Brekeke PBX

Version 3

Administrator's Guide
(Advanced)

Brekeke Software, Inc.

Version

Brekeke PBX Version 3 Administrator's Guide (Advanced)

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1.　Introduction

This document is for system administrators who have already reviewed the Brekeke PBX Administrator's Guide (Basic). This document provides more detailed and advanced information about each function of Brekeke PBX.

2.　Processes on Brekeke PBX

When Brekeke PBX is started, three processes are initiated: the Brekeke SIP Server (bundled SIP server for Brekeke PBX) process, the Brekeke PBX main process and the media server (Voicemail/IVR features) process.
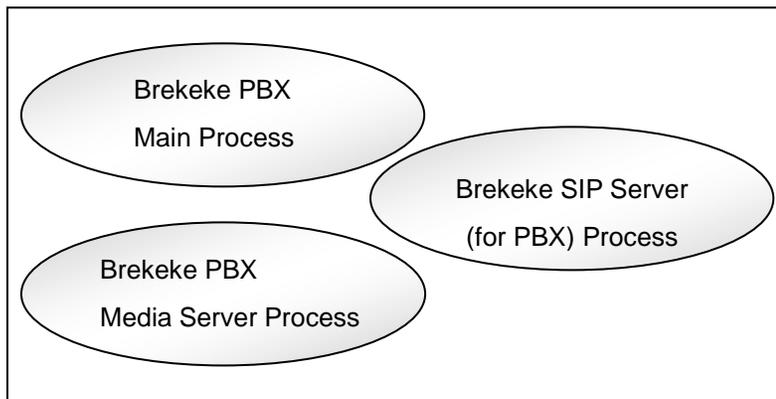
2.1.　Main Process

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

2.2.　Media Server Process

The media server process handles a number of Brekeke PBX features, such as Voicemail,　, Call Recording, and all IVR functions (including Auto Attendant).

2.3.　Brekeke SIP Server Process

The bundled SIP server for Brekeke PBX authenticates SIP requests and routes the SIP requests between Brekeke PBX and registered or external User Agents. For details about the bundled SIP server for Brekeke PBX, refer to Brekeke SIP Server Administrator's Guide.

3.   About RTP Relay

Depending on your operating environment and its requirements, and to ensure proper SIP communications, the RTP relay through Brekeke PBX may be necessary.

For example:

♦ If you are using a SIP UA that does not support the SIP-INFO (DTMF-relay) method, and you wish to use keypad commands to activate Brekeke PBX features, such as Call Forwarding, Call Park, Call Recording, etc., RTP packets need to be relayed through the 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 the Media Server.

♦ If you are using SIP devices that do not allow changing a voice codec during a call, select "on (G.711u only)" at the RTP relay setting or "0 (G.711u only)" at [Codec Priority] in the Brekeke PBX setting. Selecting "off" at the RTP relay setting may not work well depending on the SIP device. There may also be compatibility issues between different types of SIP devices. Thorough testing is recommended before deciding to set the RTP relay setting (which is not G.711u only) to "off" or to use a codec other than G.711u.

RTP relay settings can be set under several locations in Brekeke PBX: Options, ARS Settings and User Phone type Settings. The settings under User > [Phones] -> [Type] have the highest priority. When the default setting is defined in [Options] -> [Phone Type] and the Phone type is set in User -> [Phones] ->[Type], the settings under [ARS Settings] are applied. When User Phone type Settings and ARS Settings are set to their defaults, the settings at [Options] are applied.

RTP relay will be turned off only when both endpoints are set to RTP relay "off." When one of the endpoints requires RTP relay, RTP will be relayed through Brekeke PBX. By increasing the amount of RTP packets handled by Brekeke PBX, the Brekeke PBX server load will increase. Even when the RTP relay setting is set to "off," Brekeke PBX handles RTP for some features, such as Music On Hold and Voicemail. Thus, depending on the type of usage and environment, the maximum number of concurrent sessions may vary for each setting.

Brekeke PBX supports not only codec G.711 u-law, but also G.729 (Optional), G.711 A-law and

iLBC. When the RTP relay setting is set to "on," Brekeke PBX will do conversions between different codecs, so parties who are using devices with different codecs can talk each other without any problems.

4.   SIP Server Relationships

4.1.   Brekeke PBX and the Bundled SIP Server Relationship

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

4.2.   Default Dial Plans in the Bundled SIP Server

The bundled SIP server has the following default settings in the Dial Plan:

From PBX

This dial plan rule will apply to a session coming from Brekeke PBX that does not match the Dial Plan rules above it. It will disable Authentication and send the call to its destination.

To PBX from ITSP

All sessions from ITSP or SIP gateway devices with a localhost registration record will be routed to Brekeke PBX. B2BUA mode will be set to "true" and Authentication will be set to "false."

To PBX

Sessions that are initialized from registered UAs will be routed to Brekeke PBX with UDP transport and without Authentication checking.

Subscribe

All SUBSCRIBE requests for Message Waiting Indicator (MWI), Presence, BLF, SCA and for Multiline functions of SAXA IP Phones will be routed to Brekeke PBX.**Thru Register**

REGISTER requests from Brekeke PBX will be applied when Brekeke PBX is registering to a third-party SIP proxy server using Thru Registration. When a third-party SIP proxy server has a global IP address and the bundled SIP server does not have an interface for the global IP address, the bundled SIP server will not send a REGISTER request 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 problems.

Register Behind NAT

This dial plan is disabled by default. Edit and enable it to solve NAT issues for registration from some UAs. Also remove the underscore in either `&register.contact.nat=true` or `&net.registrar.adjust.expires=10`.

5.   Call Process Flow

In this section, we give a step-by-step explanation of how a call from UA1 to UA2 is processed.
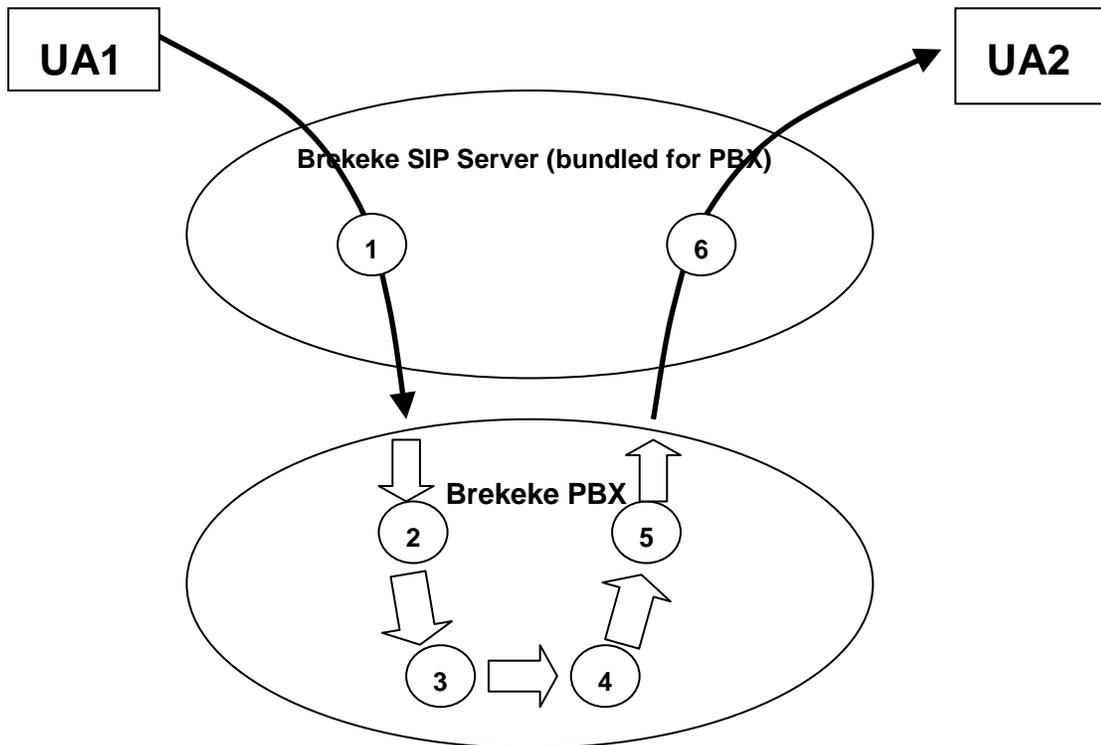
The following items describe the steps in the above image:

1)   The call initiated by UA1 matches with the Dial Plan rule "To PBX" at the bundled SIP server and is routed to Brekeke PBX.

2)   When Brekeke PBX receives the call, it searches for an ARS "Pattern IN" that matches with the call. If Brekeke 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 the "To" header includes a prefix for Call Pickup, Park Pickup or for accessing an existing call, Brekeke PBX will join this call with the corresponding existing call. If not, Brekeke PBX will treat this call as a new call.

4)   Brekeke PBX will check the settings in [Users] > [Inbound] to determine the call destination. If multiple destinations are set in the [Users] > [Inbound] > [Forward To] section or an extension is set in the [Users] > [Settings] > [Automatic Monitoring*] field, Brekeke PBX will process the call routings for multiple destinations in parallel with Step 5.

5)   Brekeke PBX searches for an ARS "Pattern OUT" that matches the call. If Brekeke 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 the bundled SIP server's Dial Plan feature and routed to UA2.

6.   Number System

6.1.   Defining a Brekeke PBX User

Brekeke PBX regards user extensions as Brekeke PBX users. It, therefore, associates a UA SIP ID with each Brekeke PBX user by looking for the [Phone ID] field in the Brekeke PBX user [Phones] page settings. All numbers entered in one user [Phone ID] field are considered as the same Brekeke PBX user.

6.2.   Understanding Prefixes

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

The prefixes used in Brekeke PBX are listed in the table below. Most prefixes cannot be modified, with the exception of the call pickup prefix. Users can include these prefixes in their settings, such as setting forwarding destinations as "vm100," which will forward calls to user extension 100's voicemail.

The table below shows the default prefix settings:

IVR prefix	ivr
Voicemail prefix	vm
Voicemail review/Setting prefix	msg

Call Pickup prefix	*

It is not possible to dial a prefix directly from most types of SIP devices. We have prepared the default ARS rule "mediaserver_prefix" so that users can dial numbers to directly access the Media Server features (Voicemail).

7.　Call Forwarding

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

7.1.　Call Forwarding

Each user can set up Call Forwarding individually. When a call is received at a user extension, the call will be forwarded to the destinations checked in the [Phone n] fields and also to the extensions set in the [Other Forwarding Destinations*] field on the [Users] > [Inbound] page.

By default, the [Other Forwarding Destinations*] field is left blank and all incoming calls are directed to the user's assigned phones checked on the [Inbound] page. If there are destinations specified in the [Other Forwarding Destinations*] field, both the checked phones and the extensions set in this field will receive calls. Each extension's phones will ring at the same time, then the first person to answer will be connected with the caller.

Different ringing times can be set to each assigned phone and extension in an extension's [Other Forwarding Destinations*] field. When no setting is input for a phone's [Ringer Time (sec)], the default ringer time setting under the [Call Forwarding] section will be applied. If there is a delay time set for a user-assigned phone, Brekeke PBX will wait the amount of time set in the phone's [Delay (sec)] field, then direct the call to this phone.

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

Forwarding settings are used in the manner described below:

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

♦　The call cannot be established or it received a busy response --> Call Forwarding (Busy)

Each user can set up forwarding for unanswered/busy calls. By default, unanswered/busy calls are forwarded to the user's voicemail inbox.

Only a single destination can be assigned in the [Forwarding destination (No answer)] and [Forwarding destination (Busy)] fields. If you need to set up multiple destinations to be used for these features, you can set a group extension number in these fields.

✓　*In the following examples, the default setting is used for each phone ringer time.*

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)	16504016636
Forwarding destination (Busy)	vm001

With this setting, if the call remains unanswered for 15 seconds, the call will be forwarded to PSTN number 16504016636 through an available ARS route. If the extension is busy, the call will be forwarded to 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)	

If a call to extension 001 remains unanswered after ringing for 15 seconds, it will be forwarded to the extension's voicemail inbox (vm001). When the [Forwarding destination (Busy)] field is left blank, Brekeke PBX will wait 15 seconds and then forward the call to the user's voicemail, even if the user's extension is busy.

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 may be useful to remember:

♦ Only a single destination can be set in the [Forwarding destination (No answer/Busy)] field.

♦ When Call Forwarding is set for an extension, the ringer time for the call depends on the setting of Call Forwarding (No answer/Busy). If no one answers the call during the Ringer time, the original call recipient's (not the forwarding destination's) [Forwarding destination (No answer)] will be applied. (Please refer to section 7.3.1 below for more information.) If none of the forwarding destinations can establish a call or if all forwarding destinations are busy, [Forwarding destination (Busy)] will be applied.

♦ After the call is forwarded by Busy Call Forwarding or No Answer Call Forwarding, the forwarding destination's setting in the [Forwarding destinations], [Forwarding destination (No answer)] and [Forwarding destination (Busy)] fields will be used and the call may be transferred to another destination. (Please refer to section 7.3.2 below for more details.)

✓ *In the following examples, the default setting is used for each phone's ringer time.*

7.3.1. Example 1: Ringer Time & Forwarding Destination (No Answer/Busy)

Extension: 001

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

Extension: 101

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

Extension: 102

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

1) All incoming calls to extension 001 are forwarded to both extensions 101 and 102.

2) Extension 101 rings for 15 seconds and Extension 102 rings for 20 seconds (not 25) if neither extension 101 nor 102 answers the call.

3) The call will then be forwarded to extension 001's voicemail inbox vm001.

User 001's Ringer time and Call Forwarding (No answer) settings are treated as the highest priority, even when extensions 101 and 102 have set up call forwarding (No answer) on all of their calls. The Ringer time for extension 001 is set for 20 seconds, so that is the maximum ringer time for any of the forwarding extensions. Even after User 101's ringer stops, extension 102 continues ringing until 20 seconds has passed. If no one answers, the call will be forwarded to extension 001's voicemail (extension vm001).

With the settings above, incoming calls to extension 001 will be transferred to both extensions 101 and 102. If neither extension 101 nor 102 answers the call, it will be forwarded to the destination specified in User 001's [Call Forwarding (No answer)] field. The [Call Forwarding (No answer)] settings for extensions 101 and 102 will not be used.

7.3.2. Example 2: Ring Group and Call Forwarding

Extension: 000

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

Extension: 001A

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

Extension: 001B

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

Extension: 100A

Group Extensions*	101, 102, 103

Extension: 200A

Group Extensions*	201, 202

Extension: 100B

Group Extensions	111, 112, 113, 114, 115, 116

Extension: 200B

Group Extensions	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 then forwarded to 100A and 200A.

4) Extensions 101, 102, 103, 201 and 202 will ring.

5) If extension 001's [Ringer Time] is more than 5 seconds, 001 will continue to ring even after 5 seconds.

6) If the call is unanswered for 10 seconds, it is forwarded to extension 001B.

7) The call is then forwarded to 100B and 200B.

8) Extensions 111, 112, 113, 114, 115, 116, 211, 212, 213 and 214 will ring.

9) If the call is unanswered for 10 seconds, it is forwarded to extension 001's voicemail inbox (vm001).

10) When vm001 answers the call, all ringing will stop.

Meanwhile, all extensions that receive the call will keep ringing until their own Ringer time settings expire or the call is answered.

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

✓ *If a caller is using a SIP UA that allows letters, the caller can make a call directly to an extension that includes letters, such as 100B.*

7.4. Call Distribution for Call Hunting Extension

Brekeke PBX features two types of call distribution for call hunting extension: 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 a specific group of extensions. In either case, the system will ring each available extension sequentially until the call is answered.

Round-Robin

Under the Round-Robin setting, you establish the order of the extensions to which an incoming call will be forwarded, and then the call will go to whichever extension is next in the list. For example, if the last call was forwarded to the second extension in the list, the next call will go to the third extension in the list, and so on.

Top-Down

Under the Top-Down setting, calls are always forwarded in the same order, starting with the left-most number in the list and moving to the right. 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.

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

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: Call Hunting

Extension: 200

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

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 vm100.

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 will be forwarded to the extension after the one that answered the previous call.

7.4.2. Example 2: Call Hunting

Extension: 200

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

Extension: 101 (Call Forwarding setting [Basic])

Other Forwarding destinations*	200,201
Ringer time (sec)*	20

Call Forwarding settings are established for 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 extensions 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 (sec)*] field.

7.4.3.   Example 3: Call Queue

Extension 200

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

The Call Queue feature is used when a time of 1 second or more is set at the [Waiting time in the queue (sec)] field.

Calls are sent to the available extensions among 100, 101 or 102, in the order established in the mode setting (Round-Robin or Top-Down). If a call is unanswered, it is placed in the queue.

When the number of calls in the queue reaches 10, the 11th call will be forwarded to vm100. The callers in the queue will hear hold music while waiting to be answered. If a 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)] field, the call at the top of the queue will be forwarded to the next available extension. When the [Single attempt] field is set to "no," the call will keep ringing until the waiting time for the queue (180 seconds) expires. When the [Single attempt] field is set to "yes," the call will be forwarded to the destination specified under [Forwarding destination (No answer)] after one failed attempt.

7.5.   Schedule Call Forwarding

Schedule call forwarding rules can be defined using conditions based on date, time, Caller ID and route information.

7.5.1.   Schedule Setup

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

To set up a schedule:

1)   Create a schedule extension at [Extensions] > [Schedule].

2) Click the [Add Forwarding Schedule] button. The menu item [Forwarding Schedule 1] will appear.

3) To define a schedule, click the [Conditions] tab. This will open the schedule menu.

4) Choose the dates and times during which you would like calls to be forwarded. When setting [Days of week], specify which day of which week in a month. For example: Checking "First" and "Sunday" means the schedule will only apply to the first Sunday in every month.

5) Set call forwarding destinations as you would under the [Forward To] tab.

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 each month during the specified term.

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

> 5-10,15-20

This setting will be applied between the 5th and 10th and also between the 15th and 20th of each 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 also between August 10th and August 20th during the specified term.

7.5.6. Example: !MM/DD-MM/DD (Exclude Date-Period)

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

The setting will be applied at all times during the specified term, except for the dates between May 10th and June 20th or between August 10th and August 20th.

7.5.7. Example: Time

08:30 – 13:30
23:30 – 00:30

This setting will be applied between 8:30 a.m. and 1:30 p.m., and also between 11:30 p.m. and 12:30 a.m.

7.5.8. Example: Caller Filter

Filter	3*
Route	From any route

By adding "*" (wildcard) after a number, you can specify all numbers that start with that number. By adding "?" after a number, you can specify any one number.

With the setting example above, when the [Matched] field is checked, all incoming calls that start with the number 3 will be forwarded according to this extension's call forwarding settings. When the [Not Matched] field is checked, all incoming calls that start with any number other than 3 will be forwarded according to this extension's call forwarding settings. When the [Filter] field is left blank, all incoming calls will be redirected according to this extension's call forwarding settings.

When the [Router] field is set to "From external line," all incoming calls through ARS routes with the [External] field checked will apply this schedule setting.

When the [Router] field is set to "Not from external line," all incoming calls through ARS routes without the [External] field checked, or calls without an ARS route applied will apply this schedule setting.

7.6. Add/Remove Forwarding Destinations (IVR)

With this feature, a user can add his/her own number to a Group extension [group extensions*] from his/her phone, instead of via Brekeke PBX Admintool.

Extension	400
Type	Add/Remove forwarding destinations
Target groups*	300,301

When user extension 200 calls extension 400, extension 200 is added to the forwarding destinations for group extensions 300 and 301. If extension 200 already exists in the forwarding destinations list for those numbers, then extension 200 will be removed as a forwarding destination (technically, remaining as a forwarding destination with a "!" 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 inbound sessions] under the Brekeke PBX user settings. Call sessions are counted either from the beginning of the conversation or when the phone begins ringing until the phone call is terminated.

8.1. Unlimited Sessions

When unlimited sessions are set at [Max inbound sessions], incoming calls will ring the extension even if the extension is in one or more sessions already. When the call remains unanswered by the extension after the specified (Ringer time), it is either forwarded to the destination set at [Forwarding destination (No answer/Busy)], or it executes that SIP UA's send response, such as "busy," "reject," etc.

8.2. Limited Sessions (0-6)

If the callee's extension already has the maximum number of sessions, as set at [Max inbound sessions], and all of those sessions are busy, an incoming call will be forwarded to the destination set at [Forwarding destination (No answer/Busy)]. If there is a session with ringing status, an incoming call will stay in pending status until all sessions are occupied or until there are no sessions remaining.

8.3. Using Maximum Session Setting

The Maximum Session setting may be useful in the following ways:

Preventing missed calls

By setting [Max inbound sessions], users can avoid missing calls. When multiple calls are coming in at the same time, all calls except the first one are forwarded to the destination set at [Forwarding destination (Busy)].

Setting the preferred number of simultaneous calls

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

9.   Call Hold/Call Transfer

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 with Brekeke PBX.

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

### 9.1.2.   Call Hold/Call Transfer Using Keypad Commands

Even if a SIP device does not support Call Hold by re-INVITE or Call Transfer by REFER, it can send commands to Brekeke PBX by using DTMF (Keypad) for Call Hold/Call Transfer. On some devices, you may be able to assign DTMF numbers to HOLD/TRANSFER buttons and use them for the proprietary Brekeke PBX method of Call Hold/Call Transfer. Keypad commands for Brekeke PBX are listed in the Brekeke PBX Users Guide.

9.2.   Call Hold

Any extension using the Call Hold feature with keypad commands must be registered as a user extension of Brekeke PBX. Call Hold permissions are set in the [Users > Edit] > [Keypad Commands] option. In this section, Call Hold refers to the unique Call Hold feature of Brekeke PBX, which is different from the Call Hold using re-INVITE request (Section 9.1).

9.2.1.   Call Hold – Activation/Deactivation

When Brekeke PBX user extensions A and B are talking with each other (Figure 1), either user can place the call on hold by using Keypad Commands (Figure 2). However, only the user who placed the call on hold can take it off hold, so if User A placed User B on hold, then the conversation will not resume until User A takes User B off hold.


Figure 1   Figure 2   Figure 3
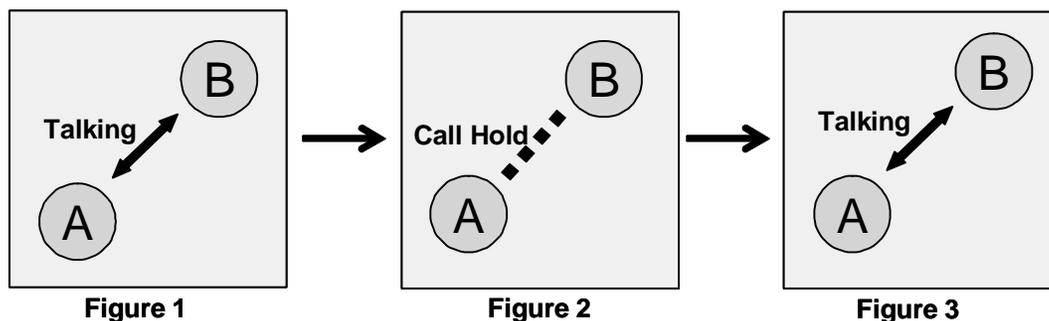
9.3. Initiating Call Transfer

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, which is different from Call Hold using a REFER request (Section 9.1).*

9.3.1. Initiating Call Transfer 1 – Calling a Third Party

User A puts a 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).

Figure 4   Figure 5   Figure 6
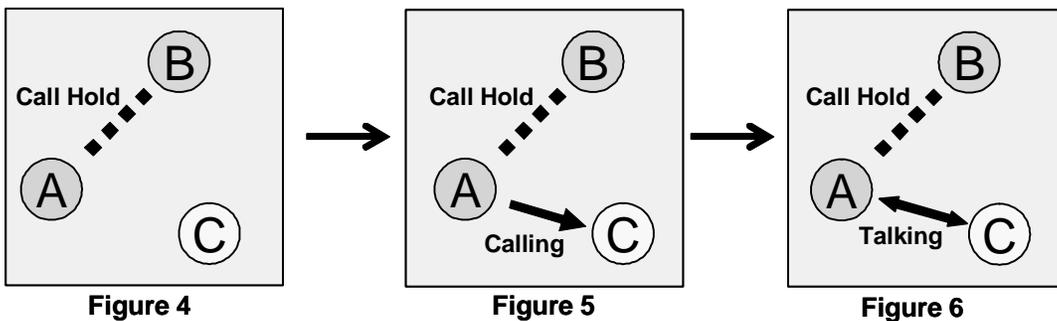
9.3.2. Initiating Call Transfer 2 – Calling Multiple Users

If User C has set [Forwarding destinations] for his/her incoming calls, the calls will be transferred to the specified destinations. In this example, User A 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.

Figure 7   Figure 8   Figure 9
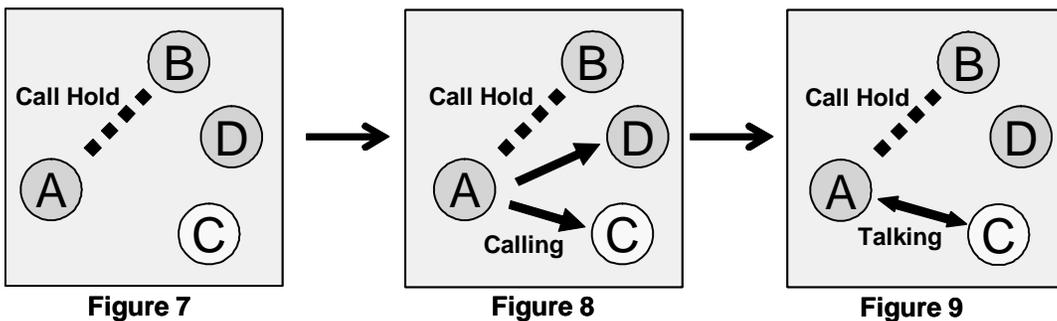
9.4. Processing Call Transfer

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

✓ *In this section, Call Hold refers to the unique Call Hold feature of Brekeke PBX, which is different from Call Hold using a REFER request (Section 9.1).*

### 9.4.1. Processing Call Transfer 1 – Attended Transfer

When User A initiates a Call Transfer, the call between Users A and B is put on hold. When User A hangs up the phone while the session between Users A and B is connecting (Figure 10, similar to Figures 6 and 9), the call between Users A and B will be transferred from User A to User C. Now Users B and C are connected.
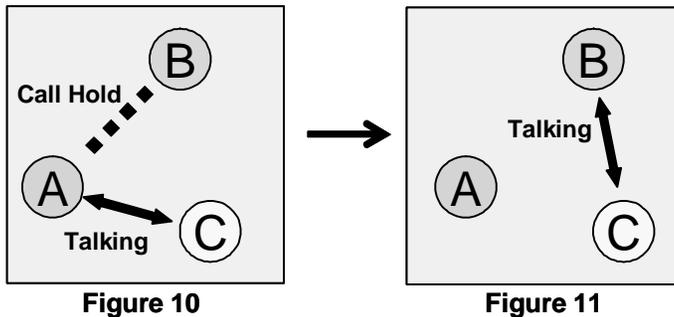
Figure 10　　　　　**Figure 11**

9.4.2. Processing Call Transfer 2 – Unattended Transfer

While the session between Users A and B is on hold, User A tries to connect with User C (Figure 12, similar to Figures 6, 9 and 10). User A hangs up the phone before User C answers the call and the call is transferred to User C, after which User B waits for User C to answer the call (Figure 13).

Figure 12　　　　　**Figure 13**　　　　　**Figure 14**
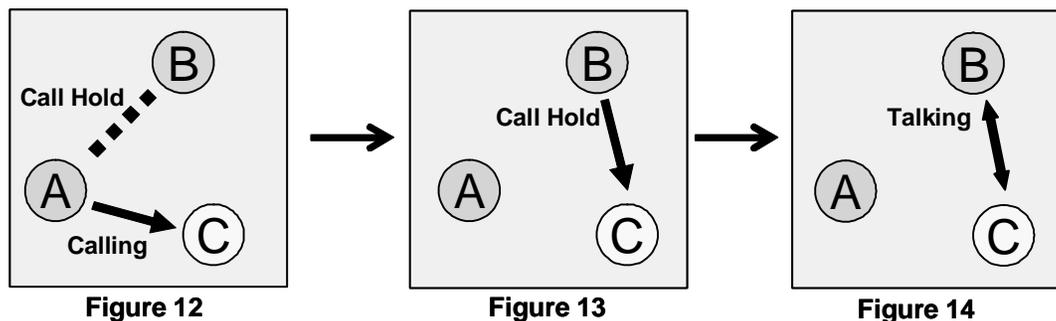
✓ *Using Unattended Transfer, if User C does not answer the phone, User A´s phone will ring to reestablish the call between Users A and B.*

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 a different destination.

✓ *In this section, Call Hold refers to the unique Call Hold feature of Brekeke PBX, which is different from Call Hold using a REFER request (Section 9.1).*

9.5.1. Canceling Call Transfer 1 – Canceling Transfer While Talking with a Third Party

There are two ways to cancel Call Transfer when you are talking with a third party:

♦ User A executes the Cancel Call Transfer command.

♦ User C hangs up the call with User A.

When the Call Transfer process between Users A and C is canceled while User B is on hold (Figure 16, similar to Figures 6, 9 and 10), the call will be put back to the hold state between Users A and B (Figure 17).

Figure 15   Figure 16   Figure 17
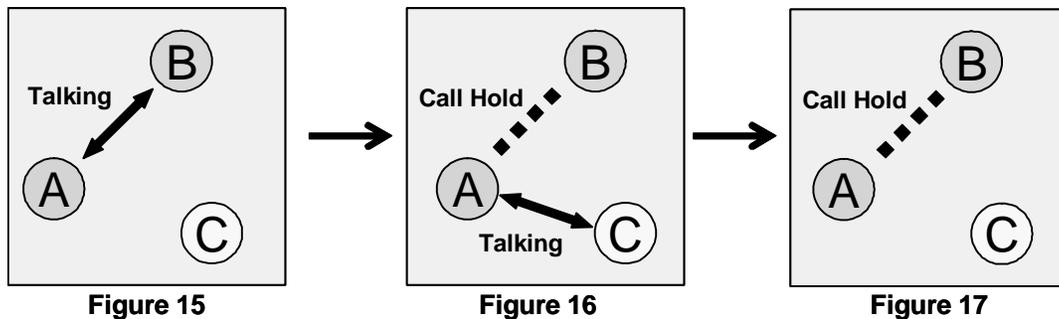
9.5.2. Canceling Call Transfer 2 – Cancel Transfer While Ringing a Third Party

There are two ways to cancel a Call Transfer while the third party's extension is ringing:

♦ User A executes the Cancel Call Transfer command.

♦ The call to User C cannot be completed for some reason, such as User C's phone is not connected, Call blocking has been set, etc.

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

| Figure 18 | Figure 19 | Figure 20 |
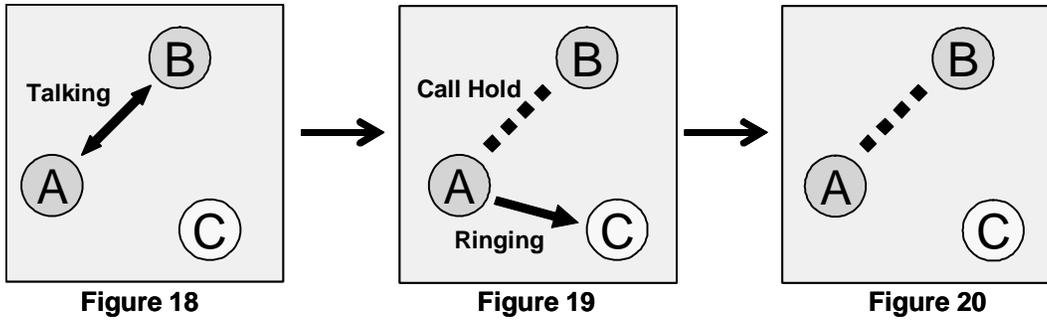
9.6.   Cautions for Call Transfer

9.6.1.   Extending Call Transfer

In the state shown in Figure 21, with regular settings, User C cannot initiate a Call Transfer for a call that has been transferred to him/her. For User C to transfer a transferred call, either user A or B first needs to hang up and then User C can transfer the call to User D.

✓ *If you need to transfer a call to a fourth party, you will need to write settings in the Dial Plan of the bundled SIP server to redirect the call from User A to C in Brekeke PBX.*

| Figure 21 | Figure 22 |
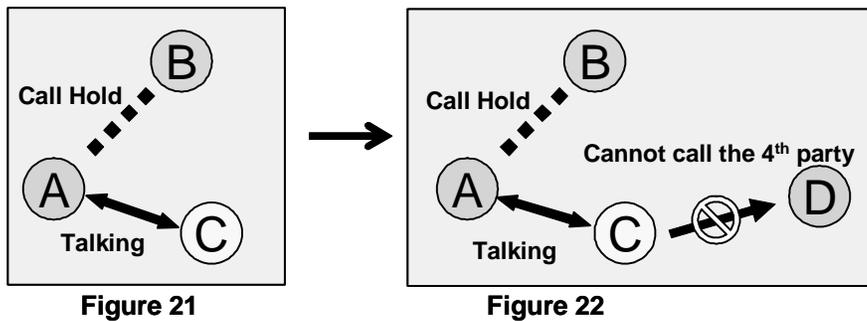
10.　Auto Attendant

Auto Attendant is a feature that interactively traffics calls without the intervention of a human operator.

10.1.　Cautions for Auto Attendant

10.1.1.　Call Transfer to Auto Attendant

When a call is transferred (Attended Transfer) to the Auto Attendant (Figure 23), Auto Attendant cannot transfer the call to a fourth party (Figure 24). Refer to section 9.6.1, "Extending Call Transfer."

Figure 23

Figure 24
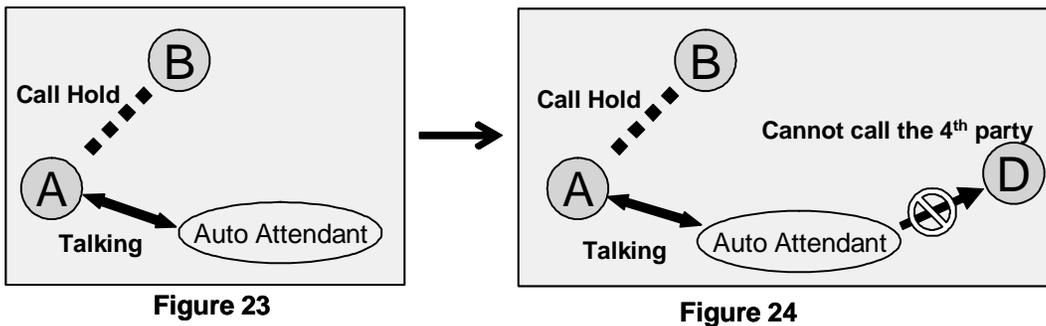
However, Call Transfer to a fourth party is possible when the original Transfer initiated from another Auto Attendant. Since Auto Attendant operates automatically and the first Call Transfer was done instantly, the second Auto Attendant can transfer the call to User D (Figures 25 and 26).

Figure 25

Figure 26
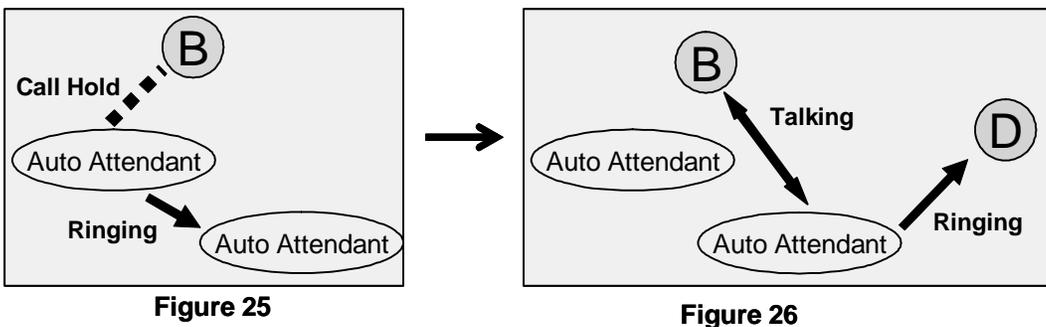
10.1.2.　Call Transfer from Auto Attendant to a Non-Brekeke PBX User

Even when [Transfer to unregistered users] is disabled under [Auto Attendant Settings], a Call Transfer from Auto Attendant can be made if the non-Brekeke PBX user's number is set at the

[Default operator] field or [Speed dial] field.

10.2. Speed Dial Settings Example for Auto Attendant

10.2.1. Simple Speed Dial Settings

Extension: 001

Auto Attendant Settings

Default operator	100
Speed dial*	6=200 7=201

Ring group extension: 200

Group Extensions*	100, 101

It can be convenient to set up [Speed dial] when using the Auto Attendant feature. For example, your Auto Attendant's 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 is pressed, the call will be immediately transferred to the appropriate destinations. Ring group extension 200 has defined multiple destinations in its [Group Extensions*] 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 any extension numbers that start with 6 or 7.*

10.2.2. Speed Dial Settings Using Regular Expressions

Extension: 001

Auto Attendant Settings

Max input digits	20
Speed dial*	6636(.+)=$1
Transfer to unregistered users	disable

When the dialed number is 6636 + <number>, the call will be transferred to <number>. Other external numbers are not allowed. Because [Transfer to unregistered users] is set to "disable," it is possible to transfer the call to an external user (such as a PSTN number), but the caller needs to know the first four digits (6636).

11. Voicemail

Only Brekeke PBX users can make use of the voicemail feature. The Brekeke PBX system treats a voicemail inbox number the same as a User ID or an extension number.

11.1. Accessing Voicemail

A user can access his/her voicemail inbox by pressing the voicemail prefix number plus his/her extension number. For details, refer to Section 6.2, "Understanding Prefixes."

Examples of accessing user extension 001's voicemail inbox:

♦ Dialing 07*001 accesses User 001's voicemail inbox directly to leave a message.

♦ Dialing 08*001 retrieves User 001's messages and allows the caller to change the voicemail inbox settings.

♦ Pressing "*" (Star) while listening to the voicemail guidance is the same as dialing 08*001.

Each user's voicemail inbox is the default destination of [Call forward (No answer)] and [Call forward (Busy)]. For details about [Forwarding destination (No answer/Busy)], refer to Section 7, "Call Forwarding."

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.

11.2. Greeting Messages

11.2.1. Recording/Uploading a Greeting Message

Users can record their personalized greeting messages through [Review/Option] (refer to a separate document called"Voicemail Navigation Map" in our website) or upload them through Brekeke PBX Admintool.

11.2.2. Selecting a Greeting Message

Users can have two personalized greeting messages: Personal Greeting and Alternative Greeting. People commonly use the "Alternative Greeting" for special occasions, such as long holidays, sick days, business trips, etc.

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

11.3. Managing Messages

The maximum length of voicemail messages that callers can leave is set in the Administrator's [Options] menu. The maximum number of voicemail messages that users can save in their inboxes is also set in the [Options] menu. Users can also set to forward voicemail messages to specific extensions or destinations. For details, please refer to Section 11.7, "Voicemail Messages: Forwarding and E-mail Notification."

11.4. Retrieving Voicemail Messages

Retrieving voicemail can be done in several ways:

♦ Download the sound file (WAV file) from the [Voice Mail] menu in Brekeke PBX Admintool and listen to it.

♦ Call your voicemail inbox and access the [Review/Option] menu.

♦ Use the e-mail notification setting and then open the sound file (WAV file) that is attached to the e-mail.

11.5. 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.

11.5.1. New Message Status

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

11.5.2. Saved Message Status

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

11.6. E-mail Notification

When a new message arrives in your voicemail inbox, an e-mail notification can be sent to a specified e-mail address.

11.6.1. Set Up E-mail Notification

Detailed settings of E-mail Notification, such as the e-mail address to receive the notification or message file, can only be set at Brekeke PBX Admintool. You can activate and deactivate e-mail notification features by calling your voicemail access number without accessing Brekeke PBX Admintool. In Brekeke PBX Admintool, you can set up your e-mail address to receive notifications and choose whether or not to receive voicemail messages as an attachment with the notifications. For details, please refer to Section 11.7, "Voicemail Messages: Forwarding and E-mail Notification."

11.6.2. Send E-mail Notification of Messages from Call Recording

Recorded messages are stored in the same location as voicemail messages. By default, e-mail notification will not be sent for messages saved with Call Recording. To send e-mail notification for recorded messages, you need to set an additional variable (described below) at [Options] > [Advanced]. Note that restarting Brekeke PBX is necessary for this change to take effect.

Variable for setting e-mail notification of call recordings:
com.brekeke.pbx.Setting.CALL_RECORDING_EMAIL_NOTIFICATION=true

11.7. Voicemail Messages: Forwarding and E-mail Notification

11.7.1. Forwarding Voicemail Messages

If you have set up Voicemail Message Forwarding at [Users] > [Settings] > [Voicemail settings] > [Message forwarding*], all voicemail messages will be forwarded to the specified destination. You can also assign multiple destinations to receive voicemail messages. If you want to keep a copy of the voicemail message, in addition to forwarding it to another destination, you will need to include your voicemail inbox number in the field as well. You can also forward voicemail messages to a group of destinations by using the forwarding settings.

11.7.2. E-mail Notification of Voicemail Messages

You may choose to receive e-mail notifications when new voicemail messages arrive in your inbox. You can also receive e-mail notifications for messages that have been forwarded to other users.

Example:

♦   User 001 sets messages to forward to User 002. Only User 002 will receive voicemail messages that come in to User 001.

♦   User 001 enters e-mail addresses for both User 001 and User 002. Both User 001 and User 002 will receive e-mail notifications when User 001 receives a voicemail message.

11.7.3.   Maximum Number of Messages

Brekeke PBX System Administrators can limit the number of messages that users can have in their voicemail inboxes in [Options] > [Media Server system setting] > [Max stored messages]. When the number of voicemail messages in the inbox reaches the maximum number, callers cannot leave new messages in your voicemail inbox. Callers will hear the system default message, "The message inbox is full."

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

11.8.   Voicemail Inbox Access Limitation

Each voicemail inbox will only allow a single user access at one time. When you or another user are accessing your voicemail inbox, no other accesses are allowed. In addition, when your voicemail inbox is being accessed by phone, your voicemail settings cannot be modified in Brekeke PBX Admintool.

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 to which Brekeke PBX connects:

♦   **Third-Party SIP Servers**

A Third-Party SIP Server is any server software other than the bundled SIP server that comes with Brekeke PBX. 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 in the [ARS] menu enables ARS to make optimal use of outside lines and of your gateways. You can define extension numbers that will receive incoming calls, as well as which lines to use for outgoing calls.

Here are some examples of uses for the ARS feature:

♦   **Select the least expensive calling options**

You could save money by sending all international calls through an inexpensive long distance carrier, or just send calls that connect to a branch office located in (for example) Taipei through that specific carrier. With the ARS feature, you can use whichever calling option is the least expensive for each call.

♦   **Effective port use of PSTN Gateways**

When there are multiple PSTN Gateways installed in an office, some ports may be used more frequently than others. You can use the ARS feature to select the most efficient 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. You can use the ARS feature to select the most efficient service provider automatically.

♦   **Network Traffic Management and Load Balancing**

When many sessions are in progress and the server load is too high or network bandwidth is insufficient, you may choose to use the ARS feature to direct some calls through PSTN Gateways.

♦ **Failover**

With ARS failover, Brekeke PBX searches for an alternate route if the specified route is unavailable or has failed, and then makes outgoing calls via the alternate route. For example, if you have multiple ITSPs (Internet Telephony Service Providers) and/or 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 will automatically switch to an analog route through the PSTN gateway. The original route can be reactivated manually from Brekeke PBX Admintool or automatically by preset settings.

12.1. Route Search Process

Two sessions are used for each call in Brekeke PBX. Please refer to Section 5, "Call Process Flow" for more details on session processing during calls.

When User A calls User B, the SIP session initiated by User A will be answered by Brekeke PBX. Brekeke PBX will then process ARS Pattern-IN for the incoming session. Brekeke PBX also initiates a session with User B and will process ARS Pattern-OUT for the outgoing session.

Route search is done by searching for Matching Patterns that apply to a call. The 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 at random. 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].

For incoming sessions, Brekeke PBX will check if any ARS route matches and will then route the session to an appropriate dial-in number. Please ensure that calls between your own extension users do not match ARS rules, because those calls do not need to be routed through your gateway and a third-party SIP provider.

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

12.2. Matching Patterns

Use regular expressions to enter conditions for the To or From headers, etc., for the matching process. The fields can be set in the same way as the [Matching Patterns] fields in the Dial Plan. If these fields are left blank, all calls will be considered to match the condition.

By changing the [User] field of an ARS Pattern-OUT setting, it is possible to restrict who can use the ARS pattern. The "User" of the call for the ARS rule will be determined as follows:

1) If the callee is a Brekeke PBX user, then that callee is considered as the "User" of the call.

2) If #1 is not applicable and the call is forwarded through a Brekeke PBX user's setting in the [Other Forwarding destinations], [Forwarding destination (No answer)] or [Forwarding destination (Busy)] field, the Brekeke PBX user will be considered as the "User" of the call.

3) If #2 is not applicable and the caller is a Brekeke PBX user, the caller will be considered as the "User" of the call.

4) If none of the above is applicable, no user will considered as the "User" of the call.

The [Plugin], [Param] and [Return] fields are used only when you want to extend the matching conditions using a JAVA plug-in. Please refer to "ARS Plug-in Developer's Guide" for more information.

12.3. Deploy Patterns

This is similar to the [Deploy Patterns] fields in the Dial Plan, however, there are several important differences:

♦ In ARS rules, numbers follow the "$" sign in [Deploy Patterns]. (Use "%" in the Dial Plan.)

♦ Conversion of a From header that matches in [Matching Patterns] is done as described in [Deploy Patterns] for the From header. Conversion of a To header is done as defined in [Deploy Patterns] for the To header, DTMF and Target field. (In Dial Plan rules, multiple lines can be defined in the [Matching Patterns] and [Deploy Patterns] fields, which are processed simultaneously.)

♦ By using &f<n>, &t<n> or &u<n> in the [Deploy Patterns] field, it is possible to use a value in parenthesis of some [Matching Patterns] fields. The symbols &f, &t and &u correspond to From, To and User, respectively. For example, &f1 refers to the value in the first parenthesis in [Matching Patterns] > [From].

♦ If the From or To field is blank, no conversion will be performed. If the DTMF field is blank, no DTMF will be sent. If the Target field is blank, no target IP address will be specified. (In most cases, the call will be routed to the domain specified in the To field.)

12.4. ARS Templates and Editing Variables

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

13.   Third-Party Call Control

Using this feature, you can connect multiple SIP UAs through an external interface. That way, you can create systems like "Click-to-Call" and "Autodialer."

Using HTTP-GET (or POST) and the Web Service interface, you can access Brekeke PBX externally. You can use HTTP-GET by doing the following:

1)   Use regular expressions to set the allowed IP patterns at Brekeke PBX Admintool > [Options] > [PBX system settings] > [Valid client IP Pattern].
2)   Access the URL (format as below) using HTTP protocol from your PC with the IP addresses set at [Valid client IP Pattern].

URL format

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

<server-ip-address>	IP address of server where Brekeke PBX is installed.
<server-port>	The same port number you use to access Brekeke PBX Admintool.
<tenant-name>	The tenant name (Multi-Tenant PBX)
<callee-number>	The callee's number or his/her SIP URI. (You can specify multiple numbers by dividing them with commas.)
<caller-number>	The caller's number or his/her SIP URI.
<caller-user>	The Brekeke PBX User that is used to call out (caller). You can omit this if it is the same as "caller-number."
<type>	Specify "1" when you want to call both destinations simultaneously. Specify "2" when you want to connect the caller before connecting the callee.

Response

Brekeke PBX will return the response: "succeeded: <ID number>." This ID will be the ID used at the [Call Status] screen for the session.