## Change History

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<th>Change Made</th>
<th>Author</th>
<th>Authorised</th>
<th>Date</th>
</tr>
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<td>1.0</td>
<td>Initial document</td>
<td>AH</td>
<td></td>
<td>18/09/2012</td>
</tr>
<tr>
<td>1.1</td>
<td>Added announcements</td>
<td>ZL</td>
<td></td>
<td>18/02/2013</td>
</tr>
<tr>
<td>1.2</td>
<td>Updated configuration of Voicemail</td>
<td>YL</td>
<td></td>
<td>26/07/2013</td>
</tr>
<tr>
<td></td>
<td>Added Extra Configuration of IVR</td>
<td></td>
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<td></td>
</tr>
<tr>
<td>1.2</td>
<td>Updated Format and added Document Number</td>
<td>DR</td>
<td>AC</td>
<td>12/02/2014</td>
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1.0 Quick Start Guide

This document will demonstrate how to setup an Asterisk server to use realtime voicemail through mysq and IVR. With OBDC storage for voicemail.
2.0 Installation

1. **Cent 5.x**
   You will need to add the Asterisk repository for CentOS from the command line run the following and accept any prompts:
   ```bash
   rpm -Uvh http://packages.asterisk.org/centos/5/current/i386/RPMS/asterisk-know-version-1.7.1-3_centos5.noarch.rpm
   
   Now the repositories are set up the packages required for the system can be installed:
   ```bash
   yum install nano mysql-server mysql asterisk asterisk-configs asterisk-addons-mysql asterisk-addons-core --enablerepo=asterisk-10
   ```

2. **Cent 6.x**
   ```bash
   cd /etc/yum.repos.d/
   wget http://repo.squire-technologies.co.uk/Squire-Testing.repo
   wget http://packages.asterisk.org/centos/centos-asterisk-11.repo
   wget http://packages.asterisk.org/centos/centos-asterisk.repo
   nano Squire-Testing.repo
   Change enabled from 0 to 1
   nano centos-asterisk-11.repo
   Change enabled from 0 to 1
   nano centos-asterisk.repo
   Change enabled from 0 to 1
   ```

3. **We now need to add a connected for mysql and odbc**
   ```bash
   cd /etc
   nano odbc.ini
   ```
   ```ini
   [asterisk]
   Description = MySQL connection to 'asterisk' database
   Driver = MySQL
   Database = svi_ms
   Server = localhost
   UserName = root
   Password =
   Port = 3306
   Socket = /var/lib/mysql/mysql.sock
   ```

4. **If using the examples from asterisk, there will already be sample configs that can be used for reference and modified located in “/etc/asterisk”**.
   Enter the directory:
   ```bash
   cd /etc/asterisk
   ```
   Open the following file for editing (You can use your preferred text editor, in these examples we will use nano).
   ```bash
   nano resconfig_mysql.conf
   ```
   Add the following to the end of the file:
This will setup the database that Asterisk will connect to, in this case it will use a local database, with the name svi_ms. We will define which tables to use in another file.

5. We will now add a trunk to the SVI to allow calls to pass in to it:
   nano sip.conf
   Then add the following to the end of the file, the “host=” will need to be changed to the IP of the SVI you are setting up.

   ```
   [vm]
   host=192.168.2.188
   type=peer
   context=vm
   rtcachefriends=yes
   disallow=all
   allow=ulaw
   allow=ilbc
   allow=alaw
   ```
   And change “udpbindaddr:
   `udpbindaddr=192.168.2.188:5061`

6. We will now tell Asterisk that it can use the realtime engine so it can search the database, we will come back to this file later to add some rules.
   nano extensions.conf
   Find the [general] tag and add the following directly under:

   ```
   [internal]
   switch => Realtime/@
   ```

7. Next we will set Asterisk to tell it to use the realtime mysql engine for voicemail only:
   nano extconfig.conf
   Add the the line below to the end of the file. This sets voicemail to pass voicemail requests to mysql using “asterisk” from res_config_mysql.conf for settings, and finally use the table called voicemails:

   `voicemail => mysql,asterisk,voicemail`
8. Next delete the voicemail.conf as we won’t require almost anything from this file, then recreate it with only the lines required. This just tells it to search elsewhere for voicemail information, ie the database that will be created:

```bash
rm voicemail.conf
nano voicemail.conf
```

Now add the following 2 lines:

```
[general]
searchcontexts=yes
```

9. `nano mgcp.conf`

Un-comment “port” under [general] and change port from 2427 to 2728:

```
[general]
Port = 2728
;bindaddr = 0.0.0.0
```

10. **SKIP THIS STEP IF USING C5 MYSQL - ONLY NEEDED IF ASTERISK RUNNING ON SPERATE BOX.**

Now we need install the SQL data, copy the following file to your / folder box using winscp, `\\linux\iscsi\linux\workspaces\Alex\Asterisk\voicemail.sql` and run:

```
service mysqld start
mysql
create database svi_ms;
exit;
cd /
mysql svi_ms < voicemail.sql
```

11. Everything is now setup and ready to use the mysql database for voicemail users, start the asterisk service by running:

```
service asterisk start
```
3.0 Config, and Useful Commands

1. You will probably now need to test the system and set up a couple of rules to allow numbers to be passed to the voicemail system. In /etc/asterisk/extensions.conf we can set dialling rules, you can add this just to the end of the config. The below example sets up the number “118118” to have calls that are unanswered or if busy to be sent to voicemail.

   [vm] ;this should match the context in sql
   exten => 118118,1,Set(TARGETNO=${EXTEN}) ;use _XXXXXX for any 6 digit number
   exten => 118118,n,Dial(SIP/${EXTEN},30)  
   exten => 118118,n,Goto(s-${DIALSTATUS},1) ; routes the call to the status priority (NOANSWER,BUSY,CHANUNAVAIL,CONGESTION,ANSWER)
   exten => s-NOANSWER,1,VoiceMail(${TARGETNO},u) ; Person at extension "is unavailable" message
   exten => s-BUSY,1,VoiceMail(${TARGETNO},b)     ; Person at extension "is busy" message
   exten => s-ANSWER,1,Hangup()                     ; To be safe, clean up the call after an answer by hanging up
   exten => _s-,1,Goto(s-NOANSWER,1) ; Handle any unhandled status the same way we handle NOANSWER

In my box, all the subscriber numbers are 8 digits, so use ”_XXXXXXXX” for any 8 digit number.

2. In some cases you will want to setup such rules for all numbers in this case change the numbers for X for example:

   exten => _XXXXXX,1,VoicemailMain

This would send all 6 digit numbers dialled in to the voicemail server to check messages, it would also prompt the user for both a username and password. In most cases if a customer is checking there voicemail it will be from there own phone, so we can avoid them having to enter a username and just prompt for a password.

Dial 4497 to receive voicemail (4497 is within my number plan). Create a subscriber 4497 in GUI (Free on net and Disable credit check).

   exten => 4497,1,VoiceMailMain(${CALLERID(num)})

Change [public]
include => vm

So now anyone with voicemail account can dial 4497 which will match their caller ID and prompt just for a password.

The final extensions.conf should look like:
[vm] ; this should match the context in sql
exten => _XXXXX,1,Set(TARGETNO=${EXTEN}) ; use _XXXXXX for any 6 digit number
exten => _XXXXX,n,Dial(SIP/${EXTEN},30)

exten => _XXXXX,n,Goto(s-${DIALSTATUS},1) ; routes the call to the status priority
(NOANSWER,BUSY,CHANUNAVAIL,CONGESTION,ANSWER)

exten => s-NOANSWER,1,VoiceMail(${TARGETNO},u) ; Person at extension "isunavailable" message

exten => s-BUSY,1,VoiceMail($TARGETNO,b) ; Person at extension "is busy" message

exten => s-ANSWER,1,Hangup() ; To be safe, clean up the call after an answer by hanging up

exten => _s-,1,Goto(s-NOANSWER,1) ; Handle any unhandled status the same way we handle NOANSWER

exten => 4497,1,VoiceMailMain(${CALLERID(num)})

3. nano /etc/my.cnf
   Add binlog_format = ROW
   The modified file should look like:
   Then restart mysql:
   service mysqld restart
4. Now we will add the odbc connection to Asterisk:

```bash
rm res_odbc.conf
nano /etc/asterisk/res_odbc.conf
```

```
[ENV]

[asterisk]
enabled => yes
dsn => asterisk
;username => myuser
;password => mypass
pre-connect => yes
```

5. Asterisk CLI

At times you may need access the Asterisk CLI to check, reload config or enable debug. You can access the asterisk CLI by running:

```bash
asterisk -r
```

Below are a list of possible useful commands, this is a very small selection of the available commands.

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>reload</td>
<td>Will reload the config files, without restarting the system.</td>
</tr>
</tbody>
</table>
### 5.6. GUI configuration

#### 6.1 VoIPStack

As RTPROUTER is always used in C5 system, we need to turn on Proxy Media in VoIPStack.

In GUI: Configuration -> Resources -> VoIPStack

Turn on Proxy Media in all the VoIPStacks shown here.

<table>
<thead>
<tr>
<th>T1</th>
<th>500</th>
</tr>
</thead>
<tbody>
<tr>
<td>T2</td>
<td>4000</td>
</tr>
<tr>
<td>T4</td>
<td>5000</td>
</tr>
<tr>
<td>D</td>
<td>32000</td>
</tr>
<tr>
<td>Tb</td>
<td>2000</td>
</tr>
<tr>
<td>Tf</td>
<td>32000</td>
</tr>
<tr>
<td>Th</td>
<td>32000</td>
</tr>
<tr>
<td>Tj</td>
<td>32000</td>
</tr>
</tbody>
</table>

- ForceSipExpiry
- SipExpiry: 3600
- Proxy Media: Enabled
- Contact Header: No User
- RPID Header: Enabled
- Append CallID
- ExpiryGrace: 10
- Description

#### 6.2 Create Customer Voicemail in Routing Tab

In GUI, setup a customer called “Voicemail” in Routing with the configuration as follows. Configure multiple IP Addresses: one is for Preferred box and the other is for Non-Preferred.
6.3 Add a routing to Voicemail.
6.4 Create a subscriber with voicemail enabled:
4.0 Adding Announcement Extensions

1. Open extensions.conf

2. Add the lines below to the bottom of the file under the voicemail extensions.

   exten => Announcement num,1,Ringing()
   exten => Announcement num,n,Progress()
   exten => Announcement num,n,Playback(Audio file,NOANSWER)

3. Change the announcement num in the config to be the number you have configured for one of your announcements.

4. Change the Audio file to be the name of the audio file which will be your announcement. You can either specify the direct path to your sound file or place it in the default sounds directory which is /var/lib/asterisk/sounds/.

5. Repeat steps 2 to 4 for the other announcements.
5.0 Extra Configuration for IVR

1. Media Gateway

   In GUI: Configuration -> Resources -> MediaGateway

   Change the value of RTO to 1000 in all the gateways shown here.

   ![Edit MediaGateway Resource](image)

2. RTPROUTER Port Range

   As the default Asterisk RTP port range is from 10000 to 20000, we need to change our RTPROUTER interface port range to avoid the conflicts. This is already done in RPM but should be double checked in the system.

   In CentOS,
   
   ```
   cd /home/squire/rtprouter
   nano RtpRouter.cfg
   ```
Change the value Interface 0 StartPort from 16000 to 28000 and the value of EndPort from 18000 to 30000. Also do the same thing to Interface 1.

<table>
<thead>
<tr>
<th>System 0 Name</th>
<th>System 0 license</th>
<th>System 0 EndpointName</th>
<th>System 0 loop</th>
</tr>
</thead>
<tbody>
<tr>
<td>RTP Router</td>
<td>6020A#520#2060#</td>
<td>SVIGateway</td>
<td>1</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Media Gateway 0 Status</th>
<th>Media Gateway 0 Type</th>
<th>Media Gateway 0 End</th>
<th>Media Gateway 0 IpSocket</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ins</td>
<td>MGCP Media Gateway</td>
<td>Media Gateway</td>
<td>1</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Media Gateway 0 Name</th>
<th>Media Gateway 0 Version</th>
<th>Media Gateway 0 THIST</th>
<th>Media Gateway 0 TMAX</th>
<th>Media Gateway 0 Max1</th>
<th>Media Gateway 0 Max2</th>
<th>Media Gateway 0 LONGTRAN</th>
<th>Media Gateway 0 RIO</th>
</tr>
</thead>
<tbody>
<tr>
<td>SVI-RIP-0</td>
<td>MGCP 1.0</td>
<td>30</td>
<td>20</td>
<td>5</td>
<td>7</td>
<td>5</td>
<td>200</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Media Gateway 0 ResolveDNS</th>
<th>Media Gateway 0 rxCommandTask</th>
<th>Media Gateway 0 destaddress</th>
<th>Media Gateway 0 destport</th>
</tr>
</thead>
<tbody>
<tr>
<td>NeverResolve</td>
<td>Router</td>
<td>192.168.8.122</td>
<td>2427</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Interface 0 Status</th>
<th>Interface 0 Type</th>
<th>Interface 0 MediaGW</th>
<th>Interface 0 StartPort</th>
<th>Interface 0 EndPort</th>
<th>Interface 0 Address1</th>
<th>Interface 0 Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ins</td>
<td>MG Ethernet</td>
<td>0</td>
<td>28000</td>
<td>30000</td>
<td>192.168.8.120</td>
<td>0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Interface 1 Status</th>
<th>Interface 1 Type</th>
<th>Interface 1 MediaGW</th>
<th>Interface 1 StartPort</th>
<th>Interface 1 EndPort</th>
<th>Interface 1 Address1</th>
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</tr>
</thead>
<tbody>
<tr>
<td>Ins</td>
<td>MG Ethernet</td>
<td>0</td>
<td>28000</td>
<td>30000</td>
<td>192.168.9.120</td>
<td>1</td>
</tr>
</tbody>
</table>

3. **dialplan.xml**

Asterisk should always listen to the local IP Address. So in the dialplan.xml, the value of “bindaddr” should be set to bond0. In “channels”, the value of “host” should be the VIP address of the system if you are using a redundant C5 system.