Speech/Audio Coding for IP networks

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Abstract
From a user's point of view, packet loss and long delay severely disrupts speech intelligibility, even for a rather low loss rates.
This paper examines the existing codecs used for real time speech/audio applications through their design and depicts problems due to their
design properties which were tailored for usage in circuit switched and wireless networks.

Consequently, speech coders which encounter packet network properties intrinsically through their design, score clearly better results on the
perceived quality tests. Thus, enabling graceful degradation of perceived speech quality, sustaining its high quality levels for as high as 30% of packet loss ratio, unlike any of the existing speech coders. Additionally, certain techniques and methods of the new approach to speech/audio coding for IP networks are being discussed in the paper.