ABSTRACT

Archival audio recordings can be found in museums and national archives, as well as in home libraries. These recordings may have limited bandwidth and dynamic range. They can also contain noise and distortions. Analog recording media are prone to wear, damage and contamination. For example, magnetic media are prone to magnetic fields, while mechanic media, such as vinyl records, are susceptible to deformation during storage and wear as a result of multiple playback process. The contamination may be caused by mishandling of the recording media. These processes results in a further increase of the noise and distortions level

Reduction of the noise and distortions, and hence the reconstruction of archival audio recordings, is a non-trivial task. High quality reconstruction, removing as much as possible distortions with the least possible original signal damage requires high level of expert knowledge and intuition. The expert has to determine the type of recording and to choose noise reduction algorithms and their parameters. After the reconstruction the results have to be evaluated, and if they are not satisfactory, the process has to be repeated.

The aim of the work was to develop new algorithms that would improve the quality and reduce the effort of audio signal reconstruction, making it as simple as possible, so they could be used also by people with less experience. For this purpose the possibility of using non-uniform sampling methods and intelligent signal processing algorithms has been explored. High computational complexity of such algorithms causes long processing time which may be reduced using parallel forms of these algorithms. Many audio signal reconstruction processes can be implemented this way using multi-core processors. GPU, FPGA or ASIC chips may be also used for this purpose.

The audio signal is a special case of a one-dimensional signal. The system developed in order to reconstruct the audio signals should also be used, with modifications, in the analysis and reconstruction of other similar signals, for example, a measurements signal, in order to detect a failure in the measured system.

The dissertation consists of six chapters. The first chapter is a brief introduction to the subject of the dissertation.

The second chapter contains the description of the nature and the structure of the sound. Then the human auditory system is described. Its parameters determinate requirements, which audio recording devices have to meet. The history of sound recording and the overview of sound recording methods used in the past are then presented. The last part of the second chapter presents selected methods of sound signal description and analysis as well as issues related to the classification of archival recordings.

The third chapter discusses most common types of noise and distortions found in archival sound recordings. Harmonic quasistationary noise, *wow*-type distortion caused by media feeding speed changes, noises caused by media impurity and recording devices imperfections, pulse noises – pops and crackles caused by mechanical damage of records, linear and non-linear distortions as well as other types of noises and distortions are described in this chapter.

The fourth chapter presents new methods and algorithms developed by the author for reduction of the noises and distortions described in the third chapter.

The fifth chapter discusses the hardware and software environments which can be suitable for implementation of the algorithms described in the fourth chapter. A program developed in Object Pascal language is also presented. The program uses algorithms presented in the fourth chapter to restore archival recordings.

The research and tests presented in the dissertation prove the possibility of using intelligent algorithms and non-uniform sampling algorithms in order to improve the process of the archival sound recordings reconstruction.