Room Response Analysis
Applications Note

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Abstract
Many modern consumer audio entertainment systems are able to superimpose the acoustic response of a concert hall on top of a piece of music that may have been recorded in the bland environment of a recording studio. In order that this acoustic response can be applied to the signal the equipment manufacturer must first of all analyse the response of the concert hall.

Acoustic Room Response Analysis
Any acoustic environment is a complex system to analyse, the walls, ceilings, floors and even bodies effect the sound as it reverberates around the enclosed space (fig. 1). These multi-path effects give a body to the sound that is often unique to that particular place. In addition to straight forward echoes and reverberations from solid objects, which effect the temporal nature of the signal, the air through which the sounds pass will often be warm and humid which will effect the frequency spectrum of the same signal.

Modern floating-point programmable Digital Signal Processors (DSPs) are ideal instruments with which to analyse these audio effects. A sample can be taken of the acoustic environment and this can be analysed in the time and frequency domains to extract the coefficients of a filter that can be applied to the signal to replicate the effect of the environment being analysed.

There are several types of signal that can be used for the analysis of an acoustic system but the signal must fulfil several primary requirements. The first concern is that the signal contains all the frequencies of interest (20 Hz to 20 KHz for high quality audio). There is also an ambiguity in the requirements of the pulse duration. A short pulse will give very accurate requirements but will fade quickly in a large environment, conversely a long pulse will not give such good definition but will be easily detected even after reflections off several objects.

The three most popular signals for this kind of analysis are : a Maximum Length Sequence (MLS), an impulse and a chirp signal, all meet the frequency content requirements. For analysing a large concert hall however the impulse signal was unable to meet the previously described energy requirements for it to be easily detected. The MLS sequence was also disregarded because it is generated from a pseudo random number generator and as such is a very non-linear signal, which could certainly damage the PA system, especially at large amplifications. The choice therefore fell to the chirp signal, which contains all the frequencies required, is a linear signal so is less likely to damage the equipment and it also contains a large amount of energy within each pulse.

The chirp signal, which consisted of 512 samples from 0 Hz to 20 KHz, was calculated during the DSP's initialisation process and placed in a buffer in DSP memory. The chirp signal was then output to the DAC, synchronously with the ADCs sampling the input from the microphone.

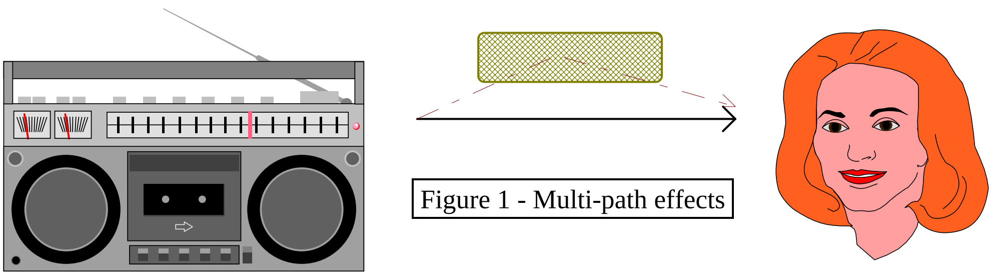


Figure 1 - Multi-path effects

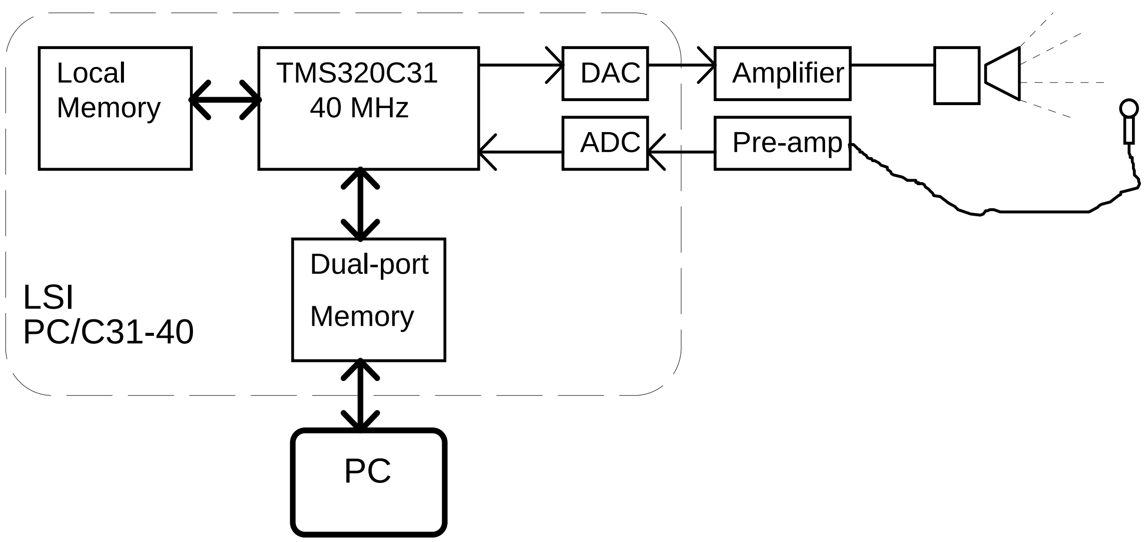
Once stored the results were analysed by three processes under the control of the host man-machine-interface, the three processes were : time domain, frequency domain and cross-correlation. The time domain function merely copied the received data to the DSP card dual-port memory, the frequency domain process windowed the data, performed an FFT and log magnitude calculation. The cross-correlation used the transmitted and received data to generate a precise display of the reverberation effects of the environment. After the calculations had been performed the results were placed in dual-port memory on the DSP card, from where they could be displayed or written to disk.

The DSP hardware (fig. 2) consisted of a Loughborough Sound Images PC/C31-40, this is a 40 MHz Texas Instruments TMS320C31 based PC plug-in card with 640KWords of local memory. The board also incorporates analogue interfaces, which can handle up to four line level inputs and outputs on a single slot PC card. The microphones were connected via pre-amplifiers to the analog inputs and the outputs were connected to the PA system. The initial system developed was a monophonic system but the LSI card can incorporate 2 synchronous stereo inputs and outputs. Tests showed however that the use of multiple directional microphone inputs was essential for the recreation of a true 3D representation of the sound space.

The results clearly showed the multi-path effects on the audio signals, especially when the cross-correlation was performed. The spectrum analysis clearly showed the effects on the high frequencies as they passed through the air and also showed up any weaknesses in the PA system used. It was also interesting to note that significantly different results were obtained by using different amplitudes of signals, this was due to the non-linearities of the transmission medium (air). For this reason the best results were achieved when the environment was analysed with the loudness level set to that which would normally be experienced at the concert.

At the present time most systems apply the required response to the signal but in the future it will be possible to first remove the response of the room in which the music is being listened to by using the same techniques. At this time it will be truly possible to sit at home close your eyes and imagine that you are in your favourite concert hall , anywhere in the world.

Figure 2 - System hardware



The software was written on a PC, in a couple of days, using the SigLib library. The DSP code was compiled for the TMS320C31, while a host program was also written for the PC, to allow it to control the DSP functionality.