Asterisk with IPv6: Seamless and Ubiquitous VoIP

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Credentials

- 20+ years in IP networking and Unix, with 10 years on IPv6...
- IP engineering standards(IETF):
  - Wrote IETF drafts and RFCs.
  - Co-chaired internationalized domain names (idn) IETF wg
- Authoring:
  - Book: Migrating to IPv6, Wiley, 2006.
  - Cisco IPv6 course (co-author)
  - Tutorials on IP, security, Ipv6, etc... at many conferences, organisations
- IPv6forum: co-founder, board member. North American Ipv6 Task Force: steering group member.
- Asterisk developer, co-ported Asterisk to IPv6.
- President of Viagénie, consulting in advanced IP networking. Helping providers, enterprises, manufacturers and governments. IPv6, VoIP, Asterisk, Security, Internationalization, etc.

Plan

- IPv6
- Why IPv6 and Asterisk
- Changes to Asterisk
- Experience running VoIP-IPv6
- Lessons learned
- Next Steps
- Conclusion

IPv6?

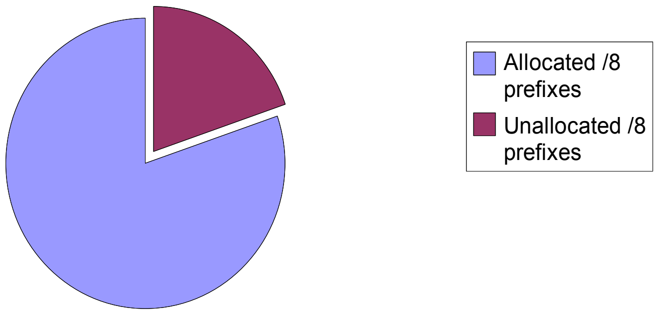
- New version of IP:
  - fixes IPv4 issues
  - adds new functionality
- Addresses:
  - No NAT!
  - 128 bits
  - Each enterprise receives /48 which can address 65536 links, each link can address an unlimited number of host.
  - Link(Subnet,vlan,...) “subnet” mask is fixed: /64
  - Uniquely assigned private address space:
    - no collision of private networks addressing.
    - No network management nightmare.
    - Full view of the networks.

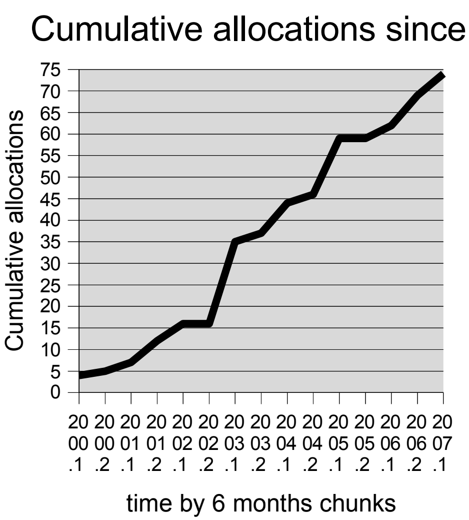
IPv6?

- Mobility:
  - keep connections up even when host changes IP address
  - Very useful for handover multiple link access (wifi, wimax,...)
- Autoconfiguration:
  - routers announce the link prefix on the link.
  - Hosts use their MAC address for the host part of the address
  - Very useful for embedded devices.
- Integrated IPsec
- Many more features

IPv4 Addresses Depletion

- < 20% of remaining address space
  - Unallocated in red on the chart
- Since 2000, average allocation rate of 4% of total address space per year.
- Last 9 months consumption rate:
  - 5% of total address space allocated
  - 20% of remaining address space allocated
- Predictions of exhaustion:
  - for 2009-2011.





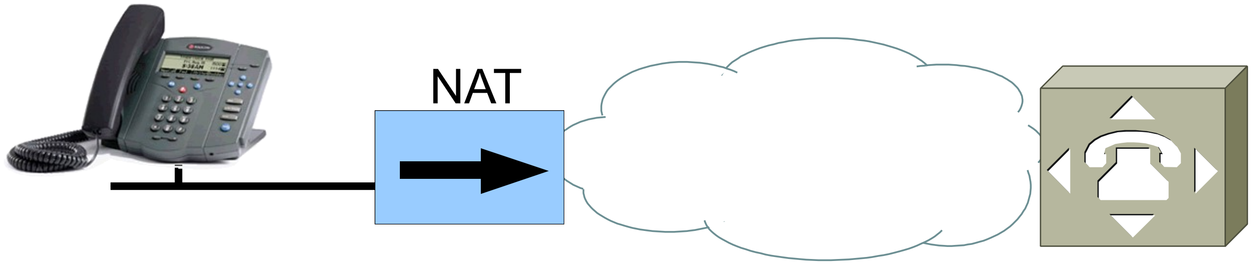
IPv6 Most Active Markets

- Asia
  - Japan: see http://www.v6pc.jp
  - China: through NGN. Olympics is important milestone.
  - Korea, Southeast Asia
- US government:
  - Mandating IPv6 for 2008 in all agencies
  - DoD is leading
- Providers (short list):
  - Teleglobe/VSNL, NTT, AT&T, GlobalCrossing,...
  - Comcast: can't address all the devices (100M+) with IPv4. Deploying IPv6. (DOCSIS 3.0 is Ipv6-ready).
- Requirement for IMS

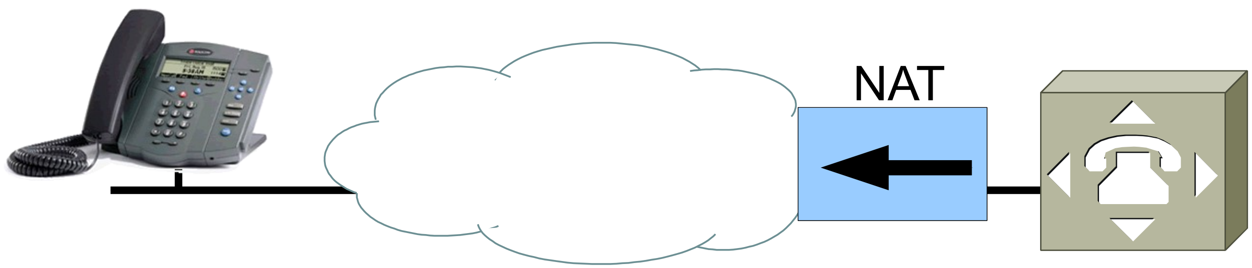
VoIP today

- SIP-based VoIP:
  - Separate signaling and media path
  - Does not work well with NAT.
  - Multiple variations of NAT traversal solutions:
    - STUN, TURN, ICE, ...
    - showed complexity and brittleness
    - and lack of support in the implementations
  - User Agent may be behind a NAT with some efforts.
  - But it is very difficult to have a SIP server (proxy,registrar, ...) to be behind a NAT.
- Important issue is reachability

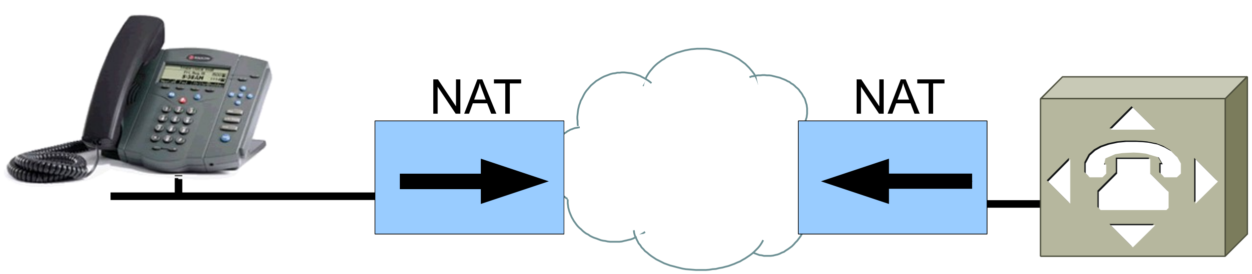
Why? Reachability



- Not all phones support this.
- Kludgy
- Does not work in all cases



- Difficult to deploy
- Sometimes mostly impossible

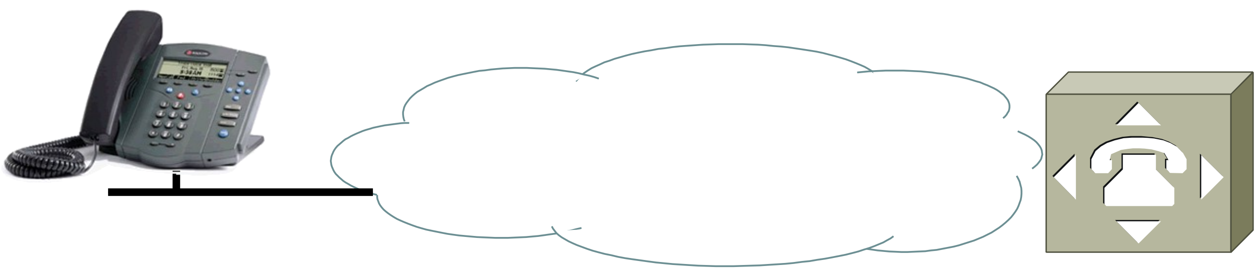


- Don't try it!

Consequences of NAT

- User consequence:
  - Calls do not go through
  - Audio is one-way
  - DTMF does not work
- Implementor consequence:
  - Very complex implementations.
  - Fragile. Difficult to debug.
  - Long cycle of development/testing.
- Deployment consequence:
  - careful planning
  - long time for deploying, testing, etc..

Why IPv6?



- It just works!

Why IPv6 and Asterisk?

- As any VoIP system, Asterisk does suffer NAT.
- Asterisk had no IPv6 support
- Viagénie has ported Asterisk to IPv6
- IPv6 and SIP
  - No NAT, No STUN, No TURN, No ICE, No MIDCOM, = no complexity, "just works".
  - True end-2-end media path.
  - Much easier to deploy. A VoIP-IPv6 deployment in Japan found important cost reductions because of the ease of installation and support.
- To have an IPv6-enabled application such as Asterisk, need to convert to the new API.

Challenges with IPv6 in Asterisk chan_sip

- Current architecture supports a single socket : 'sipsock'.
- The default source address is hardcoded to 0.0.0.0.
- The RTP socket is initialized from 'sipsock'
- Widespread use of sockaddr_in structures and short buffers (>256 bytes) to store hostnames and IP address strings.
- Many instances of similar code for parsing SIP url.

Design choices

- Use multiple sockets
  - Initial patch provides 1 socket per address family.
  - future work should include multiple sockets for each address family.
- Version independent when possible
  - Whenever possible, do not use sockaddr_in or sockaddr_in6 and never guess at the length of a sockaddr structure.
  - Only exception should be for setting socket options.

Impacts on Asterisk Code

- Files touched:
  - netsock.c/.h
  - chan_sip.c
  - rtp.c
  - Few others
- Some numbers:
  - ~25% of functions were changed/touched
  - many thousand lines changed/touched.
  - “Everywhere” in chan_sip, because: networking, logging (printing addresses) and sip url parsing.

Running Asterisk-v6 in Production

- Run at our offices with remote offices and road-warriors
- User point of view:
  - no difference. Same quality of voice, etc...
- Infrastructure point of view:
  - Dual-stack network.
  - Some phones are v4, others are dual-stack, some are restricted to v6 (for the purpose of testing)
- Deployment point of view:
  - Much easier: easier to deploy phones in home networks, for road-warriors, etc..
  - Easier to define firewall rules, since one can filter based on the source and destination addresses/prefixes (not possible with NAT)
  - Easier to troubleshoot, since easy to trace

VoIPv6 Tests

- With CounterPath EyeBeam (IPv6 version)
- And Asterisk (IPv6 version)
- Using the IPv6 Internet backbone and the VPF
-
- Viagénie(Canada) and Consulintel(Spain) conducted successful VoIPv6 calls.
- Easy config, no need to take care of NAT, STUN, etc...
- It just works!

Lessons Learned

- IPv4-IPv6 SIP in production is challenging
  - Found without trying to do:
    - IPv6 SIP signaling but media path is established using IPv4.
    - Troubleshooting is more difficult?
    - Need to investigate
- Conformance support for IPv6 SIP implementations.
- Based on deployment experience, should write a BCP paper on IPv4-IPv6 SIP deployments.

Next Steps

- Code is based on august 2006 trunk. Need to remerge to 1.4 and trunk.
- IPv6 VoIP peering
- Discuss with community how to integrate code into trunk
- Add a startup flag to Asterisk to disable IPv6.
- More testing! Especially Interop tests.
  - test with other implementations (SER, ...)
  - test with other IPv6 SIP UAs.... if you have one, please contact us.
- Improve IPv6 support in chan_sip to better handle complex scenarios.
  - implement ANAT [RFC4091, RFC4092].
  - IPv6 <-> IPv4
- Add IPv6 support to chan_iax (work in progress) and chan_*.
- Fix bugs

Conclusion

- Discussed:
  - the benefits of IPv6 and Why Asterisk benefits of being IPv6-enabled.
  - How to port an application to IPv6
  - Changes to Asterisk
  - Demo
  - Next Steps
- Information on this Asterisk-IPv6 project is available at:
  - http://www.asteriskv6.org .
  - We will be posting progress, tests with IPv6 UA, code, ...

Questions?

Contact info:

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This presentation is available at http://www.viagenie.ca/publications/

Information on this Asterisk-IPv6 project: http://www.asteriskv6.org

References

– [RFC3493] Gilligan, R., Thomson, S., Bound, J., McCann, J., and W. Stevens, "Basic Socket Interface Extensions for IPv6", RFC 3493, February 2003.

– [RFC3542] Stevens, W., Thomas, M., Nordmark, E., and T. Jinmei, "Advanced Sockets Application Program Interface (API) for IPv6", RFC 3542, May 2003.

– IPv6 Network Programming, Junichiro itojun Hagino, Elsevier, 2004, ISBN 1555583180.

– Migrating to IPv6, Marc Blanchet, Wiley, 2006, ISBN 0-471-49892-0, http://www.ipv6book.ca

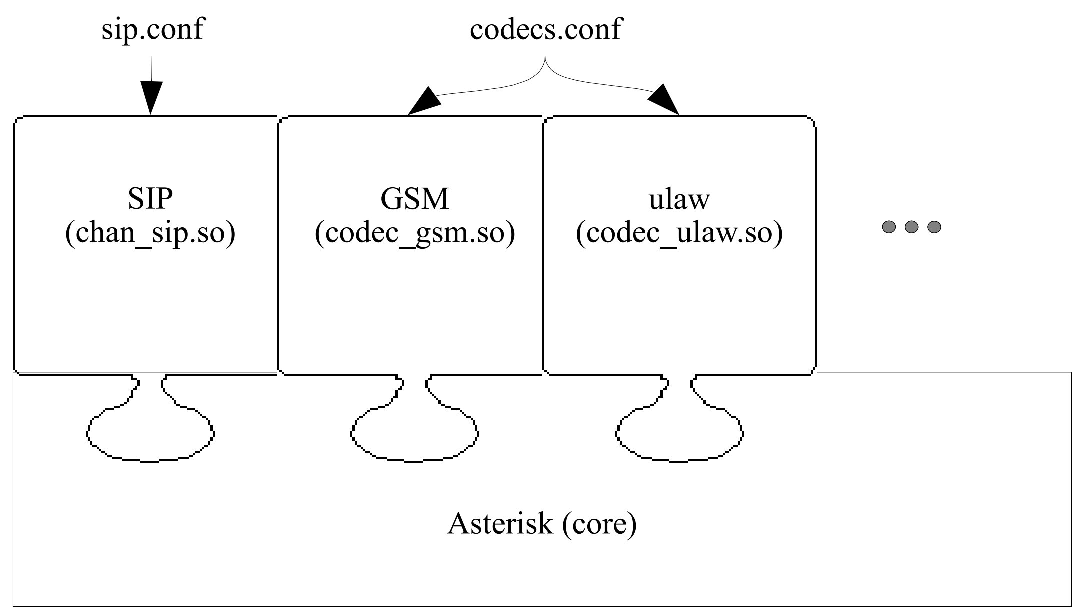
BACKUP SLIDES

IPv6 Support

- Support on OS (stack and API):
  - Same (new) API everywhere!!! ;-)
  - Since: Linux 2.4, FreeBSD 4.X, MacOSX 10.2, Windows XP, Solaris 8, ...
- Opensource Apps:
  - Apache, Sendmail/postfix, openssh, Xfree/Xorg, ...
  - Now Asterisk... ;-)
- Support on network gear:
  - Cisco, Juniper, Checkpoint, Quagga, ...

Asterisk Architecture

- Channels: SIP, IAX, MGCP, ZAP(PSTN), etc..
- Each channel is implemented as a loadable module
- SIP Channel(chan_sip) is a “monolithic” channel that does SIP and SDP.



IPv6 SIP user agents

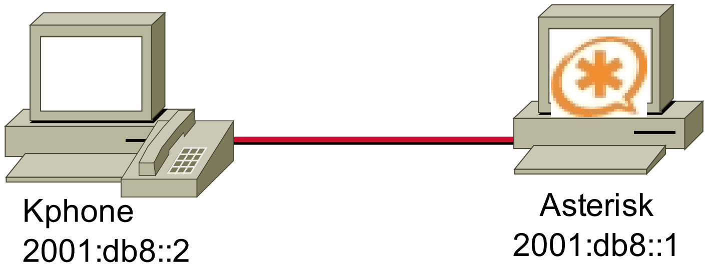
- Few open source IPv6 SIP user agents are available at this time. Many pretend to be IPv6-ready, but they were never tested or with very low number of tests. We have been sending patches to some of them.
- Makes testing and especially interop testing more limited.
- We tested 2 softphones with a 'working' ipv6 implementation:
  - kphone 3.1.1 with IPv6 patch.
  - Linphone 1.3.5
  - Both implementations contains (IPv6) bugs.
- Testing 3 commercial SIP UA IPv6-enabled: both hard and softphones. One that worked well: Counterpath Eyebeam (Windows version) (not yet released public)

Modifications to sip.conf

- 'bindaddr' now supports the address:port syntax such as:
  - 10.1.1.1
  - 10.1.1.1:5060
  - [2001:db8::1]
  - [2001:db8::1]:5060
- If no 'bindaddr' is specified for an address family, the wildcard is used (0.0.0.0 AND [::]).
- 'host' contains only the address, therefore no brackets.
- 'bindport' is still supported for backward compatibility.

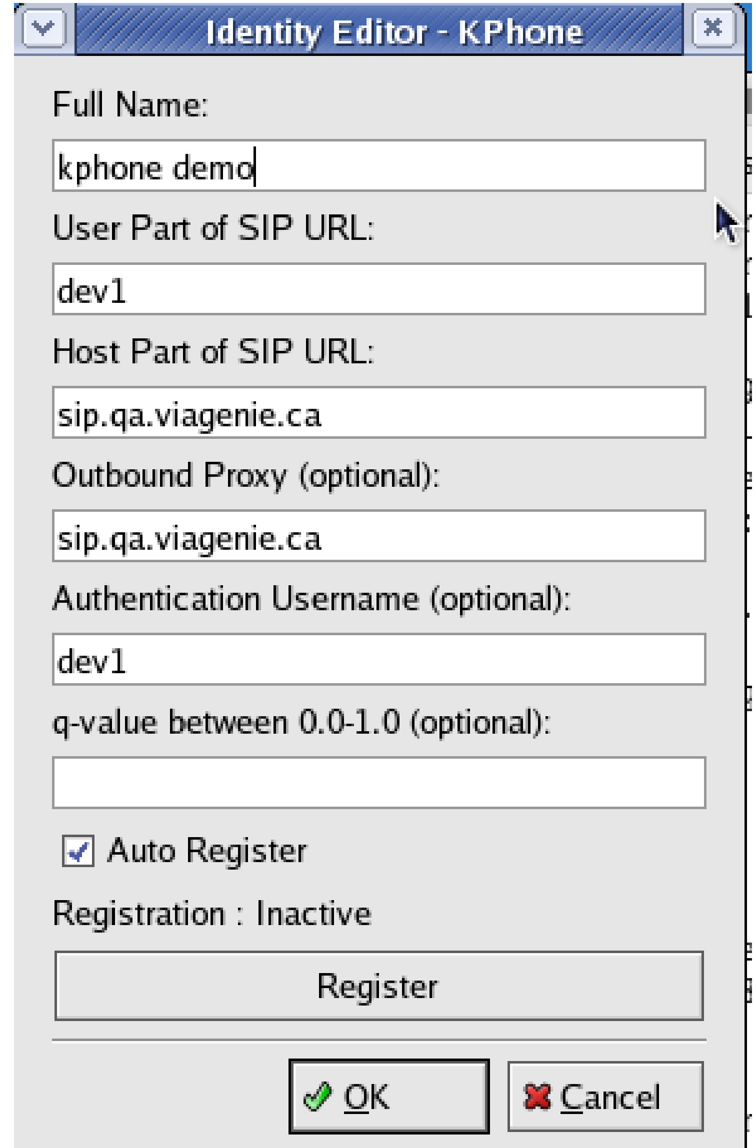
'Hello World' demo

- Uses Kphone as IPv6 SIP UA.
- Register to Asterisk.
- Make a call to play the 'Hello world' sound file.



'Hello World' demo (cont.)

```
[general]
context=internal
bindaddr=[2001:db8::1]

[dev1]
type=friend
host=dynamic
context=internal


[dev2]
type=friend
host=dynamic
context=internal
```



'Hello World' demo (cont.)

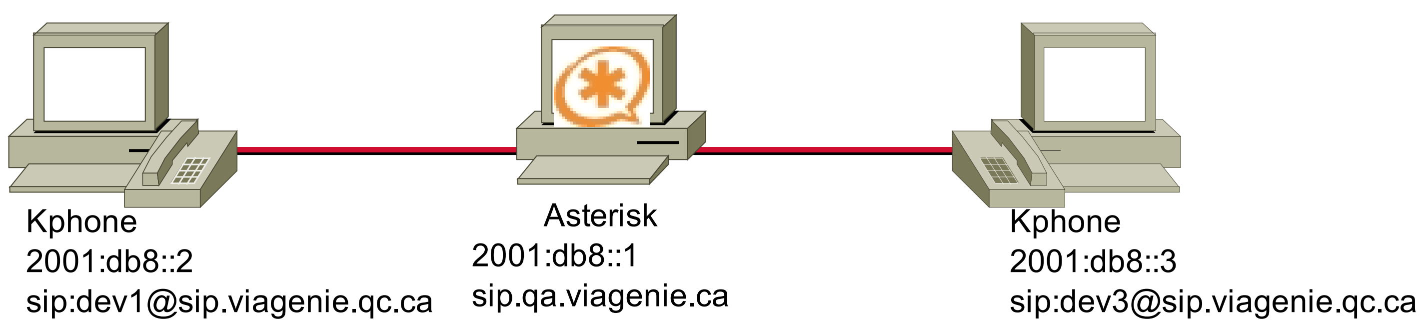
```
Reliably Transmitting (no NAT) to [2001:db8::2]:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP [2001:db8::2];received=2001:db8::2
From: "Fred" <sip:dev1@sip.qa.viagenie.ca>;tag=61617230
To: <sip:2@sip.qa.viagenie.ca>;tag=as15d09daf
Call-ID: 336600123
CSeq: 3245 INVITE
User-Agent: Asterisk PBX
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Contact: <sip:2@[2001:db8::1]>
Content-Type: application/sdp
Content-Length: 168

v=0
o=root 1406 1406 IN IP6 2001:db8::1
s=session
c=IN IP6 2001:db8::1
t=0 0
m=audio 10610 RTP/AVP 0
a=rtpmap:0 PCMU/8000
a=silenceSupp:off - - - -
a=sendrecv
```

UA1

Asterisk

2 Phones call demo

- 2 Kphone IPv6 SIP User Agents register to an Asterisk server.
- Establish a SIP call between the two user agents through an extension on Asterisk.



Bidirection call demo (cont.)

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```
Reliably Transmitting (no NAT) to [2001:db8::3]:5060:
INVITE sip:lefebvre@[2001:db8::3];transport=udp SIP/2.0
Via: SIP/2.0/UDP [2001:db8::1]:5060;branch=z9hG4bK1dc90af0;rport
From: "Fred" <sip:dev1@[2001:db8::1]>;tag=as3038e677
To: <sip:lefebvre@[2001:db8::3];transport=udp>
Contact: <sip:dev1@[2001:db8::1]>
Call-ID: 5351c608290f3c9d03ab0e346ed44a80@2001:db8::1
CSeq: 102 INVITE
User-Agent: Asterisk PBX
Max-Forwards: 70
Date: Wed, 18 Oct 2006 19:38:06 GMT
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY
Supported: replaces
Content-Type: application/sdp
Content-Length: 224

v=0
o=root 1406 1406 IN IP6 2001:db8::2
s=session
c=IN IP6 2001:db8::2
t=0 0
m=audio 32770 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=silenceSupp:off - - - -
a=sendrecv
```

UA1

Asterisk

UA2