

Development of a Sampler of Folk Instruments Based on NI Kontakt

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Abstract— In the modern music industry, samplers are used in film music, the gaming industry, advertising, theatrical sound design and sound design. Their practical value lies in quick access to timbres, the recording of which in each project would require separate performers, premises and recording equipment. The article presents the result of the development of a digital sampler for folk stringed instruments based on NI Kontakt 8. The relevance of the chosen research and the presented development is due to a specific practical task: sampler libraries are already used in sound engineering, sound design, educational projects and music production, and folk instruments in such libraries are much less represented than pianos, percussion or orchestra groups.

Keywords— folk instruments, music computer practice, music computer technologies, sampling, NI Kontakt.

I. INTRODUCTION

The work solves one of the urgent problems of sound design and modern musical sound engineering.: preserving the artistic authenticity of the sound of instruments and searching for optimal solutions for creating a controlled virtual musical instrument.

The purpose of our work was to develop a digital sampler for folk stringed instruments based on NI Kontakt 8 and to test its functional, sound and technical characteristics. The main tasks set by the authors of the study included: description of digital timbre transmission of folk instruments; consideration of NI Kontakt as a development environment; design of the library structure; preparation of audio samples; development of a tool with Volume and Reverb controls, as well as with Pluck/Tremolo switching; carrying out functional and audio technical verification of the result.

A modern sampler is wider than a regular audio player. It combines source material preparation, file editing, mapping, groups, articulations, effects, and user interface. As a result, the library works as a managed audio data system [34]. Sampler technologies are especially noticeable for folk instruments.

For example, the balalaika, domra, and bass balalaika have a timbre associated with the body, strings, plucking method,

tremolo, and performing style. At the same time, it can be difficult to find a performer, a suitable room and a high-quality tool for each creative project. A digital sampler solves this practical problem: it preserves a recognizable sound and makes it available in a MIDI environment.

To create a sampler, it is not enough to record sound on a good microphone: you need to slice WAV files, assign a root key, assemble zones in the Mapping Editor, separate the pinch and tremolo into groups, connect processing, display control on the panel and check that everything opens in Kontakt without lost files [30]. It was important for our work to get a concrete result: to transfer the recognizable sound of the balalaika prima, domra and bass balalaika to a MIDI-controlled environment [6]. The MIDI standard plays a special role in this process. For example, it transmits information about the keystroke, duration, height, strength of the performing action, controllers, and other parameters [26]. This allows the sampler to respond to the note, articulation, volume, timbre, and effect if these connections are pre-configured.

It should be noted that the problem of preserving the intonational nature of folk instrumental music during sampling has been considered in detail in the work of many musicologists and researchers (see, for example, [1]).

II. PRINCIPLES OF SOUND SAMPLING OF MUSICAL INSTRUMENTS

The sampling history shows the movement from automatic recording playback to complex software systems. The sampler combines recording, storage, processing, mapping, MIDI control, performance response, and interface. For this work, such a review is needed as a foundation: the sampler of folk instruments is located at the intersection of musical art, sound engineering and digital technologies. In this sense, it is part of a broader semantic organization of musical material in a digital environment [15; 16].

One of the main tasks when creating a sampler is to preserve the natural performance response. In real performance, the force of a pinch or impact changes the volume, spectrum, attack, duration of attenuation, and overall timbre feel. Therefore, when preparing samples, you need to take into account soft, regular and accentuated sound production, even if they are not separated into separate layers in the current version. Psychoacoustically, the listener perceives dynamics as a change in volume and timbre simultaneously [11].

When working with extended sounds, loop points and sustain loop are important. Loop points are the start and end points of the looped fragment inside the audio sample. Sustain loop is used to prolong the sound of a note without having to record a very long audio file. This technology is especially relevant for wind instruments, strings, reed instruments and

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some electronic instruments where sound can be retained for a long time. The principle is that after the initial attack, the sampler plays back a certain section of the recording in a circle as long as the key is held down. After releasing the key, the sound completion phase starts. Here, the loop must be set up so that the transition from the end of the looped fragment to its beginning is imperceptible. Incorrectly set loop points can cause clicks, ripples, phase artifacts, and a feeling of artificial sound. The technical setting of the loop is related to the general principles of digital signal processing and the prevention of artifacts at the boundaries of the audio fragment [35].

The nature of the sound is strongly influenced by the ADSR envelope: attack, decay, sustain and release. Attack sets the rise time after pressing the key; decay — the transition from the maximum level to the hold level; sustain — volume when holding the key; release — fading after releasing. Plucked and percussion instruments usually have a fast attack and natural attenuation, while wind instruments and bowed instruments have a softer beginning and are controlled by the performer. In the sampler, ADSR helps to adjust recorded samples to the logic of the game and make the instrument more predictable [27].

In addition to the basic sample organization, digital audio processing plays an important role. Filtering is used to change the spectral composition of a signal: it can be used to attenuate low, medium or high frequencies, simulate a change in tone with different dynamics, or create special sound effects. Equalization allows you to adjust the frequency balance, eliminate excessive resonances, emphasize the characteristic areas of the timbre and achieve better compatibility of the instrument with other elements of the arrangement. Compression is used to control the dynamic range: it can equalize the volume, enhance the attack, increase the sound density, or, conversely, be used minimally to preserve the natural dynamics of the original instrument. Reverberation adds a sense of space and helps to place the sampler instrument in a specific acoustic environment. Here, the processing should be commensurate with the artistic task: excessive use of effects can lead to a loss of timbre authenticity, especially when working with folk instruments. The principles of equalization, compression, and reverberation for mixing are described in detail in works on practical audio processing [24].

Before importing audio material into the sampler, you need to perform technical preparation of the samples. It includes noise reduction, editing of initial and final fragments, normalization, elimination of clicks, volume equalization and pitch control. Noise reduction reduces the level of background noise that occurs during recording, but it should be used carefully, as excessive processing can damage the natural sounds and overtones of the instrument. Normalization is used to bring the audio file level to a preset maximum value, which makes it easier to further adjust the volume inside the library. Here, normalization should preserve the musical dynamics: if all samples are artificially adjusted to the same volume without taking into account the nature of the performance, the instrument may lose its natural expressiveness. Modern sound recording manuals emphasize that editing should preserve the musically significant characteristics of the original performance [21].

The main difficulty of sampling is to preserve the natural timbre during digital playback. In a live instrument, the sound varies depending on the position of the hands, the force of the pinch, the body material, the acoustics of the room, microdynamics, noise components and resonances. If the library contains too few samples or does not take into account articulation, the sound quickly becomes mechanical. Therefore, the developer needs a technically clean recording with preserved performance characteristics of the instrument; in mastering practice, this balance is associated with the professional suitability of the audio material [25].

III. THE SPECIFICS OF DIGITAL REPRESENTATION OF FOLK INSTRUMENTS

The digital representation of a folk instrument in this work means that its main features are conveyed in the program: timbre, attack, reaction to performance, articulations, noise components and recognizable sound style. We are talking about a system of samples, mapping, groups and controls that preserves the acoustic and artistic character of the instrument [2].

The timbre of folk instruments is largely determined by the design. The electronic timbre can be assembled by generators, filters, and modulators; an acoustic folk instrument sounds through the interaction of an oscillation source, body, and room. For strings, string material, tension, body shape, soundboard, and resonance are important. Lingual ones have tongue behavior, air pressure, and a resonator system. The membrane ones have the membrane material, impact force, contact zone and housing. The wind instruments have an air flow, a channel, holes and a mouthpiece. Therefore, each folk instrument has its own sound profile, which cannot be reduced to several frequencies. The concept of the architectonics of musical sound allows us to consider timbre as a system of acoustic and artistic parameters [3].

The transmission of articulations is the most difficult. In a live performance, a folk instrument rarely sounds like a series of identical notes. Expressiveness is born from a pinch, a bump, a slip, a tremolo, a glissando, a foreshock, a tremor of sound, short transients and microdynamics. The same sound is perceived differently if it is taken softly, sharply, with an accent or with a noise component. For the sampler, this means recording the main notes along with the performance options; without them, the digital version becomes schematic. In works [3; 4] it is emphasized that the sound structure of musical material should be analyzed by pitch and timbre-temporal characteristics.

An important parameter is the sound attack, that is, the initial area after sound extraction. It is by attacking that the listener often recognizes the instrument. In folk instruments, it can be uneven, unstable, and saturated with noise elements. If you clean the beginning of the sample too much, you may lose the characteristic timbre; if you leave an extra pause or a dirty click, the instrument will be inconvenient in the MIDI part. The developer has to find a working balance between quickly launching the sample and preserving the natural character of the original sound. The zone approach to musical time

considers the beginning of a sound as a significant element of a musical phenomenon [5].

Microdynamics also affects plausibility. In a live performance, the power of sound production varies between loud and quiet notes, within short phrases, repetitions, embellishments, and transitions. These changes affect the level, spectrum, duration of attenuation, overtones, and noise components. If the sampler reproduces everything too smoothly, the instrument sounds flat. Psychoacoustic studies show that microdynamic differences are involved in the recognition of the sound source and the nature of the performance [11].

Height standardization is a separate issue. Many folk instruments developed in specific regional traditions and do not always completely coincide with a uniformly tempered system. In live practice, small pitch deviations can work as a characteristic feature of the sound. When creating a sampler, you need to choose the degree of correction: fine tuning facilitates the arrangement, and preserving intonational features enhances cultural authenticity. When sampling folk instrumental music, this intonation specificity is considered as a condition of artistic persuasiveness [3].

Standardization of volume also requires caution. Individual notes of a folk instrument can naturally vary in level due to the resonance of the body and the method of sound production; full alignment makes the instrument technically smooth, but less lively. The result is strongly influenced by the recording conditions. Folk instruments often have a complex spatial and resonant sound structure, so the microphone, its position and the acoustics of the room are important. Recording too closely highlights noises, attack, sounds, and the mechanics of sound production. Recording too far away enhances the room and makes it difficult to continue using samples in different projects. An optimal microphone scheme should maintain timbre reliability and leave the library suitable for a digital environment. Moylan considers sound recording as an artistic construction of the spatial and timbre image of an instrument [28].

The traditional context of performance also influences digital representation. The folk instrument exists as an acoustic object and as a part of musical culture: it has stable intonation turns, rhythmic formulas, decorations, phrasing and performing gestures. If the library reproduces only individual notes, it transmits the outer layer of sound. A more convincing version requires typical articulations, transitions, and ornaments. Works on the preservation of musical folklore by means of music and computer technologies show that a digital instrument must take into account the cultural context [6].

Excessive digital processing can erase the ethnic expressiveness of the instrument. Modern editors allow you to equalize pitch, volume, spectrum, attack, duration, and noise. When the processing removes all the irregularities, the instrument loses its individuality and approaches an average timbre. This is especially dangerous for folk instruments: their value is often based on timbre heterogeneity, roughness, natural resonances and performance details. Technical purity should work together with maintaining timbre recognition [5].

- The specifics of the digital representation of folk instruments consist of the acoustics of the body, articulations, attack,

microdynamics, structure, resonances and the performing context. Such instruments require careful recording, mapping, sound options, and preservation of noise components. A high-quality sampler should convey the recognizable expressiveness of the instrument and at the same time work as a convenient digital instrument. Therefore, the development of such libraries is at the intersection of sound engineering, musical acoustics, digital technologies and the preservation of intangible cultural heritage [6]. environment.

IV. THEORETICAL APPROACHES TO ASSESSING THE QUALITY OF A DIGITAL SAMPLER

Evaluation of the quality of a digital sampler is needed as a stage of practical development of a virtual musical instrument. An ordinary audio file contains a recording, and a sampler is a system: audio material, a sample distribution map, groups, articulations, a user interface, and software control mechanisms. Therefore, the quality cannot be determined solely by the impression of the sound or by the technical parameters of the recording. An integrated approach is needed, where the acoustic, performance, functional and operational characteristics of the instrument are checked [22].

The main quality criterion is timbre reliability. It shows to what extent a digital instrument retains the acoustically and artistically significant features of the original instrument: spectrum, attack, attenuation, resonances, noise components, and general recognizability. This criterion is especially important for folk instruments, because their artistic value is based on individual coloring, local style and design.

The second criterion is a natural performance response. In real performance, the power of sound extraction changes volume, timbre, duration, overtones, and attack. In the project, this criterion is tested through the persuasiveness of the recorded articulations, the evenness of the volume and the absence of sharp timbre dips. The MUSHRA technique is used to compare sound options in audio systems [23].

The absence of noticeable technical artifacts is checked separately: clicks at the sample boundaries, recording noise, digital distortion, sudden transitions, bad loops, phase problems and transposition defects. Such problems are immediately audible in solo fragments, clean parts and repetitive rhythms. At the same time, it is necessary to distinguish between recording interference and the natural noise of the instrument: the rustle of fingers, the rattle of the case, or the element of attack may be part of the timbre. The EBU recommendation R 128 sets guidelines for the normalization of volume and the permissible maximum audio signal level [13].

The next criterion is the uniformity of sound over the range. An important characteristic is expressiveness in MIDI performance. A digital instrument can be technically correct, but inconvenient or expressionless when playing from a MIDI keyboard. Expressiveness is determined by the extent to which the sampler allows the performer to use different articulations, accents, repetitions, and phrasing. In this project, the main focus is on switching groups between plucking and tremolo sounds, as well as on the correct distribution of instruments by registers [12; 22; 34].

An essential criterion is the convenience of the user interface. In this work, the interface is intentionally limited to the basic controls: volume control, reverb control, and group switching between pinch and tremolo sounds. This set corresponds to the actual implemented version of the tool and does not overload the user with unnecessary parameters. The logic of user control in Kontakt is related to the architecture of the tool and the ability to display parameters on the front panel [30].

Along with the sound and interface properties, it is necessary to evaluate the stability of the instrument in the NI Kontakt environment. Stability presupposes the correct loading of the instrument, the absence of errors when playing samples, and the correct operation of groups, zones, sound options, scripts, and interface elements. An important indicator is also the tool's ability to save settings, open correctly in different projects, and function inside a digital sound station. If a sampler crashes, downloads files incorrectly, loses connection with samples, or causes playback errors, its practical value is significantly reduced, even with good source audio. Testing in different Kontakt operating modes is described in the official user manual [31].

Another criterion is suitability for arrangement and sound design. The sampler should work as a means of creating musical material: to fit into the composition, combine with other instruments, be amenable to processing and provide material for realistic or stylized parts. In sound design, timbre often becomes the basis for further transformation: spatial processing, filtering, transposition, and texture creation. In the theory of sound recording, a musical result is evaluated based on its behavior in the mix and arrangement [28].

It is also necessary to evaluate the balance between sound quality, library size and system load. Large commercial libraries can include many files, duplicates, articulations, and microphone positions; this increases realism, but requires more RAM, disk space, and processor resources. A library that is too small saves resources, but it quickly loses its expressiveness. For such a tool, a reasonable compromise is needed: the sound must be convincing, and the project must run on a regular computer [13].

Evaluation of a digital sampler can include both objective and subjective methods. The objective ones include checking the range, analyzing the volume of samples, detecting artifacts, checking the correctness of mapping, measuring the load on the processor and RAM, testing loading and playback in NI Kontakt. Subjective methods are associated with expert listening and assessment of timbre reliability, naturalness of dynamics, convenience of performance and artistic expressiveness. A combination of these approaches is most appropriate, since a technically error-free instrument may be musically inconclusive, and an expressive sound may be accompanied by operational disadvantages. A comprehensive assessment makes it possible to identify both the strengths of the development and the directions for its further improvement. The combination of objective verification and subjective listening corresponds to the logic of the ITU-R recommendations for evaluating the quality of audio systems [23].

The choice of a software environment for the development of a digital sampler of folk instruments determines the structure of the future instrument, the quality of its sound, ease of operation and prospects for practical application. For this work, a platform is needed that allows you to play audio files, create mapping, organize groups, implement a user interface, connect effects, and use scripting logic. NI Kontakt meets these requirements [30].

One of the main arguments in favor of NI Kontakt is its prevalence in music production, sound engineering, film music, the gaming industry, and sound design. Kontakt has long been used as a standard platform for commercial and custom sampler libraries; many professional virtual instruments are created in this environment. This is important for setting a practice-oriented task, because the development in a widespread professional program connects the project with real practices of digital music production [10; 17; 18].

A significant selection criterion is the ability to create a graphical user interface. A full-fledged sampler instrument should be convenient for a musician, sound engineer, or student who will use it in practical work. In this version of the project, only the actual elements are brought to the interface: volume, reverberation, and switching groups between pinch and tremolo. This demonstrates the completeness of the development and takes it from the level of a technical experiment to the level of a practically applicable software product [8; 9].

NI Kontakt's strong point is script management via Kontakt Script Processor. KSP processes MIDI events, creates controls and switches, manages groups and connects the interface with the internal parameters of the instrument [29]. In this work, these capabilities are used for volume, reverberation, and switching of pinch/tremolo groups.

The choice of NI Kontakt can be further justified by comparing it with other software solutions. Ableton Sampler is a convenient tool inside Ableton Live and is well suited for fast work with samples within the framework of this DAW. However, its capabilities as an independent library platform are more limited, especially when it comes to creating a complete user tool with an interface, a complex articulation system, and independent distribution. Logic Sampler also has advanced sampling features, but it is closely linked to the Apple ecosystem and Logic Pro, which reduces its versatility for users of other operating systems and workstations.

Decent Sampler is a simpler and more accessible platform that can be useful for creating lightweight sampler tools and open distribution of libraries. However, in terms of the depth of professional development, the flexibility of script management, the level of interface integration, and the prevalence in the commercial industry, it is inferior to Kontakt. The SFZ format is an open and fairly flexible way to describe sampler instruments, but working with it requires a different development logic based on a textual description of the parameters. It can be effective for certain tasks, but creating a user-friendly interface and complex interactive logic in it often turns out to be less visual and less convenient. NI Kontakt is the most balanced solution. It is developed enough to demonstrate a professional approach to the development of a digital instrument, and at an accessible level you can master the main

stages of creating a sampler library: audio preparation, mapping, group setup, interface creation, effects application, and the use of a basic KSP script [19].

The use of NI Kontakt in the work does not exclude the prospect of domestic developments. At the other extreme, the analysis of the professional platform makes it possible to identify the requirements for a modern sampler: the library structure, articulation management, interface, processing and stability of work in the DAW [15].

The choice of NI Kontakt is related to technical, practical and educational reasons. The environment is widespread in the music industry, supports user libraries, provides advanced mapping and grouping of samples, allows you to create an interface, connect KSP, use built-in effects and work inside a DAW. In this work, these capabilities are applied to an instrument with a register distribution of three folk timbres, volume control, reverb control, and Pluck/Tremolo switching.

V. DESIGNING THE STRUCTURE OF A DIGITAL SAMPLER

Designing the structure of a digital sampler is the first practical step, where the sound composition, range, sample distribution logic, folder structure, and controls are specified. In this work, the sampler is assembled as a sequential project with a predefined audio file structure. This preparation reduces the risk of errors during recording, processing, and further assembly in NI Kontakt. Three folk stringed instruments were chosen as the basis of the digital sampler: the prima balalaika, the four-string domra with a fifth system, and the balalaika bass. This set covers different register zones of folk string sound and expands the overall range of the instrument. The prima balalaika gives the upper register and a bright plucked timbre, the domra forms the middle register with a more rounded and mobile sound, the bass balalaika is responsible for the low base. Due to this, the sampler is built as a combined instrument, where different folk timbres are distributed over octaves and perform different functions within the general range. The choice of balalaika and domra is connected with the tradition of folk-orchestral instrumental style, formed in Russian musical culture [2].

The sampler's sound range was determined based on the register role of each instrument. The low register, which includes a part of the octave starting with the mi note and a large octave, is assigned to the bass balalaika, since it has a dense low-frequency sound and can serve as a bass base. The middle register, covering the minor and first octaves, is assigned to a four-stringed domra with a fifth system. The upper register, which includes the second and third octaves, is assigned to the balalaika prima, which is characterized by a bright attack, sonority and expressiveness in the high range. This distribution avoids excessive transposition of samples and preserves the more natural sound of each instrument [1].

At the design stage, the main articulations presented in the sampler were identified. In this version of the instrument, two sound extraction modes are implemented: pinch and tremolo. Plucking is used as the main short method of sound production, while tremolo expands the expressive capabilities of the instrument and is especially important for balalaika and domra. Limiting articulations to two modes helps to keep the project

manageable and ensure high-quality tuning of each type of sound.

The number of samples is determined by the selected range, the number of instruments and the presented articulations. In this work, the main focus is on two modes of sound production — plucking and tremolo. The volume of the library is correlated with the actual recording, processing, and configuration capabilities in Kontakt: it is sufficient to demonstrate how the tool works, but it does not turn the project into an overly complex commercial library. Modern recording technology requires planning in advance the number of takes, articulations, and microphone fixation conditions [21].

The WAV file was chosen as the source audio file format, since this format is a common standard for uncompressed digital audio and is widely used in professional sound recording and sampler libraries. The use of WAV helps to maintain the quality of the source material without loss associated with compression. This is especially important for sampler development, as audio files can be further processed, transposed, looped, and played back under different dynamic conditions. Compressed formats, such as MP3, are impractical for such a task, as they may contain encoding artifacts that become noticeable during repeated processing or transposition. Therefore, the choice of WAV ensures the technical reliability and sound quality of the future instrument. The use of an uncompressed format meets the requirements of preserving the original audio quality during subsequent processing [32].

For the convenience of development, it is necessary to determine in advance the folder structure of the library, taking into account the logic of the organization of Kontakt tools [30]. It is better to use several main catalogs in the project: Samples, Instruments, Scripts, Resources, Documentation and Presets. The Samples folder contains the original and processed audio files, distributed by instruments and articulations. The Instruments folder is intended for Kontakt instrument files. The Scripts folder may contain fragments of the KSP code used to control the interface, articulations, and sound parameters. Resources can include graphical elements of the user panel, images, backgrounds, icons, and other supporting materials. The documentation contains a description of the library structure, terms of reference, sample tables, and explanations for using the tool. The Presets folder is suitable for storing sound settings if the project has several user configurations. The proposed structure of the set of library files is shown in Figure 4. It demonstrates the engineering organization of the project and reduces the risk of loss of links between an nki file and audio material [30].

The file naming scheme should be unified [30] and understandable, since the convenience of importing samples into Kontakt depends on it. The file name should better reflect the instrument, note, octave, articulation, and take number. For example: BalPrima_C3_Pluck_Take1. wav, where BalPrima stands for prima balalaika, C3 stands for pitch, Pluck stands for pinch articulation, Take1 stands for take number. The designation Tremolo is suitable for tremolo. This system allows you to quickly determine the contents of a file without listening and reduces the risk of mapping errors.

The recording plan provided for sequential recording of the material for each instrument. First, the room, microphones,

audio interface and the layout of the performer were prepared. Then the main notes of each instrument were recorded in the appropriate range, as well as two sound extraction modes: pluck and tremolo. During the recording process, the input signal level was monitored, clipping was eliminated, background noise was monitored, and a stable distance between the instrument and the microphone was maintained. The organization of recording as an artistic and technical process involves monitoring the signal level, space, and stability of the performance result [28].

The audio processing plan includes several sequential operations. First, the highest quality takes are selected, then samples are cut and unnecessary fragments of silence are removed at the beginning and end of the file. Next, you need to check the accuracy of the pitch, perform a moderate correction if necessary, eliminate extraneous noise, adjust the volume level, and prepare files for import into Kontakt. During processing, it is important to avoid excessive sterilization of the sound, since for folk instruments, natural sounds, attack, resonances and a slight heterogeneity of timbre are part of their expressiveness. Therefore, noise reduction and equalization should be applied carefully, while preserving the characteristic acoustic features of the instrument. At the processing stage, it is necessary to maintain a balance between technical correction and natural musical dynamics [25].

The logic of the distribution of samples on the keyboard in Kontakt was based on octaves [32] and the structure of the selected instruments. The bass balalaika was placed in the range from the fifth octave to the end of the major octave, the domra in the minor and first octaves, and the prima balalaika in the second and third octaves. Within each range, the samples were distributed based on their original height and assigned to the corresponding root note. The pinch and tremolo were organized as separate groups and switched through the user interface. The design of the user interface was subordinated to the task of conveniently managing the actual implemented parameters of the tool. In this version, the interface includes an overall volume control, a reverb level control, and group switching between pinch and tremolo sounds. This set of functions is sufficient to demonstrate the principle of interactive sampler control and does not overload the user with unnecessary elements. In an educational context, the simplicity of the interface is especially important because it facilitates the inclusion of the tool in educational and creative practice [20].

VI. RECORDING, PROCESSING, AND PREPARATION OF AUDIO SAMPLES

The practical stage was related to the recording, processing and preparation of audio samples, which became the sound basis of the instrument in NI Kontakt. The quality of the source material directly affects the timbre accuracy, expressiveness, stability and practical suitability of the sampler [21]. The recording was performed for three folk stringed instruments: the prima balalaika, the four-stringed domra with a fifth system, and the balalaika bass.

The RME UCX II audio interface was used as the main equipment, providing stable recording and high-quality analog-to-digital conversion. A Studer 89 preamp was used to

amplify the microphone signal. The quality of the sound recording circuit is essential for a sampler library, since defects in the original recording manifest themselves during repeated playback, transposition, and subsequent sample processing [21].

The microphone circuit was based on a pair of AKG C414 ULS and Neumann U87 ULS. This combination gave a detailed attack, pinch and articulation elements, as well as a tighter transmission of the instrument body. This is important for folk strings: recognition depends on the pitch, the nature of the plucking, the resonance of the body, noise sounds and natural attenuation [21].

The recording parameters were selected taking into account further work in NI Kontakt. The audio material was recorded with a sampling rate of 48,000 Hz, a 24-bit bit rate, and stored in WAV format. The 48 kHz frequency provides sufficient temporal detail of the signal, and the 24-bit bit depth helps to maintain a dynamic margin when recording acoustic instruments [32]. The uncompressed WAV format is chosen to avoid quality loss and encoding artifacts.

Two sound production modes were recorded separately — pinch and tremolo. In real performance, they differ in duration, attack, and timbre organization. Therefore, these recordings are then separated into Kontakt groups so that the user can switch between the two main sound options for folk stringed instruments.

The sound base was organized around two realized articulations — pinch and tremolo. This is enough to demonstrate the basic expressiveness of the tool and how group switching works, without stating functions that have not actually been implemented.

After the recording was completed, technical processing of the audio material was performed. First, extraneous noises unrelated to the natural sound of the instrument were removed: background room noise, random mechanical sounds, clicks, and fragments of silence. Here, noise reduction was applied selectively, since part of the sounds — the contact of the pick or fingers with the string, the initial noise of the attack, the residual resonance of the body — participates in the formation of the timbre recognition of the folk instrument [25].

A separate system for reacting to the force of a keystroke was Kontakt Script Processor allows you to create controls and switches, as well as link them to the internal parameters of the tool [29]. As part of this work, the script was used to control the volume, reverb level, and switching of pinch/tremolo groups. This made the tool more user-friendly and interactive. The result of the script is described below.

VII. TESTING, EVALUATION OF RESULTS, AND DEMONSTRATION OF SAMPLER PERFORMANCE

The created tool has passed the testing stage. The check concerned the loading of the sampler into NI Kontakt and its suitability for real musical use. The technical performance, correctness of mapping, switching of tweaks and tremolo, clarity of the interface, stability in the working environment and subjective sound were evaluated. The technique combines functional testing and listening in conditions close to practical application [23].

The operation of user interface elements was checked separately [30]. The volume control, the reverb control, and the pinch/tremolo group switch were supposed to change the corresponding parameters without delay or error. The interface was evaluated based on two criteria: technical correctness and practical clarity for the user.

Verification of spatial processing included evaluation of reverberation [25]. The processing should improve the spatial perception of the instrument without destroying the original timbre of folk instruments. Therefore, the testing was carried out both with solo listening and as part of a simple MIDI arrangement. This approach allows you to identify problems that are not always noticeable when playing individual notes: insufficient readability of the attack, excessive reverberation, or loss of timbre clarity.

The stability of the instrument was additionally tested in a digital audio workstation [31]. During the check, the download of Kontakt as a plug-in, playback of a MIDI part, saving project settings, automation work, and the overall load on the processor and RAM were evaluated. Since the library was created as a practice-oriented project, its volume should remain commensurate with the capabilities of a conventional computer. The optimal result is a configuration in which the instrument sounds expressive, but does not create an excessive load on the system.

A subjective assessment of the sound was carried out based on timbre recognition, naturalness of attack, persuasiveness of attenuation, absence of noticeable artifacts and the general suitability of the instrument for musical use. Such an assessment complements instrumental tests, because musical persuasiveness depends on perception in the performance and compositional context. This approach corresponds to the international practice of subjective assessment of the quality of audio systems [22]. This will allow comparing the prepared material with the result at the instrument output [32; 35].

VIII. CONCLUSION

The developed application was tested: screenshots of the Kontakt interface [30], a mapping scheme, a sample structure table, fragments of the KSP code, images of groups and zones, audio and MIDI demos. These materials confirm the practical development and make the verification reproducible. The testing was supplemented with a brief description of the library structure: the number of WAV files, the Pluck/Tremolo folders, the root key range, and the total volume. The check showed that the sampler is loaded into NI Kontakt, reproduces the prepared samples, distributes them into registers, applies spatial processing and gives the user three basic operations: Volume, Reverb and Pluck/Tremolo switching.

The limitations of the project are related to the small volume of the sound base, an incomplete set of articulations and the lack of an expanded expert assessment with the participation of professional performers.

Thus, the main stages of digital sampling considered by us and the developed working sampler for folk string instruments in NI Kontakt 8 made it possible to create a modern sampler — a controlled system of samples, processing methods, a user-friendly interface and script logic.

- key approaches to the organization of the sampling process have been developed;

- the parameters of digital recording, the requirements for mapping, processing and evaluation of sound quality are defined;

- digital transmission of the timbre of folk instruments is described; the library structure is designed;

- audio samples have been prepared; an instrument has been assembled with Volume and Reverb controls, as well as Pluck/Tremolo switching;

- functional and audio technical verification of the result was carried out.

The specification of folk instruments is described separately, taking into account: timbre recognition, attack, noise components, microdynamics and the balance between authenticity of sound and functionality in a MIDI environment [1; 6].

NI Kontakt is considered as a sampler instrument development environment. Mapping Editor, Group Editor, Wave Editor, Send Effects, and Kontakt Script Processor allow you to go through the entire cycle of creating a custom library: from importing WAV files to managing groups and interface parameters.

In the practical part, a combined sampler of a bass balalaika, a four-stringed domra and a prima balalaika has been designed. The instruments are divided into registers: the bass balalaika occupies a part of the control octave with notes and a large octave, the domra - the small and first octaves, the balalaika prima — the second and third octaves. Pluck and Tremolo are used as implemented sound modes. Recording and preparation of 48 kHz/24 bit WAV samples, slicing, normalization, frequency and dynamic processing, library folder structure, root key and key range settings, group creation, connection of reverberation via a send chain and a KSP script are included. To verify the result, spectral analysis and comparison of raw and processed samples were performed [21; 20; 35].

The conducted testing confirmed that the instrument is loaded into NI Kontakt, it reproduces samples over a set range, switches Pluck/Tremolo groups, adjusts volume and reverb level.

The limitations of the project are

- the small volume of the sound base,

- the lack of additional articulations and

- the need for further expert evaluation with the participation of performers on folk instruments.

Further development of the project is related to expanding the set of samples, recording additional takes, round robin, microphone positions, interface refinement, deeper KSP logic, and library preparation for use in different DAWs.

Such developments help to preserve folk timbres and introduce them into music and computer practice.

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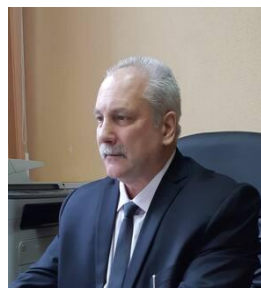
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