

The Asterisk/app_rpt Howto

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Initial Assumptions

1. The user wishing to install the package knows how to use basic Linux commands (i.e. cd, cp ...). The user knows how to bring up a terminal window in a GUI-based linux distribution (if your distribution is GUI-based)
2. The user knows how to edit text files in Linux.
3. The user has installed a Linux distribution and is comfortable doing so.
4. The user has an understanding of networking PC's under Linux, plus TCP/IP networking and Ethernet concepts in general.
5. Some way to test the software installation. You can test the software install separately with 2 phones and no Analog Radio Interface Board (ARIB).
6. The hardware requirements are met (see below).
7. If you want to test Internet Linking (which is really the whole point of this), you have to have the IP address of another node to connect to and that node must be configured to accept your IP address. You will also need to have a node number assigned which your linking peers agree to.
8. The user is familiar with interfacing repeater controllers to repeater radios, and knows how set audio levels correctly. The user has gained a basic understanding of how the ARIB board works by perusing the schematic (we haven't written a user manual for the ARIB yet!).

Hardware Requirements

A 600MHz or better Pentium, or Athlon PC with at least a 10GB hard drive, CD ROM drive, Ethernet port, and 128MB of RAM. The BIOS on this PC must be able to boot CD ROM's.

A Digium TDMx00P with at least 2 FXS ports, or a Digium T100P, T1 crossover cable, and CAC Access Bank I FXS channel bank with a 25 pair CP cable and 12 port RJ-14 Harmonica. (A Harmonica adapts the 25 pair Amphenol "Champ" connectors to modular-style plugs, and it sort of looks like a real harmonica).

One RJ14 modular jumper cable if using the channel bank, or 2 RJ11 cables if using the TDMx00P. Since the ARIB is polarity sensitive, the RJ11 cables should have the connector locking tabs straight up on both ends if the cable is laid flat on a table. If using a Channel Bank with a RJ14 Harmonica, the RJ14 cable should also have both tabs on the

top if the cable is laid flat on a table. If you have an RJ11 harmonica then the cables will be exactly the same as used for the TDMx00p.

If you are connecting to the Internet, it is advised you either have a hardware router, or set up the machine being used with a firewall . Firewall setup is beyond the scope of this howto. The hardware router must be capable of having port 4569 forwarded to the repeater controller PC on the internal private network. You will also need broadband Internet service and the hardware required to connect the service to the router.

2 telephones with modular cords to use during testing. We like to use Cortelco, or Western Electric 2500 sets with polarity guard. Those cheap phones you buy at Wally World are a poor choice. If you are using a RJ14 Harmonica, you will need a 1 to 2 splitter adapter to connect the 2 phones into one jack.

Optional – ARIB board tested and interfaced to a repeater radio.

Install RHL 9

This section contains the instructions to install Red Hat Linux (RHL) 9. These instructions are terse. You are advised to read through the RHL 9 x86 installation guide at <http://www.redhat.com/docs/manuals/linux/>

To install a useable version of RHL 9, use the following procedure:

1. Download the ISO images for RHL 9 disks 1 and 2 from one of the sites listed at <http://www.linuxhelp.net/isos/#redhat>.
2. Burn 3 CD's with these ISO images using your favorite CD burning software on a your home computer (not the computer you are going to use as the repeater controller).
3. Using the install guide, install RHL9 on the PC to be used as the repeater controller. When you are asked what to install, install everything. Also, make sure you install the kernel sources as they are needed to build Zaptel. Set up the network with a static IP address on the same subnet as the LAN side of your hardware router. Do not install the firewall; this will cause you grief if you don't know what to do to administer it.
4. Boot your new Linux system and make sure you can log in.

Install Asterisk

Asterisk is downloaded and installed using the following procedure.

1. With your Linux system booted, and logged in as root, open a terminal window so that you can execute some shell commands.
2. Change directories to /usr/src.. Execute the following shell commands:

```
export CVSROOT=:pserver:anoncvs@cvs.digium.com:/usr/cvsroot
cvs login
cvs checkout zaptel libpri asterisk
```

Note: When you execute the “cvs login” command, you will be prompted for a password. The password is anoncvs.

3. Change directories to zaptel, and type the following commands

```
make
make install
```

4. Change directories to ../libpri, and type the following commands

```
make
make install
```

5. Change directories to ../asterisk/apps. Edit the Makefile using your favorite editor (vi, emacs, joe, etc.). Uncomment the line #+=app_rpt.so by deleting the ‘#’ character. Save the file and exit the editor.
6. Move up to the parent directory (asterisk) by executing cd ..
7. Type the following commands:

```
make
make install
```

Install the Supplemental Voice Files

app_rpt requires some supplemental voice files which are not in the standard Asterisk distribution. To install these files, change directories to /var/lib/asterisk/sounds. Next, use wget to retrieve the archive:

```
wget http://www.zapatatelephony.org/rpt\_pcm.zip
```

With the archive retrieved, it will need to be unzipped:

```
unzip rpt_pcm.zip
```

Finally, rename the directory name from rpt_pcm to rpt:

```
mv rpt_pcm rpt
```

Install the Skeletal Config Files

In order to simplify the somewhat painful Asterisk configuration procedure, I made up some skeletal configuration scripts. Make a directory called /etc/asterisk, change directories to /etc/asterisk and type:

```
wget http://www.zapatatelephony.org/skelcfgs.zip
```

With the archive retrieved, it will need to be unzipped:

```
unzip skelcfgs.zip
```

One of the files, zaptel.conf will need to be moved to /etc because Asterisk expects to find it there:

```
mv zaptel.conf ..
```

Customize /etc/zaptel.conf

If you have a T100P T1 card and channel bank, you will need to uncomment the line which defines the span. This line should look something like this:

```
#span=1,0,0,d4,ami
```

To uncomment the line, remove the # symbol.

If you have a TDM400P, then the zaptel.conf file can be left as is.

Customize /etc/asterisk/extensions.conf

Look for 2 lines which look like this:

```
[radio-secure]  
exten => 1,1,rpt,111
```

Change the rightmost comma delimited number to the actual node number you are assigning to this repeater. This node number needs to match the node number you assign below in iax.conf.

Customize /etc/asterisk/iax.conf

For this step, you will need the IP address(es) your linking peers. Every line in this file with a comment which begins with an exclamation point (!) will need to be changed to a value which is meaningful. To customize this file, look for these lines:

```
[radio]
type=user
context=radio-secure
deny=0.0.0.0/0.0.0.0
permit=aaa.bbb.ccc.ddd/aaa.bbb.ccc.ddd # ! ....
permit=eee.fff.ggg.hhh/eee.fff.ggg.hhh # ! ....
```

Change the permit statements to allow your linking peers to connect to your system. Permit statements take two dotted quad values. The first dotted quad is the IP address of your peer, and the second dotted quad is the netmask. For one IP address and not a range of IP addresses, the netmask will be 255.255.255.255. If you want to permit a range of IP addresses, then the netmask will be some other value. If you only have one peer to connect to, then remove the second permit statement. If you have more than 2 peers to connect to, add additional permit statements.

Customize /etc/asterisk/rpt.conf

Open the file and search for a line similar to:

```
[111] ; ! Name of First Repeater
```

Change the number in the brackets to the node ID of the new node.

Search further down the file for lines similar to:

```
rxchannel = Zap/? ; ! Rx audio/signalling ...
txchannel = Zap/? ; ! Tx audio/signalling ...
```

Change the rxchannel to Zap/2, and the txchannel to Zap/1.

Search further down the file for a line similar to:

```
idrecording = id ; ! id recording. You will...
```

Change the id argument to *icall/r* (e.g. *iwa6zft/r*). This will set up the repeater ID'er to id in morse code with your callsign. Later when you know more *asterisk foo*, you can change this to a voice recording, by recording a .gsm file and substituting it for the morse ID.

Search further down the file for a line similar to:

```
idtalkover=|iid/r ; ! Talkover ID. you will...
```

Change the iid/r argument to *icall/r* just like you did with idrecording. By doing this, when you change your main id recording to a voice ID, the morse ID will be used when there is someone talking on the repeater so that the voice ID will not interrupt the conversation in progress.

Search further down the file for a group of lines similar to:

```
[nodes]
```

```
111 = radio@www.xxx.yyy.zzz/1 ; ! This needs...  
...  
...  
112 = radio@aaa.bbb.ccc.ddd/1 ; ! This needs ...
```

This is the place where we define our node ID and the node ID's of our linking peers.

Change the node ID 111 to the node ID assigned to this new system, then change the IP address in the argument to the IP address of the network interface installed in the system. This node number needs to match the node number you changed in extensions.conf.

Change the node ID 112 to the node ID of a linking peer you wish to be able to link to, then change the IP address in the argument to the IP address of that linking peer. This IP address should match the IP address in the permit statement contained in the iax.conf file. You might also need to change the number delimited by the / which follows the IP address if the peer you are linking with has more than 1 repeater at a given IP address. The best thing to do here is check with the administrator of the linking peer to ensure this number is correct.

If you don't have a linking peer to connect to at this time, then comment out node 112 by inserting a semicolon ';' in the line as the first character.

If you have more than one linking peer to add, copy the contents of the second line and modify it to suit your needs.

Running Asterisk

Now that everything is configured, it is time to run Asterisk to see if it will come up and give you the CLI prompt. Depending upon what hardware you have (T100p or TDMx00), the initialization procedures will differ. Also, if you have other Digium hardware not discussed in this howto in the PC, these driver initialization procedures will have to be modified to enforce the correct driver load order. This howto assumes that you only have one TDMx00 or one T100P. If you have other cards in addition to this, then you will need to refer to the Asterisk documentation to load the drivers in the proper order.

The first thing to do is load the Zaptel drivers. Make sure you are logged in as root.

If you have the TDMx00 FXS PCI adapter, then execute the following commands:

```
modprobe zaptel
modprobe wcfxs
ztcfg
```

If you have a T100P single port T1 card, then execute the following commands

```
modprobe zaptel
modprobe wct1xxp
ztcfg
```

All modules should load without any errors. You can check to see if they loaded by doing an `lsmod` and looking in the output to ensure they are listed.

Once all the drivers are loaded successfully, you can run Asterisk by typing:

```
asterisk -vvvvvvc
```

A bunch of initialization messages and warnings will scroll past, and you should get a prompt similar to:

```
CLI>
```

Now we are ready to test.

Testing

Remember those telephone sets we mentioned in the hardware requirements? We'll be using them here to test the software installation. Note: if you have an RJ14 Harmonica, you are going to need a splitter and two RJ11 cables to split the RJ14 connection into 2 separate RJ11 connections.

With both phones hung up, connect 1 telephone set to the TX port on your FXS adapter, and the other telephone set to the RX port on your FXS adapter.

Pick up the phone connected to the TX port. The line should be dead (no battery, i.e. you should hear no sidetone if you speak into the handset transmitter). While holding the TX phone's handset receiver to your ear, lift the RX phone handset from the cradle. You should hear delayed audio if you speak into the transmitter on the handset of the RX phone on the receiver on the TX phone. If you hang up the RX phone, you should hear a courtesy tone followed by the TX phone going dead (no battery).

You can now test a DTMF command. While listening to the TX handset receiver, pick up the RX receiver and dial *5 on the RX phone. You should hear something similar to:

```
"Node 111 repeat only"
```

If your system is acting as mentioned above, then your Asterisk and app_rpt software is correctly configured. If you have an ARIB board, and have interfaced it to a repeater, you should now disconnect the telephone sets and proceed to the next section.

Connecting the Repeater and Controller

This section assumes that you have an ARIB built and interfaced to your repeater radio, and that you have tested, and adjusted the ARIB with the radio and found it to be functional. If that is the case, then you now can test the both the repeater controller and the radio system together.

Before continuing, verify the PL and COR polarity settings on the ARIB are correct, and when you put a signal on the input of your repeater, the green COR led D7 illuminates.

If you have an RJ14 harmonica and RJ14 cable, insert jumpers in JP1 and JP2 on the ARIB, and connect your RJ14 cable to J1.

If you have two RJ11 cables, connect the TX cable to J1 and the RX cable to J2. Remove any jumpers on JP1 and JP2.

When everything is hooked up, try keying the the repeater to see if the repeater transmitter comes on. You should see the green COR led D7 illuminate, followed by the red led D2.

If the COR/PTT logic is working, you can now check and adjust audio levels. Audio level adjustment procedures can be non-trivial and are beyond the scope of this howto.

If you have set the audio levels to your liking, try a *5 DTMF command to see if you get a voice response.

Testing the Linking Feature

If you specified a linking peer in `iax.conf`, and `rpt.conf`. You can now test the connection, provided the far end node has modified their configuration to permit your IP address to connect, and to define your node number.

If you dial a *3 followed by the node ID of the far end node, you should get a message like:

```
"node xxx transceive"
```

If you get:

```
"node xxx remote busy"
```

or

```
"node xxx connection failed"
```

Then you have a configuration problem or network connectivity problem. Recheck your configuration files, and make sure you can reach the far end node over the network. Also, check your hardware router and make sure you forwarded port 4569 to the private IP address of the repeater controller PC.

If connections from your node to the far end node are working, make sure connections from the far end node to you also work.

Final Notes

If you get the system working in this minimal configuration, you might want to go to the sites previously mentioned and learn more about Asterisk so that you can add additional features such as an Autopatch, or additional repeaters at the same site. As mentioned earlier, the same PC can also serve as a phone system for your home or business.

It would also be worthwhile to study the notes in `rpt.conf` because this file controls most of the aspects of repeater operation.