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ACQUAINTANCE WITH SCIENTIFIC LITERATURE AND THEIR ANALYSIS WITHIN THE TOPIC OF DIGITAL AUDIO SIGNALS AND SOUND FREQUENCY CORRECTION

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ABSTRACT

Importance and application of digital audio signal analysis. The essence of digital television, the operation of sound signal synthesis to correct the psychophysiological state of a person, the results achieved by digital signal processing. Analysis of an automatic signal correction system to improve the sound quality of loudspeakers and rooms. Aspects of predictive modeling in acoustic design. Learning the basics of digital signal processing in telecommunication systems. Information systems based on biometric identification of users. Cluster information speech units models. Consider changing the timeline. Music signals, Digital signal processing filtering algorithm Sound equalization using Matlab.

Keywords: frequency, correction filters, electroencephalogram, bioelectric activity of the brain, algorithm, synthesis, harmonic sound, Fourier transform. formation, spectral analysis, spectrum extremity, , program complex.

INTRODUCTION

Digital television book presents the basics of analog and digital television theory. Signal shaping factors, methods and basic principles of video and audio data compression in digital television are widely considered. The structure and performance factors of the set of JPEG and MPEG standards for image signals and sound compression, as well as DVB, ATSC, ISDB, MMDS, LMDS, MVDS digital television standards are considered. In addition, the structure and principles of operation of receivers used in digital television broadcasting, their base of elements and the current state of digital television in Uzbekistan are presented.

Sound signal synthesis to correct the psychophysiological state of a person Research and development works dedicated to the software to synthesize a sound signal to correct the psychophysiological state of a person were considered. Spectral analysis was performed to ensure the synthesis of sound effects of the required frequency. Taking into account the separation of the local extrema of the obtained spectral function of the bioelectric signal, the necessary frequency of the sound effect was considered in accordance with the extrema of the spectral function and the synthesis, and this frequency was used to smooth the sound effect.

Algorithmically, this problem is solved as follows.

- spectrum of the background bioelectrical signal is analyzed;
- local extremes are determined in the spectrum;
- the required multiplicity coefficient to create a sound effect is checked for the selected frequency range.
- sound effects are synthesized.

Below is a description of the blocks and algorithms of the list solving software package.

Algorithm for calculation of spectral function. The spectral function is calculated for the time period T. Registration of bioelectric signal. An interval is set to determine the spectral function. spectral decomposition - the minimum time period for recording the bioelectrical signal, its magnitude is determined by the condition of providing the spectrum of the required resolution, which operates according to the frequency.

Digital signal processing. The manual discusses the basic methods of numerical theory. the type of signal processing used in signal preprocessing is common. The hand-made material is divided into 6 parts, aimed at the development of tasks given in the sixth section, which aims to provide students with practical skills in applying the basic principles of digital signal processing and in-depth analysis of its methods.

In recent decades, the rapid development of computing technology has led to the widespread introduction of digital information processing methods. Work is being carried out in almost all areas of scientific research and national economy. In addition, among the various applications of computing tools, we can determine one of the most important places in technology based on digital processing systems. Signal processing (DSP) remote sensing, biomedical research, solutions for navigation, communication, and radiophysics problems in aerospace and marine facilities are under way. In digital optics and a number of other applications, the concept of primary and secondary signal processing is analyzed in terms of the primary (primary) and secondary stages of signal processing in DSP tasks. This is due to the fact that, in general, at the system input, the DSP receives a mixture of the useful signal x(t) s(t), some noise n(t) and various interferences of different nature p(t)X (t) = S (t) +p (t) + n, where n (t) is a characteristic of the technical device, and p (t) is some disruptive effect of the physical environment in which the signal propagates (for example, attenuation).

The following tasks of digital signal processing are distinguished: to ensure optimal signal reception, what is meant here is to provide maximum suppression of various types of noise and work with noise.

Determining digital parameters of signals - measured by energy, average power, rms value, etc. What exactly is meant by ensuring optimal signal reception ensures maximum suppression of different types of interference, because in the general case, a mixture of them arrives at the input of the receiver: Decomposition of signals into a certain set of elementary compositions that can be considered separately or together in the future for output and also solves the inverse problem of signal synthesis.

Automatic signal correction to improve the sound quality of loudspeakers and rooms Acoustic correction of speaker and room characteristics using digital signal processing approaches is becoming an important part of the audio reproduction system design process. In recent years, great efforts have been made in this field, which led to the development of various algorithms for the acoustic correction of speaker and room characteristics using IIR and FIR filters. Based on this, based on the main advantages of traditional approaches, this paper presents a completely new automatic signal correction tool.

One of the main key elements of the proposed structure is the ability to achieve optimal correction of amplitude-frequency and phase-frequency characteristics with the combined use of IIR and FIR filters. In particular, IIR filters can be used to correct the signal in the lowfrequency region, which improves the amplitude-frequency response, while FIR filters can be used to correct the phase-frequency response. At the end of the article, an example of acoustic correction of room characteristics is presented to demonstrate the effectiveness of the presented approach. Predictive modeling in acoustic design The topic of research is the acoustic design of buildings. The purpose of the work is to study the application of the method of the complex article, taking into account the important aspects of the design of the acoustic room. Energy spectral and dynamic signal characteristics are used in this. Measurements are carried out in the famous Roko Halls of Moscow on pieces of sound material recorded in several fragments.

The results of his research show that there are two types of sound signal in any sound signal, which determine the evaluation of sound quality in many ways, and these formations are semantic (semantic) and aesthetic (emotional). It is suggested to use the following sound parameters during the measurements.

Differential relative average power, which indirectly determines the gradation of sound volume, the decay of the analytical envelope of the sound signal, which determines the sharpness and clarity of the sound quality; tonal purity evaluates the results using processing. Recordings of broad-band symphonic works of the total length made by the research take approximately 12 hours for the parameters used (for each of the halls), which exceeds more than a hundred intervals.

Fundamentals of digital signal processing in telecommunications systems. Provides a basic course in the fundamentals of digital signal processing. It examines spectral analysis and digital filtering methods, digital filter synthesis algorithms, the fundamentals of discrete signals and systems theory, as well as some aspects of the application of digital signal processing in communication systems are highlighted and listed.

Information systems based on biometric identification of users. Cluster information speech units models.

Dissertation research: the object of biometric identification of users' voices is the system.

The subject of the dissertation research: voice processing using a cluster model of elementary speech units by biometric user identification methods and algorithms.

The aim of the dissertation work is to improve the efficiency of the development of methods and algorithms for solving this problem based on the identification of voice information system users cluster model of elementary speech units.

To achieve this goal, the thesis was presented and the following tasks were solved and additions were proposed.

1. Research methods, algorithms, identification systems, voice of users and analysis of the weakness of modern voice biometric systems, to whom different individual voice characteristics;
2. Ways of development, formation of falsification of sound standards, users based on the cluster model of elementary speech units;
3. Development of the method and principle of statistical analysis of phonemes, collection of information to solve the problem of voice identification;

4. Biometric identification, taking into account the possibility of protecting the system against various types of attacks, in the context of the variability of speech, with the development of user identification algorithms;
5. Development of a software package for biometric voice identification of information system users; experimental studies of the developed identification complex, making recommendations for use in real practice working conditions.

Timescale Shifting Considerations Timescale Shifting (TSM) of musical signals is the act of speeding up or slowing down the playback speed of the signal without changing its pitch. In digital music production, TSM has become an indispensable tool that is now included in a wide range of music production software. Music signals are diverse and include harmonic, pulse and temporal components. There is no single TSM due to the wide range of acoustic and musical properties that can equally cope with all types of audio signals.

Digital Signal Processing Filtering Algorithm Using Matlab is an analysis of different filtering algorithms for DSP for audio equalization and how they are implemented using matlab. In addition, the study tries to use one of the basic, user-friendly, graphic equalizers in a convenient way to implement signal processing algorithms. At the same time, research tries to focus on comprehensive and practical ways of learning. Understanding the use of DSP filtering. The equalizer is implemented using the tools provided by Matlab and this software belongs to the field of DSP. Using Matlab in DSP has become the standard method. A designed equalizer consists of a set of filters that work together in their various functions. The fundamental frequencies of an analog equalizer are used to create an effect equivalent. This application includes audio data processing, Matlab. allows you to manage and display filter parameters restores a graphic image of filter channels on the screen of a working computer.

In short, several sound frequency calculation programs have been developed based on the proposed algorithm. based on the extreme frequencies of the bioelectric signal spectrum. The developed program is based on the developed algorithm included in the hardware-software complex for psychophysiological audio correction. The developed program is included in the hardware-software complex for psychophysiological audio correction Human condition Analysis of existing methods, algorithms, systems was carried out Identifying users by voice, which revealed their vulnerability in various ways to falsify individual characteristics and the ability to carry out attacks.

The construction of a standard information center allowed to reduce the number of errors in the identification of user voices with further clustering of voice standards. The result of the work is a list of the main spectrum characteristics and methods of determining them using an audio editor and spectrum analyzer plug-ins. Application of this technique allows to determine and analyze 8 test frequencies, temporal and energy characteristics of speech signals.

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