APOS Release Note

Revision   : 5.80
Date   : 2002/10/28
Target   : VoIP Gateway

Added Function

1. **Call pick up / call transfer**

    Call pickup/transfer is available between local ports.
    - Related commands
        router(config)# dial-peer call-pickup <0-9|#|*>
        router(config)# dial-p call-transfer <h|n>
    - Default value
        Disabled

2. **Voice confirmed connect**

    When an outbound call to FXO port is received from network, the Q.931 CONNECT message is sent to the caller after detecting voice on the FXO port.
    - Related commands
        router(config-vservice-voip)# voice-confirmed-connect <cr>
    - default value
        Disable

3. **Inband ringback tone**

    When the called party side gateway receives a call, this gateway sends inband ring-back tone to the source gateway(calling party side gateway) with RTP. So, the calling party side gateway doesn't need to generate local or pseudo ringback tone.

    - related command
        router(config-vservice-voip)# inband-ringback-tone

- default
    Disable

4. Easy setup

This function provides an easy and simple configuration mode for a user who is not familiar with AddPac's command line interface.

- related command
    router(config)# ez-setup

5. Auto-upgrade

This function supports to upgrade OS image and configuration file automatically through Web technology.

- related command
    router(config)# auto-upgrade ?
        action : at Auto-Upgrade (input URL and a upgrade message will be displayed)
        authentication : Web server(auto upgrade server) Login Name / Password
        auto-Reboot : after OS image upgrading, decide to reboot
        interval : retrial time setting for fail or success case
        proxy : input a proxy server IP Address
        url : input Web server (auto upgrade server) URL
        verbose : monitor display mode when upgrade
- default
    Disable

6. E1 ISDN parameter n303 (E1 PRI only) is added

A new parameter n303 which means Q.931 setup retry counter on a fail is added.
- related command
    router(config-controller-e1-1/0)# isdn n303

7. R2 DTMF (E1 R2 only) signaling type is added

This new signaling mode supports CAS with R2 and digit tone with DTMF. This is one of variant of R2 signaling.

- related command

router(config-controller-e1-1/0)# signaling-type dtmf

8. out-barred-channel command　(E1 PRI/R2 only) is added

The channel set as out-barred-channel in time slots does not permit out-bound traffic in E1 channel

- related command

router(config-controller-e1-1/0)# out-barred-channel timeslot xxxx

9. Translation hook flash

If a VoIP gateway sends a hook flash to the PABX of remote side gateway, a specific pattern (ex: #,*...) is needed. Because a normal hook flash is detected by local PABX. The gateway which receives the pattern from physical port translates this specific pattern(ex: #,*...) to a hook flash and transfers it to remote side gateway by a message to perform hook flash to the remote side PABX.

Example
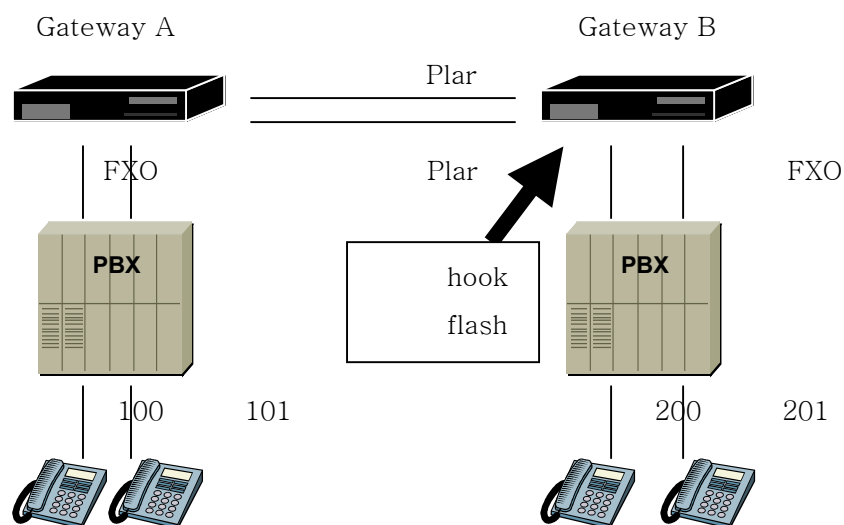
Call Scenario

Assuming that user 200 is trying to transfer the call to user 101 after user 100 call user 200. (hook flash relay)

- User 200 hook off(user 200 wants to transfer this call to user 101) → digit "#100 (assuming that "#" is set as "hook flash relay key") → Ringing "101" → user 101 hook off → user 100 and user 101 connected

Assuming that user 200 is trying to transfer this call to user 201 after user 100 call user 200. (normal case)

- User 200 hook off (user 200 wants to transfer this call to 201) → digit hook flash + "201 (in this case, it is no concern of gateway, in other words, this function is PABX feature.) → ring "201" → user 201 hook off → user 100 and user 201 connected.

- related command

router (config)# dial-peer translate-digit <0-9|#|*> <0-9#*!>

% send parameter ! means hook flash.

10. A new DTMF relay method by RTP and by H.245 signal is added

Two dtmf relay modes are added to supporting not only digit tone itself but also digit tone duration.

- dtmf-relay rtp

The dtmf tone is delivered by RTP which has encoded to the tone information. So, during detecting the dtmf, the sending gateway sends RTP packets of the tone continuously. This is actually a proprietary implementation of RFC 2833.

Caution!: This function is not compatible with other vendor.

- dtmf-relay h245-signal

The dtmf tone is delivered by H.245 signal. This method also can deliver dtmf duration as 'dtmf-relay rtp' but has less real-time characteristic than RTP. When the user push digit button long, if gateway send via h245-alphanumeric, receiving side gateway generates short tone once. But if dtmf h245-signal is used, gateway generates same tone duration (on/off time) as sending side dtmf tone.

The function is defined as the standard, so it has the compatibility.

- No dtmf-relay

  This is same as before.

- h245-alphanumeric

  This is same as before.

- related command

  router (config-dialpeer-voip-1000)#

  dtmf-relay < rtp| h245-alphanumeric| h245-signal>

11. A configurable ring cadence and detection function related to voice broadcast machine is added.

If main broadcasting equipment and broadcasting terminal react to special ring signal and VoIP gateway systems are located between them, the special ring cadence detected at FXO port of broadcasting equipment should be delivered to the FXS port of broadcasting terminal.

To solve above problem, ring detect timeout on FXO port and ring cadence generation of FXS port should be configurable by below commands.

- **Ring cadence (FXS)**

  Set up the duration(on time/off time) of ring signal that sends to FXS port which is connected with terminal. (to negate to default : no ring cadence)

- **ring detect-timeout(FXO)**

  When a call is ringing at FXO port, the call will be terminated, if next ring signal is not detected within the period of specific(set) time.

  So it is recommended to set it be longer than expected physical ring time.

  To determine hook off time on FXO port, ring number parameter is useful but the parameter will be ignored when PLAR mode is set on the FXO port.

- related command

  FXO port

  router (config-voice-ports-1/0)# ring detect-timeout <20-100>
  1=100ms

  router (config-voice-ports-1/0)# ring number <0-255>

  FXS port

  router (config-voice-ports-0/0)# ring cadence <5 – 50>

12. Support caller-id in FXS/FXO

The Bellcore type 1 (i.e., domestic (Korea) standard) caller id method is added for generation of called-id on FXS port and detecting on FXO port.

Some voice module models do not support caller-id detecting on FXO port.
- related command
  router (config-voice-ports-1/0)# caller-id enable
- default : disabled

13. Timeout parameter "tohd"(on hook delay timeout) is added.

After detecting hook on a FXS port, the call should not be terminated until this timer expires. (default=300ms).

14. Redundant RTP for packet loss network

This function is useful to a network that has a lot of packet loss. The RTP payload is copied to overcome packet loss so needs more bandwidth.
- related command
  router (config-dialpeer-voip-1000)# redundant-rtp
- default = disabled

Caution!: This function is not compatible with other vendor.

15. A configurable parameter of number of frames per packet is added for G.711

- related command
  router (config-vservice-voip)# max-frame　g711
- default = 2

16. E1/T1 channel random selection is added

- related command
  router(config-controller-e1-1/0)#chan-number-order
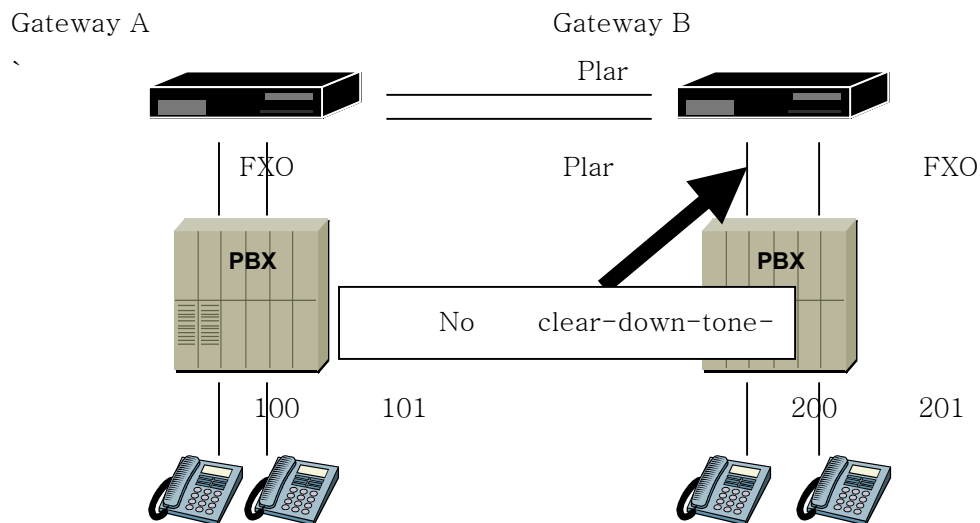 　　　　　　　　　　　<ascending|descending|**random**>
- default = descending

17. A call trace function for a specific called/calling part number is added

- Related command

router # debug voip **number** <0-9|#|*>

18. Disable of clear-down-tone-detect on FXO port is added

Example

Gateway A   Gateway B
`   Plar

FXO   Plar   FXO

PBX   **PBX**

No   clear-down-tone-

100   101   200   201

Call Scenario

Assuming that a call exists between user 100 and user 200 and at same time user 201 is busy on another call. Now, user 200 is trying to transfer call to user 201 with clear-down-tone detection enabled.

→ digit hook flash + "201 (In this time, If 201 user is busy, the gateway may receive (clear-down-tone) from PABX instead of (busy-tone) → gateway B terminate the call on detecting clear down tone.

If clear-down-tone is disabled on user 200 FXO port.

→ digit hook flash + "201 (In this time, If 201 user is busy, the gateway may receive clear-down-tone from PABX **but gateway do nothing.** → 200 user hook flash again → Inform 100 of being busy → call End

Caution! : Be careful not to get hang on by un-detecting clear down tone at both side.

- Related command

  It is available to set FXO port only. (default enabled)

  router (config-voice-ports-1/0)# no clear-down-tone-detect

19. Add a command about H.245 procedure option.

This parameter is used when H.245 procedure to be started after CONNECT.

- Related command

  router(config-vservice-voip)# h323 call channel latest

- default

  h323 call channel early

20. Fax configuration on each VoIP peer.

Now, each VoIP peer can has his own fax configuration (fax protocol / rate) as well as system wide configuration on 'voice service VoIP' fax configuration mode.

- Related command

  router(config-dialpeer-voip-1000)# fax ?

21. Permit signaling-port (default TCP 1720) changing.

- Related Command

  (config-gateway)# signaling-port xxxx

22. It is available to set max hop parameter at "call forwarding busy" in "voip peer". (For loop condition protection)

- Related Command

  #dial-peer voice 1000 voip > max-forward-hop xx   (default 4)

23. Second inter-digit timeout (tidt) parameter is added.

Now, inter digit timeout can be assigned by two stage to solve fast call processing with enough inter digit timeout.

Normally first inter-digit timeout is long enough to endure human's late pushing of digits and second timeout is short to decrease call processing time.

- Related Command

router(config-vservice-voip)# timeout   tidt <1-600> <2-32> <1-600>

parameter explanation

The first parameter <1 - 600> : The same as former version.

The 2nd parameter <2 - 32> : If it is set, the 3rd parameter must be set.
This parameter means if it is applied up to what number of digit.

The 3rd parameter <1 - 600> : Tidt value, it will be applied from 2nd
parameter.

Changed Function

1. Leased line emulation to remote branch

In some cases, VoIP is introducing to replace existing leased line to remote branch that is connected to main PBX via leased line and has a number of the PBX. In this case, as described as below, Gateway A and B set to both direction PLAR to emulate leased line. And also branch office should be able to activate/deactivate main PBX's supplementary features (i.e., call pickup / call transfer / call forwarding …).

Before： When FXO set to PLAR, call is accepted by hook off first, and after that, the call is progressed to remote side. This feature make call pickup impossible when ringing remote branch.

After： When FXO set to PLAR, call is progressing without hook off. This makes call pickup possible.

Example

Gateway A   Gateway B

Both direction PLAR

FXO   Both direction PLAR   FXS

200   201

Main   **PBX**   Branch office

200   201

100   101
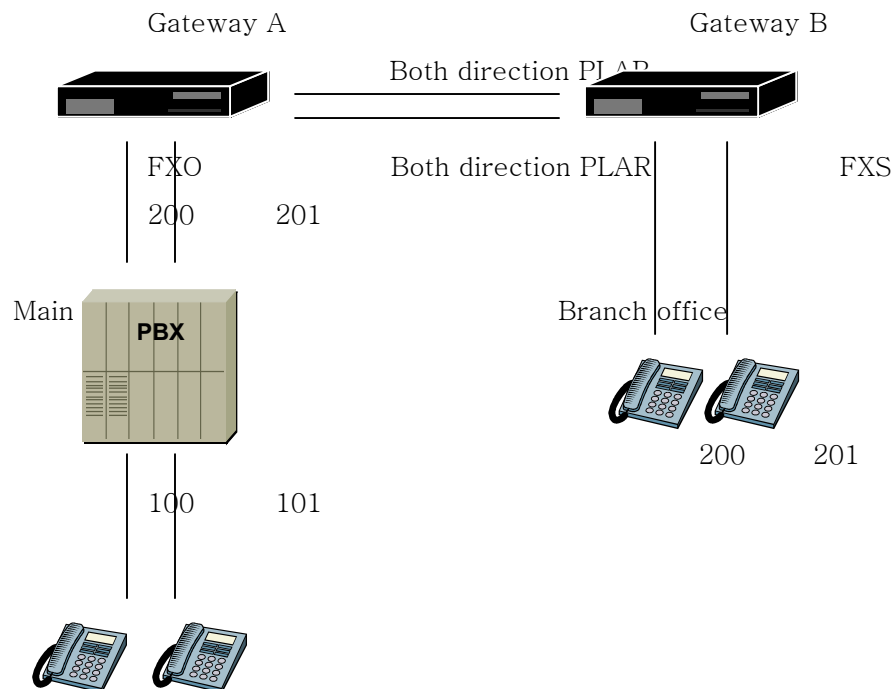
Call Scenario

- User 100 is making a call to user 200.

- When user 200 is ringing, user 201 picks up the call by digit main PBX's call pick up access code.

- When 201 user transfer the call to 200 user, Also (usually after hook-flash, digit 200), gateway relays hook flash and digits to PBX.

- Related command

2. Exception routine of failure on H.245 procedure

   Before : Terminate the call.

   After : Remain the call.

   - Related command

       None

3. RAS source port is fixed (UDP port 22000) by default

   If user wants to set with pool method, follow the bellow command.

   - Related Command

      (config-gateway)# (no) fixed-ras-port

4. Do not send URQ message to GK when received RRJ

   Before : sent URQ

   After: do not send URQ

   - Related Command

5. Add "minimize-voip-ports" command for assignment by round-robin and redundancy of port (for increasing port re-use interval.)

   Before : TCP/UDP port can be assigned with large pool or a UDP/TCP port per physical port.

   After : : TCP/UDP port can be assigned with large pool or a small pool with

redundancy.
- Related Command
router(config-vservice-voip)# minimize-voip-ports ?

6. Play busy-tone when destination port(FXS) is busy and announcement function is disabled

7. Changed default (GRQ)
Before : enabled to send GRQ before sending RRQ
After : disable to send GRQ by default
- Related Command
gateway > discovery

8. Configurable h323 call response message
When call is inbounding from network, response message could be Q.931 CALL PROCEEDING, PROGRESS, ALERTING, CONNECT messages.
Now, by default, PROGRESS message is delivered to calling party when called party's physical port is FXO, E&M, E1 and T1 and ALERTING message is delivered when called party's physical port is FXS.

Before : ALERTING message for all kinds of physical ports.
After : depends on physical port type

- Related Command
router(config-vservice-voip)# h323 call response default <cr>

9. Same RTP (UDP) port number between voice and T38 fax channel
Before : use another UDP port number when T.38 logical channel re-initiated.
After : use same UDP port number.

10. Inbound voip peer selection algorithm

When call is outbounding to network, outbound voip peer is selected by simply by destination-address field of voip peer, but in inbounding case, voip peer selection is more complicated than outbounding. But selection of a proper voip peer is important to apply configuration (codec, dtmf relay, fax , …) on the call.

Before : matching by order of IP address → answer-address → calling-number.

After : matching by order of answer-address → calling-number → IP address.

% If some gateway is set as bellows, the former and v5.80 will show deferent result.

Example

      dial-peer voice 1000 voip

        dest 20T

        sess tar ras

      dial-peer voice 1001 voip

        dest 200

        sess tar ras

      dial-peer voice 1002 voip

        sess tar 10.1.1.1

      dial-peer voice 1003 voip

        answer-address 20T

      dial-peer voice 1004 voip

        answer-address 200

When inbound call's calling number is 200 and calling gateway's IP address is 10.1.1.1 in the above example, the selected order is **dial-peer 1004 → 1003 → 1001 → 1000 → 1002**. So, voip peer 1004 will be selected.

At the same situation, at the former version, the selected order is **dial-peer 1002 → 1004 → 1003 → 1001 → 1000 .** So, voip peer 1002 was selected.

Fixed Bug

1. E&M signaling on a certain condition