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METHODOLOGY OF SPLITTING AN AUDIO FILE BY FRAMES

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ABSTRACT

In the modern world, television, as a mass media, plays a very important role in the life of every person, because it allows you to convey information to him from almost any part of the world. At the same time, digital technologies are being actively introduced in the television industry. However, when converting an analog television signal to digital form, the output stream of video data can reach 240 Mbit/s [1, 2], which is 108 GB per hour of transmission. This requires a communication channel with a bandwidth of 120 MHz for their transmission and, accordingly, does not allow transmitting such a huge amount of information either over standard 8 megahertz radio channels, or even more so over cellular communication channels with a bandwidth of 2 Mbit/s. In addition, the operations of recording and reproducing such large amounts of information on a personal computer are still fraught with serious difficulties. Therefore, to coordinate the parameters of signals and transmission channels, various methods of video compression are used, based on the elimination of redundant information from TV images. If you don't use them, the average movie will take up hundreds of gigabytes.

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In this regard, one of the most urgent tasks in the field of audio-video data processing is the development and improvement of methods for compressing audio-video data, taking into account the elimination of temporary redundancy of TV images and audio accompaniment. This problem is very relevant in the conditions of the global financial crisis, in conditions of limited frequency resources. In addition, it becomes possible to significantly reduce the time of preparing television reports for broadcast directly from the event sites by transmitting signals from TV cameras directly to the installation hardware of television centers over cellular networks, and the need to use expensive and not always available broadband communication channels disappears.

KEYWORDS:- Algorithms, splitting, audio file, frames, video images.

INTRODUCTION

As a result of the conducted research, it was found that when compressing audio files with MP3, AAC, OGG codecs, the best quality/volume ratio (data volume) is provided by the MP3 standard [1-3]. At the same time, the main compression in these codecs is performed on the basis of psychoacoustic signal processing, which removes from the stream those components that the human ear does not perceive. However, with high compression coefficients, signal losses significantly increase, which creates distortions of the restored sound in the form of clicks, noises, loss of high frequencies, rustles. A good sound quality of the above formats is provided at an output stream speed of 128-256 kbit / s. At lower speeds, it drops noticeably.

METHODOLOGY

The results of the analysis showed that these methods of audio signal compression do not take into account the temporal repeatability of individual fragments of the audio signal, as shown in Fig.1 [1-3].

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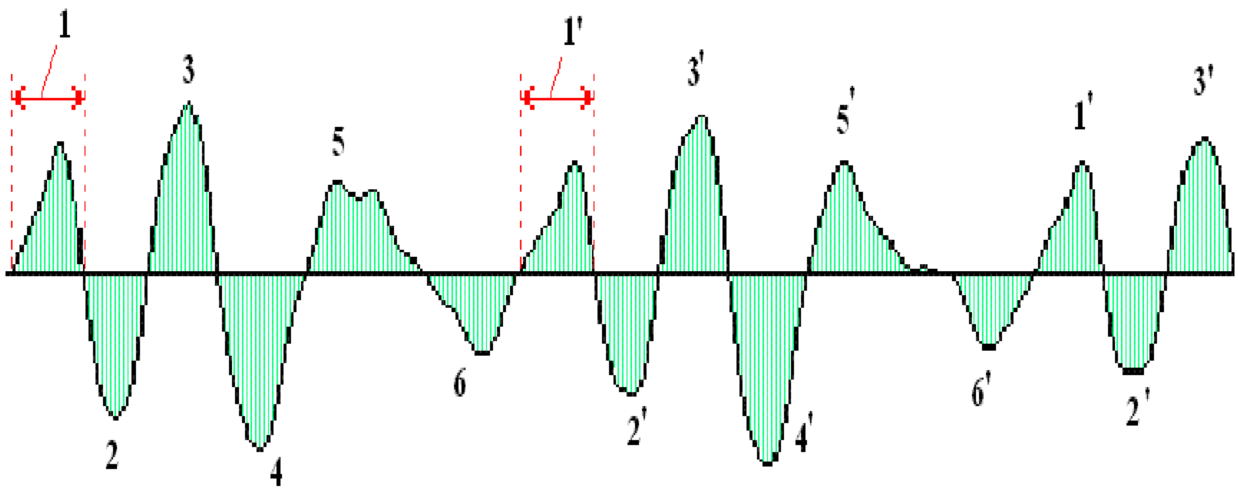

Fig 1. Temporary representation of the audio signal

As can be seen from the figure, in the given audio signal, similar repeating regions 1 - 6 and 1`- 6` can be distinguished, which we will call audio frames. Thus, if there are identical audio frames in the stream, then they can't be transmitted, but only a pointer to a reference frame of this type can be transmitted. If the correspondence of the audio frames is not complete, (5-5), then the value of the fractal difference is transmitted, which in its magnitude will be less than the values of the samples of the frame itself and therefore less code bits are required to transmit this frame. At the same time, the more such frames are found, the greater compression of the audio stream can be obtained while maintaining the sound quality of the restored signal.

In order to reduce signal distortion at bitrates less than 128 kbit / s, an audio signal compression algorithm was developed based on eliminating the temporary redundancy of the audio signal [6], the essence of which is to search for relatively similar signal fragments (audio frames), and when such fragments are found in the stream, replace the reference frame with a reference. At the same time, the larger the size of the audio frame, the greater the "gain" when compressing we get. Figure 2 shows the splitting of the original audio signal into audio frames based on the signal

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transition through 0.

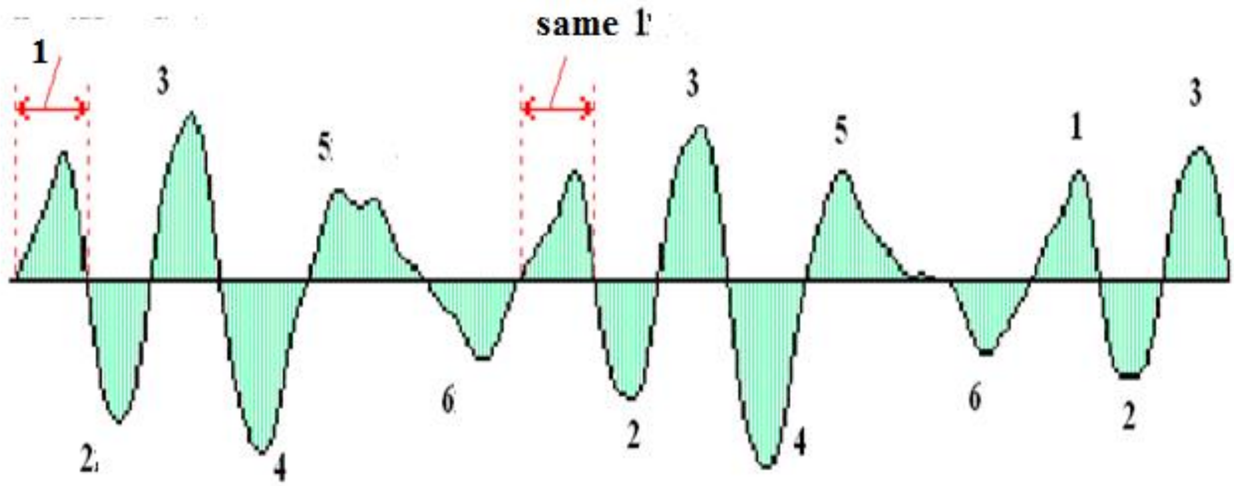

Fig. 2. Dividing the signal into audio frames.

This method does not use spectral signal transformation and is based only on the search for repeating fragments, so it is similar in structure to fractal image compression.

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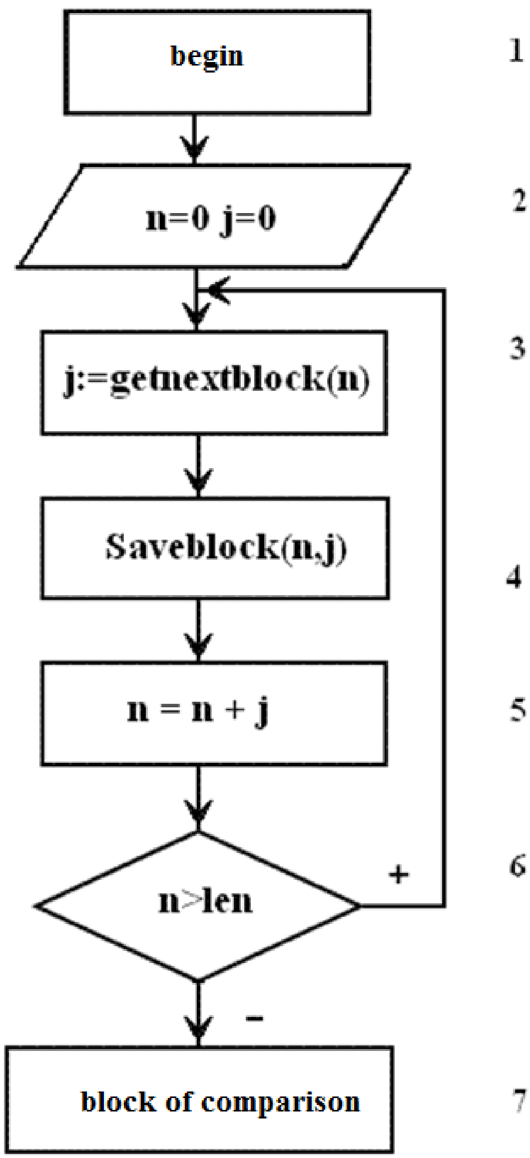

Fig.3. Block diagram of the algorithm for dividing audio data into audio frames

To increase the compression ratio, it is planned to set the error of finding identical frames in the aisles from 0 ... 10%

Figure 3 shows an audio data segmentation

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algorithm that divides the audio file data into audio frames according to the criterion of passing the sample values through 0, which works as follows.

The entire audio file is loaded into buffer memory, then after declaring variables, a loop is created – (audio frames # 3, 4, 5 and 6), where the splitting into audio frames will be performed directly.

In the 3rd block, the getnextblock function is called, which finds the next audio frame standing after the nth count. In the 4th block, the boundaries of the audio frame are saved in the form of a record of the first and last reference numbers of the audio frame. This is necessary in order to find the necessary frames in the stream when restoring an audio file. In block No. 6, a check is made for the finiteness of the cycle. If the processing is not finished, the process will be repeated again until all the audio data frames are marked up; if it is over, the control is transferred to the comparison block, where the found audio frames are analyzed for their similarity.

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