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Estimation of Late Reverberation Spectrum: Optimization of Parameters

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Abstract—Correction of speech signals distorted by reverberation is topical in building communications systems, automatic speech recognition systems, and hearing aids. The late reverberation suppression by the spectral subtraction method or the frequency correction method involves the need of estimating the late reverberation spectrum. Though the procedure of such estimation is generally developed, a number of uncertain items related to its optimization still exist. Recommendations elaborated in this study make it possible to optimize the estimation of late reverberation spectrum in terms of such criteria as the speech signal quality and the accuracy of automatic speech recognition by using computer simulation methods.

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INTRODUCTION

The speech signal indoors is practically always exposed to distorting effect of reverberation that results in significant deterioration of speech quality and intelligibility in communications systems, hearing aids and negatively affects the efficiency of automatic speech recognition (ASR) systems [1–4]. The late reverberation suppression by the spectral subtraction method with preliminary estimation of the power spectrum of late reverberation was proposed in [5]. The possibility of late reverberation suppression using the frequency correction method was shown in [6].

Unfortunately, the results obtained in [5, 6] are of preliminary character because the estimation parameters of the late reverberation spectrum are not optimized. Paper [7] initiated the elimination of this drawback and showed the existence of the boundary between early reflections and late reverberation optimal in the sense of such criteria as the speech signal quality and the accuracy of automatic speech recognition. The purpose of the current study is to refine the results of [7] and also to develop new recommendations for optimization of the estimate of late reverberation spectrum.

STATEMENT OF PROBLEM

A model of speech signal $y(t)$ distorted by reverberation can be presented in the form of convolution of "clear" speech signal $x(t)$ with the room pulse-response characteristic $h(t)$:

$$y(t) = \int_0^\infty h(v)x(t-v)\mathrm{d}v = x(t) \otimes h(t),$$

where "⊗" is the convolution symbol.

By using the Polek model [5] for pulse-response characteristic (PRC) of room

$$h(t) = \xi(t)\mathrm{e}^{-\delta t},$$

where $\xi(t)$ is the stationary white noise, $\delta = 3\ln 10 / T_{60}$ is the exponent of the decay rate of the room sound level, T_{60} is the time of reverberation, and separating areas in PRC $h(t)$ (Fig. 1) that correspond to the early and late reflections: