**Asterisk Ready.**

\*CLI> [Jan 23 12:14:17] **NOTICE**[5596]: **res\_pjsip/pjsip\_distributor.c**:**255** **log\_unidentified\_request**: Request from '"4007" <sip:4007@192.168.1.6>' failed for '192.168.1.162:57605' (callid: df825ca3d3574235bf57e00b7c103a09) - No matching endpoint found

[Jan 23 12:14:18] **NOTICE**[5596]: **res\_pjsip/pjsip\_distributor.c**:**255** **log\_unidentified\_request**: Request from '"4007" <sip:4007@192.168.1.6>' failed for '192.168.1.162:57605' (callid: df825ca3d3574235bf57e00b7c103a09) - No matching endpoint found

[Jan 23 12:14:18] **NOTICE**[5596]: **res\_pjsip/pjsip\_distributor.c**:**255** **log\_unidentified\_request**: Request from '"4007" <sip:4007@192.168.1.6>' failed for '192.168.1.162:57605' (callid: df825ca3d3574235bf57e00b7c103a09) - No matching endpoint found

\*CLI> pjsip set logger o

off on

\*CLI> pjsip set logger on

PJSIP Logging enabled

\*CLI> <--- Received SIP request (733 bytes) from UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.6 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.149;branch=z9hG4bKacGVHDplP

Max-Forwards: 70

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 1 INVITE

Contact: <sip:101@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

User-Agent: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Type: application/sdp

Content-Length: 224

v=0

o=AudiocodesGW 495397 751953 IN IP4 192.168.1.149

s=Phone-Call

c=IN IP4 192.168.1.149

t=0 0

m=audio 4000 RTP/AVP 0 96

a=rtpmap:0 pcmu/8000

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-15

a=ptime:20

a=sendrecv

<--- Transmitting SIP response (432 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacGVHDplP

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=z9hG4bKacGVHDplP

CSeq: 1 INVITE

WWW-Authenticate: Digest realm="asterisk",nonce="1421995525/33dbe749ed754930c579cd98369a59ba",opaque="3e4e8e1f40a0c8c1",algorithm=md5,qop="auth"

Content-Length: 0

<--- Received SIP request (491 bytes) from UDP:192.168.1.149:5060 --->

ACK sip:110@192.168.1.6 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.149;branch=z9hG4bKacGVHDplP

Max-Forwards: 70

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=z9hG4bKacGVHDplP

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 1 ACK

Contact: <sip:101@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

User-Agent: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Length: 0

<--- Received SIP request (989 bytes) from UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.6 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.149;branch=z9hG4bKacBrdZLHt

Max-Forwards: 70

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 2 INVITE

Authorization: Digest username="101",realm="asterisk",nc=00000001,Cnonce="0a123bcf",nonce="1421995525/33dbe749ed754930c579cd98369a59ba",opaque="3e4e8e1f40a0c8c1",uri="sip:110@192.168.1.6",qop=auth,algorithm=MD5,response="5b2c3cf95424a2f17407e674a364dfeb"

Contact: <sip:101@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

User-Agent: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Type: application/sdp

Content-Length: 224

v=0

o=AudiocodesGW 495397 751953 IN IP4 192.168.1.149

s=Phone-Call

c=IN IP4 192.168.1.149

t=0 0

m=audio 4000 RTP/AVP 0 96

a=rtpmap:0 pcmu/8000

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-15

a=ptime:20

a=sendrecv

<--- Received SIP request (989 bytes) from UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.6 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.149;branch=z9hG4bKacBrdZLHt

Max-Forwards: 70

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 2 INVITE

Authorization: Digest username="101",realm="asterisk",nc=00000001,Cnonce="0a123bcf",nonce="1421995525/33dbe749ed754930c579cd98369a59ba",opaque="3e4e8e1f40a0c8c1",uri="sip:110@192.168.1.6",qop=auth,algorithm=MD5,response="5b2c3cf95424a2f17407e674a364dfeb"

Contact: <sip:101@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

User-Agent: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Type: application/sdp

Content-Length: 224

v=0

o=AudiocodesGW 495397 751953 IN IP4 192.168.1.149

s=Phone-Call

c=IN IP4 192.168.1.149

t=0 0

m=audio 4000 RTP/AVP 0 96

a=rtpmap:0 pcmu/8000

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-15

a=ptime:20

a=sendrecv

<--- Transmitting SIP response (258 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacBrdZLHt

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>

CSeq: 2 INVITE

Content-Length: 0

<--- Transmitting SIP response (258 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacBrdZLHt

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>

CSeq: 2 INVITE

Content-Length: 0

**--** Executing [110@default:1] **Dial**("**PJSIP/101-00000000**", "**PJSIP/110**") in new stack

**--** Called PJSIP/110

<--- Transmitting SIP request (853 bytes) to UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjc12dc324-b29d-4a19-97c3-41285f752674

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>

Contact: <sip:ef4efae7-c5aa-407d-9289-d9e697b76799@192.168.1.6:5060>

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10840 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Supported: 100rel, timer, replaces, norefersub

Session-Expires: 1800

Min-SE: 90

Content-Type: application/sdp

Content-Length: 235

v=0

o=- 2038571437 2038571437 IN IP4 192.168.1.6

s=Asterisk

c=IN IP4 192.168.1.6

t=0 0

m=audio 10590 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=maxptime:150

a=sendrecv

<--- Received SIP response (562 bytes) from UDP:192.168.1.149:5060 --->

SIP/2.0 422 Session Interval Too Small

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjc12dc324-b29d-4a19-97c3-41285f752674

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c313112661

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10840 INVITE

Contact: <sip:192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

Min-SE: 7200

Server: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Length: 0

<--- Transmitting SIP response (444 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacBrdZLHt

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

CSeq: 2 INVITE

Contact: <sip:192.168.1.6:5060>

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Content-Length: 0

<--- Transmitting SIP request (330 bytes) to UDP:192.168.1.149:5060 --->

ACK sip:110@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjc12dc324-b29d-4a19-97c3-41285f752674

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c313112661

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10840 ACK

Content-Length: 0

<--- Transmitting SIP request (855 bytes) to UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjda105a77-9174-4728-a74c-bb8729644317

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>

Contact: <sip:ef4efae7-c5aa-407d-9289-d9e697b76799@192.168.1.6:5060>

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10841 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Supported: 100rel, timer, replaces, norefersub

Session-Expires: 7200

Min-SE: 7200

Content-Type: application/sdp

Content-Length: 235

v=0

o=- 2038571437 2038571437 IN IP4 192.168.1.6

s=Asterisk

c=IN IP4 192.168.1.6

t=0 0

m=audio 10590 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=maxptime:150

a=sendrecv

<--- Received SIP response (497 bytes) from UDP:192.168.1.149:5060 --->

SIP/2.0 100 Trying

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjda105a77-9174-4728-a74c-bb8729644317

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c28952301

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10841 INVITE

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

Server: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Length: 0

<--- Received SIP response (532 bytes) from UDP:192.168.1.149:5060 --->

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjda105a77-9174-4728-a74c-bb8729644317

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c28952301

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10841 INVITE

Contact: <sip:110@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

Server: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Length: 0

**--** PJSIP/110-00000001 is ringing

<--- Transmitting SIP response (444 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 180 Ringing

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacBrdZLHt

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

CSeq: 2 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Contact: <sip:192.168.1.6:5060>

Content-Length: 0

<--- Received SIP response (855 bytes) from UDP:192.168.1.149:5060 --->

SIP/2.0 200 OK

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjda105a77-9174-4728-a74c-bb8729644317

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c28952301

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10841 INVITE

Contact: <sip:110@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

Require: timer

Session-Expires: 7200;refresher=uas

Min-SE: 7200

Server: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Type: application/sdp

Content-Length: 228

v=0

o=AudiocodesGW 827619 374176 IN IP4 192.168.1.149

s=Phone-Call

c=IN IP4 192.168.1.149

t=0 0

m=audio 4090 RTP/AVP 0 101

a=rtpmap:0 pcmu/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=ptime:150

a=sendrecv

<--- Transmitting SIP request (329 bytes) to UDP:192.168.1.149:5060 --->

ACK sip:110@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPj099edbe9-248e-4c55-8dbc-45b832b26dcd

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c28952301

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10841 ACK

Content-Length: 0

**--** PJSIP/110-00000001 answered PJSIP/101-00000000

<--- Transmitting SIP response (745 bytes) to UDP:192.168.1.149:5060 --->

SIP/2.0 200 OK

Via: SIP/2.0/UDP 192.168.1.149;rport=5060;received=192.168.1.149;branch=z9hG4bKacBrdZLHt

Call-ID: 655031991fhcO@192.168.1.149

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

CSeq: 2 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Contact: <sip:192.168.1.6:5060>

Supported: 100rel, timer, replaces, norefersub

Content-Type: application/sdp

Content-Length: 224

v=0

o=- 495397 751955 IN IP4 192.168.1.6

s=Asterisk

c=IN IP4 192.168.1.6

t=0 0

m=audio 14394 RTP/AVP 0 96

a=rtpmap:0 PCMU/8000

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-16

a=ptime:20

a=maxptime:150

a=sendrecv

**--** Channel PJSIP/110-00000001 joined 'simple\_bridge' basic-bridge <5cea0917-63d8-4c99-929e-9132dff45f34>

**--** Channel PJSIP/101-00000000 joined 'simple\_bridge' basic-bridge <5cea0917-63d8-4c99-929e-9132dff45f34>

<--- Received SIP request (512 bytes) from UDP:192.168.1.149:5060 --->

ACK sip:192.168.1.6:5060 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.149;branch=z9hG4bKacShcPKxs

Max-Forwards: 70

From: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 2 ACK

Contact: <sip:101@192.168.1.149>

Supported: em,timer,replaces,path

Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE

User-Agent: Audiocodes-Sip-Gateway-MP-124 FXS/v.4.40.200.371

Content-Length: 0

**>** Bridge 5cea0917-63d8-4c99-929e-9132dff45f34: switching from simple\_bridge technology to native\_rtp

<--- Transmitting SIP request (885 bytes) to UDP:192.168.1.149:5060 --->

INVITE sip:110@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPjaa4102b5-71d6-457b-8143-96053a72e6e6

From: <sip:101@192.168.1.6>;tag=35a31505-1804-4d5e-94d6-66f45284f08a

To: <sip:110@192.168.1.149>;tag=1c28952301

Contact: <sip:ef4efae7-c5aa-407d-9289-d9e697b76799@192.168.1.6:5060>

Call-ID: 54945122-2ddb-4643-9965-91ffb154c5f3

CSeq: 10842 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Supported: 100rel, timer, replaces, norefersub

Session-Expires: 7200;refresher=uas

Min-SE: 7200

Content-Type: application/sdp

Content-Length: 236

v=0

o=- 2038571437 2038571438 IN IP4 192.168.1.6

s=Asterisk

c=IN IP4 192.168.1.149

t=0 0

m=audio 4000 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=maxptime:150

a=sendrecv

<--- Transmitting SIP request (811 bytes) to UDP:192.168.1.149:5060 --->

INVITE sip:101@192.168.1.149 SIP/2.0

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPj15b3d2f1-8308-4b6a-9eb0-8589b90dde1a

From: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

To: <sip:101@192.168.1.6>;tag=1c577729459

Contact: <sip:192.168.1.6:5060>

Call-ID: 655031991fhcO@192.168.1.149

CSeq: 14849 INVITE

Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, REGISTER, MESSAGE, REFER

Supported: 100rel, timer, replaces, norefersub

Session-Expires: 1800

Min-SE: 90

Content-Type: application/sdp

Content-Length: 225

v=0

o=- 495397 751956 IN IP4 192.168.1.6

s=Asterisk

c=IN IP4 192.168.1.149

t=0 0

m=audio 4090 RTP/AVP 0 96

a=rtpmap:0 PCMU/8000

a=rtpmap:96 telephone-event/8000

a=fmtp:96 0-16

a=ptime:20

a=maxptime:150

a=sendrecv

<--- Received SIP response (555 bytes) from UDP:192.168.1.149:5060 --->

SIP/2.0 422 Session Interval Too Small

Via: SIP/2.0/UDP 192.168.1.6:5060;rport;branch=z9hG4bKPj15b3d2f1-8308-4b6a-9eb0-8589b90dde1a

From: <sip:110@192.168.1.6>;tag=1316185c-2fcd-4030-bbd0-4c2910cc63bd

To: <sip:101@192.168.1.6>;tag=1c577729459

To: <sip:101@192.168.1.6>;tag=1c577729459

Killed