

## F X S P O R T S

FXS (Foreign eXchange Station)—These POTS ports connect to telephones, fax machines, etc.

```
R# show voice port summary
    Voice-only ports (FXO & FXS) show up w/o config
    If CME running on the router, each configured ephone-DN shows up as an EXFS port
    fxs-ls is loop start, etc.
R(config)# voice-port 0/0/0
R(config-voiceport)# signal { groundstart | loopstart }
    Loop Start default—see chapter 1 for definitions
```

Call Progress Tones—Dial tone, busy signal, etc. default to USA-style

```
R(config-voiceport)# cptone { US | FR | GB | ... }
R(config-voiceport)# station-id name 3rd floor fax
R(config-voiceport)# station-id number 5551000
    For On-Net Caller ID (name and number)
```

## F X O P O R T S

FXO (Foreign eXchange Office)—POTS “trunk” line to telco or PBX—in and out.

```
R(config-voiceport)# signal { groundstart | loopstart }
R(config-voiceport)# station-id name 3rd floor fax
R(config-voiceport)# station-id number 5551000
    Preceding three commands same as FXS config. Station-ID will be overridden on incoming
    calls if the PSTN provides ANI
R(config-voiceport)# dialtype { dtmf | pulse }
    Some countries can't handle touch tone dialing
R(config-voiceport)# ring number 3
    Number of rings before router answers—default 1
```

## D I G I T A L V O I C E P O R T S

VVIC (Voice & Wan Interface Card)—Must be configured so the router knows if the T1 is voice or data before they show up in “show voice port summary.”

In Cisco's world, T1 channels are numbered 0-23 and the time slots they occupy are numbered 1-24.

## C A S ( C H A N N E L A S S O C I A T E D S I G N A L I N G )

Channels in a group—Different collections of T1 channels (time slots) with different destinations can be split out later by a demuxer. For example, Channels 1-5 to the PBX and 6-24 to the PSTN. Groups numbers are arbitrary in the range 0-23. VVIC cards are limited to 2 groups.

Signaling Type—Telcos tend to use fxo-loop-start over T1 CAS; PBXs tend to use one of several E&M (Ear & Mouth) standards.

Configuration—use “ds0-group” to configure VWIC card.

```
R# show controllers t1
    This will show the unconfigured VWIC cards. Normal commands don't show them because the
    router doesn't know if they're voice or data devices yet
R(config)# controller t1 1/0
R(config-controller)# framing { esf | sf }
    Extended SuperFrame (ESF) or SuperFrame, dictated by telco. USA usually ESF
R(config-controller)# linecode { ami | b8zs }
    Dictated by telco; USA usually B8ZS
R(config-controller)# clock source { free-running | internal | line }
    Line=telco provides the clock, internal useful for PBX connection
R(config-controller)# ds0-group 1 1-24 type fxo-loop-start
    First number is group number (0..23 because there can't be more groups than the 24 channels
    in a T1). Second number or range tells which time slots belong to that group, numbered from 1.
    The "type" field (fxo-loop-start) is Signaling Type, details follow.
R# show voice port summary
    One line per voice port (see "voice ports" below)
```

Voice Ports—Created automatically for each time slot (channel) mentioned in the ds0-group command. Notation 1/0:1 refers to physical interface 1/0 and DS0 group number :1. This port number is what will be used to configure dial peers. The time slot (channel) is a separate column.

#### CCS (COMMON CHANNEL SIGNALING)

CCS is AKA PRI (ISDN Primary Rate Interface) use “pri-group.”

```
R(config)# isdn switch-type { primary-5ess | ... }
    Match whatever kind of ISDN switch the telco is using
R(config)# controller t1 1/0
R(config-controller)# pri-group timeslots 1-24
    No option on signaling type—router assumes ISDN PRI
    If the router doesn't have enough DSPs, it'll give an error message telling how many channels it
    can handle.
R# show voice port summary
    Sig-type shows as isdn-voice this time instead of "fxo-ls" or whatever signaling type was used
    with the CAS group.
```

Voice Ports—Notation 1/0:23, shown in the command above and used to create dial peers, refers to the physical port 1/0 and channel 23 (=time slot 24), the shared signaling channel. Channels are listed from 0-23, and time slots are 1-24. Confusingly, the time slot column, numbered 1-23, is labeled “CH.” If we were using an E1 line, channel 16 = time slot 17 would be used for signaling.

#### DIAL PEERS

Dial peers are analogous to static routes for the voice network. A CME router doesn't need a dial peer for a locally configured ephone-dn, but it can't use FXS, FXO, or T1 connections without them. An analog phone in an FXS port will receive a dial tone and can make calls without a dial peer, but can't receive calls because it doesn't have a number. The phone number is defined in a dial peer that sends calls matching that number out that analog port.

Two types of dial peers:

- POTS—Any device connected to FXS, FXO, E&M, or digital voice port
- VOIP—Any device reachable through an IP address

Voice Call Leg—A connection to *or* from a voice gateway from a POTS or VoIP source. Each hop has two legs, outgoing and incoming. Each leg needs a dial peer. (static voice route) in each direction. This can total 4 dial peers for a hop, though sometimes a single dial peer can be bidirectional.

## P O T S D I A L P E E R S

Used for things without an IP address, even if they're digital, like a T1 to the PSTN

```
R(config)# dial-peer voice 1101 pots
    This 1101 is only a tag, not a phone number. Tags are local to the router they're on.
    Only need to specify dial peer type (pots) on creation; can omit when modifying.
    To change the type, e.g. to "voip," delete the whole dial peer and recreate.
R(config-dial-peer)# destination-pattern 1101
    THIS time it's a phone number.
R(config-dial-peer)# port 0/0/0
    The destination (an FXO or FXS)
R(config-dial-peer)# port 1/0:23
    Alternate version for T1 PRI (:23 is the shared signaling channel)
R# show dial-peer voice [ summary ]
    Displays all the dial peers on the router, with a page of output each. Summary has 1 line each.
R# debug voip dialpeer
    Shows each digit, as it is dialed, and finally the rule that matched
```

Wildcards

```
R(config)# dial-peer voice 2000 pots
R(config-dial-peer)# destination-pattern 2...
R(config-dial-peer)# no digit-strip
    Send all 4 digits, don't strip off literals, i.e. the "2"
```

WILDCARD	MEANING	EXAMPLE
.	(period)	One digit ( 0-9 or the * key )
+	One or more instances of the preceding digit	5+23 can be 523 or 5523 or ...
[ ]	One of a range of digits	[1-3]22 can be 122, 222 or 322
^	Negation used with brackets	[^1-6]22 can be 022, 722, 822, 922, or *22
T	Any number (0-32) of dialed digits	For 1-32 digits, use ".T" to force ≥1 digits
,	(comma)	Inserts a 1-second pause between dialed digits

The # key isn't a wildcard; when typed on a phone, it resolves ambiguous rules by telling the router that you're done dialing. Otherwise you wait for the 10 second "interdigit timeout." This is a good reason not to use a rule "9T"—the router won't know when you're done dialing.

North American PSTN dialing patterns (p134, table 6-5) (example 6-11 shows ios config)

PATTERN	MEANING
[2-9].....	7-digit dialing areas
[2-9].[2-9].....	10-digit dialing areas
1[2-9].[2-9].....	
[469]11	
011T	International

Digit-Stripping Rule (POTS dial peers only)—Any dialed digits that are explicitly matched by the dial peer rule (literals) are stripped from the destination phone number before the call is forwarded (unless you prevent it). Great for dialing 9 for an outside line, i.e. 9.....

```
R(config-dial-peer)# no digit-strip
```

```
R(config-dial-peer)# destination-pattern 9[469]11
R(config-dial-peer)# forward-digits 3
The rightmost 3 digits, e.g. 411, are sent to the PSTN
```

```
R(config-dial-peer)# destination-pattern 9011T
R(config-dial-peer)# description International Dialing
R(config-dial-peer)# prefix 011
Allows 9011 to be stripped off automatically, then puts 011 back on the front. Perfect here because you don't know how long an international number is, so can't use "forward-digits"
```

#### V O I P D I A L P E E R S

Used for hops between routers, over an IP-based network. Main difference is “session target” command.

```
R(config)# dial-peer voice 2000 voip
R(config-dial-peer)# description Optional Description
R(config-dial-peer)# destination-pattern 2...
R(config-dial-peer)# session target ipv4:10.1.1.2
Remember the ipv4: format; context help just says "word"
Can also use dns:<name>
R(config-dial-peer)# codec g711ulaw
optional—default G.729. If mismatch between routers, the call fails with a fast busy (reorder tone)
```

PLAR (Private Line Automatic Ringdown)—Tells a port to automatically dial a number when it detects an off-hook. Can be used to send a call to the operator when it arrives on an FXO from the PSTN or if an FXS lobby phone (no dial) handset is lifted.

```
R(config)# voice-port 0/0/0
R(config-voiceport)# connection plar 1102
```

Incoming FXO calls from the PSTN have caller ID (Automatic Number Identification) but don't tell what number was dialed. Without PLAR, the caller would receive a second (internal) dial tone instead of the operator. This doesn't matter with digital PSTN lines (T1) because they *do* have DNIS information for incoming call handling with DID (Direct Inward Dial) services.

Dial Peers—Like static routing table entries, but for phone calls. As dialed digits accumulate,

- Upon match, call immediately processed. Possible longer matches are ignored
- When it has enough to make a match, the most Specific match (least wildcard involvement) wins—like routing

```
dial-peer voice 1 voip
destination-pattern 555[1-3]...
session target ipv4:10.1.1.1
```

```
dial-peer voice 2 voip
destination-pattern 5551...
```

*If rule two matches, it will win over rule 1, which also matches, but with more wildcards. A third peer with a 4-digit pattern of "5551" would win over them both because the call would be immediately processed without waiting to see if 3 more digits would have been dialed.*

```
session target ipv4:10.1.1.2
```

```
R# show dialplan number 5551234
```

*Tells which dial-peer would be matched.*

If overlapping dial plans can't be avoided, e.g. 5551 and 5551..., use the T (0-32 digits) to force the user to dial "#" or wait 10 seconds to dial the shorter. i.e., 5551T and 5551... peers.

Outbound Dial Peers—what you would expect; they compare dialed digits to determine which outbound interface to use.

Inbound Dial Peers—Matched by one of 5 methods (in order of preference):

1. Incoming Dialed Number—Match the dialed number (DNIS) using the “incoming dialed number” dial peer configuration command
2. Answer Address—Match the caller ID information (ANI) using the “answer address” dial peer configuration command
3. Destination Pattern—Match the caller ID information (ANI) using the “destination-pattern” dial peer configuration command
4. Port Dial-Peer—Match an incoming POTS dial peer using a dial peer with a “port” dest.
5. Dial Peer 0—If no other match is found. Like a default gateway only for incoming dial peers. Cannot be configured or changed; you're stuck with the following settings:
  - Any Codec—handles any inbound codec
  - No DTMF Relay—touch-tone codes stay inside the audio stream, where they may be distorted by a codec
  - IP Precedence 0—strips all QoS and treats the call like data
  - VAD (Voice Activity Detection) Enabled—noise gate with no data sent during silence
  - No RSVP (Resource Reservation Protocol)—just like the rest of QoS
  - Fax-Rate Voice—Faxes can't use any more bandwidth than voice. If you're using a highly compressed codec (low bandwidth), then faxes could be devastated.
  - No Application Support—Can't refer the call to external applications, e.g. an IVR (Interactive Voice Response) system
  - No DID (Direct Inward Dialing)—Can't auto-forward PSTN calls to internal devices

## D I G I T M A N I P U L A T I O N

COMMAND	MODE	DESCRIPTION
prefix-digits	POTS dial peer	e.g “prefix 011” adds 011 to the beginning of the dialed number
forward-digits #	POTS dial peer	Forward only the last (rightmost) # digits from the dialed number
[no] digit-strip	POTS dial peer	
num-exp	global	With “num-exp 4... 5...” dialing extension 4123 gets you 5123 With “num-exp 0 5000” dialing 0 gets you extension 5000
voice translation-profile	global and POTS or VOIP dial peer	Translation profile consists of up to 15 rules to change numbers. Created globally and applied to as many dial peers as desired (like an ACL on interfaces)

Example—WAN to PSTN failover—Dialing remote office extensions (6...) just like internal (5...), prefer to use WAN for site-to-site. Lower preference wins, identical preferences are selected from randomly. Manipulations such as digit-strip, forward-digits, and prefix are only valid for POTS dial peers.

```
R(config)# dial-peer voice 10 voip
R(config-dial-peer)# destination-pattern 6...
R(config-dial-peer)# session target ipv4:10.1.1.2
R(config-dial-peer)# preference 0
R(config-dial-peer)# exit
R(config)# dial-peer voice 11 pots
R(config-dial-peer)# destination-pattern 6...
R(config-dial-peer)# port 1/0:1
We know this is a T1/E1 with CAS (:23 for T1 or :17 for E1 could be CCS or CAS)
R(config-dial-peer)# preference 1
R(config-dial-peer)# no digit-strip
Don't strip the 6 from the 4-digit extension number.
R(config-dial-peer)# prefix 1512555
Notice that the digit manipulation is local to the dial-peer (just things going T1 to PSTN)
```

Example—Send Operator Calls (0) to the receptionist (5000). Since this transformation should always happen (IP phone, FXS port, anything), use num-exp global configuration command.

```
R(config)# voice-port 1/0/1
R(config-voiceport)# connection plar 0
R(config-voiceport)# exit
R(config)# num-exp 0 5000
```

The expansion happens first, before any attempt is made to match an inbound dial peer.

Example—Send 911 calls out remote office POTS even though all other PSTN is centralized.

```
REMOTE_RTR(config)# dial-peer voice 10 pots
REMOTE_RTR(config-dial-peer)# destination-pattern 911
REMOTE_RTR(config-dial-peer)# port 1/0/0
REMOTE_RTR(config-dial-peer)# no digit-strip
REMOTE_RTR(config-dial-peer)# exit
REMOTE_RTR(config)# dial-peer voice 11 pots
REMOTE_RTR(config-dial-peer)# destination-pattern 9911
REMOTE_RTR(config-dial-peer)# port 1/0/0
REMOTE_RTR(config-dial-peer)# forward-digits 3
The rightmost 3, i.e. 911, not 991 on dial-peer 11
REMOTE_RTR(config-dial-peer)# exit
```

To add a second PSTN port for 911 use, repeat all of the above, e.g. dial peers 12 & 13

## T R A N S L A T I O N P R O F I L E S

Translation Profiles—Change one dialed number to another, but only if it goes in (or out, based on how you apply it to the dial peer) a given port (num-exp is global to the whole router). Three steps:

- Define transformation rules
- Associate the rules into a translation profile
- Assign the translation profile to a dial peer

Format:

```
R(config)# voice translation-rule <rule_number>
R(cfg-translation-rule)# rule 1 /match/ /set/
R(cfg-translation-rule)# rule 2 /match/ /set/
R(cfg-translation-rule)# rule 3 /match/ /set/ ...
```

Example—translate DID (Direct Inward Dialing) 602-555-6xxx to internal extension range of 5xxx only if the call comes in from the PSTN T1 interface (don't affect calls to the remote office which uses the 6xxx range internally, the way num-exp would). The PSTN has already stripped all but the last 4 digits (6xxx).

- Create the Rule

```
R(config)# voice translation-rule 1
R(cfg-translation-rule)# rule 1 /6/ /5/
Rule #1 replace "6" with "5"
```

- Test the Rule

```
R# test voice translation-rule 1 6546
Tells the original number (6456) & translated number (5456)
```

- Assign Rule to a Translation Profile

```
R(config)# voice translation-profile CHANGE_DID
R(cfg-translation-profile)# translate called 1
"Called" can be { called | calling | redirect-called | redirect-target }
"Calling" would change the ANI (caller-ID)
1 is the tag for the translation rule
```

- Assign the Translation Profile—to the incoming direction only

```
R(config)# dial-peer voice 100 pots
R(cfg-dial-peer)# translation-profile incoming CHANGE_DID
```

## ORDER OF PRECEDENCE

1. num-exp
2. automatic digit-strip (POTS dial peers)
3. voice translation profiles
4. prefix digits
5. forward-digits

## CCP - CREATED DIAL - PLANS

In the folders in the left pane, Unified Communications --> Dial-Plans

- POTS
  - Create Dial Plans (tab)—3 wizards:
    - Create Incoming Dial-Plan—Inbound dial peers using PSTN trunks
    - Create Outbound Dial-Plan
      - Configure outbound dial peers using PSTN trunks
      - Specify access digits, e.g. 9
      - Specify caller ID information
    - Import Outgoing Dial-Plan Template—from a CSV file
  - Dial Peers (tab)—(no wizard) Configure dial peers through GUI pop-up window
- VOIP

## COR (CLASS OF RESTRICTION)

COR (Class Of Restriction)—Controls who can dial what.

- A phone can be associated with an incoming COR list, listing the “tags” it can access. *Incoming* is from the perspective of the phone system, not the phone.
- When the user dials, CME matches the dialed number to an outgoing dial-peer
- If that outgoing dial-peer requires a COR tag, CME checks for that tag in the dialing phone's incoming COR list & permits or denies the call accordingly

Rules—similar to access control lists

- If no outgoing COR list is applied, the call is always routed—simply don't apply any outgoing list to the 911 dial-peer
- If there is no incoming COR list applied, the call is always routed

COR List Tags—just meaningless tags for now, not yet in lists for assignment to phones (as capabilities) or dial peers (as requirements)

```
R(config)# dial-peer cor custom
R(config-dp-cor)# name 911
R(config-dp-cor)# name LOCAL
R(config-dp-cor)# name LD
```



COR Lists (to be applied outgoing)—Require the dialing phone to have these tags to complete a call (use a dial-peer which has this list applied in the outgoing direction)

```
R(config)# dial-peer cor list 911-CALL
R(config-dp-corlist)#member 911
```

```
R(config)#dial-peer cor list LOCAL-CALL
R(config-dp-corlist)#member LOCAL
```

```
R(config)#dial-peer cor list LD-CALL
R(config-dp-corlist)#member LD
```

COR Lists (to be applied incoming)—Assign these tag lists to the phone, granting it privileges

```
R(config)#dial-peer cor list 911-ONLY
R(config-dp-corlist)#member 911
```

```
R(config)#dial-peer cor list 911-LOCAL
R(config-dp-corlist)#member 911
R(config-dp-corlist)#member LOCAL
```

```
R(config)#dial-peer cor list 911-LOCAL-LD
R(config-dp-corlist)#member 911
R(config-dp-corlist)#member LOCAL
R(config-dp-corlist)#member LD
```

Outbound Assignments—presumably each of these dial-peers are already functional (have destination patterns & port values). This will add a restriction of which phones can use the peer.

```
Router(config)#dial-peer voice 10 pots
Router(config-dial-peer)#corlist outgoing 911-CALL
    Omitting this would allow anyone to dial 911 (an unprotected dial-peer)
```

```
Router(config)#dial-peer voice 11 pots
Router(config-dial-peer)#corlist outgoing LOCAL-CALL
```

```
Router(config)#dial-peer voice 12 pots
Router(config-dial-peer)#corlist outgoing LD-CALL
```

Inbound Assignments—Really, do this before outbound so callers have the ability to get through restrictions before the restrictions are created.

```
Router(config)#ephone-dn 1
Router(config-ephone-dn)#corlist incoming 911-ONLY
```

```
Router(config)#ephone-dn 2
Router(config-ephone-dn)#corlist incoming 911-LOCAL
```

```
Router(config)#ephone-dn 3
Router(config-ephone-dn)#corlist incoming 911-LOCAL-LD
    Omitting this corlist assignment would have allowed this ephone-dn (perhaps an executive) to
    make any call they want
```

In the previous example, the rules controlled what PSTN calls could be made from an internal IP phone. You could just as easily apply an incoming COR list to a PSTN dial-peer to restrict which internal extensions a PSTN caller can reach.

## C O R I N C C P

[Configure] UC → Dial Plans → Calling Restrictions (tab: Outgoing Call Types) [Create...]

*In the popup, create COR tags (meaningless for now)*

[Configure] UC → Dial Plans → Calling Restrictions (tab: Permissions) [Create...]

*Create an outgoing Cor List OR an incoming COR list—give it a name and move COR tags from list of possibilities on left to list on right.*

[Configure] UC → Dial Plans → POTS → (tab: Dial Peers) (select a dial peer) [Create... or Edit...]  
(General tab, Calling Restrictions section)

*Use drop-downs to pick an “Outgoing call restriction” and/or an “Incoming Call Restriction.”*

*The items in the drop downs are the created COR lists and “None.”*

[Configure] UC → User, Phones and Extensions → Extensions (choose an extension) [Edit...]

(Choose “Call Restrictions” in left pane of popup)

*Drop downs containing COR lists for “Incoming Permissions” and “Outgoing Permissions”*

*Note: the definitions of incoming and outgoing may be backward from the CLI and the chapter!*