AudioFix - ANALYSIS AND CORRECTION OF ACOUSTIC RESPONSE WITH DIGITAL SIGNAL PROCESSING (Acoustics, Diffusion, Sonorization)

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Abstract

This work explores a practical way of using digital signal processing to adjust the equalization curve of an audio signal, correcting the distortions produced by the acoustic response of the environment. Equipment such as signal generators, spectrum analyzers and equalizers can be used to correct these distortions, but its correct utilization requires technical knowledge not always available to the audio equipment operators.

1. Introduction

The *audio* term represents collectively all the areas where the sound technology is applied. Although it has a lot of facets, that technology basically involves capture, handling, recording and reproduction devices, such as microphones, amplifiers and speakers. The essence of all the evolution in that area is the search for fidelity of the reproduced sound in relation to the original.

The physical features of the material used in the construction of microphones and speakers can affect their fidelity in relation to sounds of different frequencies. Thus, all device presents its own characteristic response curve. The ideal transducer, capable to uniformly answer all the frequencies of the audible spectrum, practically doesn't exist; to reproduce sounds with high degree of fidelity it is used a system composed by several speakers, each one with a response curve that matches a certain band of frequencies, combined in order to reproduce the widest possible band of the audible spectrum [PHI].

The measure unit of audio level is the *decibel* (dB), the tenth part of one *bel*, that indicates the change of one order of magnitude in the power of a signal; it means to multiply or divide for ten the amount of present energy. A gain of +20dB increases 100 times the power and a gain of -20dB divides it for 100. But the amplitude of a signal is proportional to the squared root of the power, so that the same ones ±20dB mean to multiply or to divide the amplitude of the signal for ten[SMI99].

2. Presentation of the problem

The sound propagation in an environment is affected by its characteristics, such as resonant frequencies, reflection, diffraction, refraction and absorption[CYS92]. The resonant frequency of an object is the frequency in which it tends to vibrate, or vibrates with easiness, to a small received stimulation.

Each element in a certain environment, as walls, floor, ceiling, furniture and air, tends to intensify the sounds that match with its own resonant frequency, modifying the perceived intensity of sounds of different frequencies. But the features of an object can impede it to vibrate in any frequency, resulting in the absorption or attenuation of the sounds that reach them, or they can simply modify the direction of the propagation of the sounds.

To the complex combination of those effects on the sound propagation, we attribute the name of *acoustic response of an environment*, that can be seen in a graph that shows the perceived intensity of the propagated sound while we modify its frequency, staying unchanged the intensity of the generating source. In extreme cases, especially great concert halls with parallel walls and little acoustic absorption, this can greatly make it difficult to understand what is spoken, independently of the quality or power of the audio equipment.

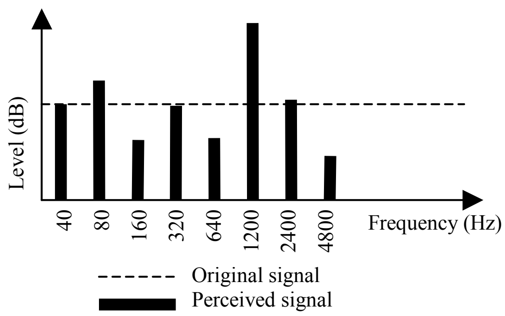

Figure 1 - Example of acoustic response

3. Current solutions

The traditional solutions for such acoustic response problems consist basically of audio generators, spectrum analyzers and equalizers. The equalizers are popular among the professionals that deal with audio equipment; audio generators with limited features are used by some maintenance workshops and the spectrum analyzers are almost rare, practically restricted to the great audio equipment manufacturers.

The basic technique consists of using a generator that produces a signal composed of all the frequencies of the audible spectrum, with uniform energy to each octave, called *pink noise*. This signal is applied to the environment, captured by a microphone and its spectrum analyzed, tracing this way the response curve of the environment. Finally, the values of that curve are used to define the equalization parameters that will be applied to the equipment.

4. Solution proposal

The proposal of *AudioFix* consists of analyzing the original composed signal that is being applied to the environment, getting its spectrum (main signal), and simultaneously, to analyze the signal captured in the same environment by a high fidelity microphone (return signal). After the compensation of the delay and the attenuation of the signal due to the distance between the system of speakers and the microphone, the two gotten spectra are compared and the observed deviations are inversely applied to a digital equalizer that acts on the original composed signal before the amplification.

Beyond allowing its use in real time, this approach is capable to also correct some distortions generated by the audio equipment itself in the audio signal spectrum.

5. Implementation

5.1. Logical Model

The digital handling of the signals was based on the *Fast Fourier Transform* (FFT) [SOR87], that makes the analysis of the input signals and the synthesis of the output signals. The analysis consists of the transpose of the signal in the time domain to the frequency domain and the opposite corresponds to the synthesis. The theoretical basis of FFT is the formula " sin (a) * sin(B) = ½ cos(A-B) - ½ cos(A+B) ", according to which a complex signal can be combined with a sine wave of known frequency and amplitude in way to indicate the participation of this sine wave in that signal. FFT can treat any digital signal; even discontinuous slopes are synthesized as the combination of multiple harmonic sine waves in decreasing levels.

The diagram below shows the logical model of *AudioFix*; the lines traced in clear color represent the functioning in "Normal/Stereo" mode, in which the system behaves as a common equalizer, without automatic fittings.

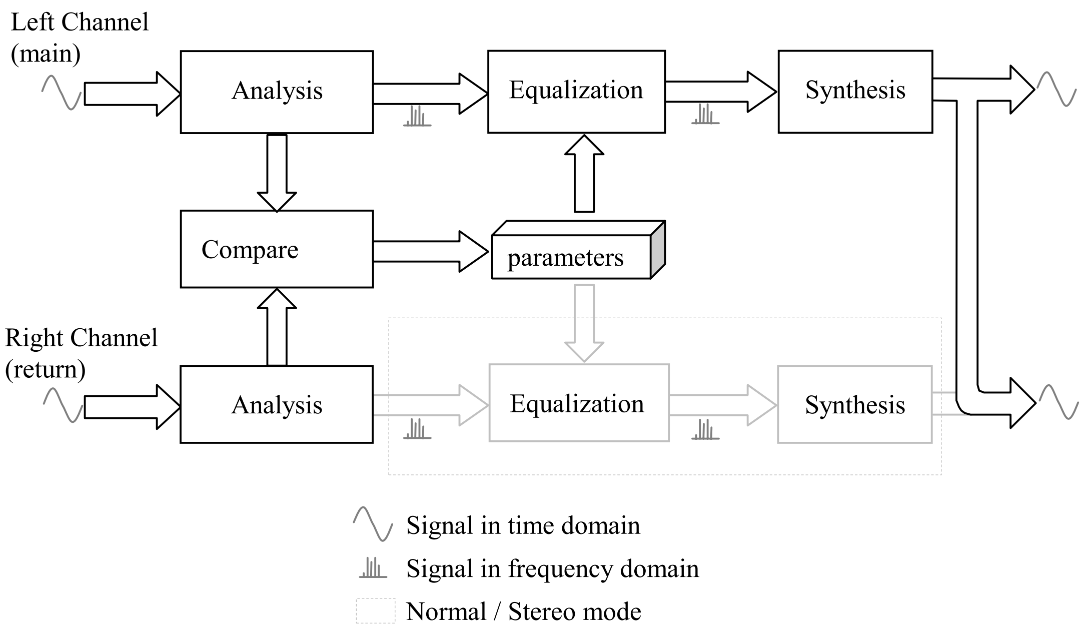

Figure 2 – Logical Model

5.2. Architecture

Implemented for Windows-95 with Borland C++ Builder, *AudioFix* has no controls of input and output level and its operation requires the use of the audio controls of the multimedia system (record/play).

The sound interface needs to be full-duplex operation capable, for simultaneous reproduction and recording; it is desirable it supports the use of professional microphones and allows to the separation of line and microphone signals in distinct channels, but that function can be supplied by an external audio equipment.

The input and output routines use the low-level API's (*Application Program Interface*) for audio signals handling. They define reading and writing buffers, open the desired devices supplying them with those buffers properly initialized and the handling routines are called by the system at each processed buffer [SIM96]. In case of the recording device (*wave in*), from which the signals are read, the handling is the analysis and storage of the data block; in case of the reproduction device (*wave out*), the handling is the synthesis of a new data block so that it can be reproduced.

To each read block, the two channels are separated in two vectors and passed as input arguments to the FFT routine in successive calls and the results are obtained in the same vectors; the average amplitude of each signal and the ratio between both are calculated, to be used to compensate the level difference between the two signals. Finally, each sample of the main signal is divided by its corresponding in the return signal already compensated, thus determining an adjustment factor stored in a parallel vector. Low amplitude samples are discarded, because they are sensitive to noises that would produce wrong adjustment factors.

For the synthesis of data blocks, that will be placed into the buffers for the output audio device (wave out), each sample of the main signal represented in the frequency domain is multiplied by the adjustment factor previously calculated. The last stage consists of applying the inverse FFT routine, passing it as argument the signal in the frequency domain with its samples already adjusted and receiving in return the signal in the time domain.

5.3. Interface

The interface of ***AudioFix*** consists of a window divided in four areas. The first one overlaps two graphs, corresponding to the main (blue) and the return (red) signals, that monitor the levels of the input signals, so that they can be are adjusted through the controls of the multimedia system available in the execution environment.

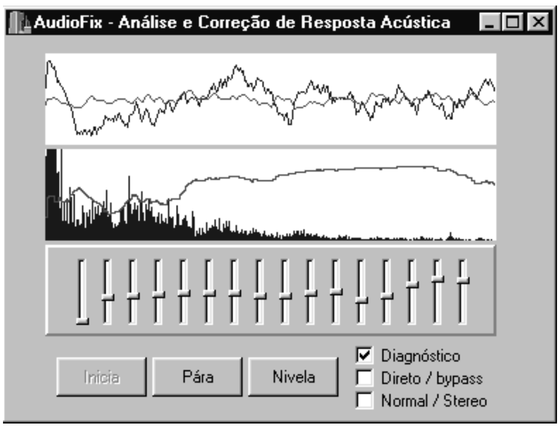

Figure 3 - Interface of *AudioFix*

In the second area, it is exhibited a histogram corresponding to the main signal spectrum (blue), obtained through analysis in narrow band, together with a line graph (green) corresponding to the equalization curve, on which we can act crawling the left button of the mouse. That area is traced in linear scale, what can cause some confusion when comparing with the controls of the equalizer in logarithmic scale, but it is interesting for allowing precise performance on resonant frequencies detected in the environment as feedback; it attenuates the problematic frequency practically without quality losses.

The third area contains the gain controls of the equalizer bands. The gain variation is of approximately ±10 dB and the number of bands can be configured. The legends of the sliding controls, indicative of the central frequencies, were substituted by hints, that are shown stopping the mouse cursor on the control.

The other controls are three buttons and three check-boxes. The button *Inicia* places the program in operation, the button *Pára* interrupts the execution and the button *Nivela* restarts the correction factors, resulting in a flat equalization curve. The *Disgnóstico* option uses the left channel as main and the right channel as return, and automatically generates the correction factors of the equalization curve; in this mode, the same signal is applied to the two output channels. The *Direto/bypass* option forces the copy of the main signal directly to the output of the sound interface, discarding the corrected signal. And the *Normal/Stereo* option enables the stereophonic mode, that deals with the two channels of the sound interface as main signal.

A pop-up menu, activated with the right button of mouse on a free area of the window, offers to the options *Configuração*, *Testes* and *Sobre*... The first one allows to define the input and output devices, the microphone to speakers distance, the number of delay blocks between input and output, the update interval of the graphs and the number of equalizer bands.

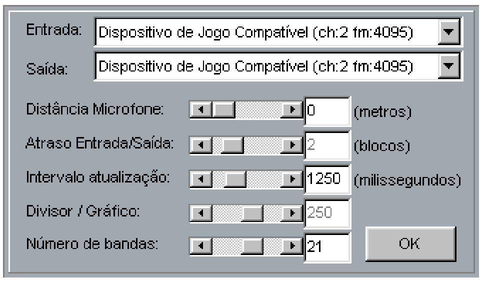

Figure 4 – Configuration frame

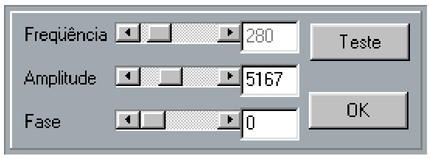

Figure 5 – Tests frame

The *Testes* option executes a simulation of sine waves handling by the program and the option *Sobre*... shows the traditional credits frame, with identification of project, author and institution.

5.4. Examples of use

A possible use of *AudioFix* consists of placing the microcomputer between an audio source and the amplifier (main signal), having a microphone in the reproduction environment to catch the resulting sound (return signal).

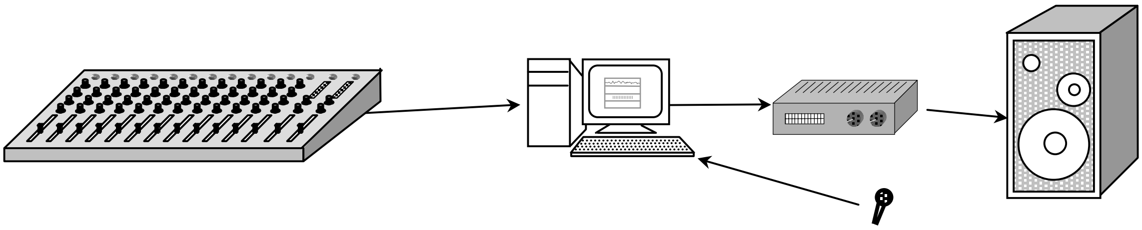

Figure 6 – Application without equalizer

We can also use a graphical equalizer between the source and the amplifier, discarding the output signal of the sound interface. Thus, the equalization curve exhibited by the program is only one referential indicating the correction levels to apply on the equalizer bands; corrected the equalization, the program will indicate a plane response curve.

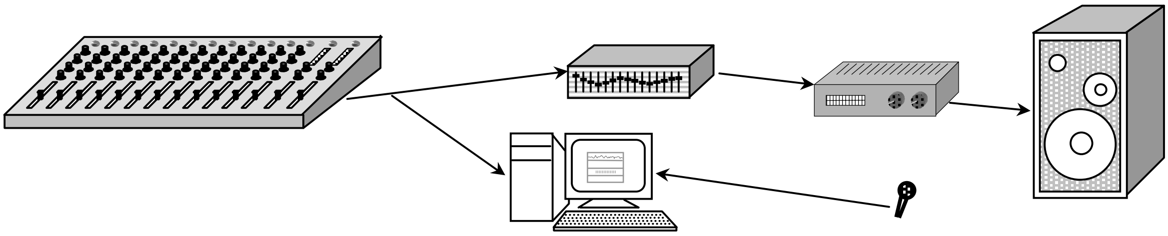

Figure 7 – Application with equalizer

6. Results

The adopted approach provokes the distortion of the signals of frequencies with incomplete cycles in the data block, because this case behaves in FFT as discontinuous slopes; thus, when deeply acting on a specific frequency it also affects the levels of other signals whose synthesis uses it, resulting in a modification of the waveform in the extremities of the data block. But this undesirable effect only appears in the ends of the level controls of the equalizer bands, not getting to invalidate the results.

In the accomplished demonstrations, the efficiency of the program could be shown when artificially modifying the answer of the microphone, acting on the tonal controls associated to it, simulating a change in the acoustic response of the environment. Reducing the level of sharp sounds of the microphone, *AudioFix* automatically increased the level of the affected frequencies, being clear the audible perception of the resulting effect.

Another demonstration, shown in figure 8, was established changing the speaker system staying unchanged the environment. The first equalization curve was generated by *AudioFix* when using a low-cost speaker system, with poor bass response; the second one was generated using a quality speaker system, with good bass response. Although using a low quality microphone, this comparison clearly shows the difference between those two speaker systems. In both cases, the extreme low band was affected by a constant low frequency noise in the environment (a noisy air conditioner) without corresponding low frequencies in the signal.

Figure 8 –Equalization curves for different speaker systems

The program only works with 44100 samples/second and 16 bits/sample. In a machine endowed with a 333 MHz Pentium-II processor, the *System Monitor* indicates more than 70% of processor utilization, having even some truncations in the audio signal stream. But the use of more powerful machines became already very popular, not having this restriction to mean a great limitation.

7. Conclusion

Although it is subjected to the distortions of the output signal, as described above, the use of the program revealed valid, especially because the intention is not to substitute a professional graphical equalizer, but to assist in the definition of the parameters for its efficient use.

REFERENCES:

[SMI99] **SMITH**, Steven W. *The Scientist and Engineer's Guide to Digital Signal Processing*. 2nd Edition. San Diego CA: California Technical Publishing, 1999.

[CYS92] **CYSNE**, Luiz Fernando O., *Áudio – engenharia e sistemas*. Rio de Janeiro : H. Sheldon, 1992

[PHI] **PHILIPS** Electronics N.V. *Sound Book – guia de áudio*. Philips Electronics N.V.

[SIM96] **SIMON**, Richard J., *Windows 95 Multimedia & ODBC API Bible*. Corte Madera, CA : Waite Group Press, 1996

[SOR87] **SORENSEN**, Henrik V., Jones, Douglas L., Heideman, Michael T. e Burrus, C. Sidney. *IEEE Transactions on Acoustics, Speech, and Signal Processing*, Vol. ASSP-35, June 1987.