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Research Article

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Unidirectional Microphone based Wireless Recorder for the Respiration Sound

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Abstract

The wireless recorder for heart and lung sounds is intuitively convenient to the medical doctors and the patients for its portability which makes the auscultation be possible in a certain range. Such a recorder facilitates increased patient movement (postures and exercises) during diagnosis. Therefore, a wireless communication system for lung sound recording has been developed in this study. Two unidirectional microphones (SM35 and Beta 98H/C, Shure), a wireless body pack transmitter (PGXD1, Shure), and a wireless receiver (PGXD4, Shure) were used in our recording system. The detected sound was transmitted to a mixing console (MG06, Yamaha) for amplification. The system characteristics are as follows: (1) offers excellent low-frequency response, (2) has unidirectional microphone sensing, (3) blocks noise from the environment, (4) records sounds of moving and exercising people, and (5) records other physiological sounds. All of SNREN>35 in tests under the environmental noises in the range of 25 to 40 dB encouraged us to neglect the influences of the environmental noise. The findings strongly recommend that the proposed recording system be adopted by groups researching lung sounds because it reduces the number of complex devices and algorithms necessary for noise reduction, offers a high SNR, and facilitates device use in various environments. Furthermore, the developed system can be used to record lung sounds because of its excellent low-frequency response, especially, using the instrumentation microphone.

Keywords: Directivity; Lung sound; Microphone; Record system; Respiration

Introduction

The characteristics of the microphone strongly influence the quality of the recorded lung sounds. The most popular microphones which the research groups employed in lung sound studies were Sony™ ECM-77B omnidirectional electret condenser microphone [1-7] and Sony™ ECM- T140 [8-13], and T150 electret condenser microphones with air coupler [14-18]. Observing the time sequence of the cited studies [1-18], the findings were that the applications of linguistics and voice recognition at first, and then the studies related to the lung sounds were continuously employed the Sony™ ECM-77B, T140, and T150 omnidirectional microphones. Was the adoption of microphones such as Sony™ ECM-77B, T140, and T150 evaluated by the paper survey or the considerations of the lung sound characteristics? This is a valuable question to be discussed.

Many previous studies have recorded sound data by using single-wired and omnidirectional electret condenser microphones. However, user activities are limited when using wired microphones, and microphone directivity causes all sounds around the microphone to be captured. Shaharum et al. [19] reviewed the reliability of sound obtained from the trachea using the wheeze data collection method in studies reported between 1985 and 2009. Wheeze is the most commonly used indicator of airway obstruction, such as asthma and chronic obstructive pulmonary disease; these obstructions are usually detected by recognizing wheezes through lung auscultation. All auscultation systems in their review were wired microphone systems.

Because the aforementioned microphones are usually used in the vocal recording of TV stations, this study proposes individual wireless and unidirectional microphones for recording sounds; the proposed microphones support various user activities and reduce the influence of environmental noises. Consequently, a recording system by using commercial products has been assembled for ensuring high measurement precision. The wireless recorder for lung sounds is intuitively convenient to the medical doctors and the patients for its portability which makes the auscultation be possible in a certain range. Such a recorder facilitates increased patient movement (postures and exercises) during diagnosis. Lu and Wu [20] proposed a real-time mobile-based auscultation (RMA) system for distant auscultation. Based on the experience of the distant auscultation, the system in this study can be pushed to realize the low noise distant auscultation.

The bandwidths of the ECM-T140 and T150 are 50–15 kHz; these microphones also offer bad low-frequency (50–100 Hz) responses. However, the vocal and instrumentation microphones have the different frequency responses. Generally, lower frequency response of the instrumentation microphone is usually enhanced. The enhanced lower frequency response is good to the record of lung sounds. Therefore, this study employed Shure Beta 98H/C instrumentation (20 to 20 KHz) and SM35 vocal (40 to 20 KHz) unidirectional microphones, and the comparison of using the vocal and instrumentation microphones for lung sound recording was examined. The results were helpful to the selection of the types of microphones for the clinic applications.

The evaluation of the applications of the frequently used microphones was presented in the section of introduction. The system architecture is described in the methods section, and the testing results are presented in the results section. In the discussion section, the examination of the use of the directional microphone is detailed. The study is summarized the contributions in the conclusion section.

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Method

System architecture

Figure 1 illustrates the recording system. The functional block diagram is presented in Figure 1a. Two unidirectional microphones (SM35 and Beta 98H/C, Shure, USA) were used in the system. The SM35 headset condenser microphone offers higher sound quality than conventional microphones. The pickup pattern that presented tight and unidirectional rejects off-axis sound sources to prevent the echoes and noises. A wireless bodypack transmitter (PGXD1, Shure) and receiver (PGXD4, Shure) were used in the assembled system. The PGXD14 wireless system delivers clear sounds and a strong, clean RF signal. The detected sound was transmitted to a mixing console (MG-06, Yamaha, Tokyo, Japan), which used three-stage cascaded amplifiers with individual adjustable gains. The amplified sound was transmitted to a personal computer (Acer, Taiwan; Intel Core2, Quad CPU Q8200 @2.33 GHz, 1 GB RAM, and MS Windows XP OS) with an on-board sound blaster. The default recording program in Windows XP was used to record the audio output by the sound blaster. Simultaneously, the amplified sound was transmitted to the plug-and-play speakers (Pebbles, JBL, USA) for real-time auscultation. The specifications of the recording system are summarized in Table 1.

Unidirectional and omnidirectional microphones

The intensity distributions of the signals received from the unidirectional and omnidirectional microphones were simulated using Matlab (Mathwork), and presented in Figure 1b and c. The simulations revealed that the unidirectional microphone effectively reduces environmental noises. In addition, the acoustic conditions were enhanced and restrictions associated with the recoding space were ameliorated. Although most research groups use omnidirectional microphones, the unidirectional microphones were strongly suggested for the subsequent tests.

Beta 98h/c instrumentation microphone versus sm35 vocal microphone

Both Beta 98H/C and SM35 are unidirectional; however, their detection ranges differ. Instrumentation microphones are placed close to the source, whereas vocal microphones are placed farther. At extremely close distances, the low-frequency responses are enhanced by both microphones, an extremely beneficial feature for recording lung sounds.

Environmental noise measurement

Noise influences the quality of the recorded lung sounds. Quiet environments were strictly maintained in previous studies. However, such an environment cannot be replicated in operating rooms [21]. In this study, environmental noise levels were recorded using "Noise Detector," a free Android application. The two unidirectional microphones, Beta 98H/C and SM35, are depicted in the left and right panels of Figure 1d, respectively. The microphones were placed in the same location in Lab. of Applied Electronics, Department of Electronic Engineering, Tungnan University, Taiwan (where all experiments were performed) to measure environmental noises. The ordinary auscultations are usually in the room with random noise conditions, therefore, the noise in Lab. of Applied Electronics meets the considerations of the ordinary auscultation. A comparison of the noises captured by the two microphones revealed their respective noise levels.

Signal-to-noise ratio

The most fundamental measure of noise components in signals is the signal-to-noise ratio (SNR) [22]. SNR can be expressed mathematically in decibels as

$$\frac{S}{N}(in\ dB) = 10\log_{10}\frac{P_s}{P_N} \tag{1}$$

where P_S and P_N denote the power of the signal and noise, respectively.

Alternatively,

$$\frac{S}{N}(in\ dB) = 20\log_{10}\frac{V_s}{V_N} \tag{2}$$

where VS and VN denote the voltage of the signal and noise, respectively. In this study, SNR was used to evaluate the performance of the recording system.

Spectrogram

Spectrograms are convenient tools for speech recognition [23] and can be employed to analyze respiratory acoustic signals [24]. An element at time τ in a spectrogram is defined as follows:

$$\left|X_\tau(j\omega)\right|^2 = \left|\int_{-\infty}^{\infty} x(t)\varnothing_{\tau,\omega}(t)dt\right|^2 \tag{3}$$

where $x(t)$ is the signal in the time domain, $\varnothing_{\hat{o},\grave{u}}(t)$ is the complex basis function, and $|X_\tau(j\omega)|$^2 is the power distribution in the frequency domain at time τ. This form of the Fourier transform, also called the short-time Fourier transform, has several applications in speech, sonar, and radar processing. The spectrogram of a sequence is the magnitude of the time-dependent Fourier transform versus time.

Results

System configurations

The wireless transmitter (PGDX1) had an audio gain adjustment of 26 dB. The transmitter gain and the distance between the transmitter and receiver were maintained constant for all experiments. Therefore, the gain in the wireless communication system was constant. The receiver (PGDX4) gain was constant and nonadjustable. The gain and frequency responses of the mixing console (MG-06) were adjusted using three and two knobs (high and low frequencies), respectively. The gains in the two microphones were maintained at adequate levels to avoid amplification weakness or saturation [25]. In addition, the frequency responses for the lung sounds were regulated because enhanced low-frequency detection results in enhanced recording; however, a reduced low-frequency response is sufficient for lung sound recording. The sound quality was evaluated at different configurations (Table 2). Configurations 1 and 2 (for lung sounds) were recorded by Beta 98 H/C and configurations 3 and 4 (for heart sounds) were recorded by SM35. For all four configurations, the gains were set to be 42 and 10 for the first and second stages of amplification, respectively. For the third stage, the gains were set to 9, 6, 10, and 10 for configurations 1, 2, 3, and 4, respectively. For configurations 1 and 3, attenuations in the high- and low-frequency responses were both −15 dB. For configurations 2 and 4 no attenuation was adjusted in these responses.

Measurements of the environmental noises in the 4 configurations

Figure 2 illustrates the environmental noises in the four tested

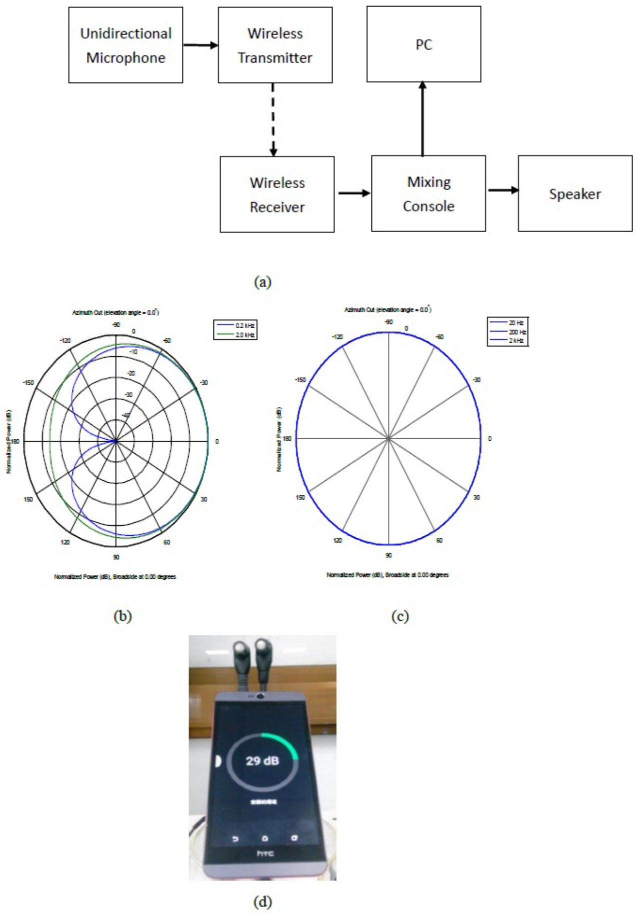

Figure 1: Recording system design: (a) Functional block diagram of wireless recording system for lung sound; (b) An demonstration of the receiving normal power distribution of unidirectional microphone (in dB); (c) An demonstration of the receiving normal power distribution of omnidirectional microphone (in dB); (d) the measurement of the environmental noises using Shure BETA 98H/C, and SM 35.

Component	Bandwidth	Features
BETA 98H/C miniature cardioid condenser instrument microphone (SHURE, USA)	20 Hz to 20 KHz	• Premier live performance microphone with Shure quality, ruggedness, and reliability • Uniform supercardioid pick-up pattern for maximum gain before feedback and superior rejection of off-axis sound
SM 35 headset cardioid condenser microphone (SHURE, USA)	40 Hz to 20 KHz	• Wearable wireframe headset fits securely and comfortably for active performers and multi-instrumentalists • Tight, unidirectional (cardioid) condenser pickup pattern rejects signal bleed and feedback for use on loud stages and behind floor monitors
PGXD1 bodypack wireless digital transmitter (SHURE, USA)	adequate	• The bodypack has 26 dB of audio gain adjustment.
PGXD4 wireless digital receiver (SHURE, USA)	adequate	• Professional quality 24-bit digital audio • Digital RF technology for rock-solid performance
MG06 mixing console, (YAMAHA, Japan)	adequate	•6-Channel Mixing Console •Max. 2 Mic / 6 Line Inputs (2 mono + 2 stereo) •+48V phantom power
Personal Computer with on-board sound blaster (Acer, Taiwan)	adequate	•Sound blaster: Realtech High Definition Audio •Intel Core2, Quad CPU Q8200 @2.33 GHz, 1 GB RAM, and MS Windows XP OS
Pebbles, plug and play computer speaker, (JBL, USA)	70 Hz to 20 KHz	•Easy, USB Plug and Play •Amazing, best-in-class stereo sound •Aux-in port for MP3 and mobile devices

Table 1: The components of the recording system.

configurations. The noises in panels a–d are those recorded in configurations 1-4, respectively. Every recording data was sampled at 8 KHz within 60 seconds. The maximal absolute values, averages (M), and standard deviations (SD) of panels (a)–(d) in Figure 2 were

Microphone	Target	Tuning knobs of mixing console					
		Configuration #	Gain 1	Gain 2	Gain 3	Higher frequency (dB)	Lower Frequency (dB)
Shure BETA 98H/C	Lung sound	1	42	10	9	-15	-15
	Heart sound	2	42	10	6	0	0
Shure SM 35	Lung sound	3	42	10	10	-15	-15
	Heart sound	4	42	10	10	0	0

Table 2: The 4 configuration settings.

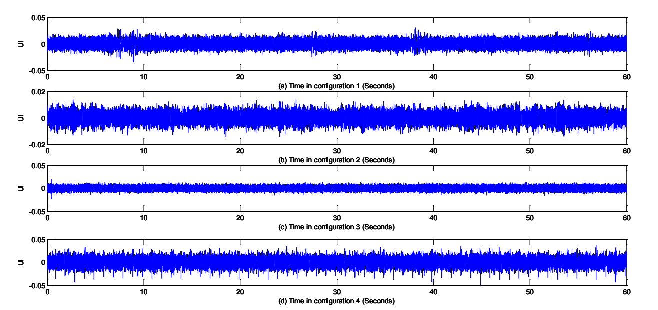

Figure 2: The environmental noises measured by the 4 configurations.

computed and are listed in Table 3. Let a set be defined as elements of C = {configuration # | 1, 2, 3, 4}; then, the following sets describe the results: Mx = {Maximum of absolute value (UI)| 0.0346, 0.0145, 0.0232, 0.0499} and MSD = {M ± S. D. (UI)| −4.0164·10^(-5) ± 0.0055, −3.4771·10^(-5) ± 0.0034, −3.7354·10^(-5) ± 0.0031, −3.4222·10^(-5) ± 0.0079}. The set, Mx, represents the sudden bursts in the signals, the probability of which is extremely low. The optimal values are in MSD. The averages in MSD are close to zero, which meets the concept of random noises. Therefore, SD becomes the basis for evaluating SNR. Furthermore, the normal probability plots indicate that the probability distributions all approach normal distribution, thus ensuring two-side p <0.05 in Figure 3. The normal probability plot is a function of various statistical analysis tools, such as Matlab, SPSS, and R programming and is used to visualize the correlation of the probability distribution of the analyzed data and its best fitting for normal distribution. These plots support the use of MSD for evaluating noise in this study.

Figure 4 presents the spectrograms of the noises recorded using the four configurations. All spectrograms exhibited harmonics of the fundamental frequency at approximately 500 Hz; this was caused by noise from an air conditioner. Physicians usually diagnose in an air-conditioned clinic room. Therefore, the actual environment of clinical auscultations were replicated in our spectrum analysis. High-frequency components (>3 KHz) were in the range −140 to −100 dB in panels (a) and (b) of Beta 98H/C but approximately −100 dB in (c) and (d) of SM35. Furthermore, configurations 1 and 3, whose spectrograms are respectively presented in (a) and (c), are recommended for lung sound recording. Compared with (b) and (d) for heart sound recording, the background noises from the environment were higher in (a) and (c) for that of lung sounds.

Configuration type	Maximum of absolute value (UI)	Mean (UI)	Standard deviation (UI)	P value
Configuration 1	0.0346	$-4.0164 \cdot 10^{-5}$	0.0055	0.05
Configuration 2	0.0145	$-3.4771 \cdot 10^{-5}$	0.0034	0.05
Configuration 3	0.0232	$-3.7354 \cdot 10^{-5}$	0.0031	0.05
Configuration 4	0.0499	$-3.4222 \cdot 10^{-5}$	0.0079	0.05

Table 3: The maximum of the absolute values, averages, and standard deviations of the environmental noises using the 4 configurations.

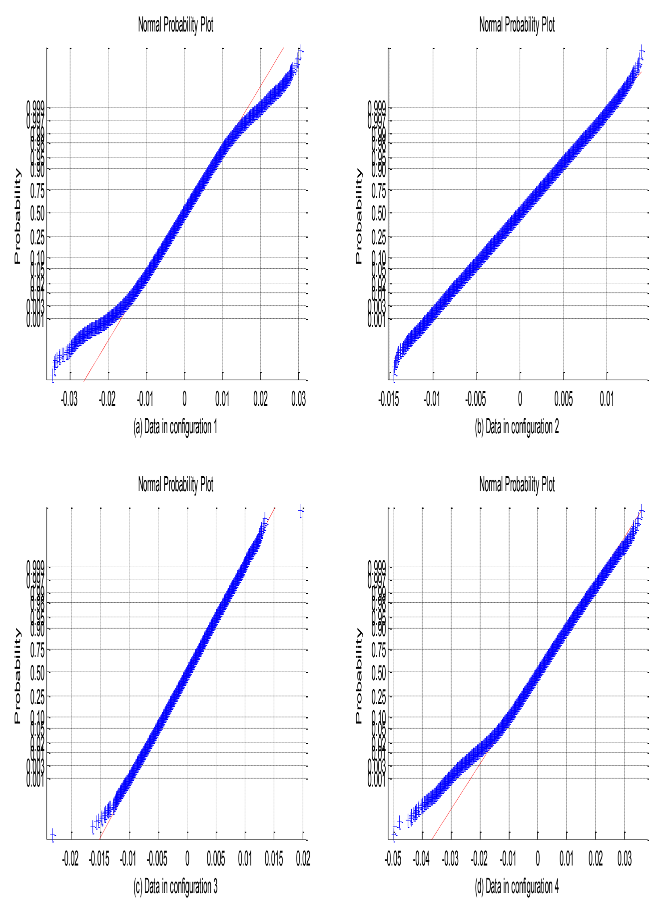

Figure 3: The normal probability plots of the noises recorded in (a) configuration 1; (b) configuration 2; (c) configuration 3; and (d) configuration 4.

Lung Sound

In Figure 5a and b illustrate the forced inspiration and expiration sounds recorded in configurations 1 and 3, respectively, and (c) and (d) illustrate quiet inspiration and expiration sounds recorded in configurations 1 and 3, respectively. This figure shows the lung sound–recording performance of the proposed system. SNR is a measurement airway during forced inspiration and expiration. The SNR_{EN} values in Table 4 are all higher than 35 dB, demonstrating that our system can be employed in a slightly noisy environment (25–40 dB; Figure 3). For SNR_{SL}, the silent level included the heart sound; therefore, the SNR_{SL} was 5–15 dB. Thus, the SM35 vocal microphone (Shure) performed better than the 98H/C instrumentation microphone (Shure).

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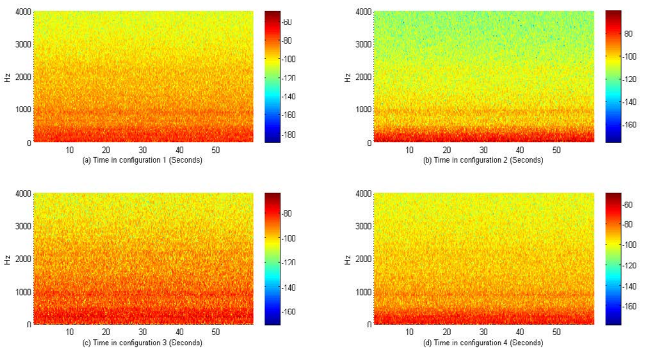

Figure 4: The spectrograms of the noises recorded in (a) configuration 1; (b) configuration 2; (c) configuration 3; and (d) configuration 4.

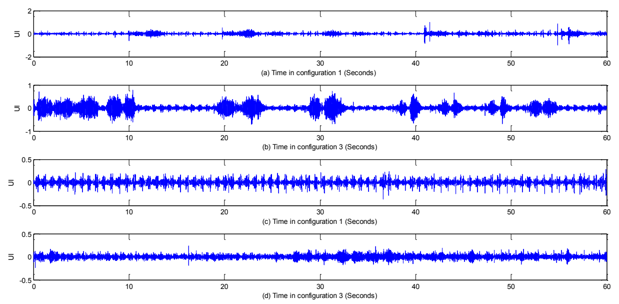

Figure 5: Lung sounds: (a) and (b) were forced inspiration and expiration sounds recorded in configurations 1, and 3, respectively; (a) and (b) were forced inspiration and expiration sounds recorded in configurations 1, and 3, respectively.

The spectrograms in Figure 5 are plotted in Figure 6. Panels (a) and (b) are forced inspiration and expiration sounds, and (c) and (d) are quiet inspiration and expiration sounds, respectively. The spectrograms (a) and (b) clearly present the information of the forced inspiration and expiration: the spectrums of the respiration sounds are much stronger than those of the heart sounds. However, in (c) and (d), the spectrums of the respiration sounds are not much stronger than those of the heart sounds. The background noises, which were lower than 500 Hz, presented low SNRs. The spectrums presented in (d) were more satisfactory than those in (c). Therefore, SM35 is optimal for recording for quiet inspiration and expiration, and Beta 98H/C is optimal for recording forced inspiration and expiration (Figure 7).

Discussion

Historic trace and reviews of the original researches employed the omnidirectional microphones

Google Scholar, one of the great database of knowledge, was employed to trace the original researches related to Sony™ ECM-77B, T140, and T150 omnidirectional microphones. The searching results were presented in Table 5 which indicated that Vieira et al. used the Sony™ T-140 to build a database for vocal fold pathologies in 1995 [26]. They are the first research group to use the Sony™ ECM-77B, T140, and T150 omnidirectional microphones for their studies. Yuchang et al. have studied the speech enhancement processing using the Sony™ ECM-77B microphone array in 1996 [27]. The vocal and speech studies [28-31] have kept employing the omnidirectional microphones as a main equipment for their studies during 1997 to 1999. In fact, Sony™ ECM-77B, T140, and T150 omnidirectional microphones are the most popular microphones for the vocal and speech researches now. However, in 1997, Kompis et al. presented the distribution of inspiratory and expiratory respiratory sound intensity on the surface of the human thorax using Sony™ ECM T-150 [32]. The research is the first study employs the Sony™ ECM-77B, T140, and T150 omnidirectional microphones to record the respiration sounds. In 1999, Kiyokawa et al. have employed the Sony™ ECM T-150 to detection of nocturnal wheezing in bronchial asthma [33].

Seriously, the acoustic sources of vocal, speech, and respiration sounds are quite different. Vocal and speech are the active associations of tongue, lips, jaw, vocal cords, and other speech organs of the human. However, respiratory sounds are the specific sounds generated by the movement of air through the respiratory system. The SNRs of vocal, speech, and respiration sounds are generally in the ranges of 60, 50, and 3 dB, respectively. Therefore, the employment of the unidirectional microphones for the respiration sounds is worth discussing.

SNR

In a communication system, the frequency response of an amplifier is defined as the range between the half power point of the lower frequency and that of the higher frequency, meaning that power decays to half the maximal power in the central frequency at the half power point (i.e., the −3 dB point) [34]. Similarly, considering acoustic signal power, $SNR_{SL} > 3$ can be regarded a clear auscultation of lung sounds. The performance of the four configurations is presented in Table 4. Because all $SNRs_{SL}$ is higher than 3, the system is suitable for lung sound recording. Furthermore, because all SNR_{EN} is higher than 35, the influence of the environmental noises can be considered negligible. The effectiveness of the adaptive directional processing for enhanced speech recognition compared with that of the non adaptive directional and omnidirectional processing was examined across the four listening environments, which aimed to simulate the real-world environments [35]. The results indicated that in all four listening conditions, adaptive and directional processing yielded a more enhanced speech recognition than did non adaptive and omnidirectional processing. Hersbach et al. [36] reported that microphone directionality elevates the performance of speech recognition in all tested noise conditions. The average speech reception threshold benefit compared with that for the standard setting was 3.7 dB for zoom and 5.3 dB for beam. Accordingly, the $SNR_{SL} > 3$ was proposed to be a criteria of the clear auscultation of lung sound.

Unidirectional microphone

The unidirectional microphones employed to detect lung sounds offered $SNR_{EN} > 35$ and considerably reduced the influence of the environmental noises which encouraged the usage of unidirectional microphones in lung sound recording systems. Speech recognition studies have revealed that a microphone array–based system has several advantages over a single microphone. For instance, a microphone array–based system which can be improved the directivity by the superposition of receiving fields of the elements in the array may be electronically aimed to capture an audio signal from a desired source location and simultaneously attenuate environmental noises. In addition, it can be used to localize a nearby active speaker, allowing computer controlled devices to provide a speaker location–aware user interface. The proposed lung sound recording system may be enhanced by using a microphone array. Briefly speaking, the effect of directivity of the microphone arrays are caused from the superposition of the sensing fields of the sound. Therefore, the performance of the microphone array is much approaching to that of unidirectional microphone. Consequently, these previous studies were obviously supported our

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Respiration	Configuration type	Maximum of absolute value (UI)	Noise (UI)		SNR (dB)	
			Figure 4	Figure 7	SNR_{EN}	SNR_{SL}
Forced inspiration and expiration	Configuration 1	1.0042	0.0055	0.2507	45.23	12.05
	Configuration 3	0.7760	0.0031	0.1406	47.97	14.84
Quiet inspiration and expiration	Configuration 1	0.3598	0.0055	0.1881	36.31	5.63
	Configuration 3	0.2351	0.0031	0.07622	37.60	9.78

Table 4: SNR_{EN}, and SNR_{SL} of lung sound measurement.

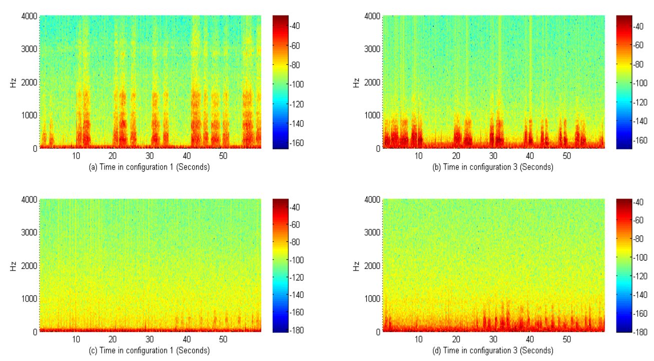

Figure 6: The spectrums of Figure 6: (a) and (b) were forced inspiration and expiration sounds. (c) and (d) were quiet inspiration and expiration sounds.

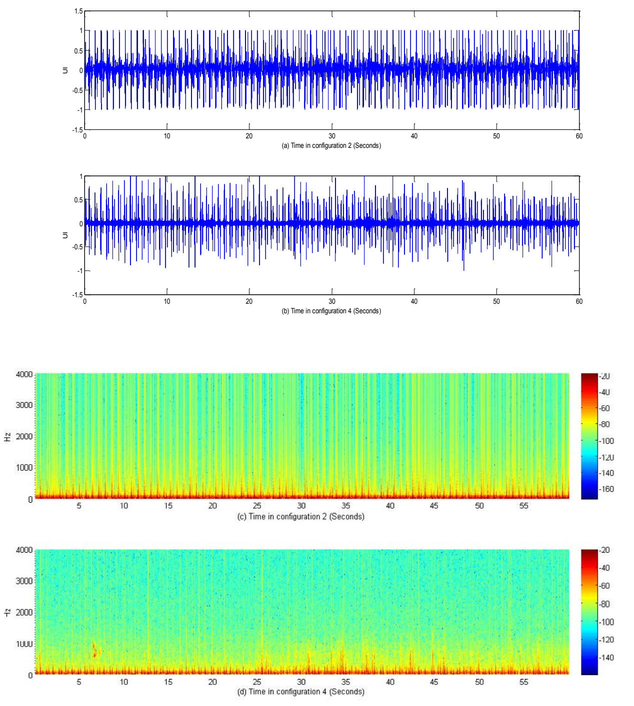

Figure 7: The acoustic signals of heart sounds: (a) recorded with configuration 2 in time domain; (b) recorded with configuration 4 in time domain; (c) spectrogram of (a); and (d) spectrogram of (a).

proposal of employing the unidirectional microphone in the lung sound recording system. Reversely, that the studies related to speech recognition use the unidirectional microphone to receive the acoustic signals is expected.

Bandwidth

This recording system integrates a wireless communication system (Shure) and a mixing console (Yamaha). Beta 98 H/C (bandwidth 20 Hz to 20 kHz) and SM35 (40 Hz to 20 kHz) microphones were employed in the recording system. The proposed system offers higher low-frequency responses than do other recording systems. For comparison [37], the bandwidths of Siemens EMT 25 C accelerometer (Siemens), PPG 201 accelerometer (PPG), Sony ECM-T150 electret condenser microphone with an air coupler (with cylindrical air chambers of 5-, 10-, and 15-mm diameter and conical air chamber of 10-mm diameter), Littman classic stethoscope head (Littman) connected to an electret condenser microphone, and Andries Tek (Andries) electronic stethoscope were evaluated. The Siemens, air coupler, and Littman devices exhibited near-identical performance, with relatively flat frequency responses from 200 to 1200 Hz. The PPG had the broadest frequency response, with useful sensitivity extending to 4000 Hz. In conclusion, the excellent lower frequency response is the crucial feature of the wireless recorder in this study.

Practicability of the wireless communication system

The proposed system is a wireless communication system; thus, it provides possibility patient movement during diagnosis. The excellent low-frequency response is conducive for detecting the lung sound of an individual who is exercising or moving. Furthermore, the application of this recording system in an ambulance can be explored in a future study to reduce the interference of ambulance noises [38-42]. Lu et al. proposed that adaptive filter was a signal processing algorithm in time domain to be developed as a real-time noise cancellation device for the auscultation on the ambulances. The proposed adaptive filter algorithm was proven by that the results showed the harmonics of the audible warning were completely cancelled, and the component of fundamental frequency of that was reduced from -30 dB to -60 dB, and was a feasible method for the auscultation on the ambulances. The further experiments using this unidirectional microphone as a hardware improvement and the aided algorithm of adaptive filter in the study [42] are much expected.

Instrumentation and vocal microphone for lung sound recording

The results of Fig. 4 disclosed that the instrumentation microphone owns the better performance than the vocal microphone's for the lung sound recording. The actual environment of clinical auscultations were replicated in our spectrum analysis. High-frequency components (>3 KHz) were in the range −140 to −100 dB in panels (a) and (b) of Beta 98H/C but approximately −100 dB in (c) and (d) of SM35. This results suggested to the previous studies of using vocal microphone to achieve the better SNR by using the instrumentation microphone. Certainly, the unidirectional microphone is crucial to improve the SNR.

Conclusion

A wireless communication system for lung sound recording was developed. The system has the following features: (1) offers excellent low-frequency response, (2) has unidirectional microphone sensing,

Year of publication	Type of Microphone	The topic of the research
1995	Sony ECM-T140	Methodological aspects in a multimedia database of vocal fold pathologies [40]
1996	Sony ECM-77B	Speech enhancement using microphone array with multi-stage processing [37]
1997	Sony ECM-T150	A validity study of an implicit phonological awareness paradigm [35]
1997	Sony ECM-T150	Distribution of inspiratory and expiratory respiratory sound intensity on the surface of the human thorax [42]
1998	Sony ECM-77B	Noise robust speech recognition using subband-crosscorrelation analysis [38]
1998	Sony ECM-T140	The effects of speaking rate on listener evaluations of native and foreign-accented speech [39]
1998	Sony ECM-T140	The effects of noise on the intelligibility of foreign-accented speech [41]
1998	Sony ECM-T150	Melon ripeness monitoring by a portable firmness tester [16]
1999	Sony ECM-T150	Detection of nocturnal wheezing in bronchial asthma using intermittent sleep tracheal sounds recording [34]

Table 5: The historic trace and review of the original researches employed the SonyTM ECM-77B, T140, and T150 omnidirectional microphones during 1995 to 1999.

(3) blocks noise from the environment, (4) records sounds of moving and exercising people, and (5) records other physiological sounds. Therefore, the results of this study strongly recommend that the proposed recording system be adopted by groups researching lung sounds because it reduces the number of complex devices and algorithms necessary for noise reduction, offers a high SNR, and facilitates device use in various environments. Furthermore, commercially available components were used in this system, thus ensuring the quality of the recorded acoustic signals. This conceptual evolution of recording systems enhances the performance of lung sound studies. Some studies have proposed the use of microphone arrays for enhancing sound quality and the use of filtering devices and algorithms for enhancing lung sound response; these techniques can be considered in future studies to further enhance the quality of the proposed system. Furthermore, the developed system can be used to record heart sounds because of its excellent low-frequency response, especially, using the instrumentation microphone; this application will be comprehensively explored in a future study.

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