An Unofficial Asterisk PBX Integration Guide − build 2004040

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1 An Unofficial Asterisk PBX Integration Guide − build 20040408

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This document provides some specific implementation details for the Asterisk PBX.

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2 1. about this document

This document attempts to address some of the finer points of Asterisk PBX Development and Administration.

It is not a complete guide to Asterisk nor is it a replacement for the many volumes of existing Asterisk documentation.

You may view this guide in the following formats: HTML (Chapters) , HTML (One page) , ASCII text, SGML source or PDF 1.2.

Big thanks to: Mark Spencer, Adtran, Mack, The Asterisk Mailing List, #asterisk on freenode.

3 2. RT & Asterisk

This section contains info about integrating Request Tracker with The Asterisk PBX.

In laymans terms this enables the creation of RT trouble tickets via telephone. It also allows the user to comment on existing RT tickets .

When RT creates a ticket it will attach the users voice message along with some call data. These messages are then available to support staff via the RT Web Gateway.

To accomplish this you must first have a working Asterisk PBX along with a working Request Tracker install, Version 3.0 or greater.

Download rt-soap-server.pl to your RT Host and configure its library path. (ie: s|/opt/rt3/lib|/www/rt3/lib|). You may also need to install SOAP::Lite and MIME::Entity from CPAN (perl -MCPAN -e'install MODNAME')

Fire up rt-soap-server.pl and make sure that it gets started after the next reboot.

Now get rt_ticket.agi onto your phone server . Edit the file to suit your environment.

You should also download these crude sounds to work with. Unpack them into /var/lib/asterisk/sounds/rt .

Next, configure your dialplan like so:

```
[rtphone]
exten => s,1,AGI(/etc/asterisk/rt_ticket.agi)
```

At this point you should hopefully be able to dial the AGI script and interact with your RT system.

Security notes By default rt-soap-server.pl runs on port 9000.

4 3. Nagios & Asterisk

This section details configuring the Nagios Monitoring System to send TTS voice alerts via an Asterisk MeetMe conference.

Nagios Configs

[contacts.cfg]

```
define contact{
        contact_name                    conference
        alias                           AsteriskMeetMeConference
        host_notification_period        24x7
        service_notification_period     24x7
        service_notification_options    c,r
        host_notification_options       d,r
        service_notification_commands   notify-by-phone
        host_notification_commands      notify-by-phone
        email                           pbx@yourdomain.com
        }
```

[contactgroups.cfg]

```
define contactgroup {
        contactgroup_name       listeners
        alias voicealert
        members                 conference
}
```

[misccommands.cfg]

```
define command {
        command_name    notify-by-phone
        command_line    /usr/bin/printf "%b" "Service $SERVICEDESC$ on Host $HOSTNAME$ sta\te cha
        }
```

[services.cfg]

```
define service{
use                             generic-service
host_name                       mail1,mail2
service_description             SMTP
is_volatile                     0
check_period                    24x7
contact_groups                  listeners
max_check_attempts              10
normal_check_interval           10
retry_check_interval            15
notification_interval           30
notification_period             24x7
notification_options            w,u,c,r
check_command                   check_smtp
}

..
```

Asterisk Configs

[extensions.conf]

```
[conf]
exten => s,1,MeetmeCount(5150)
exten => s,2,Meetme(5150|psN)
```

[iax.conf]

```
[qcall]
type=friend
context=conf
auth=md5,plaintext
secret=<IAX_PW>
host=dynamic
allow=gsm
accountcode=qcall
```

[mkqcall.pl]

---

5 4. Monitoring Asterisk

This section details monitoring your * system with Nagios .

checkcommands.cfg

```
define command{
        command_name    check_pbx
        command_line    /etc/asterisk/monitor_pbx.pl
        }
```

services.cfg

```
define service{
use                                generic-service
host_name                          pbxhost
service_description                AsteriskPBX
is_volatile                        0
check_period                       24x7
contact_groups                     janitors,helpdesk
max_check_attempts                 10
normal_check_interval              5
retry_check_interval               1
notification_interval              60
notification_period                24x7
notification_options               w,u,c,r
check_command                      check_pbx
}
```

monitor_pbx.pl, a perl script for testing * status is also available.

notes

Minimal *asterisk/manager.conf* snippet:

```
[nagios]
secret = XyXyXyXyXy
permit=127.0.0.1/255.255.255.0
read = system
write = log
```

---

6 5. dialplan visualization

The author has written a GraphViz utility to aid in the visualization of extensions.conf

An example is available, along with the code.

7 6. phones

7.1 6.1. cisco 12sp+

[Manufactured originally by Celsius.. What it is]

Cisco lists this phone as 'legacy'. They have been officially EOL'd Its is NOT a SIP compatible phone, it runs SCCP (AKA 'Skinny').

While this phone is sometimes listed as being H323−capable, please note that this is true only when integrated with Cisco Call Manager (CCM).

The 12SP+ boots via dhcp and connects to the local tftp server to download its configuration file (SEPDefault.cfg OR SEPDefault.cfg.xml OR SEPDefaultMACADDRESS.cfg (?) to recieve its configuration.

[How it behaves with Asterisk]

[chan_skinny and chan_sccp go here]

[Configuring the phone is done by pressing the sequence **#]. Network addresses are entered with the keypad , using '*' as dots. Be sure to hit # at the end of each value.

```
[ ISC dhcpd.conf stuff ]
default−lease−time 86400;
max−lease−time 604800;
boot−unknown−clients false;

option interface−mtu 1500;
ddns−update−style ad−hoc;

subnet 192.168.1.0 netmask 255.255.255.0
{
        ## asterisk server
        next−server 192.168.1.253;
        option domain−name−servers 192.168.1.254;
        option routers 192.168.1.1;
        option broadcast−address 192.168.1.255;
        range 192.168.1.200 192.168.1.251;
 }
```

```
group
 {
    host cisco12sp
      {
        hardware ethernet 00:B0:64:09:F7:CB;
        filename "OURDHCPSERVER.cnf";
        fixed−address 192.168.1.250;
      }
```

−− DHCP(yes|no), DNS, Netmask, Gateway, DNS, TFTP

7.2 6.2. GnomeMeeting

[Coming soon]

---

8 7. pbx's

8.1 7.1. definity pbx

```
[Integrating Definity VIA PRI]
```

8.2 7.2. swbell key system

```
[Integrating SW Bell System via FXO lines 1-8]
```

---

9 8. sounds

There is an excellent collection of community-contributed Allison Smith sounds.

You can check the sounds out of cvs via :pserver:anoncvs@cvs.digium.com:/usr/cvsroot . The module name is 'asterisk-sounds'.

9.1 8.1. creating .gsm files

9.1.1 8.1.1. Using SOX

$ sox inputfile.wav -r 8000 -c 1 outputfile.gsm

9.1.2 8.1.2. Helper Script

Once i started recording samples for the various PBX menus, i quickly discovered the tedium of converting and installing the WAV files by hand.

I wrote this simple script which converts WAV files to GSM and installs them in your asterisk sounds directory. Your mileage may vary.

To make use of it, call the script with a working directory of WAV files. It calls SOX to convert the files , and /usr/bin/install to install them in your asterisk system directory. You'll probably want to run this as root, or as a user that can write to your sounds directory.

```
#!/usr/bin/perl

## mike@megaglobal.net

use DirHandle;

my $sd = "/var/lib/asterisk/sounds";
my $dh = new DirHandle ".";

while (defined($_ = $dh->read()))
{
    next unless ($_ =~ /\.wav$/i);

    my $f = $_;
       $f =~ s/\.wav$/\.gsm/i;;

    my $c = "sox $_ -c 1 -r 8000 $f";
    print " Convert..";
    system $c;

    my $i = "install $f $sd";
    print " Install $sd/$f..\n";
    system $i;
```

```
}

exit 0;
```

10 9. support

How to get support for Asterisk.

10.1 9.1. documentation

10.1.1 9.1.1. official documentation

Digium.

The Official Asterisk Handbook at Digium (PDF) .

Asterisk Whitepaper by Mark Spencer .

Asterisk Documentation by Mack Allison of LSS.

10.1.2 9.1.2. un official documentation

voip-info.org, the definitive guide to Open Source Telphony.

Tilghman's Site.

James Golovich's Site.

This Document.

10.2 9.2. interweb

10.2.1 9.2.1. asterisk mailing list

http://www.marko.net/asterisk/archives.

10.2.2 9.2.2. irc support

Asterisk developers and users #asterisk on irc.openprojects.net.

Mark Spencer (Asterisk creator and otherwise programming genious:) is usually online as 'kram', when he's not idle that is :).

10.3 9.3. paid support

10.3.1 9.3.1. digium

Commercial support is available from Digium. Toll free: (877) LINUX−ME (877−546−8963) .

10.3.2 9.3.2. megaglobal support

The authors company, Megaglobal Corp, also provides * support for reasonable rates.