SOLENT UNIVERSITY SOUTHAMPTON

User adjustable loudness control for sound reproduction using Digital Signal Processing.

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Development of an algorithm allowing accurate adjustment of the "loudness" of a speakers output, altering the frequency content of audio signals in real time by modelling of the behaviours of the human ear.

Project Aims:

To design, implement, test and compare methods of enacting high fidelity, DSP loudness level control for use in commercial research and development.

Project Objectives:

- Design, develop, implement, refine and categorise at least one method of DSP loudness control on an ADI DSP development board, utilising the available loudness compensation tool blocks in SigmaStudio DSP software.
- Investigate potential methodologies for implementing loudness control using a custom algorithm on the ADI SigmaStudio DSP platform.
- Design, develop, implement, refine and categorise loudness control using a custom DSP algorithm.
- Perform subjective listening comparisons of these methods of implementing loudness control.

Background information and context:

- The Equal loudness contours (left) show how the perceived loudness of a sound varies with both its frequency and intensity.
- Each contour shows the intensity required for sine tones at different frequencies to be percieved equally loud at different overall output levels.
- The intensity required to maintain constant loudness between different frequency tones changes as the overall loudness is varied.
- In a speaker system, if a flat volume change is applied, the relative balance of bass, middle and treble frequencies will change, altering the percieved sound quality.
- The aim of a loudness control is to maintain the sound quality which the speaker was designed to achieve, regardless of the selected volume.
- By applying gain which varies with frequency and output level (relative to the volume at which the speaker gives optimum sound quality), it is possible to maintain the percieved balance of frequencies as the sound intensity is varied.
- To give a truly accurate loudness control all aspects of how humans percieve the loudness of sounds must be considered.
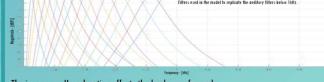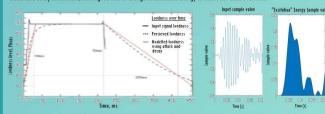
Auditory modelling

It was found that "Auditory modelling" - using algorithms that mimics the the human ear and the complex mechanisms involved in how we percieve sounds, provides the optimum method of implementing loudness control. It allows for a design which adjusts loudness dynamically, in real time, accounting for all aspects of how loudness is percieved.

The outer and middle ear

To correctly calculate how loud the signal will be percieved, it must be adjusted for the attenuation which occurs as sound passes through the outer and middle ear. This defines the minimum intensity required for sounds to be percieved - the hearing threshold. Within the developed model, the signal is reduced by a filter equal to the hearing threshold, made from 4 filters in parallel. This gives attenuation which matches that of the the acoustic channel formed by the outer and middle ear.

The inner ear - the Auditory Filters

Within the Cochlea, the auditory nerves translate sound into signals through the vibration of the "basilar membrane". The basilar membrane vibrates at various points along its length according to the input frequency of the sound. Auditory nerve excitation is then provoked in areas along the basilar membrane corresponding to a range of frequencies. This is often represented as a series of overlapping band-pass filters, the "auditory filters".

Complex tones, such as music, can vibrate multiple points along the basilar membrane. If the points fall along the membrane such so the frequency content exceeds the range of frequencies of a single auditory filter - the "critical bandwidth", the energy within multiple filter bands is summed to determine the loudness of the sound. Within the model, this effect is replicated for frequencies below 1kHz, by applying a bank of similar band-pass filters, and calculating the loudness of the signal in each of the filter bands.
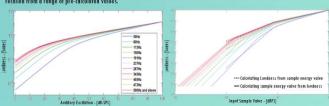
The inner ear - How duration effects the loudness of sound

The perceived loudness of sound is not only dependent on the energy within each of the filter bands, but the duration that the energy is present. Percieved loudness increases gradually over time the longer a sound remains until the peak loudness value of the signal is reached. This effect is referred to as "temporal integration". Within the model, this property of loudness is replicated using an attack and decay mechanism, limiting the rate of change of calculated energy in each filter band over time.

Calculating and controlling percieved loudness from the energy of the audio signal

From the energy in each filter band, the loudness which would be perceived by each area of auditory nerves may be calculated. Loudness is calculated in Sones, a scalable unit of loudness which changes with frequency: a loudness of 1 Sone at one frequency, is equal to 1 Sone at another frequency, and both are half as loud as 2 Sones. In the developed design, the loudness of the signal is recalled from a range of pre-calculated values.
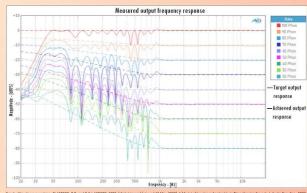
Using the calculated loudness value, the user is allowed to directly vary the signal loudness with a typical volume knob. The input value is used to scale the calculated signal loudness, giving a target loudness value (for each band). By converting the target loudness back to an energy value, the gain required to adjust the signal loudness may be calculated from the ratio of the target and actual energy values. After applying the calculated gain, the band filtered signals are then summed together to synthesize an output signal.

Block Diagram of the algorithm

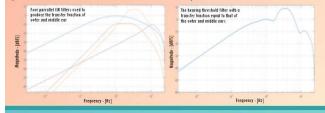## Project Outcome and Conclusions

- Measurements show the algorithm is giving loudness control roughly proportional to the equal loudness contours.
- Significant dips in output level occur at some frequencies: The phase of the different filter bands is causing the filtered signals to interfere when summed back together. Alternternative filter designs fixing this issue were simulated, but would require a more powerful DSP platform to implement.
- Comparing the sound quality of the developed algorithm and a pre-made loudness control algorithm shows that auditory modelling offers better performance than traditional loudness compensation methods.
- If the phase issues are removed, the algorithm could be used to make speaker systems sound better at lower volumes.
- Several compromises had to be made to the design in order to allow it to run on the DSP platform.
- Ultimately auditory modelling is too processing intensive for the chosen DSP platform (which is typical of the solutions used in most wireless speaker designs). This type of algorithm would therefore likely only be suited to high-end speaker designs, which could utilise high powered DSP platforms.

Graph of loudness over time: GLASBERG, B.R. and B.C.J. MOORE, 2002. [digital image] [viewed 13 May 2020] A Model of Loudness Applicable to Time-Varying Sounds. J. Audio Eng. Soc., 50(3), 25
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Output speaker: Bowers&Wilkins, n.d. Bowers&Wilkins Formation Flex. image [viewed 14 April 2020]. Available from: https://www.bowerswilkins.com/en-gb/formation/flex