# Computer Games Sound Effects
## *Recording, Postproduction and Existing Database*

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Abstract: The paper describes the process of building a new database of sound effects recordings for computer games and the first version of such product. Ways of applying signal processors for postproduction is described, as well as differences in audio edition for films and games. Some aspects of using sounds in games are also mentioned as well as the first version of the list of possible tags of the audio files in the database. Both the language of the tags and the datatabse will be substanially enlarged.

# 1 INTRODUCTION

An important factor at work in the audio creation for games is dynamics of the signal (Marks, 2001; Collins, 2008). It means the difference between the worst and best possible level of the acoustic signal to output or input of the device (Sztekmiler, 2003).

In real cases it is very difficult to record sounds with a high dynamic range. Playback and transmission of acoustic signals have their limitations as well. Some very soft sounds will be inaudible at the playback, and those with a higher level, will be too loud or distorted. Each recording or a reproducing device has a certain level of noise and maximum output level beyond which sound becomes deformed. Because of the dynamics limitations, the acoustic signal devices should be maintained above the noise level and below the threshold value of voltage distortion.

Another important issue in case of games is that in contradiction to audio edition for films, sound effects have to be ready for non-linearity and mixing during the course of the game. The effects can be stopped in a middle or overlaid on each other if a player makes some actions. In case of overlaying, a simple correlation can be not appropriate as typically sound related to events in a game but can be be more important then background music. Some sound effects have to be played in a loop, but in a way that the end point is not hearable.

# 2 SOUND EFFECTS PROCESSING

The dynamics processors were helpful in building the multimedia database. Depending on the level of the input acoustic signal they influence or reinforce the acoustic attenuation of the track in which they are placed. Dynamic processors are used, not only because of the possibility of signal distortions, but also to provide a listening comfort. If the dynamics of the recorded material is reduced, manual changes of the volume playback is not needed. In the opposite case, the quieter parts of the volume could be not noted at all, because the sound would be inaudible.

On the other hand, it could appear that the parties, in which an acoustic signal reaches a high level, are too loud. With these devices, one can manipulate the value of the effective signal, which largely determines the subjective loudness by the recipient. For the higher root mean square (RMS) value of the signal, the audio seems louder, even though the peak remains the same. We used mainly following dynamics processors: a compressor, a limiter, an expander and a gate (Lyons, 2004) while building the database.

## 2.1 Compressor

The compressor reduces the signal dynamics. Typical target of its usage is to raise the RMS of a signal. However, first, the compressor attenuates the signal, which exceeds a predetermined level. In this way, the entire recording can be strengthened without exceeding 0 dBFS peak (in the case of digital recording),

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which would override the sound track. Therefore, the compressor reduces the difference between the loudest and quietest parts of the path. The signals are amplified in a proportion corresponding to the compression ratio. Parameters that can be adjusted in most compressors are (Sztekmiler, 2003):

- *Threshold* is responsible for the moment at which the processor is on or off. Threshold sets the level of the input voltage, above which there is signal loss. The signal below is unchanged.
- The size of gain reduction is determined by the compression *ratio*

$$k = \frac{\Delta U_{in}}{\Delta U_{out}} \tag{1}$$

  where $k$ is a coefficient of compression and $\Delta U$ is the size of input/output voltage above the threshold in dB. This parameter indicates how many times the input voltage level is greater than the voltage at the output, e.g. 2:1, 10:1. This means that a proper amplification beyond the level will be reduced by two or ten.
- The rate of response (*attack*) describes how quickly after the crossing of the size of the input threshold voltage level compressor will start its function.
- *Release* time to the previous settings of earlier amplification, after going back beyond crossing the level set by the parameter threshold.
- *Make-up gain* enhances the output signal. It is used because of the reduction strengthening of the compressor.
- *Knee* is responsible for the part of the transient characteristics of the processor in the activation threshold. Soft knee is used to make compression more natural and less audible, and sharp one for strong compression.
- *Peak-RMS* - compressor reacts to the RMS or peak.

Short *attacks* of the compressor is usually used for transient signals, such as explosions. In case of long ones, the compressor would not manage to pass a fast-changing signal, so its sound would change dramatically. It could be heared as a significant beginning of the transient signal attenuation (the impact moment) and will only sustain itself. However, sometimes it can be done consciously to have this effect. Parameter *Peak / RMS* should be set also depending on the type of a signal. Selecting peak is appropriate for transient signals. Then the processor will respond to the peak value, and not an effective signal. If the sound is continuous and there are no significant changes in the level, RMS should be chosen. The processor then takes into account the effective value, and its effect will be less noticeable and more natural, which is often desirable for this type of signals.

Increasing the level of silent parts will increase the noise level recorded track, so the compressor, as well as other dynamics processors, should be used consciously, knowing precisely the effect one wants to achieve. All processes of denoising audio material should be performed before using the compressor, because it will emphasise the unwanted sounds in the recording. Dynamic processors are helpful tools for editing audio, but it is not reasonable to use them too much because they cause flattening of the sound and artificial sound recordings.

## 2.2 Limiter

The limiter is a dynamics processor, having similar characteristics to the compressor. Its characteristic feature is that the compression ratio tends to infinity, and the attack is as close to zero as possible. They are the hardcoded parameters. Limiter is used primarily to prevent the signal from clipping, reacting mainly to the peak values of the signal. Thus, it does not allow to go beyond the level of 0 dBFS. This means that after applying the limiter, output signal level will not increase regardless of the level of input signal. The processor allows to adjust the following parameters:

- *Threshold* - after crossing the threshold, limiter starts to operate at time close to zero.
- *Release* time to the original gain.

The limiter should mainly protect from distortion of audio, activating relatively rarely. Frequent cutting of the signal by a limiter would have a significant impact on quality, because the nonlinear deformations will appear. What is more, the dynamics will be significantly reduced and the sound recording can be percepted as unnatural.

## 2.3 Expander

An expander works oppositely then the compressor. It reduces the amplification, but of the quietest parts, with a level below the *threshold*. The same parameters can be adjusted, as with the compressor – the *attack* is responsible for the delay of the expander, and the *release* for the late inclusion of gain reduction when signal levels drop below the *threshold*. Too high value of the response time can result in attenuation of high, rapidly rising level, which will be heard as “ragged”. The expander is used primarily to mute noise or unwanted sounds that occur between the useful material.

Noise gate is a version of expander with ratio parameter set to 1: ∞. Then, no input signal, which falls below the level set by parameter threshold, cannot be passed to the output. Thus, the phonic circuit closes. The main goal is to get rid of unwanted sounds, such as glitches, breaks or noises which are heard in the intervals between the wanted sound. In addition to the gate parameters such as threshold, response time (time from the moment of crossing the threshold to gate opening), the recovery time (time from signal drop below the threshold to closing the gate). One can also adjust the *hold* time, which is responsible for the opening of the signal processor after falling below a preset threshold.

This processor is better than the standard denoiser built-in the most of audio editing programs, because it does not degrade the signal, and "cut" only the unnecessary parts. If noise interferes with audio during the useful sound, the gate is useless.

## 2.4 Summarisation

All of these dynamics processors not only attenuate, suppress or do not pass the signal, but also significantly affect its sound. With their help, entirely new sounds can be created, sounding very different from those played back immediately after recording. This is not always a desirable effect, but usually the correctly configured processor improves and enhances the sounds from the recording material.

Dynamic processors can operate not only across the whole band, but also in certain frequency subbands. Band compressors divide the acoustic spectrum into areas in which all the parameters can be set independently. Thus, each band is subjected to a different compression allowing more precise processing of the dynamics.

Equalizer can be also useful to adjust the frequency of acoustic signals. It can change the volume for a specific frequency. It consists of band filters, so that it is possible to amplify or to suppres of certain frequencies. Therefore, the timbre is changed. Equalizer is used, among others in shaping the sound recordings and correction of problematic frequencies.

Editing of the sound effects was performed using a digital audio workstation (DAW) developed by Steinberg - Cubase (Childs, 2007). It has a wide range of tools for recording and editing, mixing and mastering.

Depending on the type of the edited signal, different signal processors were used and different frequency corrections. It was possible to set the dynamics in order to get rid of much of the reverb, which is especially disturbing in impulse sounds. More difficult to edit were the continuous sounds, due to less possibility of interference in the recorded sample. Both graphic equalizers and parametric ones were used to obtain the desired effect.

Short duration paths were modified to allow to loop the sounds for the continuous playback of one track without noticeable discontinuity between the end and the beginning. It was necessary to extract additional fragments, where the sound was quite constant. If there would have been too many changes or pulses that stand out, the soundtrack would be somewhat rhythmic and predictable. Then the loop would have a noticeable effect, which we wanted to avoid.

In the last stage normalisation has been made by finding the samples with the highest peak and setting the level of the track, so that the highest value reached almost 0 dBFS. The signal should not cross it, because then, there would be a digital clipping. This is why a limiter was plugged at the output of the circuit to protect from the distortion.

There is no clear existing standard, which would inform what level of the effective value of the output path should have a sound effect for a computer game. The RMS value depends mainly on the scale of using dynamics procesors during mastering, in particular – the compressor. The smaller the difference in the level of quiet and noisy signals, the greater becomes the effective value of the signal. For music, these values permanently grow (Owsinski, 2006). After 2000, they reached -10 dBFS, and for some songs even -5 dBFS, especially in popular music. For classical music, where the dynamics of the song is a very important element, the effective value is approximately -30 to -20 dBFS. In this case the dynamics is not as flat as in typical entertainment music. We decided that the RMS for our sound effects will oscillate in similar limits as is the case of classical music.

# 3 COLLECTION OF SOUND EFFECTS FOR GAMES

The samples were stored in a collection of 305 sound effects, which are 45 minutes from paths which together lasted 3 h 10 min. Overall, post-production of sound effects for the collection lasted about 70 hours not including the process of recording samples, which lasted several days. One of the major obstacles was tiring of hearing, therefore, quite frequent breaks were necessary. After a two-hour work of listening to the sound effects, the perception is worse, so one can not properly perform the tasks assumed. The collection is just the preeliminary one to establish proper work routines, methods and to allow to work on soundtracing algorithms for games. The collection will be substantially increased in the following two years to reach over 10 hours of sound effects.

The file format is an uncompressed PCM with a sampling rate of 44.1 kHz and a resolution of 16 bits. Files names of the particular sound effects give an additional information about the sound: I - pulse sound, C - continuous sound, L1 - possibility of looping, L2 - possibility of looping without impression of repetitive sound, K - effect allowed for kids, A - effect for adults only, F - effect good for funny games, P - effect for positive games (Zynga type), H - effect for horrors and mysterious games.

## 4 CONCLUSIONS

The process of preparing sound effects for games was described. An existing and growing collection of such audio files have been presented as well. The collection is a result of just started project. The most important aspect of preparing sound database for games besides of quality is their usefulness in non-linear processing, because sound effects have to be mixed a lot and played in very changeable order depending on the decisions of a player.

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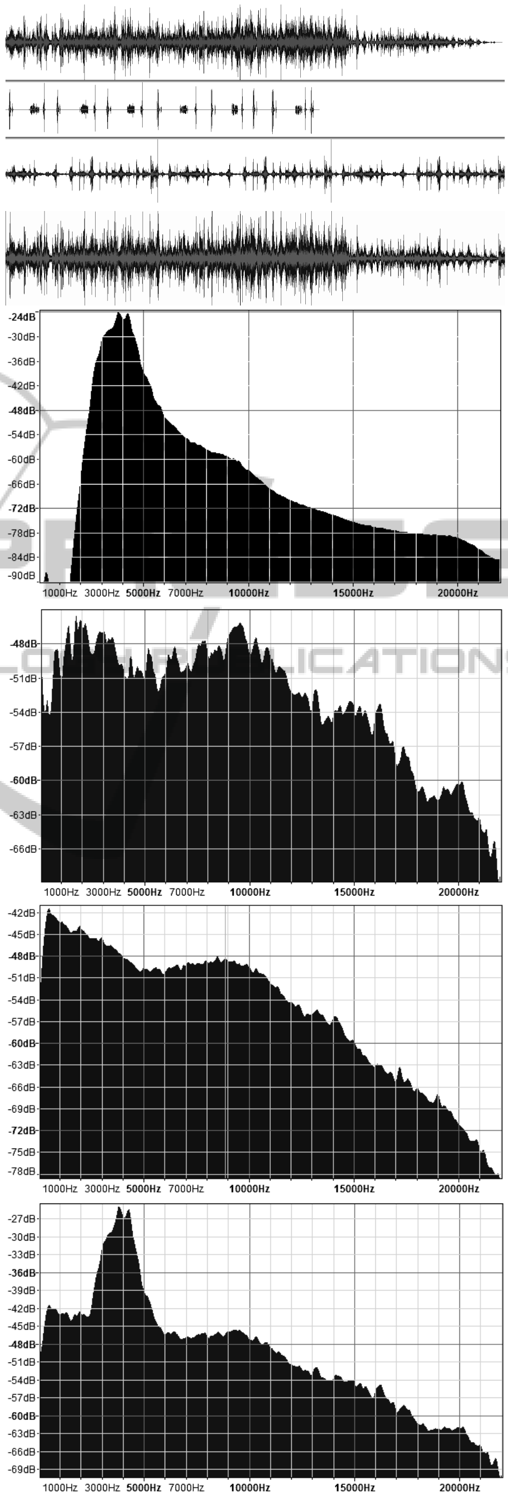


Figure 1: Recordings of three sound effects: birds (in the end intentionally fade out), a camera and someone walking on leaves and their mixture for an event in game of a person walking on leaves, taking photos of birds and going away afterwards.