A psycho acoustic bass response extension plugin - Project plan

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1 Intoroduction

Modern recording techniques and synthesizers allow recorded music and special effects to contain very low frequencies. Most speaker systems can not however reproduce these frequencies, resulting in decreased sound quality. In music, the lowest significant frequencies are usually around 30 to 40 Hz. The low B-string in a 5-string bass for example, has a fundamental frequency of 30.87 Hz, while the lowest frequencies in some pipe organs and grand pianos might be even lower[1].

Due to psycho acoustics, humans can however precieve lower frequencies than the ones produced by audio reproduction systems. This is due to the missing fundamental phenomenon, which states that the human auditory system determines the pitch of a tone according to the overtone series, more or less regardless of the actual lowest frequency precieved.

By enhancing or synthesizing overtones for low frequency components of a signal, the precieved bass content can be enhanced. This method, among others, has been discussed in the Audio Engineering Society Convention Paper 5921[2], which will be used as the basis of this project. The objective is to create a plugin for enhancing bass reproduction using the LV2 audio plugin specification[3].

2 Methods

Processing the signal consists of the following phases:

1. band-pass filter a portion of the signal
2. create harmonic content from the band-pass signal through a non-linear process
3. filter the produced signal to shape the harmonic content and remove unwanted components
4. mix the filtered signal with the original signal
5. possibly high-pass filter the resulting signal and adjust its level to avoid clipping

2.1 First stage band-pass filtering

The objective of this stage is to select the right frequencies to be processed non-linearly. To create the wanted harmonics, the portion of the signal to be processed should be below the reproduction systems low cut-off frequency (f_c). Thus the higher corner frequency of the first filter should be set at f_c. To avoid excessive intermodular distortion, the bandwidth of the filter should not be too big. The bandwidth depends on the signal material and f_c, but should not be wider than a few octaves in general. In any case, the lower cut-off frequency should not be below 16-20Hz.

2.2 Non-linear processing

The objective of this stage is to synthesize harmonics which create a sensation of the missing fundamental. Experiments have shown that for complex tones with a fundamental frequency below 200Hz, the precieved pitch is judged mainly by the difference between higher harmonics, namely the fourth

[1] http://www.contrabass.com/pages/frequency.html (retrieved 17.11.2008)
[2] AES Convention Paper 5921 - A unified approach to low- and high-frequency bandwidth extension
[3] http://lv2plug.in/ - LV2 Audio Plugin Standard

and fifth harmonic[4]. This means that it is crucial to produce both even and odd harmonics in this phase. To keep the overall sound as natural as possible the dynamics of the created harmonics should follow the dynamics of the source signal.

A simple way to achieve the above mentioned objectives is to use an integrate-and-dump algorithm, where dumping is done at every other zero-crossing (e.g. from positive to negative). This creates a pseudo-triangle-wave, which follows the frequency and amplitude of the original signal.

2.3 Filtering of the created harmonics

The harmonics created by non-linear processing need to be filtered for best results. Frequency components below f_c should be removed, as they would not be reproduced anyway. Also, the harmonics created should be filtered with a shaping filter to get best results. In addition to maximised bass perception, the objectives of the filtering are naturalness and unobtrusiveness of the final sound. The filter response will be tailored by listening tests.

2.4 Mixing and final adjustments

The harmonic content created in phases 1 to 3 are mixed with the original signal. Because new frequency components are created in the process, it might be necessary to also adjust the signal amplitude to avoid clipping. Also, any delay introduced in the processing stages should be compensated for. The signal can also be high-pass filtered at f_c if avoiding escess signal power is considered necessary.

3 Workflow

3.1 Prototyping the non-linear element

The phase responses of various filters might be critical to the overall functionality of the plugin. This is why the non-linear element will first be prototyped in Matlab with near-ideal filters in the frequency domain. Also, plotting the output of the non-linear filter with various test signals could prove to be useful. One critical task at this stage will be tuning the amplitude response and dynamics of the integrate-and-dump algortihm.

3.2 Plugin implementation and filter testing

After a basic implementation of the non-linear filter is done, the Matlab code will be ported to C++ and integrated into an LV2 plugin wrapper using the lv2-c++-tools package[5]. The workload of porting should be fairly low.

After the filter is an LV2 plugin, it can be easily tested with various tools and filters available for Linux. The harmonics shaping filter can be experimented on with a parametric or graphical equalizer. Also, the suitability of exesting LV2/LADSPA[6] filter implementations for the first band-pass filter can be tested.

[4] Plomp R. (1967) "Pitch of Complex Tones"
[5] http://ll-plugins.nongnu.org/hacking.html
[6] http://www.ladspa.org/ - Linux Audio Developer's Simple Plugin API (LADSPA), LV2 predecessor

3.3 Final implementation

After good filter types and parameters are found via experimenting, filters and controls will be added to the plugin. If an existing filter gives satisfactory results as the first stage band-pass filter, the code used in it will be copied to the plugin[7], or alternatively new filter code will be written.

3.4 Testing and analyzing

After the plugin is complete, it will be thoroughly tested with various special effects and musical material at different setting. It will also be analyzed, mainly in the frequency domain, to see how it performs in hard numbers. If prominent problems are found, they will be analyzed and possibly fixed.

[7]The plugin will be licensed under the GPL, as are most LV2 plugins.