These are some simple Asterisk configuration files to help you understand how it works.

You have to define your connections first in sip.conf or iax.conf. Sip.conf is used for hardware that supports the SIP protocol and IAX is usually used for inter system trunking but some soft phone clients like Zoiper also support it. I tend to save the original conf files as something like sip.sample and create a new file. Its cleaner and you still have the heavily documented original to refer to when necessary.

```
[general]
context=incoming
allowoverlap=no
udpbindaddr=0.0.0.0
tcpenable=no
tcpbindaddr=0.0.0.0
srvlookup=yes
externip=63.229.123.123                ; This may be needed to get around NAT issues
localnet=192.168.0.0/255.255.255.0  ; Your internal network
canreinvite=yes
registertimeout=20
registerattempts=0

; This Asterisk box is called Phoenix.  The next config registers it with a server
; called Tucson and sets up the SIP configuration for the server named Tucson to register to.

register => tucson:welcome@192.168.0.8/phoenix

[phoenix]
type=friend
secret=welcome
context=cohncom2
host=dynamic
disallow=all
allow=ulaw
qualify=yes
```

```
[authentication]

[6104]                          ; Standard config for Polycom SIP phone
type=friend
context=cohncom2
secret=6104
host=dynamic
qualify=yes
dtmfmode=rfc2833
defaultuser=6104
defaultip=192.168.0.5           ; IP of this Asterisk server
disallow=all
allow=ulaw
nat=yes
progressinband=no
mailbox = 6104
```

Next we look at the extensions configuration. I generally create a context for each function then just include them in my “customer” context as need to allow access.

```
[general]
static=yes
writeprotect=no
clearglobalvars=no

[globals]
CONSOLE=Console/dsp             ; Console interface for demo
TRUNK=DAHDI/3-1                 ; Trunk interface, port 3 on analog card one
TRUNKMSD=1                      ; MSD digits to strip (usually 1 or 0)
```

```
[trunkld]
exten => _91NXXNXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})

[trunklocal]
exten => _9NXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})

[trunktollfree]
exten => _91800NXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})
exten => _91888NXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})
exten => _91877NXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})
exten => _91866NXXXXXX,1,Dial(${GLOBAL(TRUNK)}/${EXTEN:${GLOBAL(TRUNKMSD)}})

[international]                    ; Not defined yest
ignorepat => 9
include => longdistance

[longdistance]
ignorepat => 9
include => local
include => trunkld

[local]
ignorepat => 9
include => trunklocal
include => trunktollfree

[incoming]

exten => s,1,Answer()
exten => s,2,Wait(2)
exten => s,3,Background(enter-ext-of-person)
exten => s,4,WaitExten()
```

```
exten => _XXXX,1,Set(CALLERID(num)=5201234567)
exten => _XXXX,n,Set(CALLERID(name)=Phoenix)
exten => _XXXX,n,NoOp() ; Wildcard extensions for Tucson pbx
exten => _XXXX,n,Dial(SIP/phoenix/${EXTEN})
exten => _XXXX,n,Playback(all-outgoing-lines-unavailable)
exten => _XXXX,n,Wait(2)
exten => _XXXX,n,Playback(goodbye)
exten => _XXXX,n,Hangup()

exten => 6000,1,Gosub(ani)
exten => 6101,1,Goto(cohncom2,6101,1)
exten => 6102,1,Goto(cohncom2,6102,1)
exten => 6104,1,Goto(cohncom2,6104,1)

exten => i,1,Wait(2)
exten => i,2,Playback(pbx-invalid)
exten => i,3,Goto(incoming,s,4)
exten => t,1,Playback(goodbye)
exten => t,2,Hangup()

; This is my implementation of a method to alert telemarketers that I don't want calls from them
; It uses a custom recorded message:  wget www.gwcohn.com/telemarketer.wav to download

[telemarketer]
exten => s,1,Answer()
exten => s,n,GoToIf($[${CALLERID(num)} = 5203490859]?allow)         ; Number on allowed list, bypasses message
                                                                    ; goes right to ext 6104
exten => s,n,Zapateller()
exten => s,n,Background(telemarketer)
exten => s,n,WaitExten()
exten => s,n(allow),Goto(cohncom,6104,1)
exten => 4,1,Goto(cohncom2,6104,1)
exten => i,1,Playback(pbx-invalid)
```

```
exten => i,n,Goto(telemarketer,s,1)

exten => t,1,Playback(goodbye)
exten => t,n,Hangup()

; Original demo from default extensions.conf file.  Handy for initial testing.

[demo]

;exten => s,1,Wait(1)                    ; Wait a second, just for fun
exten => s,1,Answer                      ; Answer the line
exten => s,n,Set(TIMEOUT(digit)=5)      ; Set Digit Timeout to 5 seconds
exten => s,n,Set(TIMEOUT(response)=10)  ; Set Response Timeout to 10 seconds
exten => s,n(restart),BackGround(demo-congrats) ; Play a congratulatory message
exten => s,n(instruct),BackGround(demo-instruct)        ; Play some instructions
exten => s,n,WaitExten                   ; Wait for an extension to be dialed.

exten => 2,1,BackGround(demo-moreinfo)  ; Give some more information.
exten => 2,n,Goto(s,instruct)

exten => 3,1,Set(LANGUAGE()=fr)         ; Set language to french
exten => 3,n,Goto(s,restart)            ; Start with the congratulations

exten => 1000,1,Goto(default,s,1)
exten => 1234,1,Playback(transfer,skip)         ; "Please hold while..."
                                    ; (but skip if channel is not up)
exten => 1234,n,Gosub(${EXTEN},stdexten(${GLOBAL(CONSOLE)}))
exten => 1234,n,Goto(default,s,1)               ; exited Voicemail

exten => 1235,1,Voicemail(1234,u)               ; Right to voicemail

exten => 1236,1,Dial(Console/dsp)               ; Ring forever
exten => 1236,n,Voicemail(1234,b)               ; Unless busy
```

```
exten => #,1,Playback(demo-thanks)      ; "Thanks for trying the demo"
exten => #,n,Hangup                     ; Hang them up.

exten => t,1,Goto(#,1)                  ; If they take too long, give up
exten => i,1,Playback(invalid)          ; "That's not valid, try again"

exten => 500,1,Playback(demo-abouttotry); Let them know what's going on
exten => 500,n,Dial(IAX2/guest@pbx.digium.com/s@default)        ; Call the Asterisk demo
exten => 500,n,Playback(demo-nogo)      ; Couldn't connect to the demo site
exten => 500,n,Goto(s,6)                ; Return to the start over message.

exten => 600,1,Playback(demo-echotest)  ; Let them know what's going on
exten => 600,n,Echo                     ; Do the echo test
exten => 600,n,Playback(demo-echodone)  ; Let them know it's over
exten => 600,n,Goto(s,6)                ; Start over


[voicemail]
exten => 8500,1,VoicemailMain                          ; Number to access voicemail system
exten => 8500,n,Goto(s,6)

; Default time subroutine from default extensions.conf

[time]
exten => _X.,30000(time),NoOp(Time: ${EXTEN} ${timezone})
exten => _X.,n,Wait(0.25)
exten => _X.,n,Answer()
exten => _X.,n,Set(FUTURETIME=$[${EPOCH} + 12])
exten => _X.,n,SayUnixTime(${FUTURETIME},Zulu,HNS)
exten => _X.,n,SayPhonetic(z)
exten => _X.,n,SayUnixTime(${FUTURETIME},${timezone},HNS)
exten => _X.,n,Playback(spy-local)
```

```
exten => _X.,n,WaitUntil(${FUTURETIME})
exten => _X.,n,Playback(beep)
exten => _X.,n,Return()

; Automatic Number Identification.  Plays back phone number when called.

[ani]
exten => _X.,50000(ani),NoOp(ANI: ${EXTEN})
exten => _X.,n,Wait(0.25)
exten => _X.,n,Answer()
exten => _X.,n,Wait(1.00)
exten => _X.,n,SayDigits(${CALLERID(ani)})
exten => _X.,n,Wait(1.00)
exten => _X.,n,SayDigits(${CALLERID(ani)})      ; playback again in case of missed digit
exten => _X.,n,Wait(1.00)
exten => _X.,n,Playback(goodbye)
exten => _X.,n,Return()

; My “customer” Asterisk can support multiple customers and they can’t call each other directly unless you allow it
; by including them.  Note that I have included several contexts from above that are available for this customer.

[cohncom2]
include => voicemail
include => ani
include => demo
include => incoming
include => telemarketer

exten => 6101,1,Set(CALLERID(num)=6101)                    ; Analog phone with voicemail
exten => 6101,n,Dial(DAHDI/1,20,rt)
exten => 6101,n,Voicemail(6101,u)
exten => 6101,n,dial+101,Voicemail(6101,b)
```

```
exten => 6102,1,Set(CALLERID(num)=6102)               ; Analofg phone without voicemail
exten => 6102,n,Dial(DAHDI/2,20,rt)
exten => 6102,n,Playback(nbdy-avail-to-take-call)     ; After 20 seconds it says no one available
exten => 6102,n,Wait(1)
exten => 6102,n,Playback(goodbye)                     ; Plays goodbye
exten => 6102,n,Hangup()                              ; Hangs up

exten => 4000,1,Gosub(ani)                            ; Number to call ANI
exten => 4000,n,Hangup()

exten => 2222,1,Goto(telemarketer,2222,1)             ; Number to test telemarketer
exten => 2222,n,Hangup()

exten => 6104,1,Dial(SIP/6104,20,rt);                 ; Polycom SIP phone
exten => 6104,n,Voicemail(6104,u)                     ; Voicemail (unavailable)
exten => 6104,n,Hangup()
exten => 6104,n,dial+101,Voicemail(6104,b)            ; Voicemail (busy)
```

And last but not least, the voicemail configuration.

```
[general]
format=wav49|gsm|wav
serveremail=asterisk
attach=yes
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3
emaildateformat=%A
sendvoicemail=yes


[zonemessages]
```

```
eastern=America/New_York|'vm-received' Q 'digits/at' IMp
central=America/Chicago|'vm-received' Q 'digits/at' IMp
central24=America/Chicago|'vm-received' q 'digits/at' H N 'hours'
military=Zulu|'vm-received' q 'digits/at' H N 'hours' 'phonetic/z_p'
european=Europe/Copenhagen|'vm-received' a d b 'digits/at' HM

[default]

; 1234 => 4242,Example Mailbox,root@localhost
6104 => 6104,George Cohn,gwcohn@pima.edu,,Tz=Phoenix
6101 => 6101,George Cohn,gwcohn@oima.edu,,Tz=Phoenix
```