Patton 4524 4 port FXO gateway (firmware 6.1)

Below is a sample config that will set up all 4 lines to ring the auto attendant at "100" and all outbound calls to use ports on reverse order (inbound ports 0/1 /2/3 and outbound 3/2/1/0).

You can shutdown the fxo ports not used with "shutdown" instead of "no shutdown, just remember to remove the interface in your outbound hunt group by remming out the corresponding "route call" statement.

Just copy this code and edit it to suit your network as a plain text file. Upload it to your gateway via the webgui (import) and then reload without saving changes to initialize it.

```
cli version 3.20
clock local default-offset -05:00
# this assumes your time zone is USA, New York. You can replace your clock offset to reflect your timezone and your DST periods
clock local dst-rule SPRING2011 -04:00 from 02:00 mar 13rd 2011 until 03:00 nov 6th 2011
clock local dst-rule SPRING2012 -04:00 from 02:00 mar 11st 2012 until 03:00 nov 4th 2012
clock local dst-rule SPRING2013 -04:00 from 02:00 mar 10th 2013 until 03:00 nov 3rd 2013
clock local dst-rule SPRING2014 -04:00 from 02:00 mar 9th 2014 until 03:00 nov 2nd 2014
clock local dst-rule SPRING2015 -04:00 from 02:00 mar 8th 2015 until 03:00 nov 1st 2015
clock local dst-rule SPRING2016 -04:00 from 02:00 mar 13rd 2016 until 03:00 nov 6th 2016
#best to use sipx as dns server or whatever dns sipx uses
dns-client server 192.168.54.2
webserver port 80 language en
#use sipx as timeserver or another source allowed on or through your network
sntp-client server 192.5.41.40
# this device hostname
system hostname sip-gw.voice.mydomain.loc

system

 ic voice 0
   low-bitrate-codec g729

profile ppp default

profile call-progress-tone US_Dialtone
 play 1 1000 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
 no autonomous-transitioning

profile aaa default
 method 1 local
 method 2 none

context ip router
```

```
 interface LAN
#the ip and mask of this device
   ipaddress 192.168.54.3 255.255.255.0
   tcp adjust-mss rx mtu
   tcp adjust-mss tx mtu

context ip router
#the router of this network
 route 0.0.0.0 0.0.0.0 192.168.54.1

context cs switch
 digit-collection timeout 3

 routing-table called-e164 SIP_TO_ISDN
   route default dest-service OUTBOUND

 interface sip IF_SIPX
   bind context sip-gateway GW-SIP
   route call dest-table SIP_TO_ISDN
#sipx sip domain name
   remote pbx.voice.mydomain.loc
#use your sip hostname below and your destination, the system AA at "100" is used for this example
   address-translation outgoing-call to-header user-part fix 100 host-part fix pbx.voice.mydomain.loc

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

 service hunt-group OUTBOUND
   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
```

```
   route call 1 dest-interface IF_FXO3
   route call 2 dest-interface IF_FXO2
   route call 3 dest-interface IF_FXO1
   route call 3 dest-interface IF_FXO0

context cs switch
 no shutdown

location-service SIPX_SERVER
 domain 1 sipx.voice.mydomain.loc

context sip-gateway GW-SIP

 interface IF_SIPX
   bind interface LAN context router port 5060

context sip-gateway GW-SIP
 bind location-service SIPX_SERVER
 no shutdown

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

port ethernet 0 1
 medium 10 half
 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
```