An electroacoustical testbench: characterizing an audio line

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***Abstract*— A plug-and-play audio testbench is a really useful tool for an electroacoustical emissions laboratory. This paper proposes a suite of tools and a set of configurations designed for obtaining a complete view of the most important audio and electroacoustical figures of merit. The main goal of this testbench is to provide a low-cost electroacoustical characterization system following a black box approach, easy to use in investigation research and educational environments. Furthermore, this development aims to provide a guided set of exercises for the students of the Acoustics Master of the University of Granada to be able to actually work with modern audio electronic deployments like balanced audio lines or digital signal processing toolboxes**

Keywords*— *Electroacoustical, testbench, audio measurements

I. Introduction

The current availability of high-end audio devices and their affordable prices has made possible the use of a powerful sound interface to deal with reliable acquisition of low frequency audio signals. Using a previously characterized interface, a suite of automated tools is developed. This suite is able to determine with certain accuracy and precision the major figures of merit of audio devices under test.

With these tools, it is possible to profile the response of an audio device and dive into the details of how they perform along the main audio goals.

In addition, the retrieved measurements are a great entry point for understanding the operation of an electroacoustic or acoustic device or even an audio line.

A. Audio figures of merit

Measuring and profiling is the main purpose of this development, but first and foremost, there are some definitions required in order to clarify and expose our scope.

First of all, a "black box" device which has (generally) two audio inputs and two audio outputs and some functionality has to be defined. The audio line will be manipulated by this device in a certain manner, this is the functionality.

Due to side effects, the audio signal could be modified in not expected ways. The measurement of these effects determines the quality of the device under test, henceforth DUT.

Thereupon, the sought-after figures of merit are [1]:

- Frequency response: the ratio between the input signal amplitude and the output signal amplitude for a certain frequency, usually expressed in decibels.
- THD or Total Harmonic Distortion: the ratio between the output energy of a single frequency and the energy in its harmonics.
- Cross Talk: the interference between independent input channels reflected in output channels.
- SINAD or Signal to Noise and Distortion ratio: the ratio between the energy of a signal and its harmonics plus the noise floor, usually expressed in decibels relative to carrier.
- SNR or Signal to Noise Ratio: the ratio between the energy of a signal and the floor of noise, excluding the harmonics.

II. Methods

Before any measurements are done is necessary to determine the limitations of the measurement equipment. After this procedure, if the restrictions are wide enough, the developed tools can rely on a hardware structure like the one shown in figure 1.

Once the hardware is in place, fine-tunning the whole testbench (cables, connectors, etc) was necessary in order to make the measurements reliable.

Finally, by connecting the DUT is possible to run the software testbench and obtain the measurement results.

A. Characterizing the measurement equipment

Confining the valid scope of the testbench measurements is allowed by knowing the measurement equipment boundaries and limitations.

Using the configuration shown in the figure 2, a bounded reference of these figures of merit was determined, as well as an accurate approach of the audio spectrum performance [2].

An HP 5903B Audio Analyzer as calibrated reference [3] was used to measure the flatness of the Behringer UMC404HD sound card frequency response, the signal to noise and distortion ratio, the crosstalk between channels, and the input and output balanced impedance in its XLR sockets.

Fig. 1. Block diagram of the electroacoustical testbench

Theoretically, the Behringer UMC404HD sound card is able to sample the audio signal at 192 kHz. Signals with up to 92 kHz spectrum components can be sampled with this sample rate.

In terms of sample depth, the sound card is able to register up to 24 bits precision in each sample. Taking the digital signal theory into account [4] the maximum SNR achieved with this precision is

$$SNR = 20 \times \log_{10}\left(2^{24\,bits}\sqrt{\frac{2}{3}}\right) = 142.7\text{ dB} \qquad (1)$$

Note that -142.7 dBFS is the quantization noise floor and 142.7 dB is the relation between the maximum representable signal amplitude (0 dBFS) and the noise floor.

The measurements were carried out with a low output impedance (600 Ω) and high input impedance (100 kΩ at the end of the line). Additionally, a 30 kHz low pass filter was activated in order to get measurements accurate in terms of audio performance.

Fig. 2. Profiling the measurement equipment Behringer UMC404HD using a calibrated reference: HP 5903B Audio Analyzer.

Once the characterization is completed, the Behringer UMC404HD sound card is ready to be used in the successive procedures. Having defined boundaries of its capabilities will determine the validity of the measurements.

B. *The hardware testbench*

The hardware deployment of the testbench is mainly based on the Behringer UMC404HD sound card. However, some audio and communication cables are used in order to connect the sound card to the DUT and to the main computer, respectively.

This testbench is able to perform a wide frequency band analysis to a certain audio device using noise-robust XLR balanced cables: an audio differential signaling standard.

Two pairs of “the sssnake SM10RD” cables are used to connect the Behringer sound card with the DUT. Two of them for the right channel (from sound card output to DUT input and from DUT output to sound card input) and the other two for the left. Each of these cables have an XLR male and female connectors on its ends.

An USB A to B cable is used to power and communicate the Behringer sound card.

C. *The software testbench*

The main operation of the testbench consists in playing an ordered stream of pre-generated sounds and recording the output of the DUT.

A deep analysis of the DUT behavior is possible having it response to certain audio samples with characterized attributes.

The audio samples that the testbench fires are:

D1. 5 seconds -3 dBFS stereo pure tones at various frequencies ranging from 10 Hz to 100 kHz logarithmically distributed.

D2. 5 seconds 0 dBFS stereo pure tones at various frequencies ranging from 10 Hz to 100 kHz logarithmically distributed.

D3. 5 seconds 0 dBFS mono pure tones at 1 kHz, 10 kHz and 80 kHz in each channel.

D4. -20 to 0 dBFS signals with linear or logarithmic rising pattern at frequencies ranging from 10 Hz to 25 kHz logarithmically distributed.

D5. 20 seconds and -3 dBFS white noise.

D6. 20 seconds and -3 dBFS pink noise.

D7. 20 seconds of silence to sense the noise floor.

The audio samples are crafted, played and recorded using MATLAB R2017a. The audio input and outputs are recorded as a sample array with a certain sample rate.

Some numerical procedures are executed after the play-record phase has ended:

Using data from D1, frequency response is calculated isolating the energy of the output at certain frequency and dividing it by the input energy at the same frequency.

The THD is calculated using MATLAB’s *thd()* function for each recorded sample in D2. Ten harmonics are considered in the calculation.

The SNR is calculated using MATLAB’s *snr()* function for each recorded sample in D2. As said in the documentation: this value is determined using a modified periodogram of the same length as the recorded sample. The modified periodogram uses a Kaiser window with β = 38. The result excludes the power of the first six harmonics, including the fundamental [5].

The SINAD is calculated using MATLAB's *sinad()* function for each recorded sample in D2. This function returns the signal to noise plus distortion in dBc. The measurement is determined using a modified window periodogram of the same length as the recorded signal. The periodogram uses a Kaiser window with β = 38 [6].

Using the recorded data in D3 crosstalk is determined. The monophonic signal allows comparing the energy at given frequency in the driven channel and the same energy in the undriven channel.

Using the rising amplitude patterns fired in D4 analyzing non-linear time effects is possible. These effects, such as compression knee, "pumping" effect or attack and release times.

The results of these tests are exposed in various ways: a MATLAB object ready for postprocessing, an Excel Spreadsheet for reporting purposes, and graphs for easy visual evaluation.

D. Electrical line calibration

In order to get a physical relation between the digital signal generated and the audio line signal some measures should be done before test starts.

The RMS value of the signal driven by the output on an open line will establish the relation between the output digital full scope and the electrical signal.

Connecting the sound card output to the main line input and measuring the RMS value on the connected audio line will give a relation between the input digital full scope and the electrical signal.

III. RESULTS

A. The Behringer UMC404HD calibration

The HP 5903B Audio Analyzer measurements over the Behringer UMC404HD sound card exposed a 50 kHz flat frequency response in both playing and recording.

As shown in figure 3, the selected audio card is capable of perform a SINAD greater than 40 dB and a THD lower of -40 dB on the desired frequency band.

Likewise, a -5 dBu input signal (0.6 Vrms approximately) generates a flat -29 dBFS frequency response in the digital scope of the testbench. This record has a SNR and a SINAD of 50 dB and THD lower than -50 dB, as represented in figure 4.

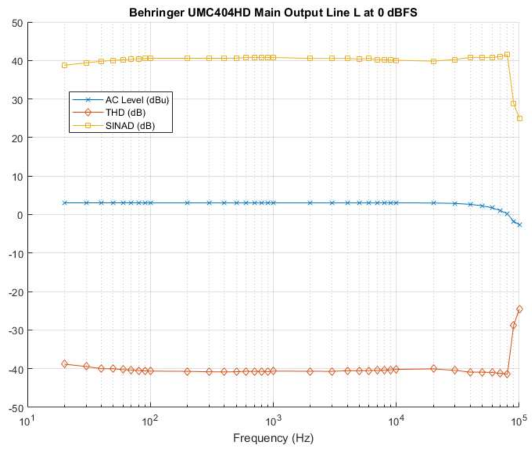

Fig. 3. Behringer UMC404HD sound card 0 dBFS playing merit figures

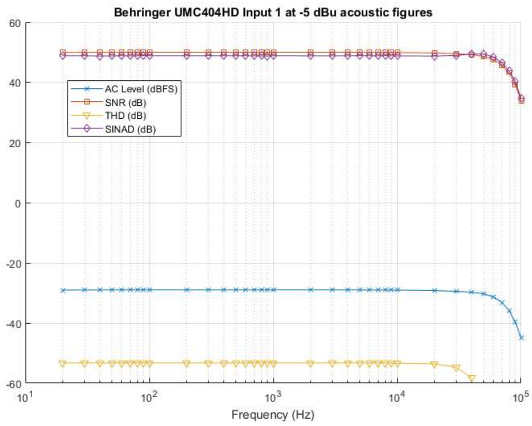

Fig. 4. Behringer UMC404HD sound card -5 dBu recording merit figures

B. The tests

In order to validate the usability of the testbench some audio devices and configurations have been tested. All of them are listed below:

T1. A through configuration consisting on an audio cable connecting the stereo output with the input. A previous testbench bypass check is allowed with this configuration.

T2. A 31 band graphical equalizer with an 80 Hz high pass filter and a 5 kHz rejection band filter on the first channel and an 8 kHz low pass filter and a 500 Hz rejection band filter on the second channel.

T3. An audio compressor with coupled stereo channels configured on 10 dB thresholds with 4:1 ratio, 300 ms attack time and 0.05 seconds of release time.

T4. A digital acoustic emission limiter with an 85 dBA emission limit and a up-to-8 dB slope on the low part of the spectrum. Implemented as a DSP (Digital Signal Processor) running a octave band continuous dynamic range compressor [7].

C. *The test results*

First of all, from the test T1, the through cable, a sound card frequency response is estimated. As shown in the figure 5 a flat 20 Hz to 40 kHz is achieved by only using the input and output interfaces of the Behringer UMC404 HD sound card.

The shown positive 3 dB represents an increase of the full scope of the input over the output.

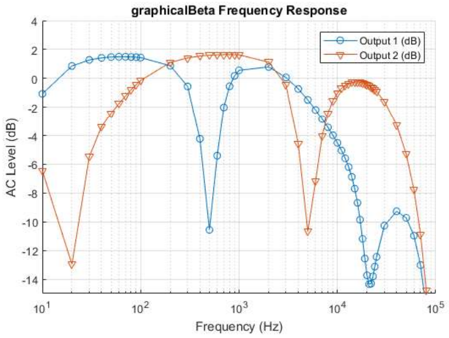

Fig. 5. Behringer UMC404 HD through configuration frequency response.

From T2, a modified frequency response can be observed on the graphical equalizer. The figure 6 shows the notch filters as well as the low and high pass filter measured during the data acquisition.

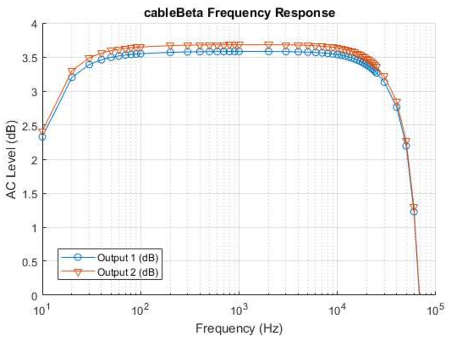

Fig. 6. Graphical equalizer frequency response.

T3 shows, an example of non-linear audio operation can be seen out of the results of the audio compressor test. Figure 6 shows how the dynamic compressor behave on 5 kHz audio patterns.

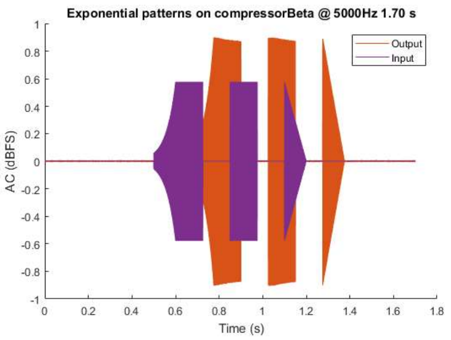

Fig. 7. Input – ouput diagram of exponential test tones on a audio dynamic compressor.

The last test, T4, determines the behavior of a complex audio device and demonstrate the conclusions that a black-box oriented test can help to make.

The measurement of a digital acoustic emission limiter shows how its performance in terms of merit figures decrease beyond the 23 kHz, as is shown in the figure 8.

This kind of behavior could be can be attributed to a 44.1 kHz digitalization of the signal in order to process it digitally in a DSP.

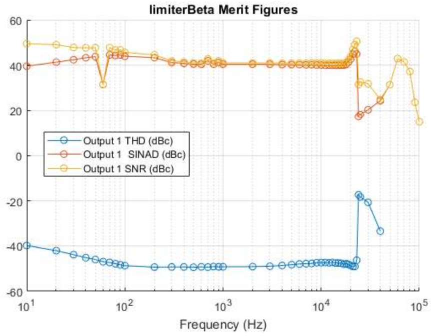

Fig. 8. Input – ouput diagram of exponential test tones on a audio dynamic compressor.

IV. Discussion

Having a wide range of measure while analyzing audio signals and devices allow a complete viewpoint of what is happening through the audio line.

In the shown examples the characterization of a graphic equalizer, the nonlinear operation of a compressor or the complex behavior of an acoustic emission limiter can be explained using obtained data and graphs. The main reason is that the understanding of these kind of complex system can be easily improved by mixing the theoretical concept with a practical experimentation.

Some interesting teaching or researching conclusions extracted from the different test due to the testbench automated results could be:

- Beside the expected 20 Hz to 40 kHz pass band of the T1 configuration, a 3 dB deviation is observed between the output and input of the Behringer sound card. The meaning of this observation is that the output dynamic range is 3 dB greater than the input's one.
- From T2, the Q factor of the individual notch filter can be measured. The neighboring frequencies attenuation is a fixed and not obvious parameter of a graphic equalizer which can be observed and measured by setting some notch filtering and executing the frequency analysis of the testbench.
- By examining the behavior of the dynamic compressor in T3 during different duration audio patterns, its attack and release time can be estimated. Some effects, such as pumping, can be observed in larger patterns.

- As explained before, T4 results show how an DSP (Digital Signal Processor) performs based on its sample frequency and how its (expected) antialiasing filters degrade the signal quality beyond the Nyquist limits. In this case, a 48 kHz audio sampling rate can be inferred from the quality of the signal before and after the 24 kHz limit.

Returning to the testbench design, the data analysis is separated from the data acquisition in order to keep the concepts accessible. Due to that, the system functionalities are easy to upgrade. Appending new tests and measurements will be just using the available measurements to perform new analysis or create new acquisition sets and implement the required analysis.

Based on the exposed concepts and the testbench workflow some other exercises are proposed as interesting upgrades:

- Use the electrical calibration (section 2.D) plus a sonometer to extend the functionality of the testbench. A realistic acoustic emission level expressed in decibel can be obtained using the combination of a loudspeaker plus microphone as DUT. A complete overview of an electroacoustical system is achieved having the linear relation between the digital full-scale magnitude to the electrical signal amplitude and the acoustic emission intensity produced by the loudspeaker.
- Profile the delay of a digital audio processing system using the testbench patterns test. In order to estimate the signal delay introduced by the DUT is necessary to estimate (mean value and standard deviation) the delay introduced by the Behringer sound card using the test T1 (through configuration). Once the base measuring delay is profiled, the DUT delay can be obtained.
- Generate an increasing amplitude test aiming to obtain an input – output diagram. Using this diagram, the threshold or the knee of a dynamic compressor can be measured accurately.

V. Conclusion

This first approach to an automated, low cost and home-made audio test bench have been designed with a teaching-oriented mind. The proposed upgrades to the system are an interesting evaluable task for a group practical lessons or a final degree project.

Due to this teaching focus, most part of the code is self-explicative. Extracted analysis are derived from data in a standard approach using basic concepts and procedures.

In the same path, stored data use common file format as .mat, .fig or .csv. This file formats allow a later data analysis using standard educational tools.

So that, this development aims to be a useful tool around which different concepts on electroacoustic can be developed. From basic concepts as SNR or SINAD to more complex ones like measuring and calibration theory or non-linear system analysis can be taught with some theoretical classes supported by practical classes using this kind of tools.

Acknowledgment

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References

[1] S. Temme, “Audio Distortion Measurements.” Brüel & Kjaer, 1992.

[2] A. Wilson and B. Fazenda, “Characterisation of Distortion Profiles in Relation To Audio Quality,” *Dafx 2014*, pp. 1–8, 2014.

[3] AES, “AES Standard method for digital audio engineering - Measurement of digital audio equipment,” *Audio Engineering Society*, vol. 1998, pp. 1–20, 1998.

[4] J. Watkinson, “The Art of Digital Audio,” *J. Electrochem. Soc.*, 1982.

[5] “MATLAB SNR Function Documentation,” *MATLAB 2017a Documentation*, 2017. [Online]. Available: https://es.mathworks.com/help/signal/ref/snr.html. [Accessed: 15-Jan-2018].

[6] “MATLAB SINAD Function Documentation,” *MATLAB 2017a Documentation*, 2017. [Online]. Available: https://es.mathworks.com/help/signal/ref/sinad.html. [Accessed: 17-Jan-2018].

[7] E. Lindemann, “The Continuous Frequency Dynamic Range Compressor,” in *Proc. 1997 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, NY*, 1997, no. 0, pp. 1–4.