

Dynamic Performance of a Matlab and Soundcard-based FFT Analyser

S.R. Sheikh Raihan

Department of Electrical Engineering, Faculty of Engineering, University of Malaysia,
50603 Kuala Lumpur, Malaysia, Email: siti_r@um.edu.my

Abstract

Normally, to determine the dynamic range of a sound card, a spectrum analyser is used. However, the spectrum analyser can be replaced by using the sound card combined with FFT (Fast Fourier Transform) analyser software. This low-cost design is implemented on a personal computer (PC). A sinusoidal signal is given as an input to the sound card, and the FFT analyser program is used in determining the dynamic range from the spectrum. This approach is able to identify the dynamic performance of the sound card.

Keywords: FFT analysis, sound card, dynamic range, Matlab programming

1.0 Introduction

A real spectrum analyser is expensive and bulky, and they are normally used in laboratories. Inexpensive sound card that is capable of recording and playing analog signal and a software to display the input spectrum is a cheaper alternative than a laboratory spectrum analyser. PC-based spectrum analyser offers the flexibility to program additional functions that are required in a spectral analysis. In [1] and [2], a sound card has been used with Matlab and the free version of Cool Edit software (now known as Adobe Audition [3]) as a PC-based audio waveform generator, an oscilloscope and analyser. However, PC-based spectrum analyser with sound card has its limitations in analysing analog signal due to the noise from the analog-to-digital-converter of the sound card that affects the quality of the signal. Additional noise, quantization errors, and harmonic distortions are some of the dynamic errors that occur during the sampling of an analog signal. To get the dynamic specifications of the analyser, the dynamic performance testing of the sound card has to be evaluated. The performance is measured by the ratio of the amplitude of the fundamental component to the amplitude of the worst harmonic. [4]

The implementation of PC-based spectrum analyser is shown in Fig.1. The sound card that is used in this system is Sound Blaster Live![5]. The analog signal, V_{in2} , is fed into the input (Line-In) of the soundcard. The FFT analyser is programmed in Matlab 6.1, and is used in analysing the signal, and the result is displayed on the PC monitor.

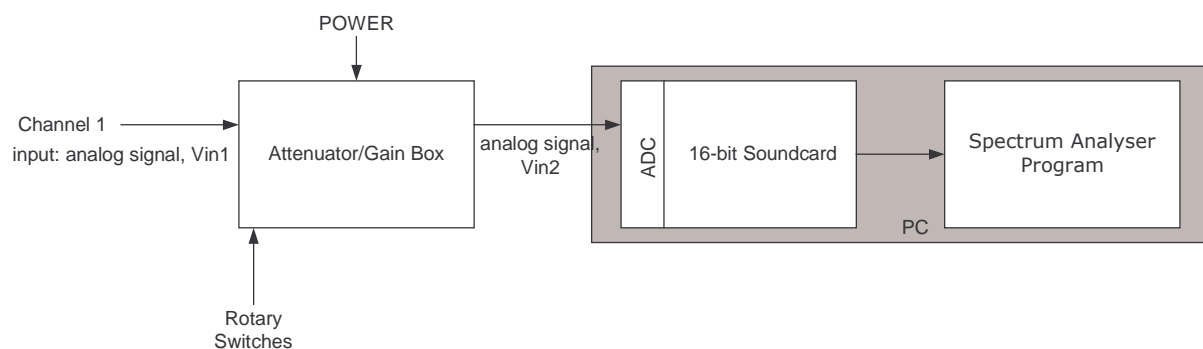


Fig. 1: The block diagram shows the PC-based spectrum analyser system.

To analyse a large or very small input signal, an attenuator/gain circuit is implemented to produce 1V amplitude as the soundcard input, V_{in2} . The rotary switches are used to select the desired input signal into the sound card. The spectrum of the input signal will display the actual amplitude, V_{in1} , on the PC monitor. The attenuator/gain circuit is powered by 5 V_{DC} from an external voltage source. As stated in [5], the sound card has full duplex capability for playing and recording digital signals simultaneously, maximum input voltage (Line Input) of 2V_{pp}, and maximum sampling frequency of 48 kHz. Thus, the maximum allowed frequency of the input signal is 20 kHz.

Fig.2 shows the block diagram of the attenuator/gain box for one channel. The input and output of the circuit is controlled by a relay that operates two switches (S1 and S1*) simultaneously as chosen in [6].

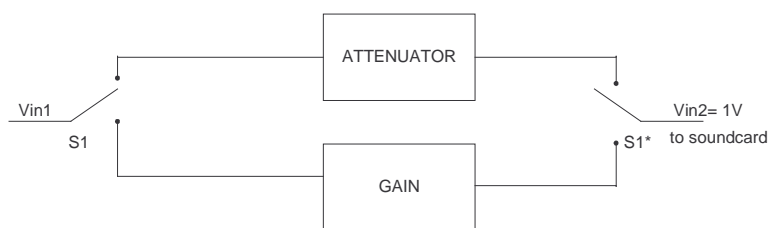


Fig.2: The block diagram of the attenuator/gain box for one channel

The gain circuit is built by using an inverting amplifier, which operates as a negative gain amplifier. Three inverting amplifiers are needed for three voltage inputs: 300mV, 100mV, and 30mV. The 1V input is fed through a voltage follower. Diodes are used as voltage limiter for safety protection. Two 0.7V diodes are used before each operational amplifier to clip the input voltage from becoming dangerously high. To improve the dynamic range of the output signal, the operational amplifiers chosen for this circuit must have low-noise, wide bandwidth, and low total harmonic distortion as specified in [7].

The attenuator circuit is a group of voltage dividers. Four voltage dividers are needed for the input voltage of 3V, 10V, 30V, and 100V. In this circuit, the input impedance is fixed at 1M Ω . The rotary switches control the relays in the attenuator/gain box.

2.0 FREQUENCY DOMAIN ANALYSIS

Dynamic specifications are usually performed in frequency domain, using the Fast Fourier Transform (FFT) or Discrete Fourier Transform (DFT). The FFT performs by averaging the frequency content of a signal over the time interval that the signal was acquired. Therefore, it is recommended to use FFT for analysing stationary signals.[8]

The concept of FFT is shown in Fig.3.

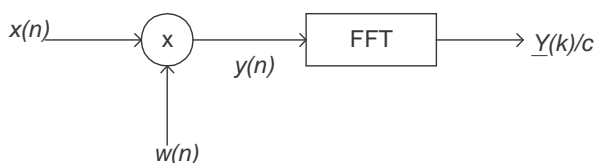


Fig. 3: FFT-based spectrum analyser

$x(n)$ denotes the sequence to be analyzed. The duration of the sequence is limited to N samples in the interval $0 \leq n \leq N - 1$, which is equivalent to multiplying $x(n)$ by a rectangular window $w(n)$ of length N .

$$y(n) = x(n) \cdot w(n) \tag{1}$$

When a window is multiplied with a time-domain signal, it introduces distortion to the signal, and changes the overall amplitude of the signal. To yield the same result as the original signal, a scaling factor, c , of the window has to be divided to the distorted signal. The window scaling factor is taken as the mean value of the window function.

The windowed signal, $y(n)$, is transformed into frequency domain using DFT with the following formula:

$$\underline{Y}(k) = \sum_{n=0}^{N-1} y(n) \cdot \exp(-j \frac{2\pi}{N} kn) \quad (2)$$

where

N = number of samples

k = number of element in the transformed series in frequency domain

Other common window functions such as the Blackman window and Hanning window can be found in [9] and [10], and the Flat Top window in [10] and [11]. The use of windowing is critical in FFT-based measurements especially during the occurrence of spectral leakage, which is undesirable. The spectral leakage often conceals the spurious signals produced by the analog-to-digital-converter (ADC) of the sound card.

3.1 Power Spectrum

Power spectrum is a powerful tool for determining the dynamic performance of the ADC of a sound card. One of the most important computations in spectral analysis is converting a two-sided to one-sided power spectrum. A time domain signal is converted to its equivalent power spectrum in frequency domain by squaring the absolute value of its FFT. Thus, the power spectrum consists of components in the positive and negative frequencies. According to [8], actual frequency analysis tools require only the positive half of the spectrum, and the spectrum of a real signal is symmetrical around DC component. Since half of the energy is distributed in the negative frequencies, converting to one-sided from two-sided spectrum requires removing the component in the negative frequencies and doubling every component except DC component in the positive frequencies. The amplitude of a signal $x(t)$ is calculated by taking the absolute value of FFT, which is normalized by the number of FFT points, N :

$$\text{Amplitude Spectrum, } M = \frac{|FFT(x(t))|}{N} \quad (3)$$

It is useful to normalize the output of FFT so that a unit sinusoid in the time-domain corresponds to unit amplitude in the frequency domain.

The phase is given as:

$$\text{Phase (degrees)} = \tan^{-1} \left(\frac{\text{imaginary}(FFT(x(t)))}{\text{real}(FFT(x(t)))} \right) \quad (4)$$

As stated in [12], the power spectrum and power density are given as:

$$\text{Power Spectrum (dB)} = 20 \times \log_{10} \left(\frac{M}{M_{ref}} \right) \quad (5)$$

where

M = amplitude of the spectrum

$M_{ref} = 0.316V$ (the reference voltage that corresponds to 0dB level)

$$Power\ Density\ (dB) = Power\ Spectrum(dB) - (10 \times \log_{10}(ENBW \times \Delta f)) \quad (6)$$

where

$$\Delta f = \frac{F_s}{N} = \frac{\text{sampling frequency}}{\text{number of FFT points}}$$

$ENBW = \text{equivalent noise bandwidth corresponding to window type}[13]$

3.2 Dynamic Range Specification

Dynamic specifications for an analog-to-digital converter (ADC) in high-speed applications (e.g. digital communications, and instrumentation) are very important for evaluating the performance of the ADC. In this case, the ADC of the soundcard is tested for its performance. Sine waves are usually used in ADC testing because the quality of the sine wave can easily be determined by spectrum analysis. Dynamic performance testing in arbitrary waveform such as ramp signals is also presented in [4]. The common specifications and techniques of testing spectrum analysers as stated in [4] are listed below:

- Signal-to-noise and distortion ratio, SINAD
- Effective number of bits, ENOB
- Spurious free dynamic range, SFDR

For determining the dynamic performance specification of a sound card using PC-based spectrum analyser, the spectral quality of a sine wave is done by measuring the SFDR. SFDR is the ratio of the amplitude of the fundamental component (maximum component) to the amplitude of the largest harmonic distortion or spurious spectral component in a specified frequency range. Harmonic distortion and spurious signals are treated as undesirable spurs in the output spectrum of a sinusoidal input. SFDR indicates the usable dynamic range of ADC of the sound card.

4.0 TESTING OF SOUND CARD

The first stage is to calibrate the measurement. The experimental setup for measurement is shown in Fig. 4. To get a good resolution of data acquisition, the entire sampling time and the frequency of the sinusoidal input signal must be such that the sampling time is equal to an integer number of periods for every spectral component of the input signal. In this measurement, the signal frequency is set to 1 kHz, the number of FFT points is 4800, and the sampling frequency is 48 kHz.

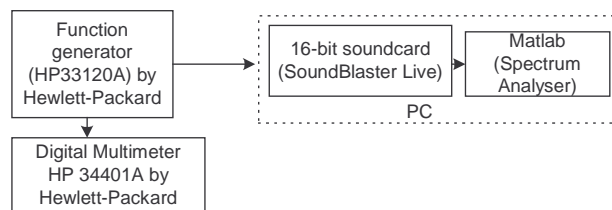


Fig. 4: Experimental setup for calibration and dynamic performance testing

The calibration of the system is done to determine the measurement in the analyser matches with the value indicated in the digital multimeter. The result is shown in Table 1.

Table 1: Difference between input in function generator and analyser for $F_s = 48000\text{Hz}$.
Average factor = $13.977/10 \approx 1.398$

Input from function generator ($V_{\text{amplitude}}$)	Input in Analyser (Actual input) ($V_{\text{amplitude}}$)	Factor (Input/Actual input)
0.1	0.0715	1.3986
0.2	0.1440	1.3889
0.3	0.2145	1.3990
0.4	0.2871	1.3932
0.5	0.3580	1.3966
0.6	0.4292	1.3979
0.7	0.5001	1.3997
0.8	0.5731	1.3959
0.9	0.6438	1.3979
1.0	0.7150	1.3986

From Table 1, it is shown that there is discrepancy by a factor of 1.398 for each input value. This factor is due from the gain of the soundcard. This value is added in the program to correct the measurement.

After having calibrated the measurement, the next stage is to measure the spurious free dynamic range (SFDR). To measure the SFDR, the input voltage of 0.5V amplitude of a sinusoidal signal is applied as the starting value with the signal frequency of 1 kHz, the number of FFT points is 4800, and the sampling frequency is 48 kHz. The input voltage is increased until clipping occurs in the signal. In this case, clipping occurs when 1.44V amplitude input is applied into the soundcard. When the signal is analyzed from the FFT analyser, the harmonics appear in the spectrum is shown in Fig. 5.

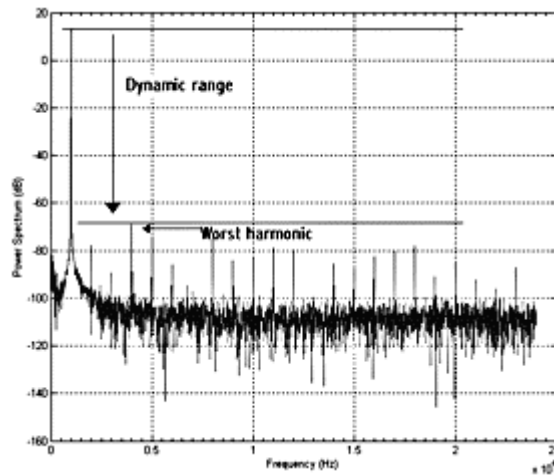


Fig. 5: Measuring the dynamic range of input signal (Power Spectrum, dB vs. Frequency, Hz)

The dynamic range is measured for each input value. Table 2 and Fig. 6 show the result of the measurement. The result shows the worst harmonic happens on the fourth harmonic for every input, and the dynamic range becomes worse when input voltage is higher than 1.31V, and clipping started to occur when the input voltage reaches 1.44V.

Table 2: Measurement of dynamic range at fundamental frequency (1kHz) vs. worst harmonic (4kHz) for sampling rate of 48 kHz.

V _{amplitude} /V	Signal Level/dB	4 th Harmonic Level /dB	Difference /dB
0.50	5.4000	-79.0000	84.4000
0.70	8.3000	-76.0000	84.3000
1.00	11.3495	-72.2858	83.6353
1.10	12.2351	-68.5355	80.7706
1.15	12.5581	-69.3342	81.8923
1.20	12.9924	-68.5623	81.5547
1.21	13.0653	-69.8153	82.8806
1.22	13.1366	-67.5448	80.6814
1.23	13.2074	-65.9186	79.1260
1.24	13.2779	-70.0385	83.3164
1.25	13.3468	-67.3497	80.6965
1.26	13.4027	-67.0029	80.4056
1.27	13.4706	-66.3555	79.8261
1.28	13.5387	-66.2619	79.8006
1.29	13.6066	-70.1146	83.7212
1.30	13.6737	-66.5466	80.2203
1.31	13.7404	-68.3495	82.0899
1.32	13.8058	-65.4334	79.2392
1.33	13.8711	-65.938	79.8091
1.34	13.9365	-60.799	74.7355
1.35	13.9938	-47.8265	61.8203
1.36	14.0465	-39.0387	53.0852
1.37	14.0956	-34.0593	48.1549
1.38	14.1392	-30.4959	44.6351
1.39	14.1836	-27.9902	42.1738
1.40	14.224	-25.7609	39.9849
1.41	14.2628	-23.8779	38.1407
1.42	14.2985	-22.1407	36.4392
1.43	14.3344	-20.7525	35.0869
1.44	14.3658	-19.5817	33.9475
1.45	14.4013	-18.5271	32.9284

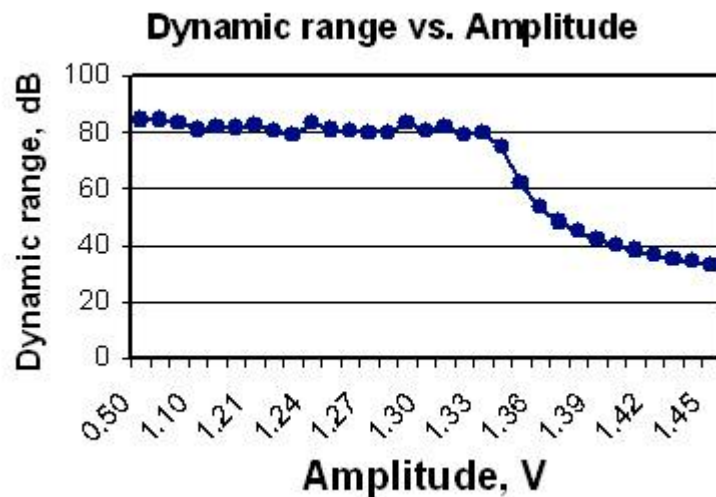


Fig. 6: Sound card performance at 48 kHz sampling frequency

From the result, the quality of the signal starts to degrade after 1.32 V. The average value of the dynamic range taken from 0.5V to 1.31V is 81.3dB. This is the range of signal level the analyzer can operate. Any signal with the dynamic range that is higher than 81.3dB will be undetected by the sound card.

5.0 CONCLUSION

The presented PC-based spectrum analyser has the ability to test the dynamic performance of a sound card. The dynamic range result of the sound card is 81.3dB for the sampling frequency of 48 kHz. Further improvement can be done by testing and evaluating the performance of the measured small and large signals through the attenuator/gain box, and adding more features in the software such as measurement in rms (root-mean-square) value to calculate SINAD. In this case, a large signal is preferable in determining SINAD.

6.0 ACKNOWLEDGEMENT

The author would like to thank Prof. Dr. Wolfgang Mayr for many useful suggestions.

REFERENCES

- [1] Hagler, M. and Mehrl, D., "A PC with Sound Card as an Audio Waveform Generator, a Two-Channel Digital Oscilloscope and a Spectrum Analyzer", *IEEE Transactions on Education*, Vol.44, Issue 2, pp.15, 2001.
- [2] Mehrl, D. and Hagler, M., "Active learning using inexpensive sound cards for circuits and communications experiments," in *Proceedings of Frontiers in Education Conference*, Volume 3, pp. 1102 – 1106, 1998.
- [3] Adobe Audition (formerly known as Cool Edit).
<http://www.adobe.com/special/products/audition/syntrillium.html>
- [4] IEEE Std 1241-2001. "IEEE Standard for Terminology and Test Methods for Analog-to-Digital Converters".
- [5] Creative Lab's Sound Blaster products and specifications.
<http://www.soundblaster.com/products/>
- [6] Low Signal Relay G5V-2 Datasheet. Internet: <http://www.omron.com/ecb/products/pdf/en-g5v2.pdf>
- [7] OPA228, High Precision, Low Noise Operational Amplifiers: Datasheet. Internet: <http://focus.ti.com/docs/prod/folders/print/opa228.html>
- [8] Matlab Signal Processing Toolbox User's Guide.
- [9] F.J. Harris, "On Use of Windows for Harmonic Analysis with Discrete Fourier Transform." *Proceedings of the IEEE* 66, No.1 pp.51-83, January 1978.
- [10] Window function – Wikipedia, free encyclopaedia.
http://en.wikipedia.org/wiki/Hann_window#Flat_top_window
- [11] Gade, Svend and Herlufsen, H., "Use of Weighting Functions in DFT/FFT Analysis (Part I)," *Brüel & Kjaer, Windows to FFT Analysis (Part I) Technical Review*, No. 3, pp. 19-21, 1987.
- [12] LeCroy Digital Oscilloscopes 9304A, 9310A & 9314A series – Operator's Manual.
- [13] J.H. McClellan, C.S. Burrus, A.V. Oppenheim, T.W. Parks, R.W. Schafer, and H.W. Schuessler. *Computer-based Exercises for Signal Processing using MATLAB 5*. Prentice Hall, 1998.

Article received: 2006-06-15