Codec Requirements

draft-ietf-codec-requirements-00.txt

Jean-Marc Valin, Octasic Inc.
Koen Vos, Skype

Introduction

- Requirements draft recently adopted as WG item
- Still a work in progress (not final)
- We present a summary of the current draft

Applications

- Point-to-point calls
- Conferencing
- Telepresence
- Teleoperations
- In-game voice chat
- Remote music performances/lessons
- (Others)

Constraints of the Internet

- The codec must:
  - Be able to re-synchronize after a lost packet
  - Have multiple bitrates with no switching artefacts
  - Have bounded decoding time (to prevent DoS)

Actual requirements

- Sampling rate:
  - 16 kHz to 48 kHz as main target
  - 8 kHz for compatibility
- Algorithmic delay:
  - 20-30 ms for most applications
  - <10 ms as “optional mode” for special applications
- Quality:
  - Better than Speex, iLBC, G.722, G.722.1[C]

Additional Considerations

- Packet loss concealment
- Complexity/footprint
- Low-complexity mixing for conferences
- Layered bit-stream
- Partial redundancy
- Time stretching/shortening
- Input robustness