

Adaptive Filter Application for the Safety of Detecting Lung Sound on Ambulance

Bing-Yuh Lu^{*,**}, Chi Wang^{***}, Hung-Wen Hung^{*}, Meng-Lun Hsueh^{****}, Jin-Shin Lai^{*****}

^{*}Department of Electronic Engineering, Tunghan University, New Taipei City, Taiwan, Republic of China

^{**}Department of Business Administration, National Taipei University of Business, Taipei, Taiwan, Republic of China

^{***}Center of General Education, Tunghan University, New Taipei City, Taiwan, Republic of China

^{****}Department of Electronic Engineering, Hwa-Hsa University of Science Technology, New Taipei City, Taiwan, Republic of China

^{*****}Department of Rehabilitation, National Taiwan University Hospital, Taiwan, Republic of China

franklinlu888@outlook.com, cwang@mail.tnu.edu.tw,

hwhung@mail.tnu.edu.tw, elic0912@gmail.com, jslai@ntu.edu.tw

Abstract—This study presented the application of the adaptive filter to reduce the interference of the sound of sirens (audible warnings) of the ambulance. The auscultation of respiration on the ambulance is usually different from that in clinic room. The environmental noise is much larger on the ambulance, especially, the sound from audible warnings. However, this study simulated the filtering effects of the adaptive filter for the lung sound auscultation on the ambulance. The results showed that the harmonics of the audible warning were completely cancelled, and the component of fundamental frequency of that was reduced from -30dB to -60dB. The adaptive filter is a signal processing algorithm in time domain. Therefore, it is very potential to develop to be a real-time noise cancellation device for the auscultation on the ambulances.

Keywords — acoustic signal, lung sound, adaptive filter, distant healthcare.

I. INTRODUCTION

When a patient is on the ambulance, it is very possible that the medical staffs need to collect the

physiological signals for monitoring and ensuring the safety of the patient [1].

Respiratory sounds occur both of the durations of inspiration and expiration. The reasons of the respiration sounds occur as the air moves in and out of the chest when a person breathes normally. Therefore, the auscultation of normal breathing sounds and abnormal or adventitious sounds can be detected by the stethoscopes [2,3]. However, the interference of the sound of sirens of the ambulance are always influent to the auscultation. Therefore, a speedy algorithm to process the cancellation of the sound of sirens is necessary to the auscultation on ambulance or sent the respiration to the distant emergency center in the hospital. Therefore, we proposed an algorithm of adaptive filter, and simulated the performance of the algorithm as the preparation before implementing the auscultation system on the ambulance.

II. METHOD

A. Conceptual investigation of the sirens

Figure 1 presented the lung sound auscultation on the ambulance, where (a) was the acoustic signal in time domain, and (b) was the

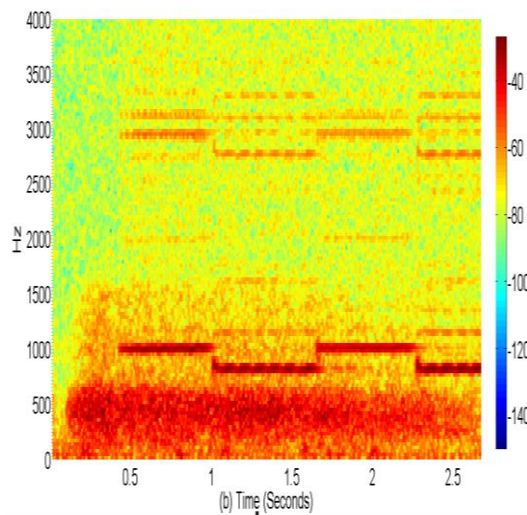
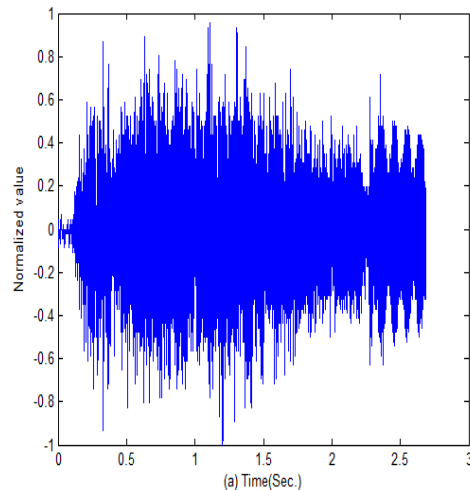


Fig. 1 Lung sound auscultation on the ambulance, where (a) was the acoustic signal in time domain, and (b) was the spectrogram.

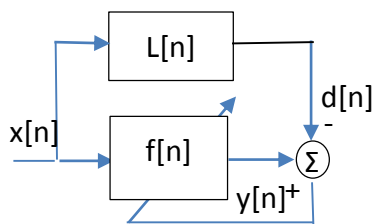


Fig. 2 the functional block diagram of the adaptive filter.

spectrogram. In the spectrogram, the fundamental frequency and its harmonics of the siren and its echo are higher than 750Hz. Therefore, design a low-pass filter to cancel the sound from siren, and then the output signal of the filter became the desired signal of the adaptive filter.

B. System block diagram included adaptive filter

Figure 2 presented the functional block diagram of the filter design. $x[n]$ denotes for the input which was recorded on the ambulance. $L[n]$ is the transfer function of the low-pass filter. $f[n]$ is the function of the adaptive filter, and $y[n]$ is the output of the adaptive filter.

C. Spectrogram

The element at the time τ in a spectrogram is defined as [4,5]:

$$|P_{\tau}(j\omega)|^2 = \left| \int_{-\infty}^{\infty} p(t) \phi_{\tau,\omega}(t) dt \right|^2 \quad (1)$$

$p(t)$ is the signal in the time domain, $\phi_{\tau,\omega}(t)$ is the complex basis function, and $|P_{\tau}(j\omega)|^2$ is the power distribution in the frequency domain at the time τ . This form of the Fourier transform, also known as the short-time Fourier transform (STFT), has numerous applications in speech, sonar, and radar processing. The spectrogram of a sequence is the magnitude of the time-dependent Fourier transform versus time.

Many researchers of the heart and lung sound analysis employed the technology of spectrogram analysis to explore the characteristics of the heart and lung sounds, because the spectrogram presents the power densities in both of time and frequency domains.

III. RESULTS

Figure 3 presented the two respiration sounds which have been processed by the adaptive filter which we have designed. (a) and (b) were the sounds in time domain, and (c) and (d) were their spectrograms, respectively. The results showed that the harmonics of the audible warning were completely cancelled, and the component of fundamental frequency of that was reduced from -30dB to -60dB. The performance of the adaptive filter can be presented in this two spectrograms.

IV. DISCUSSION

Lu, and Wu [6,7] have proposed that relatively simple trial of a potential method for transmitting audio signal derived from auscultation. Contemporary smart phones are widely used because numerous new applications (APPs) are available for the new generation of smart phones. Based on new wireless communication technologies, which include Bluetooth® technology, the modern mobile communication systems, and modern electronic stethoscopes, we developed a real-time mobile-based auscultation (RMA) that can be applied in distant health care to improve the convenience of performing real-time auscultation on patients. Consequently, the solutions of auscultation on ambulance or distant auscultation in the

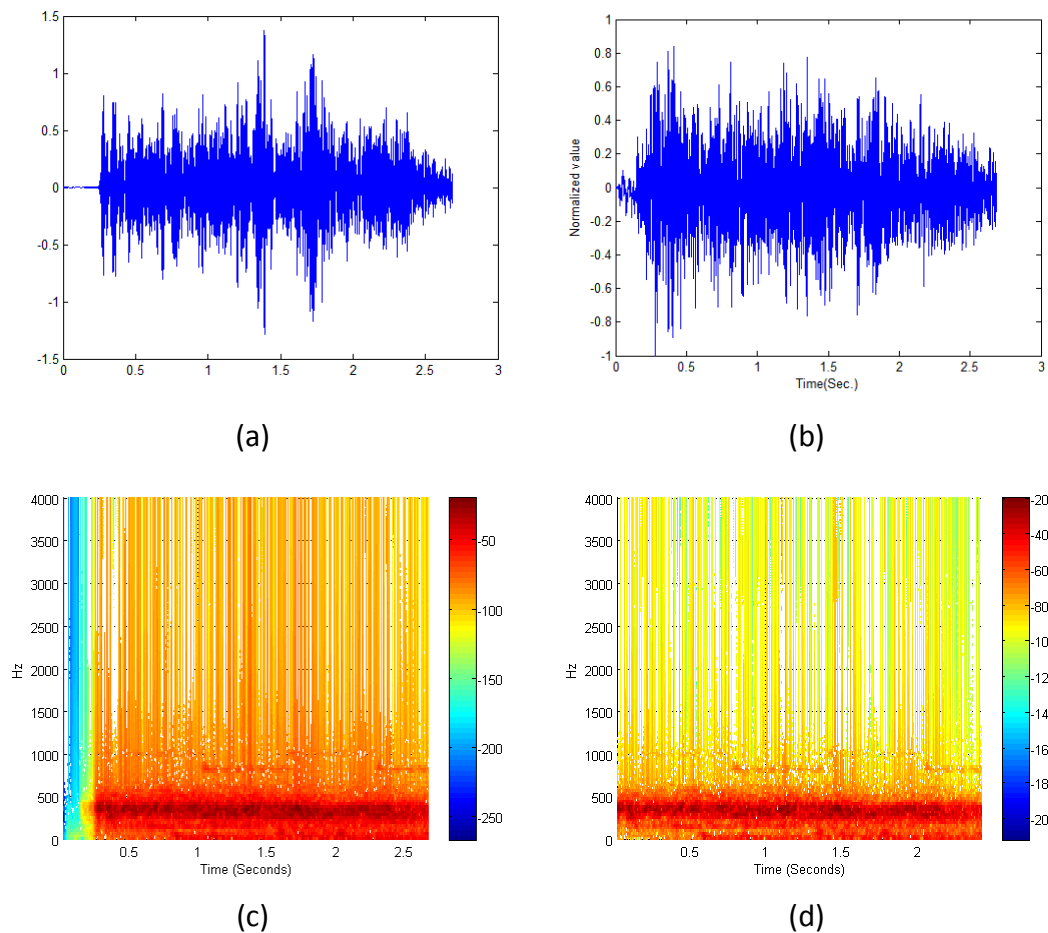


Fig. 3 The two respiration sounds which have been processed by the adaptive filter which we have designed. (a) and (b) were the sounds in time domain, and (c) and (d) were their spectrograms, respectively.

emergency center in hospital has been found to realize the system in the near future. Lu *et al.* [8] have proposed the 11.025 KHz to be fixed sampling rate for noise reduction of wheezing sound recording, and time-variant adjusting sampling rate for the optimal noise cancellation. By the way, the 8 KHz sampling rate of telephone and VoIP has been evaluated to record wheezing sound well. Therefore, the adaptive filter can be accelerated by reducing of sampling rate.

V. CONCLUSION

The adaptive filter is a signal processing algorithm in time domain. Therefore, it is very potential to develop to be a real-time noise cancellation device for the auscultation on the ambulances.

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