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METHODS OF USING ARTIFICIAL REVERBERATION IN MODERN SOUND ENGINEERING

Abstract. The article is devoted to the systematization of the main modern methods of using reverberation in sound engineering and methodological bases of its use in the educational process in higher education institutions. Scientific novelty is to clarify the classification and systematization of the main modern methods of using artificial reverberation, as well as methodological principles of mastering artificial reverberation in the process of training. The purpose of the article is to systematize the main modern methods of using reverberation in sound engineering. To achieve this goal, the following methods were used: system-analytical, art history and comparative-historical. The main results and conclusions of the research. The evolution of digital technologies in the field of sound engineering, the introduction of reverberation algorithms have greatly contributed to the expansion of the use of artificial reverberation in sound engineering. The article briefly describes "convolutional" reverberation as one that operates with the most modern reverberation algorithm. Currently, a variety of effective methods of artificial reverberation using has been developed in sound engineering, which can be used to implement various creative tasks. The article describes the methods of using artificial reverberation for sound perspective creating; for back-vocal processing; for delay processing; for "softening" the sound of individual parts of musical instruments; for different signals of the same type processing; for solo batch processing using two reverb effects. Some types of artificial reverberation were used at different stages of the music industry development. The current state of software development allows the use of several reverb algorithms simultaneously, making it possible to create original sounds. In the process of mastering of various methods of using artificial reverberation by higher education students, it is best to start their acquaintance with artificial reverberation from the simplest reverberation effects, which can be implemented in "analog" effects processor, in which only some parameters vary based on the manufacturer's algorithms. Further mastering of artificial reverberation should be carried out through software (in particular, free VST-plugins of artificial reverberation) used in digital audio workstations (DAW).

Key words: artificial reverberation; "convolutional" reverberation; reverberation algorithm; method of using artificial reverberation; artificial reverberation plugin

Introduction. The improvement of technologies in the field of sound engineering, the spread of digital sound processing devices, in particular, in our country, encourages the intensification of research to specify the methods of their use in practice. Among the sound processing technologies that have become widespread, a special place is occupied by artificial reverberation, the algorithms of which are used both in devices and in programs used by sound engineers. Accordingly, research on the systematization and specification of artificial reverberation techniques in modern sound engineering is becoming especially relevant.

Analysis of recent research and publications. Studies of reverberation of various rooms for a long time have been the subject of many scientific studies in the field of musical acoustics (N. Eyring, J. Millington, W. Sebin). The development of artificial reverberation devices has been studied in the field of music electronics (O. Julius, J. Detorro). The use and development of new reverberation algorithms is the subject of many studies in information technologies (J. Moorer, D. Taranov). Various creative and technological aspects of the use of artificial reverberation in sound engineering were considered by D. Gibson (Gibson, 2005), U. Kozyurenko, W. Moylan (Moylan, 2007), D. Moulton (Moulton, 2000), B. Owsinsky (Owsinsky, 1999), A. Farnell (Farnell, 2010) and others. However, the systematization of methods of using artificial reverberation has not been the subject of current research.

The research methodology is based on system-analytical, art history and comparative-historical methods. System-analytical method is used to systematize the experience of using artificial reverberation in sound engineering, art history method is used to determine the artistic tasks that are implemented through the use of various methods of using reverberation in sound engineering. The comparative-historical method was used to concretize the artificial reverberation used in historical retrospect in connection with the change in the concepts of constructing the space of musical works.

Accordingly, the **aim of the article** is to systematize the main modern methods of using reverberation in sound engineering.

Presentation of the main material of the research. The development of technologies in the field of sound engineering has significantly influenced the expansion of available techniques and means of sound processing both in the conditions of concert sound engineering and in sound recording. Among the new opportunities in modern digital sound engineering there is a development of the latest technologies of artificial reverberation. According to the logic of the presentation, it is advisable to specify the meaning of this concept. Reverberation is the acoustic process of preserving sound after the end of sound pulse or oscillation due to the reflection of sound waves from surfaces. Thus, "natural" reverberation exists in an environment in which there are surfaces from which sound waves can be reflected. In the process of recording, which is carried out in a concert hall, built on an acoustically designed project and decorated with relevant materials, such reflections generally have a positive effect on the recorded sound, and in some cases (when such an effect is needed on the record), in modern conditions, recording with “natural” reverberation of the concert hall is used. However, when there is a need to place sound objects in different virtual sound spaces and implement other creative tasks in the process of construction or mixing in modern conditions, it is economically impractical to record everyone in a separate concert hall with the necessary reverberation characteristics. Therefore, in the vast majority of cases, since the middle of the last century, artificial reverberation is used, because it is much easier to get sound with minimal reverberation (or no reverberation) and then use an artificial reverberator to make it "alive", than to achieve the right level of room preparation or sound recording in such room. Accordingly, artificial reverberation is a simulation of various reverberation processes created by hardware or software (in particular, those that exist in real concert halls).

In today's world, the development of modern digital reverbs (as hardware devices so computer routines (plugings)) is characterized by a certain stagnation, which is primarily due to the lack of new developments in the field of reverberation algorithms (mathematical models) (Browne, 2001). There is a rather contradictory situation in this field: on the one hand, sound engineering uses "analog" devices made more than half a century ago (for example, spring reverbs in electric guitar combo amplifiers) together with "digital" ones, which exist both in hardware and software; on the other hand, none of these reverberators, which work on algorithms developed in the last century, create reverb that is indistinguishable from natural reverberation. At the same time, modern searches for mathematical models are primarily aimed at "reproducing" the natural reverberation that occurs in different rooms. Among such modern examples of artificial reverberation, the so-called "convolutional" (pulse) reverberation is interesting, wich operates on the concept of discrete "convolution". The discrete Fourier transform of the "convolution" of the impulse characteristic of the filter and the input sequence is equal to the product of the spectrum of the input sequence and the discrete Fourier transform of the impulse characteristic. Thus, "convolution" in the time segment is equivalent to multiplication in the frequency segment. This property is used in the development of convolutional reverberators, because direct "convolution" can be performed both from the impulse characteristic of a hardware reverberator and from the impulse characteristic recorded in a real room. This method provides very plausible results and has become widespread since the beginning of the XXI century. When obtaining the pulse characteristics, it is necessary to create a pulse signal in the desired frequency band, similar to the Dirac function (δ-pulse), to apply it to the tested system and record the output signal. The response received will be a pulse characteristic of this system (Browne, 2001). Substituting the values of the readings of the obtained impulse characteristic in the formula of discrete "convolution", you can get an accurate simulation of the properties of the selected system (provided the linearity of this system). The most modern method of obtaining impulse responses of the rooms uses the method of deconvolution (inverse filtering), wich means that the creating of pulse response based on the use of special noise sequences allows you to restore the source, but a sliding tone is already as a test signal. Harmonic distortions can be filtered out of the recorded response of the room, because they will always be at frequencies above the test signal, and reverberation will be below (due to increasing frequency over time). In addition, to improve the signal-to-noise ratio, you can increase the amplitude of the low-frequency part of the test signal and take this into account in the process of deconvolution. It is worth noting that modern computers allow this procedure to be performed with exceptional accuracy, which largely ensures the compliance of such reverberation sound with, for example, the parameters of the room, the impulse characteristic of which it simulates. However, the number of multiplication and addition operations will be huge. For example, in order to collapse a 1 second signal with a standard 2 second room pulse characteristic at a sampling rate of 44.1 kHz, the number of multiplication/addition operations will be approximately 3.89×10^9, but in modern studio conditions there are signals with sampling rate of 192 kHz and above to be operated, accordingly, the use of direct "convolution" is not possible if the signal needs to be processed in real time. To solve the problem of accelerating and simplifying this process, the "convolution" in the frequency part is used on the blocks with the help of fast Fourier transform. Exactly the use of fast Fourier transform can significantly reduce the amount of computation and signal processing with a time close to real. The use of such a reverberation model led to the emergence of hardware and software "convolutional" reverberators, but they were characterized by significant cost and low variability (it was almost impossible to change the reverberation parameters). In addition, in the initial stages of development of this technology, it also had shortcomings:

1) the used impulse characteristic corresponds to only one variant of placement of the sound source and the listener, and it is obvious that the unnaturalness of sound will be noticeable in the case of spatially distributed imaginary ("virtual") sound sources, such as percussion or chorus;

2) imaginary space is treated as a stationary, invariant system, that is impossible in practice;

3) difficulties in creating libraries of impulse characteristics;

4) the most advanced models of classical algorithmic reverbs (for example, such as "Lexicon 480L") are not inferior to "convolutional".

However, the ability to use "convolutions" of the best concert halls or the best reverberators in sound engineering is seen as a significant advantage of "wrap" reverberation. In addition, at the present stage in "convolutional" reverberators there is a possibility to modify the existing pulses to achieve the desired sound of reverberation. The simplest modification wich allows to reduce the reverberation time is to shorten the pulse, more complex one is its "stretching". A significant number of modern reverberators allow you to change the relative level of early and late mappings, as well as the delay of early mappings. It is believed that for "beautiful" reverberation, the first early reflections should reach the listener 15-20 milliseconds after direct sound, and the total power of early reflections (in the range of 15-50 milliseconds) should be approximately -6 dB of direct signal power.

Pulse filtering allows you to change the timbre of reverberation. In the general case, the filtering can be time-dependent: by applying amplitude envelopes to different pulse frequency bands, you can change the attenuation rate (degree of damping) at different frequencies.

An important parameter of reverberation is the density of reflections over time, along with the random omnidirectionality of their arrival (Vecchi, McLachlan, Kohlrausch, 2020). This parameter is also called reverberation diffusion. In order to increase the density, you can add artificial simulations to the existing pulse or duplicate all pulse reflections with some filtering.

Another important parameter is the proportion of lateral reflections ("laterality"). If the reflections come from the same spatial direction as the direct sound, they can distort the spectrum of the sound, causing "comb filtering". Reflections, which independently come from the side directions, on the contrary, increase the naturalness of the sound, increasing the effect of "envelopment" by the acoustic environment. There is also a sound engineer's technique, when, while panning a direct signal to one part of the stereo, artificial reverb is panned to another part.

Reverberation of real rooms is almost linear, because the size of the room does not change, respectively, it can be quite accurately described by "convolution" with the appropriate pulse (Koumura, Furukawa (2017). However, if the auditorium is filled with spectators, there may be some randomness of the reverberation parameters over time. This randomness can be achieved by changing the parameters of the pulse over time or nonlinearly processing the resulting artificial reverberation. You can apply random level changes, dynamic processing or even frequency modulation. Not all nonlinear modifications will sound natural, but many of them can be used as additional means of expression or special effects in sound engineering. For example, height reverb modulation cannot be applied to piano recordings because it is a tempered musical instrument without modulations and vibrato. However, the same technique in some cases will "sound" good if it is applied to a vocal channel or a group of strings.

Obviously, the sound of "convolutional" reverberators is determined primarily by the pulses loaded into them (rooms or reverberators) and their means of modifying these pulses. And almost the same algorithm of "application" of reverberation in them is used.

Accordingly, there is a need to analyze the basic methods of using artificial reverberation in modern sound engineering. First of all, it is necessary to conditionally distinguish between techniques that originate from the concert industry and are used in sound engineering in real time, as well as techniques that are used in the process of compiling recordings. At the same time, modern digital technologies allow the use of reverberation effects in concert sound engineering, which previously required considerable time to calculate and therefore could be used only in the process of compiling recordings. In addition, techniques of using artificial reverberation has changed historically, primarily due to the development of technology and the introduction of appropriate devices (mid-last century), development and implementation of reverberation algorithms (last quarter of last century), as well as software improvements and the growth of computing power of computer technology (early XXI century).

In addition to technological and stylistic aspects, the use of different types of artificial reverberation correlates with the change in the "status" of performers in historical retrospect. Since the early 60's, performers in the sound space were located at a considerable distance from the listeners and were, to some extent, "out of reach". This was further emphasized by the use of reverberation of large chambers, but the gradual decline in the cost of sound engineering technologies and their active development and dissemination have helped to create opportunities for many performers to record their musical works; accordingly, famous performers are becoming much more and this significantly affects the change in "status", which is that the performer becomes much closer to the listener than before. This is emphasized by the use of sheet reverberation with significantly less time ("Plate"). The further evolution of the music industry, the spread of new multimedia technologies (the emergence of music television) changes the "status" of performers again: the listener is not only right in front of performers, but between them, which is emphasized by reverberation of very small space ("Room"), moreover, very "long" reverberation is also used as special effect, but exectly short one prevails.

It is expedient to characterize the basic methods of using artificial reverberation in modern sound engineering.

1. Using reverberation to create a sound perspective. Artificial reverb (one or different types) is added to sound objects. In the conditions of "analog" sound engineering, the use of several types of reverberation determined by the time of reverberation was introduced:

1. "room" is the "short" reverberation, used mainly to create a close-up (Ueno, Kato & Kawai, 2010);
2. "plate" is the effect of sheet reverberation, which is used to create a medium plan;
3. "hall" is the "long" reverberation, used to create a long-range plan. Moreover, in the middle of the last century in concert sound engineering for this purpose a single device of artificial reverberation was used, and plans were created by different ratios of processed and unprocessed, by the reverb, signal on each channel individually. The basic pattern is that increasing the reverberation used for a particular sound object causes the object to move away from the listener in the secondary sound space. However, to achieve a sense of common space, a single type of "short" reverb ("room") is usually used for all sound objects.

2. Using reverberation to process backing vocals. Traditionally, all vocal channels use reverberator processing. This is largely due to the fact that the distance between the singer and the microphone is small and the real space around (even if it is a concert hall) with such an arrangement of microphones is almost impossible to convey. For solo and backing vocals, the same effect of artificial reverberation is usually used, but there are cases when in the general sounding solo-vocal loses brightness against the background of backing vocals. Using the same artificial reverberation for backing vocals, but with a slightly longer reverberation time, allows you to virtually remove this part, while maintaining a certain unity with the solo. Usually, in combination with frequency processing (for example, reducing the high-frequency component of the signal), this technique gives positive results.

3. Using reverb on send delay (sequential processing of delay reverberator). Delay, as one of the common devices for time signal processing, creates signal repetitions, in which you can adjust the number of repetitions, the time between them and their "attenuation". However, such sharp repetitions of the same type can impair the perception of other sounds, but the use of delay reverberator (with a short time) will make the repetitions less bright, and their impact on "legibility" is significantly reduced. In addition, this technique allows you to "place" the delay as if in a separate sound space, which has a positive effect on the sound perspective. Also, the use of artificial reverberation, in combination with other effects of temporal signal processing, can be used as a special effect and even form a separate "party" in the process of arranging musical works of modern musical styles (e.g., ambient).

4. Using reverberation to "soften" the sound of individual parts of musical instruments. Artificial reverberation, which simulates a very small space (usually, "Room"), allows, during short time, to "soften" slighty the sound of musical instruments, especially those, for which, traditionally, reverberation was not used at all (such as bass guitar). The short reverberation time makes it inconspicuous in general sound, however, the part is no longer perceived as completely "dry". Varying the reverberation time in combination with dynamic processing allows you to significantly change the perception of "attack" in the sound of a musical instrument.

5. Using reverberation to process different signals of the same type. These are situations where the sound design of vocalists duo or any musical instruments is carried out. Similar in

content, but essentially different signals require their own approach of artificial reverberation processing. Similar signals often differ greatly in timbre. Different reverbs will make such signals even more contrasting. For example, you can create the illusion that two microphones are in a different environment. However, in the conditions of concert performance, the difference in signal processing should be noticeable, but not striking, because significant differences in reverberation can, on the contrary, worsen the perception of a musical work.

6. Using two reverb effects when processing a solo part (vocal or musical instrument). Artificial reverberation of one or more closely related species is added to the solo channel (vocal or musical instrument). At the same time, settings of each of the reverbs should be different: a slightly different time of decline, intensity (changes in settings should not be drastic, but noticeable). The signal processed by reverberators behind the panorama is not in the center, but slightly to the left and right, then these processed signals are processed by a saturator or magnetic tape emulator, which allows them to "glue" together. The performed actions will create an original effect of stereoreverberation. The solo will keep the space in the center, will not lose its expressiveness, but at the same time will gain the necessary volume and will not lose its mono compatibility.

7. Using two reverberation effects when processing different drums. A separate effect of artificial reverberation, for example, for a small (working) drum is used in parallel with the reverb on the common channel of the entire drum unit, that is a fairly common technique in the processing of drums (Cavaco, Lewicki, 2007). It allows you to further highlight the drum on the background of the entire installation and, at the same time, helps it to become more visible in the overall sound picture. It is most expedient to use this technique when working with acoustic drums, in which two microphones are used for the working drum - the upper and lower. Sound processing from the lower microphone will make the working drum denser and brighter. To implement this technique, in parallel with the raw signal from the lower microphone of the working drum, it must also be processed sequentially with a delay of about 7-15 milliseconds, and then with a reverb (with a small reverberation time) based on the music specifics. Next, combining the sound of the raw lower microphone and processed one in different proportions, it is necessary to ensure that the working drum "merged" with the whole installation, while gaining a noticeable high-frequency accent (created by the sound of strings). The same technique can be applied to volumes. The signal from the microphones of each volume should be processed by a separate reverb, which should weaken the late reflections and strengthen the early ones, then you need to combine the signals of the volumes, so that the amount of reverberation was about 10%.

In the conditions of active development and improvement of the process of professional training of sound engineers, significant stylistic differentiation in the art of music, it is obvious that currently a large number of different original methods of artificial reverberation are used; accodinly, search and selection of specific techniques for a particular piece of music, as well as means of their implementation, are the result of search and personal "vision", and experience of each sound engineer. In addition, achievment of the same "creative" result in sound engineering is possible with different "technical" means and solutions.

In the process of professional training to master the use of artificial reverberation, it is advisable to start with acquaintance with the main types of reverberation ("Room", "Plate", "Spring", "Hall", etc.) and how different effects of reverberation "sound" on the same signal. As sound material for higher education students, it is better to use pulse sources (for example, single shoots of drums), because they provide a better understanding of the nature of reverberation that is applied to them. In the future, you can use your own voice for this. The previous experience (first of all, listening) of students and their ability to "imagine" sound is important. Attending events in specially designed concert halls creates in educators their own indicators of "optimal reverberation", for example, for musical works performed by a symphony orchestra. At the same time, listening to sound recordings of musical works of a certain style forms a "reference" sound for a particular musical style, in particular, the "amount" of reverberation used in the compilation process. It is advisable to vary only the basic parameters at the initial stage, in particular the reverberation time. At this stage, even an "analog" sound effects processor can be used, as even the simplest models feature basic types of artificial reverberation and reverb time is available for adjustment. However, it is advisable to use "built-in" reverb plugins to the digital audio workstation.

Among the free VST plugins of artificial reverberation are several that are available for mastering even at the initial stage of training, and, because they are free, they can be used almost without restrictions.

1. "Abstract Chamber" is a fairly easy-to-master artificial reverberation plugin that processes sound in the stream of natural but abstract spatial impression. Pre-delay, reverb time, which gives a theoretical average frequency of RT (Reverb time) 60, and modulation, which controls the depth and speed of internal modulation, are available to the user; at low settings it is mostly depth, at high settings it is mostly speed. In addition, the plugin allows you to adjust the low-frequency and high-frequency damping, as well as the mode switch: insert mode will mix reverb with dry (raw) signal, otherwise only the processed signal will be output.

2. "AlgorVerb" is also quite easy to master stereo reverb. Within the proposed presets, "Damping", "Decay", "PreDelay", "Dry/wet levels" (ratio of processed and unprocessed signals) and “Brightness” are available for control in this plugin. Despite its simplicity, the plugin allows you to use artificial reverb to perform a wide range of tasks during the compiling process.

3. "Valhalla Supermassive" from the company "ValhallaDSP" (developer Sean Costello) is a plugin that can be used as a delay and reverb. This plugin mixes up dozens of delays and then combines them to create a reverb effect. The plugin works in eight modes, and users can control the effect with the controls of attack, decline and echo density at the same time. The user can change the delay time (up to 2 seconds), adjust the display behavior, set the ratio of processed and unprocessed signal, change the amount of reverb and process the signal in the built-in 2-band equalizer. “Warp” regulation controls the mixing of delays into a single signal, which allows the plugin to generate the simplest delay effects and rich reverberations. With the effect density adjuster, users can adjust the amount of reverb output. Additionally, the interface provides LFO (low frequency oscillator) for sound modulation, thanks to which Valhalla Supermassive can be used as a chorus and flanger.

4. "Classic reverb 8X" is an artificial reverb plugin, which implements the adjustment of 16 reverb parameters, in particular, setting the delay frequency, and also has the parameters "Room" and "Damp". The plugin has 32 internal slots for storing presets. A significant advantage is the simple interface and significant reverb settings.

5. Ambient Reverb is a plugin of artificial reverb with a wide range of reverb time (up to 100 seconds), which allows you to get different types of reverb, as well as the ability to "freeze sounds", while receiving interesting sound "pads". The plugin works on the principle of algorithmic reverb with the calculation of fairly dense sound displays over time, allowing you to get realistic reverb without the effect of "graininess" with relatively little load on the computer processor at the same time. The plugin has 16 presets, 2-band parametric equalizer, adjustment of input and output signal level.

Among the more sophisticated free VST plugins of artificial reverberation it is possible to characterize some which are expedient to master at a later stage of training, as they allow you to change a significant number of reverb settings, but at the same time they give considerable freedom of creativity to the sound engineer.

1. "MConvolutionEZ" from the company "MeldaProduction" is a plugin that contains a significant number of pulse "convolutions", but only a small number of settings at the same

time. The user can control the width of the stereo of active pulse characteristic and the shape of the tone, which is realized using several filters. The plugin allows you to download external "convolutions" in WAV and FLAC formats. Customizable and easy to use plugin allows you to consider it as one of the best for starting to master complex reverb plugins, in particular, "convolutional".

2.Convology XT from Impulse Record Convology XT is a plugin that has 70 pulse characteristics captured from many hardware reverbs. It is important to note that the plugin is able to load extraneous pulses in WAV format, respectively, "Convology XT" can be described as the plugin with the greatest potential for expansion in this list. The user is available to adjust the attack time ("Attack") and attenuation ("Decay"), "Pre-Delay" and frequency characteristics ("Frequency response"). You can also "stretch" files of the pulse characteristic and apply modulation to expand the stereo. However, the plugin does not implement the ability to adjust the delay time ("Latency"). Among the advantages of this plugin is the availability of several training videos from developers designed for users who are just starting to work with this plugin.

3. "SIR" from the company "SIR Audio Tools" is one of the first plugins of "convolutional" reverberation (therefore compatible only with 32-bit programs), characterized by a fairly simple interface and a relatively wide range of features. The user is available to configure "Pre-delay" and vary the envelope of the loaded "convolution", which can also be "stretched", which can be considered a significant advantage for the historical period when the plugin was developed. The SIR has a fixed delay time ("Latency") of 8960 samples and is optimized to reduce the load on the computer's processor.

4. "Reverberate LE" is a plugin, also developed for 32-bit programs, but a little later than "SIR". The reverb does not contain a significant number of built-in "convolutions", but you can download the pulse characteristics files in the format FLAC, AIFF, WAV (the plugin also supports stereo pulse characteristics). After loading, the pulse characteristics can be processed using the envelope and "Stretch" parameters. The plug-in also has a built-in parametric equalizer that can radically change the sound of samples of pulse characteristic, in addition, it is able to work in zero delay mode ("Zero latency mode"), while the graphics card will be used.

5."Halls Of Fame 3 Free" from the company "Best Service" is a "truncated" version of the reverb "Halls of Fame 3" - "Complete Edition", which, however, contains 27 presets and does not support external pulse characteristics, but existing "convolutions" (with EMT 240, AKG BX 20, Lexicon PCM96 and the Bricasti M7) are implemented quite well and they are enough to implement many tasks of sound engineering. Presets are configured using a detailed bypass or tone control settings. The user can set the time "Pre-delay", "Attack/release". A 3-band tone and dumping control is used to form the "reverberation tail".

In the conditions of modern development of information technologies and digital sound processing, various means (both hardware and software) which allow to master artificial reverberation are available. For most modern VST-plugins, manufacturers are creating training videos to make it easier to master the interface and capabilities of a particular product. They can be considered as an important and useful tool for beginners, which will help not only to understand the possibilities of artificial reverb and the parameters that it allows to vary, but also to develop a general understanding of artificial reverberation and its importance in sound engineering.

Conclusions and prospects for further research. Thus, the development of modern digital technologies in the field of sound engineering, the introduction of algorithms for modeling the pulse characteristics of the premises have significantly affected the possibility of using artificial reverberation in sound engineering. In the practice of sound processing, a significant number of effective methods of using artificial reverberation for the implementation of various creative tasks in the process of sound engineering have been developed. Historically, different types of artificial reverberation have been used at different stages of development of sound engineering and sound recording. At the present stage, the software allows you to use several reverb algorithms simultaneously, which, in turn, allows you to get original sounds in modern music. In the process of mastering of various methods of using artificial reverberation by higher education students, it is best to start their acquaintance with artificial reverberation from the simplest reverberation effects, which can be implemented in "analog" effects processor, in which only some parameters vary based on the manufacturer's algorithms. Further mastering of artificial reverberation should be done through software (in particular, free VST-plugins of artificial reverberation), which allows you to change a significant number of reverberation parameters. At the final stage of mastering the effects of artificial reverberation, it is advisable to use modern convolutional reverberators. The ability to use artificial reverberation is positively affected by the accumulation of listening experience both in different concert halls and from listening to different sound recordings of musical works. Further research in this area should focus on the development of technologies for combining different reverberation algorithms and methods of their use in modern sound engineering.

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МЕТОДИ ВИКОРИСТАННЯ ШТУЧНОЇ РЕВЕРБЕРАЦІЇ У СУЧАСНІЙ ЗВУКОРЕЖИСУРІ

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Анотація. Стаття присвячена систематизації основних сучасних методів використання реверберації у звукорежисурі та методичним основам її застосування у освітньому процесі у закладах вищої освіти. Наукова новизна полягає в уточненні класифікації та систематизації основних сучасних методів використання штучної реверберації, а також методичних засад опанування штучної реверберації у процесі професійної підготовки. Мета статті полягає у систематизації основних сучасних методів використання реверберації у звукорежисурі. Для реалізації мети використано методи: системно-аналітичний, мистецтвознавчий та порівняльно-історичний. Основні результати і висновки дослідження. Удосконалення цифрових технологій у галузі звукорежисури, запровадження ревербераційних алгоритмів значною мірою сприяли розширенню використання штучної реверберації у звукорежисурі. У статті коротко схарактеризовано «згорткову» реверберацію як таку, яка оперує найбільш сучасним ревербераційним алгоритмом. Наразі у звукорежисурі напрацьовано різноманітні ефективні методи використання штучної реверберації, які можуть застосовуватися для реалізації різноманітних творчих завдань. У статті схарактеризовано методи використання штучної реверберації для створення звукової перспективи; для обробки back-вокалу; для обробки дилею; для «пом'якшення» звучання окремих партій музичних інструментів; для обробки різних однотипних сигналів; для обробки партії соло з використанням двох ефектів реверберації. Окремі види штучної реверберації використовувалися на різних історичних етапах розвитку музичної індустрії. Сучасний стан розвитку програмного забезпечення дозволяє використовувати одразу декілька алгоритмів реверберації, завдяки чому наявна можливість створення оригінальних звучань. У процесі опанування здобувачами вищої освіти різних методів використання штучної реверберації найбільш доцільно розпочати їх знайомство зі штучною реверберацією з найпростіших ефектів реверберації, які можуть бути реалізовані в «аналоговому» процесорі ефектів, у якому на основі алгоритмів виробника варіюються лише деякі параметри. Подальше опанування штучної реверберації доцільно здійснювати через програмне забезпечення (зокрема, безкоштовні VST-плагіни штучної реверберації), що використовується у цифрових звукових робочих станціях (DAW).

Ключові слова: штучна реверберація; «згорткова» реверберація; ревербераційний алгоритм; метод використання штучної реверберації; плагін штучної реверберації