Using Open Source to promote an EBU standard

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Agenda

- **Why Audio Contribution over IP?**
- **EBU project group Audio Contribution over IP (N/ACIP)**
- **Reference Implementation**
- **Why Open Source?**

Why Audio Contribution over IP

- **Background**
  - Contribution of high quality audio between studios and/or production sites
  - In the past: n x 64 kb/s (ISDN) was used
- **Phase-out of ISDN**
  - Phone companies changeover to Voice over IP
  - In some parts of Europe none or difficult availability
- **IP networks are used everywhere**
  - Private and campus networks
  - Wide area and mobile networks
  - DSL connections
- **Audio over IP units available**
  - From different manufactures
  - Generally not interoperable

EBU project group Audio Contribution over IP (N/ACIP)

- **Kick-off: February 2006**
- **Recommendations and advice for**
  - IP-networks
  - Network structures
  - Protocols

  which cover streaming audio formats in broadcast quality

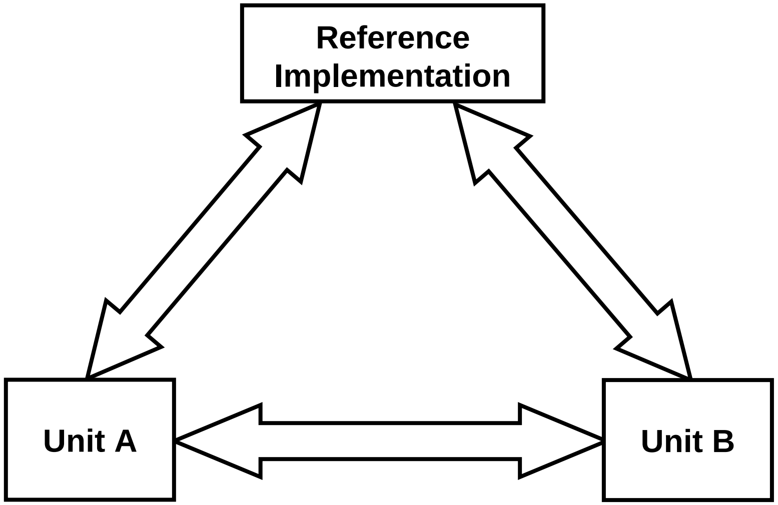
↳ **Results:**

- **Interoperability standard (version 1.0)**
  - Agreement between manufactures and broadcasters
  - Agreed in a meeting 8 September 2007 – manufacturers and EBU-members
  - To be published on www.ebu-acip.org now in October
- **Best practice for the deployment of Audio over IP (AoIP)**
  - Support to EBU-members using AoIP

Why a reference implementation?

- **Independent providing of standard**
- **As reference for interoperability tests**
  - Plugfest planned for February 2008

- The success Triangle



Based on standard protocols

- RTP: Real-time Transport Protocol (RFC3550, RFC3551)
- UDP: User Datagram Protocol
- TCP: Transmission Control Protocol is optional
- SDP: Session Description Protocol (RFC4566)
- SAP: Session Announcement Protocol (RFC2974)
- SIP: Session Initiation Protocol (RFC3261)
- Offer/Answer model for SDP (RFC3264)

Development Tools

- Open Source Tools
  - PJSIP
  - PortAudio
- Non Open Source Tools
  - ASIO Drivers
  - Steinberg SDK (freely available, but may not be distributed)

What is PJSIP?

- **Open Source SIP stack and media stack**
  - **current version 0.7.0**
  - **written in C**
  - **portable on every platform possible (even Symbian or Nintendo DS)**
  - **2 main developers and a very busy mailing-list**
  - **nearly daily updates and bugfixes ‘just in time’**

More on PJSIP

- **provides also NAT traversal methods (ICE, STUN, TURN)**
- **contains a Media Port Framework which provides easy integration of any codecs**
  - **GSM, G.711 and Speex included**
  - **incorporates PortAudio, an Open Source Cross-Platform Audio API**
- **www.pjsip.org**
- **www.portaudio.com**

Why Open Source?

- Codec extension of PJSIP
- Rapid spread of the ACIP-EBU recommendation
- Provides support by many developers
- Low error probability
- Test environment

Thank you for your attention !

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