

N/ACIP standards, recommendations, protocols for audio

EBU Networks Seminar– 19th June 2007

Lars Jonsson, Technical strategist , SR, Sweden

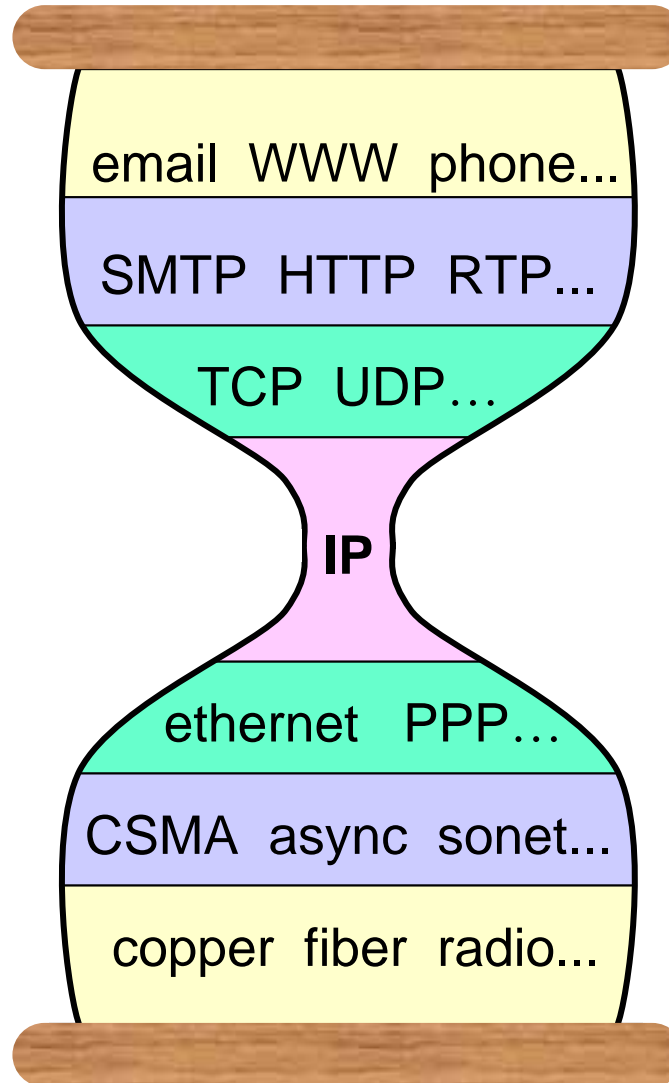
Mathias Coinchon, Senior engineer, EBU, Switzerland



Presentation plan

- **Why IP ?**
- **N/ACIP interoperability standard**
- **Protocols**
- **Networks**

Why IP ?

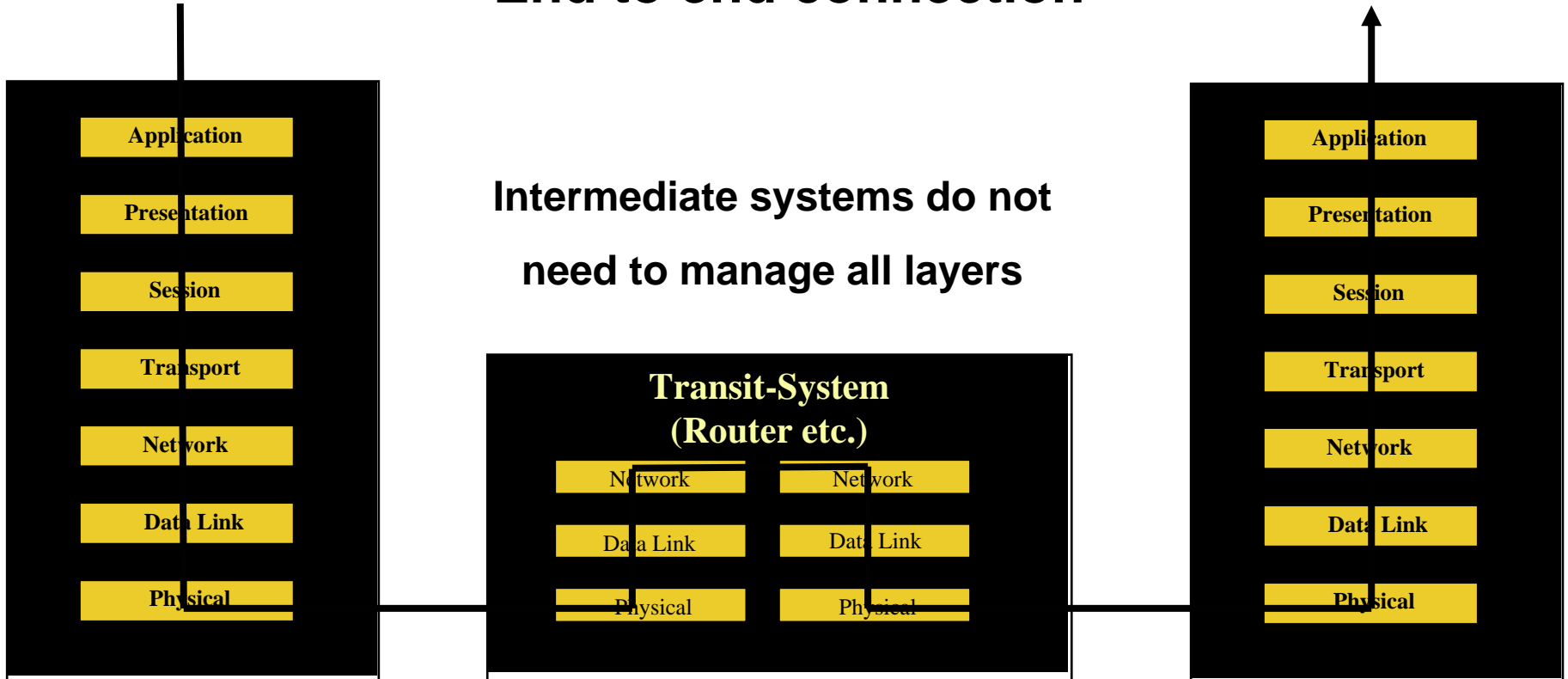


Layered approach

Why IP ?

End to end connection

Intermediate systems do not
need to manage all layers



- **Interoperability standard (now version 0.6)**
- **This document is expected to be completed during 2007, after more inputs from manufacturers and broadcasters.**
- **The standard is based on IETF RFC's for SIP and RTP over UDP**
- **A meeting with all manufacturers and EBU members will take place at IBC in September 2007**
- **Best practice for the deployment of Audio over IP to EBU members**
- **Recommendations to manufacturers: definition of common network profiles, Workflows, MMI, Security....**
- **Tests and measurements on networks and end units**

15 Manufacturers are active in our work

- **AEQ, Spain**
- **AETA, France**
- **APT, Ireland**
- **AVT, Germany**
- **Comrex, USA**
- **Digigram, France**
- **Harris, USA**
- **Mandozzi, Switzerland**
- **Mayah, Germany**
- **Musicam, USA**
- **Orban, USA**
- **Prodys, Spain**
- **Telos, USA**
- **Tieline, Australia**
- **Youcom, NL**
- **(+8 more, not yet active)**

Protocols

- **RTP: Realtime Transfer Protocol (RFC3550, RFC3551) on top of User Datagram Protocol (UDP)**
- **TCP is optional,**
 - **Pros: Reliable delivery & Passes more easily through firewalls**
 - **Cons: Causes longer delay, problems with congestion avoidance mechanism.**
- **SDP : Session Description Protocol (RFC4566)**
- **SAP : Session Announcement Protocol (RFC2974)**
- **SIP : Session Initiation Protocol (RFC3261)**
- **Offer/Answer model for SDP (RFC3264) for codec negotiation**

Audio coding

- **Minimum set of codecs for compatibility**
- **A: Mandatory codecs (A minimum set, always to be found)**
 - G.711
 - G.722
 - ISO MPEG-1 and MPEG-2 Layer II
 - PCM 16 bits, 20 bits, 24 bits
- **B: Recommended codecs**
 - MPEG-4 AAC, MPEG-4 AAC-LD
 - MPEG-1/2 Layer III
- **C: Optional codecs**
 - Enhanced APT-X, MPEG-4 HE-AACv2...

Two types of equipment

- General contribution equipment for all types of contribution (fixed or remote)
- Portable



Ethernet, 3G, HSPA

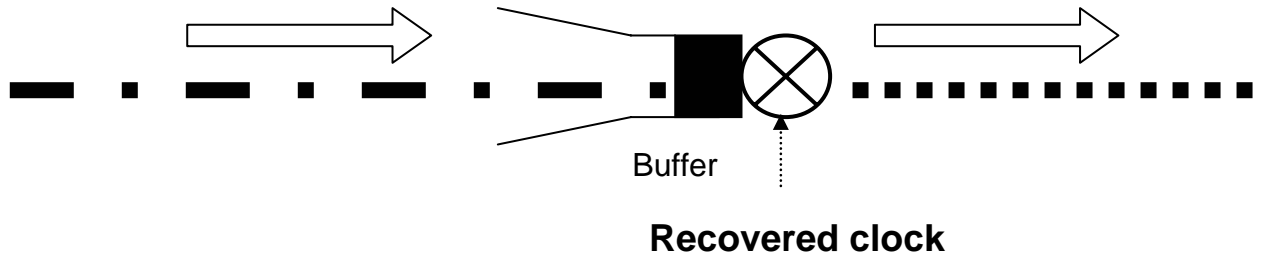


Packet loss recovery



- **Packet loss recovery methods**
- **FEC: Forward Error Correction**
 - Redundancy packets for loss reconstruction at receiver side
 - No return channel necessary
 - IPTV: Pro MPEG COP-3 (SMPTE 2022-1), Digital Fountain Raptor codes
- **Retransmission**
 - Receiver ask sender for retransmission
 - Increased delay
 - Use of RTCP messages (RFC4588, need RTP/AVPF)

Clock recovery



- **IP networks are asynchronous: no clock**
- **Receiver clock must be estimated and reconstructed**
- **Difficulty: estimate the long term drift avoiding the network jitter delay.**
- **Clock adaptation: Sampling rate conversion/adaptation or Frame drop**
- **Other approach: External clock source (GPS, etc)**

Delay

- **The delay is always longer than in conventional networks**
- **A short delay is needed for two-way communication**
- **There is a trade-off between delay and robustness.**
 - **A short delay (20 ms) means a short jitter buffer. Higher network jitter or delay jumps will create interruptions.**
 - **A very long jitter buffer (seconds) can compensate a bad network. This can be OK for a one-way circuit**
- **Packet recovery mechanisms always increase the delay**
- **Many suggest to use Error Concealment (instead of FEC)**

Signalling with SIP



- **Signalling and control of session**
 - Parameters from the stream
 - Connection setup and termination
 - Audio codec negotiation
 - Presence

SIP wireless handheld



In 3 or 5 years time SIP will be common in mobile devices
Interoperability with broadcast contribution units

Network Issues



The Internet



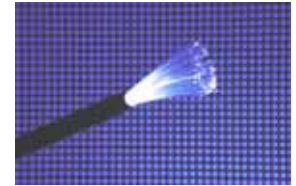
- **The Internet, a public IP network**
 - **Common router queues, congestions**
 - **Internet “noise”: Flood attacks (DoS: Denial of Service), scans,...**
- **Interconnections in Internet Exchange Points**
- **Dynamic routing (BGP)**
 - **Sudden latency increase, losses, packet disorder**
- **Absolutely no end-to-end guaranty is possible nowadays**
- **Guarantee is possible only on dedicated managed IP networks**

Networks

- **Sufficient robustness with audio over IP depends mainly on the quality of the IP network.**
- **The Internet is unreliable and so unsuitable for long live transmission.**
- **Internet is used more and more, for news, due to its availability and low cost, with a calculated risk of loss of the signal.**
- **Internal, well managed IP networks, can provide a sufficiently good Quality of Service.**
- **The WAN can be used both for conventional computer traffic, audio file transfer and streaming for live contribution of programmes.**
- **There are reliable methods, such as MPLS, to separate computer IP traffic from the audio streams.**
- **Audio over IP will gradually replace many of the traditional circuits that are in use today. Telcos will phase out old systems and move to IP.**
- **Both PCM Linear and various compressed formats at higher bitrates can be used for a stable contribution over IP. Don't go down too low bit-rate!**

Last mile access

- Last mile to the end user
- Fiber optic
 - High quality but expensive
- Copper with xDSL
 - ADSL: Asymmetrical uplink/downlink (bandwidth),
SDSL: Symmetrical uplink/downlink
 - Bit errors lead to single packet losses
- Mobile (3G/UMTS, Wimax)
 - Unreliable, no solutions with guaranteed QoS nowadays
 - HSDPA/HSUPA: shared channel
- Satellite
 - Long delays, high costs, often shared bandwidth
- Wireless
 - Wifi: no guaranty due to frequency sharing



Summary

- **ACIP interoperability standard proposal driven by manufacturers and broadcasters**
- **Based on common IETF standards (RTP, SIP, etc)**
- **Audio over IP will replace ISDN codecs**
- **SDH/J.57 and other older systems will be phased out**
- **The Internet will be unreliable for a long time**
- **Managed private networks provide broadcast robustness**
- **Similar issues for video are handled by the N/VCIP group**

Thank you ! – Questions ?

Lars Jonsson

Swedish Radio

lars.jonsson@sr.se

Mathias Coinchon

EBU

coinchon@ebu.ch