Icecom SwitchBoard

Version 3.0

Asterisk Cofiguration Manual

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PostgreSQL Database Management System
(formerly known as Postgres then as Postgres95)

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INTRODUCTION

This configuration manual was made for Icecom SwitchBoard version 3.0, for configuring Asterisk. The starting point for the manual is that the Icecom SwitchBoard software and Asterisk be already installed on the computer. The installation instructions for the Icecom SwitchBoard software can be found in a separate manual.

This document explains what the Asterisk server configuration files must include in order for the Asterisk message to be transmitted to the Icecom SwitchBoard.

All the files to be configured can be found in the file ***/etc/asterisk***.

In the manual, text marked with the Verdana font signifies lines to be entered in the configuration files. Computer-specific variables for the configuration have been marked inside tags <> and comment lines have been marked to start with a semicolon ;. In addition, the names of files and directories have been written *in italics*.

At the end of the document, there are the example files for a configuration, in which there are 20 extensions besides the switchboard attendant (appendices 1 – 6)

Terms used commonly in the manual:

context	*extensions.conf* file element marked inside square brackets. A context ends when new square brackets begin.
extension (exten)	A function marked for a single extensions in the *extensions.conf* file. An extension begins with the character sequence ”exten =>”, which is followed by an ID identifying the extension and then the priority (=executing order) and function divided by a comma. (for example exten => 200,1,Dial(SIP/200))

1 MANAGER.CONF

The Asterisk is given the direction to use the manager interface from port 5038.

```
[general]
enabled = yes
port = 5038
```

Add user manager, whose username is **login** and password is **passwd** (**<u>Note!</u>** Username and password must not be changed!), and the rights to use the functions set:

```
[login]
secret = passwd
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
```

2 EXTENSIONS.CONF

2.1 Setting up a common prefix to supplement extensions

A common prefix for the extensions is set in the beginning of the file in [**globals**] context. The prefix is set as the variable ”start”.

```
[globals]
start=<common prefix for extensions>
; the common prefix can be for example 5277
```

The prefix is used later when setting an extension for placing an outside call (chapter 2.4.7).

2.2 Setting up queue music

The accompanying three files can be set as the queuing music:

- *fur-elise.gsm*
- *turkishm.gsm*
- *bourree-pizzicato.gsm*

The files must be saved in the directory */var/lib/asterisk/sounds/***.**

For the queuing music, extensions are created in the file’s [**default**] context.
There are four extensions:

- **music**: for incoming calls queue
- **hold**: for queued calls queue
- **nite**: for night switching
- **busy**: for busy line queue

Any personal messages/recordings saved in the system must be files of the .gsm type, and they must reside in the directory */var/lib/asterisk/sounds*.

```
[default]
exten => music,1,Answer
exten => music,2,Playback(you_are_in_the_queue)
;	you_are_in_the_queue-wait_a_moment: own recording
exten => music,3,Playback(fur-elise)
exten => music,4,Playback(turkishm)
exten => music,5,Playback(bourree-pizzicato)
exten => music,6,Goto(music|3)

exten => hold,1,Answer
exten => hold,2,Playback(receiver_busy)
;	receiver_busy: own recording
exten => hold,3,Goto(music|3)
;	caller is put on hold

exten => nite,1,Answer
exten => nite,2,Playback(we_are_open_from_8_to_16)
;	we_are_open_from_8_to_16: own recording
exten => nite,3,Hangup

exten => busy,1,Answer
```


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```
exten => busy,2,Playback,(person_is_busy)
;        person_is_busy: own recording
exten => busy,3,Hangup
exten => busy,400,Busy
```

2.3 Setting up attended transfers between external calls

When wanting to support attended transfer between two external calls in the system, an extension named **conference** must be created in the file's [**default**] context.

```
exten => conference,401,Meetme(1|q)
exten => conference,501,Meetme(2|q)
exten => conference,601,Meetme(3|q)
```

Please note!
The priorities of the "conference" extension must be the same as the priorities set in the meetme table in the swb database of postgresql. By default, four priorities have been set up in the database: 401, 501 and 601.

2.4 Setting up extensions

Extensions can be set up for two different types of users:

- switchboard attendants
- web users

All extensions (except macro) are placed under the [**default**] context by default.

2.4.1 Extension for switchboard attendant

There are two lines in a switchboard attendant's extension. On the first one the attendant's phone rings. On the second one, when the attendant is busy (priority 102), the call is transferred to a call queue titled "music". (Defined in chapter 2.2).

Example 1. Extension for two switchboard attendants (extensions 201 and 202):

```
exten => 201,1,Dial(SIP/${EXTEN})
exten => 201,102,Goto(music|2)
;
exten => 202,1,Dial(SIP/${EXTEN})
exten => 202,102,Goto(music|2)
```

2.4.2 Extension for virtual switchboard attendant

When calling a joint number for two switchboard attendants, both their extensions will ring at the same time. The numbers must be defined as explained in chapter 2.4.1. Then an individual physical phone device is not required for the joint number.

Example 2. When calling joint number 200, extensions 201 and 202 ring:

```
exten => 200,1,Dial(SIP/201&SIP/202)
exten => 200,102,Goto(music|2)
```

2.4.3 Extension for web user

A web user's extension is defined by one user-specific line, stating the user's extension number, referring to the common macro between the users and defining other user-specific values.

<u>Example 3.</u> Defining a web user's extension:

```
exten => 203,1,Macro(sipcall,20,200)
```

In example 3, the user's extension number is 203 and sipcall refers to the macro defined in example 4. 20 is the time that the phone rings (in example 4 $ {ARGS1}), and 200 the number to which the call is transferred if the number is busy in example 4 ${ARGS2}).

2.4.4 Macro context shared by all web users

A macro is placed into its own context, where its name is put in square brackets [ ] and started by the prefix **macro**.

<u>Example 4.</u> Macro named sipcall:

```
[macro-sipcall]
exten => s,1,Dial(SIP/${MACRO_EXTEN}|${ARGS1})
exten => s,2,Wait(1)
exten => s,3,Voicemail(u${MACRO_EXTEN} )
;       alternatively queue transfer can be used:
;       exten => s,3,Goto(default, ${ARGS2},1)
exten => s,102,Wait(1)
exten => s,103,Goto(default, ${ARGS2},1)
;       default means the [default] context defined in the file, where
;       the switchboard attendant's extension has been set by default
```

2.4.5 Extensions for call transfers

Transferring a call requires own extension. Extensions must be included for every call type, SIP- and ZAP –calls. Priority must not be changed!

```
exten => _2XX,123,Macro(sip-transfer,${EXTEN},20)

[macro-sip-transfer]
exten => s,1,Dial(SIP/${ARG1}|${ARG2})
exten => s,2,Goto(music|3)
exten => s,102,Goto(music|3)

exten => _XXX.,123,Macro(zap-transfer-out,${EXTEN},20)

[macro-zap-transfer-out]
```


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```
exten => s,1,SetCallerId,${CALLERIDNUM}
exten => s,2,Dial(ZAP/g1/${ARG1}|${ARG2})
exten => s,3,Goto(music|3)
exten => s,103,Goto(music|3)
```

2.4.6 Attended transfer to internal number

The extension for an attended transfer is presented with a single line. The first number to be placed after the underline character refers to the first number of the whole extension group. In example 5, 2XX refers to extensions 200 – 299 (i.e. the number sequence for extensions 100 – 199 would be 1XX, for extensions 300 – 399 3XX, etc.)

Example 5. Attended transfer set for extensions 200 – 299:

```
exten => _2XX,401,Meetme(${EXTEN}|q)
```

2.4.7 Extension for a call placed outside

The extension defined for a call to be placed outside is marked with the **_XXX.** symbol sequence **(Note!** a dot at the end of the extension). Then the number to be called must be at least 3 digits long and may contain the numbers 0 – 9.

In example 6, it is first checked if the call is placed from Icecom SwitchBoard, and then the caller name is changed if necessary. The actual call is placed in priorities 4 or 8. In priority 400, the call is transferred to the switchboard attendant, if it could not be made in the previous stage. In the example, the variable ”start” is a four-character sequence (see chapter 2.1), and ”CALLERIDNUM:0:4” is marked in priority 6. If the length of the variable is something else, the length is marked in priority 6.

Example 6. Call placed outside:

```
exten => _XXX.,1,GotoIf($[${CALLERIDNUM} = asteriskCaller]?2:6)
exten => _XXX.,2,AGI,Lookup.agi  ;see chapter 2.4.7
exten => _XXX.,3,SetCallerId,${start}${CIDNUM}  ;start: see chapter 2.1
exten => _XXX.,4,Dial,Zap/g1/${EXTEN}
exten => _XXX.,5,Goto(${EXTEN}|400)
exten => _XXX.,6,GotoIf($[${CALLERIDNUM:0:4} = ${start}]?8:7)
exten => _XXX.,7,SetCallerId,${start}${CALLERIDNUM}
exten => _XXX.,8,Dial,Zap/g1/${EXTEN}
exten => _XXX.,9,Goto(${EXTEN}|400)
exten => _XXX.,400,Goto(200|1)
```

2.4.8 Lookup.agi for outgoing calls

The `Lookup.agi` script changes the caller’s callerId from one called asteriskCaller into the caller’s own, when the call is placed from the Icecom SwitchBoard. If this script is not executed, asteriskCaller is shown as the callerId of all calls started from the Icecom SwitchBoard.

Lookup.agi and Lookup.class and files must be copied to directory /var/lib/asterisk/agi-bin/. Switchboard.properties file must be copied to directory /etc/asterisk/. Switchboard.properties file includes Icecom SwitchBoard's internal user agent's name (asteriskCaller by default). Please check the access rights for the files (required commands are "chmod a+rxw Lookup.agi" and "chmod a+rxw Lookup.class"). These three files can be downloaded from the download-section of Icecom SwitchBoard website (www.icecomswitchboard.com).

Check JAVA_HOME variable from Lookup.agi –file.

2.4.9 Changing SBSMM –user's presence status

Users that have been added through Exel-file have SBSMM- web user interface in their use. This web user interface enables users to change their presence status. SBSMM users can also change their presence status by using SIP –phone. Below are the instructions how to use this function in question.

Following lines are added to the *Extensions.conf* -file:

```
;	Available
exten => *01,1,Answer
exten => *01,2,Playback,
;	(Changing presence status into "Available"): own recording
exten => *01,3,Hangup

;	Busy
exten => *02,1,Answer
exten => *02,2,Playback,
;	(Changing presence status into "Busy"): own recording
exten => *02,3,Hangup
;
exten => occupied,1,Answer
exten => occupied,2,Playback,
;	(Person's presence status is "Busy"): own recording
exten => occupied,3,Goto(sip-call,8060,1)

;	Away
exten => *03,1,Answer
exten => *03,2,Playback,
;	(Changing presence status into "Away"): own recording
exten => *03,3,Hangup
;
exten => away,1,Answer
exten => away,2,Playback,
;	(Person's presence status is "Away"): own recording
exten => away,3,Goto(sip-call,8060,1)

;	Out to lunch
exten => *04,1,Answer
exten => *04,2,Playback,
;	(Changing presence status into "Out to lunch"): own recording
exten => *04,3,Hangup
;
exten => lunch,1,Answer
```


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```
exten => lunch,2,Playback,
;         (Person's presence status is "Out to lunch"): own recording
exten => lunch,3,Hangup

;         Do not disturb
exten => *05,1,Answer
exten => *05,2,Playback,
;         (Changing presence status into "Do not disturb"): own recording
exten => *05,3,Hangup
;
exten => dnd,1,Answer
exten => dnd,2,Playback,
;         (Person's presence status is "Do not disturb"): own recording
exten => dnd,3,Goto(sip-call,8060,1)

;         Be right back
exten => *06,1,Answer
exten => *06,2,Playback,
;         (Changing presence status into "Be right back"): own recording
exten => *06,3,Hangup
;
exten => brb,1,Answer
exten => brb,2,Playback,
;         (Person's presence status is "Be right back"): own recording
exten => brb,3,Hangup
```

Please note!
In the database's sbsmmextensions -board's messageextension -column's additions needs to be named in the same way as they are in extension-lines in *extensions.conf* –file.

Example 8. Lunch extension:

```
exten => lunch,1,Answer
exten => lunch,2,Playback,
          (Person's presence status is "Out to lunch"): own recording
;         exten => lunch,3,Hangup
```

Please note!
In the database's sbsmmextensions -board's extension-column's additions needs to be named in the same way as they are in extension-lines in *extensions.conf* –file.

Example 9. *06 extension:

```
exten => *04,1,Answer
exten => *04,2,Playback,
;         (Changing presence status into "Out to lunch"): own recording
exten => *04,3,Hangup
```

Please note!
It is not allowed to change the information in the database's sbsmmextensions –board's status -column.

Example 10. sbsmmextensions-board:

extension	status	messageextension
*01	available	
*02	busy	occupied
*03	away	away
*04	lunch	lunch
*05	dnd	dnd
*06	brb	brb

When a user calls the number in the Extension- column she/he hears saved recordings. After the recording call ends and user's presence status changes.

When a user has changed her/his presence status and she/he receives a call, the caller hears a recording matching user's presence status which has been added to messageextension – column. After playing recording the call can be defined to end or to transfer to switchboard's number.

Please note!
Contents in the extension- and messageextension –columns can be named in any chosen way.

3 SIP.CONF

Create an own context for Icecom SwitchBoard's internal user agent (by the same name as set in the database, asteriskCaller by default).

```
[asteriskCaller]
type=friend
host=dynamic
insecure=yes
```

An individual context must be created for every user (switchboard attendants and web users), with the user's extension number as the context name. The variable "secret" is a password with which the user's phone registers to Asterisk.

```
[name]
type = friend
secret = password
host = dynamic
nat = yes
```

Please note!

When wanting to change the name of the user agent called asteriskCaller into another one, the same change must be made also to the asterisksettings table of the postgresql `swb` database.

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4 CDR_PGSQL.CONF

In the file *Cdr_pgsql.conf*, insert the IP address where the database resides and the super-user set up for it. By default the username is postrgres, with **no password**. The name of the database is *swb*.

```
[global]
hostname=<database IP address>
dbname=swb
password=
user=postgres
port=5432
```

This configuration gives the Asterisk the command to use cdr table in the PostgreSQL database called *swb*.

5 MEETME.CONF

For attended transfer, add a conference number for every user to the *meetme.conf* file.
(See user priority in example 5.)

Example 11. Add user extension numbers 203 and 204:

```
conf => 203
conf => 204
```

Example 12. Add own attended transfer numbers for calls coming in from the outside:

```
conf => 1
conf => 2
conf => 3
conf => 4
```

6 MODULES.CONF

The use of the PostgreSQL database is added to the modules used by the Asterisk server. Existing configurations must be kept in the file and the following ones added:

```
[modules]
load => cdr_pgsql.so
…

[global]
cdr_pgsql.so=yes
…
```

APPENDIX 1. Example of meetme.conf file configuration

```
[rooms]
conf => 200
conf => 201
conf => 202
conf => 203
conf => 204
conf => 205
;
conf => 1
conf => 2
conf => 3
```

APPENDIX 2. Example of cdr_pgsql.conf file configuration

```
[global]
hostname=213.255.100.10          ; change IP address
dbname=swb
password=
user=postgres
port=5432
```

APPENDIX 3. Example of modules.conf file configuration

```
; Asterisk configuration file
;
; Module Loader configuration file
;
[modules]
autoload=yes
noload => pbx_gtkconsole.so
noload => pbx_kdeconsole.so
noload => app_intercom.so
noload => chan_oss.so
load => cdr_pgsql.so
load => res_parking.so
;
[global]
cdr_pgsql.so=yes
```

APPENDIX 4. Example of manager.conf file configuration

```
; Asterisk Call Management support
;
[general]
enabled = yes
port = 5038
;
[login]
secret=passwd
read=system,call,log,verbose,command,agent
write=system,call,log,verbose,command,agent
```


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