| Category | S/W Release Version | Date |
|---|---|---|
| General | 6.200 | 13 Aug. 2003 |

# Title: Trunk Emulation Feature is added for E1/T1/Audio endpoints

Before APOS version 6.200, the trunk (tie-line) emulation feature is supported for FXS, FXO, and E&M endpoints. A trunk (tie-line) is a permanent point-to-point communication line between two voice ports. When set voice port to trunk by voice port command **connection trunk-initiate** or **connection trunk-answer**, the voice port deactivate POTS event such as hook on and hook off and make a permanent VoIP connection between trunk initiation endpoint and trunk answer endpoint that are being monitored for recovering the connection after recovering of power failure or network failure.

One of typical usage of the trunk emulation is conventional audio broadcasting system. Before introducing VoIP, the PSTN tie line is used for trunk line between audio broadcasting center and branches. With trunk emulation feature of VoIP, the PSTN tie line could be replaced to VoIP economically.

In case of Audio endpoint, the endpoint acts similar as other analog endpoint. One of the trunk endpoint could be connecting to microphone and the other endpoint could be connecting to speaker system.

In case of E1 or T1 endpoint, signal type of the controller should be DTMF mode and several channel groups for virtual voice port should be created for multi point-to-point VoIP connection per channel. Even though a channel group of more than one channel could be designated to trunk, typical usage is making a channel group of one channel for trunk connection to remote endpoint. The remote endpoint, i.e., the counter part of trunk connection of E1/T1 channel could be any kind of endpoints such as Audio, E1, T1, FXS, FXO, E&M endpoint.

## Example

There are four gateways – Gateway A, B, C, and D. Among them, Gateway A has one T1 port with E.164 number of 1xx , Gateway B has a FXS port of E.164 of 200, Gateway C has a FXS port of E.164 of 300, and Gateway D has AUDIO port of E.164 of 400.

A configuration example shows as bellows,

**Gateway A**

**At first, set signaling type to DTMF.**

```
ap2520(config)# controller t1 0/0
ap2520(config-controller-t1-0/0)# signalling ?
dtmf                DTMF signalling
isdn                ISDN PRI signalling (default)
r2                  R2 signalling
ap2520(config-controller-t1-0/0)# signalling dtmf
When you change the signalling type, system should be rebooted after save configuration.
ap2520(config-controller-t1-0/0)# end
ap2520# wr
Do you want to WRITE configuration ? [y|n] y
Writing configuration....done
ap2520# reb
System Reboot...
```

**Create channel groups each of them has one channel. When creating channel group virtual voice port is created automatically.**

```
ap2520(config)# controller t1 0/0
ap2520(config-controller-t1-0/0)# channel-g t 1 1
ap2520(config-controller-t1-0/0)# channel-g t 2 2
ap2520(config-controller-t1-0/0)# channel-g t 3 3
ap2520(config-controller-t1-0/0)# show voice p s
```

| Port | LineType | Status | InGain | OutGain | TieType | TieDigits | CallNum | Tcalled | Tcalling |
|---|---|---|---|---|---|---|---|---|---|
| 0/ 0: 0 | T1 | Link Up | 0 | 0 | none | | -1 | -1 | -1 |
| 0/ 0: 1 | T1 | Link Up | 0 | 0 | none | | -1 | -1 | -1 |
| 0/ 0: 2 | T1 | Link Up | 0 | 0 | none | | -1 | -1 | -1 |
| 0/ 0: 3 | T1 | Link Up | 0 | 0 | none | | -1 | -1 | -1 |

**Create dial peers for each created virtual voice port.**

```
ap2520(config)# dial-p v 1 pots
ap2520(config-dialpeer-pots-1)# dest 101
ap2520(config-dialpeer-pots-1)# port 0/0 1
ap2520(config-dialpeer-pots-1)# dial-p v 2 p
ap2520(config-dialpeer-pots-2)# dest 102
ap2520(config-dialpeer-pots-2)# port 0/0 2
ap2520(config-dialpeer-pots-2)# dial-p v 3 p
ap2520(config-dialpeer-pots-3)# dest 103
ap2520(config-dialpeer-pots-3)# port 0/0 3
ap2520(config-dialpeer-pots-3)# show dial-p v s
```

| Tag | DestPattern | Prefix | Admin | Port | Regist | Inbound | Pref | Hstop | Tcalled | Tcalling |
|---|---|---|---|---|---|---|---|---|---|---|
| 1 | 101 | | up | 0/ 0: 1 | yes | -1 | 0 | no | -1 | -1 |
| 2 | 102 | | up | 0/ 0: 2 | yes | -1 | 0 | no | -1 | -1 |
| 3 | 103 | | up | 0/ 0: 3 | yes | -1 | 0 | no | -1 | -1 |

**Set trunk for each created virtual voice port. You should set trunk answer for remote side before set trunk-init at this endpoint.**

```
ap2520(config-dialpeer-pots-3)# voice-p 0/0 1
ap2520(config-voice-port-0/0:1)# conn trunk-init 200
ap2520(config-voice-port-0/0:1)# voice-p 0/0 2
ap2520(config-voice-port-0/0:2)# conn trunk-init 300
ap2520(config-voice-port-0/0:2)# voice-p 0/0 3
ap2520(config-voice-port-0/0:3)# conn trunk-init 400
ap2520(config-voice-port-0/0:3)# show voice p s
```

| Port | LineType | Status | InGain | OutGain | TieType | TieDigits | CallNum | Tcalled | Tcalling |
|---|---|---|---|---|---|---|---|---|---|
| 0/ 0: 0 | T1 | Link Up | 0 | 0 | none | | -1 | -1 | -1 |
| 0/ 0: 1 | T1 | Link Up | 0 | 0 | trunk-initiate | 200 | -1 | -1 | -1 |
| 0/ 0: 2 | T1 | Link Up | 0 | 0 | trunk-initiate | 300 | -1 | -1 | -1 |
| 0/ 0: 3 | T1 | Link Up | 0 | 0 | trunk-initiate | 400 | -1 | -1 | -1 |

```
ap2520(config)# show voice p 0/0
```

```
Voice port slot(0)/port(0) group(0)
    line type = T1
    input gain = 0 db
    output gain = 0 db
    tie connection = none
    description =
    T1 wait wink = 550 msec
    T1 wink duration = 200 msec
    T1 wink wait = 200 msec
    translate incoming called-number = -1
    translate incoming calling-number = -1
    comfort noise generation = enabled
    dial tone generation = disabled
    echo cancellation = enabled
    announcement = enabled
    low dtmf gain = -8
    high dtmf gain = -5
    busyout action = none
    compand type = a-law

      ChannelNum   Status          AssocCall Direction
     ==================================================
Voice port slot(0)/port(0) group(1)
    line type = T1
    input gain = 0 db
    output gain = 0 db
    tie connection = trunk-initiate 200
    description =
    T1 wait wink = 550 msec
    T1 wink duration = 200 msec
    T1 wink wait = 200 msec
    translate incoming called-number = -1
    translate incoming calling-number = -1
    comfort noise generation = enabled
    dial tone generation = disabled
    echo cancellation = enabled
```

```
    announcement = enabled
    low dtmf gain = -8
    high dtmf gain = -5
    busyout action = none
    compand type = a-law

      ChannelNum   Status            AssocCall Direction
     ==================================================
      1            Idle              -1        Both
Voice port slot(0)/port(0) group(2)
    line type = T1
    input gain = 0 db
    output gain = 0 db
    tie connection = trunk-initiate 300
    description =
    T1 wait wink = 550 msec
    T1 wink duration = 200 msec
    T1 wink wait = 200 msec
    translate incoming called-number = -1
    translate incoming calling-number = -1
    comfort noise generation = enabled
    dial tone generation = disabled
    echo cancellation = enabled
    announcement = enabled
    low dtmf gain = -8
    high dtmf gain = -5
    busyout action = none
    compand type = a-law

      ChannelNum   Status            AssocCall Direction
     ==================================================
      2            Idle              -1        Both
Voice port slot(0)/port(0) group(3)
    line type = T1
    input gain = 0 db
    output gain = 0 db
```

```
    tie connection = trunk-initiate 400
    description =
    T1 wait wink = 550 msec
    T1 wink duration = 200 msec
    T1 wink wait = 200 msec
    translate incoming called-number = -1
    translate incoming calling-number = -1
    comfort noise generation = enabled
    dial tone generation = disabled
    echo cancellation = enabled
    announcement = enabled
    low dtmf gain = -8
    high dtmf gain = -5
    busyout action = none
    compand type = a-law

      ChannelNum   Status          AssocCall Direction
     ==================================================
      3            Idle            -1        Both

ap2520(config)# show controller 0/0
Controller T1 slot(0)/port(0)
    T1 Link is DOWN
        Alarm detected.
        Applique type is Channelized T1.
        Framing is SF, Line Code is AMI, Cable Length is Short 110.
        Signalling type is DTMF.
        3855 Line Code Violations, 0 Framing Bit Errors
        0 Out Of Frame Errors, 0 Bit Errors
    signalling type = dtmf
    clock source = master
    channel group 0 =
    channel group 1 = 1
    channel group 2 = 2
    channel group 3 = 3
                                   1       2       3
```

```
    allocated timeslots = YYYNNNNNNNNNNNNNNNNNNNNNNNNNNNNN
    outgoing barred channel group =
    channel order = descending
    b-channel negotiation = exclusive
    overlap receiving = enabled
    protocol side = network
    R2 get calling number = disabled
    ISDN virtual connect = disabled
    T1 cable length = short 110
    T1 framing = sf
    T1 line code = ami
    T1 CAS type = immediate
```

**Gateway B (Gateway C and D are similar to Gateway B)**

**At first, create POTS peer of E.164 as 200.**

```
GatewayB(config)# dial-p v 200 pots
GatewayB(config-dialpeer-pots-200)# dest 200
GatewayB(config-dialpeer-pots-200)# port 1/0
```

**Set voice port as trunk answer and waits call from trunk initiator.**

```
GatewayB(config-dialpeer-pots-200)# voice-p 1/0
GatewayB(config-voice-port-1/0)# conn trunk-answer 101
GatewayB(config-voice-port-1/0)# show voice p 1/0
Voice port slot(1)/port(0)
    line type = FXS
    status = Busy
    input gain = 0 db
    output gain = 0 db
    ring frequency = 25 Hz
    ring cadence = 1000 msec on, 2000 msec off
    polarity inverse = disabled
```

```
tie connection = trunk-answer 101
description =
translate incoming called-number = -1
translate incoming calling-number = -1
comfort noise generation = enabled
dial tone generation = enabled
echo cancellation = enabled
announcement = enabled
low dtmf gain = -8
high dtmf gain = -5
caller ID = disabled
caller ID type = bellcore
caller ID NAME = enabled
busyout action = none
associated call number = 1769
```

```
GatewayB(config-voice-port-1/0)# show voice p s

Port      LineType  Status     InGain OutGain  TieType       TieDigits     CallNum  Tcalled Tcalling
-----------------------------------------------------------------------------------------------
1/ 0      FXS       Busy       0      0        trunk-answer  101           1769     -1      -1
1/ 1      FXS       Idle       0      0        none                        -1       -1      -1
1/ 2      FXO       Idle       0      0        none                        -1       -1      -1
1/ 3      FXO       Idle       0      0        none                        -1       -1      -1
```