Reduction of the Noise in the Respiration Sound Recording by the Optimal Sampling Rate of Sound Card: The Verification by Simple Filters

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Abstract — The simple first and second order low pass filter were employed to verify the effect on the reducing the sampling rate of the respiration sound. This study presented a theoretical analysis to prove that there exists an optimal sampling frequency to reduce the noise in the signal to reduce the processes of acoustic digital signal processing. The respiration sounds have been recorded by 2 to 44.1 KHz or higher sampling rates, however, the optimal frequency might be decided by the noise from environment. This study solves the problem from the point of theoretical view. Finally, the study discussed the literatures of noises of the lung sound recording, and examined the effect of the optimization of sampling rate in the processes of moving average.

Keywords — noise reduction, sampling frequency, acoustic signal, lung sound, sound card.

I. INTRODUCTION

In the digital electronics era, the sampling is the basic process of the signal processing. The modern lung sound and heart sound recorders, biomedical sensors, digital image systems, storage devices, digital communication systems get the acquisition of raw data from the humans by sampling [1,2]. The computing speeds of the micro-processors are enhanced by the semiconductor manufacturing technology which allowed the very high speed internal clock to control the units of the processors. Therefore, the sampling rate of the electronic devices has been accelerated in the past decades. For example, the limitation of sampling rate of analog-to-digital converters were 10 MHz 30 years ago, but that of the advanced communication signal converters are several GHz now [16].

The Computerized Respiratory Sound Analysis (CORSA) [3] pointed out the survey of respiratory sounds research, and the sampling rate was listed as a parameter of the method of digitalization [4]. The word lengths, and sampling rates were considered to record the lung sound perfectly by using the sound blaster cards or commercial multi-channel signal acquisition cards. The simplest process of the acquired raw data is moving average as a low pass filter to reduce the noise. The algorithm of adaptive filters acquired the environment noise to cancel the noise in the signals [5]. Furthermore, in the computation of the spectrogram, the overlapped data decided its resolution [6]. Therefore, the total error in a specific duration of lung sound can be a dominant factor of precision.

The adjustable sampling rate has been employed to
reduce the volume in data storage and communication. Some literatures presented the adjustable sampling rate can enhance the signal-to-noise ratio [7,8]. In some cases, the weak signal with lower sampling rate, and the strong signal with higher sampling rate are the rules to reduce the data volume, and enhance the SNR. Therefore, the experimental results can be discussed by the theoretical analysis. The following proof proposed a direct solution to the optimal sampling rate for reducing the noise.

II. METHOD

If the signal x(t) with noise n(t), and the sampling rate is $\omega_s = \frac{2\pi}{T}$, where $T$ is the period. The sampling pulse train is denoted as $p(t)$. Therefore, the sampled signal is:

$$y_p(t) = (x(t) + n(t)) \ast p(t)$$

(1)

where $p(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT)$

In this study, the proof gives a solution to reduce noise by taking an optimal sampling frequency. Taking the Fourier transform [9], we have:

$$Y_p(j\omega) = X(j\omega) + N(j\omega) \ast p(j\omega)$$

(2)

Generally, a low-pass filter with the transfer function of $H(j\omega)$ is always designed as be a pre-processing device to cancel the noise. Therefore, the noise can be optimal reduced by some specific sampling rate in the condition of

$$-\frac{d}{d\omega} (Y(j\omega) - X(j\omega) \ast p(j\omega) \ast H(j\omega)) = 0$$

(3)

That is:

$$-\frac{d}{d\omega} \left( \frac{1}{T} \sum_{k=-\infty}^{\infty} N(j(\omega-k\omega_s)) \right) = 0$$

(4)

Then

$$-\frac{jk}{T} \sum_{k=-\infty}^{\infty} N'(j(\omega-k\omega_s)) = 0$$

(5)

The optimal $\omega_s$ depends on the solution of the differential of $\sum_{k=-\infty}^{\infty} N'(j(\omega-k\omega_s)) = 0$. It exists the solution of $\omega_s$ to reduce the noise.

To verify the proof, we design a computer simulation to show the statistical view of the reduction of noise. Ideally, the respiration can be simplified as a sine wave whose half cycle for inspiration, and the other for expiration. The noise was the additive white Gaussian noise (AWGN) which is the common noise in the communication system [14]. The parameters of computer simulation were listed in Table 1.

| TABLE 1. THE PARAMETERS of the SIMULATED RESPIRATION SIGNAL. |
|-----------------|----------------|
| Parameter       | Value          |
| Shape           | Sine wave      |
| Period          | 3 sec.         |
| Signal to noise ratio (SNR) of AWGN | 20, 30, 40, and 50 dB |
| Sampling rate   | 44.1(Group 1), 22.05(Group 2), 11.025(Group 3), 5.5125(Group 4), and 2.76125(Group 5) KHz |
| Transfer function of the first order low-pass filter | $H(z)=1/(1-0.1z^{-1})$ |
| Transfer function of the second order low-pass filter | $H(z)=1/(1-0.707z^{-1}+z^{-2})$ |

III. RESULTS

The ideally pure sine wave and that with the AWGN of signal to noise ratio (SNR) =40 dB were shown in Fig. 1 (a) and (b). The sampling rate was 44.1 KHz. The root-mean-square (RMS) error was computed by different sampling rates of 44.1(Group 1), 22.05(Group 2), 11.025(Group 3), 5.5125(Group 4), and 2.76125(Group 5) KHz, respectively. We run 100 times of computer simulations by the same parameters in Table 1 to compute the RMS errors. The box plots were presented in Fig. 2.
Figure 1. Computer simulation: (a) the ideally pure sine wave and (b) that with AWGN.

Figure 2 presented the statistical results. Each box contained 100 trials of RMS errors. In (a), the SNR of AWGN is 20 dB. The box plots displayed that Group 5 (sampling rate = 2.76125 KHz) has gotten the minimum RMS errors. Compared with the original signal (Group 1), the operation of reducing sampling rate cancelled about 1/4 RMS errors from the point of statistical view. In (b), (c), and (d), the SNRs were 30, 40, and 50 dB. All results show that the smallest sampling rate (Group 5) gets the smallest RMS errors. The reduction ratios of RMS errors are 0.32, 0.21, and 0.33, respectively. Based on the proof, there exists an optimal sampling rate to reduce the noise. From the box plots, we find that the lower sampling rate got the lower RMS errors by statistical analysis.

Figure 2. The box plots of the RMS errors of Groups 1, 2, 3, 4, and 5 of 100 times. The AWGN level were (a) SNR=20, (b) SNR=30, (c) SNR=40, and (d) SNR=50. (The sampling rates: 44.1(Group 1), 22.05(Group 2), 1.025(Group 3), 5.5125(Group 4), and 2.76125(Group 5) KHz) with the first order low-pass filter.
The results of noise reeducation of the second order low-pass filter were presented in Fig. 3. The function of the second order low-pass filter was better than the first order filter. Anyway, we can find the trend of the figures was that the errors got smaller from group 1 to group 5. That is the lower sampling rate is good to record the respiration sound.

The filter design is very advanced in the digital era. However, we proposed the popular filter to reduce the complexity of computation, and reduce the sampling rate to cancel the noise. Therefore, the condition of simulation is very pure. It disclosed the effect of the reduced sampling rate in noise reduction.

![Box plots of the RMS errors of Groups 1, 2, 3, 4, and 5 of 100 times. The AWGN level were (a) SNR=20, (b) SNR=30, (c) SNR=40, and (d) SNR=50. (The sampling rates: 44.1(Group 1), 22.05(Group 2), 1.025(Group 3), 5.5125(Group 4), and 2.76125(Group 5) KHz) with the second order low-pass filter.](image)

IV. DISCUSSION

The Fourier series described that a periodic signal is the linear combination of the fundamental signal and its harmonics. Therefore, the simulated ideally pure sine wave is significant, because of the theory of the linear time invariant
(LTI) system [1]. The box plots can be expanded to explain the geniality of the operation of the reducing sampling rate. Chang and Lai [11] have studied on the performance evaluation and enhancement of lung sound recognition system in two real noisy environments. The results presented that the frequencies of noises were higher than that the main episodes in the spectrogram. Therefore, if the sampling rate of the experiment reduced to a specific frequency and processed the raw data with moving average, most of the environmental noise can be much cancelled. However, the experienced medical staff usually records the lung in silent environments whose noises compose with the sounds from air conditioners, fans, computer systems and so on. Based on the sampling theory, the energy of noise would be reduced by the lower sampling rate that shows the possibility of the optimal sampling rate can be found in some specific environments. Furthermore, the processing of the digital data can be accelerated by decreasing the data. The benefits of selecting a better sampling rate can achieve the better performance of signal processing. CORSA indicated that the wheeze was contained in the domain frequency at 400 Hz, but a number of investigators have suggested that the range is actually between 80–1,600 Hz and 350–950 Hz by filter theory [12,13]. Lu et al [14] have synthesized the normal breath and wheezing sounds whose main components were under 2 KHz. All the papers provided that the evidence of the sampling rate can be reduced to cancel the noise. However, the crackle includes the higher frequency components, the optimal sampling rate can be higher than normal breath and wheezing sounds. This is the limitation of using the lower sampling rate for lung sound recording. Furthermore, the standardized sampling rates applied in many industry standard sound facilities are 44.1, 22.05, 11.025 or 5.5125 KHz as standards. Most modern research groups selected 44.1 or 22.05 KHz to record the lung sound. However, if the cases of the environmental noises can be controlled as the environment conditions of CORSA description [15], the sampling rate can be reduced or optimized to cancel the noise which can be proven by the box plots of this study. We will analyze the component of the real lung sound and proposed the better sampling rate for various kinds of the sounds of respiration diseases.

V. CONCLUSION

Through the experimental results of the previous studies showed the sampling rate effects on the SNR, this study presented a direct theoretical analysis to prove it. The proof can be employed to simplify the filter design, if we get the smallest error from the simulated signal by the optimal sampling frequency. In addition the reasons of noises in physiologic signals are usually band-limited, therefore, the optimal sampling frequency can be used to simplify the system design. In the green energy era, the reduction the power consumption by simplifying the portable system design is good to the world to achieve the purposes of the medicine or health care of humans. The filter design is very advanced in the digital era. However, we proposed the popular filter to reduce the complexity of computation, and reduce the sampling rate to cancel the noise. Therefore, the condition of simulation is very pure. It disclosed the effect of the reduced sampling rate in noise reduction.

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REFERENCES


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