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FEATURES OF DIGITAL DATA PRESENTATION IN TECHNOLOGIES MULTIMEDIA

The analysis of the main features of digital data presentation, in particular sound signals, in technologies of multimedia is made. Classification of multimedia-applications is proved and the main methods and widespread hardware of processing of sound signals in multimedia technologies are considered. The assessment of a noise stability of data processing is carried out.

Keywords*: technology multimedia, application multimedia, processing of signals, analog-digital transformation, digital-to-analog transformation, discretization, oversampling, quantization, quantization noise accuracy of reproduction of data*

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ОСОБЕННОСТИ ЦИФРОВОГО ПРЕДСТАВЛЕНИЯ ДАННЫХ В ТЕХНОЛОГИЯХ МУЛЬТИМЕДИА

Выполнен анализ основных особенностей цифрового представления данных, в частности звуковых сигналов, в технологиях мультимедиа. Обоснована классификация мультимедиа-приложений и рассмотрены основные методы и распространенные аппаратные средства обработки звуковых сигналов в мультимедиа технологиях. Проведена оценка помехоустойчивости обработки данных.

Ключевые слова*: мультимедиа технологии, мультимедиа-приложения, обработка сигналов, аналого-цифровое преобразование, цифро-аналоговое преобразование, дискретизация, передискретизация, квантование, шумы квантования, точность воспроизведения данных*

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ОСОБЛИВОСТІ ЦИФРОВОГО ПОДАННЯ ДАНИХ В ТЕХНОЛОГІЯХ МУЛЬТИМЕДІА

Виконано аналіз основних особливостей цифрового подання даних, зокрема звукових сигналів, в технологіях мультимедіа. Обґрунтовано класифікацію мультимедіа-додатків та розглянуто основні методи та поширені апаратні засоби щодо обробки звукових сигналів в мультимедіа технологіях. Здійснено оцінку завадостійкості обробки даних.

Ключові слова*: мультимедіа технології; мультимедіа-додатки, обробка сигналів, аналого-цифрове перетворення, цифро-аналогове перетворення, дискретизація, передискретизація, квантування, шуми квантування, точність відтворення даних*

Abstract. The term "multimedia" has become firmly established in the lexicon of the user, and without it is difficult to imagine a modern computer world. The term "multimedia" (consists of two parts Lat. multi - and media, medium) refers to a set of hardware and software tools that allow the user to work interactively with heterogeneous data (graphics, text, sound, video and animation) organized in the form of a unified information environment. Thus, multimedia combines multiple types of heterogeneous data (text, sound, video, graphics and animation) into a single unit. In this sense, media can be regarded as a kind of technology.

In turn, the use of multimedia technology involves consideration for some of the starting points: First, the media should be seen as a new approach to the storage of different types of information into a single digital form. Secondly, the media should be seen as a certain set of equipment for the processing and storage of information. Third, the media - is software that allows you to combine the above four pieces of information (sound, graphics, etc.) into a complete multimedia application.

Multimedia technologies are one of the most promising and popular areas of computer science. They aim to create a product containing "a collection of images, text and data, accompanied by sound, video, animation and other visual effects (Simulation), which includes an interactive interface and other control devices. This definition formulated in 1988, the largest of the European Commission dealing with the problems of implementation and use of new technologies. (In this case, the "interactivity" refers to the property to respond to user actions, including control by the user). The apparent advantage and feature of the technology are, in particular, the following multimedia features, which are widely used in the presentation of information:

— storing large amounts of a wide range of information on the same media;

— increase (detail) is on the screen or the most interesting fragments ("magnifier" mode) while maintaining image quality;

— comparison of the image processing and its various software tools with scientific research or educational purposes;

— selection in the accompanying text or other visual material "hot words (areas)", on which get immediate help or any other explanatory (including visual) information (technology of hypertext and hypermedia);

— the implementation of continuous audio language appropriate static or dynamic visual range;

— the use of video clips from movies, videos, etc."Memorization distance traveled" and create "bookmarks" on the screen that interests "page";

— automatically scroll through the content of the product ("slide show") or create an animated and voiced "guide-guide" on the product ("talking and showing the user's manual");

– thus the multimedia product - the most effective form of information among computer information technology. It allows you to piece together the vast and disparate amounts of information.

The purpose. The purpose of this paper is to review the concept of signals and how to deal with the information of the video signals like (processing, compressing, transmitting information through the use of distinct electronic or optical pulses that represent the binary digits 0 and 1, storing, deleting, adding or reducing the space of storage).

Classification of multimedia applications. Multimedia – is the interaction of visual and audio effects with running interactive software. In other words, media – is a combination of text, graphics, sound, animation and video elements. According to the definitions given above, multimedia can be classified from different points of view: by supporting the interaction; through the use of various multimedia communications technology.

Fields of application multimedia. Among the applications of multimedia technology should highlight the following: video encyclopedia, interactive travel guides, trainers, situation-role-playing games, etc.; information and advertising service; internet-broadcasting; virtual reality systems; presentation; industry and technology (touch screens); in the research area; this electronic archives and libraries, their cataloging and scientific description, the creation of "insurance copies", the automation of search and storage, etc.

Hardware and software for multimedia technology. Multimedia technology is specialized hardware and software. To construct a multimedia system requires additional hardware support analog-to-digital and digital to analog converters to convert analog audio and video signals into a digital equivalent and back, video processors to convert the signals to a form reproduced on a display screen, a special integrated circuits for data compression file size allowed and etc.

Multimedia software tools consist of three components

— system software, i.e. a set of programs that are part of the operating system (OS) on your computer, and manage multimedia devices (and this management is implemented at two levels - the physical input-output control information at a low level with the help of machine instructions and user management features of devices using a graphical interface depicting remote control device, such as sound volume, tone, stereo balance, etc.);

— software tools, i.e. programs to modify media files, and create multimedia applications (graphics editors are fixed; tools for creating animated GIF-files, facilities for audio and video editing, the means of creating presentations, means OCR imposed a scanner, means of the creation of training programs, and a voice recognition system that convert audio files into text, etc.);

— aplication software, i.e. usually sold software systems on media (eg, CD or DVD) movies, textbooks, encyclopedias, books, virtual museums, guides, promotional materials, etc.

The main part. The following will focus on the features of audio applications of multimedia technologies.

Analog to digital conversion. Converting an analog signal to a digital audio includes several stages. First, an analog audio signal is supplied to an analog filter which limits the bandwidth of the signal and eliminates interference and noise. Then, the analog signal using circuits "fetch / store" stand counts: with a certain periodicity is memorized instantaneous level of an analog signal. Further, these samples arrive at the analog-to-digital converter (ADC) which converts the instantaneous value of each sample into a digital code (or numbers). The resulting sequence of digital code bits, in fact, is the acoustic signal into digital form. The transformation of a continuous analog audio signal is converted to digital (binary), both in time and magnitude (level).

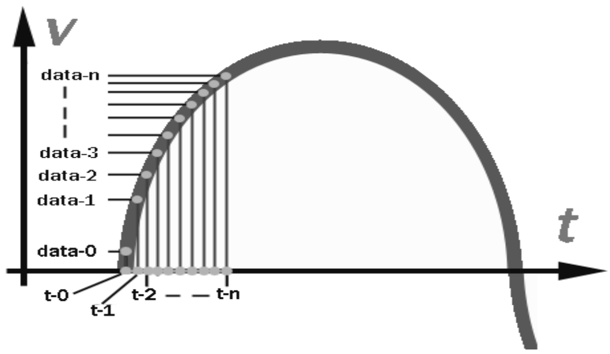

Fig. 1. Analog-to-Digital Conversion ADC

Discretization (sampling). The most important step analog-to-digital conversion of the analog signal is sampling. Instead of "sampling" in the technical literature sometimes use the term "sample", and the literature on sound processing the notion of "sampling" (literally "sample." So the word in multimedia and professional terminology has some values to indicate different types of "samples". Most often referred to as the sampled interval between two measurements of the analog signal). By definition, the sampling - the process of taking samples of the continuous time signal at equally spaced points in time. After a predetermined time interval, which is called the sampling interval, the procedure is repeated. For qualitative converting analog to digital signals to produce a sufficiently large number of samples, the sampling frequency cannot be arbitrary. Indeed, the sampling frequency actually determines the bandwidth of the signal which can be recorded by the digital system used. The width of this band cannot be greater than half the sample rate. This theorem is essential in the art recording and sound transmission in digital form. The theorem states that the frequency spectrum of the signal which occupies an area from $F_{\min}$ to $F_{\max}$ can be fully represented by its discrete samples at intervals T_d if T_d does not exceed $1/2F_{\max}$. In other words, the sampling frequency $f_d = 1/T_d$ during the conversion process should be at least, twice the highest frequency audible signal $F_{\max}$, i.e. $f_d > 2F_{\max}$. This follows from the fact that the spectrum of the signal converted by the ADC to digitize is periodic. According to Fourier's theorem, any signal shape can be represented as a sum of elementary sinusoidal oscillations of different frequencies and amplitudes. After the analog-to-digital conversion, audio signal represented in digital form, comprises, besides low frequency corresponding to the original analog signal, even high-frequency components. These components are low-frequency spectrum signal repetition in the form of lateral bands centered at multiples of the sampling frequency (f_d, $2f_d$, $3f_d$, $4f_d$ and next). If the sampling rate decrease, it will overlay (overlap) of low frequency range and a side band with a center at a point. Aliasing lead to new spectral components in the signal, and therefore its inability to properly recover.

A classic example of aliasing is when watching a movie when it seems that the wheel is spinning a moving carriage with speed that does not match the speed of the carriage, or even in the opposite direction. The occurrence of this effect is due to the fact that the frame rate (the image sampling frequency) is small compared with the angular velocity of rotation.

For recording the audio signal to avoid aliasing, filter before the ADC is set low (LPF), which suppresses all frequency lying above the sampling rate. So, the result is the discrete sampling time signal representing a sequence of samples - the instantaneous level of the analog signal. The higher the sampling rate, the more accurate will be restored sound. Sampling Procedure technically implemented using the device "fetch / store". As the storage element that commonly used capacitor is charged up to the voltage level of the input signal. The potential of the charge on the capacitor corresponds to the instantaneous value of the voltage signal. The voltage on the capacitor remains unchanged for a certain length of time, called the store. In the ideal case, the taking of reference must be instantaneous, real as the duration of this process is about $1 \mu S$.

Quantization. second stage of analog-to-digital conversion - quantization samples. In the quantization process we can measure instantaneous values of signal levels obtained in each sample, where it is carried out with a precision which depends on the number of bits used to record the level value. If given the long N – codeword record signal level using binary numbers, the number of possible values will be equal to 2N. The same number can be quantization level. For example, if the reference amplitude is a 16-bit codeword, the maximum gradation level signal (quantization level) is equal to $65536\left(2^{16}=65536\right)$.When 8-bit representation will be respectively, $256\left(2^{8}=256\right)$ gradation level.

Quantization noise. Quantization noise is a model of quantization error introduced by quantization in the analog-to-digital conversion (ADC) in telecommunication systems and signal processing. It is a rounding error between the analog input voltage to the ADC and the output digitized value. The noise is non-linear and signal-dependent. It can be modeled in several different ways.

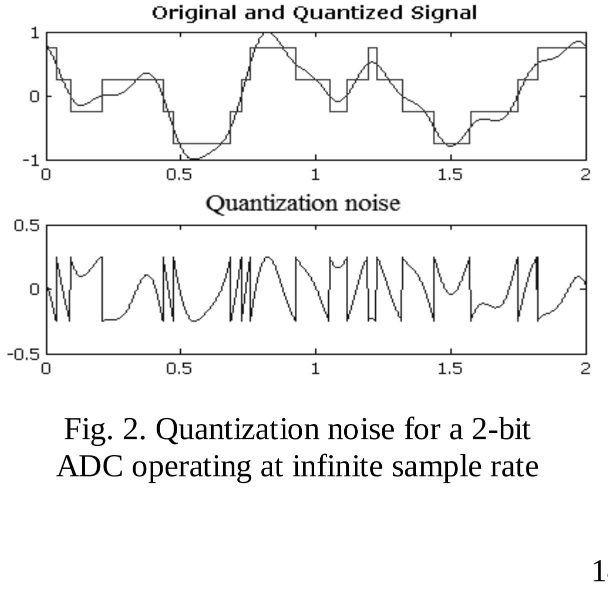

Fig. 2. Quantization noise for a 2-bit ADC operating at infinite sample rate

We can explain the Quantization noise in the following scheme.Conversion from an analog signal to digital form can be made only with a degree of accuracy, while a higher sampling frequency and bit ADC, the conversion occurs precisely. Quantization noise can be regarded as specific signal distortion, particularly noticeable it at low levels. The level of quantization noise is typically measured in the presence of a signal level relative to the maximum signal value. The lower the level, the better the sound quality. Achievable noise level is determined by the bit quantization and sampling rate.

Oversampling. In order to perform analog-to-digital conversion with high quality, you must fulfill several conditions. First of all, when you digitize the audio signal, this signal can be used as a higher sampling rate: the higher the sampling rate, the higher quality will be restored to the original signal. Unfortunately, increasing the sampling frequency is proportional to the flux in the digital data recording channel, and the amount of memory required to store an audio signal in digital form. Another condition of the analog-digital conversion is that the need to limit the range of sampling the input signal with a low pass filter (LPF). It should remove all the harmonics with frequencies above the frequency of sampling and thus to prevent aliasing. In modern ADC, problem of filtering to eliminate high frequency components of the spectrum is achieved by oversampling – sampling at a higher frequency. As a result, unwanted high frequency components are eliminated, while the high frequency components of the original audio signal will be stored.

Digital to analog conversion. To play an audio signal stored in digital form and to convert it to analog form, must implement a digital-to-analog signal conversion. The first stage of the digital data stream via a digital-to -analog

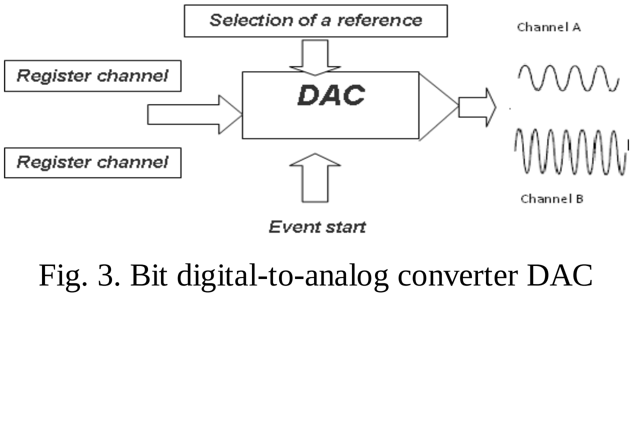

Fig. 3. Bit digital-to-analog converter DAC

converter (DAC) is isolating signal samples, the following sampling frequency f_d.

The second stage of the digital samples generated by smoothing (interpolation) continuous analog signal. This operation is equivalent to an ideal low-pass filtering a signal that suppresses the components of the spectrum of the sampled periodic signal. Immediately after the first stage of digital-to-analog conversion, signal will be a series of narrow pulses having numerous high frequency spectral components. We can explain the algorithms of converting audio signals in the following scheme.

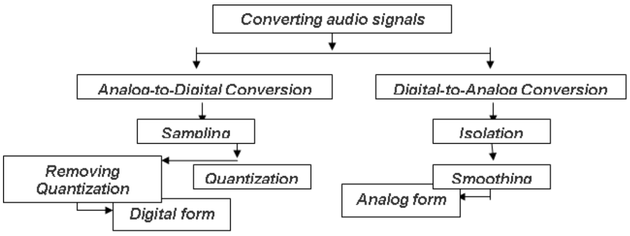

Fig. 4. General steps of algorithms for converting the audio signals (ADC and DAC)

Conclusion. The paper presents an analysis of multimedia technologies, and therefore considered appropriate classification, hardware and software. In particular, it highlights questions digital conversion of audio signals in multimedia technologies and solutions. Overview on this paper, using of digital representation of audio signals increases resistance to noise while in analog form less resistance to noise, the second result, using of digital representation of audio signals provides stability and reproducibility.

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