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An Enhanced Predictive Multipulse LPC Speech Coder at 2.4 kbits/s

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A 2.4 kbits/s Predictve Multipulse LPC coder,which could produce better speech quality than conventional LPC-10 coder, has been proposed in the paper. A simple multipulse LPC coder can not operate with acceptable good speech quality below 6–8 kbits/s, because to transmit a minimial number of multipulses amplitudes and locations requires at least 6 kbits/s. The performance of MPLPC coder degrades rapidly below this lower bound. The quality of a CELP coder also dramatically deteriorates below 4.8 kbits/s, beccause of insufficient representation of pitch harmonics,when a Gaussian excitation codebook is used.The main features of the proposed Predictive Multipulse LPC coder are as follows: 1. Input speech frames each with 180 samples are first divided into subframes, referred to as prediction intervals. The locations and magnitudes of three multi pulses optimized by a closed-loop search in first subframe are transmitted. Multipulses in other subframes are interpolated by a linear time varying synthesis filter to trace the time-varying characteristics of the vocal tract for consecutive subframes in one frame. The first subframe length and the multipulses are jointly determined by minimizing the mean-square error (MSE) between the orignal and the synthesized speech. 2. Speech signals are classified into voicing/unvoicing categories by an analysis-by-synthesis(ABS) method. Based on a speech database, a SNR threshold has been found for the V/UV classification. A frame is defined as an unvoicing one,if the SNR is less than 1.05 dB. This novel algorithm for classification has improved SNR by about 3 dB during transition periods,compared to a pitch predictive MPLPC coder. 3. Vector quantization for LPC parameters of tree structure with branching factor of 32 is used to tailor the proposed algorithm to work at 2.4 kbits/s. The two-stage tree codebook has reduced computation load down to 1/16 with only slightly increased memory requirement. Cepstrum distance (CD) with average 2.91 dB and informal listening test of several phonetically balanced sentences have shown that the proposed algrithm of an enhanced Predictive MPLPC coder could obtain better quality than present LPC-10 vocoder.