

Brekeke PBX

Version 2.0

Administrator's Guide (Advanced)

Brekeke Software, Inc.

Version

Brekeke PBX v2.0 Administrator's Guide (Advanced)

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1. Introduction

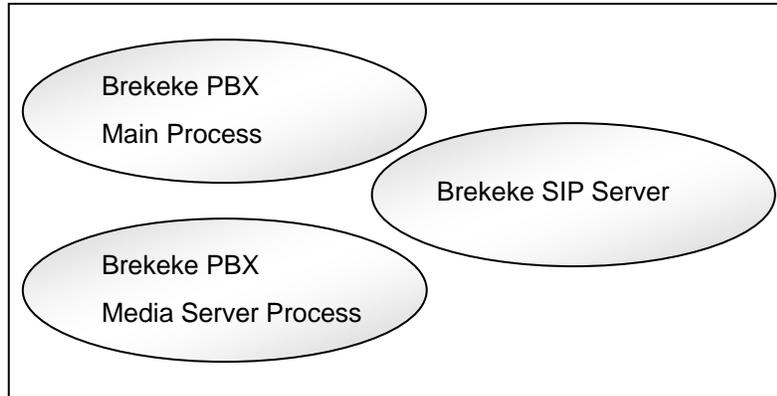
Our award-winning product, Brekeke PBX is a full-featured IP-PBX system easily managed through a web-based administrative interface. Brekeke PBX is feature rich and fully scalable to meet the needs of any size organization. Brekeke PBX solutions are flexible and affordable, from Basic with traditional 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.

One of the many benefits of using Brekeke PBX is that compatibility issues can be minimized because the software mediates between different types of IP phones, allowing you a wider range of IP phones to select from. There's no need to purchase expensive, multi-function IP phones in order to access all the features of Brekeke PBX. With just the touch-tone dialing pad of a common phone (0-9,*,#), you can operate the same functions as a legacy PBX phone. Easy interoperability, and the use of simple IP phones, can reduce overall costs to implement and manage Brekeke PBX.

This document is intended for the use of those who have already reviewed the Brekeke PBX Administrator's Guide (Basic) and want more advanced information about the details of each function of Brekeke PBX.

2. Processes on Brekeke PBX

When Brekeke PBX is started, three processes are initiated; SIP Server process, Brekeke PBX main process, and the process for media server (voicemail/IVR features).



✓ *Brekeke SIP Server can be installed separately from Brekeke PBX.*

2.1. Main Process

The Brekeke PBX main process receives and dispatches SIP sessions between SIP User Agents (UAs). All call sessions between caller and callee are mediated by the Brekeke PBX main process.

2.2. Media Server Process

The media server process handles RTP (Real Time Protocol) packets (multi-media data, including sound data). It also handles some PBX features, such as voicemail, auto attendant, and setup. This process is separate from the Brekeke PBX main process; however, with the current version of Brekeke PBX you may not run this process on a separate PC from the main process.

3. About RTP Relay

In accordance with your operating environment, and its requirements, and to ensure proper SIP communications, the RTP relay through Media Server (one of the process of Brekeke PBX) may be necessary. The media server process was handling the RTP relay in older versions of Brekeke PBX (1.3 – 1.5). From version 2.0, the PBX main process handles the RTP relay.

- *If you are using a SIP UA that does not support the SIP-INFO (dtmf-relay) method, and wish to use Brekeke PBX features operated by dial key pad commands, such as Call Forwarding, Call Park, Call Recording, etc., RTP packets need to be relayed through a Media Server.*
- *If you are using SIP devices (SIP UA, SIP proxy server, etc.,) that do not support changing RTP sender information by receiving re-INVITE requests, RTP packets need to be relayed through a Media Server.*
- *If you are using SIP devices that do not allow changing a voice codec during a call, “on (G.711u only)” at RTP relay setting or “0 (G.711u only)” at Coded priority in PBX setting need to be selected. Switching voice codec during a call can cause trouble for many types of SIP devices. Thorough testing is recommended before deciding to set RTP relay setting to “off” (which is not G.711u only) options or use a codec other than G.711u.,*

RTP relay settings can be set under several locations in Brekeke PBX: Options, ARS Settings, and User Settings. The settings under [User Settings] have the highest priority. When the default is set at [User Setting], settings under [ARS Settings] are applied. When User Settings and ARS Settings are set to the defaults, the settings for [Options] are applied.

RTP relay will be turned off only when both end points are set to RTP relay “off”. When one of the end points require RTP relay, RTP will be relayed through PBX. By decreasing the amount of RTP packets handled by PBX, the maximum number of concurrent sessions that PBX can handle is increased.

Even when the RTP relay setting is set to “off”, PBX handles RTP for some PBX features such as Music on Hold and Voicemail. Thus, depending upon the type of usage and environment, the maximum number of concurrent session can be varied upon each setting.

Brekeke PBX supports not only codec G.711 Ulaw but also G.729 (Option), G.711 alaw and iLBC. When iLBC or G.729 is used, the PBX machine load will increase because Brekeke PBX performs codec conversion.

4. SIP Server Relationships

4.1. Brekeke PBX and SIP Server Relationship

Brekeke SIP Server is embedded in Brekeke PBX. If you log in to Brekeke PBX as an Administrator, you can also access and manage Brekeke SIP Server. Brekeke SIP Server is a SIP proxy server which provides call session management in a VoIP network and processes SIP requests and responses. Brekeke PBX receives a call session and initiates another session with the appropriate destination according to user settings, such as forwarding destinations, then joins the two sessions.

4.1.1 Default Dial Plans in Brekeke SIP Server

Brekeke SIP Server has the following default settings in the Dial Plan.

PBX Prefix

Matching Patterns	Deploy Patterns
<pre>\$port=15062 \$localhost=true \$request=^INVITE To=(^.*sip:)pbx(.+\$)</pre>	<pre>To=%1%2 \$target=localhost:15060 \$auth=false</pre>

If the session that has come from PBX has the prefix “pbx” in To header, the prefix will be stripped off and the session will be routed to PBX again. Authentication for this session is disabled.

From PBX 1

Matching Patterns	Deploy Patterns
<pre>\$port=15062 \$localhost=true \$request=^INVITE X-PBX-Target=(.+)</pre>	<pre>\$auth=false \$target=%1 X-PBX-Target= &net.sip.hide.loopback=true</pre>

If the session has come from PBX and has the “X-PBX-Target” header which value is set in the [Target] field at ARS Pattern-OUT, the session will be sent to the destination set in [Target] field. Authentication for this session is disabled.

From PBX2

Matching Patterns	Deploy Patterns
<pre>\$port=15062 \$localhost=true \$request=^INVITE</pre>	<pre>\$auth=false &net.sip.hide.loopback=true</pre>

Other sessions from PBX will be applied. Authentication is disabled.

To PBX

Matching Patterns	Deploy Patterns
<code>\$request=^INVITE</code>	<code>\$target=localhost:15060</code>

The sessions that are not from PBX will be routed to PBX. To header field will not change.

Subscribe

Matching Patterns	Deploy Patterns
<code>\$request=^SUBSCRIBE</code> <code>Event=^x-tthd.* ^message-summary\$</code>	<code>\$target=localhost:15062</code>

SUBSCRIBE requests for Message Waiting Indicator (MWI) and for Multiline functions of SAXA IP Phones will be routed to PBX.

Thru Register

Matching Patterns	Deploy Patterns
<code>\$port=15060</code> <code>\$localhost=true</code> <code>\$request=^REGISTER</code>	<code>\$action=register</code> <code>&net.registrar.thru.catchhere=false</code> <code>&net.registrar.onlyglobal=true</code>

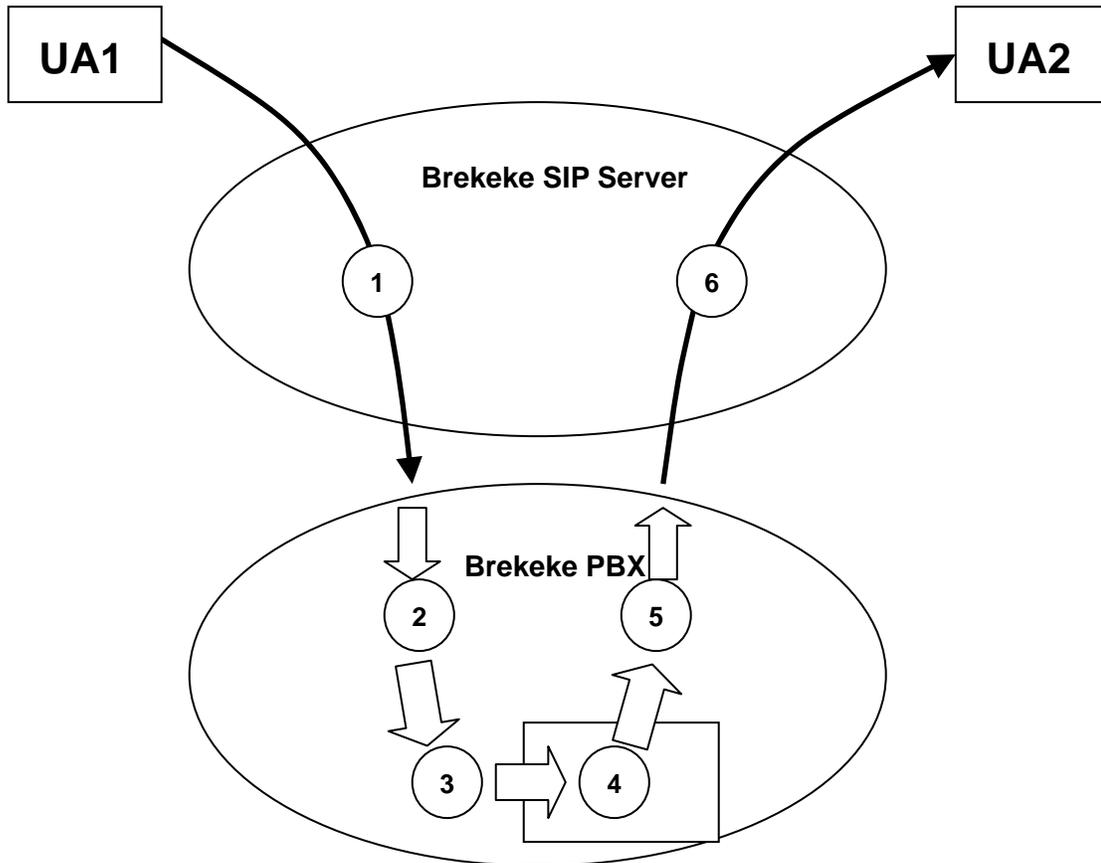
REGISTER requests from PBX will be applied. When PBX is registering to the third party SIP proxy server using Thru Registration, the PBX will not be registered at the Brekeke SIP Server.

When the third party SIP proxy server has a global IP address, and if Brekeke SIP Server does not have interface for the global IP address, the Brekeke SIP Server does not send REGISTER requests to the third party SIP proxy server. This is to prevent problems with ITSPs since having a local IP address registered at an ITSP's proxy server tends to cause unnecessary problems.

- ✓ Dial Plan called "To MediaServer" is not required from Brekeke PBX v2.0. When the prefix for media server is set at PBX, it directly connects with the media server without having SIP proxy in between.

5. Call Process Flow

In this section, we explain how a call from UA1 to UA2 is processed by Brekeke PBX and Brekeke SIP Server. This graphic represents the user agent's call step-by-step.



The following items correlate to the graphic presented above:

1. The call initiated by UA1 matches with the Dial Plan rule “to PBX” at Brekeke SIP Server and is routed to Brekeke PBX.
2. When Brekeke PBX receives the call, it searches an ARS Pattern IN that matches with the call. If PBX finds a Pattern, it will change the caller’s address (FROM) and call recipient’s address (TO) as specified in the Deploy Patterns.
3. If TO header includes a prefix for Call Pickup, Park Pickup, or for joining an existing call, PBX will make the call join the corresponding existing call. If not, PBX will treat the call as a new call.
4. PBX will check the setting in [User Setting]> [Forwarding destination] or [Forwarding destination (No answer/Busy)] and determine the call destination. If multiple

destinations are set in [Forwarding destination] > [Forwarding destination (No answer/Busy)] or some extension is set in [Automatic Monitoring] field, PBX will process the call routings for multiple destinations in parallel from Step 5.

5. Brekeke PBX searches an ARS Pattern OUT that matches the call. If PBX finds a Pattern, it will change the caller's address (FROM) and call recipient's address (TO) as specified in the Deploy Patterns.
6. The call is processed by Brekeke SIP Server's Dial Plan feature and routed to UA2.

6. Number System

A well-designed number system is indispensable for a user-friendly PBX system.

6.1. Defining a Brekeke PBX User

Brekeke PBX recognizes each extension number as an Brekeke PBX user ID. An Brekeke PBX user ID is the same as the user-info part in the SIP URI (e.g., In the SIP URI “sip:001@brekeke.com”, 001 is considered as the Brekeke PBX user ID). PBX users can use PBX features, such as Call Transfer, Call Hold, Call Forwarding, and Voicemail.

6.2. Understanding Prefixes

To provide easy access to PBX features such as Voicemail and Auto Attendant, Brekeke PBX uses a prefix system. By adding the appropriate prefix to an extension number, users can directly access Voicemail or Auto Attendant.

Please note that the default prefixes have changed in version 2.0.

The table below shows the default prefix settings:

IVR prefix	ivr
Voicemail prefix	vm
Voicemail review / Setting prefix	msg
Call Pickup prefix	*

If you upgraded the PBX from version 1.X to 2.X and kept all configurations, the prefixes are the same as before.

IVR prefix	6
Voicemail prefix	7
Voicemail review / Setting prefix	8
Call Pickup prefix	*

Though it is possible to change prefix values in the [Options] menu, we do not recommend changing the values unless necessary for compatibility with older versions.

If you change the prefix (number or alphabet) value, you will need to change the Brekeke SIP Server Dial Plan appropriately. The default Brekeke SIP Server Dial Plan is set with default prefix settings, and calls are routed accordingly.

From version 2.0, the prefixes are not numbers. So it is not possible to dial the prefix directly. We have prepared the default ARS rule “mediaserver prefix” so that users can dial numbers to directly access the Media Server features (Voicemail/IVR).

7. Call Forwarding

Call Forwarding also includes Call Forwarding (No Answer) and Call Forwarding (Busy).

7.1. Call Forwarding

[Administrative settings] > [Type of Call Forwarding] = Basic

Each user can set up call forwarding individually. When a call is received at the user's extension, the call will be forwarded to the destination set in the **[Forwarding destinations]** field in the **[User Setting]** menu.

By default, the **[Forwarding destinations]** field is left blank and all incoming calls are directed to the user's extension. If there are destinations specified in the field, only those extensions will receive calls.

If multiple destinations are set, incoming calls will be forwarded to all destinations (Ring Group). Each extension's phone will ring at the same time, and the first person to answer will be connected with the caller.

7.1.1. Example 1: Forwarding Calls to a Single Extension

Extension: 001

Forwarding destinations*	002
---------------------------------	-----

With this setting, every call that comes into extension 001 is transferred to extension 002. Extension 001 will not receive any calls that are dialed for extension 001.

7.1.2. Example 2: Forwarding Calls to Multiple Extensions (Ring Group)

Extension: 001

Forwarding destinations*	001, 002
---------------------------------	----------

With this setting, every call that comes into extension 001 will be transferred to extensions 001 and 002. If there is no forwarding set at the extension 002, both extension 001 and 002 will receive calls that comes into extension 001. The first person that answers the phone will be connected with the caller.

7.1.3. Example 3: Forwarding Calls to Multiple Group Extensions (Multiple - Ring Group)

Extension: 001

Forwarding destinations*	100, 200
---------------------------------	----------

Extension: 100

Forwarding destinations*	101, 102, 103
---------------------------------	---------------

Extension: 200

Forwarding destinations*	201, 202, 203
---------------------------------	---------------

With this setting, incoming calls to extension 001 are forwarded to extensions 100 and 200. Extensions 100 and 200 in turn forward calls to extensions 101,102, 103, 201, 202, and 203. If the users on these extensions do not have call forwarding set up, these 6 extensions will ring. The first person that picks up the phone will be connected with the caller.

7.1.4. Example 4: Enter SIP URI in the Forwarding Calls Field

Extension: 001

Forwarding destinations*	sip:6636@brekeke.com
---------------------------------	----------------------

You can specify SIP URI in the **[Forwarding destinations]** field under the **[user setting]** menu.

7.1.5. Example 5: Ignore Destination Number's Call Forward Settings-Forward Call Only to the Specified Extension Here

Extension: 001

Forwarding destinations*	%002
---------------------------------	------

With this setting, the calls directed to extension 001 are forwarded to extension 002. All of the call forwarding settings at extension 002 will be ignored, and the settings at 001 will be applied. Also, forwarded calls will be recorded on extension 001's call log.

7.1.6. Example 6: Ignore

Extension: 001

Adding "!" before an extension instructs the system to ignore that extension. This setting is useful when a certain extension needs to be excluded from the setting temporarily. Note that the call will be disconnected when all of the destination extensions have "!"

7.1.7. Example 7: Disconnection

Extension: 001

Forwarding destinations*	!
---------------------------------	---

If only ! is set in the field, [Forwarding destination (Busy)] setting for the extension 001 will be applied. If none is set in the [Forwarding destination (Busy)] , the call will be disconnected.

7.2. Call Forwarding (No answer)/ Call Forwarding (Busy)

Use this forward setting in the following situations:

The UA send responses other than 200OK response (e.g., busy) > Call Forwarding (Busy) or

No answer after the ringer timeout occurs > Call Forwarding (No answer)

Each user can set up forwarding for unanswered/busy calls. By default, unanswered/busy calls are forwarded to the user's voicemail inbox. (For the extension 001, "vm001" is set for the **[Forwarding destination (No answer)]** and **[Forwarding destination (Busy)]** field.

When multiple destinations are set in [Forwarding destinations] field:

1) [Call Forwarding (no answer)] will be applied in the following case:

- ✓ When no phone has answered the call for the period of [Ringer time].

2) [Call Forwarding (busy)] will be applied in the following cases:

- ✓ All of the forwarding destinations returned a response other than 200 OK.
- ✓ All [Ringer time] timeout of the forwarding destinations has reached before the extension's [Ringer time] timeout.

Unlike regular call forwarding settings, you can only specify a single destination here. If you need to set up multiple destinations here, you can use the method described in [13.1.3 Example 3: Forwarding Calls to Multiple Group Extensions (Multiple - Ring Group)].

7.2.1. Example 1: Forwarding Unanswered/Busy Calls to Voicemail

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Incoming calls to extension 001 are forwarded to the user's voicemail inbox (extension vm001) if the call is busy or remains unanswered after ringing for 15 seconds.

7.2.2. Example 2: Forwarding Unanswered/Busy Calls to Another Extension

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	916504016636
Forwarding destination (Busy)	vm001

With this setting, the call will be forwarded to 916504016636 if the call remains unanswered for 15 seconds. If the call is busy, the call will be forwarded to the extension 001's voicemail.

7.2.3. Example 3: Busy Call Forwarding is empty

Extension: 001

Ringer time (sec)	15
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	

Incoming calls to extension 001 are forwarded to the user's voicemail inbox (extension vm001) if the call remains unanswered after ringing for 15 seconds. Even for the case the call is busy, PBX will wait for 15 seconds and then the call will be forwarded to the user's voicemail.

7.3. More Advanced Forwarding Settings

By combining Call Forwarding, Call Forwarding (No answer), Call Forwarding (Busy) and Ringer time, you may create more advanced call forwarding settings.

Here are a few rules that are useful to remember:

- When Call Forwarding is set for the extension, the Ringer time for the call depends on the setting on the extension that calls are forwarded to. If no one answers the call for the Ringer time, original call recipient (not forwarding destination)'s [Forwarding destination (No answer)] will be applied (Please refer to 7.3.1.) If all of the forwarding destinations returned a response other than 200 OK or if all [Ringer time] timeout of the forwarding destinations has reached, [Forwarding destination (Busy)] will be applied.
- Only a single destination can be set in the [Forwarding destination (no answer/busy)] field.
- After the call is forwarded by Busy Call Forwarding or No Answer Call Forwarding, forwarding destination's [Forwarding destinations], [Forwarding destination (No answer)], [Forwarding destination (Busy)] settings will be used and the call can be transferred to another destination. (Please refer to 7.3.2).

7.3.1. Example 1: Ringer Time & Forwarding Destination (no answer/busy)

Extension: 001

Forwarding destinations*	101, 102
Ringer time (sec)	20
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Extension: 101

Forwarding destinations*	
Ringer time (sec)	15
Forwarding destination (No answer)	vm101
Forwarding destination (Busy)	vm101

Extension: 102

Forwarding destinations*	
Ringer time (sec)	25
Forwarding destination (No answer)	vm102
Forwarding destination (Busy)	vm102

- 1) All incoming calls to the extension 001 > forwarded to both extensions 101 and 102
- 2) Extension 101 rings for 15 seconds and Extension 102 rings for 25 seconds > no answer
- 3) The call will be forwarded to the 001's voicemail inbox vm001

User 001's settings, Ringer time and Call Forwarding (no answer), are treated as highest priority even when the user 101 and 102 have set up call forwarding (no answer) on all of his/her calls.

With the settings above, incoming calls to extension 001 will be transferred to both extensions 101 and 102. If the call is not answered by extension 101 or 102, it will be forwarded to the destination specified in User 001's **[Call Forwarding (no answer)]** field. The forwarded call will ignore **[Call Forwarding (no answer)]** settings for extensions 101 and 102.

Ringer time for extension 001 is set for 20 seconds. Even after User 101's ringer stops, extension 102 continues ringing for 20 seconds. Then the call will be forwarded to extension 001's voicemail inbox (extension vm001). When the call is forwarded to the voicemail inbox, the ringer will stop.

7.3.2. Example 2: Ring Group and Call Forwarding

Extension: 000

Forwarding destinations*	001
Ringer time (sec)	5
Forwarding destination (No answer)	001A
Forwarding destination (Busy)	001A

Extension: 001A

Forwarding destinations*	100A, 200A
Ringer time (sec)	10
Forwarding destination (No answer)	001B
Forwarding destination (Busy)	001B

Extension: 001B

Forwarding destinations*	100B, 200B
Ringer time (sec)	10
Forwarding destination (No answer)	vm001
Forwarding destination (Busy)	vm001

Extension: 100A

Forwarding destinations*	101, 102, 103
---------------------------------	---------------

Extension: 200A

Forwarding destinations*	201, 202
---------------------------------	----------

Extension: 100B

Forwarding destinations*	111, 112, 113, 114, 115, 116
---------------------------------	------------------------------

Extension: 200B

Forwarding destinations*	211, 212, 213, 214
---------------------------------	--------------------

- 1) All incoming calls to extension 000 are forwarded to extension 001.
- 2) When a call is unanswered for 5 seconds, it is forwarded to 001A.
- 3) The call is forwarded to 100A and 200A.
- 4) Extensions 101, 102, 103, 201, and 202 will ring.
- 5) If the call is unanswered for 10 seconds it is forwarded to extension 001B.
- 6) The call is forwarded to 100B and 200B.
- 7) Extensions 111, 112, 113, 114, 115, 116, 211, 212, 213, and 214 will ring.
- 8) If the call is unanswered for 10 seconds, it is forwarded to the extension 001's voicemail inbox vm001.
- 9) When vm001 answers the call, all ringing will stop.

Meanwhile all extensions that receive the call will keep ringing until their own Ringer time setting expires or the call is answered.

With this example, we used letters for the extension numbers in order to clarify different groups.

Important Note: If a caller is using a SIP UA which allows letters, the caller can make a call directly to a number such as 100B.

7.4. Round Robin / Top-down Call Distribution

Brekeke PBX features two types of call distribution: Round Robin and Top-Down. The round robin setting is useful if you want to distribute calls equally within a specific group of extensions. The top-down setting is useful if you want to distribute calls in a certain order within the specific group of extensions. In either case, the system will ring each available extension sequentially until the call is answered.

Round Robin

Under round robin settings, the extension that rings first will keep changing in the order as specified. Every time the call will be forwarded to the next specified extension of the one that answered the last call.

Top-Down

Under top-down setting, calls are always forwarded in a specified order. With Brekeke PBX, the number specified in the left most in the field is given the highest priority. The same setting can be configured under the basic call forward settings, but top-down settings are more simplified.

Single Attempt

When the Single Attempt setting is set to “no” (default setting), calls will keep ringing the next destination even after the [Ringer time] has expired. A similar setting is possible by specifying multiple destinations under “Basic” settings located in the [Type of Call Forwarding] section; however, using this feature is more straightforward.

When the Single Attempt setting is set to “yes”, after one failure to establish a connection, the calls will be forwarded to the destination set at [Forwarding destination (No answer)] or [Forwarding destination (Busy)].

Call Queue

Using the Call Queue feature, calls are kept on hold until a specified user answers the call. If the Single Attempt setting is set to “yes”, the unanswered call will be forwarded to the destination set at [Forwarding destination (No answer)] after one attempt.

7.4.1 Example 1: Round Robin / Top-Down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Forwarding destination (No answer)	vm100
Waiting time in the queue (sec)	0
Single attempt	No

- **Top-Down Setting**

- 1) Extension 100 starts ringing.
- 2) If the call is unanswered at extension 100 after 10 seconds, extension 101 starts ringing.
- 3) If the call is unanswered at extension 101 after 5 seconds, extension 102 starts ringing.
- 4) If the call is unanswered at extension 102 after 5 seconds, the call is forwarded to 7100.

- **Round Robin Setting**

The Call Forwarding order is the same as the top-down setting for the first call, however, from the second call on, the call is forwarded to the extension after the one that answered the most recent call.

7.4.2 Example 2: Round Robin / Top-Down

Extension: 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	10,5,5
Forwarding destination (No answer/Busy)	vm100
Waiting time in the queue (sec)	0

Extension: 101 (Call Forwarding setting [Basic])

Forwarding destinations*	200,201
Ringer times (sec)*	20

Call Forwarding settings are enabled at each extension individually, regardless of the settings of the extension the call is forwarded from. With the above example, when a call is forwarded to extension 101, it is directed to extension 200 and 201 simultaneously. If the call is unanswered at both extensions after 5 seconds, extension 102 will ring. Extension 200 and 201 will keep ringing until they reach the time specified in the Ringer times.

7.4.3 Example 3: Call Queue

Extension 100

Forwarding destinations*	100,101,102
Ringer times (sec)*	20
Forwarding destination (No answer/Busy)	vm100
Waiting time in the queue (sec)	180
Max number of calls in the queue	10

The Call Queue feature is used when the time, more than 1 second, is set at the [Waiting time in the queue (sec)] menu.

Calls are answered by the available extensions among 100, 101, or 102. The order the extensions answer will depend upon the mode setting (Round Robin or Top-Down). If the call is unanswered, it is placed in the queue.

When the calls in the queue exceed 10, the 11th call is forwarded to vm100. The callers in the queue will hear hold music while waiting to be answered. If the call is not answered after the caller waits 180 seconds in the queue, it will be forwarded to vm100.

After the period set in the **[Call interval (msec)]**, the call in the top of the queue will be forwarded to the available extension. When [Single attempt] setting is set for “no”, the call will keep calling until the waiting time for the queue (180 seconds) expires. To avoid missing calls, the calls in queue are designed to call different extensions for each attempted calls. Thus, sometimes the call at the top of the queue may not be answered first.

7.5. Schedule Call Forwarding

When Schedule is selected for the **[Type of Call Forwarding]**, call forwarding rules can be defined using conditions based on date, time, and Caller ID information.

7.5.1 Schedule Setup

Schedule setting is applied when the condition (period, date, hour, or Caller ID) matches. (The action will be same as the basic call forwarding setting.)

To set up a schedule:

- 1) Select *schedule* as the **[Type of call forwarding]** in the **[User Setting]** menu.
- 2) Click the **[Save]** button.
- 3) The menu items **[Schedule Settings 1]** and **[Schedule Setting 2]** will appear.
- 4) To define a schedule, click the **[...]** button. This will open the schedule menu as shown below.
- 5) Choose the dates and times during which you would like calls to be forwarded.
- 6) Set call forwarding destinations as you would with the other call forwarding settings.

Term 2004 1 1 - 2027 12 31

Days of week/Weeks of month

Sun Mon Tue Wed Thu Fri Sat

Days Week1 Week2 Week3 Week4 Week5

DD | DD-DD | MM/DD | MM/DD-MM/DD (separated by commas)

Time 15 : 00 - 16 : 00

00 : 00 - 00 : 00

00 : 00 - 00 : 00

OK CANCEL

7.5.2 Example: DD (Date of months)

5,10,15,20,25

This setting will be applied on the 5th, 10th, 15th, 20th, and 25th of every month during the specified term.

7.5.3. Example: DD-DD (Date-Duration)

5-10,15-20

This setting will be applied on between the 5th and 10th; and the 15th and 20th of every month during the specified term.

7.5.4. Example: MM/DD (Specific Date)

5/10,5/20,6/10,6/20

This setting will be applied on May 10th, May 20th, June 10th, and June 20th.

7.5.5. Example: MM/DD-MM/DD (Specific Date- Period)

5/10-6/20,8/10-8/20

The setting will be applied between May 10th and June 20th, and between August 10th and August 20th during the specified term.

7.5.6. Example: Time

08:30 – 13:30

23:30 – 00:30

This setting will be applied between 8:30am and 1:30pm; and between 11:30pm and 12:30am.

7.5.7. Example: Caller ID information

Applies to (Caller numbers)*	3*
Not Applies to (Caller numbers)*	3001,3002

By adding * (wildcard) after a number, you can specify all numbers that starts with that number.

By adding "?", you can specify any one number.

With the setting example above, all the coming calls that start with 3 except 3001 and 3002 will be forwarded according to this schedule call forwarding setting. The numbers specified in the "Not Applies to (Caller numbers)" field override "Applies to (Caller numbers)". If the "Applies to (Caller numbers)" field is left blank, all incoming calls will be directed according to this schedule call forwarding setting.

7.6. Add/Remove Forwarding Destinations (IVR)

With this feature, a user can add or remove their own number in **[Forwarding destinations*]** from their phone instead via Web Admintool page.

7.6.1 Setting Up Add/Remove Forwarding Destinations

Extension: 100

Forwarding destinations	6100
IVR	Add/Remove forwarding destinations
Target users*	300,400

When extension 200 calls into extension 100, extension 200 is added to the forwarding destinations for extension 300 and 400. If extension 200 already exists in the forwarding destinations for those numbers, extension 200 will be invalid as a forwarding destination (remain as forwarding destination with "!" in the front of the extension number).

8. Session Management

You can set a maximum number of sessions that an extension can receive in **[Max sessions]** under the PBX settings. Call sessions are counted between the beginning of the conversation until the phone call is terminated, or when the phone begins ringing until the phone call is terminated.

8.1. Unlimited Sessions

When unlimited sessions are set at **[Max sessions]**, incoming calls will ring the extension even if the extension is in session(s).

The call is forwarded to the destination set at **[Forwarding destination (No answer/Busy)]** when it is unanswered by the extension after the specified time (Ringer time) or conditions (SIP UA send response as “busy”, “reject”, etc.).

When **[Type of Call Forwarding]** is set for “Round robin/Top-down”, the call will only ring the available extensions.

8.2. Limited Sessions (0-6)

If the callee’s extension already has the maximum number of sessions on hold (set at **[Max sessions]**), the incoming call will be in a pending status.

When **[Type of Call Forwarding]** is set for “Round robin/Top-down”, the call will only ring the available extensions thus it will not be in pending status.

The calls will stay in pending status until the callee’s extension is available to receive sessions or **[Forwarding destination (No answer/Busy)]** setting is in effect.

8.3. Using Maximum Session Setting

This topic covers when to use Maximum Session settings.

- **Preventing missed calls**

By setting **[Max sessions]**, users can avoid missing calls. When multiple calls are coming in at the same time, all calls, except the first call, are forwarded to the destination set at **[Forwarding destination (No answer/Busy)]**.

- **Setting the preferred number of simultaneous calls**

For the SIP UAs that have capacity to handle multiple sessions, users can limit the preferred number of simultaneous calls and make other lines available for outgoing sessions.

9. Call Hold / Call Transfer

Brekeke PBX users can use Call Hold and Call Transfer features.

9.1. Different Methods of Call Hold / Call Transfer

9.1.1 Call Hold by re-INVITE/Call Transfer by REFER

Many SIP devices support Call Hold by re-INVITE (HOLD button) and/or Call Transfer by REFER (TRANSFER button). You can use these features of SIP devices with Brekeke PBX. Brekeke PBX will send "Music on Hold" to the other party when the Call Hold by re-INVITE (with send only or IP address is 0.0.0.0 in SDP) is made by one party (PBX did not send "Music on Hold" in the older version 1.x). When a member pushes the Hold button during a conference, other members can continue the conversation. When only one member does not place the call on hold and all of other members place the call on hold, the only one member who does not place the call on hold will hear Music on Hold.

When using Brekeke PBX, re-INVITE/REFER requests are processed at Brekeke PBX and not processed at the other party. Thus, Call Hold and Call Transfer will work even if the other party's device does not support re-INVITE or REFER.

9.1.2 Brekeke PBX Proprietary Ways of Call Hold/ Call Transfer

Even if SIP devices do not support the method of Call Hold by re-INVITE / Call Transfer by REFER, these devices can send commands to Brekeke PBX by using DTMF (dialing pads) for Call Hold / Call Transfer. On some devices, you may be able to assign DTMF numbers to HOLD/TRANSFER buttons and use them for Brekeke PBX proprietary method of Call Hold / Call Transfer.

- ✓ *In the following sections, we will explain the details of Brekeke PBX proprietary methods of Call Hold / Call Transfer.*

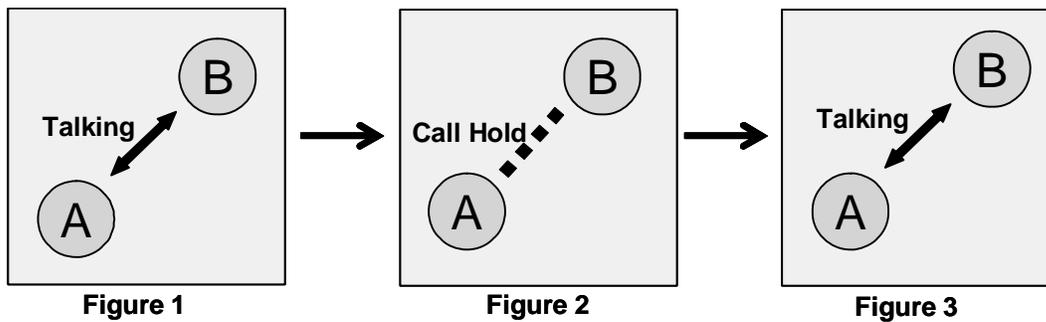
9.2. Call Hold

Any extension using the Call Hold feature must be registered as a user of Brekeke PBX.

Call Hold permissions can be set in the **[User Setting]** menu, at the **[transfer/hold]** option. In this section, Call Hold is referring to the unique Call Hold feature of Brekeke PBX, and it is different from Call Hold using re-INVITE requests (Section 9.1).

9.2.1 Call Hold – Activation / Deactivation

When Brekeke PBX Users A and B are on the phone with each other. (Figure 1), either user can place the call on hold by entering the hold command (Figure 2). Taking a call off hold can only be done by the user who placed the call on hold, so the conversation will not be resumed until User A takes User B off hold.



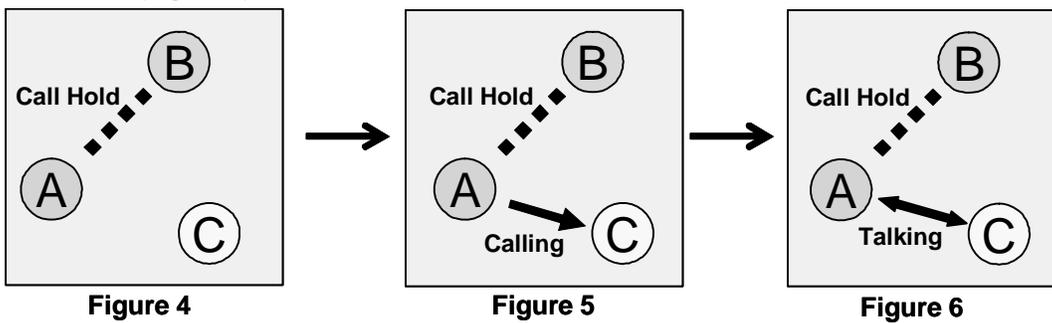
9.3. Initiating Call Transfer

After putting a call on hold, Brekeke PBX users can transfer a call to other users.

- ✓ *In this section, Call Hold is referring to the unique Call Hold feature of Brekeke PBX, and it is different from Call Hold using REFER requests (Section 9.1).*

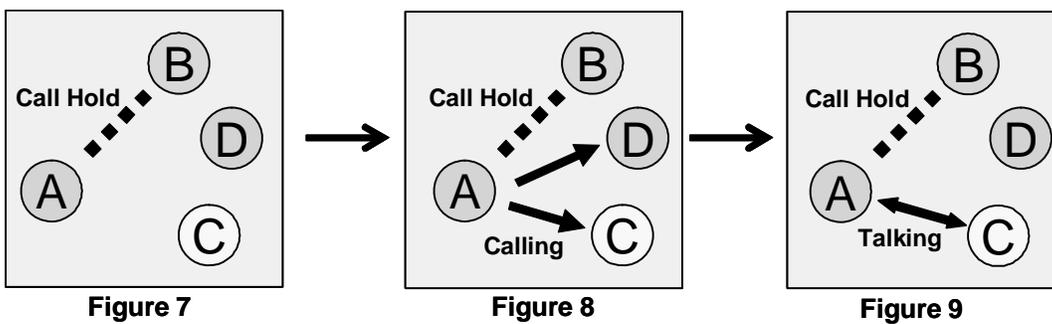
9.3.1 Initiating Call Transfer 1 – Calling the Third Party

User A puts the call with User B on hold (Figure 4), and initiates a new session with User C (Figure 5). When User C answers the call, the session between User A and User C will be connected (Figure 6).



9.3.2 Initiating Call Transfer 2 – Calling Multiple Users

If User C sets a [Forwarding destinations] for his/her incoming calls, the calls will be transferred to the specified destinations. In this example, User C has set User C and User D for the destinations. (Figure 7 and Figure 8) The first user who answers the phone will be connected with User A.



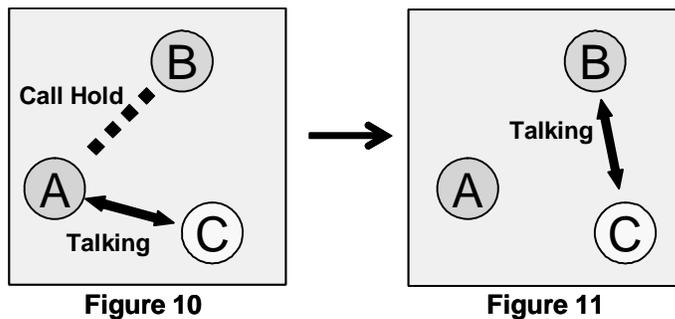
9.4. Processing Call Transfer

After talking with the transfer target user, you will have the opportunity to go back to the original session with User B.

- ✓ *In this section, Call Hold is referring to the unique Call Hold feature of Brekeke PBX, and it is different from Call Hold using REFER requests (Section 9.1).*

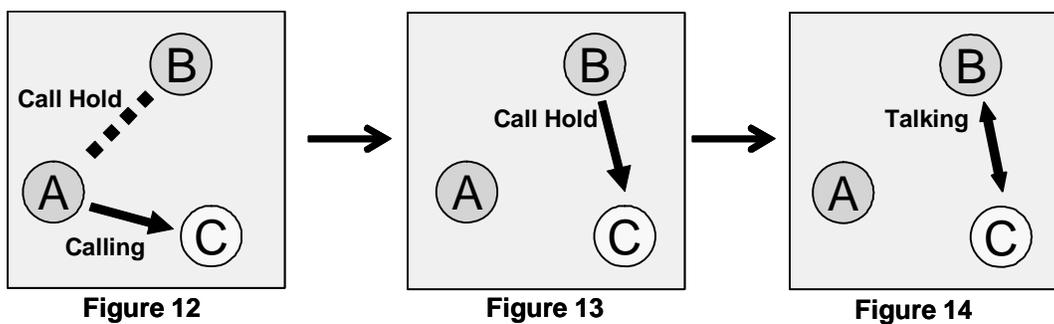
9.4.1 Processing Call Transfer 1 – Attended Transfer

After User A initiated Call Transfer, the call between Users A and B was put on hold. When User A hangs up the phone while Users A and B's session is connected (Figure 10, similar situation with Figure 6 and 9), the call with Users A and B will be transferred to User C. Now Users B and C are connected.



9.4.2 Processing Call Transfer 2 – Unattended Transfer

While the session between Users A and B are on hold, User A tries to connect with User C. (Figure 12, similar situation with Figure 6, 9, and 10) User A hangs up the phone before User C answers the call, then the call will be transferred to User C and User B wait for User C to answer. (Figure 13)



- ✓ *Using Unattended Transfer, User A cannot retrieve the call with User B even when User C did not answer the phone.*

9.5. Canceling Call Transfer

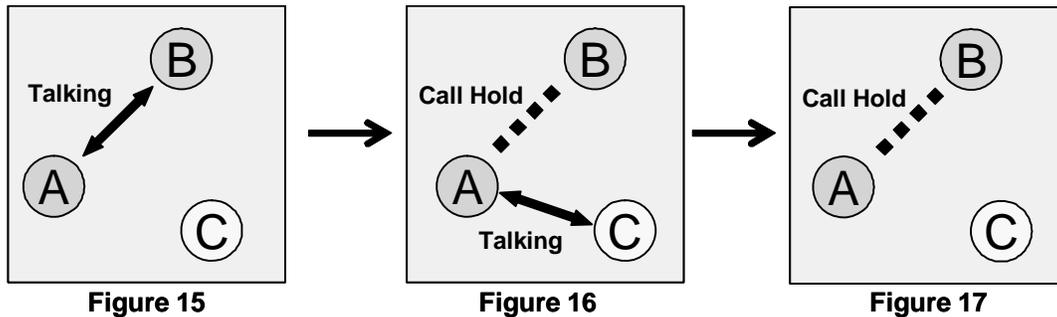
You may cancel Call Transfer in the middle of the process, and either go back to talk with the original caller or transfer the call to the different destination. In this section, Call Hold is referring to the unique Call Hold feature of Brekeke PBX and is different from Call Hold using REFER requests (Section 9.1).

9.5.1 Canceling Call Transfer 1 – Canceling Transfer While Talking with a 3rd Party

There are two ways to cancel Call Transfer when you are talking with a 3rd party.

- User A executes the cancel Call Transfer command
- User C hangs up call with User A

When the Call Transfer process was canceled, while User B was on hold (Figure 16, similar situation with Figure 6, 9, and 10), the call will be put back to the hold state with User A and B. (Figure 17)

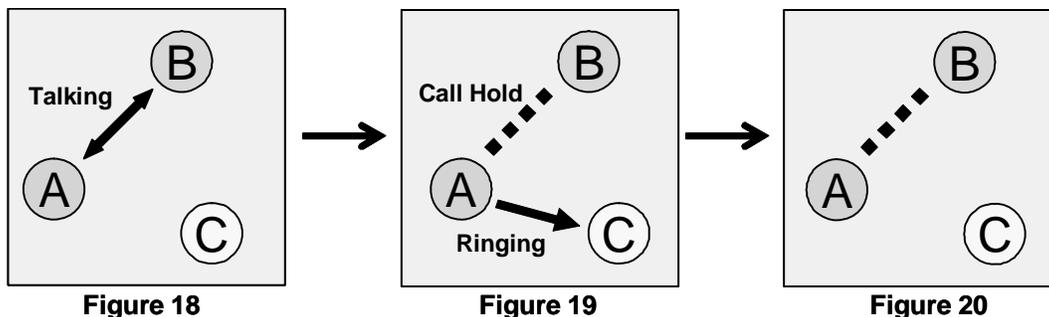


9.5.2 Canceling Call Transfer 2 – Cancel Transfer While Ringing the 3rd party

There are two ways to cancel Call Transfer when session is ringing for the 3rd party

- User A executes the cancel Call Transfer command
- When a call to User C cannot be completed (User C's phone is not connected, Call blocking, etc.,)

You can cancel Call Transfer while the phone is ringing at the destination, User C (Figure 19, similar situation at Figure 5 and 8). The call status goes back to the hold status between Users A and B.



9.6. Cautions for Call Transfer

- ✓ In this section, *Call Hold* is referring to the unique *Call Hold* feature of Brekeke PBX, and it is different from *Call Hold* using *REFER* requests (Section 9.1).

9.6.1 Extending Call Transfer

In the state of Figure 21 with regular settings, User C cannot initiate a Call Transfer with the call transferred to him/her. For User C to transfer the transferred call, either User A or B need to first hang up, then User C can transfer call to User D.

(If you really need to transfer the call to the 4th party, you need to write settings in the Dial Plan of Brekeke SIP Server to redirect the call from User A to C to Brekeke PBX.)

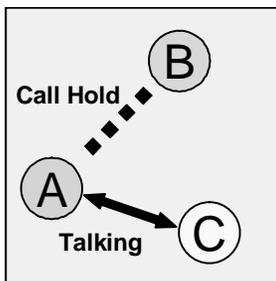


Figure 21

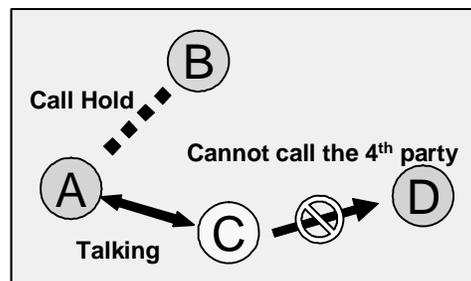


Figure 22

10. Auto Attendant

The Auto Attendant is essentially an automated receptionist system. Calls made to a specific extension are directed to a greeting system which plays a recorded message and asks for the extension of the party the caller wishes to reach--calls are then forwarded to the requested extension.

10.1. Setting Up Auto Attendant

Set up for Auto Attendant is done by the System Administrator through the Brekeke PBX Admintool.

10.1.1. Accessing Auto Attendant

User: 001

Choose [User Setting]

Forwarding destinations	%ivr001
IVR	Auto Attendant

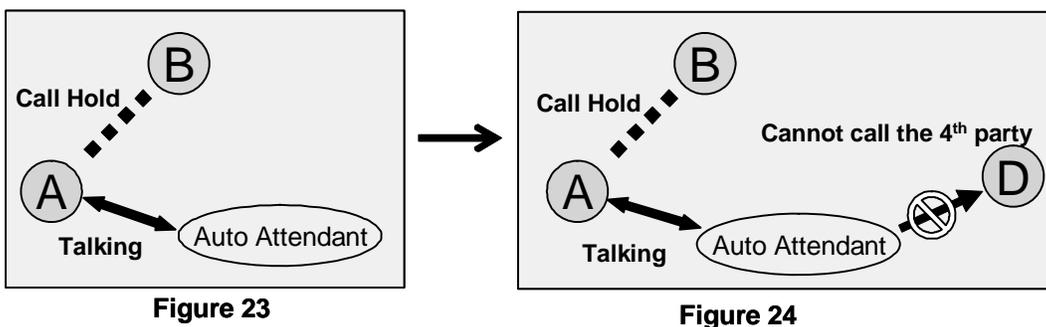
Add % and IVR prefix to the extension 001 and set it to [Forwarding destinations]. By adding % in front, user settings of the user 001 will be applied for the call to this auto attendant. (Please refer to the section “7.1.5. Example 5: Ignore Destination Number’s Call Forward Settings-Forward Call Only to the Specified Extension Here”).

- ✓ %ivr001 will be automatically set when you select the value other than “None” for the [IVR] setting.

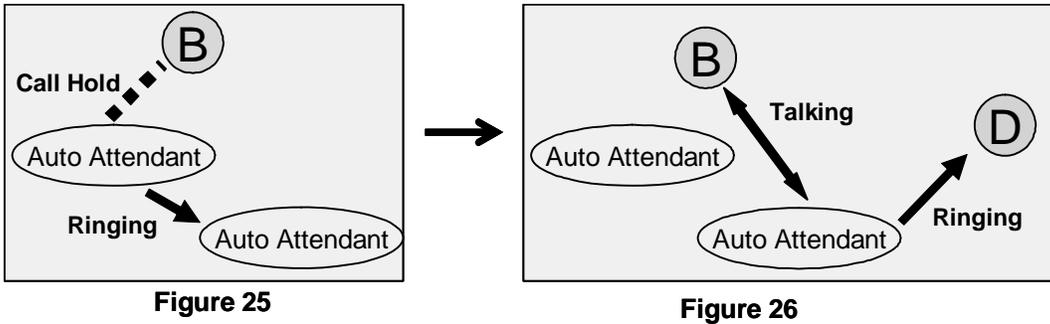
10.2. Cautions for Auto Attendant

10.2.1 Call Transfer to Auto Attendant

When a call is transferred (Attended Transfer) to the Auto Attendant (Figure 23), Auto Attendant cannot transfer the call (Figure 24). Refer to section called “Extending Call Transfer”.



However, Call Transfer to the 4th party is possible when the Call Transfer is initiated from another Auto Attendant. Since Auto Attendant operates automatically and the first Call Transfer is done instantly, 2nd Auto Attendant can transfer the call to the User D. (Figures 25 and 26)



10.2.2 Call Transfer from Auto Attendant to the Non-Brekeke PBX User

Even when **[Transfer to unregistered users]** is disabled under **[User Setting]**, Call Transfers from Auto Attendant can be made if the non-Brekeke PBX User's number or SIP URI are set at **[Default operator]** field or **[Speed dial]** field in the **[Auto Attendant]** section of the **[User Setting]** menu.

10.3. Speed Dial Setting Example for Auto Attendant

10.3.1 Simple Speed Dial settings

User: 001

Auto Attendant Setting

Default operator	001
Speed dial*	6=001, 7=101

User: 001

Forwarding destinations*	001, 002
---------------------------------	----------

It is handy to set up **[Speed dial]** for using the Auto Attendant feature.

Your Auto Attendant guidance might say:

"If you know your party's extension, please enter it now. For sales, press 6. For support, press 7."

If 6 or 7 are pressed, the call will be immediately transferred to the appropriate destinations. User 001 has defined multiple destinations for its **[Forwarding destinations]** setting as Ring Group setting. If there is no entry from the caller, the call will be transferred to the destination that is set in the **[Default operator]**.

✓ *With this setting, you will not be able to use the numbers that start with 6 or 7 for extension numbers.*

10.3.2 Speed Dial settings using Regular Expressions

User: 001

Auto Attendant Setting

Speed dial*	6636(.*)=\$1,(.*)=
Transfer to unregistered users	enable

Only when the dialed number is 6636 + <number>, the call will be transferred to <number>. Other numbers are not allowed. Because **[Transfer to unregistered users]** is set to enable, it is possible to transfer the call to an external user (such as PSTN number) but the caller need to know the first 4 digits (6636).

11. Voicemail

Each Brekeke PBX User can have a voicemail inbox.

11.1. Voicemail Inbox Number

The Brekeke PBX system treats a voicemail inbox number the same as a User ID or extension number.

11.2. Accessing Voicemail

Users can access their voicemail inbox by pressing the voicemail prefix number plus their extension number. For details, refer to Section 6.2, Prefix.

Example: Extension (or User ID) --- 001

- Dialing 07*001 → access the user 001's voicemail inbox directly to leave messages (If the configurations of version 1.X are kept after upgrading to 2.X, the dialing number will be 7001.)
- Dialing 08*001 → retrieve the user 001's messages, change settings for voicemail inbox (If the configurations of version 1.X are kept after upgrading to 2.X, the dialing number will be 8001.)
- Pressing "*" (Star) while listening to the voicemail guidance → same as dialing 08*8001

By default, each user's voicemail inbox is set for the destination of **[Call forward (No answer)]** and **[Call forward (Busy)]**. For details about **[Forwarding destination (No answer/Busy)]**, refer to the section called "Call Forwarding / Call Forwarding (No Answer/Busy)".

When a voicemail inbox receives a call, specified guidance is given to the caller. You can choose voicemail guidance from two personalized greetings and the default greeting.

11.3. Greeting Messages

11.1.1. Recording / Uploading Greeting Messages

Users can record their personalized greeting messages through **[Review / Option]** (Refer to "Voicemail Navigation Map") or upload them through Brekeke PBX Web Admintool.

11.1.2. Selecting a Greeting Message

Users can have two personalized greeting messages, "Personal Greeting" and "Alternative Greeting." Common use of the "Alternative Greeting" is for special occasions such as long

holidays, sick days, and business trips.

You can also select a greeting message by calling the voicemail inbox and accessing **[Review / Option]**, or through the Brekeke PBX Admintool.

11.4. Managing Messages

The maximum length of the voicemail message that callers can leave is set in the Administrator's **[Option]** menu. The maximum number of voicemail messages that users can save in their inboxes is also set in the **[Option]** menu. Users can also set to forward the voicemail messages to specific extensions or destinations. For details, please refer to the section "Voicemail Messages: Forwarding and Email Notification."

11.5. Retrieving Voicemail Messages

Retrieving voicemail can be done several ways.

- You can download the sound file (WAV file) from the **[User setting]** in Brekeke PBX Admintool. Once you download the file, you can listen to the file.
- By calling your voicemail inbox and accessing **[Review / Option]** menu
- By using Email notification setting, and opening sound file (WAV file) attached to the email

11.6. Voicemail Message Status

Your voicemail has two possible states, New message or Saved message.

- New message: a message that you have not saved or erased including newly arrived messages
- Saved message: a message that you have chosen to save on the system

11.6.1 New Message Status

This status is for the newly arrived voicemail messages, including forwarded messages from other users. This message will remain in the inbox up to the number of days set in the **[Days to keep unsaved messages]** in the **[Option]** menu.

11.6.2 Saved Message Status

Users can save voicemail messages when they are in message review. Saved messages will **not** be deleted even after the days set in the **[Days to keep unsaved messages]**.

11.7. Retrieving Voicemail Messages

Users can listen to their voicemail messages by accessing the **[Review / Option]** menu.

11.8. Email Notification

When a new message arrives at your voicemail inbox, an email notification will be sent to the specified email address.

11.8.1 Set Up Email Notification

Enabling or Disabling the email notification features can also be done in the **[Review / Option]** menu. Other settings can only be done through the Brekeke PBX Admintool. In the Brekeke PBX Admintool, you can set up your email address to receive notifications and choose whether or not to receive voicemail messages as an attachment with the notification or not. For details, please refer to the section Voicemail Messages: Forwarding and Email notification.

11.9. Voicemail Messages: Forwarding and Email Notification

11.9.1 Forwarding Voicemail Message

If the User has set up voicemail message forwarding, (**[User Setting]** menu > Voicemail setting **[Message forwarding*]**), all voicemail messages will be forwarded to the specified destinations. You can assign multiple destinations to receive the voicemail messages. If you want to leave a copy of voicemail message as well as forwarding to destinations, you will need to specify your voicemail inbox number in the field as well.

You can also forward voicemail messages, to forward to a group of destinations, by using the forwarding settings.

11.9.2 Email Notifications

You may choose to receive email notifications when new voicemail messages arrive in your inbox. You can receive email notifications for messages that have been forwarded to other users as well.

Example:

- User 001 sets messages to forward to User 002 → Only User 002 will receive voicemail message that came in to User 001
- User 001 enters both email addresses of User 001 and 002 → Both User 001 and 002 will receive email notifications when User 001 receives a voicemail message

11.9.3 Max Messages

Brekeke PBX System Administrators can set the limit of the number of messages that users can have in their voicemail inboxes. **[Option] > [Media Server system setting] > [Max messages]** When the number of voicemail messages in the inbox reaches the maximum number, callers cannot leave messages in your voicemail inbox. Callers will hear the guidance, “The person you are trying to reach is unavailable. To leave a message, please wait for the tone.”

A message forwarded to a user who has reached the maximum number of messages in their inbox will not be accepted. If you are forwarding your voicemail messages to multiple destinations, the message will also be forwarded to inboxes that have not reached the maximum number of messages.

11.10. Voicemail Inbox Access Limitation

Each voicemail inbox will only allow a single user access at a time. When you are accessing voicemail inboxes using a phone, you cannot modify your settings through the Brekeke PBX Admintool. When some other person is using your voicemail inbox, your voicemail will be locked to other accesses. Sometimes the lock-outs may persist for half a minute or so after a user exits from his/her voicemail inbox.

11.11. Creating / Deleting Voicemail Inboxes

This feature is only available to the system administrators. Creating or deleting a voicemail inbox is the same as creating or deleting an Brekeke PBX User. You may create/delete voicemail inboxes as follows:

Using the phone to create / delete voicemail inboxes:

- 1) In the Brekeke PBX Admintool choose **[User Setting] > [IVR] > [Setup]**.
- 2) Call the number (IVR prefix) + (User number) (ex. 6001 for User 001)

Follow the guidance to create/delete the voicemail inboxes (For details, please refer to “Voicemail Navigation Map.”)

12. Automatic Route Selection (ARS)

Most of the call management rules for outgoing and incoming calls through outside lines are set at Automatic Route Selection.

Common outside lines are connected to:

- **Third Party SIP Servers**

Any SIP server that is not specified as the SIP server for Brekeke PBX (defined at **[Options] > [SIP settings] > [SIP Proxy Address]**) is considered to be third party SIP server. Common third party SIP servers are those that are located at VoIP service providers, or other office locations.

- **Gateways**

The gateways referred to here are SIP PSTN Gateways that connect digital and analog PSTN lines or software gateways that convert between different protocols.

Defining rules enables ARS to make optimal use of outside lines, and your gateways. In the **[ARS]** menu, you can define extension numbers that will receive incoming calls, and which lines to use for outgoing calls.

Here are some examples of uses for ARS feature:

- **Select the least expensive calling options**

For example, you can send all international calls through a specific call long distance carrier, or to connect to the branch office located in Taipei. With ARS feature, the least expensive calling option can be utilized.

- **Effective port use of PSTN Gateways**

When there are multiple PSTN Gateways installed in the office, some ports may be used more frequently than others. The ARS feature can select the most effective PSTN Gateway port automatically.

- **Effective Use of Outside Lines**

Similar to the above example, when you subscribe to multiple VoIP service providers, some phone numbers may be used more frequently than others. The ARS feature can select the most effective service provider automatically.

- **Network Traffic Management and Load Balancing**

When many sessions are in progress and network bandwidth is insufficient, some calls can be directed through PSTN Gateways.

- **Failover**

With ARS failover, Brekeke PBX seeks an alternate route if the specified route is unavailable or failed, and makes outgoing calls via an alternative route. For example, if you have multiple ITSPs (Internet Telephony Service Providers) or/and PSTN (Public Switched Telephone Network) gateways, Brekeke PBX allows for backup connections if the VoIP (Voice over Internet Protocol) or internet connection is lost. In this situation, Brekeke PBX would automatically switch to an analog route through the PSTN gateway. The route can be re-activated manually from the Admintool or automatically by preset settings. (This feature is not supported by the SmallOffice Edition in 1.X versions.)

12.1. Route Search Process

Two sessions are used for one call in Brekeke PBX. Please refer to the section “Call Process Flow” for the detail of session processing during calls.

For a simple call between two users, two route searches are done. The first search checks for possible route matches, and the second search selects a route to use. The first route search rules are defined in the IN pattern settings, and the second route search rules are defined in the OUT pattern settings.

Route Search is done by searching for Matching Patterns that apply to a call. A Route Search starts from the highest priority (lowest number) as defined by the ARS rules. If multiple patterns share a priority level, a pattern is selected randomly. When a matching pattern is found, the process will check if the Max Sessions value has been reached. If the pattern has sessions available, the process will select the pattern and convert the FROM and TO header fields according to the settings defined in **[Deploy Patterns]**.

Routes of sessions incoming to Brekeke PBX cannot be changed. It is possible to define conditions for changing routes of incoming calls using priority, but to do so may have no effect. For incoming sessions, ARS is only used for checking route matches and setting a Dial-in number. You can choose to forward incoming calls to multiple extensions or to the Auto Attendant in the **[User Setting]** menu. You also need to take care to ensure that calls between extension users do not match ARS rules, because those calls need not be routed through your gateway and third party SIP provider.

For outgoing calls, it is necessary to set priorities for effective routing depending on the usage of lines.

12.2. Matching Patterns

Using regular expressions enter conditions of TO or FROM headers for the matching process. If the field is left blank, all calls will be considered to match the condition. This field can be set in the same way as the Dial Plan Matching pattern in Brekeke SIP Server Admin tool.

Using [User] field of ARS Pattern-OUT setting, it is possible to restrict users for using the ARS pattern. "User" of the call for the ARS rule will be determined as follows:

- 1) If the destination number is PBX user, the destination number is considered as "User" of the call.
- 2) If 1 is not applicable and also if the calls is forwarded through some PBX user's [Forwarding destinations], [Forwarding destination (No answer)] or [Forwarding destination (Busy)] setting, the PBX user will be considered as "User" of the call.
- 3) If 2 is not applicable and also if the caller is PBX user, the caller will be considered as "User" of the call.
- 4) If none of the above are not applicable, no user will considered as the "User" of the call.

12.3. Deploy Patterns

This is similar to Brekeke SIP Server Dial Plan's Deploy patterns; however, there are several important differences:

- Numbers follow the "\$" sign in deploy patterns (Brekeke SIP Server's Dial Plan uses %)
- Conversion of From header that matches Matching Pattern is done as described in Deploy Pattern for From header. Conversion of To header is done as defined in Deploy Pattern for To header, DTMF and Target field. (Multiple lines can be defined in Matching Pattern and Deploy Pattern in an Brekeke SIP Server's Dial Plan rule and it is process at the same time)When FROM and TO field are blank, no conversion is performed. Also when DTMF field is blank, DTMF will not be sent.
- By using &f,&t,&u before number, it is possible to use a value of each Matching Pattern field in Deploy Pattern. &f, &t, &u correspond to From, To, User.
- If FROM, TO field is blank, no conversion will be performed. If DTMF field is blank, no DTMF will be sent. If Target field is blank, no target IP address will be specified (The call will be routed to the domain specified in To field in most cases).

12.4. ARS Templates and Editing Variables

It is possible to create a template in ARS Settings for multiple ARS rules that have same configuration patterns. In **[Running Status]**, real rules (not template) will be displayed. In the template (excluding fields, **[Route name]**, **[Descriptions]**, and **[Parameters]**) , you can use the variables &v1, &v2,....., \$v9. You can specify real values in **[Edit variables]**.

Templates for the Internet Telephony Service Providers (ITSPs) for which Brekeke Software has confirmed interoperability, are all configured in ARS Settings and all are disabled by default. Users can easily start using one of the ITSPs with Brekeke PBX by just enabling the ITSP's template and entering the user's account information in **[Edit variables]**.

13. Third Party Call Control

Using this feature, you can connect multiple SIP UAs through external interface. With this, you can create systems like “Click-to-Call” and “Autodialer”.

Using HTTP-GET (or POST) and Web Service interface, you can access Brekeke PBX externally. Here we will explain the method using HTTP-GET. For the specification using web service interface, please refer to XXXX manual.

Access the URL (format as below) using HTTP protocol. (You may do this easily by entering the URL at the address bar at web browser applications.)

```
http://<server-ip-address>:<server-port>/pbx/3pcc?to=<callee-number>&from=<caller-number>
&user=<caller-user>&type=<type>
```

<server-ip-address>	IP address of server where Brekeke PBX is installed on
<server-port>	The same port number you use accessing PBX Admintool
<callee-number>	Callee's number, or its SIP URI. You can specify multiple numbers by dividing number using comma.
<caller-number>	Caller's number or its SIP URI.
<caller-user>	PBX User which is used to call out (caller). You can omit this if this is same with “caller-number”.
<type>	Specify “1” when you would like to call destinations simultaneously. Specify “2” when you want to connect Caller before connecting Callee.

Response

Brekeke PBX will return the response: “succeeded: ID number”. This ID will be the ID used at **[Call Status]** screen for the session.