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