[May 20 02:47:30] Reliably Transmitting (NAT) to 88.131.158.171:5060:

OPTIONS sip:siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK4f6175ed;rport

From: "asterisk" <sip:asterisk@192.168.1.67>;tag=as3c1440fb

To: <sip:siptrunk.phonera.se;cpd=on>

Contact: <sip:asterisk@192.168.1.67>

Call-ID: 0f57fa2c3543bedf60afa3ce16236c5c@192.168.1.67

CSeq: 102 OPTIONS

User-Agent: Asterisk PBX

Max-Forwards: 70

Date: Mon, 19 May 2014 23:47:30 GMT

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

Supported: replaces

Content-Length: 0

---

[May 20 02:47:30]

<--- SIP read from 88.131.158.171:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK4f6175ed;rport=64074

Contact: <sip:leica@88.131.158.171:5060>

To: <sip:siptrunk.phonera.se;cpd=on>;tag=0a8e812d

From: "asterisk"<sip:asterisk@192.168.1.67>;tag=as3c1440fb

Call-ID: 0f57fa2c3543bedf60afa3ce16236c5c@192.168.1.67

CSeq: 102 OPTIONS

WWW-Authenticate: Digest nonce="1400543250:ab9fbe93a438d3bd72fd1b7553cc2847",algorithm=MD5,realm="192.168.1.67",qop="auth",stale=false

Content-Length: 0

<------------->

[May 20 02:47:30] --- (9 headers 0 lines) ---

[May 20 02:47:30] Really destroying SIP dialog '0f57fa2c3543bedf60afa3ce16236c5c@192.168.1.67' Method: OPTIONS

[May 20 02:47:31] -- Executing [0774424246@default:1] **AGI**("**SIP/8001-00000000**", "**agi://127.0.0.1:4577/call\_log**") in new stack

[May 20 02:47:31] -- AGI Script agi://127.0.0.1:4577/call\_log completed, returning 0

[May 20 02:47:31] -- Executing [0774424246@default:2] **Dial**("**SIP/8001-00000000**", "**SIP/0774424246@C901||tTo**") in new stack

[May 20 02:47:31] Audio is at 192.168.1.67 port 11244

[May 20 02:47:31] Adding codec 0x4 (ulaw) to SDP

[May 20 02:47:31] Adding non-codec 0x1 (telephone-event) to SDP

[May 20 02:47:31] Reliably Transmitting (NAT) to 88.131.158.171:5060:

INVITE sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7413c57b;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 102 INVITE

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Date: Mon, 19 May 2014 23:47:31 GMT

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

Supported: replaces

Content-Type: application/sdp

Content-Length: 211

v=0

o=root 4177 4177 IN IP4 192.168.1.67

s=session

c=IN IP4 192.168.1.67

t=0 0

m=audio 11244 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=sendrecv

---

[May 20 02:47:31] -- Called 0774424246@C901

[May 20 02:47:31]

<--- SIP read from 88.131.158.171:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7413c57b;rport=64074

Contact: <sip:leica@88.131.158.171:5060>

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=3b1af102

From: "8001"<sip:0720627616@192.168.1.67>;tag=as32270419

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 102 INVITE

WWW-Authenticate: Digest nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b",algorithm=MD5,realm="192.168.1.67",qop="auth",stale=false

Content-Length: 0

<------------->

[May 20 02:47:31] --- (9 headers 0 lines) ---

[May 20 02:47:31] Transmitting (NAT) to 88.131.158.171:5060:

ACK sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7413c57b;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=3b1af102

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 102 ACK

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Content-Length: 0

---

[May 20 02:47:31] Audio is at 192.168.1.67 port 11244

[May 20 02:47:31] Adding codec 0x4 (ulaw) to SDP

[May 20 02:47:31] Adding non-codec 0x1 (telephone-event) to SDP

[May 20 02:47:31] Reliably Transmitting (NAT) to 88.131.158.171:5060:

INVITE sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7a5b5135;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 103 INVITE

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Authorization: Digest username="atelo.lin2..phonera.se", realm="192.168.1.67", algorithm=MD5, uri="sip:0774424246@siptrunk.phonera.se;cpd=on", nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b", response="281c1c5f1729d51f784bc6b3b67c51c4", qop=auth, cnonce="3933e8ff", nc=00000001

Date: Mon, 19 May 2014 23:47:31 GMT

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

Supported: replaces

Content-Type: application/sdp

Content-Length: 211

v=0

o=root 4177 4178 IN IP4 192.168.1.67

s=session

c=IN IP4 192.168.1.67

t=0 0

m=audio 11244 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=sendrecv

---

[May 20 02:47:31]

<--- SIP read from 88.131.158.171:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7a5b5135;rport=64074

Contact: <sip:leica@88.131.158.171:5060>

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=98c98e0a

From: "8001"<sip:0720627616@192.168.1.67>;tag=as32270419

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 103 INVITE

WWW-Authenticate: Digest nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b",algorithm=MD5,realm="192.168.1.67",qop="auth",stale=false

Content-Length: 0

<------------->

[May 20 02:47:31] --- (9 headers 0 lines) ---

[May 20 02:47:31] Transmitting (NAT) to 88.131.158.171:5060:

ACK sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK7a5b5135;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=98c98e0a

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 103 ACK

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Content-Length: 0

---

[May 20 02:47:31] Audio is at 192.168.1.67 port 11244

[May 20 02:47:31] Adding codec 0x4 (ulaw) to SDP

[May 20 02:47:31] Adding non-codec 0x1 (telephone-event) to SDP

[May 20 02:47:31] Reliably Transmitting (NAT) to 88.131.158.171:5060:

INVITE sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK4f33385c;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 104 INVITE

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Authorization: Digest username="atelo.lin2..phonera.se", realm="192.168.1.67", algorithm=MD5, uri="sip:0774424246@siptrunk.phonera.se;cpd=on", nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b", response="8c682ccb349c2454a6e9cdab98702165", qop=auth, cnonce="0d928091", nc=00000002

Date: Mon, 19 May 2014 23:47:31 GMT

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

Supported: replaces

Content-Type: application/sdp

Content-Length: 211

v=0

o=root 4177 4179 IN IP4 192.168.1.67

s=session

c=IN IP4 192.168.1.67

t=0 0

m=audio 11244 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=sendrecv

---

[May 20 02:47:31]

<--- SIP read from 88.131.158.171:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK4f33385c;rport=64074

Contact: <sip:leica@88.131.158.171:5060>

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=05d67463

From: "8001"<sip:0720627616@192.168.1.67>;tag=as32270419

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 104 INVITE

WWW-Authenticate: Digest nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b",algorithm=MD5,realm="192.168.1.67",qop="auth",stale=false

Content-Length: 0

<------------->

[May 20 02:47:31] --- (9 headers 0 lines) ---

[May 20 02:47:31] Transmitting (NAT) to 88.131.158.171:5060:

ACK sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK4f33385c;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=05d67463

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 104 ACK

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Content-Length: 0

---

[May 20 02:47:31] Audio is at 192.168.1.67 port 11244

[May 20 02:47:31] Adding codec 0x4 (ulaw) to SDP

[May 20 02:47:31] Adding non-codec 0x1 (telephone-event) to SDP

[May 20 02:47:31] Reliably Transmitting (NAT) to 88.131.158.171:5060:

INVITE sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK57a9aa31;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 105 INVITE

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Authorization: Digest username="atelo.lin2..phonera.se", realm="192.168.1.67", algorithm=MD5, uri="sip:0774424246@siptrunk.phonera.se;cpd=on", nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b", response="222181abeb8c7c34080d6d233cb0b61c", qop=auth, cnonce="4498e065", nc=00000003

Date: Mon, 19 May 2014 23:47:31 GMT

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

Supported: replaces

Content-Type: application/sdp

Content-Length: 211

v=0

o=root 4177 4180 IN IP4 192.168.1.67

s=session

c=IN IP4 192.168.1.67

t=0 0

m=audio 11244 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

a=ptime:20

a=sendrecv

---

[May 20 02:47:31]

<--- SIP read from 88.131.158.171:5060 --->

SIP/2.0 401 Unauthorized

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK57a9aa31;rport=64074

Contact: <sip:leica@88.131.158.171:5060>

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=8b01d63e

From: "8001"<sip:0720627616@192.168.1.67>;tag=as32270419

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 105 INVITE

WWW-Authenticate: Digest nonce="1400543251:67c81d08870352ea4a9c32a2d5c0e08b",algorithm=MD5,realm="192.168.1.67",qop="auth",stale=false

Content-Length: 0

<------------->

[May 20 02:47:31] --- (9 headers 0 lines) ---

[May 20 02:47:31] Transmitting (NAT) to 88.131.158.171:5060:

ACK sip:0774424246@siptrunk.phonera.se;cpd=on SIP/2.0

Via: SIP/2.0/UDP 192.168.1.67:5060;branch=z9hG4bK57a9aa31;rport

From: "8001" <sip:0720627616@192.168.1.67>;tag=as32270419

To: <sip:0774424246@siptrunk.phonera.se;cpd=on>;tag=8b01d63e

Contact: <sip:0720627616@192.168.1.67>

Call-ID: 34f9e9c84a0826a83439fa787c79883a@192.168.1.67

CSeq: 105 ACK

User-Agent: Asterisk PBX

Max-Forwards: 70

Remote-Party-ID: "8001" <sip:0720627616@192.168.1.67>;privacy=off;screen=no

Content-Length: 0

---

[May 20 02:47:31] **NOTICE**[4228]: **chan\_sip.c**:**13470** **handle\_response\_invite**: Failed to authenticate on INVITE to '"8001" <sip:0720627616@192.168.1.67>;tag=as32270419'

[May 20 02:47:31] -- SIP/C901-00000001 is circuit-busy

[May 20 02:47:31] == Everyone is busy/congested at this time (1:0/1/0)

[May 20 02:47:31] -- Executing [0774424246@default:3] **Hangup**("**SIP/8001-00000000**", "") in new stack

[May 20 02:47:31] == Spawn extension (default, 0774424246, 3) exited non-zero on 'SIP/8001-00000000'

[May 20 02:47:31] -- Executing [h@default:1] **DeadAGI**("**SIP/8001-00000000**", "**agi://127.0.0.1:4577/call\_log--HVcauses--PRI-----NODEBUG-----21-----CONGESTION----------**") in new stack

[May 20 02:47:31] -- AGI Script agi://127.0.0.1:4577/call\_log--HVcauses--PRI-----NODEBUG-----21-----CONGESTION---------- completed, returning 0

[May 20 02:47:31] Really destroying SIP dialog '34f9e9c84a0826a83439fa787c79883a@192.168.1.67' Method: INVITE

go\*CLI>