AddPac VoIP Gateway Series

Release Note

V6.052

AddPac Technology Co., Ltd.

Phone (82 2)568-3848
Fax (82 2)568-3847
E-mail : info@addpac.com
http://www.addpac.com

[INDEX]

1. Added Function

1.1. IP sharing between a gateway and a PC with same public IP

Before providing this feature, IP sharing by NAT/PAT was the only IP sharing technology of our gateway. Now, Addpac's gateway series also provides public IP sharing between a gateway and a PC(or other equipment) that is connecting to LAN side of gateway. This feature compensates a restriction of IP sharing by NAT/PAT and the best solution for user who has only one PC or already has IP sharer.

- Related command
- router(config)# ip-share ?

  enable : ip-share function activation

  interface ?

  net-side : VoIP link for interface set

  local-side : Private link for interface set

  config ?

  fin-timeout : Set Timeout value after TCP FIN

  group-static-entry : Add group IP Share static entry

  icmp-timeout : Set Timeout value for ICMP entry

  static-entry : Add single IP Share static entry

  syn-timeout : Set Timeout value after TCP SYN

  tcp-timeout : Set Timeout value for TCP entry

  udp-timeout : Set Timeout value for UDP entry

Example

- Cable Modem(Allocated Public IP using DHCP)

  router(config)# dhcp-list 0 type server

  router(config)# dhcp-list 0 address server interface ether0.0

  router(config)# dhcp-list 0 option dhcp-lease-time 600

  router(config-ether0.0)# ip address dhcp

  router(config-ether1.0)# no ip address

  router(config-ether1.0)# ip dhcp-group 0

  router(config)# ip-share interface net-side ether0.0

  router(config)# ip-share interface local-side ether1.0

  router(config)# ip-share enable

- ADSL MyIP(Static ip used)

router(config-ether0.0)# ip add 123.46.44.3 255.255.255.0

router(config-ether1.0)# no ip address

router(config)# route 0.0.0.0 0.0.0.0 123.46.44.1

router(config)# ip-share interface net-side ether0.0

router(config)# ip-share interface local-side ether1.0

router(config)# ip-share enable

- ADSL(Allocated Public IP PPPoE)

router(config)# dhcp-list 0 type server

router(config)# dhcp-list 0 address server interface ether0.0

router(config)# dhcp-list 0 option dhcp-lease-time 600

router(config-ether0.0)# no ip address

router(config-ether0.0)# encapsulation pppoe

router(config-ether0.0)# ppp authentication pap callin

router(config-ether0.0)# ppp pap sent-username <string> password <string>

router(config-ether1.0)# no ip address

router(config-ether1.0)# ip dhcp-group 0

router(config)# ip-share interface net-side ether0.0

router(config)# ip-share interface local-side ether1.0

router(config)# ip-share enable

- default

Only AP200, AP1100 enable

1.2. Flexible IP ToS setting for providing voice traffic QoS

The IP ToS(Type Of Service) field could be used for prioritizing voice traffic at WAN network if the Internet provider has a capability to control the IP ToS field. Now, Addpac's VoIP gateway series provides flexible setting on the IP ToS and also setting of appliance for RTP or VoIP signaling or both.

- Related command

router(config)# ip-tos ?

default : General Data Packet precedence setting

delay : Request Low Delay

throughput : Request High Throughput

reliability : Request High Reliability

precedence : Specify Datagram Precedence

value : Specify the value directly

forward : Transmit to VoIP about packet that enter from local by recedence setting

bypass : Transmit to VoIP about packet DS field that enter from local clear : Transmit to VoIP about packet DS field that enter from local after 0x00 setting

set : Transmit to VoIP about packet DS field that enter from local by user value setting

rtp : UDP/RTP packet about precedence setting

sig : H.323/SIP signaling packet about precedence setting

- default : disable

1.3. Changeable service port

When applying the public IP sharing feature, in order to solve port confliction between the gateway and the PC, gateway's well-known service port can be changed by this command.

- Related command

router(config)# service-port ?

ftpd <port>: FTP server port change

httpd <port> : HTTP server port change

snmpd <port> : SNMP agent port change

telnetd <port> : Telnet Server port change

tftpd <port> : TFTP server port change

- default

ftpd : 21, httpd : 80, snmpd : 161, telnetd : 23, tftpd : 69

1.4. Setting of VLAN priority for voice traffic QoS

VLAN(IEEE802.P) priority field tagging function.

- Related command

router(config)# vlan-pri ?

default <0-7>: General data packet VLAN priority setting

rtp <0-7>: UDP/RTP data packet VLAN priority setting

sig <0-7>: H.323/SIP signaling packet VLAN priority setting

- default : disable

1.5. encapsulation pptp added

Added encapsulation type pptp(point to point tunneling protocol) on the LAN interface.

- Related command

router(config-ether0.0)# encapsulation pptp

router(config-ether0.0)# pptp ?

ip local : local ip address setting

ip remote : remote ip address setting

mode ppp : Higher level protocol is ppp

mode dhcp : Higher level protocol is dhcp

- default : disable

1.6. ipcp default-route

Using this command, It is available to get default-router information from ADSL RAS.

- Related command
  router(config-ether0.0)# ppp ipcp default-route <cr>
- default : disable

1.7. ipcp ms-dns

Using this command, It is available to get dns-server information from ADSL RAS.

- Related command
  router(config-ether0.0)# ppp ipcp ms-dns <cr>
- default : disable

1.8. ETSI(European Telecommunications Standards Institute) type Caller-ID added

Three type of ETSI Caller-ID are supported now.

- Related command

router(config-voice-ports-0/0)# caller-id ?

enable : Caller-ID function enable

type : Caller-ID type change

bellcore (default)

etsi : ETSI type 1

etsi-dtmf : ETSI type 1 DTMF

etsi-dtmf-prior-ring : ETSI type1 DTMF prior ring

- default : disable(default caller-id type is **bellcore**)

1.9. Change Tone Frequency

The tone generation in a system can be changed to adjust for each country. Assigned tone information can be checked by command **show tone**.

- Related command

router(config)# voice class ?
dial-tone : dial-tone setting change
ring-back-tone : ring-back-tone setting change
line-busy-tone : line-busy-tone setting change
reorder-tone : reorder-tone setting change
line-lock-tone : line-lock-tone setting change

- subcommand
    1. (1-3980) low frequency(Hz)
    2. (0) single tone / (1-3980) high frequency(Hz)
    3. (0-10000) on time(msec)
    4. (0-10000) off time(msec)
    5. (cr) / (0-10000) second on time(msec)
    6. (cr) / (0-10000) second off time(msec)
    7. (cr) / (-31 – 3) level(db)
- default

Tag	Low(Hz)	High(Hz)	On1(ms)	Off1(ms)	On2(ms)	Off2(ms)	dBm	Description
-	350	440	10000	0	0	0	-12	Dial tone
-	440	480	1000	2000	0	0	-12	RingBack tone
-	480	620	500	500	0	0	-12	LineBusy tone
-	480	620	300	200	0	0	-12	Reorder tone
-	1400	2060	100	100	0	0	0	LineLock tone

1.10. TCP/UDP port for VoIP can be configurable

Now all VoIP related ports could be configurable to meet multi-gateway operation in a private network. In case of installation of more than one gateway in a IP shared private network, it the IP sharer is operating at PAT mode, gateway's port assignment should not be conflicted.

You can check port assignment by end of **show gateway**.

- Related command

router(config-vservice-voip)# minimize-voip-ports service ?
signal-tcp-src : H.225 signalling source port limit setting
control-tcp-src : H.245 control source port limit setting
control-tcp-listen : H.245 contro listen port limit setting
rtp-udp-listen : RTP/RTCP port limit setting

- default : disable

1.11. ARQ option of attachment h323id's domain is added

When sending ARQ, if the h323id has a form of name with domain like acct@addpac.com and this option enabled, source address could be 5551234@addpac.com when calling party's E.164 is 5551234.

- Related command

router(config-gateway)# arq attaché-domain <cr>

- default : disable

1.12. LRQ option is added

Most of case, LRQ(Location Request) is not necessary in gateway, but for some cases, LRQ could be sent with or without ARQ following this option.

- Related command

router(config-gateway)# lrq ?

no-arp : Only LRQ transmit

with-arq : First transmit LRQ, transmit ARQ sent by fail

domain-with-arp : Transmit H.323 domain with LRQ

- default : disable

1.13. Display option is added

H.323-ID is used for Q.931 display info field by default. If user want to replace it with e164 ,OR remove display field from Q931 message, this function is used.

- Related command

router(config-vservice-voip)# display send ?

h323id : h323-id to display field setting

e164 : E.164 to display field setting

none : display field to not setting

- default : h323-id

1.14. Codec variant can be supported.

The most popular codec using at VoIP is G.723.1 and G.729 that has variant on silence suppression and/or codec complexity.

The G.723.1 has codec variant on silence suppression (i.e., SID packet sending or not) and the G.729 has variants on silence suppression and complexity. Using below commands, more precise codec negotiation could be possible.

- Related command

router(dialpeer-1000-voice)# codec-variant ?

g7231 ?

standard : G7231 standard type, no SID

Annex-a : G7231 annex A type, on SID

G729 ?

Standard : G729 standard type, no SID, High Complexity

Annex-a : G729 annex A type, no SID, low Complexity

Annex-b : G729 annex B type, on SID, High Complexity

Annex-ab : G729 annex AB type, on SID, low Complexity

- default

G7231 : annex A type

G729 : G729 standard type

1.15. Added new DTMF relay method by RTP Or H245 signal.

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

dtmf-relay rtp-2833

The dtmf tone is delivered by RTP which has encoded to the tone information. So, the dtmf is transferred via RTP channel then tone is generated by remote gateway itself. Because the remote gateway plays not only tone but also duration, user may feel just like inband tone. This has compatibility with other vendor which support RFC-2833

The function is defined as the RFC-2833 standard, so it has the compatibility.

dtmf-relay rtp

From an operation point of view, it is the same as dtmf-relay rtp-2833, but 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.

Even though the user push digit button long duration, when gateway send via h245-alphanumeric, receiving side gateway generates short tone once. But using the dtmf h245-signal, gateway generates same tone duration (on/off time) as sending side dtmf tone. So user may feel just like inband tone.

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

No dtmf-relay

This is the same as before.

h245-alphanumeric

This is the same as before.

- Related command

  router (config-dialpeer-voip-1000)#

  dtmf-relay < rtp| rtp-2833 | h245-alphanumeric| h245-signal>

1.16. All-up-ace[1](ez-setup for GUI) program added

Please refer to All-up-ace(Ez-setup program) description document on AddPac Home Page.

[1] All-up-ace is KT's service trade mark

2. Changed Feature

2.1. PSTN Backup port model (AP200, AP1000) – Set busy out monitor as default

(Before) All models are disabled the busyout monitor as default including PSTN backup port models(AP200, AP1000).

(After) In case of PSTN backup port equipped model, when PSTN backup port is enabled by busyout monitoring of VoIP WAN network interface is downed or gatekeeper is downed. So, busyout monitor option is enabled by default for convenience.

2.2. fixed-ras-port parameter

Before : It was just only available to set if it fix RAS source port(22000) or not.

After : Added Parameter which set specific UDP port number for RAS source port

- Related command

router (config-gateway)# fixed-ras-port <<0-65536>|<cr> >

3. Fixed Bug

3.1. Problem : Lost of configuration.

When set second inter-digit timeout parameter of 'timeout tidt' – inter digit timeout – with default inter-digit timeout value, can not be saved the configuration.

→ Fixed.

3.2. Problem : Can't free back when busy out condition has been released.

When busyout monitoring of gatekeeper and voip-interface is set and no gatekeeper registration information is set, even downed voip-interface is restored the gateway has a status of busyout status.

→ Fixed.

3.3. Problem : Web server's abnormal operation on too long URI

When gateway's http server receives a message with too long URI, sometimes gateway works abnormal.

→ Fixed.