Interactive, real-time media flows

- Audio/video flows:
  - streaming audio/video
  - use buffering at receiver
- Interactive real-time:
  - only limited receiver buffering
  - delay <200ms
  - jitter <200ms
  - keep loss low
- Effects of loss:
  - depend on application, media, and user
- Audio:
  - humans tolerant of “bad” audio for speech
  - humans like “good” audio for entertainment
- Video:
  - humans tolerant of “low” quality video for business
  - humans like “high” quality video for entertainment
- Audio and Video have different requirements
- Audio – video sync:
  - separate flows?

Audio

QoS requirements

- Delay < 400ms:
  - including jitter
- Low loss preferable:
  - loss tolerant encodings exist
- Data rates:
  - speech ≤ 64Kb/s
  - “good” music ≥ 128Kb/s

- Time domain sampling
- Example – packet voice:
  - 64Kb/s PCM encoding
  - 8-bit samples
  - 8000 samples per second
  - 40ms time slices of audio
  - 320 bytes audio per packet
  - 48 bytes overhead
    (20 bytes IP header)
    (8 bytes UDP header)
    (20 bytes RTP header)
  - 73.6Kb/s

Example audio encoding techniques

G.711

- PCM (non-linear)
- 4KHz bandwidth
- 64Kb/s

G.722

- SB-ADPCM
- 48/56/64Kb/s
- 4-8KHz bandwidth

G.728

- LD-CELP
- 4KHz bandwidth
- 16Kb/s

G.729

- CS-ACELP
- 4KHz bandwidth
- 8Kb/s

G.723.1

- MP-MLQ
- 5.3/6.3Kb/s
- 4KHz bandwidth

GSM

- RPE/LTP
- 4KHz
- 13Kb/s

Video

QoS requirements

- Delay < 400ms:
  - including jitter
  - same as audio
  - inter-flow sync
- Loss must be low
- Data rate – depends on:
  - frame size
  - colour depth
  - frame rate
  - encoding

Example video encoding techniques

MPEG1

- upto 1.5Mb/s

MPEG2

- upto 10Mb/s (HDTV quality)

MPEG4

- 5-64Kb/s (mobile, PSTN)
- 2Mb/s (TV quality)
- MPEG7, MPEG21

H.261 and H.263

- $n \times 64$Kb/s, $1 \leq n \leq 30$