Asterisk Voicemail and IVR Install

Document SQ01507

Version 1.2

Change History

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

Contents

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:
   rpm -Uvh http://packages.asterisk.org/centos/5/current/i386/RPMS/asterisknow-version-1.7.1-3_centos5.noarch.rpm
   Now the repositories are set up the packages required for the system can be installed:
   yum install nano mysql-server mysqll asterisk asterisk-configs asterisk-addons-mysql asterisk-addons-core --enablerepo=asterisk-10

2. **Cent 6.x**

   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
   /cd /etc
   nano odbc.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:
   cd /etc/asterisk
   Open the following file for editing (You can use your preferred text editor, in these examples we will use nano).
   nano res_config_mysql.conf
   Add the following to the end of the file:

```
[asterisk]
dbhost = 127.0.0.1
dbname = svi_ms
dbuser = root
dbpass =
dbport = 3306
dbsock = /tmp/mysql.sock
dbcharset = latin1
requirements=warn ; or createclose or createchar
```

This will setup the 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:
   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\vociemail.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/extenstions.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
"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
```

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 extenstions.conf should look like:

```
[vm]  ;this should match the context in sql
exten => _XXXXXXXX,1,Set(TARGETNO=${EXTEN}) ;use _XXXXXX for any 6 digit number
exten =>  _XXXXXXXX,n,Dial(SIP/${EXTEN},30)
exten =>  _XXXXXXXX,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 => _XXXXXXXX,1,VoicemailMain
exten => 4497,1,VoiceMailMain(${CALLERID(num)})
```

[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

```
[mysqld]
datadir=/var/lib/mysql
socket=/var/lib/mysql/mysql.sock
user=mysql
# Default to using old password format for compatibility with mysql 3.x
# clients (those using the mysqlclient10 compatibility package).
old_passwords=1

server-id = 1
replicate-same-server-id = 0
auto-increment-increment = 2
auto-increment-offset = 1

master-host = 192.168.2.187
master-user = slave1_user
master-password = squireSVI
master-connect-retry = 60
replicate-do-db = svi_ms

slave-skip-errors = 1062

log-bin = /var/log/mysql/mysql-bin.log
binlog-do-db = svi_ms
binlog_format = ROW

relay-log = /var/lib/mysql/slave-relay.log
relay-log-index = /var/lib/mysql/slave-relay-log.index

expire_logs_days         = 10
max_binlog_size          = 500M

[mysqld_safe]
log-error=/var/log/mysqld.log
pid-file=/var/run/mysqld/mysqld.pid
```

4. Now we will add the odbc connection to Asterisk:
   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:
   asterisk -r

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

Command	Description
reload	Will reload the config files, without restarting the system.

voicemail show users for vm	Will output all users for the context "vm", you can change this to any context you use.
sip set debug on	Enables sip debug.
dialplan debug vm	Shows dialplan for "vm", you can change this for your dialplan.
core set debug 4	Sets asterisk debug to level 4 (you can set this between 0-4)
exit	Takes you back to the linux shell, it will not stop asterisk.

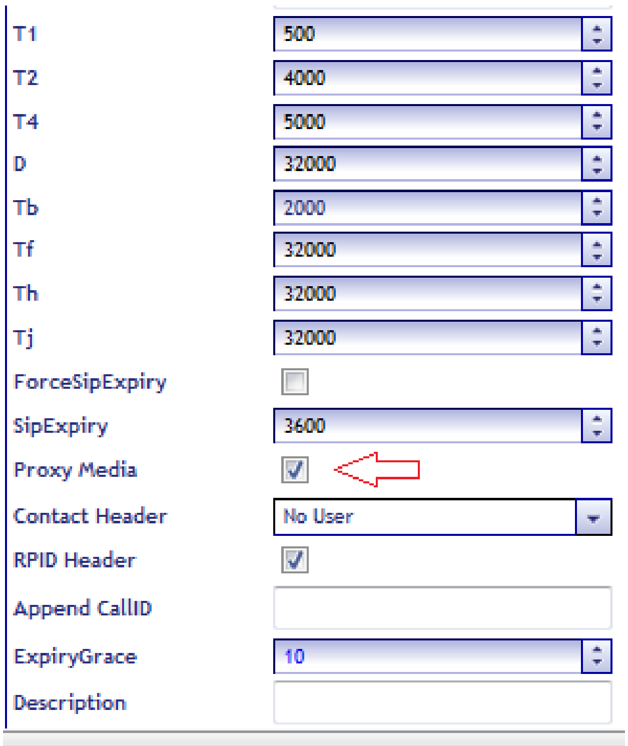
6. GUI configuration

6.1 VoIPStack

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

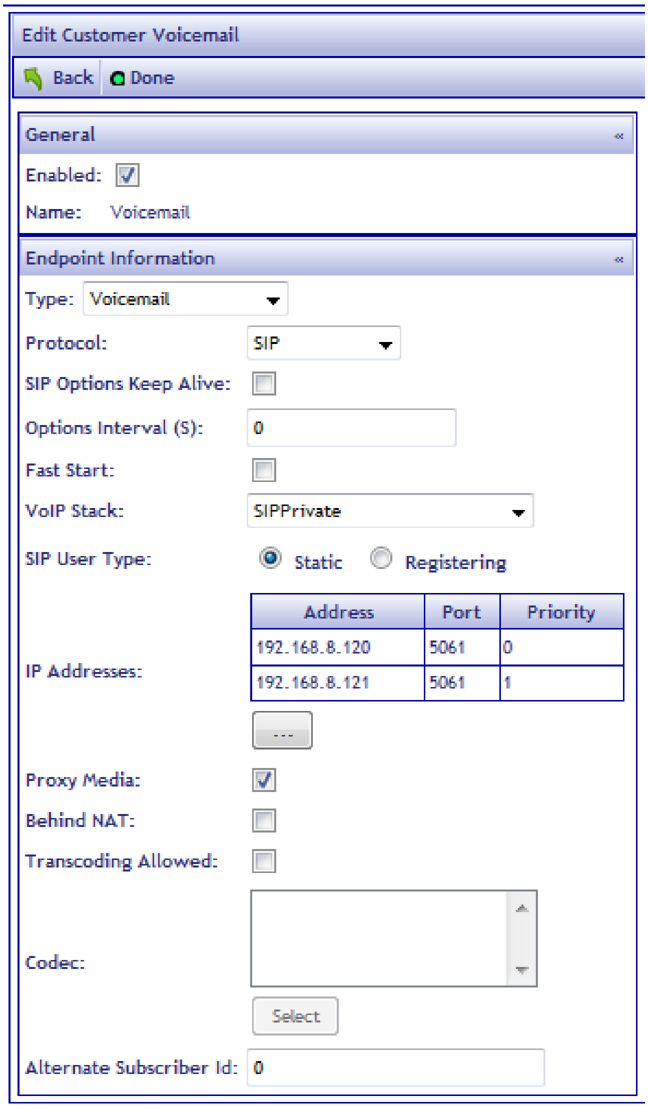
In GUI: Configuration -> Resources -> VoIPStack

Turn on Proxy Media in all the VoIPStacks shown here.

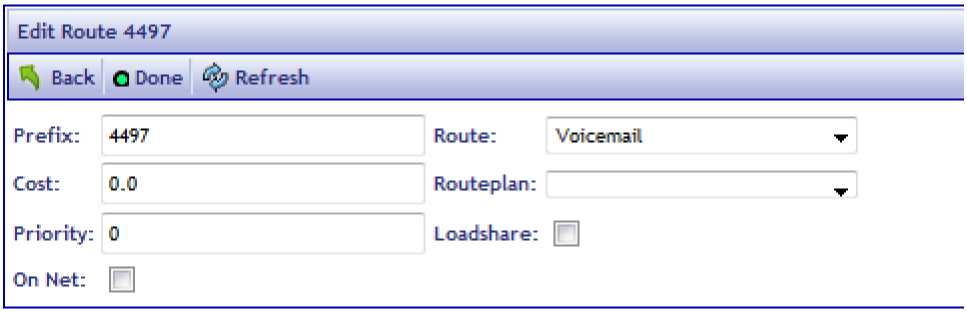


6.2 Create Customer Voicemail in Routing Tab

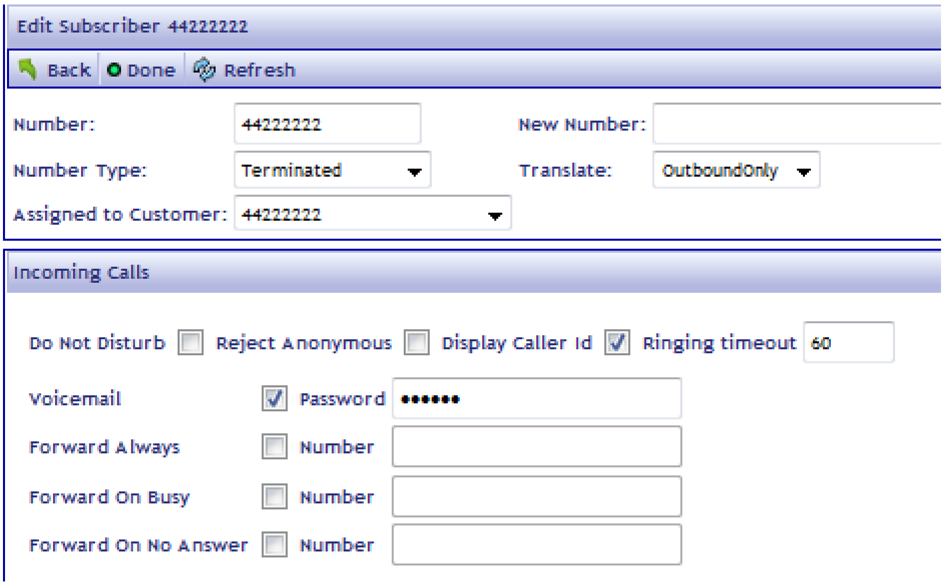
In GUI, setup a customer called "Voicemail" in Routing with the configuration as follows. Configure multiple IP Addresses: one is for Preferred box and the other is for Non-Preferred.



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()

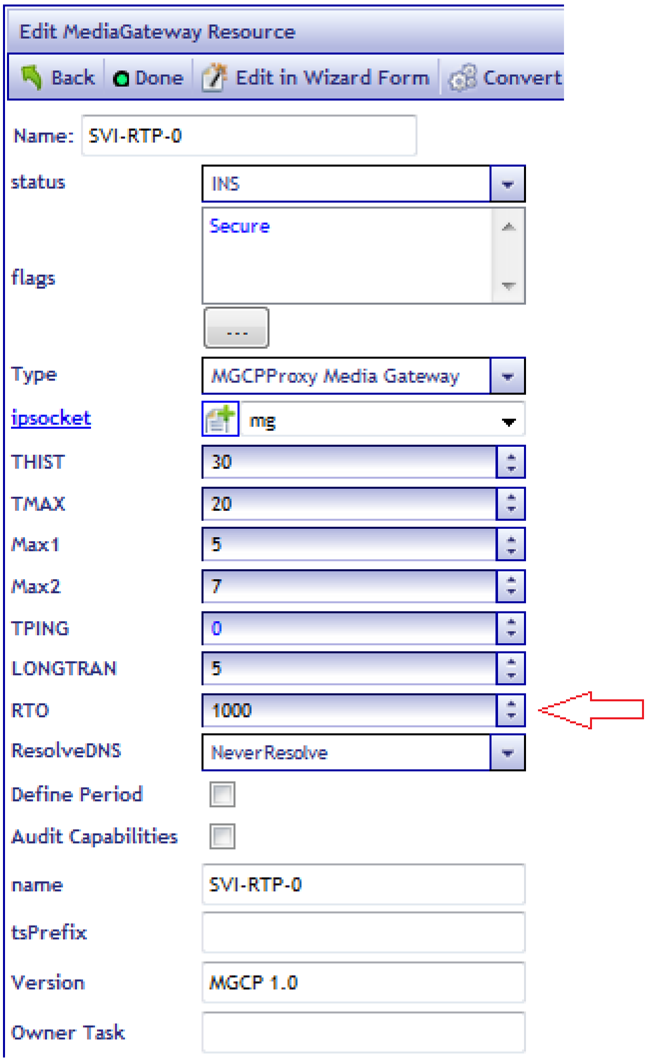
   exten => ***Announcement num***,n,Playback(***Audio file***,NOANSWER)
3. Change the announcement num in the config to be the number you have configured for one of your announcements.
4. Change the Audio file to be the name of the audio file which will be your announcement. You can either specify the 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.



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.

```
**************** System Resources ******************
System 0 Name                            RTP Router
System 0 license                         8020A#520#2060#
System 0 EndpointName                    SVIGateway
System 0 loop                            1
*********** Resources on Media Gateways ************
Media Gateway 0 Status                   Ins
Media Gateway 0 Type                     MGCP Media Gateway
Media Gateway 0 End                      Media Gateway
Media Gateway 0 IpSocket                 1
Media Gateway 0 Name                     SVI-RTP-0
Media Gateway 0 Version                  MGCP 1.0
Media Gateway 0 THIST                    30
Media Gateway 0 TMAX                     20
Media Gateway 0 Max1                     5
Media Gateway 0 Max2                     7
Media Gateway 0 LONGTRAN                 5
Media Gateway 0 RTO                      200
Media Gateway 0 ResolveDNS               NeverResolve
Media Gateway 0 rxCommandTask            Router
Media Gateway 0 destaddress              192.168.8.122
Media Gateway 0 destport                 2427

Interface 0 Status                       Ins
Interface 0 Type                         MG Ethernet
Interface 0 MediaGw                      0
Interface 0 StartPort                    28000
Interface 0 EndPort                      30000
Interface 0 Address1                     192.168.8.120
Interface 0 Name                         0

Interface 1 Status                       Ins
Interface 1 Type                         MG Ethernet
Interface 1 MediaGw                      0
Interface 1 StartPort                    28000
Interface 1 EndPort                      30000
Interface 1 Address1                     192.168.9.120
Interface 1 Name                         1
```

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.

```
<ivr:ivrSetup xmlns:ivr="http://squire-technologies.co.uk/ivrSchema">
    <ivr:settings>
        <ivr:audioFormat>alaw</ivr:audioFormat>
        <ivr:sql host="localhost" database="svi_ms" username="root" password=""/>
        <ivr:sip>
            <ivr:parameter key="allowguest" value="no"/>
            <ivr:parameter key="allowoverlap" value="yes"/>
            <ivr:parameter key="allowtransfer" value="yes"/>
            <ivr:parameter key="bindaddr" value="bond0"/>
            <ivr:parameter key="bindport" value="5061"/>
            <ivr:parameter key="tcpenable" value="no"/>
            <ivr:parameter key="tcpbindaddr" value="127.0.0.1"/>
            <ivr:parameter key="transport" value="udp"/>
            <ivr:parameter key="srvlookups" value="yes"/>
            <ivr:parameter key="pedantic" value="yes"/>
            <ivr:parameter key="disallow" value="all"/>
            <ivr:parameter key="allow" value="ulaw"/>
        </ivr:sip>
        <ivr:ssh>
            <ivr:connection host="192.168.8.120" username="root" password="SQ!tec5v1"/>
            <ivr:connection host="192.168.8.121" username="root" password="SQ!tec5v1"/>
        </ivr:ssh>
    </ivr:settings>
    <ivr:channels>
        <ivr:channel id="channel01">
            <ivr:parameter key="host" value="192.168.8.122"/>      VIP Address
            <ivr:parameter key="type" value="peer"/>
            <ivr:parameter key="call-limit" value="200"/>
            <ivr:parameter key="context" value="transfer"/>
        </ivr:channel>
    </ivr:channels>
```