

# COMPSCI 314 S1 C

Streaming Audio & Video,  
VoIP, Internet Telephony

*Not in 2006 exam!*

## Audio Compression & Encoding

- A **codec** **encodes** audio/video/etc. for transmission, and **decodes** it upon reception. Each codec's behaviour is defined by a standard
- Audio codecs
  - Use various physical models [Halsall A.2.4]
  - Trade *bandwidth* against *quality* (measured using MOS)
  - G.721, G.722: ITU-T standards, used in telephone networks
  - MPEG audio: ISO standard, MP3 “CD quality at 64kb/s per channel”
  - Skype: proprietary, software implementation said to be “very good”
- Standards summary: Halsall Table A.5

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## Video Compression

- Motion JPEG simply presented a sequence of JPEG frames
  - Easy to edit, not very much compression
- MPEG uses sequences of I, P and B frames to get higher compression
  - I frames carry a whole picture
  - P frames carry forward differences, use those + preceding frame(s) to reconstruct the pictures
  - B frames, differences between preceding and following frame. More work to encode/decode, highest level of compression
- MPEG2 (ISO 13818) for studio-quality audio and video
  - Four levels of video resolution

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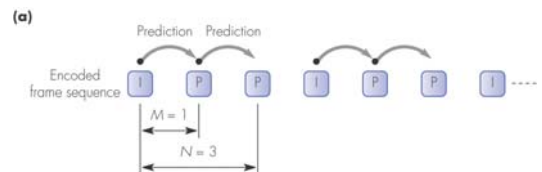


Figure A.17 Example frame sequences with: (a) I- and P-frames only

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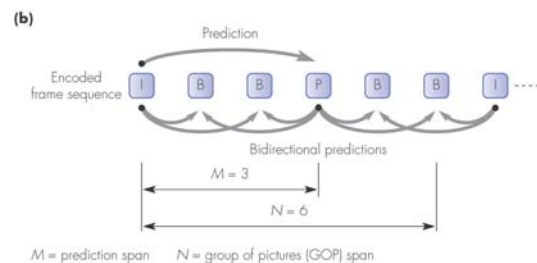


Figure A.17 Example frame sequences with: (b) I-, P- and B-frames

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## RTP: Realtime Transport Protocol

- Halsall 7.5.1
- An **application** protocol, carried over UDP
- Each packet has a **sequence number**
  - Used to detect lost and out-of-sequence packets
- And a **timestamp**
  - Returned to source, used to track RTT
- RTP receiver delivers audio/video via a **playout buffer**
- Comment about **jitter**: it can mean different things
  - Computer Science: difference in successive transit times
  - Electrical Engineering: differences from expected arrival time
  - Safer to use the term **delay variation**

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Figure 1 illustrates the RTP protocol architecture. It shows two nodes, each with an 'Audio and/or video in/out' interface, an 'Encoder/Decoder' block, a 'Sending RTP' block, a 'Receiving RTP' block, and a 'UDP' block. The nodes are connected via an 'IP' network. On the right, a detailed view of a packet shows its structure: 'Digital stream' (Block 1, Block 2, ...), 'RTP header' (Block 1), 'RTP packet', 'UDP header' (Block 1), 'UDP stream', and 'IP header' (Block 1), 'IP packet'.

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(b)

Diagram illustrating the structure of an RTP packet header and payload:

- Header Fields:**
  - V (1 bit):** Version
  - P (1 bit):** Padding
  - X (1 bit):** Extension flag
  - CC (2 bits):** CSRC count (N ≤ 15)
  - M (2 bits):** Marker bit
  - Payload type (5 bits):** Timestamp
  - Sequence number (16 bits):** Synchronization source (SSRC) identifier
  - Contributing source (CSRC) identifier (1):** Contributing source (CSRC) identifier (1)
  - Contributing source (CSRC) identifier (N):** Contributing source (CSRC) identifier (N)
- Payload:** Data (1-N audio and/or video frames)

Legend:

- V = version
- P = pad
- X = extension flag
- CC = CSRC count (N ≤ 15)
- M = marker bit

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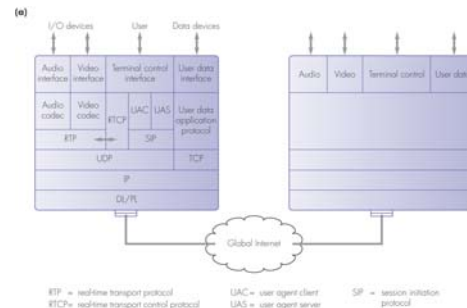
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- Streamed video
  - QuickTime, RealAudio, Windows Media ...
- Early work
  - Desktop audio/video conferencing, CuSeeMe, etc.
- Telephony
  - H.323 (ITU-T)
  - SIP: Session Initiation Protocol
    - User names like email addresses, e.g. [sip:nevil@auckland.ac.nz](mailto:sip:nevil@auckland.ac.nz)
  - E.164 addresses: ITU-T recommendation for PSTN numbering
    - *enum* maps E.164 numbers to domain names, e.g.
      - +64 373 7599 maps to 9.9.5.7.3.7.3.4.6.e164.arpa

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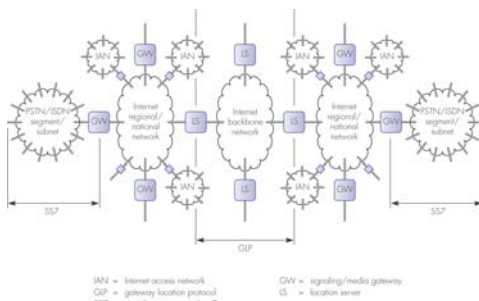
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- A Peer-to-Peer (P2P) network for voice calls
- Uses that P2P network for 'best' routing of voice calls over UDP (or TCP if it can't find a non-firewalled UDP path)
- Uses its own user namespace - skype usercodes
- Has a good audio codec of its own
- Has gateways to PSTN
- Cheap and widely used

but ...

- Proprietary!

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