Asterisk IPv6 Implementation
Test Report

Presented to Digium
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1. Introduction

This report describes the test plan and lists the results of the tests that were made against the IPv6 port of Asterisk done during spring 2010. Some comparisons between the trunk IPv4 version of Asterisk and the IPv6 port based on that trunk version are also described.

The document use the following notation:

- 4 means IPv4
- 6 means IPv6
- so 464 means a call between IPv4 to IPv6 to IPv4.

Section 5. provides a summary of test results.

2. Test Setup

Figure 1 shows the test setup used throughout this document. When the IPv4-only trunk version of Asterisk is being tested, the IPv6 user-agents (UA's) are not registered.

Figure 1: Test setup comprising three IPv4 user-agents, three IPv6 user-agents, and a dual-stack Asterisk server.

UA1, UA2, UA3 are IPv4-only. UA4, UA5, UA6 are IPv6-only.

Asterisk is running on a testing machine with the following specs:

- Intel Core Solo T1500 2.0 GHz
- 2 GB RAM
- Marvell RDK-8053 Ethernet controller
  - Network to tester system is gigabit full-duplex.
  - Checksum offloading is disabled with "ethtool -K".
- Fedora 13
  - kernel-2.6.33.5-112.fc13.i686
  - glibc-2.12-2.i686

Asterisk versions tested:

- trunk: r270974, available at: http://svn.digium.com/svn/asterisk/trunk
- v6-new: r271228, available at: http://svn.digium.com/svn/asterisk/team/group/v6-new

Asterisk's configuration is the following:

2.1. modules.conf

```
[modules]
load => app_dial.so
load => chan_sip.so
load => func_channel.so
load => func_strings.so
load => pbx_config.so
load => res_rtp_asterisk.so
```

2.2. sip.conf

```
[general]
udpbindaddr=[xyz::123]:5060[1]
disallow=all
allow=ulaw

[1]
type=friend
host=dynamic

[2]
type=friend
host=dynamic [2]

[3]
type=friend
host=dynamic

[4]
type=friend
host=dynamic

[5]
type=friend
host=dynamic

[6]
type=friend
host=dynamic
```

2.3. extensions.conf

```
[default]
exten => 1,1,Dial(SIP/1)
exten => 2,1,Dial(SIP/2)
exten => 3,1,Dial(SIP/3)
exten => 4,1,Dial(SIP/4)
exten => 5,1,Dial(SIP/5)
exten => 6,1,Dial(SIP/6)
```

There is no other file in /etc/asterisk.

1 Here, "xyz::123" is the IPv6 address of the system running Asterisk. When testing the IPv4-only trunk, we use "udpbindaddr=a.b.c.d" instead, where "a.b.c.d" is its IPv4 address.

2 When running performance tests with *sipp*, UA2 does not register. Instead, its address and port are statically configured with "host=<address>" and "port=<number>".

3. Functional Testing

We have used "pjsua", included in the open-source PJSIP SIP stack[3], for performing the functional tests.

The command line used in IPv4 mode is:

```
$ pjsua --registrar=sip:a.b.c.d --id=sip:x@a.b.c.d
```

where "a.b.c.d" represents the IPv4 address of the system under test and "x" represents the extension number being registered.

The command line used in IPv6 mode is:

```
$ pjsua --ipv6 --registrar=sip:[abc::def] --id=sip:x@[abc::def]
```

where "abc::def" represents the IPv6 address of the system under test and "x" represents the extension number being registered.

Once pjsua is running, the "m" command is used to make calls and the "x" command is used to transfer calls. The SIP URL entered is identical to the value of the "--id" flag from the corresponding pjsua instance.

3.1. IPv4 REGISTER

Test description:

UA1 sends an IPv4 REGISTER request to Asterisk, which replies with a 200 Ok.

Results:

```
REGISTER sip:206.123.31.104 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5060;rport;branch=z9hG4bKPjnX7-BbWizEQtFk-
Z2IXJV3PurHXPF4cy
Max-Forwards: 70
From: <sip:1@206.123.31.104>;tag=2iq01YD2esl-i7sXLvp-FngprmEJm2Rr
To: <sip:1@206.123.31.104>
Call-ID: bp6gOZnsQaqXvyahYPH1WRA3scMEWNTQ
CSeq: 29478 REGISTER
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Contact: <sip:1@206.123.31.67:5060>
Expires: 300
Content-Length:  0
```

3 http://www.pjsip.org/pjsua.html

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.67:5060;branch=z9hG4bKPjnX7-BbWizEQtFk-
Z2IXJV3PurHXPF4cy;received=206.123.31.67;rport=5060
From: <sip:1@206.123.31.104>;tag=2iq01YD2esl-i7sXLvp-FngprmEJm2Rr
To: <sip:1@206.123.31.104>;tag=as62f63d5c
Call-ID: bp6gOZnsQaqXvyahYPH1WRA3scMEWNTQ
CSeq: 29478 REGISTER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Expires: 300
Contact: <sip:1@206.123.31.67:5060>;expires=300
Date: Thu, 17 Jun 2010 20:11:58 GMT
Content-Length: 0
```

Asterisk replied with 200 Ok. The test passed.

3.2. IPv6 REGISTER

Test description:

UA4 sends an IPv6 REGISTER request to Asterisk, which replies with a 200 Ok.

Results:

```
REGISTER sip:[2620:0:230:c000:216:cbff:fe92:2504] SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5060;rport;branch=z9hG4bKPjwqR7RQkr7YJqUol4L8kW6DyHtVskb737
Max-Forwards: 70
From: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=4ElbHnZCpvBEZ2rts8ELsfLS2pWHpLsK
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>
Call-ID: ALT4aXktRNdVl7JNUqOiCWolwhB.gkz0
CSeq: 48566 REGISTER
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Contact: <sip:4@[2620:0:230:c000::67]:5060>
Expires: 300
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5060;branch=z9hG4bKPjwqR7RQkr7YJqUol4L8kW6DyHtVskb737;received=262
0:0:230:c000::67;rport=5060
From: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=4ElbHnZCpvBEZ2rts8ELsfLS2pWHpLsK
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as278b31e4
Call-ID: ALT4aXktRNdVl7JNUqOiCWolwhB.gkz0
CSeq: 48566 REGISTER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Expires: 300
Contact: <sip:4@[2620:0:230:c000::67]:5060>;expires=300
Date: Thu, 17 Jun 2010 20:37:14 GMT
Content-Length: 0
```

Asterisk replied with a 200 Ok. The test passed.

3.3. IPv4-to-IPv4

Test description:

UA1 dials extension 2. Asterisk relays to UA2, then issues a re-INVITE so that IPv4 RTP flows directly.

Results:

```
INVITE sip:2@206.123.31.104 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5061;rport;branch=z9hG4bKPjkK4dM3ESLEbD9i.yw15g-
aFbr5pyZ3fp
Max-Forwards: 70
From: sip:1@206.123.31.104;tag=2kzz6lhbpnNuzjF8Xd4RVajXhOG0COzh
To: sip:2@206.123.31.104
Contact: <sip:1@206.123.31.67:5061>
Call-ID: nxJXYSuSbBLE9aqh0fXqCmlQoS7QbvGh
CSeq: 18180 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Type: application/sdp
Content-Length:   462

v=0
o=- 3485797623 3485797623 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4024 RTP/AVP 103 102 104 109 3 0 8 9 101
a=rtcp:4025 IN IP4 206.123.31.67
a=rtpmap:103 speex/16000
a=rtpmap:102 speex/8000
a=rtpmap:104 speex/32000
a=rtpmap:109 iLBC/8000
a=fmtp:109 mode=30
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.67:5061;branch=z9hG4bKPjkK4dM3ESLEbD9i.yw15g-
aFbr5pyZ3fp;received=206.123.31.67;rport=5061
From: sip:1@206.123.31.104;tag=2kzz6lhbpnNuzjF8Xd4RVajXhOG0COzh
To: sip:2@206.123.31.104
Call-ID: nxJXYSuSbBLE9aqh0fXqCmlQoS7QbvGh
CSeq: 18180 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:2@206.123.31.104:5060>
Content-Length: 0
```

```
INVITE sip:2@206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK4dacf41d
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as1a7e139f
To: <sip:2@206.123.31.67:5062>
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 52c8f43d70709d0862ff89f31b8eee96@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Thu, 17 Jun 2010 21:07:03 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 288

v=0
o=root 42482439 42482439 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 5818 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK4dacf41d
Call-ID: 52c8f43d70709d0862ff89f31b8eee96@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as1a7e139f
To: <sip:2@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK0ea0c637
Call-ID: 3b5624b93c314b08226479844fd05a29@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as01850306
To: <sip:2@206.123.31.67>;tag=FT.eLk2Ep2vrHnhSPMKUj2SxApg0FjCh
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485797598 3485797599 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4006 RTP/AVP 0 101
a=rtcp:4007 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK11883983
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as01850306
To: <sip:2@206.123.31.67:5062>;tag=FT.eLk2Ep2vrHnhSPMKUj2SxApg0FjCh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 3b5624b93c314b08226479844fd05a29@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.67:5061;branch=z9hG4bKPjh07ohJDmyc2qxEdU8YgUbpNBKssT8-
gA;received=206.123.31.67;rport=5061
From: sip:1@206.123.31.104;tag=kIMy9t9UBu1QbrtL37hSZSaOaZHIMVtN
To: sip:2@206.123.31.104;tag=as5cbc0613
Call-ID: 7YCfPv7EmbpRmOba1FCOt640GtZryaV-
CSeq: 11071 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:2@206.123.31.104:5060>
Content-Type: application/sdp
Content-Length: 293

v=0
o=root 2032642976 2032642976 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 21638 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:2@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjBGFs8sqavt7.GL.782nV4EpqZD7ufWYB
Max-Forwards: 70
From: sip:1@206.123.31.104;tag=kIMy9t9UBu1QbrtL37hSZSaOaZHIMVtN
To: sip:2@206.123.31.104;tag=as5cbc0613
Call-ID: 7YCfPv7EmbpRmOba1FCOt640GtZryaV-
CSeq: 11071 ACK
Content-Length:  0
```

```
INVITE sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK4498c429
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as01850306
To: <sip:2@206.123.31.67:5062>;tag=FT.eLk2Ep2vrHnhSPMKUj2SxApg0FjCh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 3b5624b93c314b08226479844fd05a29@206.123.31.104:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 290

v=0
o=root 2018166449 2018166450 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4018 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK4498c429
Call-ID: 3b5624b93c314b08226479844fd05a29@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as01850306
To: <sip:2@206.123.31.67>;tag=FT.eLk2Ep2vrHnhSPMKUj2SxApg0FjCh
CSeq: 103 INVITE
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485797598 3485797600 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4006 RTP/AVP 0 101
a=rtcp:4007 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK45173adc;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as5cbc0613
To: sip:1@206.123.31.104;tag=kIMy9t9UBu1QbrtL37hSZSaOaZHIMVtN
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 7YCfPv7EmbpRmOba1FCOt640GtZryaV-
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 290

v=0
o=root 2032642976 2032642977 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4006 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
206.123.31.104:5060;rport=5060;received=206.123.31.104;branch=z9hG4bK45173adc
Call-ID: 7YCfPv7EmbpRmOba1FCOt640GtZryaV-
From: <sip:2@206.123.31.104>;tag=as5cbc0613
To: <sip:1@206.123.31.104>;tag=kIMy9t9UBu1QbrtL37hSZSaOaZHIMVtN
CSeq: 102 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:1@206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485797598 3485797599 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4018 RTP/AVP 0 101
a=rtcp:4019 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK6aee4b21
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as01850306
To: <sip:2@206.123.31.67:5062>;tag=FT.eLk2Ep2vrHnhSPMKUj2SxApg0FjCh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 3b5624b93c314b08226479844fd05a29@206.123.31.104:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK1760804b;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as5cbc0613
To: sip:1@206.123.31.104;tag=kIMy9t9UBu1QbrtL37hSZSaOaZHIMVtN
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 7YCfPv7EmbpRmOba1FCOt640GtZryaV-
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

Asterisk sent a re-INVITE and RTP flows directly between UA1 and UA2. The test passed.

3.3.1. 444 Transfer

Test description:

UA2 transfers the IPv4 call to extension 3.

Results:

```
REFER sip:1@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5062;rport;branch=z9hG4bKPj7Tq0ZpYxBpBTQPwrvzZeY5eQQFR7KDCl
Max-Forwards: 70
From: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
To: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
Contact: <sip:206.123.31.67:5062>
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 10548 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:3@206.123.31.104
Referred-By: <sip:2@206.123.31.67>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
206.123.31.67:5062;rport;branch=z9hG4bKPj7Tq0ZpYxBpBTQPwrvzZeY5eQQFR7KDCl;received=206.1
23.31.67
From: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
To: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 10548 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:1@206.123.31.104:5060>
Content-Length: 0
```

```
NOTIFY sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK02b02ea1
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
To: <sip:2@206.123.31.67:5062>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 105 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=10548
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK02b02ea1
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
To: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
CSeq: 105 NOTIFY
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK28f71f1c
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
To: <sip:2@206.123.31.67:5062>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 106 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=10548
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK2d82ac24;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as1f672c0f
To: sip:1@206.123.31.104;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 291

v=0
o=root 894060185 894060187 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 10870 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
206.123.31.104:5060;rport=5060;received=206.123.31.104;branch=z9hG4bK2d82ac24
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
From: <sip:2@206.123.31.104>;tag=as1f672c0f
To: <sip:1@206.123.31.104>;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
CSeq: 103 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:1@206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485857416 3485857418 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4018 RTP/AVP 0 101
a=rtcp:4019 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK28f71f1c
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
To: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
CSeq: 106 NOTIFY
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:1@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5062;rport;branch=z9hG4bKPjtGSk-nji-
THDycyTqyUkLne1nOOOP48V
Max-Forwards: 70
From: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
To: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 10549 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
INVITE sip:3@206.123.31.67:5063 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK079a60b5
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67:5063>
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 13:43:44 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 292

v=0
o=root 1721407837 1721407837 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 6454 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK33495a99;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as1f672c0f
To: sip:1@206.123.31.104;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK079a60b5
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP 206.123.31.67:5062;rport;branch=z9hG4bKPjtGSk-nji-
THDycyTqyUkLne1nOOOP48V;received=206.123.31.67
From: <sip:2@206.123.31.67>;tag=7VhXemxWvaxmfnsKi7Ot3es.lmLMev.e
To: "1" <sip:1@206.123.31.104>;tag=as2af00e6f
Call-ID: 041a41dd20022092515f5be252ce9196@206.123.31.104:5060
CSeq: 10549 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK079a60b5
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67>;tag=B6OrxmuLaQR4nw9jKBzFBmPgtBjRAtJh
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5063>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485857424 3485857425 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4008 RTP/AVP 0 101
a=rtcp:4009 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5063 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK373fe179
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67:5063>;tag=B6OrxmuLaQR4nw9jKBzFBmPgtBjRAtJh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK2a33c93f;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as1f672c0f
To: sip:1@206.123.31.104;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
CSeq: 104 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 288

v=0
o=root 894060185 894060188 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4008 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
INVITE sip:206.123.31.67:5063 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK5dfd7830
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67:5063>;tag=B6OrxmuLaQR4nw9jKBzFBmPgtBjRAtJh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 290

v=0
o=root 1721407837 1721407838 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4018 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
206.123.31.104:5060;rport=5060;received=206.123.31.104;branch=z9hG4bK2a33c93f
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
From: <sip:2@206.123.31.104>;tag=as1f672c0f
To: <sip:1@206.123.31.104>;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
CSeq: 104 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:1@206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485857416 3485857419 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4018 RTP/AVP 0 101
a=rtcp:4019 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK5dfd7830
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67>;tag=B6OrxmuLaQR4nw9jKBzFBmPgtBjRAtJh
CSeq: 103 INVITE
Contact: <sip:206.123.31.67:5063>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485857424 3485857426 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4008 RTP/AVP 0 101
a=rtcp:4009 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK3f781444;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as1f672c0f
To: sip:1@206.123.31.104;tag=aYhvq-9eWh55s5H3vq8wEWrHAb5j7K9I
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 57mmevb6MgiDCQpXyTeQk.VaLLNeT3.E
CSeq: 104 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:206.123.31.67:5063 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK27777d2f
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as70d5020d
To: <sip:3@206.123.31.67:5063>;tag=B6OrxmuLaQR4nw9jKBzFBmPgtBjRAtJh
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0ca245d23bfb705f3614a6c22899a8d1@206.123.31.104:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was successfully transferred. After the transfer, Asterisk issued a re-INVITE and the RTP is flowing directly between UA1 and UA3. The test passed.

3.3.2. 446 Transfer

Test description:

UA2 transfers the IPv4 call to extension 4, on IPv6 UA4.

Test results:

```
REFER sip:1@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5062;rport;branch=z9hG4bKPj5L6DtoecKbyysoDmo9nh4AiyDICckuOG
Max-Forwards: 70
From: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
To: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
Contact: <sip:206.123.31.67:5062>
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 14386 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:4@206.123.31.104
Referred-By: <sip:2@206.123.31.67>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
206.123.31.67:5062;rport;branch=z9hG4bKPj5L6DtoecKbyysoDmo9nh4AiyDICckuOG;received=206.1
23.31.67
From: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
To: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 14386 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:1@206.123.31.104:5060>
Content-Length: 0
```

```
NOTIFY sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK139a13ed
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
To: <sip:2@206.123.31.67:5062>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 105 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=14386
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK139a13ed
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
To: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
CSeq: 105 NOTIFY
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK3453797a
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
To: <sip:2@206.123.31.67:5062>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 106 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=14386
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK095973c8;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as722e49c6
To: sip:1@206.123.31.104;tag=USQlBoW6ja1xFGxQcz7sbD5xr7wG86zr
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 4N7y3f.GJNJeblJsCZnnzfA4FkvXQYyq
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 293

v=0
o=root 1489242479 1489242481 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 13984 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
206.123.31.104:5060;rport=5060;received=206.123.31.104;branch=z9hG4bK095973c8
Call-ID: 4N7y3f.GJNJeblJsCZnnzfA4FkvXQYyq
From: <sip:2@206.123.31.104>;tag=as722e49c6
To: <sip:1@206.123.31.104>;tag=USQlBoW6ja1xFGxQcz7sbD5xr7wG86zr
CSeq: 103 INVITE
Contact: <sip:1@206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485797608 3485797610 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4022 RTP/AVP 0 101
a=rtcp:4023 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK3453797a
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
To: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
CSeq: 106 NOTIFY
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:1@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5062;rport;branch=z9hG4bKPjscCPRmu2WyAQ7GP3f6-
twj2r3Rc5Jsoy
Max-Forwards: 70
From: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
To: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 14387 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
ACK sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK7dcb2036;rport
Max-Forwards: 70
From: sip:2@206.123.31.104;tag=as722e49c6
To: sip:1@206.123.31.104;tag=USQlBoW6ja1xFGxQcz7sbD5xr7wG86zr
Contact: <sip:2@206.123.31.104:5060>
Call-ID: 4N7y3f.GJNJeblJsCZnnzfA4FkvXQYyq
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP 206.123.31.67:5062;rport;branch=z9hG4bKPjscCPRmu2WyAQ7GP3f6-
twj2r3Rc5Jsoy;received=206.123.31.67
From: <sip:2@206.123.31.67>;tag=e0sefxiwJmUFcOuz8baiBxA-AkudhOxx
To: "1" <sip:1@206.123.31.104>;tag=as2dfa177b
Call-ID: 0baa55b70b2f92da0c685edb32d2ae42@206.123.31.104:5060
CSeq: 14387 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK444282f1
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as503331c9
To: <sip:4@[2620:0:230:c000::67]:5064>
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 443050b6205e0315525892f7391ce972@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 13:31:01 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 332

v=0
o=root 1875550393 1875550393 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 9786 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK444282f1
Call-ID: 443050b6205e0315525892f7391ce972@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as503331c9
To: <sip:4@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 206.123.31.67:5061;branch=z9hG4bKPjkK4dM3ESLEbD9i.yw15g-
aFbr5pyZ3fp;received=206.123.31.67;rport=5061
From: sip:1@206.123.31.104;tag=2kzz6lhbpnNuzjF8Xd4RVajXhOG0COzh
To: sip:2@206.123.31.104;tag=as60b9e00e
Call-ID: nxJXYSuSbBLE9aqh0fXqCmlQoS7QbvGh
CSeq: 18180 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:2@206.123.31.104:5060>
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK444282f1
Call-ID: 443050b6205e0315525892f7391ce972@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as503331c9
To: <sip:4@[2620:0:230:c000::67]>;tag=Zq8c5hm5dVCpalIJ1H9HsrSrut4NUtDk
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485856661 3485856662 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK271ff4c6
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as503331c9
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=Zq8c5hm5dVCpalIJ1H9HsrSrut4NUtDk
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 443050b6205e0315525892f7391ce972@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was successfully transferred to UA4. Asterisk did not send a re-INVITE and is relaying the media between UA1 (IPv4) and UA4 (IPv6). The test passed.

3.4. IPv6-to-IPv6

Test description:

> UA4 dials extension 5. Asterisk relays to UA5, then issues a re-INVITE so that IPv6 RTP flows directly.

Results:

```
INVITE sip:5@[2620:0:230:c000:216:cbff:fe92:2504] SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjRmUwB9whwQsSW3Shm33lGLfRougbk-bn
Max-Forwards: 70
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
To: sip:5@[2620:0:230:c000:216:cbff:fe92:2504]
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 18567 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Type: application/sdp
Content-Length:   480

v=0
o=- 3485858577 3485858577 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 103 102 104 109 3 0 8 9 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:103 speex/16000
a=rtpmap:102 speex/8000
a=rtpmap:104 speex/32000
a=rtpmap:109 iLBC/8000
a=fmtp:109 mode=30
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;branch=z9hG4bKPjRmUwB9whwQsSW3Shm33lGLfRougbk-
bn;received=2620:0:230:c000::67;rport=5064
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
To: sip:5@[2620:0:230:c000:216:cbff:fe92:2504]
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 18567 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
INVITE sip:5@[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK6b55dcee
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:02:57 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 333

v=0
o=root 1842960611 1842960611 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 29244 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK6b55dcee
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK6b55dcee
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485858577 3485858578 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4010 RTP/AVP 0 101
a=rtcp:4011 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK422b80ac
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;branch=z9hG4bKPjRmUwB9whwQsSW3Shm33lGLfRougbk-
bn;received=2620:0:230:c000::67;rport=5064
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
To: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 18567 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Type: application/sdp
Content-Length: 331

v=0
o=root 628066661 628066661 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 27254 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPj4rI9atA7r.hh9sRduFbdVGYS270mMEGB
Max-Forwards: 70
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
To: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 18567 ACK
Content-Length:  0
```

```
INVITE sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK555ddaac
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 302

v=0
o=root 1842960611 1842960612 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4090 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK555ddaac
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
CSeq: 103 INVITE
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485858577 3485858579 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4010 RTP/AVP 0 101
a=rtcp:4011 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK4fcd9bc0;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 300

v=0
o=root 628066661 628066662 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4010 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;rport=5060;received=2620:0:230:c000:216:cbff:f
e92:2504;branch=z9hG4bK4fcd9bc0
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
From: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7a562e25
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
CSeq: 102 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485858577 3485858578 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK3274f8d0
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
Contact:
```

```
 <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK2f202ace;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was successfully established. Asterisk sent a re-INVITE and the RTP media is flowing directly between UA4 and UA5, over IPv6. The test passed.

3.4.1. 666 Transfer

Test description:

UA5 transfers the IPv6-IPv6 call from UA4 to extension 6 over IPv6.

Results:

```
REFER sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjL9a6RUeipVFTqMvK9c5RvsnEYWngM1Yw
Max-Forwards: 70
From: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
Contact: <sip:[2620:0:230:c000::67]:5065>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 30464 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:6@[2620:0:230:c000:216:cbff:fe92:2504]
Referred-By: <sip:5@[2620:0:230:c000::67]>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjL9a6RUeipVFTqMvK9c5RvsnEYWngM1Yw;receiv
ed=2620:0:230:c000::67
From: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 30464 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK1b656c8c
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 105 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=30464
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK1b656c8c
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
CSeq: 105 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK30efadc7
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 106 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=30464
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK30efadc7
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
To: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
CSeq: 106 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK491dbd2d;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 331

v=0
o=root 628066661 628066663 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 27254 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
BYE sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjoZ03RS4SYP.3G9iZEPy.3bNOjrElMAwR
Max-Forwards: 70
From: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 30465 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;rport=5060;received=2620:0:230:c000:216:cbff:f
e92:2504;branch=z9hG4bK491dbd2d
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
From: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7a562e25
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
CSeq: 103 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485858577 3485858579 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjoZ03RS4SYP.3G9iZEPy.3bNOjrElMAwR;receiv
ed=2620:0:230:c000::67
From: <sip:5@[2620:0:230:c000::67]>;tag=xwxEso79zUtlo3AcScHFxGoL.DYqtFU7
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as02a23c08
Call-ID: 426bb00d7c3058eb611181ab57bdb25a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 30465 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:6@[2620:0:230:c000::67]:5066 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK6fcd2863
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]:5066>
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:14:26 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 331

v=0
o=root 325551124 325551124 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 18032 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK6fcd2863
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
ACK sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK3330b84c;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK6fcd2863
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]>;tag=KIBHx1RhswZnqsp5Dlq2WhdNK6i48ZaW
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5066>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485859266 3485859267 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4050 RTP/AVP 0 101
a=rtcp:4051 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5066 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK52e5c624
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]:5066>;tag=KIBHx1RhswZnqsp5Dlq2WhdNK6i48ZaW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK51894ce9;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 104 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 300

v=0
o=root 628066661 628066664 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4050 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
INVITE sip:[2620:0:230:c000::67]:5066 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK76e077f6
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]:5066>;tag=KIBHx1RhswZnqsp5Dlq2WhdNK6i48ZaW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 300

v=0
o=root 325551124 325551125 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4090 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;rport=5060;received=2620:0:230:c000:216:cbff:f
e92:2504;branch=z9hG4bK51894ce9
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
From: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7a562e25
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
CSeq: 104 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485858577 3485858580 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK76e077f6
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]>;tag=KIBHx1RhswZnqsp5Dlq2WhdNK6i48ZaW
CSeq: 103 INVITE
Contact: <sip:[2620:0:230:c000::67]:5066>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485859266 3485859268 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4050 RTP/AVP 0 101
a=rtcp:4051 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK54bfa607;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7a562e25
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=H.c4j3lTeq4IQQZtf8B0R5f-9vHvEp5N
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 6wD9LAxwPLU551D6pn089hMgxHEDtZaz
CSeq: 104 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:[2620:0:230:c000::67]:5066 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK50df423a
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3ffe4f58
To: <sip:6@[2620:0:230:c000::67]:5066>;tag=KIBHx1RhswZnqsp5Dlq2WhdNK6i48ZaW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 7e102e6d632550c04a4754e903e13c8d@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was transferred successfully. Asterisk sent a re-INVITE and the RTP media is flowing directly between UA4 and UA6 over IPv6. The test passed.

3.4.2. 664 Transfer

Test description:

UA5 transfers the IPv6-IPv6 call from UA4 to to extension 1 over IPv4.

Results:

```
REFER sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjwom4vXjJbjEtiPOgFd-
Gund.44vZhXCd
Max-Forwards: 70
From: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
Contact: <sip:[2620:0:230:c000::67]:5065>
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 4465 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:1@206.123.31.104
Referred-By: <sip:5@[2620:0:230:c000::67]>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP [2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjwom4vXjJbjEtiPOgFd-
Gund.44vZhXCd;received=2620:0:230:c000::67
From: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 4465 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK787f2f06
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 105 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=4465
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK787f2f06
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
To: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
CSeq: 105 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK0594eaf3
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 106 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=4465
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK0594eaf3
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
To: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
CSeq: 106 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK387da7e9;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7afd5e91
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=yUr3P7hQix7eZJkqxT7.-PpxTZJbr6ER
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 9rXXq7-Ox9U0ZP8YF40i3hXsy0Gp.FwG
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 331

v=0
o=root 508106258 508106260 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 30470 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
BYE sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjyDN5YunN8kusbgj2zUWSbz0yUSG7J4Sa
Max-Forwards: 70
From: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 4466 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;rport=5060;received=2620:0:230:c000:216:cbff:f
e92:2504;branch=z9hG4bK387da7e9
Call-ID: 9rXXq7-Ox9U0ZP8YF40i3hXsy0Gp.FwG
From: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7afd5e91
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=yUr3P7hQix7eZJkqxT7.-PpxTZJbr6ER
CSeq: 103 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485860656 3485860658 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4120 RTP/AVP 0 101
a=rtcp:4121 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5065;rport;branch=z9hG4bKPjyDN5YunN8kusbgj2zUWSbz0yUSG7J4Sa;receiv
ed=2620:0:230:c000::67
From: <sip:5@[2620:0:230:c000::67]>;tag=LMGwBen1RYDNahmwaemmfyqRFf38KDNd
To: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as3f5f2b6b
Call-ID: 5c936cb726f5d25a0957c34f63db5216@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 4466 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK1b267085
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as27426839
To: <sip:1@206.123.31.67:5061>
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 391d2ea9159b3db832411b9e36be5cdc@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:37:45 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 293

v=0
o=root 1811197849 1811197849 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 26100 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK5b0b1e65;rport
Max-Forwards: 70
From: sip:5@[2620:0:230:c000:216:cbff:fe92:2504];tag=as7afd5e91
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=yUr3P7hQix7eZJkqxT7.-PpxTZJbr6ER
Contact: <sip:5@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 9rXXq7-Ox9U0ZP8YF40i3hXsy0Gp.FwG
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK1b267085
Call-ID: 391d2ea9159b3db832411b9e36be5cdc@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as27426839
To: <sip:1@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK1b267085
Call-ID: 391d2ea9159b3db832411b9e36be5cdc@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as27426839
To: <sip:1@206.123.31.67>;tag=PX-NMvrlIrxOrYk-fyuVhjYFGopWv52M
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485860665 3485860666 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4018 RTP/AVP 0 101
a=rtcp:4019 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK73a00942
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as27426839
To: <sip:1@206.123.31.67:5061>;tag=PX-NMvrlIrxOrYk-fyuVhjYFGopWv52M
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 391d2ea9159b3db832411b9e36be5cdc@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was transferred successfully. Asterisk did not send a re-INVITE and is relaying the media between UA4 (IPv6) and UA1 (IPv4). The test passed.

3.5. IPv4-to-IPv6

Test description:

> UA1 dials extension 4. Asterisk relays to UA4. There is no re-INVITE. Asterisk converts RTP from IPv4 to IPv6.

Results:

```
INVITE sip:4@206.123.31.104 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5061;rport;branch=z9hG4bKPj86HshUQRvvibLlAMPPsyo4iqZ-
y3uh9Z
Max-Forwards: 70
From: sip:1@206.123.31.104;tag=EQ2g2LB3ivVhMmyIvA.6xHUPwouTq7Ks
To: sip:4@206.123.31.104
Contact: <sip:1@206.123.31.67:5061>
Call-ID: 6itKeY8W4Hvas3bLpvrfjwmk0AFadz-0
CSeq: 12295 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Type: application/sdp
Content-Length:   462

v=0
o=- 3485861016 3485861016 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4024 RTP/AVP 103 102 104 109 3 0 8 9 101
a=rtcp:4025 IN IP4 206.123.31.67
a=rtpmap:103 speex/16000
a=rtpmap:102 speex/8000
a=rtpmap:104 speex/32000
a=rtpmap:109 iLBC/8000
a=fmtp:109 mode=30
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.67:5061;branch=z9hG4bKPj86HshUQRvvibLlAMPPsyo4iqZ-
y3uh9Z;received=206.123.31.67;rport=5061
From: sip:1@206.123.31.104;tag=EQ2g2LB3ivVhMmyIvA.6xHUPwouTq7Ks
To: sip:4@206.123.31.104
Call-ID: 6itKeY8W4Hvas3bLpvrfjwmk0AFadz-0
CSeq: 12295 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:4@206.123.31.104:5060>
Content-Length: 0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK28207214
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as15452aa2
To: <sip:4@[2620:0:230:c000::67]:5064>
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 4ea2f619607406e4746eaa2e40e7ff06@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:43:36 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 331

v=0
o=root 174821655 174821655 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 25374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK28207214
Call-ID: 4ea2f619607406e4746eaa2e40e7ff06@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as15452aa2
To: <sip:4@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK28207214
Call-ID: 4ea2f619607406e4746eaa2e40e7ff06@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as15452aa2
To: <sip:4@[2620:0:230:c000::67]>;tag=Tw-G1f735hbbfyIBtPiblcfbCGnkDFj1
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485861016 3485861017 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK1b293260
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as15452aa2
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=Tw-G1f735hbbfyIBtPiblcfbCGnkDFj1
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 4ea2f619607406e4746eaa2e40e7ff06@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.67:5061;branch=z9hG4bKPj86HshUQRvvibLlAMPPsyo4iqZ-
y3uh9Z;received=206.123.31.67;rport=5061
From: sip:1@206.123.31.104;tag=EQ2g2LB3ivVhMmyIvA.6xHUPwouTq7Ks
To: sip:4@206.123.31.104;tag=as18b5f524
Call-ID: 6itKeY8W4Hvas3bLpvrfjwmk0AFadz-0
CSeq: 12295 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:4@206.123.31.104:5060>
Content-Type: application/sdp
Content-Length: 293

v=0
o=root 2062179349 2062179349 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 10814 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:4@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5061;rport;branch=z9hG4bKPjd4FN1bw2-BeP-8L6q0mGI4V8-
EwDyZLh
Max-Forwards: 70
From: sip:1@206.123.31.104;tag=EQ2g2LB3ivVhMmyIvA.6xHUPwouTq7Ks
To: sip:4@206.123.31.104;tag=as18b5f524
Call-ID: 6itKeY8W4Hvas3bLpvrfjwmk0AFadz-0
CSeq: 12295 ACK
Content-Length:  0
```

The call was successfully established. Asterisk did not send a re-INVITE and is relaying the media between UA1 (IPv4) and UA4 (IPv6). The test passed.

3.5.1. 464 Transfer

Test description:

UA4 transfers the IPv4-IPv6 call from UA1 to extension 2 over IPv4.

Results:

```
REFER sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjxukBV3zITVvspwtW9bAMCmqASCZb0A9q
Max-Forwards: 70
From: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
Contact: <sip:[2620:0:230:c000::67]:5064>
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 20504 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:2@[2620:0:230:c000:216:cbff:fe92:2504]
Referred-By: <sip:4@[2620:0:230:c000::67]>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjxukBV3zITVvspwtW9bAMCmqASCZb0A9q;receiv
ed=2620:0:230:c000::67
From: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 20504 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK4ac1957c
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=20504
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK4ac1957c
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
To: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
CSeq: 103 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK230f3418
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 104 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=20504
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK230f3418
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
To: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
CSeq: 104 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjYTElPk6ePcZProQzPjj3ioiB7T34r-ZH
Max-Forwards: 70
From: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 20505 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
INVITE sip:2@206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK05860745
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67:5062>
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:47:26 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 292

v=0
o=root 1619468355 1619468355 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 6602 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjYTElPk6ePcZProQzPjj3ioiB7T34r-
ZH;received=2620:0:230:c000::67
From: <sip:4@[2620:0:230:c000::67]>;tag=WpXQjEsJJrk79eFpDn31jDA7RrGnTU2F
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as679871e5
Call-ID: 37d324e1601fe9a22f5ef7666c5a1a47@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 20505 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK05860745
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK05860745
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67>;tag=B4.34y8ePw79-XAcK6kTeF1Q12yCG8Gj
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485861246 3485861247 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4002 RTP/AVP 0 101
a=rtcp:4003 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK5e16b9bd
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67:5062>;tag=B4.34y8ePw79-XAcK6kTeF1Q12yCG8Gj
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK15bdf40a;rport
Max-Forwards: 70
From: sip:4@206.123.31.104;tag=as477958e8
To: sip:1@206.123.31.104;tag=GaOuf6J0dlPtoNp3VZ5I02LPWmwhSpnC
Contact: <sip:4@206.123.31.104:5060>
Call-ID: zQ4vmRwrKtx2rzu-jmFASVBpR9p1ItOV
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 290

v=0
o=root 1917552604 1917552605 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4002 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
INVITE sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK57228893
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67:5062>;tag=B4.34y8ePw79-XAcK6kTeF1Q12yCG8Gj
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 290

v=0
o=root 1619468355 1619468356 IN IP4 206.123.31.67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.67
t=0 0
m=audio 4026 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
206.123.31.104:5060;rport=5060;received=206.123.31.104;branch=z9hG4bK15bdf40a
Call-ID: zQ4vmRwrKtx2rzu-jmFASVBpR9p1ItOV
From: <sip:4@206.123.31.104>;tag=as477958e8
To: <sip:1@206.123.31.104>;tag=GaOuf6J0dlPtoNp3VZ5I02LPWmwhSpnC
CSeq: 102 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:1@206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485861218 3485861219 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4026 RTP/AVP 0 101
a=rtcp:4027 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK57228893
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67>;tag=B4.34y8ePw79-XAcK6kTeF1Q12yCG8Gj
CSeq: 103 INVITE
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485861246 3485861248 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4002 RTP/AVP 0 101
a=rtcp:4003 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK15be8e53;rport
Max-Forwards: 70
From: sip:4@206.123.31.104;tag=as477958e8
To: sip:1@206.123.31.104;tag=GaOuf6J0dlPtoNp3VZ5I02LPWmwhSpnC
Contact: <sip:4@206.123.31.104:5060>
Call-ID: zQ4vmRwrKtx2rzu-jmFASVBpR9p1ItOV
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK685d0bd7
Max-Forwards: 70
From: "1" <sip:1@206.123.31.104>;tag=as64494f36
To: <sip:2@206.123.31.67:5062>;tag=B4.34y8ePw79-XAcK6kTeF1Q12yCG8Gj
Contact: <sip:1@206.123.31.104:5060>
Call-ID: 1d9e8f75626622290d3bac295a2dc894@206.123.31.104:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was transferred successfully. Asterisk sent a re-INVITE and the media is flowing directly between UA1 and UA2 over IPv4.

3.5.2. 466 Transfer

Test description:

UA4 transfers the IPv4-IPv6 call from UA1 to extension 5 over IPv6.

Results:

```
REFER sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPj5zE97E3YmZCxv7LZlstlBdfZI5av5rMo
Max-Forwards: 70
From: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
Contact: <sip:[2620:0:230:c000::67]:5064>
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 12627 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:5@[2620:0:230:c000:216:cbff:fe92:2504]
Referred-By: <sip:4@[2620:0:230:c000::67]>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPj5zE97E3YmZCxv7LZlstlBdfZI5av5rMo;receiv
ed=2620:0:230:c000::67
From: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 12627 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK1298cba5
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=12627
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK1298cba5
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
To: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
CSeq: 103 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK7392ae5a
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
To: <sip:4@[2620:0:230:c000::67]:5064>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 104 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=12627
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK7392ae5a
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
To: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
CSeq: 104 NOTIFY
Contact: <sip:[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjDT0RrcoR-
0Y6lFWIf76ocSy1LMYOvFwy
Max-Forwards: 70
From: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 12628 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP [2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjDT0RrcoR-
0Y6lFWIf76ocSy1LMYOvFwy;received=2620:0:230:c000::67
From: <sip:4@[2620:0:230:c000::67]>;tag=e8zmQ1qbmEOE.-kyZP-5D7luUo5GYDdh
To: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as1c325238
Call-ID: 4dbed9b1729757f96e64eff43a66ec3a@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 12628 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:5@[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK1ec5c2a4
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as562d1e3c
To: <sip:5@[2620:0:230:c000::67]:5065>
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 79a50cde417879c122aad9a3207e7cfa@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 14:52:12 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 332

v=0
o=root 1616274050 1616274050 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 9976 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK1ec5c2a4
Call-ID: 79a50cde417879c122aad9a3207e7cfa@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as562d1e3c
To: <sip:5@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK1ec5c2a4
Call-ID: 79a50cde417879c122aad9a3207e7cfa@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as562d1e3c
To: <sip:5@[2620:0:230:c000::67]>;tag=eiLLgKzBnKZf5ZPWoeS32RNriDnJR5hp
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485861532 3485861533 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4010 RTP/AVP 0 101
a=rtcp:4011 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK77ecf1eb
Max-Forwards: 70
From: "1" <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as562d1e3c
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=eiLLgKzBnKZf5ZPWoeS32RNriDnJR5hp
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 79a50cde417879c122aad9a3207e7cfa@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was transferred successfully. Asterisk did not send a re-INVITE and is relaying the media between UA1 (IPv4) and UA5 (IPv6). The test passed.

3.6. IPv6-to-IPv4

Test description:

> UA4 dials extension 1. Asterisk relays to UA1. There is no re-INVITE. Asterisk converts RTP from IPv6 to IPv4.

Results:

```
INVITE sip:1@[2620:0:230:c000:216:cbff:fe92:2504] SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjkrkMy6c0RLt.Rv9DPK1GGy7HKlHLpOKU
Max-Forwards: 70
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
To: sip:1@[2620:0:230:c000:216:cbff:fe92:2504]
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 12559 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Type: application/sdp
Content-Length:   480

v=0
o=- 3485862250 3485862250 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 103 102 104 109 3 0 8 9 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:103 speex/16000
a=rtpmap:102 speex/8000
a=rtpmap:104 speex/32000
a=rtpmap:109 iLBC/8000
a=fmtp:109 mode=30
a=rtpmap:3 GSM/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;branch=z9hG4bKPjkrkMy6c0RLt.Rv9DPK1GGy7HKlHLpOKU;received=262
0:0:230:c000::67;rport=5064
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
To: sip:1@[2620:0:230:c000:216:cbff:fe92:2504]
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 12559 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Length: 0
```

```
INVITE sip:1@206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK0464801a
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67:5061>
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 15:04:10 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 293

v=0
o=root 1910473687 1910473687 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 22484 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK0464801a
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK0464801a
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485862250 3485862251 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4022 RTP/AVP 0 101
a=rtcp:4023 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK3b0c0851
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67:5061>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;branch=z9hG4bKPjkrkMy6c0RLt.Rv9DPK1GGy7HKlHLpOKU;received=262
0:0:230:c000::67;rport=5064
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
To: sip:1@[2620:0:230:c000:216:cbff:fe92:2504];tag=as6547f6c2
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 12559 INVITE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Require: timer
Session-Expires: 1800;refresher=uas
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Content-Type: application/sdp
Content-Length: 333

v=0
o=root 1249917348 1249917348 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 11480 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
ACK sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060 SIP/2.0
Via: SIP/2.0/UDP
[2620:0:230:c000::67]:5064;rport;branch=z9hG4bKPjyoElTlYiBzRClenDoKaHZ4GFS0AinTE5
Max-Forwards: 70
From: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
To: sip:1@[2620:0:230:c000:216:cbff:fe92:2504];tag=as6547f6c2
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 12559 ACK
Content-Length:  0
```

The call was successfully established. There was no re-INVITE and Asterisk is relaying media between UA4 (IPv6) and UA1 (IPv4). The test passed.

3.6.1. 646 Transfer

Test description:

UA1 transfers the IPv6-IPv4 call from UA4 to extension 5 over IPv6.

Results:

```
REFER sip:4@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjJ3sPC9uTFh22Jzmn0veqgXpK.BCN7JcV
Max-Forwards: 70
From: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
To: "4" <sip:4@206.123.31.104>;tag=as6c418b38
Contact: <sip:206.123.31.67:5061>
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 15995 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:5@206.123.31.104
Referred-By: <sip:1@206.123.31.67>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjJ3sPC9uTFh22Jzmn0veqgXpK.BCN7JcV;received=206.1
23.31.67
From: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
To: "4" <sip:4@206.123.31.104>;tag=as6c418b38
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 15995 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:4@206.123.31.104:5060>
Content-Length: 0
```

```
NOTIFY sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK28b1fdbc
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67:5061>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 103 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=15995
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK28b1fdbc
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
CSeq: 103 NOTIFY
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK04a4fdf3
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67:5061>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 104 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=15995
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK04a4fdf3
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as6c418b38
To: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
CSeq: 104 NOTIFY
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:4@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjd8Pecjejzvk75MCta.VbceUNTnu0Kz6-
Max-Forwards: 70
From: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
To: "4" <sip:4@206.123.31.104>;tag=as6c418b38
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 15996 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjd8Pecjejzvk75MCta.VbceUNTnu0Kz6-;received=206.1
23.31.67
From: <sip:1@206.123.31.67>;tag=waZKu3YULkUuZP6HeM323gwKvSrEhnXB
To: "4" <sip:4@206.123.31.104>;tag=as6c418b38
Call-ID: 2abbc5ed2f1416e47c9067e20ea01974@206.123.31.104:5060
CSeq: 15996 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:5@[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK14f5aa8f
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]:5065>
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 15:09:26 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 330

v=0
o=root 737268360 737268360 IN IP6 2620:0:230:c000:216:cbff:fe92:2504
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000:216:cbff:fe92:2504
t=0 0
m=audio 7204 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK14f5aa8f
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK14f5aa8f
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]>;tag=ba3CaiDPfEJp.qvmPQ-iyCQMGhS79oyW
CSeq: 102 INVITE
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485862566 3485862567 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4020 RTP/AVP 0 101
a=rtcp:4021 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK2adcd944
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=ba3CaiDPfEJp.qvmPQ-iyCQMGhS79oyW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
INVITE sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK69cf454b;rport
Max-Forwards: 70
From: sip:1@[2620:0:230:c000:216:cbff:fe92:2504];tag=as6547f6c2
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Require: timer
Session-Expires: 1800;refresher=uas
Min-SE: 90
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 302

v=0
o=root 1249917348 1249917349 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4020 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
INVITE sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK6a17cb6a
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=ba3CaiDPfEJp.qvmPQ-iyCQMGhS79oyW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 300

v=0
o=root 737268360 737268361 IN IP6 2620:0:230:c000::67
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP6 2620:0:230:c000::67
t=0 0
m=audio 4090 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;rport=5060;received=2620:0:230:c000:216:cbff:f
e92:2504;branch=z9hG4bK69cf454b
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
From: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as6547f6c2
To: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
CSeq: 102 INVITE
Session-Expires: 1800;refresher=uas
Contact: <sip:4@[2620:0:230:c000::67]:5064>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485862250 3485862251 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4090 RTP/AVP 0 101
a=rtcp:4091 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP
[2620:0:230:c000:216:cbff:fe92:2504]:5060;received=2620:0:230:c000:216:cbff:fe92:2504;br
anch=z9hG4bK6a17cb6a
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]>;tag=ba3CaiDPfEJp.qvmPQ-iyCQMGhS79oyW
CSeq: 103 INVITE
Contact: <sip:[2620:0:230:c000::67]:5065>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   272

v=0
o=- 3485862566 3485862568 IN IP6 2620:0:230:c000::67
s=pjmedia
c=IN IP6 2620:0:230:c000::67
t=0 0
a=X-nat:0
m=audio 4020 RTP/AVP 0 101
a=rtcp:4021 IN IP6 2620:0:230:c000::67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:4@[2620:0:230:c000::67]:5064 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK724e9fd7;rport
Max-Forwards: 70
From: sip:1@[2620:0:230:c000:216:cbff:fe92:2504];tag=as6547f6c2
To: sip:4@[2620:0:230:c000:216:cbff:fe92:2504];tag=ytNuAsLO2aZcMFlsVs9c6hyN.gURrqPM
Contact: <sip:1@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: N06PQgKIp0bgoQDtV.c.t8q5Cso9-D8r
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

```
ACK sip:[2620:0:230:c000::67]:5065 SIP/2.0
Via: SIP/2.0/UDP [2620:0:230:c000:216:cbff:fe92:2504]:5060;branch=z9hG4bK22dd22da
Max-Forwards: 70
From: "4" <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]>;tag=as7cc12036
To: <sip:5@[2620:0:230:c000::67]:5065>;tag=ba3CaiDPfEJp.qvmPQ-iyCQMGhS79oyW
Contact: <sip:4@[2620:0:230:c000:216:cbff:fe92:2504]:5060>
Call-ID: 75213cac5db834477aa083c27f3c03c5@[2620:0:230:c000:216:cbff:fe92:2504]:5060
CSeq: 103 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was successfully transferred. Asterisk sent a re-INVITE and the media is flowing directly between UA4 and UA5 over IPv6.

3.6.2. 644 Transfer

Test description:

UA1 transfers the IPv6-IPv4 call from UA4 to extension 2 over IPv4.

Results:

```
REFER sip:4@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.67:5061;rport;branch=z9hG4bKPjpCAIEtPksSv81bQ9r2CmyX-
sA3MZgiWd
Max-Forwards: 70
From: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
To: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
Contact: <sip:206.123.31.67:5061>
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 23797 REFER
Event: refer
Expires: 600
Accept: message/sipfrag;version=2.0
Allow-Events: presence, message-summary, refer
Refer-To: sip:2@206.123.31.104
Referred-By: <sip:1@206.123.31.67>
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 206.123.31.67:5061;rport;branch=z9hG4bKPjpCAIEtPksSv81bQ9r2CmyX-
sA3MZgiWd;received=206.123.31.67
From: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
To: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 23797 REFER
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Contact: <sip:4@206.123.31.104:5060>
Content-Length: 0
```

```
NOTIFY sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK619de2ac
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
To: <sip:1@206.123.31.67:5061>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 103 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=23797
Subscription-state: active
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 21

SIP/2.0 183 Ringing
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK619de2ac
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
To: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
CSeq: 103 NOTIFY
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
NOTIFY sip:206.123.31.67:5061 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK31f54d26
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
To: <sip:1@206.123.31.67:5061>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 104 NOTIFY
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Event: refer;id=23797
Subscription-state: terminated;reason=noresource
Content-Type: message/sipfrag;version=2.0
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 16

SIP/2.0 200 Ok
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK31f54d26
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
To: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
CSeq: 104 NOTIFY
Contact: <sip:206.123.31.67:5061>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length:  0
```

```
BYE sip:4@206.123.31.104:5060 SIP/2.0
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjvBqCXFTn3JM9BrRwp8kaE0wrpFvmPDvE
Max-Forwards: 70
From: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
To: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 23798 BYE
User-Agent: PJSUA v1.6-trunk/x86_64-unknown-linux-gnu
Content-Length:  0
```

```
SIP/2.0 481 Call leg/transaction does not exist
Via: SIP/2.0/UDP
206.123.31.67:5061;rport;branch=z9hG4bKPjvBqCXFTn3JM9BrRwp8kaE0wrpFvmPDvE;received=206.1
23.31.67
From: <sip:1@206.123.31.67>;tag=1Lk.MKoq2A35eFpM76HyeDLmbaRfc6xk
To: "4" <sip:4@206.123.31.104>;tag=as1a8456ed
Call-ID: 27c8be9e7014f2f1725a612f1bb7faf4@206.123.31.104:5060
CSeq: 23798 BYE
Server: Asterisk PBX UNKNOWN__and_probably_unsupported
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Length: 0
```

```
INVITE sip:2@206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK6ab949d1
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as364a141d
To: <sip:2@206.123.31.67:5062>
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 3671f1426063933616aae7476dd12dbd@206.123.31.104:5060
CSeq: 102 INVITE
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Date: Fri, 18 Jun 2010 15:19:03 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH
Supported: replaces, timer
Content-Type: application/sdp
Content-Length: 291

v=0
o=root 617328266 617328266 IN IP4 206.123.31.104
s=Asterisk PBX UNKNOWN__and_probably_unsupported
c=IN IP4 206.123.31.104
t=0 0
m=audio 25860 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=ptime:20
a=sendrecv
```

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK6ab949d1
Call-ID: 3671f1426063933616aae7476dd12dbd@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as364a141d
To: <sip:2@206.123.31.67>
CSeq: 102 INVITE
Content-Length:  0
```

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 206.123.31.104:5060;received=206.123.31.104;branch=z9hG4bK6ab949d1
Call-ID: 3671f1426063933616aae7476dd12dbd@206.123.31.104:5060
From: "4" <sip:4@206.123.31.104>;tag=as364a141d
To: <sip:2@206.123.31.67>;tag=58MJZwRek-lEllziFS9ka7EP6G79PC0Y
CSeq: 102 INVITE
Contact: <sip:206.123.31.67:5062>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE,
OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Type: application/sdp
Content-Length:   254

v=0
o=- 3485863143 3485863144 IN IP4 206.123.31.67
s=pjmedia
c=IN IP4 206.123.31.67
t=0 0
a=X-nat:0
m=audio 4004 RTP/AVP 0 101
a=rtcp:4005 IN IP4 206.123.31.67
a=rtpmap:0 PCMU/8000
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

```
ACK sip:206.123.31.67:5062 SIP/2.0
Via: SIP/2.0/UDP 206.123.31.104:5060;branch=z9hG4bK2b2d2695
Max-Forwards: 70
From: "4" <sip:4@206.123.31.104>;tag=as364a141d
To: <sip:2@206.123.31.67:5062>;tag=58MJZwRek-lEllziFS9ka7EP6G79PC0Y
Contact: <sip:4@206.123.31.104:5060>
Call-ID: 3671f1426063933616aae7476dd12dbd@206.123.31.104:5060
CSeq: 102 ACK
User-Agent: Asterisk PBX UNKNOWN__and_probably_unsupported
Content-Length: 0
```

The call was transferred successfully. There was no re-INVITE and Asterisk is relaying the media between UA4 (IPv6) and UA2 (IPv4). The test passed.

4. Performance Testing

4.1. REGISTER Performance

Test description:

> From UA4, send REGISTER requests to Asterisk. Measure the maximum rate at which Asterisk can process REGISTER requests over IPv6. Compare with UA1 registering to an unmodified Asterisk over IPv4.

For testing over IPv4, we are using the following command line:

```
$ sipp -s 1 -sf register.xml -l 10 -r 1000 -m 1000 a.b.c.d
```

where "a.b.c.d" is the IPv4 address of the Asterisk server.

For testing over IPv6, we are using the following command line:

```
$ sipp -s 4 -sf register.xml -l 10 -r 1000 -m 1000 -i \[abc::def\]
\[xyz::123\]
```

where "abc::def" is the IPv6 address of the tester system and "xyz::123" is the IPv6 address of the Asterisk server. The register.xml file is in Appendix A.

This test generates 1000 REGISTER requests at a rate of up to 1000 per second. We impose a limit of 10 simultaneous requests to ensure that we do not drop packets and create retransmissions.

Results:

Code base	Protocol	Requests per second	Relative to trunk
trunk	IPv4	23.5	100 %
v6-new	IPv4	24.3	103.4 %
v6-new	IPv6	24.7	105.3 %

Since the new code's performance is within 10% of trunk, this test passed.

4.2. RTP Relaying Performance

Test description:

> From UA4, establish 96 concurrent IPv6 calls to extension 5, with RTP being relayed by Asterisk. Compare CPU and memory usage with an unmodified Asterisk and UA1 dialing extension 2 over IPv4.

For testing over IPv4, we are using the following command lines:

```
# sipp -sn uas -rtp_echo -i a.b.c.d -p 5061
# sipp -sf uac_pcap.xml -s 2 -m 96 -r 100 -i a.b.c.d x.y.z.w
```

where "a.b.c.d" is the IPv4 address of the tester system and "x.y.z.w" is the IPv4 address of the system running Asterisk. The "uac_pcap.xml" scenario file is in Appendix A.

For testing over IPv6, we are using the following command lines:

```
# sipp -sn uas -rtp_echo -i \[abc::def\] -p 5061
# sipp -sf uac_pcap.xml -s 4 -m 96 -r 100 -i \[abc::def\] \[xyz::123\]
```

where "abc::def" is the IPv6 address of the tester system and "xyz::123" is the IPv6 address of the system running Asterisk.

Results:

Code base	Protocol	CPU usage		Memory usage	
		Absolute	**Relative to trunk**	**Absolute**	**Relative to trunk**
trunk	IPv4	18.83 s	100 %	67,520 kB	100 %
v6-new	IPv4	19.56 s	103.9 %	70,736 kB	104.8 %
v6-new	IPv6	19.93 s	105.8 %	70,784 kB	104.8 %

Since all relative CPU and memory usage figures are within 10% of trunk, the test passed.

5. Summary

The following table summarizes the tests that have been run and their results.

Section	Test name	Result
3.1.	IPv4 REGISTER	PASS
3.2.	IPv6 REGISTER	PASS
3.3.	IPv4-to-IPv4	PASS
3.4.	IPv6-to-IPv6	PASS
3.5.	IPv4-to-IPv6	PASS
3.6.	IPv6-to-IPv4	PASS
3.3.1.	444 Transfer	PASS
3.3.2.	446 Transfer	PASS
3.4.1.	666 Transfer	PASS
3.4.2.	664 Transfer	PASS
3.5.1.	464 Transfer	PASS
3.5.2.	466 Transfer	PASS
3.6.1.	646 Transfer	PASS
3.6.2.	644 Transfer	PASS
4.1.	REGISTER Performance	PASS: Less than 10% difference with IPv4
4.2.	RTP Relaying Performance	PASS:Less than 10% difference with IPv4

6. Conclusion

This test report shows the tests conducted on the Asterisk IPv6 port. All instances of the test plan passed according to specs.

Appendix A

Contents of the *register.xml* sipp scenario file used for performance testing (see section 4.1.):

```
<?xml version="1.0" encoding="ISO-8859-1" ?>
<!DOCTYPE scenario SYSTEM "sipp.dtd">

<scenario name="Simple Register">
  <send retrans="500">
    <![CDATA[

      REGISTER sip:[service]@[remote_ip]:[remote_port] SIP/2.0
      Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
      From: sipp <sip:sipp@[local_ip]:local_port]>;tag=[pid]SIPpTag00[call_number]
      To: sut <sip:[service]@[remote_ip]:[remote_port]>
      Call-ID: [call_id]
      CSeq: 1 REGISTER
      Contact: sip:sipp@[local_ip]:[local_port]
      Max-Forwards: 70
      Subject: Performance Test
      Content-Type: application/sdp
      Content-Length: [len]

    ]]>
  </send>

  <recv response="200" rtd="true">
  </recv>

</scenario>
```

Contents of the *uac_pcap.xml* scenario file used for performance testing (see section 4.2.):

```
<?xml version="1.0" encoding="ISO-8859-1" ?>
<!DOCTYPE scenario SYSTEM "sipp.dtd">

<!-- This program is free software; you can redistribute it and/or      -->
<!-- modify it under the terms of the GNU General Public License as     -->
<!-- published by the Free Software Foundation; either version 2 of the -->
<!-- License, or (at your option) any later version.                    -->
<!--                                                                    -->
<!-- This program is distributed in the hope that it will be useful,    -->
<!-- but WITHOUT ANY WARRANTY; without even the implied warranty of     -->
<!-- MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE.  See the      -->
<!-- GNU General Public License for more details.                       -->
<!--                                                                    -->
<!-- You should have received a copy of the GNU General Public License  -->
<!-- along with this program; if not, write to the                      -->
<!-- Free Software Foundation, Inc.,                                    -->
<!-- 59 Temple Place, Suite 330, Boston, MA  02111-1307 USA             -->
<!--                                                                    -->
<!--                 Sipp 'uac' scenario with pcap (rtp) play           -->
<!--                                                                    -->

<scenario name="UAC with media">
  <!-- In client mode (sipp placing calls), the Call-ID MUST be         -->
  <!-- generated by sipp. To do so, use [call_id] keyword.                -->
  <send retrans="500">
    <![CDATA[

      INVITE sip:[service]@[remote_ip]:[remote_port] SIP/2.0
      Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
      From: sipp <sip:sipp@[local_ip]:[local_port]>;tag=[pid]SIPpTag09[call_number]
      To: sut <sip:[service]@[remote_ip]:[remote_port]>
      Call-ID: [call_id]
      CSeq: 1 INVITE
```

```
    Contact: sip:sipp@[local_ip]:[local_port]
    Max-Forwards: 70
    Subject: Performance Test
    Content-Type: application/sdp
    Content-Length: [len]

    v=0
    o=user1 53655765 2353687637 IN IP[local_ip_type] [media_ip]
    s=-
    c=IN IP[local_ip_type] [media_ip]
    t=0 0
    m=audio [auto_media_port] RTP/AVP 0 101
    a=rtpmap:0 PCMU/8000
    a=rtpmap:101 telephone-event/8000
    a=fmtp:101 0-11,16

  ]]>
</send>

<recv response="100" optional="true">
</recv>

<recv response="180" optional="true">
</recv>

<!-- By adding rrs="true" (Record Route Sets), the route sets         -->
<!-- are saved and used for following messages sent. Useful to test   -->
<!-- against stateful SIP proxies/B2BUAs.                             -->
<recv response="200" rtd="true" crlf="true">
</recv>

<!-- Packet lost can be simulated in any send/recv message by         -->
<!-- by adding the 'lost = "10"'. Value can be [1-100] percent.       -->
<send>
  <![CDATA[

    ACK sip:[service]@[remote_ip]:[remote_port] SIP/2.0
    Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
    From: sipp <sip:sipp@[local_ip]:[local_port]>;tag=[pid]SIPpTag09[call_number]
    To: sut <sip:[service]@[remote_ip]:[remote_port]>[peer_tag_param]
    Call-ID: [call_id]
    CSeq: 1 ACK
    Contact: sip:sipp@[local_ip]:[local_port]
    Max-Forwards: 70
    Subject: Performance Test
    Content-Length: 0

  ]]>
</send>

<!-- Pause 10 seconds so that all SIP calls can be established. -->
<pause milliseconds="10000"/>

<!-- Play a pre-recorded PCAP file (RTP stream)                       -->
<nop>
  <action>
    <exec play_pcap_audio="pcap/g711u.pcap"/>
  </action>
</nop>

<!-- Pause 37 seconds, which is approximately the duration of the      -->
<!-- PCAP file                                                        -->
<pause milliseconds="37000"/>

<!-- Pause 10 seconds so that all SIP calls can be shut down. -->
<pause milliseconds="10000"/>

<!-- The 'crlf' option inserts a blank line in the statistics report. -->
```

```
  <send retrans="500">
    <![CDATA[

      BYE sip:[service]@[remote_ip]:[remote_port] SIP/2.0
      Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
      From: sipp <sip:sipp@[local_ip]:[local_port]>;tag=[pid]SIPpTag09[call_number]
      To: sut <sip:[service]@[remote_ip]:[remote_port]>[peer_tag_param]
      Call-ID: [call_id]
      CSeq: 2 BYE
      Contact: sip:sipp@[local_ip]:[local_port]
      Max-Forwards: 70
      Subject: Performance Test
      Content-Length: 0

    ]]>
  </send>

  <recv response="200" crlf="true">
  </recv>

  <!-- definition of the response time repartition table (unit is ms)   -->
  <ResponseTimeRepartition value="10, 20, 30, 40, 50, 100, 150, 200"/>

  <!-- definition of the call length repartition table (unit is ms)     -->
  <CallLengthRepartition value="10, 50, 100, 500, 1000, 5000, 10000"/>

</scenario>
```

The "g711u.pcap" file containing RTP packets is available on demand.