Audio Contribution over IP

Swiss AES meeting – 12th June 2008 - Bern

Mathias Coinchon

European Broadcasting Union

Technical department

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Plan of the presentation

1st part

- The European Broadcasting Union
- Audio contribution over IP background
- Overview of N/ACIP activities

2nd part

- Protocols
- Network aspects

The European Broadcasting Union

- **World largest association of broadcasters**
- **75 members from 56 countries**
- **45 associate members around the world**
- **470 TV channels and 904 Radio channels**
- **EBU activities: Radio/TV production, legal, technical, worldwide media exchange network, news,..**

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EBU Technical department

- Technical project groups with EBU members
- Management Committees
  - Network Management Committee (NMC), Spectrum Management Committee (SMC), Delivery Management Committee (DMC), Production Management Committee (PMC)
- Permanent staff in headquarters in Geneva

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ISDN is dying !

Radio and TV broadcasters use ISDN for sound contribution

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Radio contribution needs

- High quality and reliability
- Low delay
- Temporary and fixed circuits
- ‘Last minute’ circuit setup
- Wide geographical availability
- Simple use
- Low cost

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Type of equipment

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

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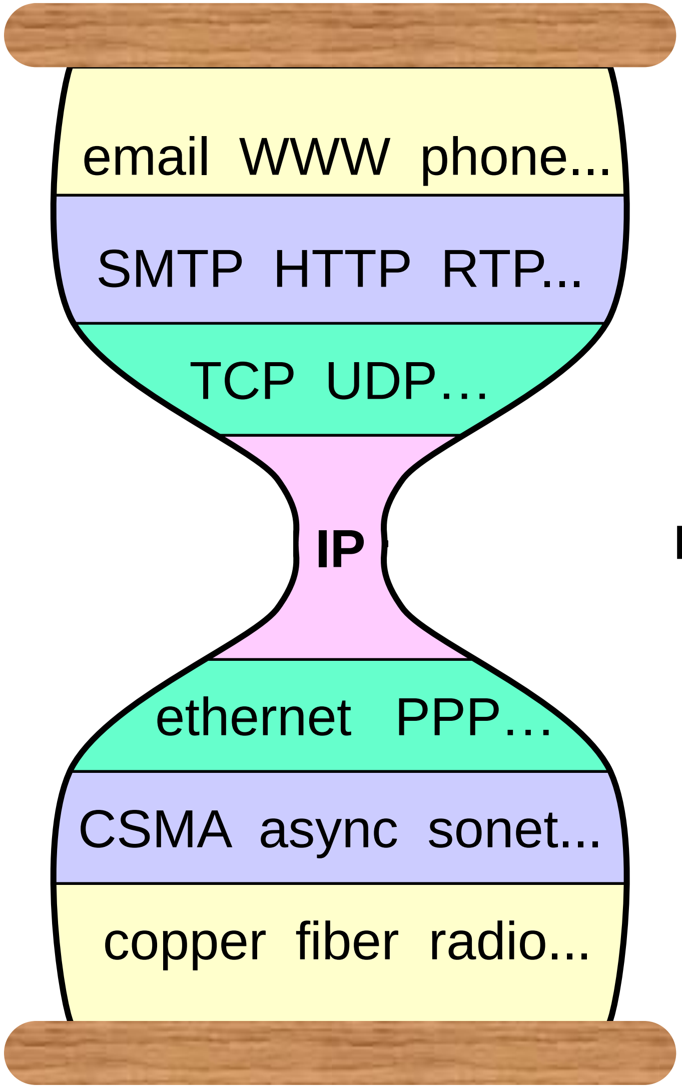
Why IP ?



Layered approach

Source: http://www.cs.columbia.edu
(Henning Schulzrinne)

From synchronous to asynchronous...

- Circuit based
- Synchronous (clocked)
- Fix bandwidth, delay
- Example: ISDN, SDH, E1,..

- Packet based
- Asynchronous (no clock)
- Best effort delivery
- High flexibility

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EBU project group N/ACIP

- Audio Contribution over IP project group
- Interoperability standard
  - Common framework defined by manufacturers and EBU members
  - The standard is based on Internet Engineering Task Forces (IETF) standards (RFC documents)
- Best practices for the deployment of Audio over IP
- Recommendations to manufacturers
- Tests and measurements on networks and end units

Why a new standard ?

- **Audio over IP solutions already exist ?**
  - **Audio LAN solutions used in studios**
  - **Internet streaming: shoutcast, realaudio, ...**
- **Broadcaster needs**
  - **Choice of audio coding algorithms depending on application**
  - **Connection over wide area networks**
  - **Bidirectional links (low delay)**
  - **Dynamic links, automatic negotiation**
  - **INTEROPERABILITY**

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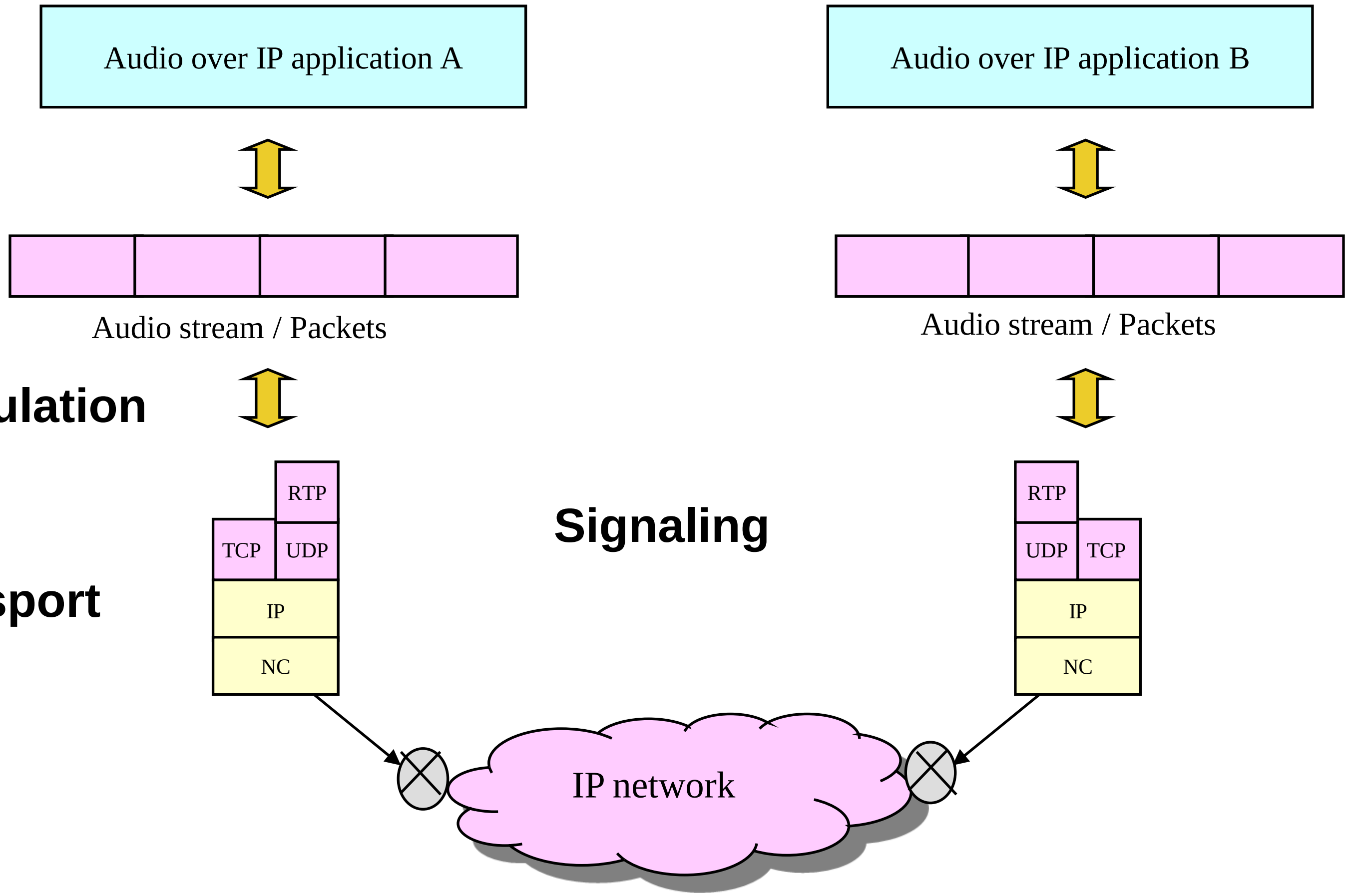
Overview

Audio coding

Encapsulation

Transport



Network profiles

Meetings with manufacturers

Meetings during IBC

More then 15 manufacturers

Consensus on interoperability

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Manufacturers

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

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EBU TECH3326 specification

- Audio coding formats
- Transport protocols
  - RTP: Realtime Transfer Protocol on top of User Datagram Protocol (UDP)
  - TCP (Transmission Control Protocol) is optional
- RTP media payload formats according to RFC specifications
- Signalling protocols
  - SDP : Session Description Protocol (RFC4566)
  - SIP : Session Initiation Protocol (RFC3261)
  - SAP : Session Announcement Protocol (RFC2974)
  - Offer/Answer model for SDP (RFC3264) for codec negotiation

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Audio coding

- Minimum set of codecs for compatibility
- A: Mandatory codecs (A minimum set, always to be found)
  - G.711
  - G.722
  - MPEG-1/2 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, Dolby AC-3...

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N/ACIP interoperability

- Interoperability document published: EBU TECH3326
- Website: http://www.ebu-acip.org
- Open source reference implementation by IRT/BBC R&D in development
- No certification !
- Plug tests workshop in February 2008

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Plug test workshop at IRT in Munich

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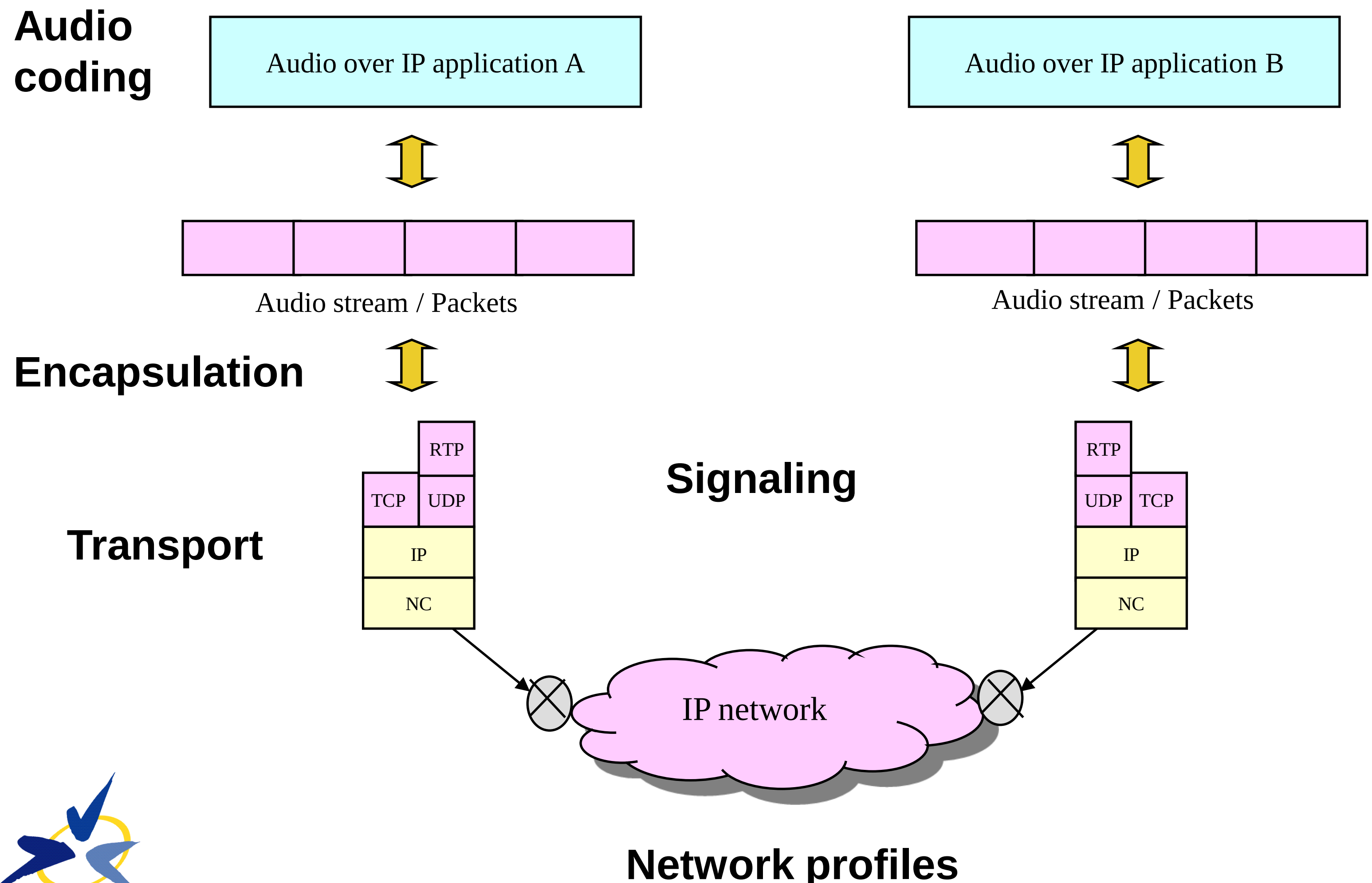
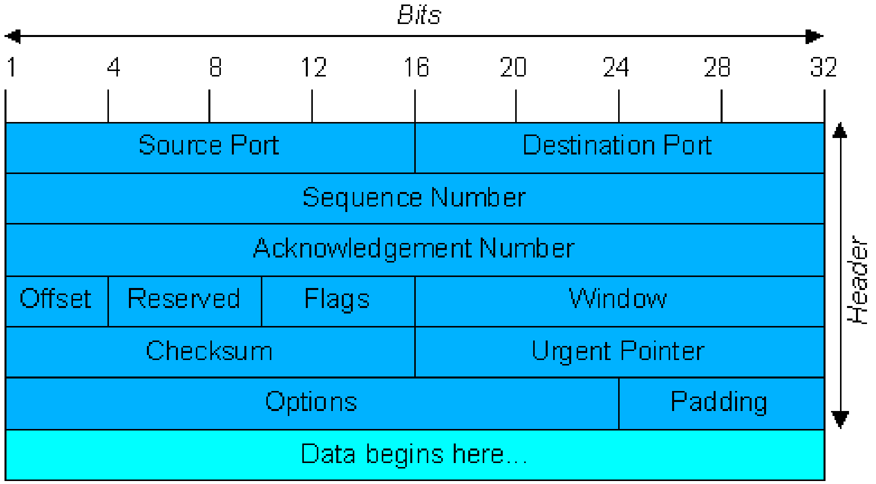
Overview


Audio coding
Audio over IP application A
Audio over IP application B
Audio stream / Packets
Audio stream / Packets
Encapsulation
RTP
TCP
UDP
IP
NC
RTP
UDP
TCP
IP
NC
Transport
Signaling
IP network
Network profiles

TCP: Transport Control Protocol



- Statefull: connection establishment, control, termination
- Reliable delivery with acknowledgement, retransmission
- Congestion control: send rate adaptation based on congestion estimation
- TCP used for HTTP (web), FTP application layer protocols
- Long delay

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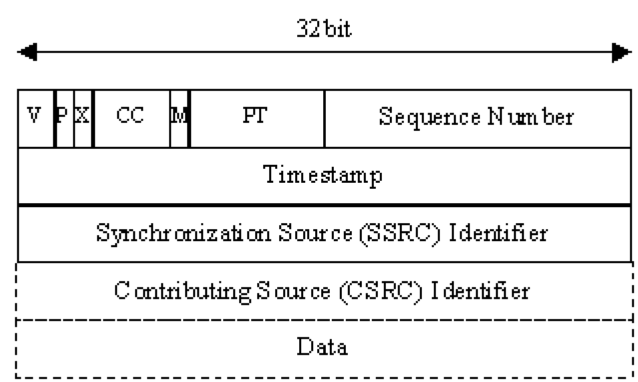
RTP: Real-time Transport Protocol

- Application layer protocol designed for audio and video real-time transport (RFC3550)
- RTP packets are encapsulated in UDP
- Timestamps: sampling instant for the first media byte in packet
  - Useful for clock recovery
- Sequence number
  - Useful for loss detection, reordering
- PT: Payload type
  - Useful for packet content description
- SSRC, CSRC: source identification
- Total overhead:
  IP+UDP+RTP headers= 40 bytes



No flow control

No protection against losses

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RTP associated protocols

- **RTCP: RTP Control Protocol**
  - **Sender receipt: used for synchronisation**
  - **Receiver receipt: used for loss notification to the source**
  - **New extended RTP profile for RTCP based feedback: RTP/AVPF (RFC4585)**
    - More immediate RTCP feedback

- **SRTP: Secure RTP**
  - **Encryption and authentication features**

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RTP Payload format (media encapsulation)

- **Definition of media frame encapsulation into RTP packets**
  - **Important for interoperability**
  - **Impact on performance with losses (packet independence, ms of media in a packet) and overhead**
- **Some payload formats**
  - **RFC2250: MPEG-2 (Transport stream or Elementary Streams)**
  - **RFC3640: MPEG-4 Elementary Streams (MPEG-4 AAC audio,..)**
  - **Others: http://www.ietf.org/html.charters/avt-charter.html**

RTP Header (12)	MPEG-2 TS (188)	MPEG-2 TS (188)	MPEG-2 TS (188)	MPEG-2 TS (188)	MPEG-2 TS (188)	MPEG-2 TS (188)	MPEG-2 TS (188)

Source: Cisco

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IP networks impairments

- **Packet loss**
  - **Network congestion**
  - **Line errors: 1 error = 1 lost packet**
    No packet with errors because of UDP/TCP checksum
- **Latency: end to end fix delay**
  - **Buffering delay**
  - **Propagation time**
- **Jitter: delay variation**
  - **Statistical multiplexing (queuing)**
  - **Route change, route lookup in routers**
- **Reordering: packets received in a different order as sent**
  - **Route change**

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Packet loss recovery

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

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Signalling

- **Signalling and control of session**
  - **Parameters from the stream: port, audio coding, bitrate, mode,...**
  - **Control: connection setup and termination**
  - **Parameter negotiation**

Signalling protocols

- RTSP: Real Time Streaming Protocol (RFC2326)
- SDP: Session Description Protocol (text) (RFC4566)
- SAP: Session Announcement Protocol (RFC2974)
  - Multicast, unidirectional
- SIP: Session Initiation Protocol (RFC3261)
  - Bidirectional, used for Voice over IP

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SDP: Session Description protocol

```
v=0
o=alice 2890844526 2890844526 IN IP4
host.anywhere.com
s=
c=IN IP4 host.anywhere.com
t=0 0
m=audio 49232 RTP/AVP 98
a=rtpmap:98 L16/48000/2
```

Example:
Linear audio 16 bits
48kHz stereo

- File format
- Describe content of RTP sessions
- Need a transport protocol => SIP, SAP or others (ftp, ...)

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SAP: Session Announcement Protocol

- Simple transport format
- No session management, just information
- Messages sent regularly (example every minute)

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SIP: Session Initiation Protocol (RFC 3261)

- Session management (establish, maintain, end)
- Commonly used for voice over IP
- Hosts listen to messages and respond
- Audio coding negotiation
  - Base on RFC3264 (Offer/Answer model with the SDP)
  - Caller propose a list of available audio coding algorithms
  - Receiver respond with available possible audio coding algorithms

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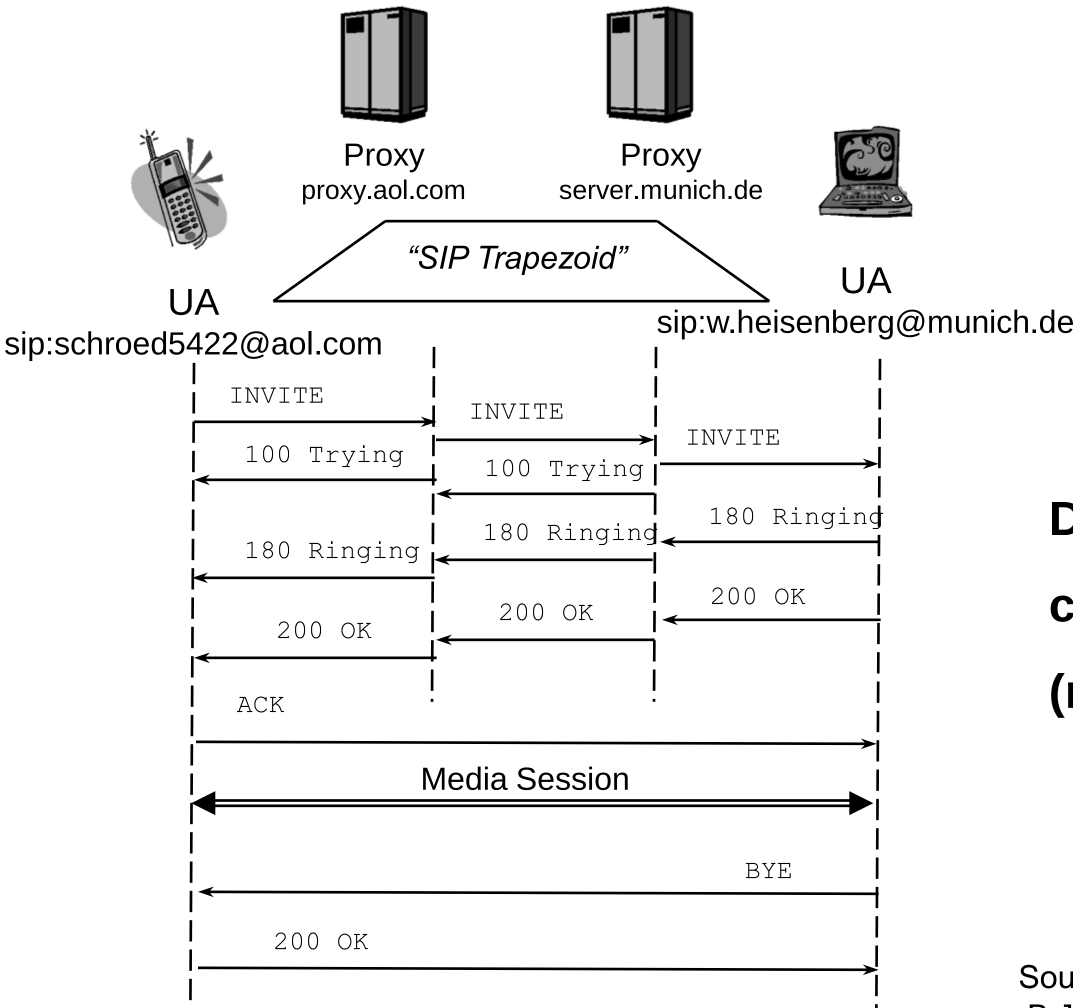
SIP call flow example



Direct host to host call are possible (no proxies)

Source: "Introduction to SIP"
B.Johnston, H. Sinnreich

Network Issues

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
  - Even it is working well in many situations
- Guarantee is possible only on dedicated managed IP networks

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IP ≠ INTERNET

Managed IP networks providers

- QoS: Quality of Service
- QoS can be expressed in Service Level Agreement (SLA) contracts with providers
- Requirements must be clearly expressed
  - Performances (latency, jitter,..)
  - Availability (99.9% 99,99%)
  - Provisioning delay (1 week ? 1 month ?)
- Measurement method definition is important
  - QoS classes validation, contracts validation

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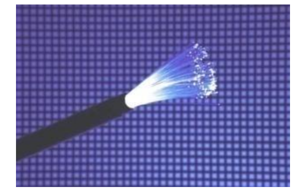
IP ≠ Internet

Last mile access

- Last mile to the end user
- Fiber optic
  - High quality but expensive



- Copper with xDSL
  - ADSL: Asymetrical uplink/downlink (bandwidth), SDSL: Symetrical 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

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Delay chain

- Audio encoding/decoding delay (~ 1- 500 ms)
  - Audio encoding algorithm + implementation
- Packetisation delay (~ 1 – 100 ms or more)
  - Trade-off: Delay ⬄ Overhead, packet rate (network stress)
- Network delay (~ 1 – 500 ms)
  - Due to distance, buffering, packet routing, (tunnelling)
- Receive buffer delay (~ 10 – 500 ms or more)
  - Trade-off: Delay ⬄ Network quality (jitter compensation)
- + Optional FEC encoding and decoding

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Conclusion

- EBU TECH 3326 interoperability framework for audio over IP
  - Common work from manufacturers and broadcasters
  - Early implementations are already available from some codecs manufacturers
  - Possible interoperability with SIP based VoIP systems
- EBU TECH 3329, tutorial on Audio contribution over IP
- Rely on managed IP networks

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Download links on http://www.ebu-acip.org

Thank you ! – Questions ?

Mathias Coinchon

EBU

coinchon@ebu.ch

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