

# GSM VoIP Gateway Series

## SIP Protocol Debugging Service Overview



**AddPac**

**AddPac Technology**

2011, Sales and Marketing

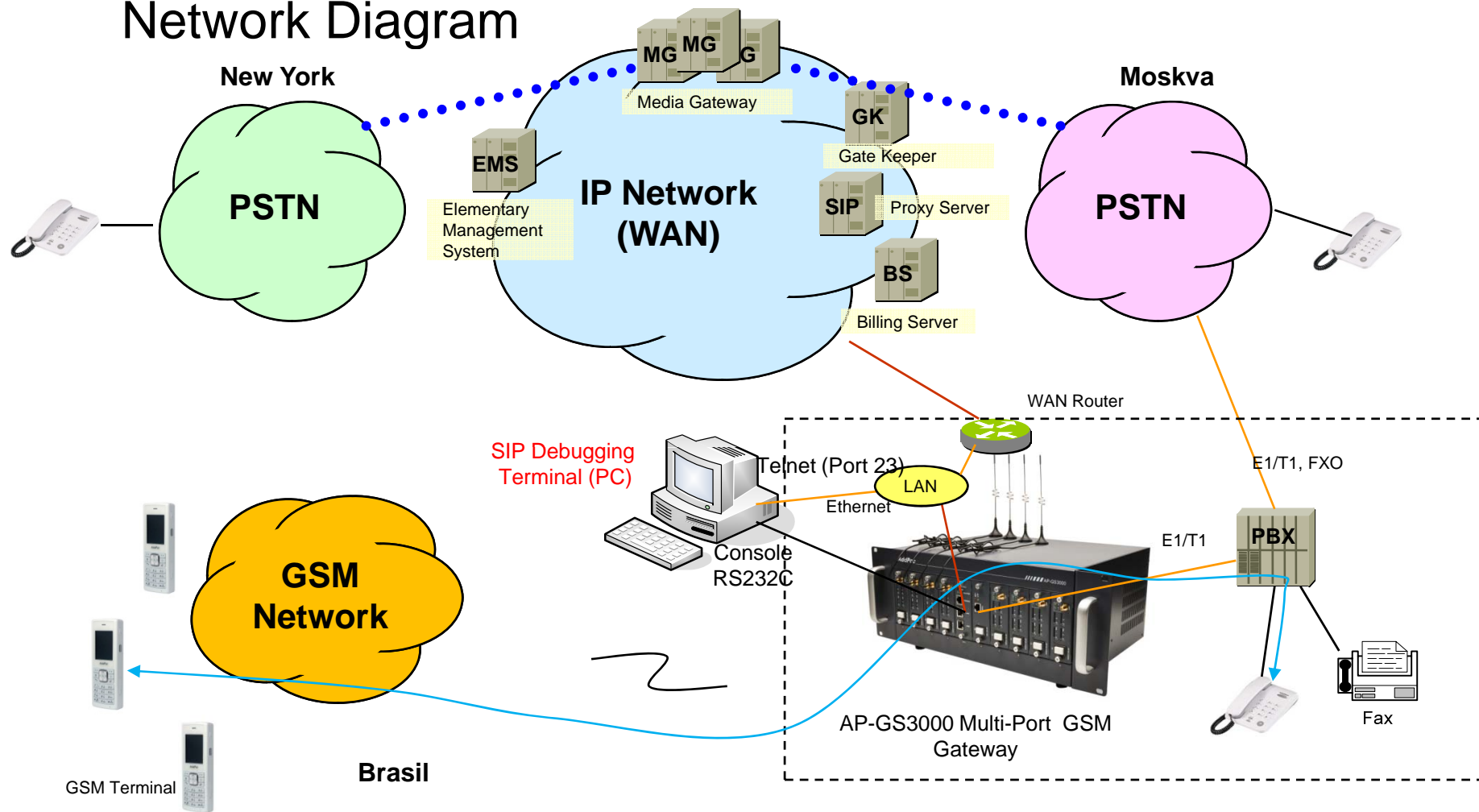
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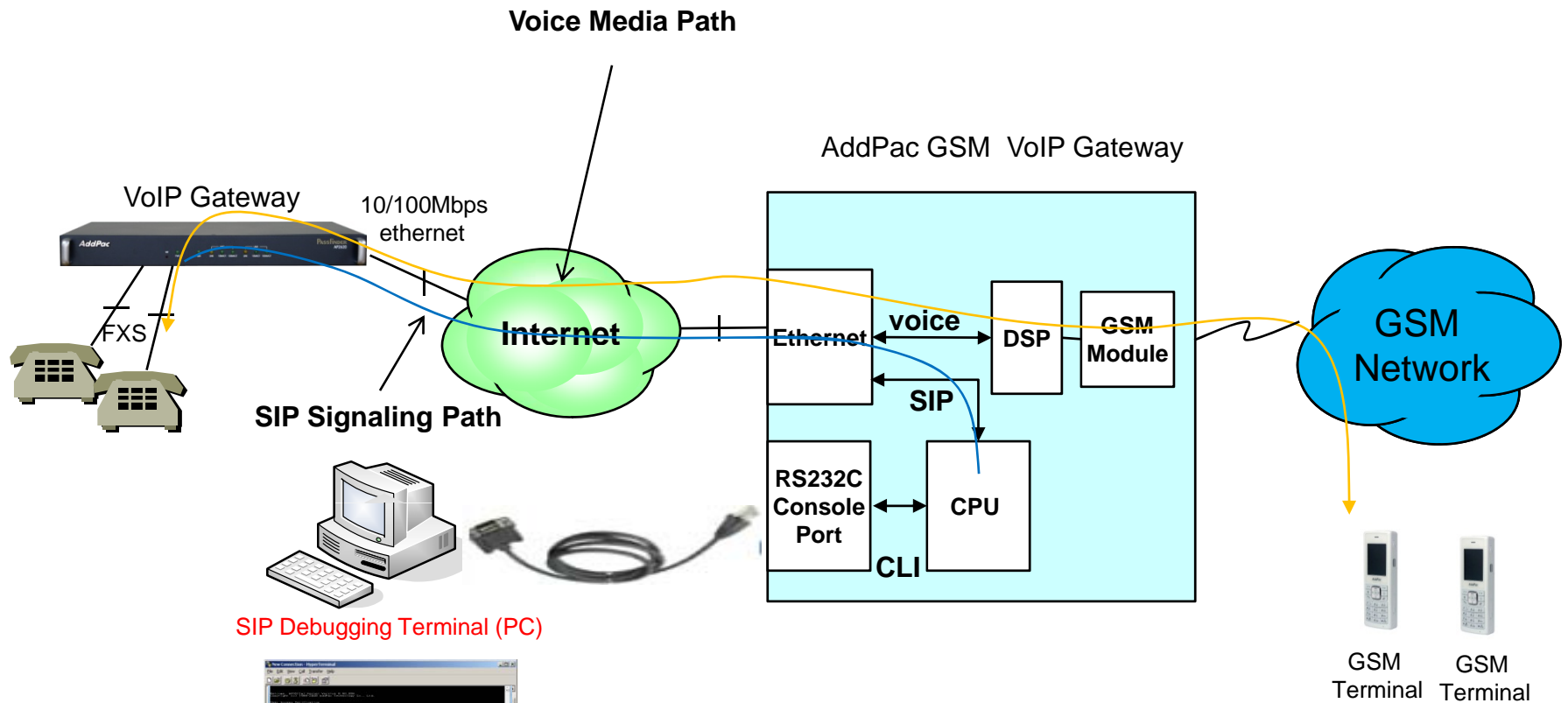
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# Network Diagram for SIP Debugging

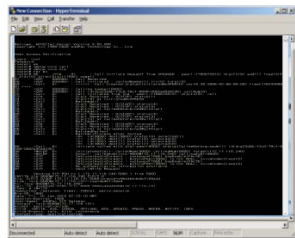
## Network Diagram



# SIP Debugging Access Method via Console Port



SIP Debugging Terminal (PC)

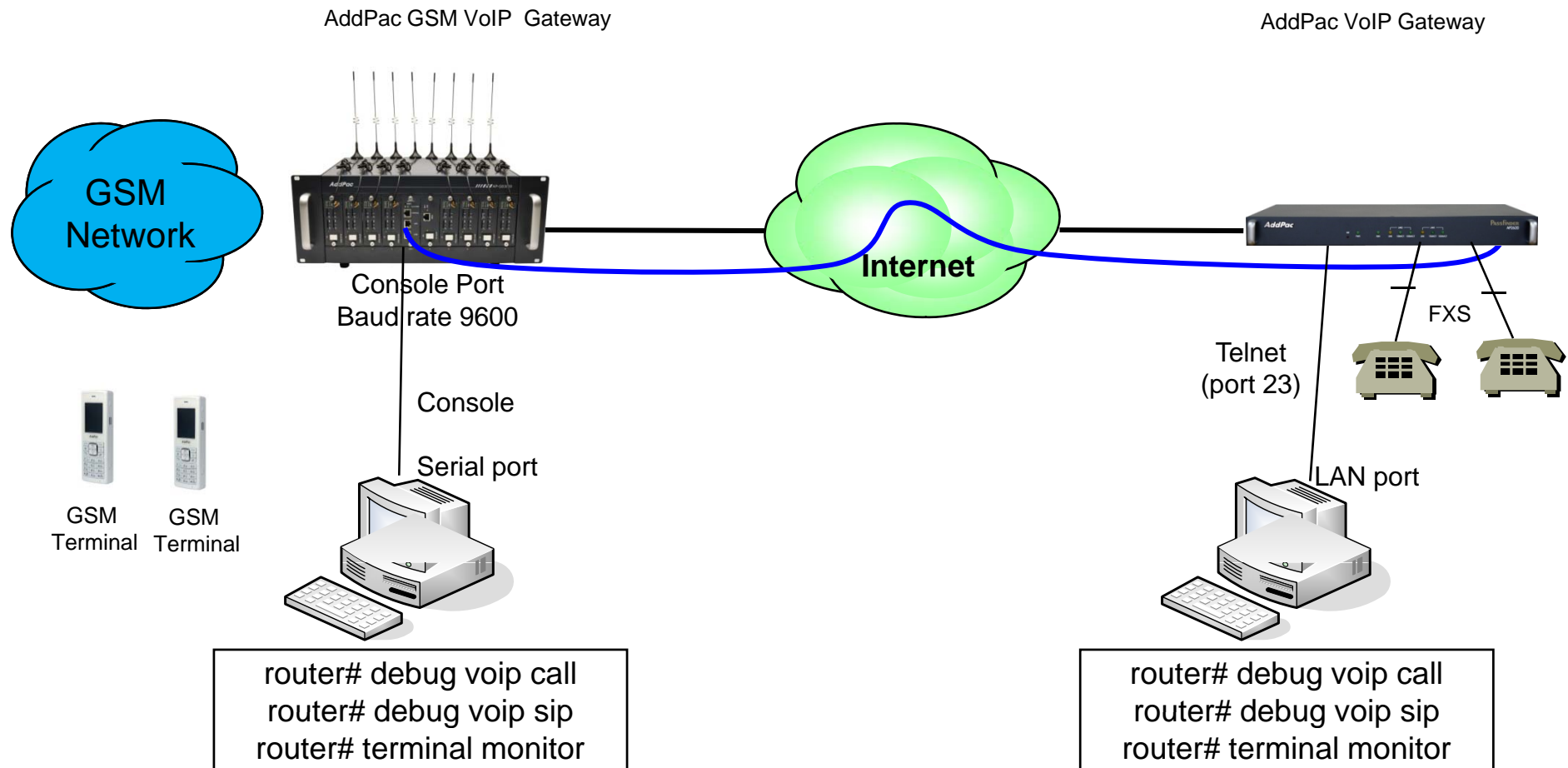


Windows HyperTerminal

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
```

# GSM VoIP Gateway Series

# Thank you!

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