IETF Developments on IP Multimedia

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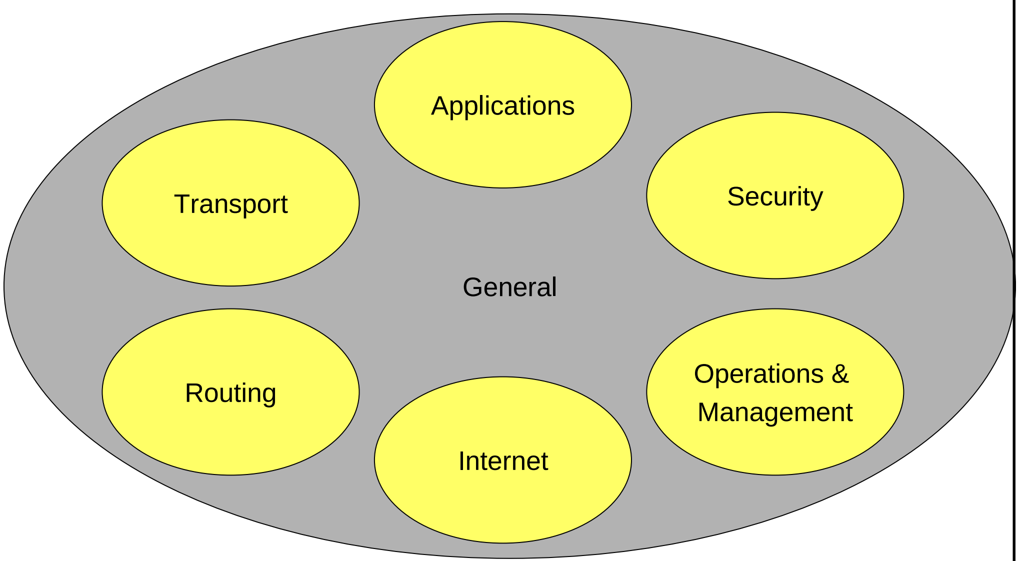
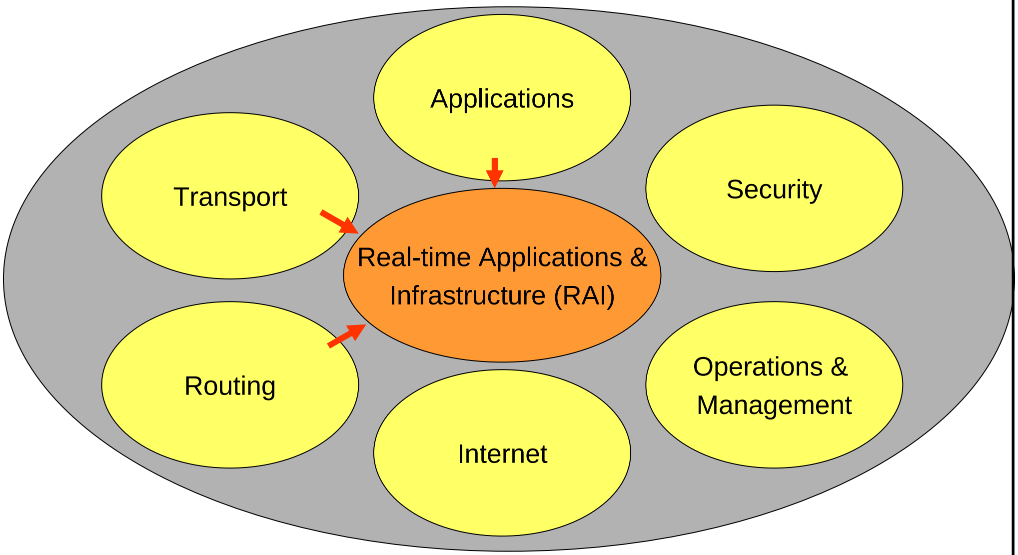
HELSINKI UNIVERSITY OF TECHNOLOGY
DEPARTMENT OF COMMUNICATIONS AND NETWORKING

IETF Areas



IETF Areas



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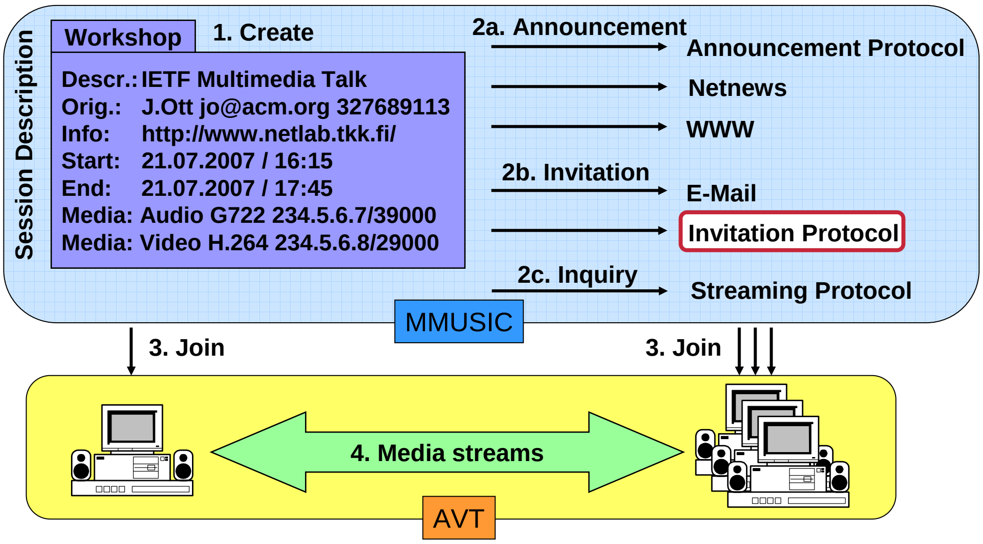
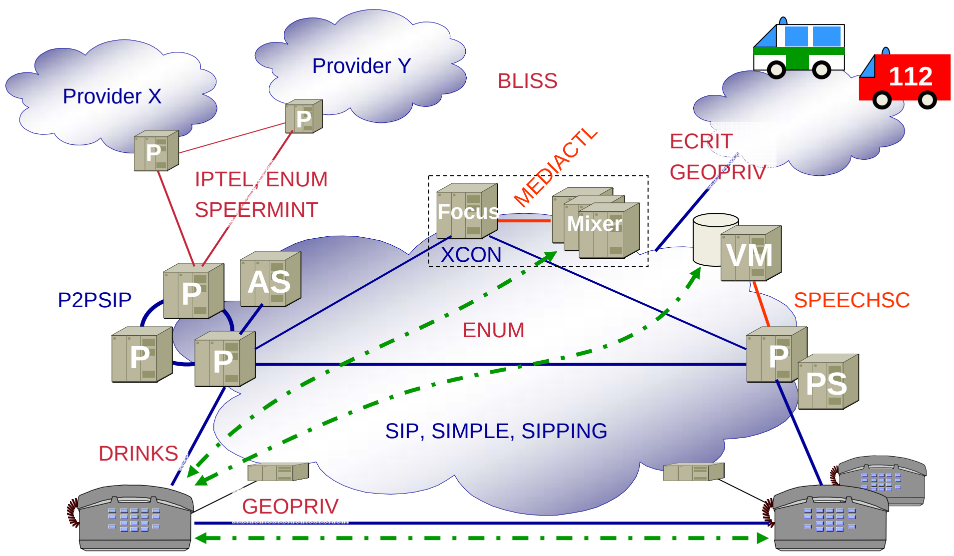
RAI Area Working Groups

- avt — Audio/Video Transport: RTP + payload formats
- mmusic — Multiparty Multimedia Session Control: RTSP, SDP, ICE
- sigtran — Signaling Transport: SCTP
- sip — Session Initiation Protocol: SIP + extensions
- p2psip — Peer-to-Peer Session Initiation Protocol: peer-to-peer architecture & protocols
- simple — SIP for Instant Messaging and Presence Leveraging Extensions
- sipping — Session Initiation Proposal Investigation: usages of SIP
- xcon — Centralized Conferencing: conferencing
- speechsc — Speech Services Control
- mediactrl — Media Server Control
- bliss — Basic Level of Interoperability for SIP Services
- drinks — Data for Reachability of Inter/tra-NetworK SIP
- ecrit — Emergency Context Resolution with Internet Technologies
- geopriv — Geographic Location/Privacy
- iptel — IP Telephony
- enum — Telephone Number Mapping
- speermint — Session PEERing for Multimedia INTerconnect

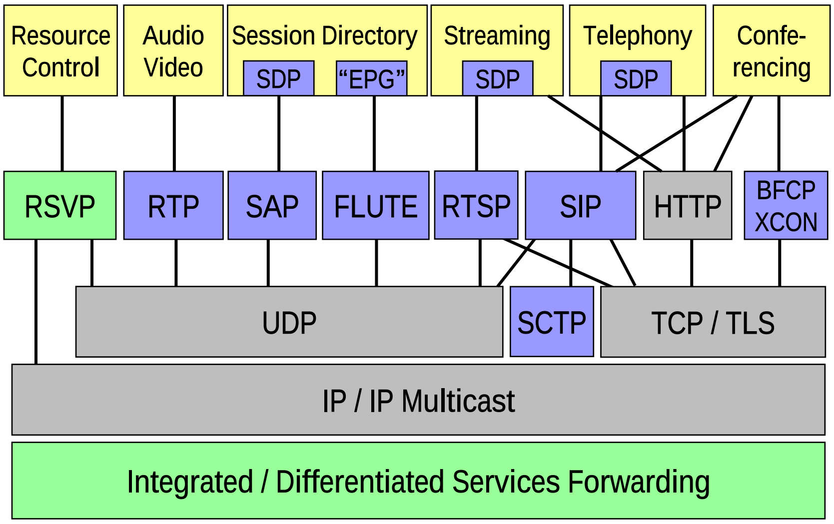
Basic IETF Multimedia



SIP*

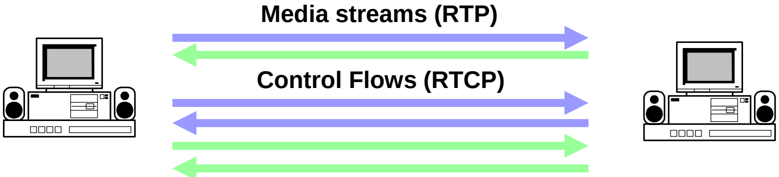


IETF Multimedia (Conferencing) Architecture



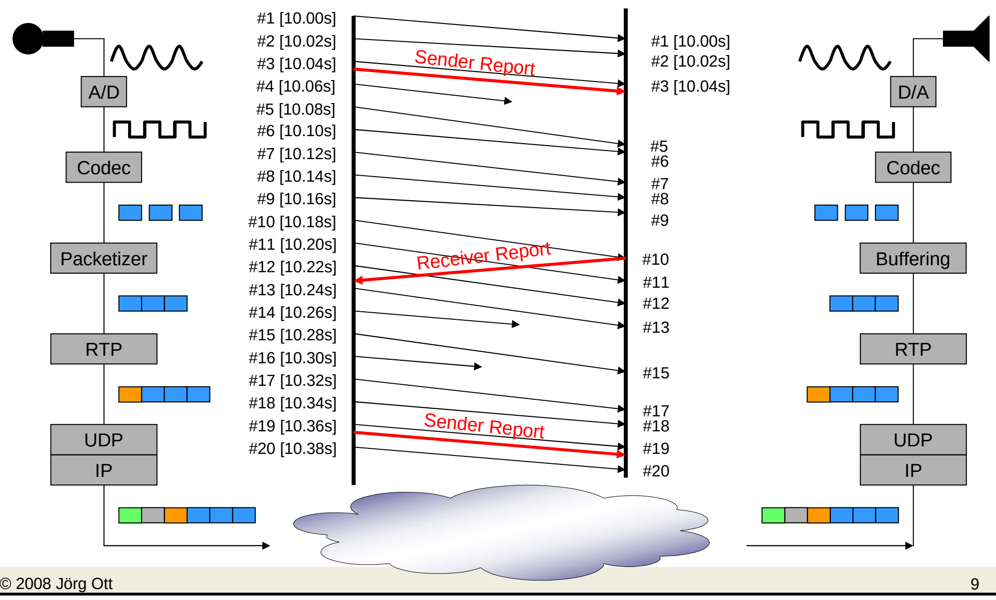
AVT: Real-time Transport (1)

- RTP + RTCP Functionality (RFC 3550)
  - Framing for audio/video information streams
  - Packetization formats for loss tolerant decoding
  - Preserve intra- and inter-stream timing; loss detection
  - Feedback about network quality (long-term + instant)
  - ➲ RTP sessions: media carried in UDP or DCCP packets (or via TCP, SCTP)



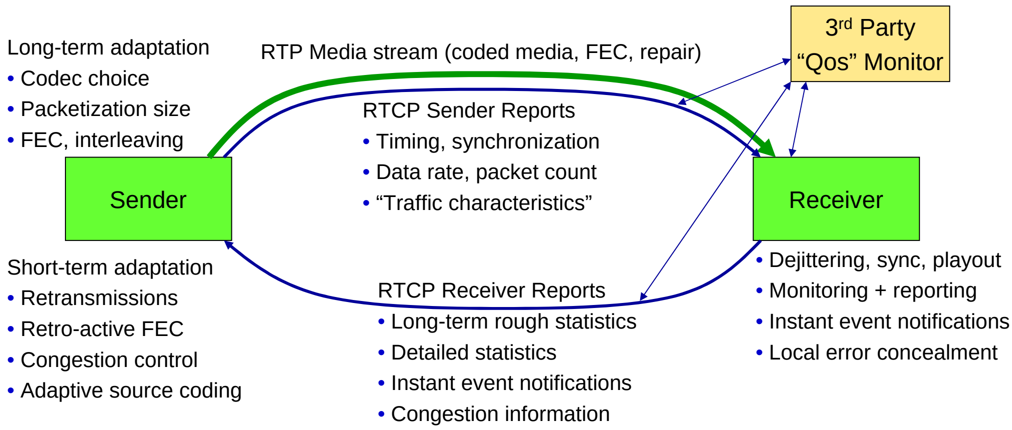
- Unicast: (S)IP telephony
- Multicast: Large-scale IPTV content distribution

Example: Audio over RTP



AVT: Real-time Transport (2)

- Adaptive real-time applications over IP
  - Tunable feedback loop for individual and group communications
  - From reporting per 5s and more to event-driven to once per RTT



MMUSIC: Basics for IP Multimedia

- Mbone: Media descriptions and session announcements
  - Session Announcement Protocol
  - Session Description Protocol
- Media streaming: RTSP (v1 in 1996, v2 in progress…)
- Origin of SIP (1996 – 1999)
- Media description and capability negotiation for SIP
  - Offer/answer model (RFC 3264)
  - Plethora of extensions to SDP
    - Attributes and fields: Addresses, media formats, parameters
    - Structuring descriptions: alternatives, dependencies, relations, semantics
  - Recently: True capability negotiation framework
- NAT traversal: Interactive Connectivity Establishment (ICE)

Some Further Information

http://www.ietf.org/html.charters/wg-dir.html
#Real-time%20Applications%20and%20Infrastructure%20Area

http://www.softarmor.com/sipwg/
http://www.softarmor.com/sippingwg/
http://www.softarmor.com/simple/
http://www.p2psip.org/

http://www.cs.columbia.edu/sip/
http://www.sipit.net/

and more…