Patton 4114 (4 FXO)

The following is a configuration for a Patton 4114 4 port FXO gateway.

This is a simple file assuming you are using three digit extensions
and using all 4 ports for FXO use and a device with a sigle ethernet port.
it starts dialing out port three and hunts to port 1 for outbound calls.
make sure you bind the ethernet port of your sip gateway
to the proper ethernet port as shown below.

```
gateway sip GW-SIP
bind interface eth0 router
```

After importing the config file, remeber to save and copy to running-config.

Remove all of the above lines when ready to save the config and import it.

#--#

1. #
2. SN4114/JO/EUI #
3. R3.20 2006-07-27 H323 SIP FXS FXO #
4. 1970-01-01T00:15:40 #
5. Generated configuration file #
6. #
   #--#

cli version 3.20

1. set your password in the system first, then do an EXPORT config, REMOVE THIS LINE
2. replace the line below with your username and encrypted password REMOVE THIS LINE
   administrator admin password xxxxxxxx encrypted
3. replace these nameservers with your own REMOVE THIS LINE
   dns-client server 198.6.1.5
   dns-client server 198.6.1.2
   dns-relay
   webserver port 80 language en
4. good idea to change the snmp read string from default of "public" REMOVE THIS LINE
   snmp community public ro
   sntp-client
5. put in your own ntp server(s) REMOVE THIS LINE
   sntp-client server primary 192.5.41.41 port 123 version 4
   sntp-client server secondary 192.5.41.40 port 123 version 4
   sntp-client poll-interval 36000
   sntp-client local-clock-offset
6. put in your own timezone REMOVE THIS LINE
   sntp-client gmt-offset - 05:00:00
7. change the hostname to your own for this device REMOVE THIS LINE
   system hostname patton.mydomain.com

system

ic voice 0
low-bitrate-codec g729

profile napt NAPT

profile ppp default

profile call-progress-tone US_Dialtone
play 1 0 350 -13 440 -13

profile call-progress-tone US_Alertingtone
play 1 2000 440 -19 480 -19
pause 2 4000

profile call-progress-tone US_Busytone
play 1 500 480 -24 620 -24
pause 2 500

profile tone-set default

profile tone-set US
map call-progress-tone dial-tone US_Dialtone
map call-progress-tone ringback-tone US_Alertingtone
map call-progress-tone busy-tone US_Busytone
map call-progress-tone release-tone US_Busytone
map call-progress-tone congestion-tone US_Busytone

profile voip default
codec 1 g711alaw64k rx-length 20 tx-length 20
codec 2 g711ulaw64k rx-length 20 tx-length 20

profile pstn default

profile sip default

profile aaa default
method 1 local
method 2 none

context ip router

1. ip address and mask of this device REMOVE THIS LINE

interface eth0
ipaddress 192.168.1.11 255.255.255.0
tcp adjust-mss rx mtu
tcp adjust-mss tx mtu

context ip router

1. default route of this device REMOVE THIS LINE
   route 0.0.0.0 0.0.0.0 192.168.1.1 0

context cs switch
digit-collection timeout 3

routing-table called-e164 TAB-OUT
route .%T dest-interface IF-SIP

routing-table called-e164 TAB-IN
route .%T dest-service FXOHUNT

interface sip IF-SIP
bind gateway GW-SIP
service default
route call dest-table TAB-IN

1. your sipxpbx SIP Domain Name (hostname if not using SRV records) REMOVE THIS LINE
   remote mydomain.com

interface sip IF-SIP1
bind gateway GW-SIP
service default
route call dest-table TAB-IN

1. your sipxpbx SIP Domain Name (hostname if not using SRV records) REMOVE THIS LINE
   remote mydomain.com
2. change the '100' to the number for your auto attendant REMOVE THIS LINE
   address-translation outgoing-call to-header user-part fix 100 host-part interface

interface fxo PSTN
ring-number on-caller-id
use profile tone-set US

interface fxo IF_FXO0
route call dest-interface IF-SIP1
disconnect-signal busy-tone
ring-number on-caller-id
dial-after timeout 2
mute-dialing
use profile tone-set US

interface fxo IF_FXO1
route call dest-interface IF-SIP1
disconnect-signal busy-tone
ring-number on-caller-id
dial-after timeout 2
mute-dialing
use profile tone-set US

interface fxo IF_FXO2
route call dest-interface IF-SIP1
disconnect-signal busy-tone
ring-number on-caller-id
dial-after timeout 2
mute-dialing
use profile tone-set US

interface fxo IF_FXO3
route call dest-interface IF-SIP1
disconnect-signal busy-tone
ring-number on-caller-id
dial-after timeout 2
mute-dialing
use profile tone-set US

service hunt-group FXOHUNT
drop-cause normal-unspecified
drop-cause no-circuit-channel-available
drop-cause network-out-of-order
drop-cause temporary-failure
drop-cause switching-equipment-congestion
drop-cause access-info-discarded
drop-cause circuit-channel-not-available
drop-cause resources-unavailable
drop-cause user-busy
drop-cause destination-out-of-order
route call 1 dest-interface IF_FXO3
route call 2 dest-interface IF_FXO2
route call 3 dest-interface IF_FXO1
route call 4 dest-interface IF_FXO0

context cs switch
no shutdown

gateway sip GW-SIP
bind interface eth0 router

service default

1. your domain, dns_srv records (host name if not using SRV) may be different REMOVE THIS LINE
   domain mydomain.com
2. your sipxpbx sip domain (host name if not using SRV) REMOVE THIS LINE
   default-server mydomain.com loose-router

gateway sip GW-SIP
no shutdown

port ethernet 0 0
medium auto
encapsulation ip
bind interface eth0 router
no shutdown

port fxo 0 0
flash-hook-duration 50
use profile fxo us
caller-id format bell
encapsulation cc-fxo
bind interface IF_FXO0 switch
no shutdown

port fxo 0 1
flash-hook-duration 50
use profile fxo us
caller-id format bell
encapsulation cc-fxo
bind interface IF_FXO1 switch
no shutdown

port fxo 0 2
flash-hook-duration 50
use profile fxo us
caller-id format bell
encapsulation cc-fxo
bind interface IF_FXO2 switch
no shutdown

```
port fxo 0 3
flash-hook-duration 50
use profile fxo us
caller-id format bell
encapsulation cc-fxo
bind interface IF_FXO3 switch
no shutdown
```