# Audio/Video Transport Working Group

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### **Agenda**

15:10	Introduction	(Chairs)
15:15	VoIP Shim for RTP Payload Format	(Johansson)
15:30	DTLS extension to establish keys for SRTP	(McGrew)
15:45	ZRTP	(Zimmermann)
16:00	Open discussion	
16:30	End	

#### **Intellectual Property**

- When starting a presentation you MUST say if:
  - There is IPR associated with your draft
  - The restrictions listed in section 5 of RFC 3978 apply to your draft
- When asking questions or commenting on a draft:
  - You MUST disclose any IPR you know of relating to the technology under discussion

• Reference: RFC 3978/3979 and the "Note Well" text

#### **VoIP Shim**

#### draft-johansson-avt-rtp-shim-01.txt

- Basic proposition: add signalling bytes to payload
  - redundancy level
  - frames per packet
  - application-dependent codec mode word
- Mechanism: designated payload type
- Justification: avoid fragmentation and packet loss on voice-optimized radio links
  - RTCP packets too big
  - header extension also adds bytes
- Issue: violates architectural principle
  - signalling does not relate directly to content of current packet
- What is the response time requirement?

## **DTLS for SRTP Key Establishment** draft-mcgrew-tls-srtp-00

- Open issue: transporting key management messages
- Proposition: separate protocol multiplexed into RTP port
- Justification:
  - lack of deployment of RTCP
  - inappropriate to carry in media packets (using header extension)

#### **ZRTP**

#### draft-zimmermann-avt-zrtp-02.txt

- Proposition: use header extension to carry keying information in media stream
- Justification:
  - avoid use of second port to simplify NAT passage
- Issue: violates architectural principle
  - signalling does not relate directly to content of current packet
- Alternative: carry in RTCP
  - issue of limited bandwidth, implying setup time of several seconds
  - hard to relax in QoS-controlled systems
    - typically end up over-providing RTCP bandwidth for duration of session