Multimedia Services on the Internet and on Cellular Networks - Current Trends -

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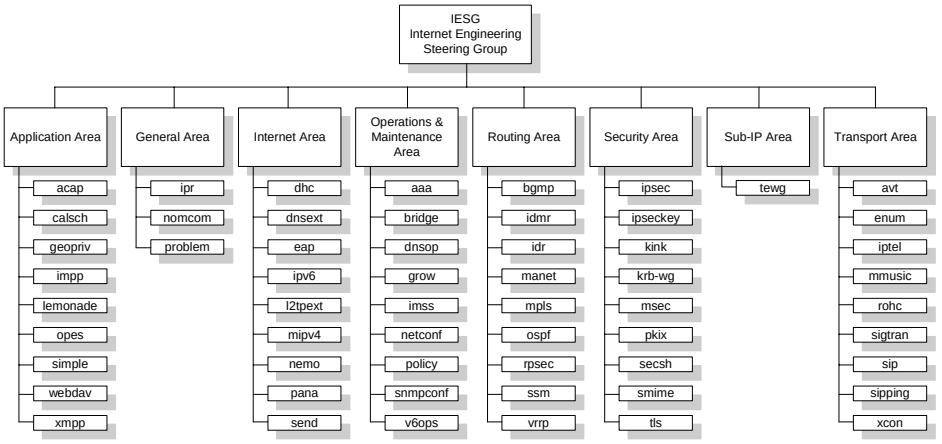
Outline

- Introduction to the IETF
- Where are multimedia protocols developed?
- Current work in
  - MMUSIC
  - SIPPING
  - SIMPLE
  - XCON
- ICE
- Session Policies
- Push-to-Talk

IETF Structure

- Areas
  - Area directors + IETF chair = IESG
- Working Groups
  - Chairs
  - Charter
  - Mailing list
- IAB

IETF Structure



Multimedia Work

- Traditionally in the Transport Area
  - AVT developed RTP
  - MMUSIC developed SDP, RTSP, SAP, and SIP
  - SIP, SIPPING, IPTEL, ENUM, and XCON were chartered later
- Application Area
  - SIMPLE
  - XMPP

Current Work in MMUSIC

- Internet Media Guide
  - Analysis of existing delivery protocols
- SDNng
  - XML-based
- ICE (Interactive Connectivity Establishment)
  - End-to-end STUN

SIPPING Design Teams

- Transcoding
  - B2BUA invocation
- Conferencing using SIP
  - Non-SIP work done in XCON
- Application Server interaction
  - Security
- Emergency calls
  - Routing
  - Location information conveyance

More SIPPING Work

- Configuration framework
- Session Policies
- End-to-middle security
- Request history
- Middle-to-end security
- Exploders

Current Work in SIMPLE

- MSRP
  - IM sessions
- XCAP
  - Configuration
- Different XML-based formats
- Partial notifications
- Filters
  - Limit contents and frequency

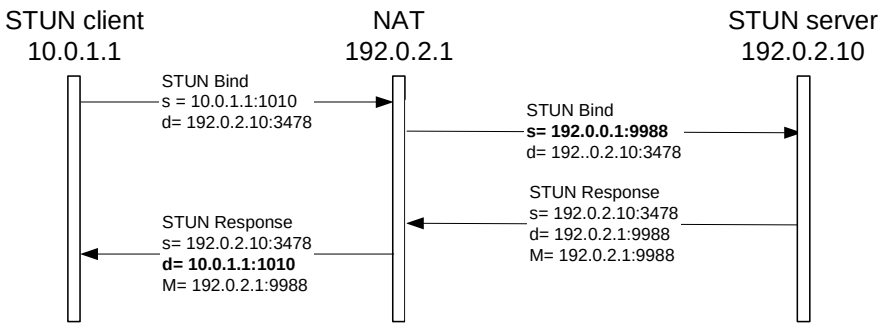
Current Work in XCON

- Conference Policy Control Protocol
- Floor Control Protocol

ICE

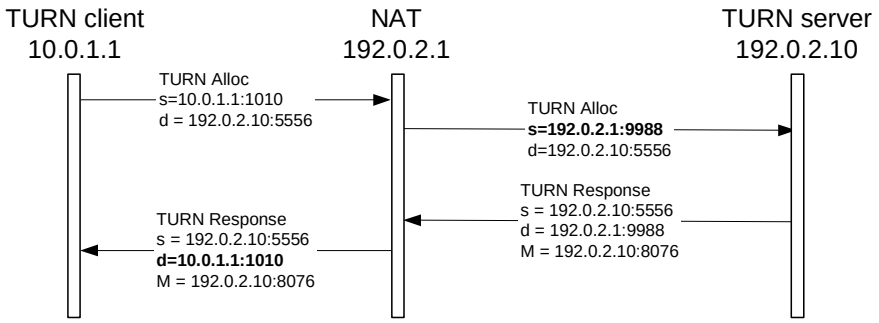
- ICE is a set of procedures to achieve connectivity in presence of NATs
  - The UAs gather a set of IP addresses
    - Local addresses
    - Using STUN, TURN, etc
  - Perform an offer/answer exchange
  - End-to-end STUN
    - May result in the discovery of extra IP addresses

STUN



s = source address
d= destination address
M = mapped address

TURN



s = source address
d = destination address
M = mapped address

Offer

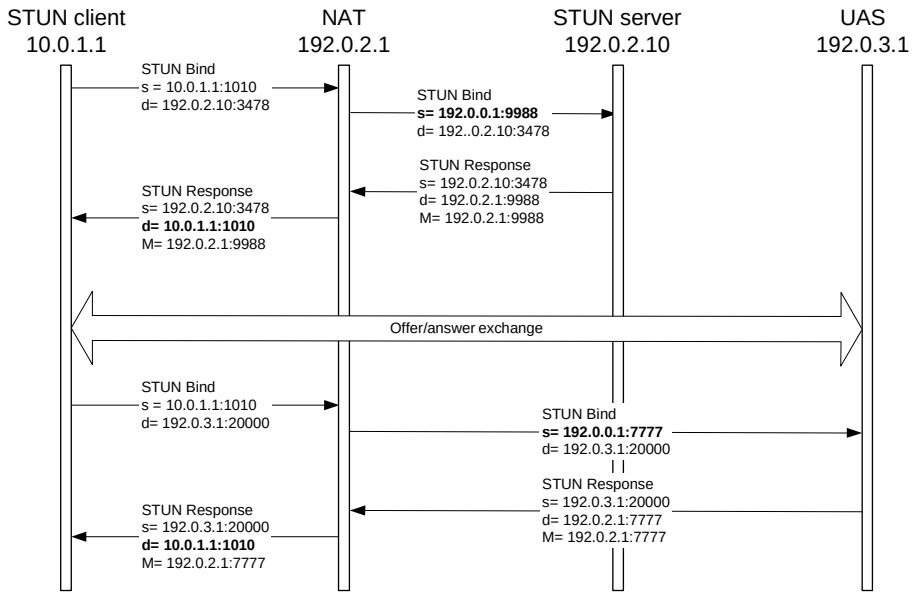
v=0
o=alice 2890844730 2890844731 IN IP4 host.example.com
s=
c=IN IP4 192.0.2.10
t=0 0
m=audio 8076 RTP/AVP 0
a=alt:1 1.0 : user 9kksj== 10.0.1.1 1010
a=alt:2 0.8 : user1 9kksk== 192.0.2.1 9988 192.0.2.1 9990
a=alt:3 0.4 : user2 9kksl== 192.0.2.10 8076

- TURN-derived address is the most likely to work
- RTP and RTCP ports are not adjacent in the STUN-derived address (the client did STUN twice)

Offer Arrives at the UAS

- UAS does STUN to
  - 10.0.0.1:1010
  - 192.0.2.1:9988 192.0.2.1:9990
  - 192.0.2.10:8076
- UAC does STUN to the UAS's addresses
- Checks connectivity
- May find new addresses
  - Which may have higher q values
  - E.g., symmetric NAT and a UAS with a public (or STUN-derived IP address)

End-to-End STUN



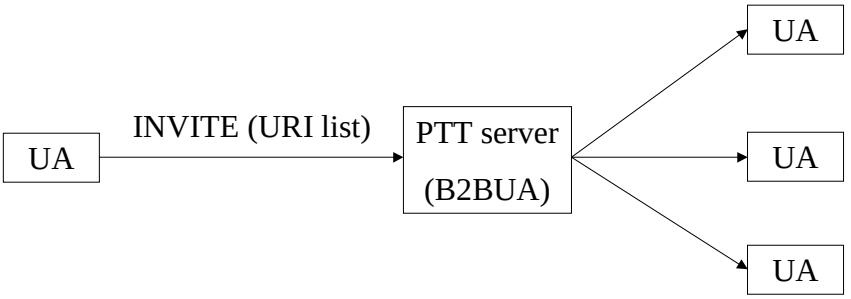
Session Policies

- Offer/answer happens between two UAs
  - But the network may have something to say
  - E.g., do not use high-bandwidth codecs
- Session policies allow the network to send policies to UAs
  - Session independent policies
  - Session specific policies
- The mechanism is work in progress

Push-to-Talk

- Half-duplex group communication
- Uses the IMS infrastructure
- Does not need conversational radio bearers
- Standardized by OMA
- Technology wise, a Push-to-Talk server is a conference server

SIP Signalling



XML-Based URI List

```
--boundary1
Content-Type: application/resource-lists+xml
Content-Length: 315
Content-ID: <cn35t8jf02@example.com>

<?xml version="1.0" encoding="UTF-8"?>
<resource-lists xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
  <list>
    <entry uri="sip:bill@example.com" />
    <entry uri="sip:joe@example.com" />
    <entry uri="sip:ted@example.com" />
    <entry uri="sip:bob@example.com" />
  </list>
</resource-lists>
--boundary1--
```

Floor Control