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Loudspeaker Frequency Response Measurements and Bass Reproduction

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Abstract— **In this paper, three types (small-, medium-, and large-size) of multimedia loudspeakers' frequency response measurements signals, findings, and analysis are presented. Measurement signals of maximum length sequence (MLS), and Swept Sine are reviewed, compared, and the reasons for choosing Swept Sine are stated. Measurement hardware setup and software tools used are also presented in details. Measurement results reveal that small-sized loudspeakers have no ability to produce bass at all, whereas medium-sized loudspeakers have poor bass response. Ways to solve the low-frequency reproduction problem of small-, and medium-sized loudspeakers by means of psychoacoustic bass processing are proposed as discussions at the end of this paper.**

Keywords— **Loudspeakers, Frequency Response Measurement, Impulse Response Measurement, Audio Bass, MLS, Swept Sine.**

I. Introduction

Ideally, the audio reproduction devices such as loudspeakers, and headphone frequency response should be flat across audio frequency range from 16 Hz to 20 kHz so that they can reproduce the original audio content without loss of fidelity, and intensity. In reality, however, to achieve such flat frequency response across audio frequency band is a great engineering challenge, and is almost impossible due to the physical limitation of size, and acoustic reproduction power of multimedia loudspeakers.

From 16 Hz to 250 Hz frequency range is considered audio bass region that can be further subdivided into Sub-Bass (16-60 Hz), and Bass (60-250 Hz) [1]. To reproduce the audio frequency contents in Sub-Bass, we need sub-woofers with large cabinet size, and good acoustic reproduction capability at very low frequency. Multimedia loudspeakers can never reproduce audio frequencies in Sub-Bass. In Bass region, not all loudspeakers can reproduce equal amount of acoustic power, and possess the same frequency response curve.

In addition, very small loudspeakers cannot reproduce audio frequency in the Bass region at all, whereas medium-sized loudspeakers reproduce bass frequencies poorly. Thus, depending on the frequency responses of such small-, medium-, and large-sized multimedia loudspeakers, audio quality, fidelity, and pleasantness differ tremendously. In this paper, we investigate the small-, medium-, and large-sized loudspeakers' frequency response characteristics by actual measurements. Measurements were performed in living environment a.k.a. normal laboratory environment. Results were recorded, studied, and discussed. At the end of this paper, the alternate ways to overcome such kind of audio bandwidth limitation problem of loudspeakers are presented.

This paper is organized as follows. Section II describes the two types of measurement signals such as Maximum Length Sequence (MLS), and Swept Sine. The comparisons between the two signals are stated, and the reasons for choosing Swept Sine as the test signal are presented. Section III presents the measurement setup — hardware, and software. Section IV describes the measurement results of large-, medium-, and small-sized loudspeakers. In Section V, the recent research findings in overcoming poor reproduction problem of such small-, and medium-sized loudspeakers by means of psychoacoustics are discussed. Section VI concludes this paper.

II. Measurement Signals

The frequency response is the frequency domain representation of the impulse response of the system under investigation. As their names imply, the impulse response is in the time domain, and the frequency response is in the frequency domain. If a system is a linear and time-invariant (LTI) system, it can be completely described by an impulse response that is measured at the output of the system [2]. The input to the system is the impulse function (Dirac delta function) which can be defined as follows.

$$\delta(t)=\begin{cases}\infty & ,t=0\\ 0 & ,t\neq 0\end{cases}, \qquad (1)$$

$$\int_{-\infty}^{\infty}\delta(t)=1. \qquad (2)$$

Referring to Fig. 1 (a), theoretically, the device under test (DUT) is fed by the input signal, $x(t)$, which is the impulse function, $\delta(t)$. The output signal, $y(t)$ is measured, and is called the impulse response. Fig. 1 (b) shows the Fourier transform equivalence of this impulse response convolution operation. It is noted that the convolution in the time domain is the multiplication in the frequency domain. The above-mentioned operations can be mathematically presented as,

$$y(t)=h(t)*x(t)=h(t)*\delta(t)=h(t). \qquad (3)$$

$$Y(f)=H(f).X(f)=H(f), \qquad (4)$$

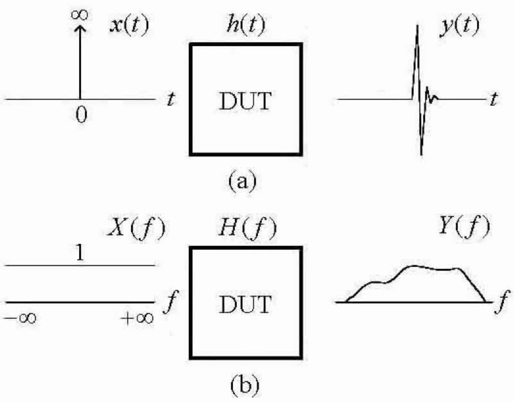

Fig. 1 (a) Time-domain illustration of impulse response generation (b) Frequency-domain illustration of impulse response.

since the Fourier transform of the impulse signal is the unity as

$$X(f)=\int_{-\infty}^{\infty} e^{-j2\pi ft}\delta(t)dt=1 . \tag{5}$$

From (3), and (4), $h(t)$ is the impulse response, and $H(f)$ is the frequency response of the DUT. In addition, the impulse response can also be computed as

$$h(t)=F^{-1}\left\{\frac{Y(f)}{X(f)}\right\}=F^{-1}\left\{\frac{F(y(t))}{F(x(t))}\right\}, \tag{6}$$

where $F\{.\}$, and $F^{-1}\{.\}$ are defined as Fourier and inverse Fourier transform operators. The impulse signal of (1), however, can never be realized in the actual measurement because the signal amplitude is infinity, and the duration is infinitely short.

The alternative signal to measure the frequency response is the white noise [3]. Unlike (1), white noise is a non-deterministic signal that can only be described using statistics. The white noise signal is given by

$$R_{XX}(\tau)=E\{x(t)x(t+\tau)\}=\begin{cases}\sigma_X^2 & ,\tau=0\\ 0 & ,\tau\neq 0\end{cases}, \tag{7}$$

and

$$S_{XX}(f)=F\{R_{XX}(\tau)\}=\int_{-\infty}^{\infty}R_{XX}(\tau)e^{-j2\pi f\tau}d\tau=\sigma_X^2, \tag{8}$$

where $R_{XX}(\tau)$ is the autocorrelation function, $S_{XX}(f)$ is the power spectrum density (PSD) function, and σ_X^2 is the power density value of the input white noise. $E\{.\}$ is the expectation operator, and τ is the time lag parameter.

From (8), note that white noise has equal power density at all frequencies. As compared to (5), (8) has adjustable power density, and white noise is more realizable than the impulse. On the other hand, to compute the impulse response, it has been known that

$$S_{XY}(f)=H(f)S_{XX}(f) . \tag{9}$$

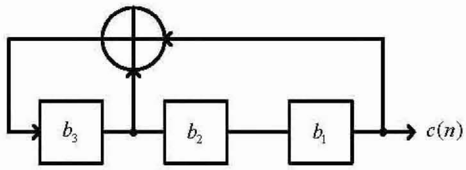

Fig. 2 MLS generator using three bits binary feedback shift registers.

Thus, the impulse response can be computed by

$$h(t)=F^{-1}\left\{\frac{S_{XY}(f)}{S_{XX}(f)}\right\}=F^{-1}\left\{\frac{F\{R_{XY}(\tau)\}}{F\{R_{XX}(\tau)\}}\right\}, \tag{10}$$

where $R_{XY}(\tau)$ is the cross-correlation function between input and output. $S_{XY}(f)$ is the Fourier transform of $R_{XY}(\tau)$.

Alternatively, the impulse response can be computed in the time domain by just cross-correlating between input and output as

$$\begin{aligned}R_{XY}(\tau)&=R_{XX}(\tau)*h(\tau)\\&=\sigma_X^2\delta(\tau)*h(\tau)\\&=\sigma_X^2h(\tau).\end{aligned} \tag{11}$$

The equation (11) shows that the impulse response is proportional to the cross-correlation, meaning that by cross-correlating between input and output, for a white noise input, the impulse response of the LTI system can be recovered. After the impulse response is obtained, the frequency response can be easily computed using signal processing tool such as fast Fourier transform (FFT) in the digital domain.

In order to cross-correlate between input and output, to recover the impulse response, a pseudo-random noise is preferred to the white noise. A pseudo-random noise, such as binary maximum-length sequence (MLS), is a deterministic signal unlike white noise which is purely random in nature. Hence, if MLS is used, the cross-correlation between the known input sequence, and the measured output sequence is easier than white noise. This fact is one of the advantages of MLS over white noise. A brief overview of MLS is presented in next sub-section.

A. Maximum-Length Sequence

MLS can be generated as simple as shown in Fig. 2 [4],[5]. Binary feedback shift registers, $\{b_1,b_2,b_3\}$, are used to store and generate the MLS sequence according to the rule as

$$b_k(n+1)=\begin{cases}b_1(n)\oplus b_3(n) & ,k=3\\ b_{k+1}(n) & ,k=1,2\end{cases}, \tag{12}$$

where n is the time index, k is the binary shift register's position, and $\oplus$ is the modulo-2 sum, or exclusive-or. The output of the binary MLS generator is denoted as $c(n)$ that has only two values, $\{1,0\}$. To generate MLS signal, $c(n)$ is mapped to $s(n)$ that has values, $\{1,-1\}$. The mapping rule is

$$s(n)=\begin{cases}+1 & \text{if } c(n)=1\\ -1 & \text{if } c(n)=0\end{cases}, \tag{13}$$

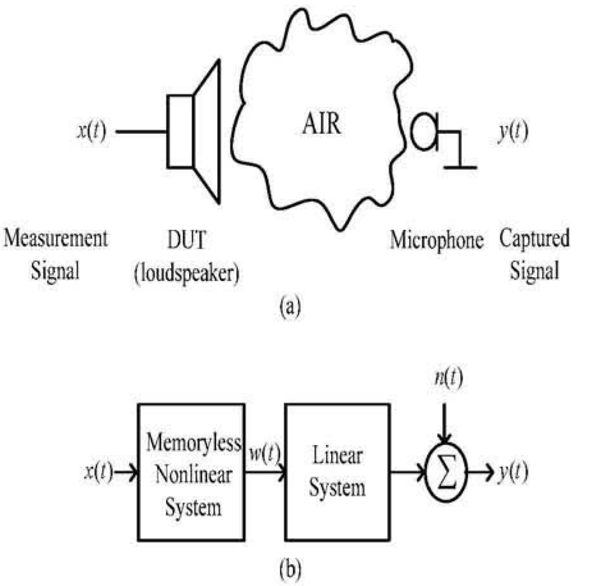

Fig. 3 (a) Loudspeaker measurement signal chain from the measurement signal to the capture signal (b) the model for the signal chain.

All the bit registers are initialized to all one's. MLS generator does not use the stage when all the bit register are zero's. Hence, the length of the MLS is $2^m - 1$, where m is the number of bit registers. In Fig. 2, $m = 3$, and the length of MLS is 7.

MLS is a pseudo-random signal that can be repeated precisely because it is periodic, and deterministic, though its power spectrum is flat, as in the case of white noise. It has been shown that the impulse response can be computed by cross-correlation. Cross-correlation of MLS requires no multiplication; only additions and subtractions are required. Fast transforms such as fast Hadamard transform can be used, and the computational speed is even faster than FFT, and inverse FFT (IFFT) [4].

B. Swept Sine

Swept Sine signal has significant advantages over MLS [7]-[9]. To review these concepts, understanding of the signal path in the measurement, and system theory are presented. First, any LTI system can be described by a convolution operation as

$$\begin{aligned} y(t) &= x(t) * h(t) \\ &= (x * h)(t) \\ &\triangleq \int_{-\infty}^{\infty} x(\tau) h(t-\tau) d\tau \\ &= \int_{-\infty}^{\infty} x(t-\tau) h(\tau) d\tau, \end{aligned} \tag{14}$$

where $h(t)$ is the impulse response, as defined by (3), $x(t)$ is the input signal in time, and $y(t)$ is the output signal in time. Second, the system is time-invariant, with memory, and nonlinear, Volterra series is used, instead, to represent the input-output relation as follows [6]:

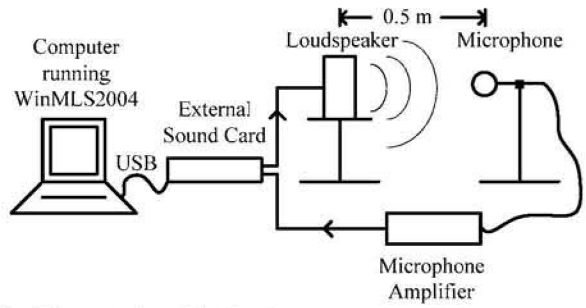

Fig. 4 Computer-based loudspeaker measurement system setup.

$$\begin{aligned} y(t) &= \int_{-\infty}^{\infty} h_1(\tau_1) x(t-\tau_1) d\tau_1 \\ &+ \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_2(\tau_1,\tau_2) x(t-\tau_1) x(t-\tau_2) d\tau_1 d\tau_2 \\ &+ \int_{-\infty}^{\infty}\int_{-\infty}^{\infty}\int_{-\infty}^{\infty} h_3(\tau_1,\tau_2,\tau_3) x(t-\tau_1) x(t-\tau_2) x(t-\tau_3) d\tau_1 d\tau_2 d\tau_3 \\ &+ \cdots + \int_{-\infty}^{\infty} \cdots \int_{-\infty}^{\infty} h_n(\tau_1,\ldots,\tau_n) x(t-\tau_1) \cdots x(t-\tau_n) d\tau_1 \cdots d\tau_n \\ &+ \cdots, \end{aligned} \tag{15}$$

in which $h_n, n = 1, 2, \ldots$ are the multidimensional impulse responses. However, loudspeakers are memoryless nonlinear systems. In this case, (15) can be rewritten as, referring to Fig. 3 (b),

$$\begin{aligned} w(t) &= \underbrace{\int_{-\infty}^{\infty} h_1(\tau) x(t-\tau) d\tau}_{x(t)*h_1(t)} + \underbrace{\int_{-\infty}^{\infty} h_2(\tau) x^2(t-\tau) d\tau}_{x^2(t)*h_2(t)} + \\ &+ \cdots + \underbrace{\int_{-\infty}^{\infty} h_n(\tau) x^n(t-\tau) d\tau}_{x^n*h_n(t)} + \cdots, \end{aligned} \tag{16}$$

where $w(t)$ is the output of the memoryless nonlinear system. Hence, as shown in Fig. 3(b), the captured signal, $y(t)$, contaminated by the noise, $n(t)$, can be modelled as

$$\begin{aligned} y(t) &= w(t) * h(t) + n(t) \\ &= \int_{-\infty}^{\infty} h(\tau) w(t-\tau) d\tau + n(t) \\ &= x(t) * \overbrace{h_1(t) * h(t)}^{\tilde{h}_1(t)} + x^2(t) * \overbrace{h_2(t) * h(t)}^{\tilde{h}_2(t)} \\ &+ \cdots + x^n(t) * \overbrace{h_n(t) * h(t)}^{\tilde{h}_n(t)} + \cdots + n(t) \\ &= x(t) * \tilde{h}_1(t) + x^2(t) * \tilde{h}_2(t) + \cdots + x^n(t) * \tilde{h}_n(t) + \cdots + n(t), \end{aligned} \tag{17}$$

where $\tilde{h}_1(t)$ is the linear impulse response, and $\tilde{h}_n$, $n = 2, 3, \ldots$ are nonlinear impulse responses respectively [7]. The Swept Sine signal can not only obtain the linear impulse response, but also recover the nonlinear impulse responses of harmonics. MLS does not have such kind of property. Swept Sine signal can be categorized into linear sweep and logarithmic sweep [8],[9].

1) *Linear Sweep:* The instantaneous angular velocity is defined as [10],

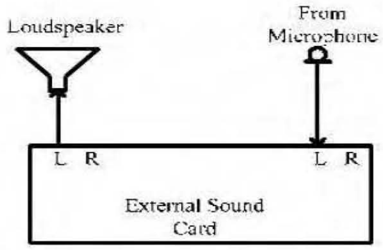

Fig. 5 Cable connection schematic for the one channel measurement using WinMLS2004 [12].

$$\omega(t) = 2\pi f(t) = \frac{d\phi(t)}{dt} \quad (18)$$

where the instantaneous frequency is

$$f(t) = \frac{1}{2\pi} \cdot \frac{d\phi(t)}{dt}, \quad (19)$$

and the instantaneous phase is

$$\phi(t) = 2\pi \int_{-\infty}^{t} f(\tau) d\tau = 2\pi \int_{0}^{t} f(\tau) d\tau. \quad (20)$$

In linear sweep, the instantaneous frequency varies linearly with time [11], i.e.,

$$f(t) = f_0 + kt, \quad (21)$$

in which f_0 is the starting frequency at time, $t = 0$, and k is the rate of change of frequency with respect to time, the sweep rate. Hence, using (20), and (21), the linear sweep equation is given by

$$\begin{aligned} x(t) &= \sin(\phi(t)) = \sin\left(2\pi \int_0^t f(\tau) d\tau\right) \\ &= \sin\left(2\pi \int_0^t (f_0 + k\tau) d\tau\right) \\ &= \sin\left(2\pi (f_0 + \frac{kt}{2}) t\right). \end{aligned} \quad (22)$$

2) *Logarithmic Sweep*: The logarithmic swept sine can be derived first from the exponential sweep that instantaneous frequency varies exponentially with time [11].

$$f(t) = f_0 k^t, \quad (23)$$

where k is exponential rate, and f_0 is the starting frequency at time, $t = 0$. Substituting (23) into (20), the exponential sweep is

$$\begin{aligned} x(t) &= \sin\left(2\pi \int_0^t f_0 k^\tau d\tau\right) \\ &= \sin\left(\frac{2\pi f_0}{\ln(k)} (k^t - 1)\right). \end{aligned} \quad (24)$$

Substituting $k = e$, and defining f_1 as the final frequency when $t = T$, where T is the measurement time, from (23):

$$\ln\left(\frac{f_1}{f_0}\right) = \ln e^T = T, \quad (25)$$

and (24) becomes

$$x(t) = \sin\left(\frac{2\pi f_0 T}{\ln(f_1/f_0)}\left[\exp\left(\frac{\ln(f_1/f_0)t}{T}\right) - 1\right]\right). \quad (26)$$

which is the equation of logarithmic sweep [7].

III. Measurement Setup

A. Hardware Setup

The conceptual hardware setup diagram is shown in Fig. 4. A condenser microphone is used, and is connected to the sound card through a microphone preamplifier. The external soundcard is connected to the computer via a universal serial bus (USB) cable. The hardware tools are listed in Table I.

TABLE I
Measurement Hardware Tools

Item	Manufacturer and Product
Sound Card	Creative Sound Blaster Extigy
Microphone	Earthworks M30 High Definition Microphone
Mic Preamp	Earthworks Lab1

The measured loudspeakers are listed in Table II.

TABLE II
Measured Loudspeakers

Size	Manufacturer and Product
Small	Sonic Gear, 2GO NoW!
Medium	Altec Lansing 120i
Large	Genelec model 1029A

B. Software Setup

For transmitting the measurement signal, logarithmic sweep, capturing the impulse response, and plotting the frequency response of the loudspeakers, *WinMLS2004* software is used [12]. Basic setup is used without any calibration. The measurement set up is one channel measurement, as shown in Fig. 5. The software setting is as shown in Fig. 6.

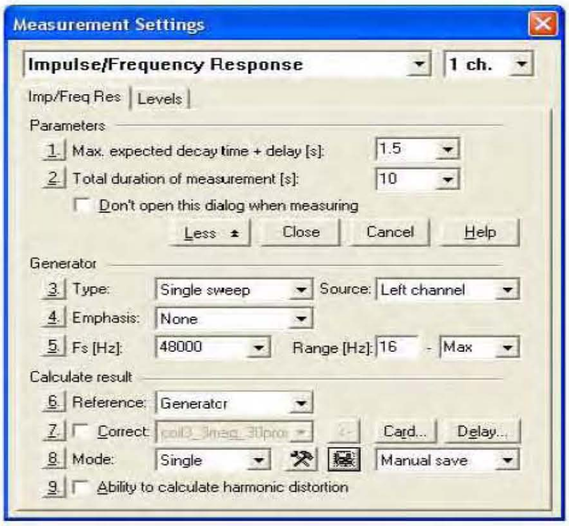

Fig. 6 WinMLS2004 measurement settings are shown. The test signal is the logarithmic sweep. The left channel is used for transmitting signal. The sampling frequency is 48 kHz.

IV. Measurement Results

The measurement results for the three loudspeakers, listed in Table II, are presented in this section. The three frequency

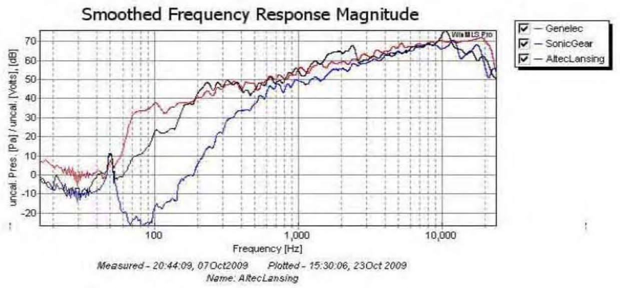

Fig. 7 The three loudspeakers' frequency responses are plotted in a single plot. The lower cut-off frequencies of the three loudspeakers are different, and of interest for bass reproduction.

response plots are combined into a single plot as shown in Fig. 7 for easy comparison. The parameter of interest for this measurement is the lower cut-off frequencies of the three loudspeakers.

From Fig. 7, the lower cut-off frequencies are found to be 70 Hz for the large loudspeaker, 200 Hz for the medium-sized loudspeaker and 400 Hz for the small-sized loudspeakers.

V. Discussion

Since the audio bass frequency region is from 16 Hz to 250 Hz, for the small loudspeakers, there is no or very little bass reproduction, and for the medium-sized loudspeakers, there is a steep low-frequency roll-off in the bass region. Only the large-sized and high quality loudspeaker can extend the bass reproduction down to 70 Hz. Hence, the small loudspeakers sound lack of bass, and booming effect, because it cannot simply reproduce the bass frequencies at all. The medium-sized loudspeakers, though it can reproduce some bass, do not sound as pleasant as large-sized loudspeakers do, because it cannot fully reproduce the bass frequencies, and there is a steep roll-off in the bass frequency region.

Normally, large-sized sub-woofers are used to compensate the low frequencies which cannot be reproduced by normal loudspeakers. If the sub-woofers are not used, there are ways to enhance the degraded audio quality of the small-, and medium-sized loudspeakers. One way is physically to equalize the low-frequency response. Although it might work for the medium-sized loudspeakers, there is a danger that loudspeakers may be overloaded, and damaged. This problem is known as audio bandwidth limitation.

The other way is to use psychoacoustic bass enhancement system. The basic idea is to extend the audio low frequency bandwidth in human's brain instead [13]-[18]. There is a psychoacoustic finding that human can reconstruct the fundamental frequency even if harmonics are presented in the brain. This phenomenon is known as Missing Fundamental. Using this fact, psychoacoustic bass enhancement system uses harmonic generators or synthesizers to generate harmonics of the bass frequencies in the mid-range where loudspeakers can reproduce the audio frequencies well. When these harmonics are mixed together with the original audio stream, human can perceive the bass even if physically bass may be missing due to the bandwidth limitation of small/medium loudspeakers.

VI. Conclusion

In this paper, basics of measurement signal theory are reviewed. The generation of MLS and Swept Sine signals are presented in mathematical equations. The three loudspeakers of small, medium, and large sizes are measured for their frequency responses, and the cut-off frequencies are identified. From that, the loudspeaker bandwidth limitation problem is discussed, and psychoacoustic bass enhancement idea is introduced. From the understanding of the theory to the actual measurements, and analysis, this paper is served as the starting point of more intensive loudspeaker measurements, and psychoacoustic bass enhancement algorithm developments.

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