Design and implementation of a software DSP system to correct digital audio reproduction artifacts.

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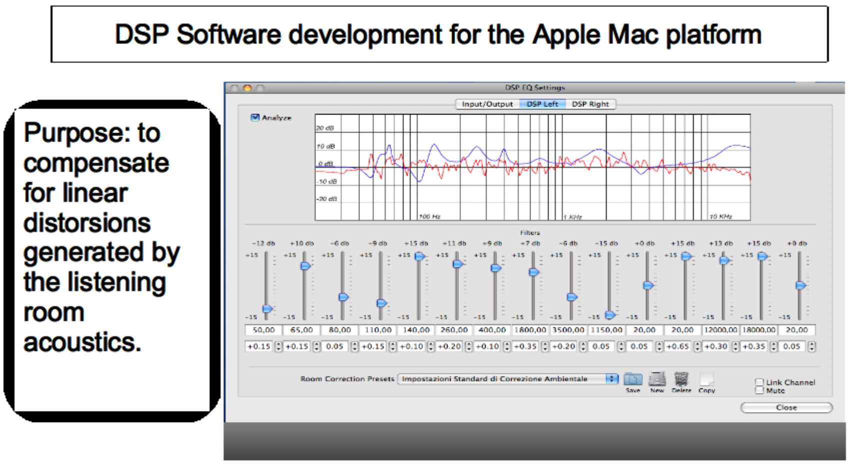
Abstract: In this work the listening room's acoustic distorsion phenomenon (frequency response nonlinearity) has been studied and attacked in order to improve the perceived sound subjective quality characteristics. Two different techniques of Spectral Analysis have been tested and evaluated in order to assess the frequency response of different environments. Within a collaboration between Istituto di Scienza e Tecnologie dell'Informazione (ISTI-CNR) and TangerineTech Engineering, a Room Correction procedure has been perfected making use of digital parametric high-resolution IIR filters. The result of this research is a Software DSP which can process digital audio and compensate the filtering behavior of listening rooms. The whole application has been released for the Apple Macintosh platform.

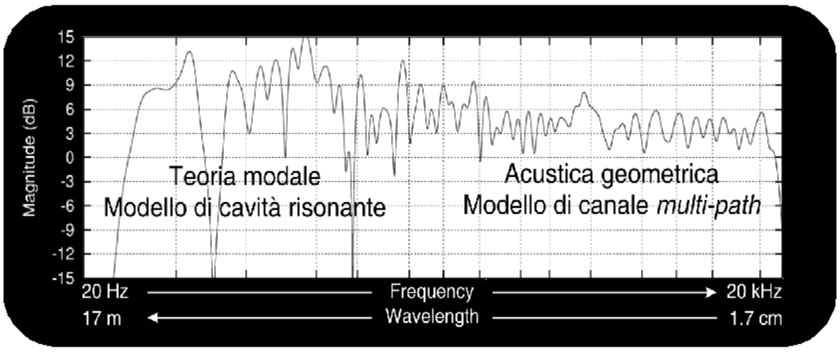
Keywords: room correction, listening test, computer audio

1. Introduction



In this paper we will introduce a software application allowing to measure the environmental frequency response and to correct its magnitude in order to minimize audible artifacts in an audio domestic system. This software has been developed for the Apple Macintosh platform and it allows to compensate for the linear distortions which the listening room superimposes to the reproduced sound. These distortions alter the timbre characteristics of sound in a heavy way.

2. Listening rooms behavior



The above figure conveys an idea about how much listening rooms acoustics interfere with the audio systems characteristics.

In the lower spectrum region, standing wave phenomena prevail, or resonant cavity effect. Rooms behave like tuned resonators which give origin to amplitude responses characterized by peaks and valleys.

In the higher region spectrum, geometric acoustics, or reflexion phenomena, are prevailing and render the amplitude response extremely variable.

In the center zone of the audio spectrum, both behaviors coexist.

3. The 'Theoretical' approach

Theoretical approach to room correction

- **FIR Digital Room Correction**

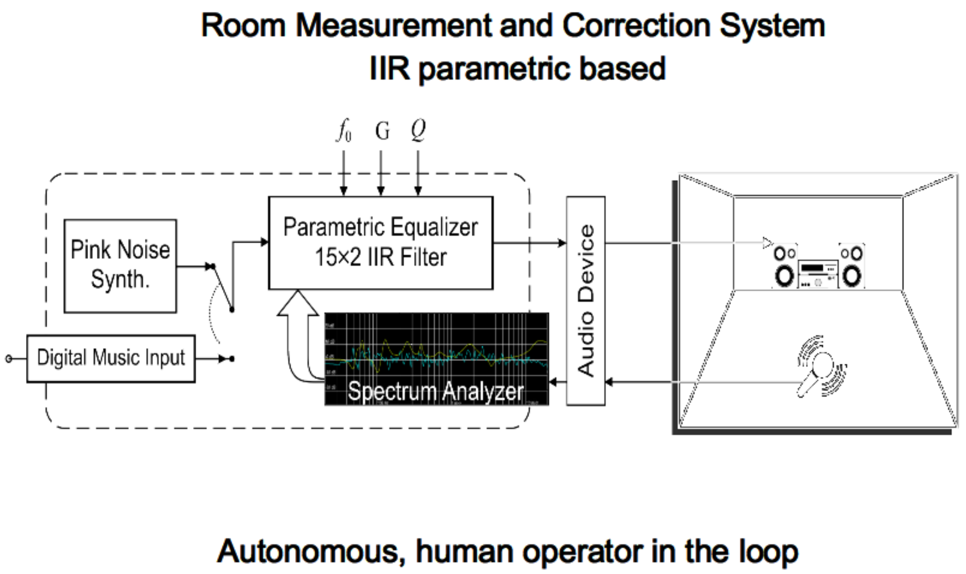
-Through FFT of the impulse response series the room frequency response is obtained

-This information is used to calculate the coefficients of an inverse FIR filter

-The FIR filter is employed to 'deconvolute' the environmental amplitude and phase response

Even though room correction technology is used since analog equalizers, these last years have seen to take off a more theoretical approach to this question. Through FFT of the impulse response series the room frequency response is obtained, then this information is used to calculate the coefficients of an inverse FIR filter, employed to 'deconvolute' the environmental amplitude and phase response. Problems linked to this approach derive essentially from the use of impulsive signals, which can induce nonlinear anomalies in the subsequent chain, and from the heavy computational load which the processing system has to bear with when working at high sampling frequency. Moreover phase correction has not been demonstrated audible in blind tests. [1]

4. System Structure



The system here presentes uses a full-parametric digital IIR equalizer for room correction and integrates a digital measurement noise generator and Constant-Q spectrum analyzer. The measurement noise (white or pink) is first processed by the EQ stage, then sent to digital or analogue outputs in order to be reproduced by the audio system. The environmental response is acquired by a measurement microphone connected to one of the system inputs and processed by the spectrum analyzer in real time.

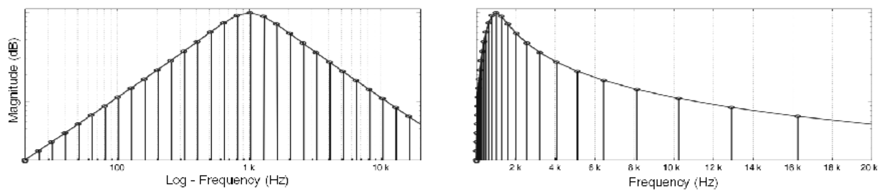
Room Measurement and Correction System
IIR parametric based

- ✓ Spectrum analyzer eases operation
- ✓ Pink noise is wideband but has low amplitude fluctuations
- ✓ Scalable system: if more power is needed, we can add more filters, maintaining low latency and high stream speed.

Parametric filters' control is made easy by the spectrum analyzer which gives a feedback about the system's transfer function correction process. Pink noise is a wideband noise, but maintains roughly a static amplitude, minimizing the risks of nonlinear behavior by the audio system, and in the end the system is very scalable, low latency and optimally functional even when processing high speed audio data streams (32 bits float / 192 kHz).

5. Signal Analyzer

Frequency analysis



Log and linear frequency domains

The amplitude spectrum of the room response acquired by the measurement microphone is represented on a diagram which bears frequency (Hz) on the x axis and amplitude (dB) on the y axis. We'll neglect the phase information because *in normal conditions the human ear has not been proven sensitive to it*. [1] When working into the audio band usually the x axis is logarithmic while the y axis is linear. The image is displayed on the computer screen and refreshed every 300 mS.

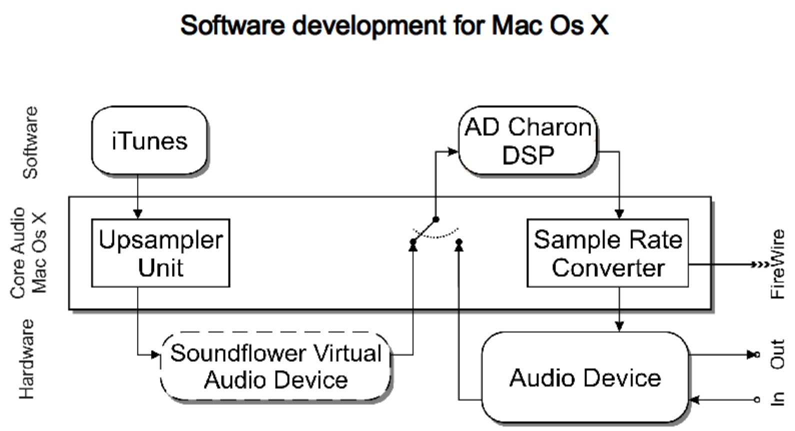
Innovation: Constant Q analysis

	FFT	Constant Q
Frequenze	$k \cdot \Delta f$ lineari in *k*	$(2^{1/24})^k \cdot f_{min}$ esponenziali in *k*
# Campioni	costante = *N*	variabile = $N_k = \frac{f_{SR}}{f_k} \cdot Q$
Risoluzione Δf	costante = f_{SR}/N	variabile = f_k/Q
$f_k/\Delta f$	variabile = *k*	costante = *Q*

Constant Q analysis allows for a simple solution

The FFT transform spectrum analysis usually employed in these cases isn't quite apt to this task. In particular, we have too little analysis points in low frequency compared to the optimal mapping we obtain in high frequency. A solution to this problem has been proposed by Brown in 1991 and is known as Constant-Q Transform. Its implementation as source code was not public before the present work, so it has been necessary by our research group to implement it fully starting from theory and definitions. In our case we have demonstrated its equivalence with a 1/24 oct filter bank, coupled with high flexibility and computational efficiency. The Constant-Q Transform Algorithm so obtained has now been open sourced.[2]

6. Software development



The above image is a working diagram of the whole software process. Multimedia files are read by a software (ie. iTunes) and if necessary upsampled to the desired sample rate. By using the Soundflower open source virtual audio device the signal is then fed to the DSP unit which performs the room correction function. After processing the data stream is sample rate adapted to the receiving D/A Converter (either internal or Firewire / USB external) and sent to it. When measuring a room response, the DSP unit is fed by the digital noise generator and a microphone input is taken to the Constant-Q signal analyzer for visualization.

Software development for Mac Os X

-COREAUDIO: a reliable 'audio engine' which operates *real-time.*

-Up to 32 bits/192 kHz even on less-capable hardware.

-Libraries used: Carbon *framework*, PortAudio A.P.I.

-Developed on: Xcode, Interface Builder.

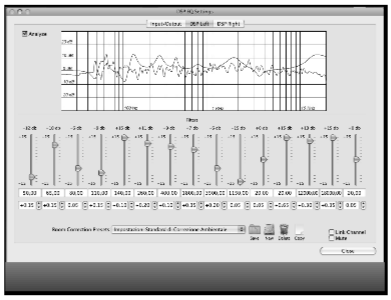
The software has been developed for Mac OS X taking advantage of its COREAUDIO technology which is an 'audio engine' dedicated to real time processing.

The sound reproduction is separated from its management, allowing a fast processing of digital streams up to 32 bits / 192 kHz even with limited hardware resources.

The libraries employed in the program building have been Carbon and PortAudio and the programming kit used has been Xcode + Interface Builder.

User interface

- Main Window
  - Preset
- Settings Window
  - Preset Save
- DSP Window
  - Spectrum analyzer
  - Parametric DSP EQ



The software user interface is divided into three main parts: the Main Window, which allows the preset choice (inputs / outputs selections, EQ selections), the Settings Window, allowing the inputs / outputs presets creation and the DSP Windows, allowing to create presets for the full-parametric room correction unit using the built-in Constant-Q signal analyzer.

7. Conclusions

Conclusions

Computer as a multimedia tool:

Pros:

-Ease, networkable

-Archival, track management

***Perceived* cons (largely NOT real):**

-Associated to 'consumer' music (MP3, compact stereo, etc.)

-More complex to the end-user

The advent of affordable Personal Computers has allowed audio information to be treated as any other data type. It is transmitted, stored, shared and networked as it has never been before.

The other side of the question is the progressive deterioration of perceived audio quality following technical and commercial choices by the main players in the market, choices which in some cases have been dictated by the requirements of the new popular media consumption ways (cellphone music, portable players, etc).

Audio and music professionals have often used DSP systems in order to restore to its maximum the audio signal quality, and it's only logical to employ the newfound sheer processing power of modern PC CPUs to perform this important task.

Conclusions

Development of a software for audio DSP room correction

-Allows to correct loudspeaker response

-Support for hi-quality external DACs

The results of this work have been released as a Room Correction Software which makes use of the technologies described before and can be interfaced to external high quality audio peripherals [4] in order to achieve the best sonic performances available in the domestic digital audio domain today.

References

[1] Toole, Floyd E., "The Acoustics and Psychoacoustics of Loudspeakers and Rooms - The Stereo Past and the Multichannel Future," 109th AES Conv., Los Angeles, Sept 2000.

[2] http://www.tangerinetech.net/downloads/constant_Q.pdf

[3] PortAudio website: http://www.portaudio.com

[4] Jakob Ludwig Firewire DAC1: http://www.jakobludwig.com

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