hostname of the SIP Server the Brekeke PBX uses as a SIP Proxy.

7.1.3. Phone Number Settings

Name	Default value	Description
IVR prefix	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 to user's IVR.
Voicemail prefix	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 access directly to user's voicemail inbox, dial "07*".
Voicemail review/ Setting prefix	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).

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 assigned number for retrieving parked calls. (The number is assigned randomly.)
Park number (max)	89	The maximum assigned number for retrieving parked calls. (The number is assigned randomly.)

7.1.4. 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 user agents	Depends on the edition	The maximum number of user agents (SIP UAs) that PBX can handle. (Cannot be modified)
Min Port	11000	Minimum port number the RTP Protocol uses for sending voice data.
Max Port	11999	Maximum port number the RTP uses for sending voice data.
RTP relay	on	on – RTP is handled by PBX. off – RTP is not handled by PBX. (Unless specified at the User settings or on the ARS.)
Codec priority	0	G.711 ulaw (PCMU) is used by default. Use a comma (,) when specifying multiple payload. The following payload type can be used at the 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	no	Use codec setting that is preferred at the remote SIP

Preferred Codec		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)	120000(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.
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)
Valid client IP Pattern	Blank	web service security -- future use
Java VM arguments	Blank	Parameters to pass to VM

7.1.5. 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	Depends upon the	Maximum number of concurrent sessions for Media

session limit	license/edition	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 (Section 7.1.4).
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 seting is blank, this value will be used.
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.
Ringng Timeout (ms)	120000(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	Blank	Parameters to pass to VM

✓ ms = 0.001 second

7.1.6. Email Settings

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

	length(sec):{recording-length h	
--	------------------------------------	--

7.1.7. Advanced

Select the menu [Options] > [Advanced]. You can 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.

7.1.8. Notes

Select the menu [Options] > [Notes]. PBX plug-ins can use the text in the notes. You can use this just 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.

7.2. 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.

7.2.1. 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

7.2.2. 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.)

7.3. 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.

7.3.1. 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.

7.3.2. 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 (✖).

7.3.3. Viewing an Active Route

To view active ARS route:

- Choose **[ARS]** menu.
- Select **[Running Status]**.

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

7.3.4. General

Name	Default value	Description
Route name	Blank	The name for the route
Description	Blank	The description for the route
Disabled	OFF	Disable this setting

Name	Default value	Description
Register URI	Blank	SIP URI that is used to register Brekeke PBX with a registrar server. Leave this blank when there is no need to register Brekeke PBX to other registrar server.
Realm	Blank	Realm that is used for authentication. This field is optional.
Proxy Address	Blank	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	The length of time till REGISTER expires.
User	Blank	User ID for authentication account. This field is optional field when authentication is not being used.
Register Update Period (%)		The percentage value of the interval until re-register occurs is calculated from the length specified in the Register Expire setting.
Password	Blank	Password for authentication account. This field is optional when authentication is not being used.
Session interval (ms)	Blank	Set interval period between sessions for some VoIP FXO Gateways that require pausing between sessions.
Line key	uncheck	Check if you use Line keys. (Optional feature)

7.3.5. Pattern - IN

Name		Default value	Description
Priorities		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		Off	Disable this pattern
Matching patterns	From	Blank	Specify a matching rule for From header using regular expressions. When the field is left blank, all calls will be considered as matched.
	To	Blank	Specify a matching rule for To header using regular expressions. When the field is left blank, all calls will be considered as matched calls.
	Plugin	Blank	The java class name for the plug-in
	Param	Blank	The parameters which will be used by the plug-in
	Return	Blank	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	Blank	Specify replace patterns for From header using regular expressions.
	To	Blank	Specify replace patterns for To header using regular expressions.
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	Blank	Specify codec to be used. Use a comma (,) when specifying multiple payload. The following payload type can be used at the 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	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
	Remove SDP (18x)	no	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.

7.3.6. Patterns - OUT

Name	Default value	Description
Priorities	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		Off	Disable this pattern
Matching patterns	From	Blank	Specify a matching rule for From header using regular expressions. When the field is left blank, all calls will be considered as matched.
	To	Blank	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	Blank	Mostly designed for multiple ITSP accounts. It is used for specifying which account this ARS rule is applying.
	Plugin	Blank	The java class name for the plug-in
	Param	Blank	The parameters which will be used by the plug-in
	Return	Blank	The pattern of the value returned by the plug-in
Deploy patterns	From	Blank	Specify replace patterns for From header using regular expressions.
	To	Blank	Specify replace patterns for To header using regular expressions.
	DTMF	Blank	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.
	Target	Blank	Destination IP address. May omit entry when the destination IP address is specified in To header domain.
Parameters	RTP relay	Default	Select RTP relay ON/OFF. If "Default" is selected, [Option] menu > [RTP relay] (Unless specified at the User settings). "on" – on – RTP is handled by PBX. off – RTP is not handled by PBX.
	Codec Priority	Blank	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)	1800	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)
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 all OUT patterns on failure	yes	Disable all OUT patterns when the route using this OUT pattern failed. (Pro Edition only)

7.4. 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.

7.5. Users > New Edit

Brekeke PBX Admintool > View 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:

7.5.1. General Settings

Name	Default value	Description
Name	Blank	The description of this user. This field is optional.
Language	Ask your SA (System Administrator) ⁱⁱⁱ	Selected Language will be used by the Brekeke PBX Admintool and as the default voicemail guidance prompt. Options: English/Japanese
Password	Ask your SA ^{iv}	Password for the Brekeke PBX Admintool login and accessing the user's voicemail inbox.
Password (confirm)	Ask your SA ^v	Reenter password for confirmation.

ⁱⁱⁱ These options will be set as the Administrator who created this user.

^{iv} These options will be set as the Administrator who created this user.

^v These options will be set as the Administrator who created this user.

7.5.2. Call Forwarding Settings

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

Name	Default value	Description
Forwarding destinations*	Blank	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.
Forwarding destination (No answer) Voicemail	vm + extension # Voicemail Prefix ^{vi}	Phone number or SIP-URI to which the call will be forwarded when Ringer timeout has occurred. 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) Voicemail	vm + extension # Voicemail Prefix ^{vii}	Phone number or SIP-URI to which the call will be forwarded when the called Phone number or SIP-URI is busy. 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
Transfer / Hold	on	Enable/disable this user to use call transfer/hold features. Options: on/off

^{vi} This value is set in the [Option] menu.

^{vii} This value is set in the [Option] menu.

Call Pickup group	Blank	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).

7.5.3. Call Forwarding Settings

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

Name	Default value	Description
Forwarding destinations*	Blank	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 or SIP-URI 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.
Transfer/Hold	on	Enable/disable this user to use call transfer/hold features.

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	Blank	Enables one touch Call Pickup by assigning the user to a preset Call Pickup group.

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

7.5.4. Call Forwarding Settings

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

Name	Default value	Description
Forwarding destinations*	Blank	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 or SIP-URI to which the call will be forwarded when Ringer timeout has occurred.
Forwarding destination (Busy)	vm + extension #	Phone number or SIP-URI to which the call will be forwarded when called Phone number or SIP-URI is busy.
Schedule	Blank	Specify schedule information by which to forward incoming calls.
Applies to (Caller numbers)*	Blank	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)*	Blank	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)

7.5.5. Call Forwarding Settings

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

Name	Default value	Description
Forwarding destinations*	Blank	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. An asterisk (*) and a questions mark (?) can be used for matching metacharacters. An asterisk (*) means zero (0) or more characters and a question mark (?) means one character.
Transfer/Hold	on	Grants the user permission to use transfer and hold functions. Options: on/off
Call Pickup group	Blank	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.)

7.5.6. 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 forwarding*	Empty	The extension number(s) to which received voicemail messages will be forwarded. Multiple numbers can be specified using a comma (,) delimiter.
Email address*	Empty	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.
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✓ * This category allows multiple entries (divided with commas).

7.5.7. 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

7.5.8. 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 PBX. off – RTP is not handled by PBX. (Unless specify at the User settings or on the ARS.)
Codec priority	Blank	Specify codec to be used. Use a comma (,) when specifying multiple codec.
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	Blank	<p>Allow user to “monitor” another user’s conversation.</p> <p>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.</p> <p>(Pro Edition only)</p> <p>If a tilde (~) is placed before the extension, user 1000 can not speaks during the conversation.</p> <p>If a caret (^) is placed before the extension, user 1000 can not listens to the conversation.</p> <p>If both (~) and (^) are placed before the extension, user 1000 can not speaks nor listens to the conversation.</p>
Max sessions	unlimited	Specify the maximum received session numbers for the user.

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

7.5.9. 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	Empty	Default destination (phone number or SIP-URI) for an incoming call that has not specified a call recipient.
Speed dial*	Empty	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.

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

Button Name		Description
Target users*	Blank	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.




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

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

7.6. 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, and () to delete.

7.6.1. User Messages

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 and click [Download] button. The file will be downloaded to your PC.
Delete		To delete a voicemail message, select the desired message from the pull-down list and click [Delete] button. The message will be deleted from the voicemail inbox.

7.7. Voice Prompts

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

7.7.1. 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.*

7.8. Notes for Sound Files

Uploaded sound files must be formatted as below.

Sample rate	8000kHz
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.

8. Uninstall (Windows)

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

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.

9. Uninstall (Red Hat Linux)

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