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THE TECHNIQUE OF WORKING WITH MICROPHONES WHEN RECORDING TO PRESERVE THE NATURAL SOUND OF THE SOUND SOURCE

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Abstract

The article is devoted to important aspects of sound recording, which contribute to the transmission of the most natural and authentic sound. It discusses the different types of microphones and their characteristics, as well as how to use them correctly, depending on the recording context. Special attention is paid to the techniques of microphone placement, the choice of orientation, as well as the influence of acoustic conditions on the recording quality. The article offers practical recommendations for setting up equipment and sound processing in order to achieve the best results and preserve the natural sound of the source. Common mistakes that should be avoided during the recording process are also investigated. This work will be useful for both novice sound engineers and experienced professionals seeking to improve the quality of their recordings.

Keywords: *equalization, timbre formation, proximity effect, directional microphones, postequalization, natural sound*

Often in sound engineering, the process of signal equalization is associated with post-processing through a device such as an equalizer. However, in practice, the timbre formation of an instrument or voice begins long before its processing at the mixing stage. The main part of the audio signals is recorded by means of microphone recording. This statement may seem somewhat outdated in modern practice: it is not necessary to record acoustic instruments, for example, percussion, in the presence of sampler libraries, it is not necessary to record the sound of electric musical instruments such as electric guitar in the traditional way, in the presence of digital guitar solutions, etc.

But if we look at the issue comprehensively, we can conclude that the sampler libraries of acoustic instruments were somehow originally fixed by means of microphones and microphone arrays, and in guitar processors the cabinet-microphone system is the final link of the timbre-forming tract. Therefore, microphones, even virtual ones, in most cases participate in the process of sound formation. Only signals synthesized from scratch in oscillators – sound generators – synthesizers are not recorded by microphones. As a rule, these are the simplest wave forms: sinusoidal, triangular, sawtooth, etc.

Excessive abuse of postequalization, however, like any other sound processing device, often leads to a subjective feeling of a re–processed timbre – a timbre that sounds not just unnatural, but even to some extent artificial. In relation to the re-processed timbres, the listener can often use such epithets as "plastic", "cardboard", "fake", etc. In this regard, even at the stage of sound recording, maximum attention should be paid to such a phenomenon as natural signal equalization.

So, we have considered an approach to a priori evaluation of most audio signals as recorded by means of a microphone or microphone system; let's focus on this thesis in more detail. The microphone is the first link in the chain of natural equalization of the audio signal. At this stage, natural equalization has four main stages:

- Microphone selection;
- Microphone position;
- Proximity effect;
- acoustic properties of the room.

Let's look at each stage in more detail. Microphone Selection Different microphones differ from each other in a number of parameters. One of the most important parameters of the difference between microphones is its amplitude-frequency response, which largely forms the characteristic "sound" of a particular microphone. Nonlinear distortions introduced by the microphone at the stage of recording the audio source also play a significant role in this process. In this regard, they are usually combined with the frequency response of the microphone in the general concept of the initial stage of natural signal equalization. Each microphone has its own frequency response (as well as the nonlinear distortions it introduces into the signal), often represented as a graph resembling an equalization curve. It is logical to use such a microphone, for example, when recording a pop voice or an acoustic guitar, and it is completely impractical to use it when recording, for example, a large drum. To record it, it is much more efficient to use a microphone optimized for recording low-frequency signals, such as the AKG D112.

It is the selection of the "correct" microphone signal suitable for a specific sound source at the stage of its fixation that is the first and most important stage of the timbre formation of a musical signal in recording. At the same time, the price category of the microphone has only an indirect relation to the final sound of the source; how well the microphone "fits" a particular voice or musical instrument is a disproportionately more important criterion for its choice. The main rule in this case is categorically not to rely on postequalization when fixing the source of the audio signal. The original recorded material should be as timbral as possible similar to the desired final result. Postequalization does not "transform" one microphone into another, and therefore choosing the "right" microphone at the recording stage remains relevant to this day. Anyway, the formation of the required timbre of a musical instrument or vocal strictly at the stage of its recording is not fully achievable in every case. The main purpose of selecting a specific microphone in the process of fixing the sound source is to minimize the amount of subsequent sound processing required at the mixing stage. The choice of microphone position is determined by several acoustic patterns, primarily related to the peculiarities of sound wave propagation. When moving away from the listener (or in the case of sound fixation, a microphone), the spectrum of the sound source loses part of its high–frequency component. A similar effect occurs when the microphone is rotated azimuthally relative to the source or when the microphone is blocked by a physical object at some distance from the source. This phenomenon is called natural shading of the signal and in the case of moving the microphone away or turning it relative to the source of the audio signal, it is expressed in a specific frequency drop in the upper part of the frequency spectrum from approximately 4.5–5.5 kHz. For example, if the microphone used is excessively "bright" for this particular vocalist, and replacing it with a "darker" one is not possible – the most correct recording strategy would be to move the microphone away from the vocalist (if recording is performed in a muffled room), turn it away along the horizontal axis, and, if possible, use a denser pop filter. The proximity effect is expressed in the amplification of the woofer component of the signal when the microphone approaches the sound source. This effect is observed only with directional microphones: cardioid, supercardioid, bidirectional, etc. The closer the microphone's directional pattern is to the bidirectional one, the more the proximity effect is manifested during recording. In addition, the manifestation of the proximity effect is most evident when the microphone is directed directly at the source. When you turn the microphone away from the sound source, the proximity effect decreases. So, the proximity effect is observed only in directional microphones. This is explained by the fact that due to the phenomenon of diffraction, sound reaches the microphone membrane from both the front and back sides. The pressure difference between the front and back sides creates movement of the microphone membrane. The distance from the front to the back of it is usually within 1–2 cm. At low frequencies, the wavelength of which can reach several meters, the pressure drop within 1–2 cm of the total duration of the sound wave is small. At higher frequencies, this difference increases. In addition to the gradient, the proximity effect is also affected by the law of inverse squares. In the field of acoustics, the inverse square law states that the sound intensity decreases by about 6 dB (i.e. twice) when the distance from the sound source is doubled. Accordingly, as you approach the source, the sound intensity increases in the same proportion.

In practice, proximity effect control can solve spectral problems of audio signal sources in the low-frequency region. For example, if the sound engineer finds the sound of an electric guitar recorded from the guitar cabinet by a microphone to be thin and immense, then one should not hope for postequalization at the mixing stage. It is worth bringing the microphone closer to the cabinet, if its location allows it. Similarly, if the snare drum is listened to excessively voluminously and "droningly" during recording, the best way to combat this phenomenon in this case is to move the microphone away from the drum to weaken the effect of proximity. Of course, these examples are hypothetical, and the use of the proximity effect can be applied to any source of an audio signal, depending on the goals and objectives of the timbre formation of a particular musical instrument or voice. Despite the fact that modern reverberation algorithms allow you to simulate almost any parameters of enclosed spaces and create a very convincing reverberation effect, some sound signals are still recorded with the capture of natural space, where recording is performed in real time (for example, when recording acoustic percussion, academic instruments, etc.).

What is this being done for? In traditional sound engineering, it is considered that an uncluttered room in which recording is performed reacts timbral in a specific way to a specific acoustic signal sounding in it, enriches the sound of the source timbral, making it "large" and "voluminous". At the same time, reverberation algorithms focus primarily on modeling the process of reflection and re-reflection of sound waves from room surfaces, without paying due attention to the timbral component. In practice, capturing a natural acoustic space allows you to use the "sound" of the room, fixed in the form of a mono track or stereo pair, as an additional degree of freedom of natural equalization, which adds the ability to control the total volume of a particular musical instrument. This has an extremely positive effect on "thin-sounding" timbres, whose spectral properties are almost impossible to compensate for postequalization due to the weakly expressed frequency range in their spectrum, which is responsible for the total volume. Of course, such a technique is not feasible in muffled rooms and is applicable primarily in "live", i.e. non-muffled rooms. The listed stages of natural signal equalization make it possible to reduce the amount of subsequent equalization required, but this does not mean that equalization at the stage of information completely loses its relevance. The purpose of natural equalization is to minimize the use of postequalization and, as a result, to avoid the effect of overprocessing during reduction.

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