Quality Elemets mainly from a networking perspective

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Outline

- Codecs
- Delay
- Jitter
- Packet loss

Codecs

- To reduce the bandwidth costs of voice transmission
- Coding often result in a perceivable degradation because the speech codec can be viewed as a weak nonlinear time-variant system.

Two types of distortions:

- ✓ Linear (frequency)
- ✓ Nonlinear

In addition:

- ✓ Codecs tandeming → additional impact (multiple coding)
- ✓ Impact of coding algorithms is described by Equipment Impairment Factor $\boldsymbol{I_e}$ in E-model.

Overall Transmission Delay

Results from:

- **Algoritmic delay** (applied by encoder and error correction schemes)
- **Delays introduced by the network**
- **Delays introduced by jitter-buffers and decoder**
- **Delays introduced by additional signal processing components** (like EC)

Overall Transmission Delay

Delay Source	Typical Range (ms)
Recording	10–40
Coder	10–20
Internet delivery	70–120
Jitter buffer	50–200
Decoder	10–20
Total	150–400

Table 1: Contributions of VoIP processing steps to overall transmission delay (adopted by Raake)

Overall Transmission Delay

- Large amount of delay reduce the communicability as well as degrade speech quality.
- Impact of delay depends on conversation situation and also the experience of users with delay-type degradations.

Jitter

- Because of different traffic load at subsequent links of connection, the packets within one talkspurts may arrive at destination with varying delay (jitter)
- Defined as a difference in time domain between two consecutive packets
- Has a significant impact on speech quality
- Has to be compensated for

Jitter

- The compensation is usually done in the network at the receiver-side by jitter buffers (more about jitter buffer principles in previous part).
- The above-mentioned process introduces additional delay and potentially loss. (loss/delay trade-off)

Packet loss

Packet loss can occur in the network or at the receiver site due:

- ✓ Excessive delay in case of network congestion or traffic load changing → dropped packets
- ✓ Bit errors introduced on link (mainly in radio networks) → lost packets (not entire packet is lost, only part of its)

Packet loss

The impact of packet loss depends on several factors:

- ✓ **Packet Loss Location** (Loss obtained in talkspurt or silence periods)

- ✓ **Packet Loss Distribution**
  - ❑ **Independent (Random)** → Bernoulli model
  - ❑ **Dependent (Bursty) losses)** → Gilbert model

Packet loss

- ✓ **Packet Size**
  - ❑ Sizes from 10-60ms
  - ❑ Larger packets → increase of transmission delay and lower speech quality in case of packet loss

- ✓ **Packet Loss Recovery Mechanisms** (like PLC, FEC, LBR)

References

- Raake, A.: **Speech Quality of VoIP: assessment and prediction**, John-Wiley & Sons. UK, 2006, ISBN 0-470-03060-7.

Thank you for your attention !

Questions ?