

VoIP Gateway Series

SIP Protocol Debugging Service Overview



AddPac

AddPac Technology

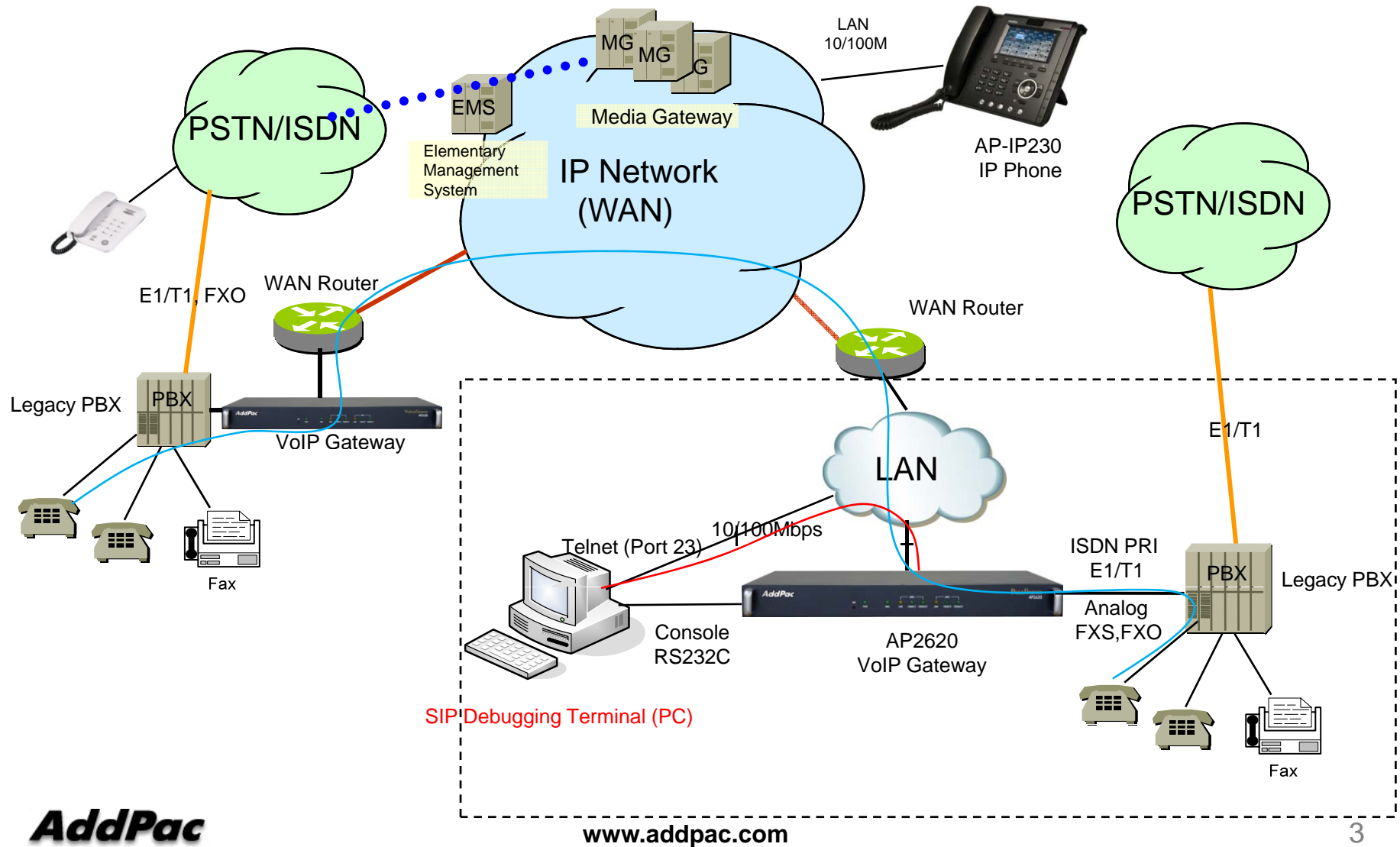
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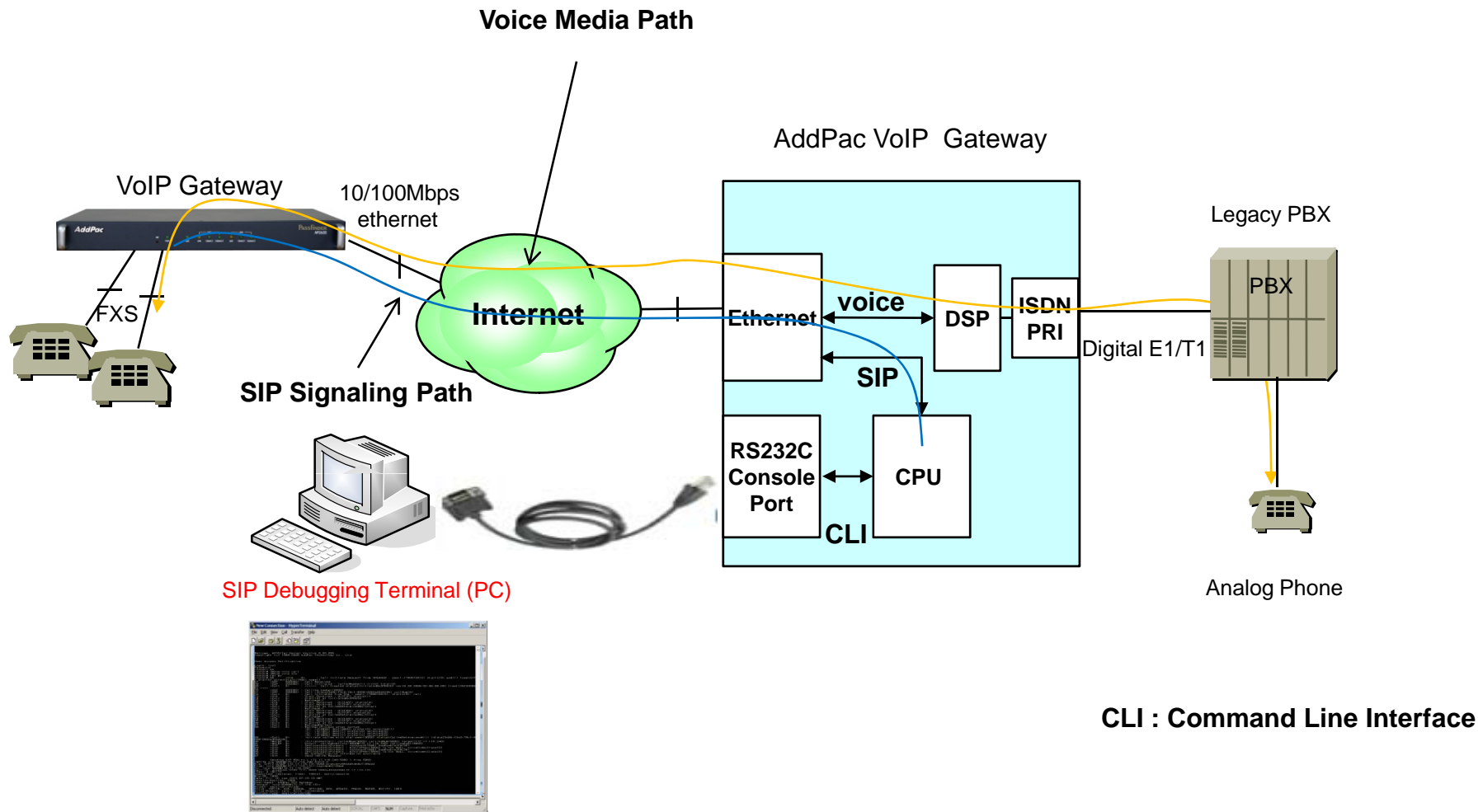
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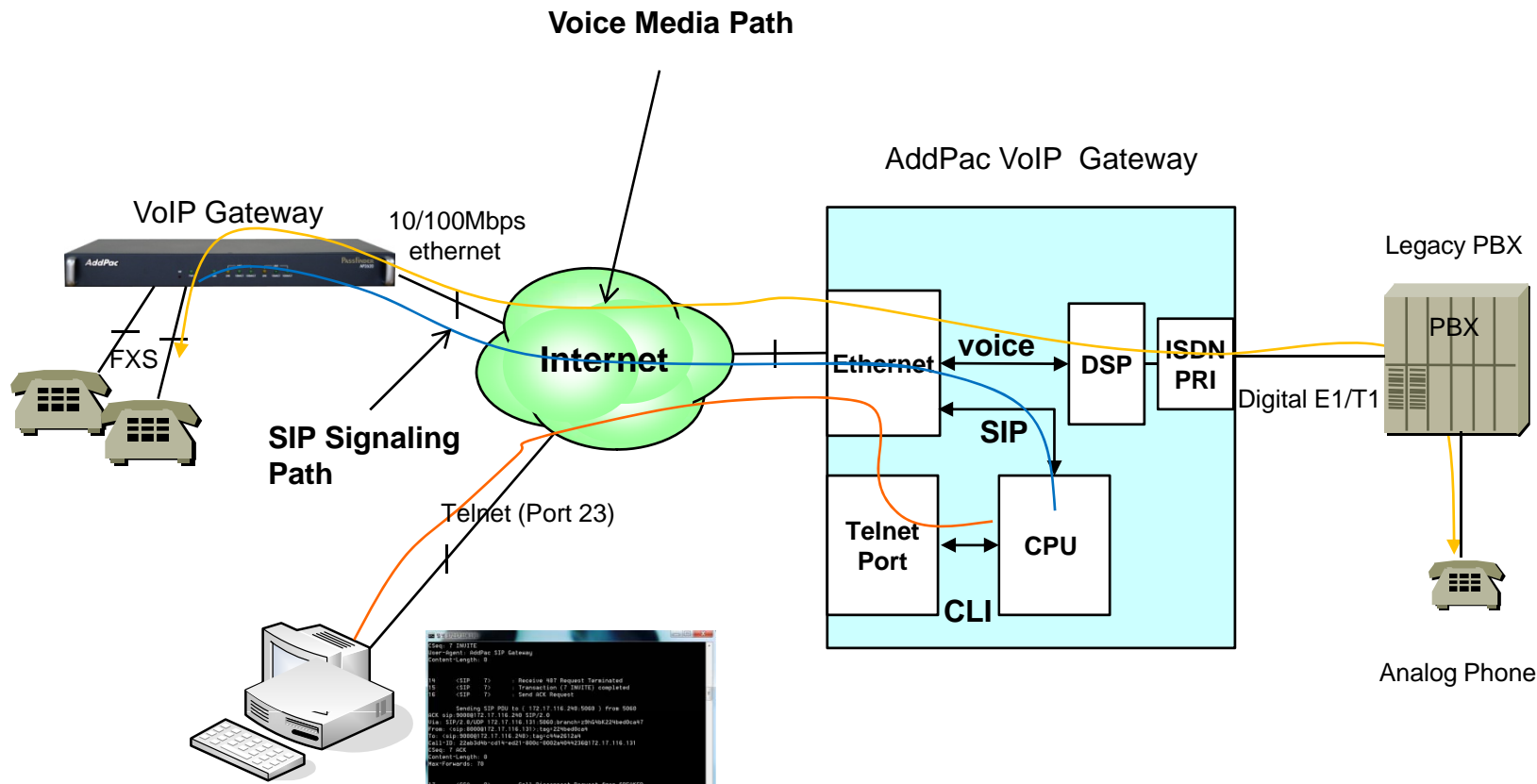
Network Diagram for SIP Debugging



SIP Debugging Access Method via Console Port



SIP Debugging Access Method via Telnet Port

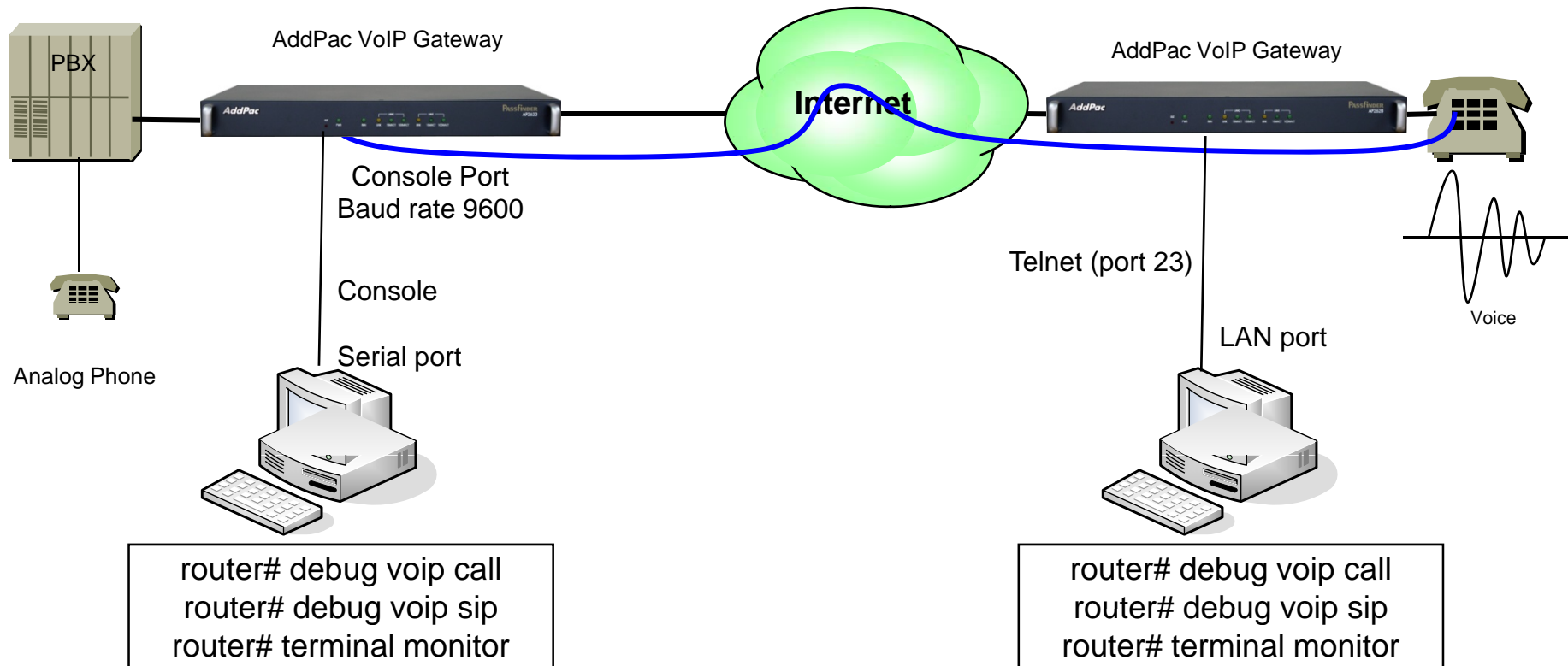


SIP Debugging Terminal (PC)

Usable with telnet access program such as TeraTerm, window command, Putty, CRT

CLI : Command Line Interface

Real-time SIP Debugging Environment



VoIP Debugging Commands

Major Command

Login: root

Password:

router> en

router# debug voip ?

-> VoIP Debugging command

h225-asn1 H.225 ASN.1 trace

h245-asn1 H.245 ASN.1 trace

ras-asn1 RAS ASN.1 trace

call Call trace

-> Call trace Debugging Command

mgcp MGCP message trace

number debug on a specific number (calling or called party number)

port debug on a specific voice port

sip SIP message trace

-> SIP Message Debugging

router# **terminal monitor**

-> Display SIP debug message through terminal

Useful when debugging by remote accessing

router# **no terminal monitor**

-> Use when deactivate of debug command (valid when accessing telnet)

SIP Debugging Commands (Example)

```
Welcome, APOS(tm) Kernel Version 8.50.006.
Copyright (c) 1999-2008 AddPac Technology Co., Ltd.

User Access Verification

Login: root
Password:
router> en
router# debug voip call
router# debug voip sip
router# terminal monitor
router# 1 <CCA 0> : Call Initiate Request from SPEAKER , peer(-
1760673404) digits() addr() type(SIP) plar(0) directDigit(FALSE)
name()
2 <CEP 000000> : Call Received
3 <CEP 000000> : Call Initiated : calledNumber() crv(0) total(0)
4 <Call 8> : ***** Call Created status(InitiatedBySPEECH)
ver(8.28:2006-02-06-00-00) time(1262390967) ****
5 <CEP 000000> : Calling number(8000)
6 <CEP 000000> : Call id(b78e3e4b-3c3f-2691-800e-0002a4044236)
callNum(8)
7 <Call 8> : Call Initiated from CCA : peer(-1760673404), digits(), ip()
8 <CCA 0> : Digit Received : 9(START) status(1)
9 <CCA 0> : Digit Received : 9(STOP) status(4)
10 <Call 8> : Digit(9) at InitiatedBySPEECH
11 <Call 8> : MatchedAll
12 <CCA 0> : Digit Received : 0(START) status(4)
13 <CCA 0> : Digit Received : 0(STOP) status(4)
14 <Call 8> : Digit(0) at CalleeDeterminedWaitDigit
15 <Call 8> : MatchedAll
16 <CCA 0> : Digit Received : 0(START) status(4)
17 <CCA 0> : Digit Received : 0(STOP) status(4)
18 <Call 8> : Digit(0) at CalleeDeterminedWaitDigit
19 <Call 8> : MatchedAll
20 <CCA 0> : Digit Received : 0(START) status(4)
21 <CCA 0> : Digit Received : 0(STOP) status(4)
22 <Call 8> : Digit(0) at CalleeDeterminedWaitDigit
23 <Call 8> : MatchedPerfect
24 <Call 8> : MatchAllProcess After Sorted
```

```
<0> id(9999) dest(9000) prefer(0) selected(3)
    <1> id(1001) dest(T) prefer(0) selected(0)
    <2> id(1002) dest(T) prefer(1) selected(0)
    <3> id(3000) dest(T) prefer(2) selected(0)
25 <Call 8> : Initiate callee with dial-peer(9000)
    status(CalleeDeterminedAll) id(b78e3e4b-3c3f-2691-800e-
    0002a4044236)
26 <NetEP 8> : InitiateOutCall: calledNum(9000) callingNum(8000)
    target(172.17.116.240)
27 <NetEP 8> : DoCall: calledAddr(sip:9000@172.17.116.240)
    callingAddr(8000)
28 <SIP 8> : SetLocalAudioFormats : outbound(TRUE) hqaEnable(FALSE)
29 <SIP 8> : SetLocalAudioFormats : myVoipPeer(9999) is not NULL,
    voiceCodecClass(0)
30 <SIP 8> : SetLocalAudioFormats : outbound(TRUE) hqaEnable(FALSE)
31 <SIP 8> : SetLocalAudioFormats : myVoipPeer(9999) is not NULL,
    voiceCodecClass(0)
32 <SIP 0> : No authentication information available
33 <SIP 8> : Send INVITE Request
    Sending SIP PDU to ( 172.17.116.240:5060 ) from 5060
INVITE sip:9000@172.17.116.240 SIP/2.0
Via: SIP/2.0/UDP 172.17.116.131:5060;branch=z9hG4bKbb4b310fa49
From: <sip:8000@172.17.116.131>;tag=bb4b310fa4
To: <sip:9000@172.17.116.240>
Call-ID: bb8e3e4b-0fa6-314b-800f-0002a4044236@172.17.116.131
CSeq: 9 INVITE
Supported: replaces, timer, 100rel, early-session
Min-SE: 1800
Date: Sat, 02 Jan 2010 00:09:31 GMT
Session-Expires: 1800
User-Agent: AddPac SIP Gateway
Contact: <sip:8000@172.17.116.131>
Accept: application/sdp
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, UPDATE, PRACK, REFER, NOTIFY,
    INFO
Allow-Events: talk, hold, conference
Content-Type: application/sdp
Content-Length: 304
Max-Forwards: 70
```


SIP Debugging Commands (Example)

```
v=0
o=8000 1262390971 1262390971 IN IP4 172.17.116.131
s=AddPac Gateway SDP
c=IN IP4 172.17.116.131
t=1262390971 0
m=audio 23016 RTP/AVP 0 8 18 4 2 9
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=rtpmap:4 G723/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:9 G722/8000
Received SIP PDU from ( 172.17.116.240:5060 )
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.17.116.131:5060;branch=z9hG4bKbb4b310fa49
From: <sip:8000@172.17.116.131>;tag=bb4b310fa4
To: <sip:9000@172.17.116.240>
Call-ID: bb8e3e4b-0fa6-314b-800f-0002a4044236@172.17.116.131
CSeq: 9 INVITE
User-Agent: AddPac SIP Gateway
Content-Length: 0
34 <SIP 8> : Receive 100 Trying
35 <SIP 8> : Transaction (9 INVITE) proceeding

Received SIP PDU from ( 172.17.116.240:5060 )
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.17.116.131:5060;branch=z9hG4bKbb4b310fa49
From: <sip:8000@172.17.116.131>;tag=bb4b310fa4
To: <sip:9000@172.17.116.240>;tag=384e3113a4
Call-ID: bb8e3e4b-0fa6-314b-800f-0002a4044236@172.17.116.131
CSeq: 9 INVITE
Supported: timer, replaces, early-session
User-Agent: AddPac SIP Gateway
Contact: sip:9000@172.17.116.240
RSeq: 223744
Require: 100rel
Content-Type: application/sdp
Content-Length: 434
```

```
Received SIP PDU from ( 172.17.116.240:5060 )
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.116.131:5060;branch=z9hG4bKbb4b310fa49
From: <sip:8000@172.17.116.131>;tag=bb4b310fa4
To: <sip:9000@172.17.116.240>;tag=384e3113a4
Call-ID: bb8e3e4b-0fa6-314b-800f-0002a4044236@172.17.116.131
CSeq: 9 INVITE
Supported: timer, replaces, early-session
Session-Expires: 1800;refresher=uac
User-Agent: AddPac SIP Gateway
Contact: sip:9000@172.17.116.240
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, UPDATE, REFER, NOTIFY, INFO
Require: timer
Content-Length: 0

43 <SIP 8> : Receive 200 OK
44 <SIP 8> : Received INVITE OK response
45 <SIP 8> : Send ACK Request

Sending SIP PDU to ( 172.17.116.240:5060 ) from 5060
ACK sip:9000@172.17.116.240 SIP/2.0
Via: SIP/2.0/UDP 172.17.116.131:5060;branch=z9hG4bKbb4b310fa49
From: <sip:8000@172.17.116.131>;tag=bb4b310fa4
To: <sip:9000@172.17.116.240>;tag=384e3113a4
Call-ID: bb8e3e4b-0fa6-314b-800f-0002a4044236@172.17.116.131
CSeq: 9 ACK
Content-Length: 0
Max-Forwards: 70

router#
router# no terminal monitor
```

VoIP Gateway Series

Thank you!

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