```
<--- SIP read from UDP:172.16.18.3:5060 --->
INVITE sip:5555555555@10.0.0.229:5060;user=phone SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102827-2025231428-1351659
420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102827 INVITE
Contact:<sip:172.16.18.3:5060>
P-Asserted-Identity:"First Last"<sip:4444444444@172.16.18.3;user=phone>
Privacy:none
Supported:100rel
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept:application/media_control+xml,application/sdp,application/x-broadworks-call-center+xml,multipart/
mixed
Max-Forwards:10
Content-Type:application/sdp
Content-Length:278

v=0
o=BroadWorks 562265 1 IN IP4 172.16.18.130
s=-
c=IN IP4 172.16.18.130
t=0 0
a=sendrecv
m=audio 10658 RTP/AVP 9 0 8 18 127
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:127 telephone-event/8000
<------------->
--- (15 headers 13 lines) ---
Sending to 172.16.18.3:5060 (NAT)
Using INVITE request as basis request - BW045700245311012-1928226041@172.16.18.3
Found peer 'bw1' for '4444444444' from 172.16.18.3:5060
   == Using SIP RTP CoS mark 5
Found RTP audio format 9
Found RTP audio format 0
Found RTP audio format 8
Found RTP audio format 18
Found RTP audio format 127
Found audio description format G722 for ID 9
Found audio description format PCMU for ID 0
Found audio description format PCMA for ID 8
Found audio description format G729 for ID 18
Found audio description format telephone-event for ID 127
Capabilities: us - 0x4 (ulaw), peer - audio=0x110c (ulaw|alaw|g729|g722)/video=0x0 (nothing)/text=0x0
(nothing), combined - 0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x1 (telephone-event|), combined - 0x1
(telephone-event|)
Peer audio RTP is at port 172.16.18.130:10658
```

```
Looking for 5555555555 in sip-subscriber (domain 10.0.0.229)
list_route: hop: <sip:172.16.18.3:5060>

<--- Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102827-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102827 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Length: 0


<------------>
    -- Executing [5555555555@sip-subscriber:1] Macro("SIP/bw1-00000006",
"nanp-outbound,5555555555") in new stack
    -- Executing [s@macro-nanp-outbound:1] GotoIf("SIP/bw1-00000006", "0?directory,1") in new stack
    -- Executing [s@macro-nanp-outbound:2] Set("SIP/bw1-00000006", "extensions=""") in new stack
    -- Executing [s@macro-nanp-outbound:3] Set("SIP/bw1-00000006", "extensions_home=") in new
stack
    -- Executing [s@macro-nanp-outbound:4] GotoIf("SIP/bw1-00000006", "0?congest") in new stack
    -- Executing [s@macro-nanp-outbound:5] GotoIf("SIP/bw1-00000006", "0?internal-remote,1") in new
stack
    -- Executing [s@macro-nanp-outbound:6] Set("SIP/bw1-00000006", "to="npa=720,state=CO,"") in
new stack
    -- Executing [s@macro-nanp-outbound:7] Set("SIP/bw1-00000006", "to_state=CO") in new stack
    -- Executing [s@macro-nanp-outbound:8] GotoIf("SIP/bw1-00000006", "0?tollfree,1") in new stack
    -- Executing [s@macro-nanp-outbound:9] GotoIf("SIP/bw1-00000006", "0?international,1") in new
stack
    -- Executing [s@macro-nanp-outbound:10] Goto("SIP/bw1-00000006", "interstate,1") in new stack
    -- Goto (macro-nanp-outbound,interstate,1)
    -- Executing [interstate@macro-nanp-outbound:1] Dial("SIP/bw1-00000006",
"SIP/5555555555@siptrunk") in new stack
  == Using SIP RTP CoS mark 5
Audio is at 10846
Adding codec 0x4 (ulaw) to SDP
Reliably Transmitting (NAT) to 10.0.1.46:5060:
INVITE sip:5555555555@10.0.1.46 SIP/2.0
Via: SIP/2.0/UDP 10.0.0.229:5060;branch=z9hG4bK258648fd;rport
Max-Forwards: 70
From: "First Last" <sip:4444444444@10.0.0.229>;tag=as04147867
To: <sip:5555555555@10.0.1.46>
Contact: <sip:4444444444@10.0.0.229:5060>
Call-ID: 75a383061cbb71fe796576eb7c3227d4@10.0.0.229:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX 1.8.15.0
Date: Wed, 31 Oct 2012 04:57:00 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
```

```
Content-Length: 177

v=0
o=root 1496572978 1496572978 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 10846 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=ptime:20
a=sendrecv

---
    -- Called SIP/5555555555@siptrunk

<--- SIP read from UDP:10.0.1.46:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 10.0.0.229:5060;branch=z9hG4bK258648fd;received=10.0.0.229;rport=5060
From: "First Last" <sip:4444444444@10.0.0.229>;tag=as04147867
To: <sip:5555555555@10.0.1.46>
Call-ID: 75a383061cbb71fe796576eb7c3227d4@10.0.0.229:5060
CSeq: 102 INVITE
Content-Length: 0

<------------->
--- (7 headers 0 lines) ---

<--- SIP read from UDP:10.0.1.46:5060 --->
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 10.0.0.229:5060;branch=z9hG4bK258648fd;received=10.0.0.229;rport=5060
From: "First Last" <sip:4444444444@10.0.0.229>;tag=as04147867
To: <sip:5555555555@10.0.1.46>;tag=781f0bfbbb650cbab569b061d424caa9
Call-ID: 75a383061cbb71fe796576eb7c3227d4@10.0.0.229:5060
CSeq: 102 INVITE
Contact: <sip:5555555555@10.0.1.46:5060;transport=udp>
Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS
Content-Length: 184
Content-Disposition: session; handling=required
Content-Type: application/sdp

v=0
o=Sonus_UAC 146100 14610001 IN IP4 10.0.1.38
s=SIP Media Capabilities
c=IN IP4 10.0.1.38
t=0 0
m=audio 28436 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendrecv
a=maxptime:20
<------------->
--- (11 headers 9 lines) ---
list_route: hop: <sip:5555555555@10.0.1.46:5060;transport=udp>
Found RTP audio format 0
Found audio description format PCMU for ID 0
Capabilities: us - 0x4 (ulaw), peer - audio=0x4 (ulaw)/video=0x0 (nothing)/text=0x0 (nothing), combined -
```

```
0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x0 (nothing), peer - 0x0 (nothing), combined - 0x0 (nothing)
Peer audio RTP is at port 10.0.1.38:28436
    -- SIP/siptrunk-00000007 is making progress passing it to SIP/bw1-00000006
Audio is at 17410
Adding codec 0x4 (ulaw) to SDP
Adding non-codec 0x1 (telephone-event) to SDP

<--- Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102827-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102827 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Type: application/sdp
Content-Length: 231

v=0
o=root 771287585 771287585 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 17410 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
a=fmtp:127 0-16
a=ptime:20
a=sendrecv

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

<--- SIP read from UDP:10.0.1.46:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.0.0.229:5060;branch=z9hG4bK258648fd;received=10.0.0.229;rport=5060
From: "First Last" <sip:4444444444@10.0.0.229>;tag=as04147867
To: <sip:5555555555@10.0.1.46>;tag=781f0bfbbb650cbab569b061d424caa9
Call-ID: 75a383061cbb71fe796576eb7c3227d4@10.0.0.229:5060
CSeq: 102 INVITE
Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed
Contact: <sip:5555555555@10.0.1.46:5060;transport=udp>
Allow:
INVITE,ACK,CANCEL,BYE,REGISTER,REFER,INFO,SUBSCRIBE,NOTIFY,PRACK,UPDATE,OPTIONS
Require: timer
Supported: timer
Session-Expires: 7200;refresher=uac
Content-Length: 184
Content-Disposition: session; handling=required
Content-Type: application/sdp
```

```
v=0
o=Sonus_UAC 146100 14610001 IN IP4 10.0.1.38
s=SIP Media Capabilities
c=IN IP4 10.0.1.38
t=0 0
m=audio 28436 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=sendrecv
a=maxptime:20
<------------->
--- (15 headers 9 lines) ---
list_route: hop: <sip:5555555555@10.0.1.46:5060;transport=udp>
set_destination: Parsing <sip:5555555555@10.0.1.46:5060;transport=udp> for address/port to send to
set_destination: set destination to 10.0.1.46:5060
Transmitting (NAT) to 10.0.1.46:5060:
ACK sip:5555555555@10.0.1.46:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 10.0.0.229:5060;branch=z9hG4bK05c0ca34;rport
Max-Forwards: 70
From: "First Last" <sip:4444444444@10.0.0.229>;tag=as04147867
To: <sip:5555555555@10.0.1.46>;tag=781f0bfbbb650cbab569b061d424caa9
Contact: <sip:4444444444@10.0.0.229:5060>
Call-ID: 75a383061cbb71fe796576eb7c3227d4@10.0.0.229:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX 1.8.15.0
Content-Length: 0


---
    -- SIP/siptrunk-00000007 answered SIP/bw1-00000006
Audio is at 17410
Adding codec 0x4 (ulaw) to SDP
Adding non-codec 0x1 (telephone-event) to SDP

<--- Reliably Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102827-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102827 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Type: application/sdp
Content-Length: 231

v=0
o=root 771287585 771287586 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 17410 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
```

```
a=rtpmap:127 telephone-event/8000
a=fmtp:127 0-16
a=ptime:20
a=sendrecv

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

<--- SIP read from UDP:172.16.18.3:5060 --->
ACK sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102827A2025231428-135165
9420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102827 ACK
Contact:<sip:172.16.18.3:5060>
Max-Forwards:10
Content-Length:0

<------------->
--- (9 headers 0 lines) ---

<--- SIP read from UDP:172.16.18.3:5060 --->
INVITE sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102828-2025231428-1351659
420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102828 INVITE
Contact:<sip:172.16.18.3:5060>
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported:
Accept:application/media_control+xml,application/sdp,application/x-broadworks-call-center+xml,multipart/
mixed
Max-Forwards:10
Content-Type:application/sdp
Content-Length:195

v=0
o=BroadWorks 562265 2 IN IP4 172.16.18.130
s=-
c=IN IP4 172.16.18.130
t=0 0
a=inactive
m=audio 10658 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
a=inactive
<------------->
--- (13 headers 10 lines) ---
Sending to 172.16.18.3:5060 (NAT)
Found RTP audio format 0
Found RTP audio format 127
```

Found audio description format PCMU for ID 0
Found audio description format telephone-event for ID 127
Capabilities: us - 0x4 (ulaw), peer - audio=0x4 (ulaw)/video=0x0 (nothing)/text=0x0 (nothing), combined - 0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x1 (telephone-event|), combined - 0x1 (telephone-event|)
Peer audio RTP is at port 172.16.18.130:10658

<--- Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102828-2025231428-1351659420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102828 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Length: 0

<------------>
Audio is at 17410
Adding codec 0x4 (ulaw) to SDP
Adding non-codec 0x1 (telephone-event) to SDP

<--- Reliably Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102828-2025231428-1351659420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102828 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Type: application/sdp
Content-Length: 231

v=0
o=root 771287585 771287587 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 17410 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
a=fmtp:127 0-16
a=ptime:20
a=inactive

```
<------------>
    -- Started music on hold, class 'none', on SIP/siptrunk-00000007

<--- SIP read from UDP:172.16.18.3:5060 --->
ACK sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102828A2025231428-135165
9420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102828 ACK
Contact:<sip:172.16.18.3:5060>
Max-Forwards:10
Content-Length:0

<------------->
--- (9 headers 0 lines) ---

<--- SIP read from UDP:172.16.18.3:5060 --->
INVITE sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102829-2025231428-1351659
420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102829 INVITE
Contact:<sip:172.16.18.3:5060>
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported:
Accept:application/media_control+xml,application/sdp,application/x-broadworks-call-center+xml,multipart/
mixed
Max-Forwards:10
Content-Type:application/sdp
Content-Length:244

v=0
o=BroadWorks 562270 1 IN IP4 10.1.75.183
s=-
c=IN IP4 10.1.75.183
t=0 0
m=audio 14160 RTP/AVP 0 8 9 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
<------------->
--- (13 headers 12 lines) ---
Sending to 172.16.18.3:5060 (NAT)

<--- Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 100 Trying
```

```
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102829-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102829 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Length: 0


<------------>
Audio is at 17410
Adding codec 0x4 (ulaw) to SDP
Adding non-codec 0x1 (telephone-event) to SDP

<--- Reliably Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102829-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102829 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Type: application/sdp
Content-Length: 231

v=0
o=root 771287585 771287587 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 17410 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
a=fmtp:127 0-16
a=ptime:20
a=inactive

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

<--- SIP read from UDP:172.16.18.3:5060 --->
ACK sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102829A2025231428-135165
9420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
```

```
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102829 ACK
Contact:<sip:172.16.18.3:5060>
Max-Forwards:10
Content-Length:0

<------------->
--- (9 headers 0 lines) ---
Really destroying SIP dialog 'BW042053603311012-2026918305@172.16.18.3' Method: BYE

<--- SIP read from UDP:172.16.18.3:5060 --->
INVITE sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102830-2025231428-1351659
420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102830 INVITE
Contact:<sip:172.16.18.3:5060>
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Supported:
Accept:application/media_control+xml,application/sdp,application/x-broadworks-call-center+xml,multipart/
mixed
Max-Forwards:10
Content-Type:application/sdp
Content-Length:290

v=0
o=BroadWorks 562265 3 IN IP4 172.16.18.130
s=-
c=IN IP4 172.16.18.130
t=0 0
a=sendrecv
m=audio 10658 RTP/AVP 0 9 8 18 127
a=sendrecv
a=rtpmap:0 PCMU/8000
a=rtpmap:9 G722/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:127 telephone-event/8000
<------------->
--- (13 headers 14 lines) ---
Sending to 172.16.18.3:5060 (NAT)
Found RTP audio format 0
Found RTP audio format 9
Found RTP audio format 8
Found RTP audio format 18
Found RTP audio format 127
Found audio description format PCMU for ID 0
Found audio description format G722 for ID 9
Found audio description format PCMA for ID 8
Found audio description format G729 for ID 18
Found audio description format telephone-event for ID 127
Capabilities: us - 0x4 (ulaw), peer - audio=0x110c (ulaw|alaw|g729|g722)/video=0x0 (nothing)/text=0x0
```

```
(nothing), combined - 0x4 (ulaw)
Non-codec capabilities (dtmf): us - 0x1 (telephone-event|), peer - 0x1 (telephone-event|), combined - 0x1
(telephone-event|)
Peer audio RTP is at port 172.16.18.130:10658

<--- Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102830-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102830 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Length: 0


<------------>
Audio is at 17410
Adding codec 0x4 (ulaw) to SDP
Adding non-codec 0x1 (telephone-event) to SDP

<--- Reliably Transmitting (NAT) to 172.16.18.3:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102830-2025231428-1351659
420245-;received=172.16.18.3;rport=5060
From: "First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To: <sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID: BW045700245311012-1928226041@172.16.18.3
CSeq: 446102830 INVITE
Server: Asterisk PBX 1.8.15.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:5555555555@10.0.0.229:5060>
Content-Type: application/sdp
Content-Length: 231

v=0
o=root 771287585 771287588 IN IP4 10.0.0.229
s=Asterisk PBX 1.8.15.0
c=IN IP4 10.0.0.229
t=0 0
m=audio 17410 RTP/AVP 0 127
a=rtpmap:0 PCMU/8000
a=rtpmap:127 telephone-event/8000
a=fmtp:127 0-16
a=ptime:20
a=sendrecv

<------------>
    -- Stopped music on hold on SIP/siptrunk-00000007
```

```
<--- SIP read from UDP:172.16.18.3:5060 --->
ACK sip:5555555555@10.0.0.229:5060 SIP/2.0
Via:SIP/2.0/UDP
172.16.18.3;branch=z9hG4bKBroadWorks.-oqcorr-10.0.0.229V5060-0-446102830A2025231428-135165
9420245-
From:"First Last"<sip:4444444444@172.16.18.3;user=phone>;tag=2025231428-1351659420245-
To:<sip:5555555555@10.0.0.229:5060;user=phone>;tag=as1e956584
Call-ID:BW045700245311012-1928226041@172.16.18.3
CSeq:446102830 ACK
Contact:<sip:172.16.18.3:5060>
Max-Forwards:10
Content-Length:0

<------------->
--- (9 headers 0 lines) ---
```