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Sound Design Exploration in Educational Multimedia Publications Applying Computing Platform

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***Abstract*—The commonly used practices of preparing a balanced sound support for educational multimedia publications have been examined. The methods for distinguishing key sound sources when modeling sound design have been streamlined and expanded. On basis of determined means of polar characteristics describing of meaningful sound fragments, typical and cardioids' sound fixation pattern were analyzed, which are laid as a basis of developed analytical apparatus of projected microprocessor system for researching of sound content quality for multimedia publication.**

Keywords—educational content, multimedia publication, computing platform, sound design, sound fixation pattern

I. Introduction

The practice of preparing balanced sound support for educational multimedia publications is an important stage in methodical content quality improving aimed at effective learning ensuring. Sound design in multimedia projects has a significant impact on the student's perception and interaction with field-oriented training materials, facilitation to increase their interest and attention. Adequate audio fragments selection helps to enhance the recipient's overall experience, creating a targeted academic environment, as well as supporting effective communication of messages and information [1].

The high-quality audio support of the educational publication clarifies the researched subject area and strengthens the immersive degree [2], providing a deeper impression of content. The established hierarchy and importance of selected methodical aspects contributes to the convenience and efficiency of the user's cooperation with the content, determines priorities and emphasis for drawing attention to important events in interaction with textual or visual elements.

The situational relevance of audio materials to the educational scenario and their technical performance are of great importance to ensure proper perception of audio recordings. When creating and editing audio tracks, it is important to consider the context and purpose of the educational multimedia publication. Sound design should be aimed at achieving educational goals and contribute to a comprehensive understanding of training content [3]. For example, informative video presentations can use clear and professional voiceovers that explain key concepts and ideas. In training games or interactive tasks, it is advisable to use sound effects that emphasize correct and incorrect answers, providing unobtrusive feedback. Next, for music-oriented content, audio tracks can be specialized for skills development of instruments sounding recognizing, understanding the musical form of analyzed composition, approbation of different genres and styles. In language education, audio tracks should it is worth specialized for improving language skills, teaching pronunciation and developing vocabulary, understanding different language variants, dialects and accents.

When demonstrating natural sounds, the specialization of soundtracks helps to audio effects conceive that occur during the course of physical phenomena or the reaction of chemical compounds. So, typical categories distinguishing the operation sounding of engines, gears, roller bearings, etc. allows engineering degree students to expand their understanding of structure, kinematics and dynamics of mechanisms and systems when studying the scientific foundations of their design. The tracks' categorization in context of signals, oscillation and other aspects of electronic devices will allow visualizing the abstract concepts of wave theory in the study of electrical and magnetic phenomena.

Thus, particular sources distinguish in audio tracks helps students to separate and recognize different situations in subject area, which contributes to development of their professional competences and analytical skills, understanding and perception of concepts and phenomena related to specific engineering branch. This opportunity allows students to focus on specific sound elements and study their role in researched stages of technological process. Therefore, hardware and software complex development for sound design study of multimedia publications is timely and relevant, it will ensure the improvement of technical quality of electronic training content and strengthen the multisensory interaction of recipients with industry-oriented methodical materials.

II. Methods of Distinguish Key Sound Sources in Educational Multimedia Publications Preparation

Particular sources distinguish in soundtracks is carried out by using and combining a number of methods that allow to virtually place sound sources in space, creating a sense of volume and realism. As noted, soundtracks can be specialized for a wide range of educational purposes depending on the specific requirements and learning context.

A. Limits of spatial sounding effects applying

An effective means to distinguish particular sound sources in an audio track is *stereophonic sounding* using, allowing different sound sources to place in spatial position.

Sound objects can be placed across of virtual scene width, reproducing their spatial localization: this creates the presence effect and realism, and also helps the listener to distinguish and identify particular sound sources more easily.

Panning as a method to distinguish particular sound sources involves moving the source from one channel to another or mixing it in different proportions between channels, moving from left to right or from foreground to background. Panning allows to precisely controlling the position and movement of particular sound sources in space, which makes the sound scene more detailed and realistic.

Amplitude dynamically changing of particular sources is another effective means of distinguish key audio content. Adjusting the volume allows giving a specific sound source a more prominent or muted role in the audio track, emphasizing important elements or leveling out the value of a fragment where distant sources have a lower volume. Volume changing affects the recipient's perception and attention, helping to focus on certain sound fragments.

The use of *reverberation*, *echo*, *chorus*, etc. can also help to particular sound sources distinguish. These effects create a sense of spaciousness and depth of the soundstage, bringing additional acoustic attributes to individual sound sources. So, reverberation gives the impression of presence in a certain environment, and echo emphasizes the volume or movement of particular sound sources.

The presented methods to key sources distinguishing in audio tracks can be used to create more target and effective sound design in engineering disciplines context. They help improve the perception, understanding and analysis of learning material, creating a more immersive and meaningful educational audiovisual experience. In order to prepare adequate audio content for educational multimedia publication, it is necessary to research the hardware for of particular sources distinguishing in audio tracks.

B. Hardware categorization for processing sound sources

When analyzing hardware for particular sound sources distinguishing, it is worth considering its ability to accurately and realistically reproduce sound, control volume levels and spatial processing. The use of appropriate hardware will help to achieve high quality and efficiency in the creation of means of particular sources distinguishing in audio tracks in researched subject area context of professionally oriented academic discipline.

An *audio interface* with the ability to record and reproduce multi-channel sound is used to particular sources distinguishing in soundtracks, which allows for more precise control of the spatial position of sound sources. This device provides communication between computer or other sound recording devices and, in fact, fixation object. Proper listening of audio tracks is essential for effective particular sources distinguishing. For this, in the technological process of sound design, it is necessary to provide high-quality *audio monitors* or professional *headphones* that transmit sound information accurately and with high resolution. This will make it possible to adequately assess the position and placement of sound sources in space.

Sound processors with spatial processing functions can create immersive acoustic effects that help imagine the spatial location of sound sources. *Equalizers* allow adjusting of individual frequencies levels in sound spectrum, which can also be useful for certain sound sources distinguishing. *Sound controllers* and *mixers* are used to precisely control the volume levels of individual sound sources in audio tracks, allowing adjusting the amplitude, panning and other parameters of each source individually, aiding in dedicated sound effects creation. However, among all categories of researched hardware for processing audio effects, *microphones* are of decisive importance as primary means of key sources distinguishing and fixation for sound design of multimedia publication.

III. Polar Characteristics' Description Means of Key Sources of Sound Content

Appropriate microphones choice for distinguishing key sound sources depends on specific needs and thematic purpose of the multimedia publication. The correct microphones setting will ensure high-quality sound recording, accurate sources space localization, helping to achieve the desired sound effect. An important aspect when choosing microphones and when placing sound sources in space is the sound fixation pattern. Also known as audio specimen, this term is used to describe the polar characteristics of a microphone or sound source, indicating the way the sound wave is spatially distributed around the source.

A. Analysis of typical sound fixation patterns

The typical *spherical* pattern of universal microphone is characterized by uniform sensitivity to sounds from all directions [4]. It describes the sound signal coming from any angle within sensitive area (Fig. 1, *a*). Having no noise suppression mechanisms, the spherical pattern guarantees a wide perspective of audio recording without introducing a negative proximity effect. Such parameters provide the devices with a small size and compact design, which allows them to be used in various portable systems. However, in conditions when it is necessary to focus on a specific sound source or when fixing an object in a noisy environment, it is worth expanding this design, for example, with *side addresses* (Fig. 1, *b*), or use other specialized patterns that provide better directionality.

Equal focus on sounds coming from two opposite directions provides a *shaped* pattern with almost no response to sounds coming from the side (Fig. 1, *c*). Formed by special design with two built-in frontal microphone capsules, this pattern provides the strongest proximity effect. Providing the narrowest gather angle, *lobar* pattern works with adjustable zones (Fig. 1, *d*) and allows catching focused sounds at particularly long distances thanks to some width of lateral directions.

B. Cardioid modifications of sound fixation patterns study

The maximum sensitivity of audio specimen cardioid modifications is directed in sound source direction, while sensitivity decreases from sides and back [5]. The main sound fixation area in *cardioid* pattern is in frontal direction, forming a wide, up to 120°, sound capture angle. Such pattern in inverted heart shape (Fig. 2, *a*) allows to effectively sound source distinguishing located in microphone front, while reducing the coverage of unwanted sounds from sides and behind.

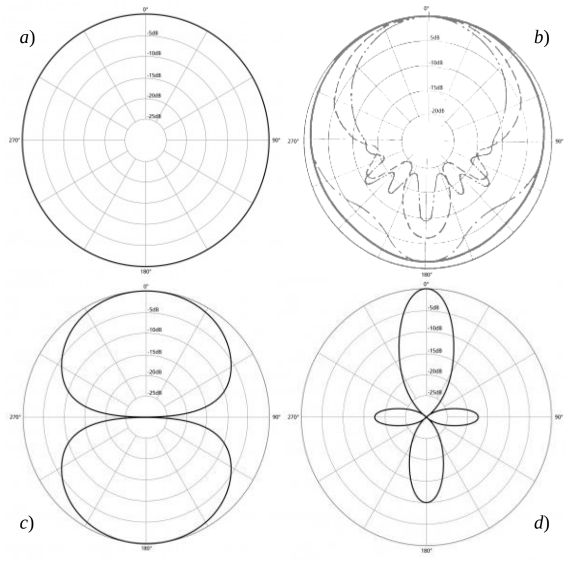

Fig. 1. Diagrams of typical sound fixation patterns

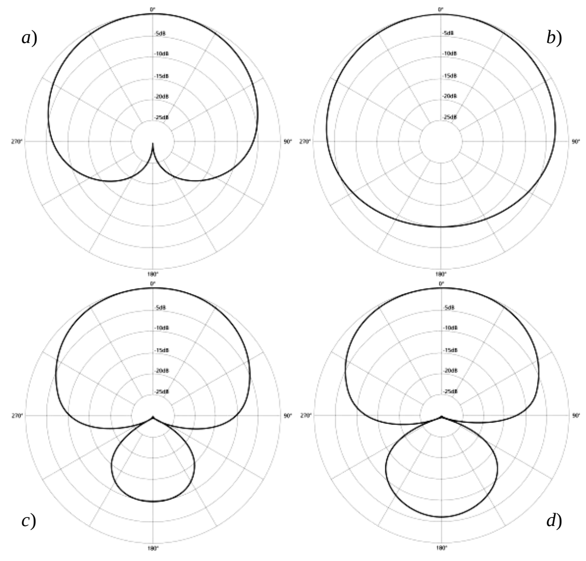

Fig. 2. Diagrams of modifications of cardioid sound fixation patterns

This makes cardioid microphones ideal for situations where it needs to reduce background noises, main sound source separate from environment, or focus on sources in a narrow area. Cardioid microphones applying can be found in many educational situations, including recording live lectures or public reports, conferences and webinars, industrial videography, and field experiments. They provide clear and directional sound capture, which helps to improve the quality of recording and transmission of sound information [4].

Subcardioid microphone pattern has a greater forward directivity (Fig. 2, *b*) and several features that affect its parameters. Characterized by increased sensitivity in the direction of the sound source located directly in front of the microphone, the subcardioid pattern has a wider sound capture angle, fixating sound objects at a short distance from each other and generally located slightly outside the direct direction. By reducing the influence of side and background sounds, such devices guarantee transparent sound with less approximation effect and preservation of natural low frequencies in relatively quiet scenes for rooms with controlled acoustics. However, there is also the possibility of triggering a feedback loop with the speakers [5].

Narrow and accurate sound capture with a larger zone-capturing forward direction and strong sound attenuation from the sides and back provides a *supercardioid* pattern (Fig. 2, *c*), allowing to capture sound sources located far enough from the microphone. Providing good isolation from unwanted background noise and sounds from other directions, distinguishing key fragments and minimizing the differences between them, this pattern type is very useful in situations where need to focus on a specific scene and ensure minimal influence of the environment.

The main sensitivity zone of the *hypercardioid* microphone is also directed forward (Fig. 2, *d*), which allows to accurately fixed the sound source in front of the microphone, but muffles the sounds coming from the sides. At the same time, there is a certain sensitivity to sounds coming from the rear direction, giving advantages in terms of feedback. The hypercardioid means use allows achieving maximum accuracy, high purity of sound and detail.

IV. Generalized structural scheme of the project

To perform computing tasks in the sound patterns researching, the resources of the computing platform as a complex software and hardware environment are used. Such a platform covers unified physical equipment and provides an integrated development environment that allows designing service utilities for managing and distributing project computing resources. The computing platform provides the power to perform complex computing operations, data processing, modeling, analysis, and other tasks that require significant computing resources.

A. Peripheral equipment

To optimal audio specimen research for sound design modeling of educational multimedia publications, it was decided to use the Arduino Uno computing platform [6, 7] based on the ATmega328P microcontroller.

The Keyes KY-038 electret microphone shield chosen for the project contains an audio module with a built-in amplifier and filters to capture pure sound [8].This module (Fig. 3) allows reading audio signals from the environment and carry out their further analysis. The module's power LED turns on when power is applied to the board.

The microphone digital input is connected to serial pin TX0 of computing platform [9]. The digital signal sensitivity is adjusted using from cyan potentiometer. A low signal level keeps the audio module in standby mode, and a high signal level indicates that the microphone has detected a sound pattern. At the same time, the signal trigger LED on the audio module turns on.

The researching of the physical nature of the detected pattern is carried out by the built-in ADC of the computing platform [10], which has the ability to receive amplified analog signals from the microphone through the channel connected to the A4 pin (Fig. 3). KY-038 is able to detect sound patterns in the range of up to 100 dB, which is quite enough for our project (Fig. 1, 2).

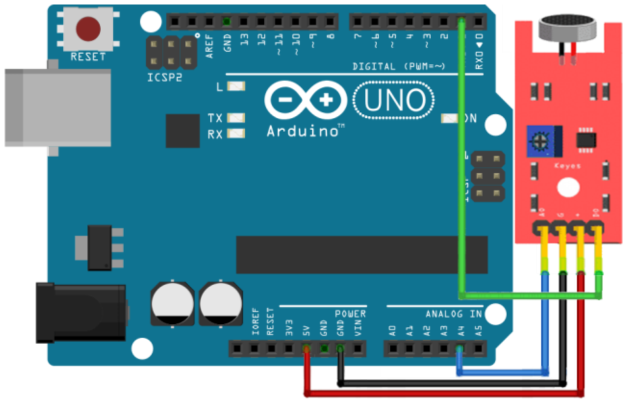

Fig. 3. Connecting the digital audio module board to computing platform

B. *Programming the sound pattern research project*

In the software of the project for the researching of sound patterns, first of all, the levels of the analog signal [11] coming from the audio module is measured. Further, it is converted into a digital signal using sampling by the analog-to-digital converter of the computing platform and is reduced to decibels. ArduinoFFT.h library was used for signal processing, which allows Fast Fourier Transform to be performed when analyzing the sound spectrum.

The Fourier transform makes it possible to decompose a signal in the time domain into its frequency components in the frequency domain. With supplementary analysis of the sound spectrum, FFT allows revealing the frequency components of the input audio signal. Since the audio spectrum represents the distribution of the signal's energy over different frequencies, applying the FFT to an audio signal produces a spectrogram that shows the intensity of the signal at each frequency versus time. It allows identifying the main components of the sound, such as the fundamental frequency, harmonics, noise components and other sound artifacts.

By revealing the frequency component of the audio signal, the Fourier transform allows the analyzer to identify key components of the pattern, such as bases tones and other features. As a result, a spectrogram is obtained, which visualizes the distribution of signal energy over the frequency spectrum. Next, using the Fourier transform for each moment in time of the corresponding audio fragment, a time-frequency representation of the sound is obtained. This makes it possible to detect changes in the frequency structure of sound depending on time, which is useful when analyzing the dynamics of sound patterns. That is, FFT is a powerful mathematical algorithm for converting data from the time domain to the frequency domain.

In addition to the considered ArduinoFFT.h, other libraries are also involved in the project. Yes, the Wire.h library allows the computing platform to interact with the peripheral via the I2C/TWI interface. On the Arduino Uno board, the SDA data line associated with this interface is located on pin A4 (Fig. 3). Thus, when researching sound patterns, it is possible to analyze different sound sources and their spectral characteristics by adjusting the size of the FFT window, frequency range and other parameters to obtain the desired results [10]. The coverage area identified by sound intensity is displayed on the simulator's serial plotter.

V. CONSOLUTIONS

In the presented study, hardware and analytical apparatus to audio specimens' analysis in sound design modeling for educational multimedia publications are proposed. The spectral information obtained as a result is suitable for further processing on a personal computer, in particular for intelligent comparison of reference target sound patterns and obtained diagrams. The coupling of the microprocessor system and the computer expands the possibilities of the project and allows carrying out more complex studies of sound patterns, capturing more data and interacting with them with the help of additional functions and network connection.

Fourier transformation is also planned to be used for sound signal processing, noise filtering, and amplification of certain frequency components or selection of characteristic features. This helps extract information from the sound that can be used for further analysis and development of sound models.

In the perspective of further research, it is planned to build an expert system of vicious elimination of bugs and integrate the project into built-in complexes of the Industrial Internet of Things.

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