



IETF Developments on IP Multimedia

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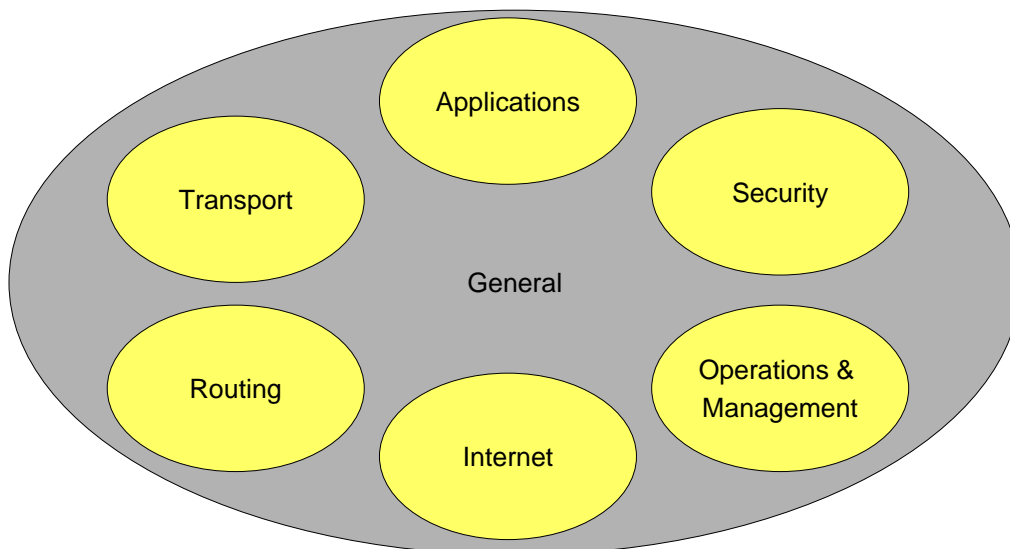
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EuroView 2008

21 July 2008

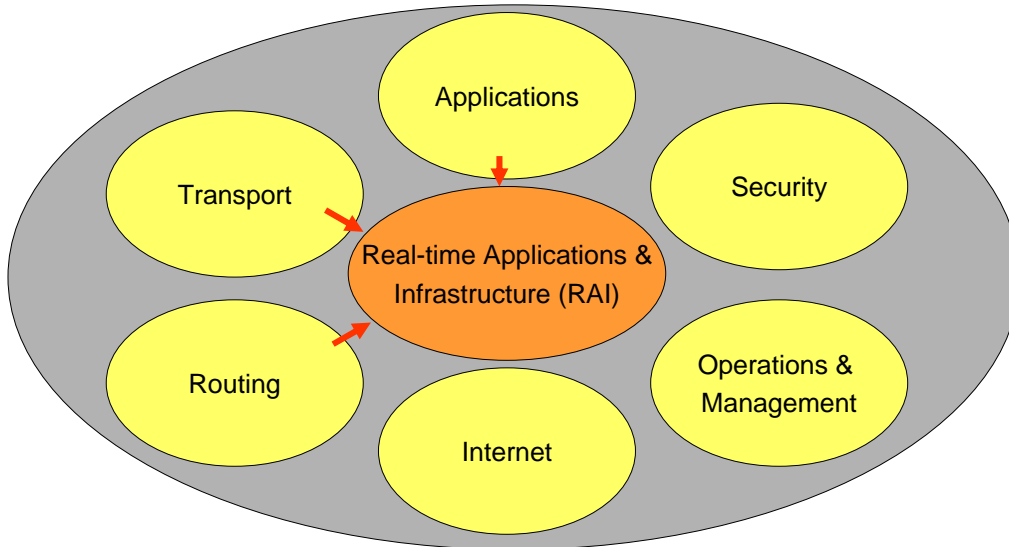


IETF Areas





IETF Areas



RAI Area Working Groups

- ▶ [avt](#) Audio/Video Transport: **RTP + payload formats**
- ▶ [mmusic](#) Multiparty Multimedia Session Control: **RTSP, SDP, ICE**
- ▶ [sigtran](#) Signaling Transport: **SCTP**

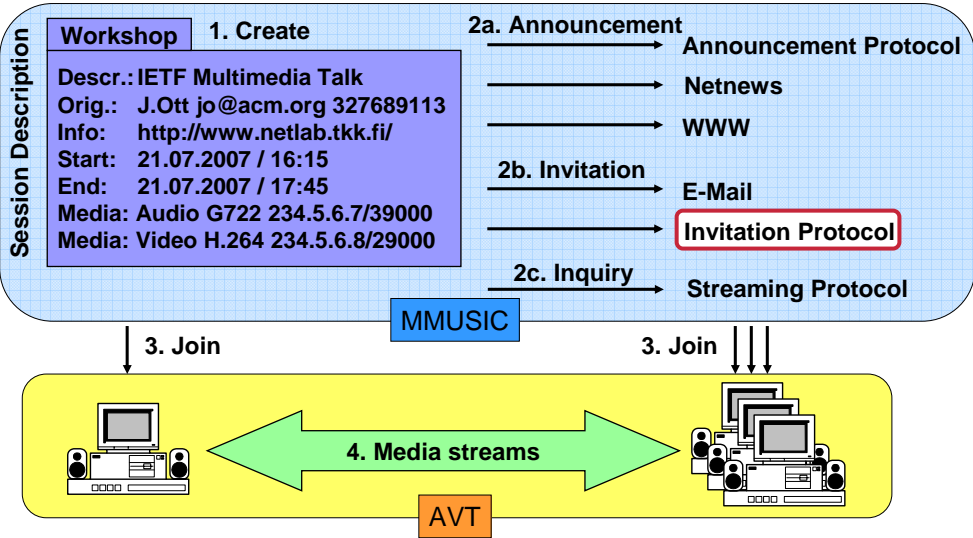
- ▶ [sip](#) Session Initiation Protocol: **SIP + extensions**
- ▶ [p2psip](#) Peer-to-Peer Session Initiation Protocol: **peer-to-peer architecture & protocols**
- ▶ [simple](#) SIP for Instant Messaging and Presence Leveraging Extensions
- ▶ [sipping](#) Session Initiation Proposal Investigation: **usages of SIP**
- ▶ [xcon](#) Centralized Conferencing: **conferencing**

- ▶ [speechsc](#) Speech Services Control
- ▶ [mediactrl](#) Media Server Control

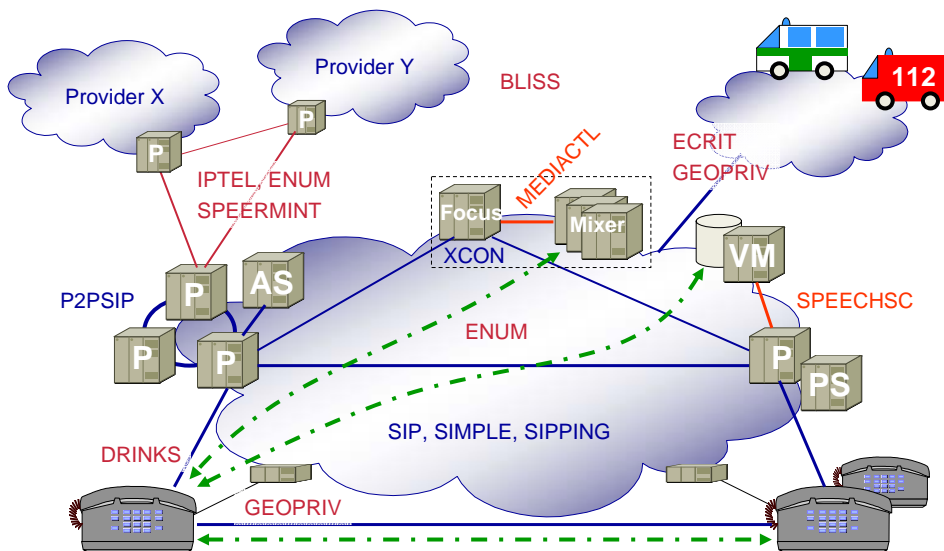
- ▶ [bliss](#) Basic Level of Interoperability for SIP Services
- ▶ [drinks](#) Data for Reachability of Inter/tra-Network SIP

- ▶ [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

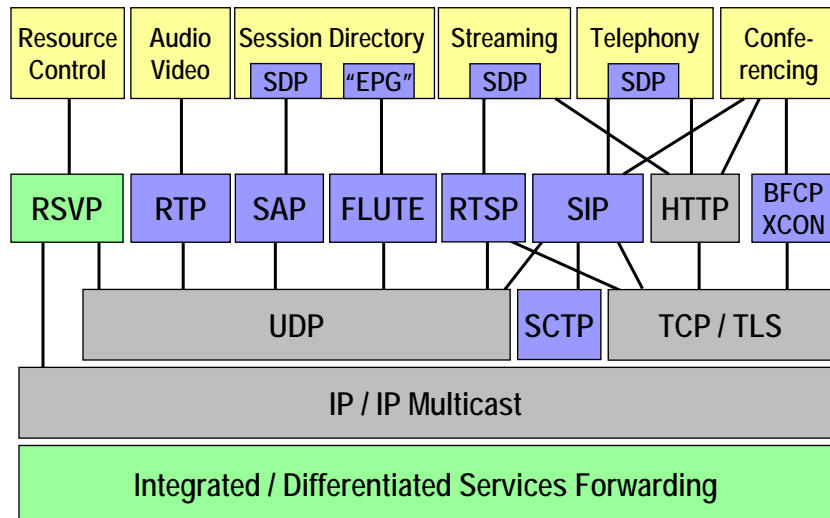


SIP*



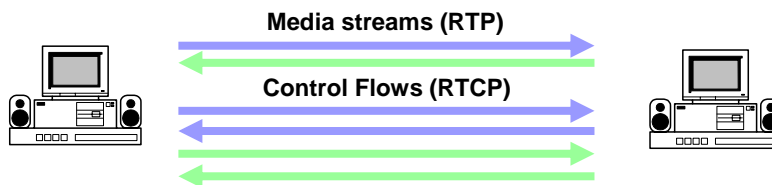


IETF Multimedia (Conferencing) Architecture



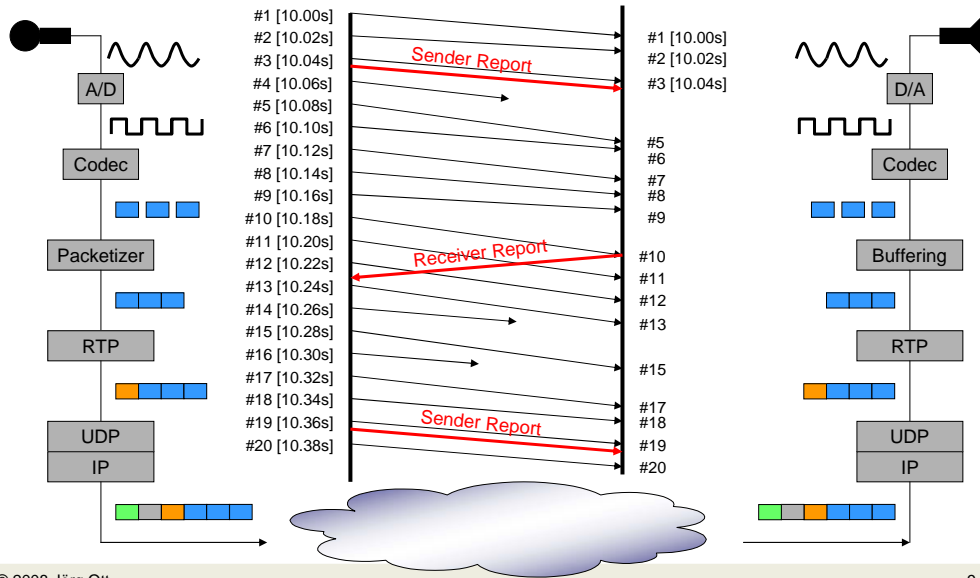
AVT: Real-time Transport (1)

- ▶ 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

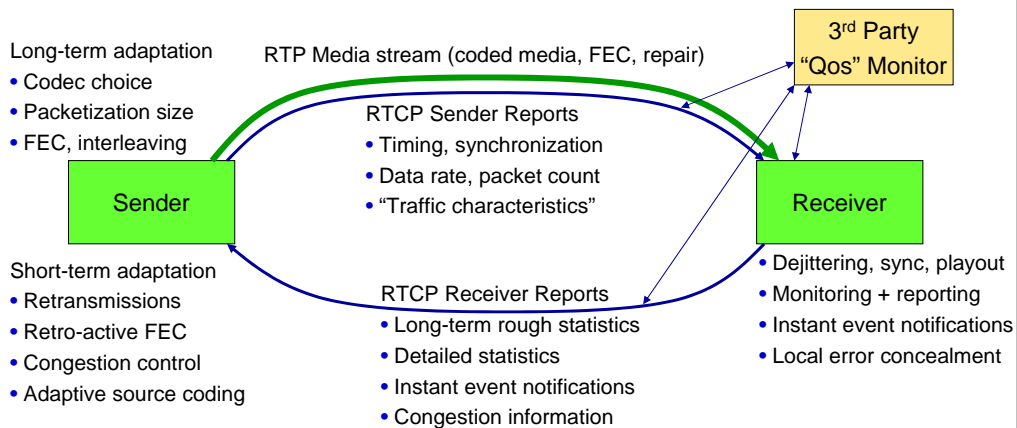
Example: Audio over RTP



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.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...