SIP Protocol Debugging Guide For AddPac VoIP Product

AddPac

AddPac Technology

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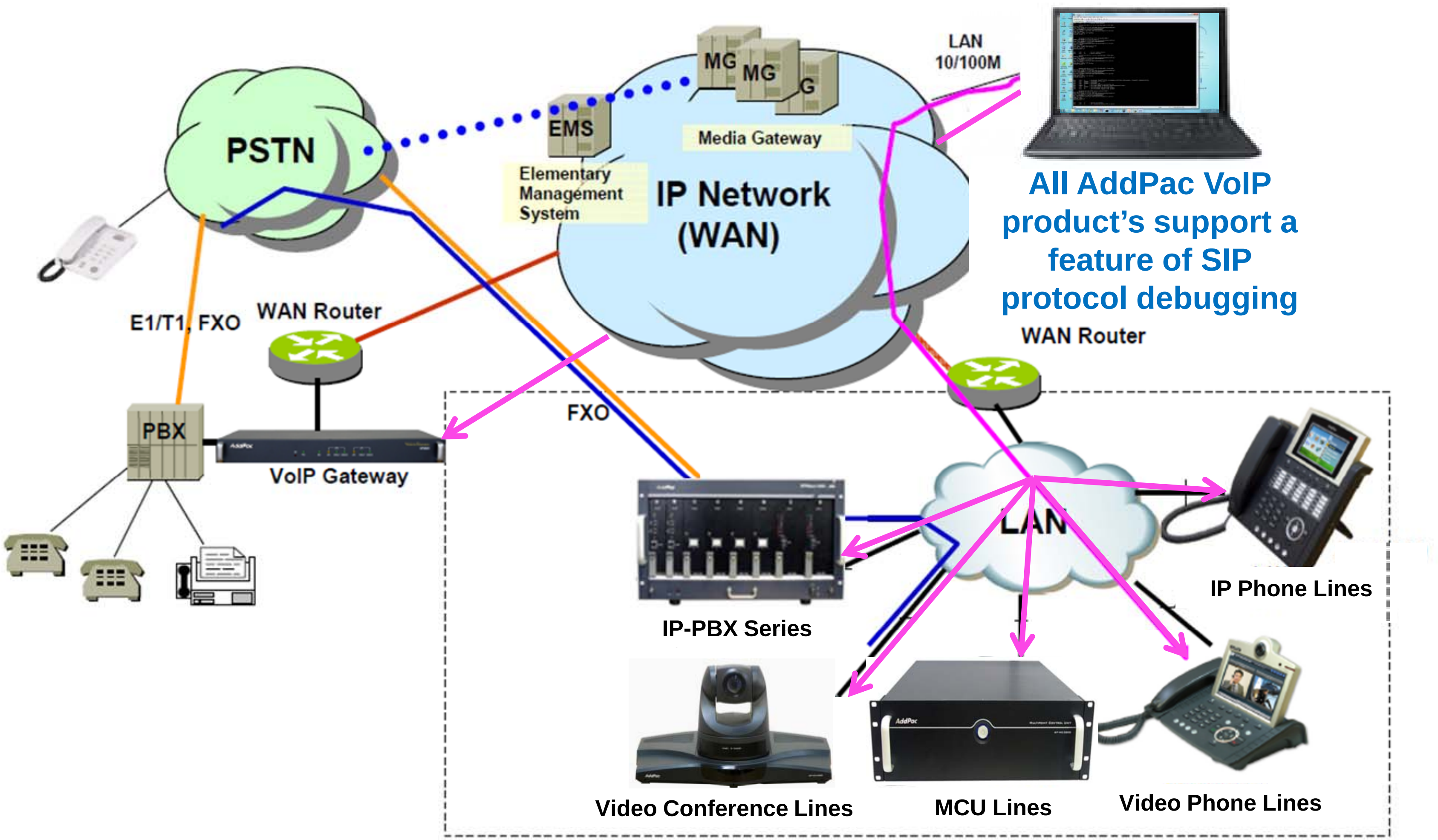
www.addpac.com

System Diagram of SIP Usage



Debugging Features

- All AddPac VoIP product's support a feature of SIP protocol debugging
  - IP-PBX Series
    - IPNext20, 50, 100, 150(SOHO Call Manager)
    - IPNext200, 500, 600, 700, 1000, 3000, 5000, 10000
  - Video Conference Lines
    - VC200, 1000, 2000, 5000
    - MC1000, 2000, 3000, 5000
  - IP/Video Phone Lines
    - IP90, 100, 120, 150, 160, 230, 300
    - VP120, 150, 200, 230, 250, 280, 300, 350, 500
  - All AddPac gateway product

Debugging Feature(Cont.)

Access Method for AddPac Gateway Debugging

Access with AddPac Device Console Port

AddPac Gateway

Windows HyperTerminal

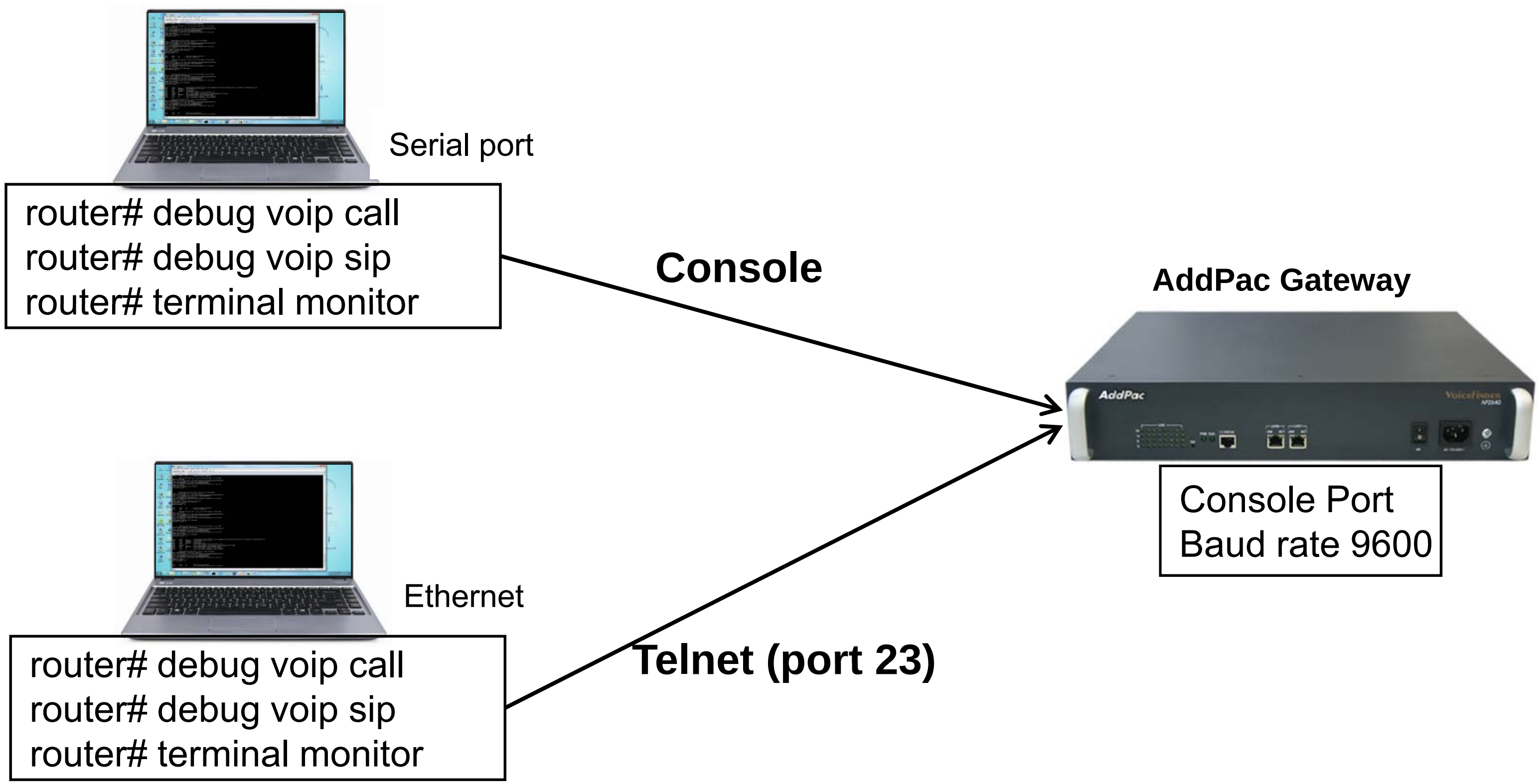
Access with AddPac Device telnet

AddPac Gateway

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

Debugging Feature(Cont.)

Debugging is possible in real time regarding AddPac gateway trace information



Debugging Feature(Cont.)

AddPac VoIP products support debugging regarding call trace and SIP protocol

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

Debugging Example

```
Welcome, APOS(tm) Kernel Version 8.50.006.
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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
```

Debugging Example(Cont.)

```
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
```

Thank you!

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