

# **ACQUISITION AND REPRODUCTION OF FACTORY NOISE WITH LOW-FREQUENCY EXTENDED BANDWIDTH: A CASE STUDY**

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## **Abstract**

A case study of the problem of collection of noise sound signals in a typical factory shop floor, down from 2 Hz up to 20 kHz, and subsequent reproduction in a biologically controlled small reverberant closed-room is presented. The purpose is to allow an *in-vivo* investigation of the biological effects of acoustical noise radiation. The project of the electro-acoustic set-up for recording and reproduction is described, with discussion of the encountered solutions. The resulting system is based on a PC, with a DT2823 data acquisition and a SB Live 5.1 cards, one B&K 4165 microphone with preamplifier, one 2-channel power amplifier, 16 monitor-type and 1 sub-woofer loudspeakers. The operation software was designed under the LabVIEW system. Sound signals processing was done offline, partly in LabVIEW and partly in Matlab. The system is capable of recording and reproducing noise sounds with monitoring of saturation level in the amplitude dynamic range. Signal acquisition and processing methodologies are presented and explained. Total signals' duration is 1 hour. Frequency and amplitude characterisation of signals was done. Reproduction of sounds at the original levels with spectrum very near the original is the following topic addressed in the paper with a description of the problems of equalisation and distribution of sound output in the room. The system has been in operation for 6 months in a laboratory of the Instituto de Ciências Biomédicas Abel Salazar, of the University of Porto.

## INTRODUCTION

Investigation of the effects of noise over living organisms is the area where the present engineering work is inscribed. The objective was stated in a very simple way: to reproduce in a biological laboratory environment, as accurately and economically as possible, the acoustical events that exist in an industrial shop floor. This reproduction should be automatic, giving a sound output with a regular daily work-life rhythm, that is 8 hours per day, 5 days per week operation. A top-level reliability was requested together with a traceable operation. The target frequency range was defined as 20 Hz to 20 kHz, in order to include SS (sub-sonic) vibration as well.

The ambient noise during labour hours in a typical industry (textile) that was selected is a quasi-random signal with no apparent periodicity, due to the large number of machines that act as noise sources. The short-term and long-term average sound powers were confirmed to be approximately constant. The average SPL values, the frequency contents and the average form and peak factors remained to be checked as well as the signal's amplitude probability density function (pdf).

Admitting that a gaussian type of pdf applies then peak instantaneous amplitudes about 3 or more times the average rms value should be encountered. This corresponds to 9.5 or more dB above the short-term average SPL. Considering a value of 95 dB maximum average SPL, a value of 104.5 dB peak instantaneous value should be allowed in the signal path without saturation. Due to the nature of the signal, a reference value of 60 dB for the signal path signal-to-noise ratio (SNR) was considered adequate. The final recording gain adjustment was adjusted for a  $3\sigma$  peak amplitude allowance, giving a probable saturation in 0,3% of recording time.

The acoustic field nature in this case is basically diffuse due to the diversity, number and positions of sources and to the reverberant enclosure of the factory hall. The sound pick-up should be done in an omni-directional way. The place of pick-up should be near the places where workers stay during labour hours.

Duration of signal recording was selected according to statistical relevance when compared to the temporal pattern of the average level, that is to say not less than a probable period of non-stationarity of the signal. This period is not larger than 1 day and is probably less than 1 hour due to the periodic repetition of work tasks and machine cycling. Therefore a period of 1 hour was considered to be adequate to include the relatively small SPL and frequency contents variations, and this was the selected total recording time. The data quantity when digitally recorded was still considered to be manageable.

The recording system should allow an easy data collection as well as a flexible retrieval and reproduction under software control. As the target reproduction system was specified to work automatically, that is booting-up and starting sound reproduction and later in the day stopping and shutting down, the most appropriate architecture was a PC

based system with analogue data output system. The sound recording should be done also on this platform, eventually with some hardware differences.

Software control followed the Virtual Instrument approach, including real-time operation monitoring and event logging. A non-interruptible power supply was employed to power the whole system and a log file was created and regularly updated during operation time. This logging facility has proven to be very valuable due to the many “crashes” that the software system underwent during the 6-month operation. In this way, the sound exposure could then, eventually, be completely recovered by completing the programmed sound output period duration, after a “crash” or a power-down event, each day. In spite of the careful selection of components, the common PC-based system has proven to be not very reliable in terms of time stability of operation.

Data acquisition and retrieval was done by means of a data acquisition board with some additional external equipment as needed. A sound acquisition board was initially considered, but the input bandwidth did not reach 2 Hz. A general-purpose instrumentation data acquisition board was therefore selected for the input process.

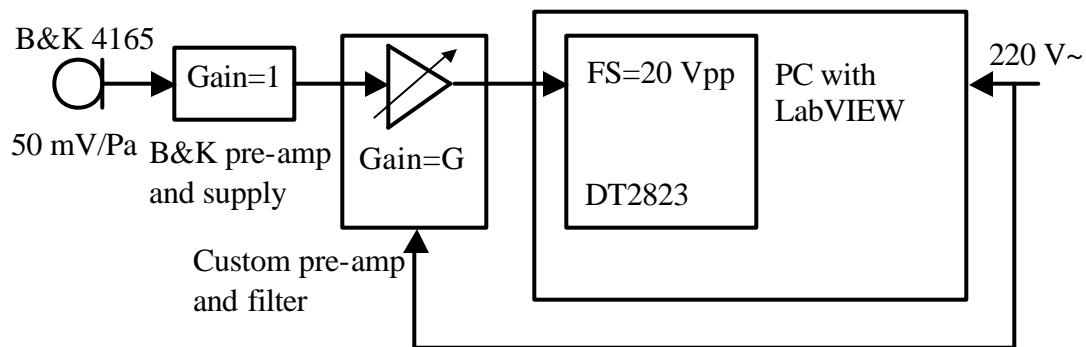


Figure 1- Configuration of the sound acquisition system.

For the sound output process, however, the sound board was good enough, extending its bandwidth down to DC. The loudspeaker system should need the use of separate drivers for the higher audible and for the lower and sub-sonic ranges, leading to crossover filtering for separation into 2 channels. The availability of 2 channels in the sound output board allowed the output of the 2 digitally crossover-filtered signals, one for each loudspeaker range.

## SOUND RECORDING AND ANALYSIS

The recording system should allow easy data collection and prepare for flexible retrieval and reproduction under software control. The configuration of the system that was used can be observed in Figure 1

A set of virtual instruments was designed to control the data recording operation, allowing a preparation phase for gain adjustment, by use of an amplitude histogram, to determine the necessary input gain. The gain was adjusted such that a 1-minute long record would show a percentage of saturation time near the 0,3% target value.

A nominal gain of  $G=104$  was initially foreseen for a shop floor noise sound level of 90 dB (0,63 Pa) to produce a FS signal excursion at the DT2823 input. This value was slightly decreased in-site due to the higher sound pressure level encountered. A representation of the amplitude histogram of a 1-minute long signal can be observed in figure 2. The shape is similar to a gaussian curve as expected.

The input sample rate was selected with the highest value available in the DT2823, 100 ksamples/sec, in order to simplify the construction of the anti-alias filter. This should

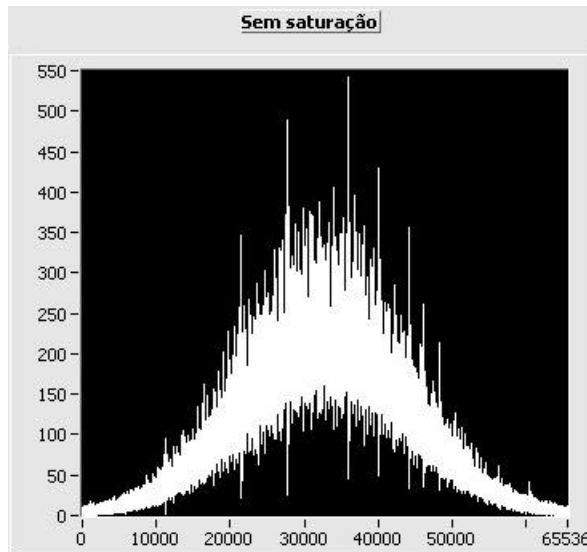


Figure 2 – Representation of the amplitude histogram of a 1-minute record excluding limit values.

bring the input signal's amplitude down to  $-63$  dB @ 50 kHz relative to the mid-band value. The required attenuation from the filter is considerably smaller due to the natural sound pressure roll-off with increasing frequency. A rough estimate of  $-40$  dB SPL value relative to 500 Hz, based on a  $-20$  dB/decade roll-off was used for design. In practice a substantially different roll-off of approximately  $-40$  dB/decade starting at 5 kHz was observed but anyhow converged to a similar attenuation at 50 kHz. An additional attenuation of 23 dB was required (value reachable with a 3<sup>d</sup> order design) for the anti-alias filter, so that the processing noise ceiling of  $-60$  dB could apply from 0 to 50 kHz. A subsequent decimation was needed for signal reproduction. This is described in

the next section.

Considering equal quantization and background rms noise levels, the requirements lead to a minimum resolution equivalent to 11 bits. The resolution available in the DT2823 is equivalent to 16 bits, with a 5 LSB peak-to-peak system noise, giving an intrinsic maximum SNR of 79 dB. This was largely sufficient for the present purpose and allowed a good margin for the error contributions of the other stages.

During the recording operation, 60 1-minute long consecutive records (720 Mbytes) were made, together with periodic measurement of the SPL, using a calibrated sound-meter; showing values around 92 dB. Data analysis with calculation of rms

values and amplitude histograms was done to check saturation levels before further processing, so avoiding possible underestimation of peaks' amplitudes produced by subsequent processing. An example of amplitude histogram of a 1-minute file can be observed in figure 2. This file presented a 0.44% saturation time.

The signals were then ready for the post-processing in preparation for the reproduction task. This preparation was by its turn composed of a 2:1 decimation to convert to 50 ks/s and a spectral analysis and crossover filtering for 2-channel reproduction.

Spectral analysis was performed by means of an 1/3-octave filter bank in the digital domain, except for the lower frequencies in the band from 2,46Hz to 9,84Hz, in which 1-octave bands were used in order to decrease the specific filters' settling times. Chebyshev filters were chosen with 50 dB selectivities. The octave band filters were based on a 50 dB selectivity Butterworth approximation because Chebyshev filters tended to become numerically unstable. A total of 3 octave and 32 1/3-octave filters were used, the highest central frequency was 20159 Hz.

This filter bank was also used for crossover filtering of sound paths and equalization of the loudspeaker-room response. The global frequency response represented in figure 3 was obtained by means of the joint plot in Matlab of the amplitudes of the DFTs of the impulse responses of the individual filters, calculated in LabVIEW.

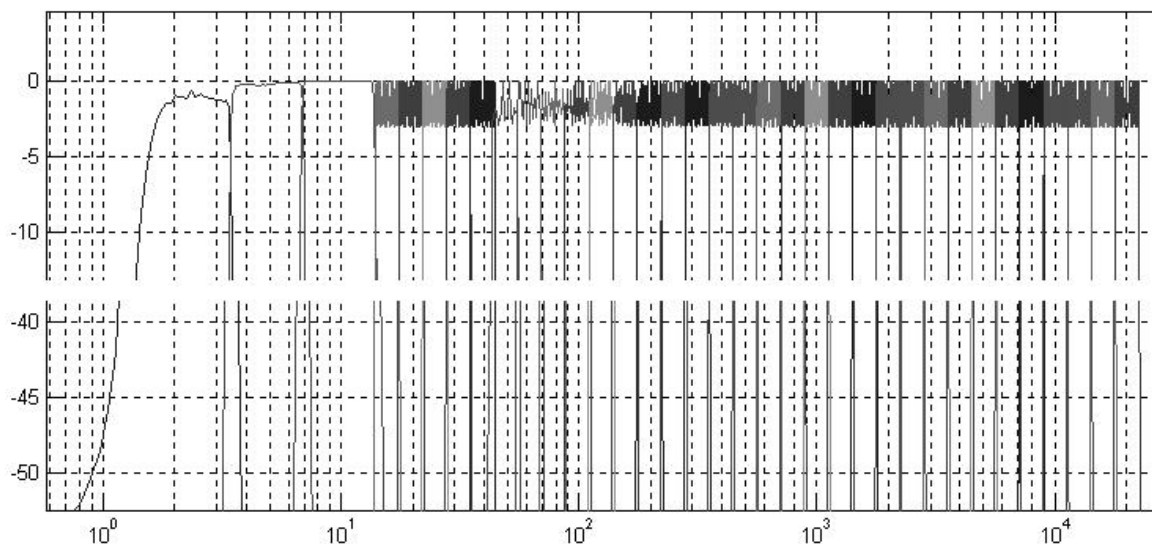
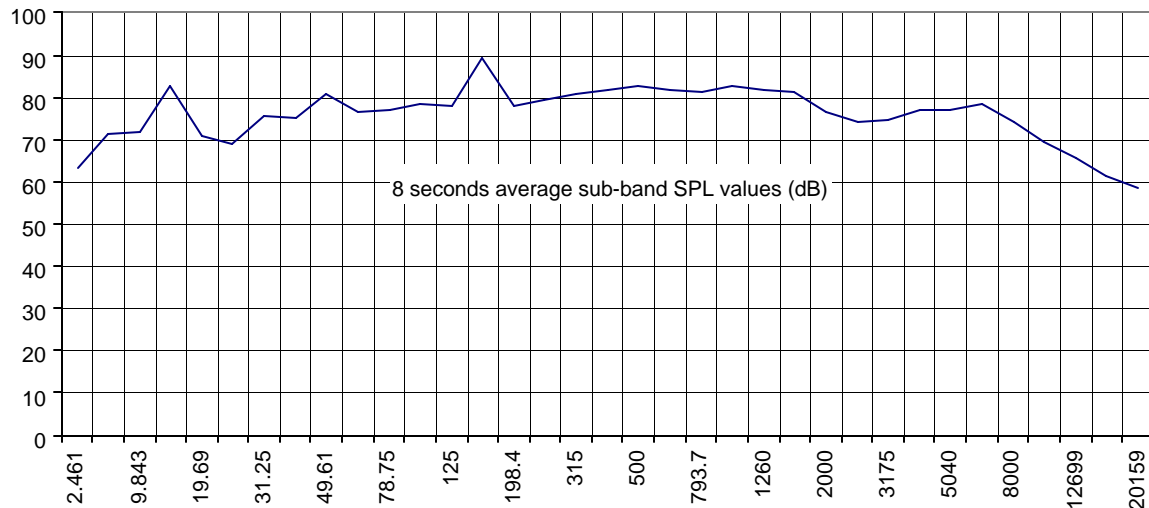


Figure 3 – Ensemble amplitude frequency responses of the filter bank in dB.

The amplitude function of the spectrum of the recorded signals was determined by passing them through the filter bank, under LabVIEW followed by sub-band rms values calculation of features in Excel for 1, 2, 4 and 8 seconds segments with  $\sin x/x$

correction.. The plot of figure 4 shows the average spot frequency distribution of the data blocks of 8 seconds duration.

It should be noted that the plotted values in figure 4 represent the SPL calibrated output values of each individual filter. For this calibration to be possible the complete sound acquisition system was tested beforehand with a calibrator.



*Figure 4 – Plot of the average SPL in function of the 1/3-octave central sub-band frequencies. For the 3 lower sub-bands the bandwidth was 1-octave, therefore if a comparison is required the values need to be adjusted*

The calibration curve's average value was 15 dB. An inspection of amplitude densities ( $\text{dB}/\sqrt{\text{Hz}}$ ) requires scaling is through division by the square root of the specific filters' bandwidths. The 3 1-octave filters produce larger outputs than would appear if they followed the general trend of 1/3-octave, like all the others. In the subsequent operations however this was not a source of error because the same filters were used for all the operations. The approximately uniform filters' outputs at mid-frequencies indicate that the noise in question is approximately pink.

## **SOUND PREPARATION AND REPRODUCTION**

The final reproduction of the sound in the target  $45 \text{ m}^3$  laboratory was prepared in order to achieve the least possible variation of SPL amongst the different useful places of the room (basically, two rows of 2-levels bookshelves were to be used for experiments). The final SPL results showed a  $\pm 3 \text{ dB}$  variation. The series of tasks of preparation started by the evaluation of the room's and bookshelves acoustical characteristics. The reverberant nature of the enclosure advised to use an array of 16 spatially distributed loudspeakers

for the higher frequency bands and one sub-woofer for the lower ones. Careful insulation of room openings was also done, taking in consideration the ventilation facilities.

Availability of high-range loudspeakers of the monitor type allowed a relatively easy design. Oppositely, non-existence of sub-sonic devices demanded a custom solution. As power levels involved weren't large it was decided to overload the sub-woofer, making it operate for subsonic frequencies. This overcharge would be done through equalisation of the programme material in each sub-band, producing a raise in excursion level of (a respectable) 40 dB/decade for frequency decrease below resonance. It was however confirmed by calculation that the driver would not saturate neither suffer excessive power dissipation. Figure 5 represents the reproduction system.

A crossover frequency of 111 Hz was selected, in coincidence with the boundary between filters 12 and 13. Next, the overall gain of the reproduction system and the gain of each filter in order to reach the same SPL in each band as in the original source sound were adjusted. Sound acquisition, simultaneously with sound output, was used to control in each specific band the spatially averaged output from the reproduction chain, and to calculate a calibration gain curve. The 2 output channels of the sound board were used for output, under LabVIEW control.

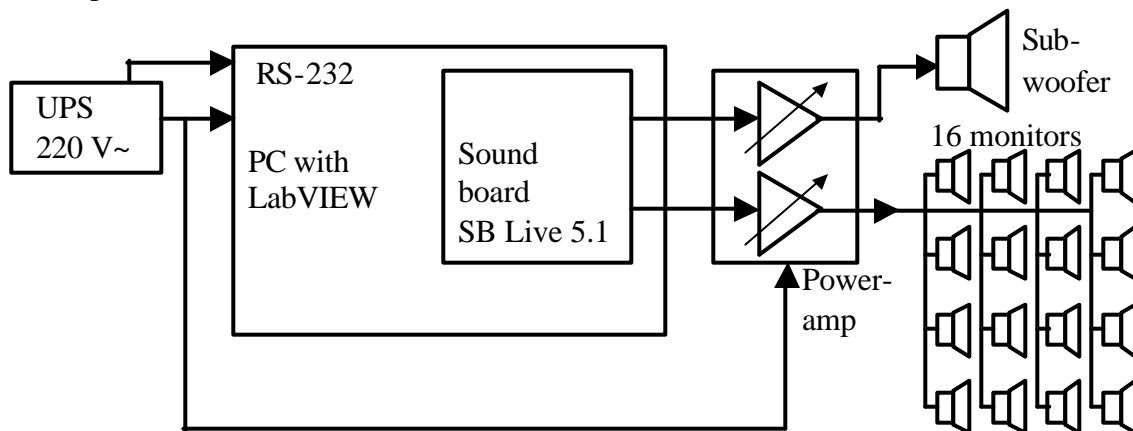


Figure 5 - Configuration of sound output system.

Determination of the equalising gains for each band was done in three steps. Firstly, the global average transfer function for each channel was determined from the sound data to the spatially averaged SPL in 12 points carefully defined in the room.

The sub-woofer and the full-range monitors were excited with one 10-seconds long signal and sound was recorded in both cases at the 12 points of interest and then passed through filters 1 to 12 (1.95 – 111 Hz) and filters 13 to 35 (111Hz – 22.63 kHz) and the average difference was calculated for the 12 rms values from the value calculated for the original sound to obtain the gain to use for each filter band. The set of gains gives an estimate of the transfer function for each of the 12 points and the average is the

equalizing function. The highest dispersion values amongst the 12 selected points was  $\pm 5$ dB, in the lower frequency channels 7, 8, 9, 11, 12, 13 and 14. In the other channels the deviations were below  $\pm 2.5$ dB. The amplifier gain was then adjusted.

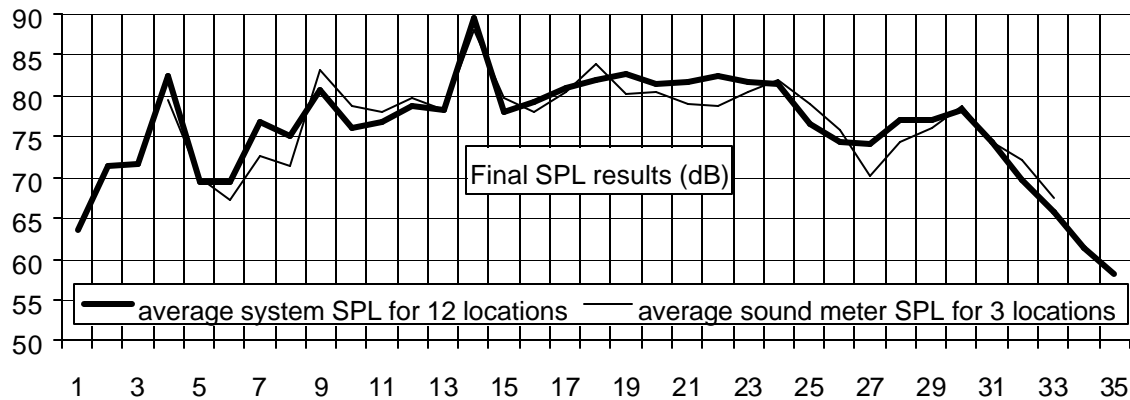


Figure 6 – Plot of the final SPL values measured by the system and by a calibrated sound level meter, showing a reasonable agreement.

Data was subsequently processed by summation of the individual band filtered signals using the equalising gains as weight coefficients. Finally, the equalised sound was reproduced and measured again by means of the acquisition system and a calibrated sound meter for verification of the spectral envelope. The final average differences remained less than  $\pm 1.5$  dB between the average reproduced sound and the original sound in all the bands. Comparison with the calibrated sound meter in the available measuring bands from 16 Hz to 12.5 kHz showed spot differences not higher than 5 dB with a good agreement of the spectral envelopes in general, as can be seen in figure 6.

## CONCLUSION

The above report illustrates the possibility of simulation of random acoustic noise of extraordinarily large bandwidth in a small reverberant closed room by means of digital processing with filter banks, allowing an acceptable spatial and frequency accuracy in terms of SPL and amplitude pdf. Additional work would be needed to design the reproduction system with an improved SPL spatial uniformity in the low bands.

## REFERENCES

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