

Asterisk on RPI

Starting with Debian 3.2.27+

RPI-Debian # uname -a

Linux RPI-Debian 3.2.27+ #1 PREEMPT Sat Sep 22 06:10:01 EDT 2012 armv6l GNU/Linux

And after making a dd copy on 10/21/2012, (3767 seconds) lets add on asterisk

```
apt-get -y install make gcc g++ libxml2 libxml2-dev ssh libncurses5
libncursesw5 libncurses5-dev libncursesw5-dev linux-libc-dev sqlite libnewt-
dev libusb-dev zlib1g-dev libmysqlclient15-dev libsqlite0 libsqlite0-dev
bison openssl libssl-dev libeditline0 libeditline-dev libedit-dev mc sox
libedit2 libedit-dev curl libcurl4-gnutls-dev apache2 libapache2-mod-php5
php-pear openssl-server build-essential openssl-client zlib1g zlib1g-dev
libtiff4 libtiff4-dev libnet-telnet-perl mime-construct libipc-signal-perl
libmime-types-perl libproc-waitstat-perl mpg123 libiksemel-dev
```

Update Asterisk Install Pre-requisites

To start with, update the system and install the required component's by copying the command below:

```
apt-get update
```

```
apt-get -y install make gcc g++ libxml2 libxml2-dev ssh libncurses5 libncursesw5 libncurses5-dev libncursesw5-dev
linux-libc-dev sqlite libnewt-dev libusb-dev zlib1g-dev libmysqlclient15-dev libsqlite0 libsqlite0-dev bison openssl
libssl-dev libeditline0 libeditline-dev libedit-dev mc sox libedit2 libedit-dev curl libcurl4-gnutls-dev apache2
libapache2-mod-php5 php-pear openssl-server build-essential openssl-client zlib1g zlib1g-dev libtiff4 libtiff4-dev
libnet-telnet-perl mime-construct libipc-signal-perl libmime-types-perl libproc-waitstat-perl mpg123 libiksemel-dev
php5 php5-cli mysql-server php5-mysql php-db libapache2-mod-php5 php5-gd php5-curl mysql-client vim
```

```
apt-get -y install asterisk-core-sounds-en asterisk-core-sounds-en-g722 asterisk-core-sounds-en-gsm asterisk-
core-sounds-en-wav
```

Download and Extract Asterisk

[illegible]

Change interface ETH0 to 192.168.5.1/24, change metric on eth0 and wlan0

vi /etc/network/interfaces

The primary network interface

auto eth0 If using ifplugd, don't use auto here

iface eth0 inet static

#metric 10

metric 200

address 192.168.5.1

netmask 255.255.255.0

network 192.168.5.0

broadcast 192.168.5.255

dns-nameservers 192.168.1.1

#address 192.168.0.89

#gateway 192.168.0.1

#netmask 255.255.255.0

#network 192.168.0.0

#broadcast 192.168.0.255

#dns-nameservers 192.168.1.1

The wireless interface

auto wlan0

iface wlan0 inet dhcp

metric 10

#metric 200

dns-nameservers 192.168.1.1

/etc/init.d/networking restart

Install and enable DHCP

apt-get update

apt-get install isc-dhcp-server

vi /etc/default/isc-dhcp-server

Edit the following line

```
INTERFACES="ETH0"
```

```
vi /etc/dhcp/dhcpd.conf
```

```
option domain-name "rpi.jj3601.com";  
option domain-name-servers 192.168.1.1, 4.4.4.1;
```

```
default-lease-time 86000;  
max-lease-time 86000;
```

```
subnet 192.168.5.0 netmask 255.255.255.0 {  
  range 192.168.5.100 192.168.5.199;  
  option routers 192.168.5.1;  
  option subnet-mask 255.255.255.0;
```

```
option broadcast-address 192.168.5.255;  
#option domain-name-servers 192.168.1.1;
```

```
option ntp-servers 192.168.0.20;  
#option netbios-name-servers 10.0.0.1;  
#option netbios-node-type 8;  
}
```

```
service isc-dhcp-server restart
```

Set up Linksys PAP-2T

PAP-2T runs on 5V DC so I made up a USB adapter cable to provide power. Plug a phone into the analog Phone 1 jack and (touch-tone only) enter **** to enter the configuration mode.

Reset the PAP-2T

Press 73738# followed by 1 to confirm.

Set up the PAP-2T for DHCP

Enable DHCP by pressing 101# (Enter 1 to Enable and 0 to Disable DHCP)
Press 1 # to confirm

Verify IP address on PAP-2T

Enter 110#

Verify PAP-2T gets a DHCP address

Plug the PAP-2T into the RPI (a straight cable works fine)

Verify it got an address by executing the arp command on the RPI.

Settings- Cisco/Linksys PAP2t

The Cisco/Linksys PAP2T is a very popular 2 line Internet Phone Adapter or ATA device which can be connected up to your router. An analog phone can be connected to each of the two phone ports and if enabled with your VoISP the Cisco/Linksys PAP2T will support both lines. Each phone port operates independently with separate phone service and their own phone numbers, line 1 and line 2. The Cisco/Linksys ATA can be accessed from connecting a PC and typing the IP address of the ATA into a browser. To determine the current IP address connect a phone into phone port 1 while the ATA is powered up and connected to your router's LAN port and dial 4 stars **** then 110#. The IVR will say the IP address which you will need to type in the address of your browser.

Click on the "Admin Login" button near the top right side of the screen, then click on the "Line 1" tab. The following instructions are typical instructions for line 1. In most cases you will need only modify a few parameters from the normal factory default settings.

LINKSYS®
A Division of Cisco Systems, Inc.

Firmware Version: 3.1.9(LSo)

Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info

System

SIP

Regional

Line 1

Line 2

User 1

User 2

Basic View (switch to advanced view)

User Login

System Configuration

Internet Connection Type

Optional Network Configuration

Enable Web Server:

yes

User Password:

Passw0rd

DHCP:

yes

Static IP:

NetMask:

Gateway:

HostName:

RPI-PAP2T

Domain:

Primary DNS:

Secondary DNS:

DNS Query Mode:

Parallel

Syslog Server:

Debug Server:

Debug Level:

0

Save Settings

Cancel Settings

CISCO SYSTEMS

LINKSYS®
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Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info

System

SIP

Regional

Line 1

Line 2

User 1

User 2

Basic View (switch to advanced view)

User Login

Response Status Code Handling

RTP Parameters

SDP Payload Types

SIT1 RSC:

SIT2 RSC:

SIT3 RSC:

SIT4 RSC:

RTP Port Min:

16384

RTP Port Max:

16482

NSE Dynamic Payload:

100

AVT Dynamic Payload:

101

G726r16 Dynamic Payload:

98

G726r24 Dynamic Payload:

97

G726r40 Dynamic Payload:

96

G729b Dynamic Payload:

99

Save Settings

Cancel Settings

CISCO SYSTEMS

Voice

Info

System

SIP

Provisioning

Regional

Line 1

Line 2

User 1

User 2

Advanced View (switch to basic view)

User Login

Configuration Profile

Provision Enable:

no

Resync On Reset:

no

Resync Random Delay:

2

Resync Periodic:

3600

Resync Error Retry Delay:

3600

Forced Resync Delay:

14400

Resync From SIP:

yes

Resync After Upgrade Attempt:

yes

Resync Trigger 1:

Resync Trigger 2:

Resync Fails On FNF:

no

Profile Rule:

Profile Rule B:

Profile Rule C:

Profile Rule D:

Log Resync Request Msg:

\$PN \$MAC -- Requesting resync \$\$SCHEME://\$SERVIP:\$P

Log Resync Success Msg:

\$PN \$MAC -- Successful resync \$\$SCHEME://\$SERVIP:\$P

Log Resync Failure Msg:

\$PN \$MAC -- Resync failed: \$ERR

Report Rule:

Firmware Upgrade

Upgrade Enable:

no

Upgrade Error Retry Delay:

3600

Regional Settings (Advanced View)

Ring and Call Waiting Tone Spec

Ring Name:

Service 12

Ring Name:

Service 10

Ring Waveform:

Sinusoid

Ring Frequency:

20

Ring Voltage:

90

CWT Frequency:

440@-10

Synchronized Ring:

no

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Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info

System

SIP

Regional

Line 1

Line 2

User 1

User 2

Basic View (switch to advanced view)

User Login

SIP Settings

Proxy and Registration

Subscriber Information

Supplementary Service Subscription

Line Enable:

yes

SIP Port:

5060

Proxy:

192.168.5.1

Register:

yes

Make Call Without Reg:

no

Register Expires:

3600

Ans Call Without Reg:

no

Display Name:

RPI_Phone_1

User ID:

1111

Password:

1111

Use Auth ID:

no

Auth ID:

Call Waiting Serv:

yes

Block CID Serv:

yes

Block ANC Serv:

yes

Dist Ring Serv:

yes

Cfwd All Serv:

yes

Cfwd Busy Serv:

yes

Cfwd No Ans Serv:

yes

Cfwd Sel Serv:

yes

LINKSYS®
A Division of Cisco Systems, Inc.

Firmware Version: 3.1.9(LSc)

Voice

Phone Adapter with 2 Ports for Voice-Over-IP

PAP2

Info

System

SIP

Regional

Line 1

Line 2

User 1

User 2

Basic View (switch to advanced view)

User Login

SIP Settings

Proxy and Registration

Subscriber Information

Supplementary Service Subscription

Line Enable:

yes

SIP Port:

5061

Proxy:

192.168.5.1

Register:

yes

Make Call Without Reg:

no

Register Expires:

3600

Ans Call Without Reg:

no

Display Name:

RPI_Phone_2

User ID:

2222

Password:

2222

Use Auth ID:

no

Auth ID:

Call Waiting Serv:

yes

Block CID Serv:

yes

Block ANC Serv:

yes

Dist Ring Serv:

yes

Cfwd All Serv:

yes

Cfwd Busy Serv:

yes

Cfwd No Ans Serv:

yes

Cfwd Sel Serv:

yes

The Preferred codec drop down box has several choices. Typically G711, which is an uncompressed codec would be the choice, but to conserve on bandwidth you might change Line 1 "Preferred Codec" to G729a. The "Use Pref Codec Only" setting to No would allow the call to negotiate the codec.



Preferred Codec:	G729a	Silence Supp Enable:	no
Use Pref Codec Only:	no	FAX CED Detect Enable:	yes
DTMF Tx Method:	Auto		

Note: The Cisco/Linksys PAP2 or PAP2T does not support usage of simultaneous calls using the G.729 codec. When one line port uses G.729, the other line port will use G.711. (Using compression like G729 does is processor intensive and both these ATA models do not have a capacity to handle simultaneous G729 calls.)

Asterisk Configuration

First, save off the original configuration files.

```
cd /etc/asterisk
cp iax.conf iax.conf.orig
cp sip.conf sip.conf.orig
cp extensions.conf extensions.conf.orig
```

Add the following line to /etc/asterisk/sip.conf

```
#include "/etc/asterisk/rpi_sip.conf"
```

Create /etc/asterisk/rpi_sip.conf with the following content.

```
[1111]
secret=1111
type=friend
```

deny=0.0.0.0/0.0.0.0
permit=192.168.5.0/255.255.255.0

host=dynamic
port=5060
qualify=yes
canreinvite=no
dtmfmode=inband
nat=no
context=internal
allow = all
mailbox = 1111

[2222]
secret=2222
type=friend

deny=0.0.0.0/0.0.0.0
permit=192.168.5.0/255.255.255.0

host=dynamic
port=5061
qualify=yes
canreinvite=no
dtmfmode=inband
nat=no
context=internal
allow = all
mailbox = 2222

From the Asterisk console, enter sip reload and sip show peers

You should see output similar to that listed below

RPI-Debian*CLI> sip reload

Reloading SIP

== Parsing '/etc/asterisk/sip.conf': == Found
== Parsing '/etc/asterisk/rpi_sip.conf': == Found
== Parsing '/etc/asterisk/users.conf': == Found
== Using SIP CoS mark 4
== Parsing '/etc/asterisk/sip_notify.conf': == Found

RPI-Debian*CLI> sip show peers

Name/username	Host	Dyn	Forcerport	ACL	Port	Status
1111/1111	192.168.5.100	D	A	5060	OK (12 ms)	
2222/2222	192.168.5.100	D	A	5061	OK (12 ms)	

2 sip peers [Monitored: 2 online, 0 offline Unmonitored: 0 online, 0 offline]

RPI-Debian*CLI>

On the PAP-2T, you should get dial tone on both ports.

Set up extensions.conf Insert this just before ;[context]

```
[internal]
exten => 1111,1,Answer()
exten => 1111,n,Set(CALLERID(num)=1111)
exten => 1111,n,Set(CALLERID(name)=Raspberry)
exten => 1111,n,Dial(SIP/1111,20,tr)
exten => 1111,n,VoiceMail(1111,u)
exten => 1111,n,Hangup

exten => 2222,1,Answer()
exten => 2222,n,Set(CALLERID(num)=1111)
exten => 2222,n,Set(CALLERID(name)=Raspberry)
exten => 2222,n,Dial(SIP/2222,20,tr)
exten => 2222,n,VoiceMail(2222,u)
exten => 2222,n,Hangup
exten => 8888,1,VoiceMailMain() ; to retrieve voicemail
```

Set up voicemail.conf Insert this just after [default]

```
; Added by JJ
;extension number => password, User Name, email to send wav file to
1111 => 12345,Phone1,user@user.com
2222 => 12345,Phone2,user@user.com
```

Reference

Linksys PAP2T ATA Adapter Reset Procedure:

Sometimes it will be very helpful to reset your linksys ATA adapter to factory default settings. If you are using a used ATA adapter, then resetting your adapter to factory default settings is highly recommended. The procedure to reset your linksys adapter is as follows.

Step1: Connect a telephone connected to line1 of the ATA unit and power up your ATA unit using its power adapter.

Step2: Disconnect your PAP2T adapter from the internet connection or in other words just unplug the ethernet cable from the PAP2T hardware unit. Resetting with internet connection may mess up the unit making it completely useless.

Step3: Dial ****, and wait for the Interactive Voice Menu (IVM) to get activated. After hearing this message, type in the following number with the # symbol.

73738# This number spells RESET.

Step4: After this, you will be asked to confirm this by pressing 1.

Your linksys ATA unit will now go back to it factory default settings.

Dial 73738#

Password: 7756112# or 8995523#

enter 1 to confirm press # Hang up

Linksys PAP2T IP Configuration Procedure:

During the time where there is no ethernet cable connected to the PAP2T the 1 long 2 short flashing green light is normal because that means that there is no ethernet detected -- reset the PAP2T one more time while the ethernet is disconnected and phone configuration is accessible.

then try to set a static IP, network mask and default gateway on the PAP2T

-- disable DHCP by pressing **** 101 (Enter 1 to Enable and 0 to Disable DHCP)

-- after you have successfully disabled DHCP press **** 111

-- Enter IP Address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point

-- set a Network Mask by pressing **** 121

-- enter value using keypad

-- set Default Gateway by **** 131

-- enter value using keypad

-- set Primary DNS by **** 161

of course the IP settings you will enter on the PAP2T needs to be in the same range as that of your existing network

```
mount -t ntfs-3g -o rw,users /dev/sda1 /mnt/
```

Additional Steps

NOTE: It is necessary with ZoneEdit to create a new host at www.zoneedit.com before an update can be applied. Zoneedit will NOT create a new A record from an update. Go there and create a new record for raspberry.cnetvoip.com, assign IP address 1.1.1.1 and Save and Publish.

Add Dynamic DNS

apt-get update
apt-get install ddclient

A menu is auto-launched with choices for DDNS providers. I use Zoneedit.

Provide zoneedit username
Provide zoneedit password
Provide Fully Qualified Domain Name raspberry.cnetvoip.com

Edit /etc/rc.local and add the following line to auto start ddclient and run it every 20 minutes
/usr/sbin/ddclient -daemon 1200 -syslog

Here is the working /etc/ddclient.conf

```
# added by JJ
#ssl=yes
protocol=zoneedit1
#use=if, if=wlan0 #comment out this line to not use the private IP address
use=web, web=myip.dnsomatic.com

login=jjones74, password='badpassword74'
raspberry.cnetvoip.com
```

To debug

```
ddclient -daemon=0 -debug -verbose -noquiet
```

To force an update

```
rm /var/cache/ddclient/ddclient.cache
```

```
ddclient -daemon=0
```

Here is a config using dnsomatic Note: you need to set up an account at dnsomatic.com with Zoneedit.com credentials and an entry for raspberrypi.cnetvoip.com

```
# Configuration file for ddclient generated by debconf
#
# /etc/ddclient.conf
ssl=yes
protocol=dyndns2
# added by JJ
#use=if, if=wlano #comment out this line to not use the private IP address
use=web, web=myip.dnsomatic.com
server=updates.dnsomatic.com, login=jjones74, password='badpassword74'
all.dnsomatic.com
#raspberrypi.cnetvoip.com
```

ADD C*Net Macros to Extensions.conf

; Insert in [globals]

```
;
;Added by JJ
CNETANI=16877710          ;
OFFICECODE=687
MYNAME=John Jones;        for CallerID.
```

; The following is added for CNET

```
; This is the extensions.conf file. It should be placed in the /etc/asterisk directory.
; THIS CONFIGURATION PROVIDES SIP AND IAX2 CONNECTIVITY TO LEGACY
; OR COLLECTIBLE TELEPHONE SYSTEMS.
```

```
;
; THIS EXAMPLE FILE ASSUMES A CERTAIN CONFIGURATION, BUT CAN EASILY BE
; MODIFIED TO PROVIDE FOR DIFFERENCES WITH INDIVIDUAL PHONE SWITCHES.
```

```
;
; THIS SAMPLE SYSTEM IS CONFIGURED AS FOLLOWS:
;   + THE CNET NETWORK HAS NO AREA CODE.
;   + THIS TANDEM HAS THE ASSIGNED OFFICE CODE OF 687.
;   + 2-DIGIT DIALING TOWARDS ELECTROMECHANICAL PHONE SYSTEM
;   + 7-DIGIT DIALING RECEIVED FROM ANALOG SWITCH FOR CNET
;     CALL DESTINATIONS IN NORTH AMERICA.
;   + 011 + COUNTRY CODE + CITY CODE + NUMBER RECEIVED
;     FROM ANALOG SWITCH FOR INTERNATIONAL CNET DESTINATIONS
;
; =====
```

```
; Macros can be called from other contexts. What they do is perform a certain
; function, and then return to the originating context.
```

```
;
[macro-dialswitch]
```

```
;
;
[macro-dialcnet]          ;version bj1.01 - asterisk 1.4 and asterisk 1.2
```

```
exten => s,1,Set(NUMBER=${ARG1})          ;Store number to be called
exten => s,2,GotoIf($[ ${ARG1:0:1} = "+" ]?search)    ;Is number prefixed with '+'?
exten => s,3,Set(ARG1=+${ARG1})          ;Prefix number with '+',required to properly retrieve US ENUM
entries!!!
```

```

exten => s,4(search),Set(ENUM=${ENUMLOOKUP(${ARG1},ALL,,1,std.ckts.info)}) ;Search ENUM
database
exten => s,5,GotoIf($${LEN(${ENUM})}=0)?no_uri ;Is ENUM record found?
exten => s,6,GotoIf($${DB_EXISTS(ELBD/${NUMBER})}?cdata) ;Does backup entry exist?
exten => s,7,Set(DB(ELBD/${NUMBER})=${ENUM}) ;No existing backup so store backup ENUM
data
exten => s,8(test),GotoIf($${ENUM:0:3} = iax ]?iaxuri) ;Yes-IAX2 protocol
exten => s,9,GotoIf($${ENUM:0:3} = sip ]?sipuri) ;Yes-SIP protocol
exten => s,10,GotoIf($${ENUM:0:3} = h32 ]?h32uri) ;Yes-H323 protocol
exten => s,11(no_uri),GotoIf($${DB_EXISTS(ELBD/${NUMBER})}?seek) ;ENUM="", is there a backup
entry?
exten => s,12,Macro(invalid-office-code,${NUMBER}) ;No valid ENUM and no backup entry
exten => s,13,Wait(5) ;Pause
exten => s,14,Hangup ;Done - failed to make call - Goodbye
exten => s,15(seek),Set(ENUM=${DB_RESULT}) ;Set ENUM to backup base entry
exten => s,16,GotoIf($${REGEX("iax2|sip|h323"${ENUM})}?test) ;If it's a single backup proceed to dial
out
exten => s,17,Set(NE=${CUT(ENUM,":",1)}) ;Get backup entry field 1
exten => s,18,Set(LU=${CUT(ENUM,":",2)}) ;Get backup entry field 2
exten => s,19,GotoIf($${NE}>${LU}?grec) ;Was last used the last entry in list?
exten => s,20(frec),Set(LU=0) ;Reset entry pointer
exten => s,21(grec),Set(LU=${LU}+1) ;Increment last used pointer
exten => s,22,Set(ENUM=${DB(ELBD/${NUMBER}/${LU})}) ;Read ENUM backup entry
exten => s,23,Set(DB(ELBD/${NUMBER})=${NE}:${LU}) ;Update last used record
exten => s,24,Goto(test) ;Proceed to dial out
exten => s,25(iaxuri),Set(DIALSTR=IAX2/${ENUM:5}) ;IAX2
exten => s,26,Goto(dodial) ;Make call
exten => s,27(sipuri),Set(DIALSTR=SIP/${ENUM:4}) ;SIP
exten => s,28,Goto(dodial) ;Make call
exten => s,29(h32uri),Set(DIALSTR=H323/${ENUM:5}) ;H323
exten => s,30,Macro(invalid,${NUMBER}) ;Make Call
exten => s,31(dodial),NoOp(Outbound Caller ID is ${CALLERID(all)})
exten => s,32,Dial(${DIALSTR}) ;Dial Out
exten => s,33,Hangup ;Done -call attempted - Goodbye
exten => s,34(cdata),GotoIf(${"${DB_RESULT}"="${ENUM}"}?test) ;Does entry have a single backup?
exten => s,35,Set(SR=${DB_RESULT}) ;Backup does not match current
exten => s,36,GotoIf($${REGEX("iax2|sip|h323"${SR})}?shuffle) ;Is more than one backup entry stored?
exten => s,37,Set(NE=${CUT(SR,":",1)}) ;Get backup entry field 1
exten => s,38,Set(LU=${CUT(SR,":",2)}) ;Get backup entry field 2
exten => s,39,Gosub(rent) ;Check multiple backup entries
exten => s,40,Set(DB(ELBD/${NUMBER})=${NE}:${MR}) ;Store no. of entries and match as entry last
used
exten => s,41,GotoIf($${MR}>0)?test ;If backup exists proceed to dial out
exten => s,42,Set(NE=${NE}+1) ;Increment entries pointer
exten => s,43,Set(DB(ELBD/${NUMBER})=${NE}:${NE}) ;Update entry count and usage
exten => s,44,Set(DB(ELBD/${NUMBER}/${NE})=${ENUM}) ;Store nth backup entry
exten => s,45,Goto(test) ;Nth entry stored - proceed to dial out
exten => s,46(shuffle),Set(DB(ELBD/${NUMBER}/1)=${SR}) ;Move stored entry to backup entry /1

```

```

exten => s,47,Set(DB(ELBD/${NUMBER}/2)=${ENUM}) ;Create second backup entry /2
exten => s,48,Set(DB(ELBD/${NUMBER})=2:2) ;Store no. of backup entries & last called
exten => s,49,Goto(test) ;Alternative backup entry stored - proceed to dial out
exten => s,50(rent),Set(i=0) ;Load test counter
exten => s,51,Set(MR=0) ;Load match store
exten => s,52,While(${i}<${NE}) ;Check multiple entries for a NUMBER looking for a
match
exten => s,53,GotoIf("${DB(ELBD/${NUMBER}/${i+1})"="${ENUM}")?mark) ;Mark entry
matching ENUM result
exten => s,54,Set(i=${i+1}) ;Increment counter
exten => s,55,EndWhile ;Look at next entry if one exists
exten => s,56,Return ;Done searching for ENUM result backup record
exten => s,57(mark),Set(MR=${i+1}) ;ENUM result already backed up
exten => s,58,Set(i=${NE}) ;Load counter with no. of last entry
exten => s,59,ContinueWhile ;Stop searching

```

[macro-invalid]

```

exten => s,1,Answer
exten => s,2,Wait(1)
exten => s,3,Playtones(480*20/2000,0/4000)
exten => s,4,Wait(8)
exten => s,5,Playtones(!950/330,!1400/330,!1800/330,0)
exten => s,6,Wait(2)
exten => s,7,Playback(discon-or-out-of-service)
exten => s,8,Wait(1)
exten => s,9,SayDigits($OfficeCode)
exten => s,10,Wait(5)
exten => s,11,Hangup
;

```

[macro-Milliwatt]

```

exten => s,1,Answer
exten => s,2,Playtones(1004)
exten => s,3,Wait(${ARG1})
exten => s,4,Hangup
;

```

[cnet-out]

```

exten => _1NXXXXXX,1,Set(CALLERID(name)='John Jones') ; make sure you dont use any space character

```

```

exten => _X.,n,NoOp(${EXTEN})
exten => _X.,n,NoOp(${CALLERID(num)})

```

```

exten => _1NXXXXXX,n,Set(CALLERID(number)=16877710) ; make sure you dont use any space
character;

```

```

exten => _1NXXXXXX,n,Macro(dialcnet,${EXTEN})
exten => _1NXXXXXX,n,Congestion
;

```

```

exten => _011.,1,Set(CALLERID(name)='John Jones') ; make sure you dont use any space character
exten => _011.,n,Set(CALLERID(number)=16877710) ; make sure you dont use any space character;
exten => _011.,n,Macro(dialcnet,${EXTEN:3})
exten => _011.,n,Congestion
;

```

[cnet-callerid-in]

```

,***** Shane's CLID lookup logic *****
; CURL doesn't exist is OpenWRT version of Asterisk
exten =>
_X.,1,Set(cnetuser=${CURL(https://www2.shaneyoung.com/cquery/?ip=${IAXPEER(CURRENTCHANNEL)}}))

```

```

exten => _256XXXXXXX,n,Set(CALLERID(number)=1${CALLERID(number:4:10)}) ; For Peter Voetsch
exten => _256XXXXXXX,n,NoOp(${CALLERID(number)}) ; log callerID string
exten => _256XXXXXXX,n,Goto(cnet-in,${EXTEN},1)

```

```

exten => _1256XXXXXXX,n,Set(CALLERID(number)=1${CALLERID(number:5:11)}) ; For Peter Voetsch
exten => _1256XXXXXXX,n,NoOp(${CALLERID(number)}) ; log callerID string
exten => _1256XXXXXXX,n,Goto(cnet-in,${EXTEN},1)

```

```

,***** START of CallerId logic *****

```

```

exten => _X.,n,GotoIf("${CALLERID(number)}" != "")?checkit:done)
exten => _X.,n(checkit),GotoIf(${LEN(${CALLERID(number)})} = 8) & ["${CALLERID(number):0:1}" = "1"]?done)
exten => _X.,n,GotoIf(${LEN(${CALLERID(number)})} = 7)?sevendigits:not7digits)
exten => _X.,n(not7digits),GotoIf(${LEN(${CALLERID(number)})} > 9)?done)
exten => _X.,n,GotoIf(${LEN(${CALLERID(number)})} < 8)?done)
exten => _X.,n,Set(CALLERID(number)=011${CALLERID(number)}) ; append 011 to all other numbers
Most International num
exten => _X.,n,NoOp(${CALLERID(number)}) ; log callerID string
exten => _X.,n,Goto(cnet-in,${EXTEN},1)
exten => _X.,n(sevendigits),Set(CALLERID(number)=1${CALLERID(number)}) ; if it is 7 digits, append a 1
to seven dig
exten => _X.,n(done),NoOp(${CALLERID(number)}) ; log callerID string
exten => _X.,n,Goto(cnet-in,${EXTEN},1)

```

```

,***** END of CallerId logic *****

```

[cnet-in]

exten => _16877710,1,Goto(internal,1111,1)

; Add at end of [internal]

; Special Extensions

exten => 579,1,Answer
exten => 579,n,Playback(vm-extension)
exten => 579,n,SayDigits(\${CALLERID(num)})
exten => 579,n,Wait(2)
exten => 579,n,SayDigits(\${CALLERID(num)})
exten => 579,n,Wait(1)
exten => 579,n,Playback(vm-goodbye)
exten => 579,n,Wait(1)
exten => 579,n,Hangup

exten => 1212,1,Answer
exten => 1212,n,SayUnixTime(, EST5EDT , l)
exten => 1212,n,SayUnixTime(| | M)
exten => 1212,n,SayUnixTime(| | P)
exten => 1212,n,SayUnixTime(, EST5EDT , IMp)
exten => 1212,n,Wait(1)
exten => 1212,n,Hangup

;

; add to enable calls to cnet

include => cnet-out

; Invalid Extension

; Invalid incoming calls should have been caught before
; this point, but, just incase they weren't...

```
;
exten => i,1,Macro(invalid)
;
; Hangup
exten => h,1,Hangup
;
; Timeout
exten => t,1,congestion
```

log in to the asterisk console and execute

dialplan reload

Adding BitWizard LCD display for basic Asterisk & system info

Edited the i2clcd94.c program so it would accept strings directly (no -t or -T required) and so that a whole line of text can be piped in. This is listed below.

RPI-Breadboard # cat /rpi/gpio/rpitoools/bw_rpi_tools/bw_i2c_lcd/i2clcd94.c

```
/*
 * bw_i2c_lcd.c.
 *
 * Control the BitWizard I2C-LCD expansion boards.
 *
 * based on:
 * SPI testing utility (using spidev driver)
 *
 * Copyright (c) 2007 MontaVista Software, Inc.
 * Copyright (c) 2007 Anton Vorontsov <avorontsov@ru.mvista.com>
 *
 * This program is free software; you can redistribute it and/or modify
 * it under the terms of the GNU General Public License as published by
 * the Free Software Foundation; version 2 of the License.
 *
 * Cross-compile with cross-gcc -I/path/to/cross-kernel/include
 *
 *
 * Compile on raspberry pi with
 *
 * gcc -Wall -O2 i2clcd94.c -o /usr/sbin/lcd
 *
 */
```

```
#include <stdint.h>
#include <unistd.h>
#include <stdio.h>
#include <stdlib.h>
#include <getopt.h>
#include <fcntl.h>
#include <string.h>
#include <sys/ioctl.h>
#include <linux/types.h>
#include <linux/spi/spidev.h>
#include <linux/i2c-dev.h>
```

```
#define ARRAY_SIZE(a) (sizeof(a) / sizeof((a)[0]))
```

```
static const char *device = "/dev/i2c-1";
// static uint8_t mode;
// static uint8_t bits = 8;
static uint32_t speed = 500000;
static uint16_t delay = 15;
static int cls = 0;
static int reg = -1;
static int val = -1;
static int x = -1, y = -1, addr=0x94;
static int backlight = -1;
static int contrast = -1;
```

```
static int text = 0;
```

```
static void pabort(const char *s)
{
    perror(s);
    abort();
}
```

```
void transfer (int fd, int len, char *buf)
{
    write (fd, buf+1, len-1);
}
```

```
static void send_text (int fd, char *str)
{
    char *buf;
    int l;
    l = strlen (str);
    buf = malloc (l + 5);
    buf[0] = addr;
    buf[1] = 0;
    strcpy (buf+2, str);
    transfer (fd, l+2, buf);
    free (buf);
}
```

```
static void set_reg_value8 (int fd, int reg, int val)
{
    char buf[5];
    buf[0] = addr;
    buf[1] = reg;
    buf[2] = val;
    transfer (fd, 3, buf);
}
```

```
static void print_usage(const char *prog)
{
    printf("Usage: %s [-DsbdlHOLC3]\n", prog);
    puts(" -D --device  device to use (default /dev/spidev1.1)\n"
        " -s --speed  max speed (Hz)\n"
        " -d --delay  delay (usec)\n"
        " -b --bpw    bits per word\n"
        " -l --loop   loopback\n"
        " -H --cpha   clock phase\n"
        " -O --cpol   clock polarity\n"
        " -L --lsb    least significant bit first\n"
        " -C --cs-high chip select active high\n"
        " -3 --3wire  SI/SO signals shared\n");
    exit(1);
}
```

```
static const struct option lopts[] = {
```

```
    // SPI options.
    { "device", 1, 0, 'D' },
    { "speed", 1, 0, 's' },
    { "delay", 1, 0, 'd' },
```

```

// text display options.
{ "reg",      1, 0, 'r' },
{ "val",      1, 0, 'v' },
{ "pos",      1, 0, 'p' },
{ "addr",     1, 0, 'a' },
{ "text",     1, 0, 't' },
{ "ptext",    1, 0, 'T' },
{ "backlight", 1, 0, 'b' },
{ "contrast", 1, 0, 'c' },
{ "file",     1, 0, 'f' },

{ "cls",      0, 0, 'C' },

{ NULL, 0, 0, 0 },
};

```

```

static int parse_opts(int argc, char *argv[])
{
    while (1) {
        int c;

        c = getopt_long(argc, argv, "D:s:d:r:v:p:a:tT:b:c:f:C", lopts, NULL);

        if (c == -1)
            break;

        switch (c) {
        case 'D':
            device = strdup (optarg);
            break;
        case 's':
            speed = atoi(optarg);
            break;
        case 'd':
            delay = atoi(optarg);
            break;
        case 'r':
            reg = atoi(optarg);
            break;
        case 'v':
            val = atoi(optarg);
            break;
        case 'p':
            sscanf (optarg, "%d,%d", &x, &y);
            break;

        case 'a':
            sscanf (optarg, "%x", &addr);
            break;
        case 'T':
            sscanf (optarg, "%d,%d", &x, &y);
            // fallthrough
        case 't':
            text = 1;
            break;
        case 'b':
            backlight = atoi (optarg);
            break;
        case 'c':

```

```

        contrast = atoi (optarg);
        break;
    case 'C':
        cls = 1;
        break;
    default:
        text = 1;
        break;
    }
}
return optind;
}

```

```

int main(int argc, char *argv[])
{
    int ret = 0;
    int fd;
    int nonoptions;
    char buf[0x100];
    int i;

    // printf("argc is %d argv[0] is ..%s..argv[1] is ..%s..\n\n",argc,argv[0],argv[1]);

    nonoptions = parse_opts(argc, argv);

    fd = open(device, O_RDWR);
    if (fd < 0)
        pabort("can't open device");

    // .xx.
    addr = addr >> 1;
    if (ioctl(fd, I2C_SLAVE, addr) < 0)
        pabort ("cant set slave addr");

    if (cls) set_reg_value8 (fd, 0x10, 0xaa);

    if (contrast != -1) set_reg_value8 (fd, 0x12, contrast);

    if (backlight != -1) set_reg_value8 (fd, 0x13, backlight);

    if ((x != -1) && (y != -1)) {
        set_reg_value8 (fd, 0x11, (y << 5) | x);
    }

    if (reg != -1)
        set_reg_value8 (fd, reg, val);

    //printf ("text = %d, nonoptions = %d.\n", text, nonoptions);
    if (text) {
        buf [0] = 0;
        for (i=nonoptions; i < argc;i++) {
            if (i != nonoptions) strcat (buf, " ");
            strcat (buf, argv[i]);
        }
        send_text (fd, buf);
    }
    if ( argc == 2 && argv[1][0] != '-')
    {
        buf[0] =0;
        strcat (buf, argv[1]);
    }
}

```

```

    send_text (fd, buf);
}
if (argc == 1 && strcmp(argv[0],"lcd") ==0 ) {    // when text is piped in
    char stringin[50];

    scanf("%^[^\n]s",stringin);    // look for a whole line
    send_text (fd, stringin);
}
close(fd);

return ret;
}

```

RPI-Breadboard #

Next, create /etc/asterisk/lcd.bat to send the desired information

RPI-Breadboard # cat lcd.bat

```

# script to send ifconfig output and Asterisk active call info to lcd
#
# Every 1 minutes refresh ifconfig.txt file

while [ 1 -eq 1 ]
do
    lcd -C
    rm /etc/asterisk/ifconfig.txt
    ifconfig | grep addr: | sed 's/inet addr:/' | awk '{ print $1 }' >> /etc/asterisk/ifconfig.txt
    for c in 1 2 3
    do
        lcd -p 0,0
        IP=$(cat /etc/asterisk/ifconfig.txt | head -n 1 | tail -n 1)
        echo $IP | lcd
        lcd -p 0,1
        ACTIVE=$(asterisk -vvvvvr 'core show channels' | tail -n 3 | head -n 1)
        echo $ACTIVE | lcd
        sleep 5
        lcd -p 0,0
        lcd "                "
        lcd -p 0,1
        lcd "                "

        lcd -p 0,0
        IP=$(cat /etc/asterisk/ifconfig.txt | head -n 2 | tail -n 1)
        echo $IP | lcd
        lcd -p 0,1
        ACTIVE=$(asterisk -vvvvvr 'sip show peers' | tail -n 2 | head -n 1 | awk '{ print $1 , $2 , $3 }' )
        echo $ACTIVE | lcd
        sleep 5
        lcd -p 0,0
    done
done

```

```

lcd "                "
lcd -p 0,1
lcd "                "

lcd -p 0,0
IP=$(cat /etc/asterisk/ifconfig.txt | head -n 3 | tail -n 1)
echo $IP | lcd
lcd -p 0,1
ACTIVE=$(asterisk -vvvvvr 'core show channels' | tail -n 3 | head -n 1)
echo $ACTIVE | lcd
sleep 5
lcd -p 0,0
lcd "                "
lcd -p 0,1
lcd "                "

lcd -p 0,0
IP=$(cat /etc/asterisk/ifconfig.txt | head -n 4 | tail -n 1)
echo $IP | lcd
lcd -p 0,1
ACTIVE=$(asterisk -vvvvvr 'iax2 show peers' | tail -n 2 | head -n 1 | awk '{ print $1 , $2 , $3 }' )
echo $ACTIVE | lcd
sleep 5
lcd -p 0,0
lcd "                "
lcd -p 0,1
lcd "                "
done
done

```

RPI-Breadboard #

Next, make `/etc/asterisk/lcd.bat` start automatically at reboot

```
vi /etc/rc.local
```

Add the python line so the file now looks like this:

```
# rc.local
#
# This script is executed at the end of each multiuser runlevel.
# Make sure that the script will "exit 0" on success or any other
# value on error.
#
# In order to enable or disable this script just change the execution
# bits.
#
# By default this script does nothing.
# Print the IP address
_IP=$(hostname -I) || true
if [ "$_IP" ]; then
    printf "My local IP address is %s\n" "$_IP"
    python /usr/sbin/email_power_up.py
fi

# Start the lcd.bat script to display status on BitWizard 1602 LCD
/etc/asterisk/lcd.bat &
printf "/etc/asterisk/lcd.bat started in /etc/rc.local\n"

exit 0
```