International Journal of Innovative Studies in Medical Sciences (IJISMS)

ISSN 2457-063X (Online) www.ijisms.com Volume: 3 Issue: 5 | 2019

Spectrum Distortion of a Directional Microphone and its Removal for Hearing

Xubao Zhang

Engineer, R & D, Sonova Unitron; Oticon Canada;

Associate Professor, EE Department, Xi'an Electronic Science and Technology University, China

Abstract: *We illustrate extensive experimental data, graphs and related statistics to prove spectrum distortion and waveform damage of a directional microphone(DM) with speech signals. We researched a solution for this problem: processing with a multiband filter bank plus magnifiers and an adder. The multiband filter bank is of three types: eight equal bandwidth(BW) FIR filters, eight logarithmic BW FIR filters and sixteen equal BW FIR filters. These magnifiers were assigned to balance the sloped-frequency response of the DM. Accordingly, the designed balanced DMs also have three types. Following observations were obtained in the waveforms and spectra measurements at Meadow SL Lab. ① The balanced DMs of the three types can remove the spectrum distortion effectively; ② eight equal BW FIR filters, sixteen equal BW FIR filters and eight logarithmic BW FIR filters have delay times of 2.9 ms, 5.9 ms and 0.374~9.25 ms, respectively; ③ but the DM with logarithmic BW filters slightly much changes the waveform construct or naturalness of the speeches, we refer to the change as the time-delay distortion; ④ moreover, the balanced DMs perform reasonable S/N improvement: in competing traffic noise, a few dB better than Omni mic does; and in talking interference, over 15 dB better than Omni mic does.*

Keywords: *directional microphone, multiband processing, spectrum distortion, time-delay distortion*

1. Introduction

A directional microphone(DM) provides more benefits than other signal processing strategies in noise/ interference suppression, so hearing aid users and manufacturers are interested in it[1]. Flynn designed a voice-priority system with tri-pattern beamformer of parallel processing. Such an adaptive processor could make reliable decisions in a varying environment, and could implement the optimal DM mode[2]. Moeller et al. noted that the conventional DM was less sensitive in low-frequency region than Omi microphone(mic), so their strategy to counter this loss was to use band-split directionality, i.e., to activate Omni or DM processing in a channel lower or higher than the band-split frequency[3]. Their results showed that the test hearing aids enhanced SNR and preserved speech quality.

A conventional DM which is composed of a cardioid (or hyper-cardioid) DM and a multiband processor always causes spectrum distortion. Phonak presented a declined sensitivity response of a double-mic DM, i.e., 6 dB/octave down-slope from mid to low frequency regions[4]. That has implicitly shown us that the spectrum distortion exists. Zhang measured some waveforms and spectra of talking voices before and after the voices went through a conventional DM. The comparison of his measurements explicitly indicated the existence of the spectrum distortion[5]. The distortion removal and speech quality guard have not been popularly known. In practice, directional hearing aids have largely incorporated multiband magnifiers' processing to balance the sloped frequency response. However, there is no open reference to show how the spectrum distortion is removed and what the effectiveness is. By extensive experimental measurements, we intended to provide more waveforms and spectra with talking voices to show the distortion existence and speech quality decline when the voices went through a conventional DM. Then, we studied how to remove the distortion and to design the balanced DMs with the multiband magnifiers' processing. The effectiveness of the filter banks of different types, equal BW and logarithmic BW, to remove the distortion was a key concern. Furthermore, we conducted a simulating experiment to check performance of the balanced DMs to suppress the traffic noise and talking interference. Time–delay distortion was our additional finding during the experiment of the logarithmic BW filter bank.

2. A conventional DM and its S-gain characteristics

Fig.1 shows architecture of a cardioid or hyper-cardioid DM. In a real-world hearing aid, the conventional DM always combines such a DM with multiband processing so as to implement gain control in the frequency domain. In this figure, the solid arrows represent the 0° incidence(front), and the dashed arrows represent the non-zero degree incidence. AS everyone known, an essential specification of an Omni mic and a DM is the sensitivity curve, whose unit is dB re 1V/Pa, where Pa a unit of sound pressure[6]. To study DM's performance

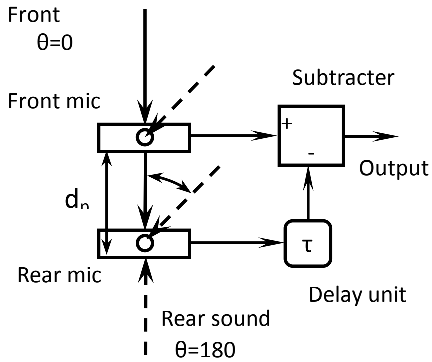


Fig.1 Basic architecture of a directional microphone

conveniently, we define DM sensitivity/Omni mic sensitivity as sensitivity(S)-gain of the DM, and its unit is dB only. Then, we can derive an equivalent gain equation for the DM: S-gain=DM output/Omni mic output. Our concern with a DM, as a beamformer, is its polar pattern. Assume that in a sound field, there exists a pure tone of frequency f; given delay time of the spacing of the two-mic ports is Δ, delay time between signals of the rear mic and the front mic is $\delta(\theta) = \Delta\cos(\theta)$, and delay unit parameter is τ. Based on these assumptions, the S-gain of the DM of Fig.1 is

$$g_{dm}(\theta,f)=2\sin[\pi f(\tau+\Delta\cos(\theta))] \qquad (1)$$

which is related to the frequency, ports spacing and incident angle. Fig.2 shows the S-gain polar patterns of the cardioid DM (τ=Δ) with three tones at frequencies 5k, 2k and 500 Hz. Each pattern has a zero notch at 180°. In pure tone measurement, we used 5k Hz to represent the high frequency region; 2k Hz, the mid frequency region; and 500 Hz, the low frequency region. The outer one in Fig.2 results from 5k Hz tone, having front gain 6 dB(max) at 0°; the inner one, from 500 Hz tone, having front gain -10.7 dB at 0°. The lower the frequency is, the less the gain is.

We calculated the gain response by means of (1) with the ports spacing 16 mm and incidence 0°. Fig.3 shows the S-gain frequency response of the cardioid DM and an Omni mic. We observe that the S-gain response of the cardioid DM has 0~6dB gain at the frequencies ≥1.78k Hz, and -18~0dB gain at the frequencies <1.78k Hz; the response of the Omni mic is a 0 dB horizontal line. This fact tells us that when the frequency is less than 1.78k Hz, speech enhancement of the DM is less than that of the Omni mic, and vice versa. From Fig.2 and Fig.3, we can conclude that ① the cardioid DM performs very strong suppression to interference/noise from the back side, especially

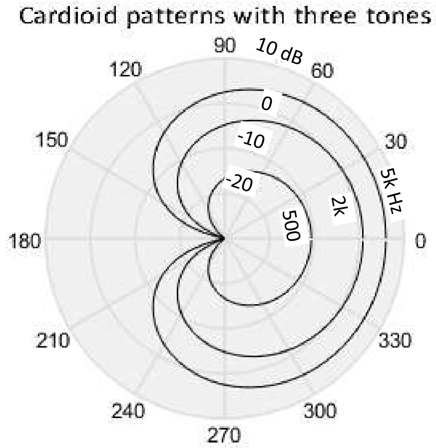


Fig. 2 Gain patterns of a cardioid DM with pure tones

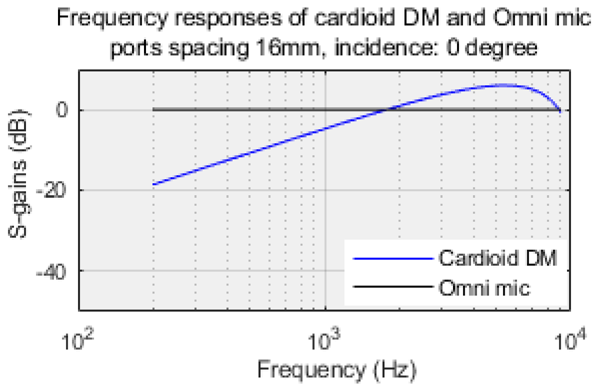


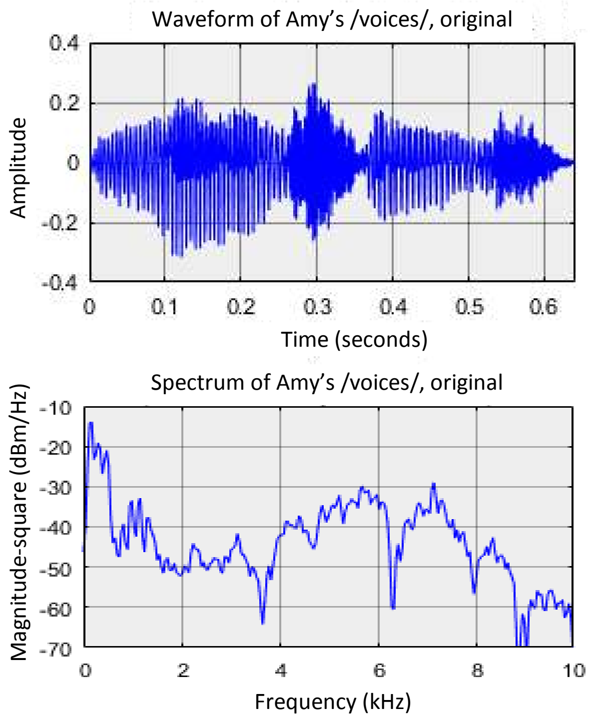
Fig.3 Frequency responses of a cardioid DM and Omni mic at incidence 0°

at 180° incidence; ② the cardioid DM has a sloped gain response, 6 dB/octave down-slope from mid to low frequency regions, and this must cause spectrum distortion of speech.

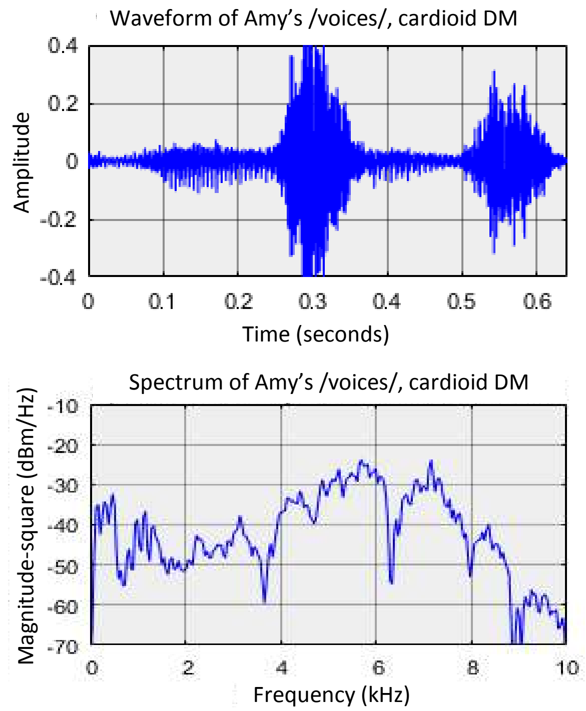
3. Spectrum distortion of a conventional DM

In Fig.3, we observe the S-gain frequency responses of the cardioid DM and Omni mic. How does the 6 dB down-slope response affect speech quality of the DM output when talking voices go through the cardioid DM? To answer this question, we selected real-world voices from wave files of Amy's and Brian's talking[7]. The talking "voices" contains four phones /voi/, /c/, /e/ and /s/, and the entire /voices/ lasts 0.641 s. Note that we use double slashes /a/ to represent the sound of a. We can view the waveforms by means of Adobe Sound Booth or Time Scope in SimuLink. For details of measuring the waveforms, refer to reference [9]. When /voices/ enters the cardioid DM, the waveform and spectrum of the front/rear mic outputs are shown in Fig.4 (a). Durations of the phones /voi/, /c/, /e/ and

/s/ are about 0.259, 0.109, 0.151 and 0.122 s, respectively. At the cardioid DM output, we measured and recorded the waveform and spectrum, as shown in Fig.4 (b).Comparing the waveforms in (b) to in (a), we observe that the levels of /voi/ and /e/ significantly decline, but the levels of /c/ and /s/ are significantly enhanced. In Fig.4 (b), the spectrum of the Amy's /voices/ declines significantly in the low and mid frequency regions, and rises obviously in the high frequency region. Table 1 lists peak-peak values and RMS(root-mean-square) of the phones /voi/, /c/, /e/ and /s/. We listened to sounds of the waveforms in Fig.4 (a) and (b) separately; the two sounds were much different, the pitch of the waveform in (b) got higher than that of the waveform in (a). This perception was consistent with the statistics in Table 1.



(a) Front/rear mics' outputs



(b) Cardioid DM output

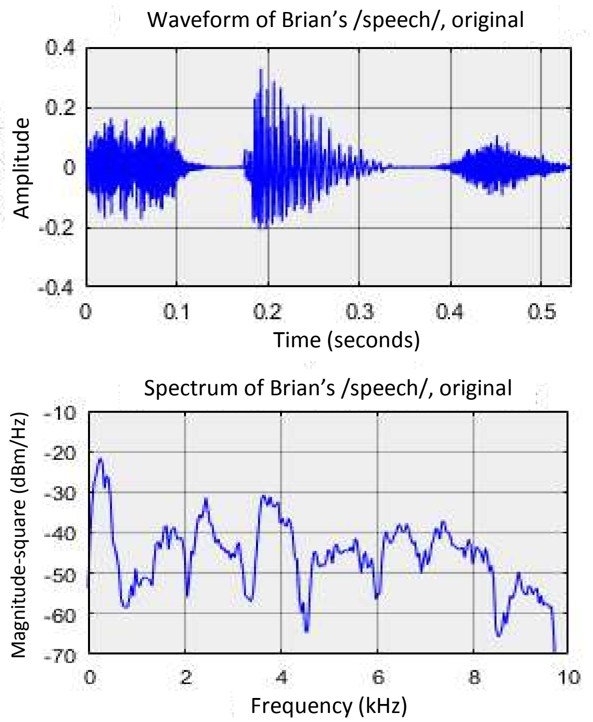
Fig. 4 Waveforms and spectrum of Amy's talking "voices" before and after going through a cardioid DM

Table 1 Peak-peak and RMS of phones of /voices/

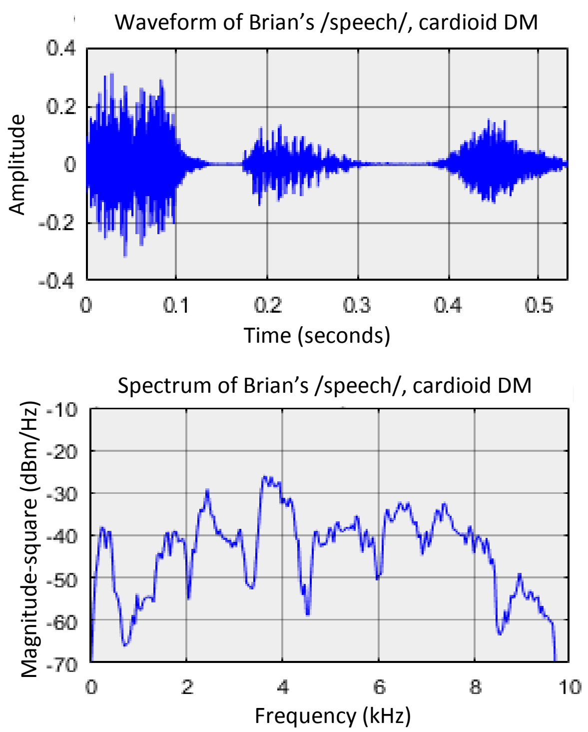
	phones	peak-peak (mv)	RMS (mv)
original Amy's /voices/	/voi/	532	104
	/c/	526	62.7
	/e/	369	76.3
	/s/	328	39
after going through cardioid DM	/voi/	232	21.8
	/c/	1007	116
	/e/	160	16.2
	/s/	632	70.5

Brian's talking "speech" lasts 0.533 s, and contains three phones /s/, /pee/ and /ch/. Fig.5 (a) shows the waveform and spectrum of the front/rear mics' outputs. When the entire /speech/ entered the DM, we measured durations of /s/, /pee/ and /ch/, which were about 0.132, 0.163 and 0.155 s, respectively. Note that we ignored the silent gaps between the phones. Fig.5 (b) shows the waveform and spectrum of the DM output. Comparing the waveforms in (b) to in (a), we can observe that level of each phone changes significantly: the /s/ and /ch/ are enhanced, but the /pee/ declines. The spectrum change is similar to that in Fig.4. Table 2 lists peak-peak values and RMS of these phones, the statistics and their change rule are similar to those in Table 1.

Through the cardioid DM, the voice waveforms change significantly, depending on the spectral components of each phone. When a phone contains more high frequency components, the DM gives it a higher gain; and vice versa. This behavior is consistent with the S-gain response in Fig.3, i.e., until >5.36k Hz, the higher the frequency of the tone is, the higher the output of the DM is. By conducting listening checks, we also perceived that the overall pitches of these voices became obviously high after processing with the DM. We refer to the change of speech pitch as the spectrum distortion caused by the DM. Evidently, the spectrum distortion cannot be ignored because it may damage the speech construct/naturalness, then affect the speech comprehensibility.



(a) Front/rear mics' outputs



(b) Cardioid DM output

Fig. 5 Waveforms and spectrum of Brian's talking "speech" before and after going through a cardioid DM

Table 2 Peak-peak and RMS of phones of /speech/

	phones	peak-peak (mv)	RMS (mv)
original Brian's /speech/	/s/	339	42.3
	/pee/	538	67.5
	/ch/	201	19.4
after going through cardioid DM	/s/	632	73.9
	/pee/	278	26.9
	/ch/	290	28.9

4. Spectrum distortion removal of a conventional DM

Although a conventional DM causes speech spectrum distortion, its strong suppression to the interference from the back side still attracts hearing researchers. To remove the spectrum distortion, balancing/flatting the frequency response is an applicable and effective way. A common hearing aid always contains multi-channel/band processing, which was used for noise reduction and feedback cancellation etc. When magnifiers are inserted between the multi-bands and a following adder, and the magnifiers provide the mid and low frequency bands with desired gains, overall frequency response of the modified DM can be balanced, and the distortion can be removed. Such a DM is called a balanced DM. The frequency bands, also called a band splitter, can have a few types of filter banks, e.g., equal BW FIR filters, logarithmic(octave) BW FIR filters and an FFT plus IFFT processor. We designed three types of multi-band filter banks: eight equal BW FIR filters' bank, eight logarithmic BW FIR filters' bank and sixteen equal BW FIR filters' bank at our Meadow SL Lab. The FFT plus IFFT processor[8] will be designed later. All the filter banks met the following basic requirements: ① frequency coverage 100~8k Hz, ② as much short delay time as possible, ③ as much low frequency response ripples as possible.

4.1 Eight equal BW filters' bank

Currently, SimuLink in MatLab provides extremely convenient function blocks to simulate various FIR, IIR and octave filters. Equal BW FIR filter bank is a common multiband splitter. To avoid long delay time, we selected the Chebyshev II, Direct form FIR in DSP Toolbox of Simulink[10]. Eight equal BW filters' bank is composed of eight equal BW FIR filters, easy to be implemented, and hence we named it EB8 type bank. For this filter bank, the filters each have BW 1k Hz, and their center frequencies are 500, 1.5k, ..., 7.5k Hz, respectively. Fig. 6 shows the frequency response of the EB8 bank. Its ripples are a little, within ±1.0 dB; the eight FIR filters have almost the same delay time of 2.9 ms. The less the response ripples are, the longer the delay times are. For details of the FIR filter design, refer to reference [9].

Magnifiers are assigned to meet requirements to balance the frequency response of a conventional DM. The gain of a band magnifier is equal to the desired gain(6 dB) minus the cardioid DM gain at the band

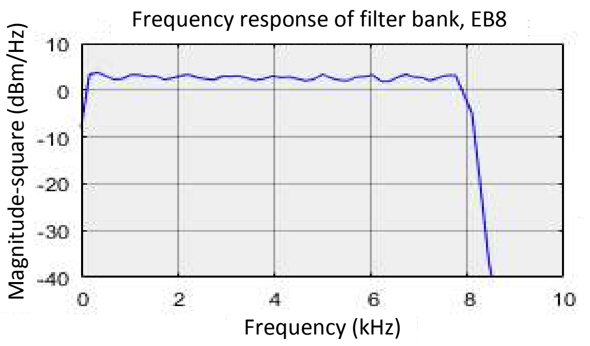


Fig.6 Frequency response of eight equal BW filters' bank

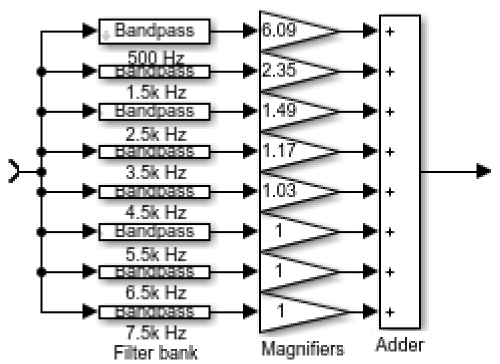


Fig.7 EB8 multiband filter bank plus magnifiers and adder

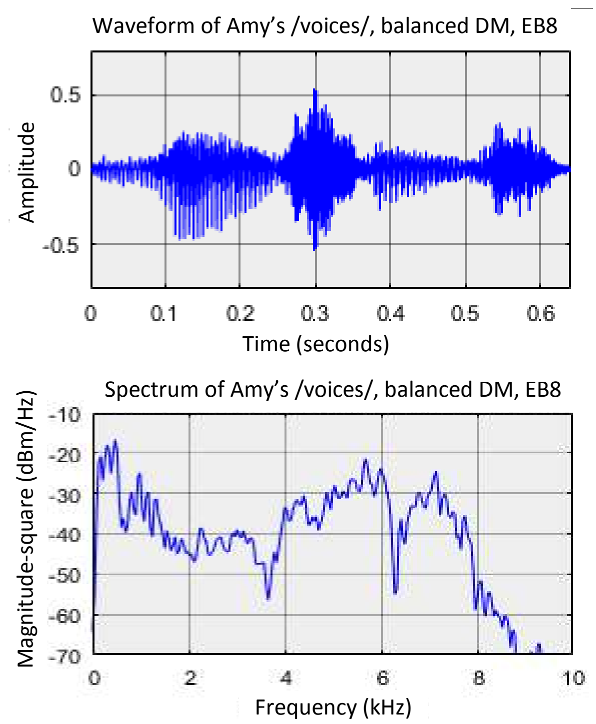


Fig.8 Waveform and spectrum of balanced DM EB8 output

frequency. As a result, the gains of the magnifiers are higher in the low frequency region than in the mid frequency region. Fig.7 shows a multiband filter bank connecting eight magnifiers, whose gains are 6.09, 2.35, 1.49, 1.17, 1.03, 1.0, 1.0 and 1.0, respectively. Considering the spectral characteristics of human speech and the specific hardware of practical design, the 1st band gain was adjusted to be 5 dB, and the last bands maintained the cardioid DM gain, and the other band gains, to be 6 dB. When the output of the cardioid DM of Fig.1 was connected to the input of the multiband filter bank plus the magnifiers and adder, we obtained a balanced DM. It contained the EB8 bank, and hence it was named the balanced DM of EB8 type. To confirm the spectrum distortion removal, we used the same talking voices in section 3 as the test source, and the measured waveform and spectrum of the DM output with Amy's /voices/ are shown in Fig.8. From this figure, we observe that the waveform and spectrum approximate to those of the original Amy's /voices/ in Fig.4 a). We also achieved another confirmation with Brian's /speech/, and obtained the same conclusion.

4.2 Eight logarithmic BW filters' bank

Logarithmic BW FIR filter bank is reasonable to achieve frequency domain processing because the human auditory sensitivity/resolution to audio frequencies follows logarithm rule[11]. So, we designed another FIR filter bank, which complies with 2/3 octave frequency band standard, center frequencies of the eight filters are 250, 400, 630, 1k, 1.6k, 2.5k, 4k and 6.3k Hz, respectively, and their band edgings are 200, 315, 500, 800, 1.25k, 2k, 3.15k, 5k and 8k. For differentiating from the EB8 type, we named this logarithmic BW bank as LB8 type. Given the large difference between these filter bandwidths, delay times caused by the filters are quite different, from 0.374 to 9.25 ms. Fig. 9 shows the frequency response of the LB8 bank, and its ripples are a little enough, within ±1.25 dB. From the cardioid DM's slope in Fig.3, we calculated gains of the magnifiers for the LB8 bank as 10.8, 7.61, 5.44, 3.45, 2.21, 1.49, 1.08 and 1.0. Fig.10 shows the multiband filter bank connecting eight magnifiers and an adder. A cardioid DM of Fig.1 was combined with these eight logarithmic BW filters plus the magnifiers and adder, we obtained a balanced DM of LB8 type. To confirm the spectrum distortion removal, we used Amy's /voices/ and Brian's /speech/ again. When /voices/ went through the LB8 balanced DM, we measured the waveform and spectrum of this DM output, as shown in Fig.11. Compared to Fig.4 (a), the waveform and spectrum of this DM with Amy's /voices/ approximate to those of Fig.4 (a), but the result of Fig.8 is less distorted than that of Fig.11. Another confirmation with Brian's /speech/ resulted in the same conclusion.

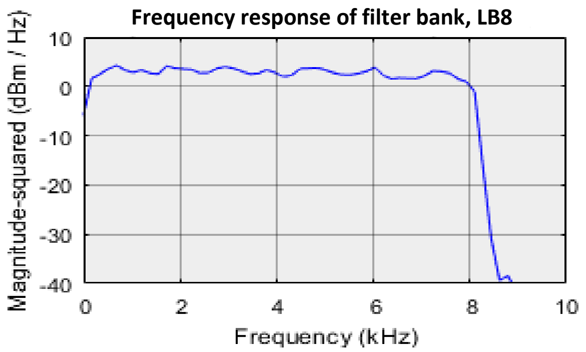


Fig. 9 Frequency response of eight logarithmic BW filters' bank

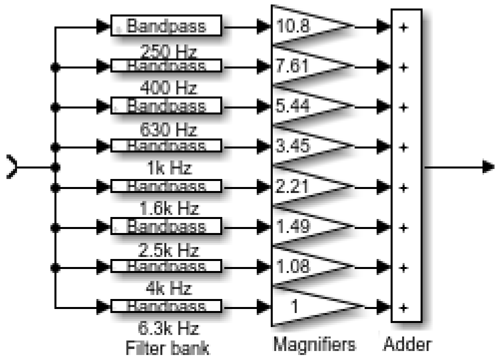


Fig.10 LB8 multiband filter bank plus magnifiers and adder

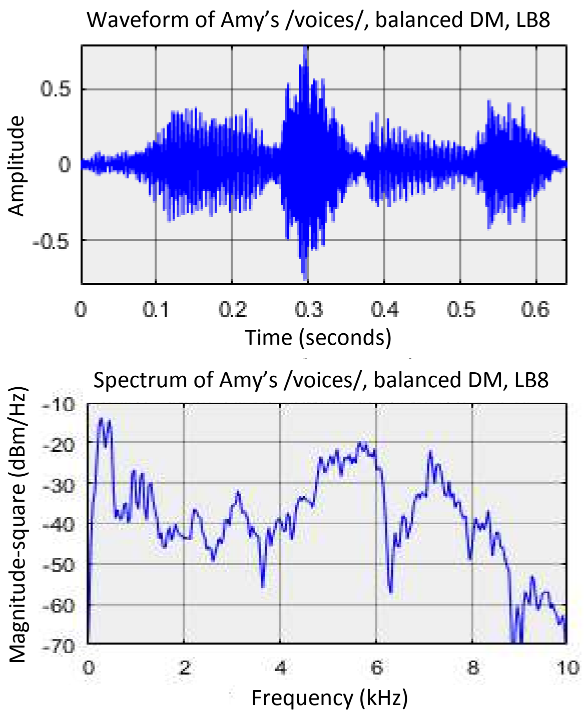


Fig.11 Waveform and spectrum of balanced DM LB8 output

4.3 Sixteen equal BW filters' bank

In order to achieve better spectrum distortion removal, we tried to use sixteen equal BW FIR filters' bank rather than EB8 type. Its design was exactly the same as in section 4.1, except parameters of the FIR filters and magnifiers. The filters each have BW 500 Hz, and their center frequencies are 250, 750, 1.25k, ..., 7.75k Hz, respectively. It characters sixteen equal BWs, so we named it EB16 type. Fig. 12 shows the frequency response of the filter bank EB16, its ripples are quite a little, within ±0.6 dB; all the FIR filters have almost the same delay time of 5.8 ms.

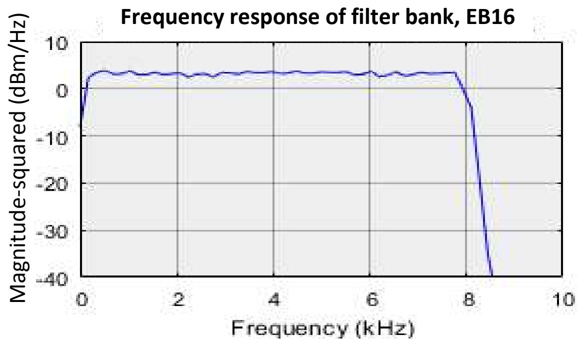


Fig.12 Frequency response of sixteen equal BW filters' bank

Gains of magnifiers were calculated as 12.1, 4.58, 2.79, 2.03, 1.63, 1.38, 1.22, 1.12, 1.05, 1.01, 1.0, 1.0, 1.0, 1.0, 1.0 and 1.0. The 1st magnifier gain was to meet 5 dB for its band, the last 6 bands maintained the cardioid DM's gain in the high frequency region, and all the other magnifier gains were to meet 6 dB for their bands. Fig.13 shows the EB16 multiband filter bank connecting sixteen magnifiers and an adder. To confirm improvement of this EB16 filter bank, we used the same test voices as in section 4.1; the waveform and spectrum of this balanced DM EB16 output were measured, as shown in Fig.14. From this figure, we observe that the waveform and spectrum with Amy's /voices/ are reconstructed better than those of the balanced DM EB8. Another confirmation with Brian's voice /speech/ resulted in the same conclusion.

In summary, the balanced DM EB8 removes the spectrum distortion well, with the same delay time of each band; the balanced DM LB8 removes the spectrum distortion not better than the EB8 does, with a quite different delay time of each band; the balanced DM EB16 removes the spectrum distortion better than the EB8 does, with an larger, the same delay time of each

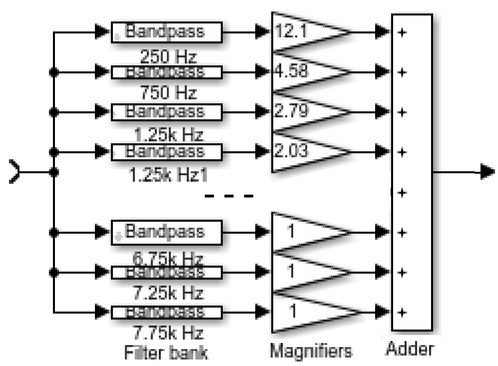


Fig.13 EB16 multiband filter bank plus magnifiers and adder

band.

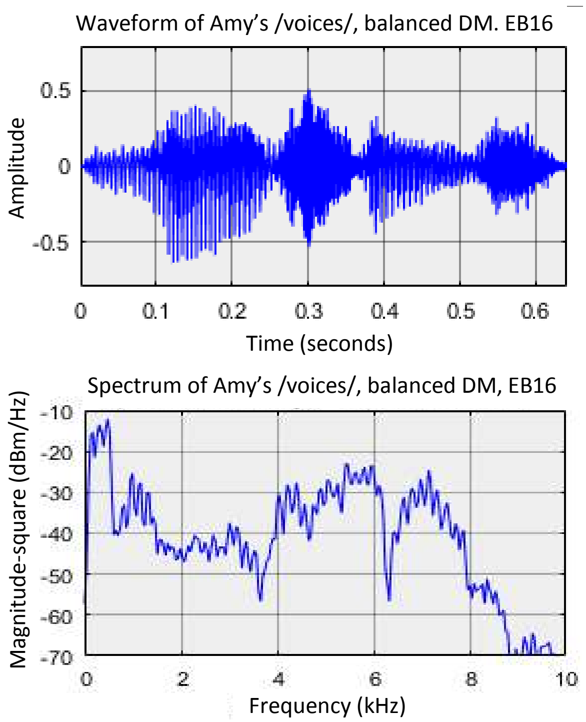


Fig.14 Waveform and spectrum of balanced DM EB16 output

5. Noise suppression of the balanced DMs

After removing the spectrum distortion using these balanced DMs, we needed to check their performance to suppress interference and noise, and to compare the effectiveness of the balanced DMs in this respect. We managed the test conditions: three balanced DMs were designed fairly, the hearing aid hardware was considered as much as possible; and an exactly balanced response of each DM is not necessary. We balanced the gains in mid and low frequency regions, maintained original gain of the cardioid DM in the high frequency region, and dropped 1~2 dB gain at the frequencies ≤300 Hz. The test audio signals had sampling rate of 44.1k Hz; the mic-port spacing was 16 mm, as in section 3.

Polar patterns of the balanced DMs of the three types EB8, LB8, EB16 were measured at Meadow SL Lab., as shown in Fig.15. We observe that unlike Fig.2, the three balanced DMs exhibit excellent directivity with three frequencies. At incidence 0°, they all have a 6 dB gain for speech enhancement, independent of the frequency; at incidence 180°, they all have a deep notch, also independent of the frequency, which can suppress interferences from back side very well. Theoretically, the gains of the three types' DMs are the same because their frequency responses all are balanced on the desired 6 dB. Practically, however, these responses may have differences to a certain extent. Fig.16 shows the frequency responses of the three balanced DMs. We

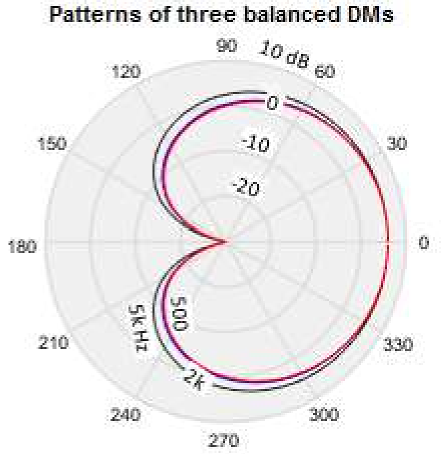


Fig.15 Polar patterns of the balanced DMs with pure tones

can observe that curves behave with sawtooth fluctuation along with 6 dB gain, so they have greater speech enhancement than an Omni mic has. The sawtooth of the response of the DM EB8 is big; the DM LB8, small; and the DM EB16, mid. Note that hardware of the DMs' implementation will smooth the theoretical sawtooth curves.

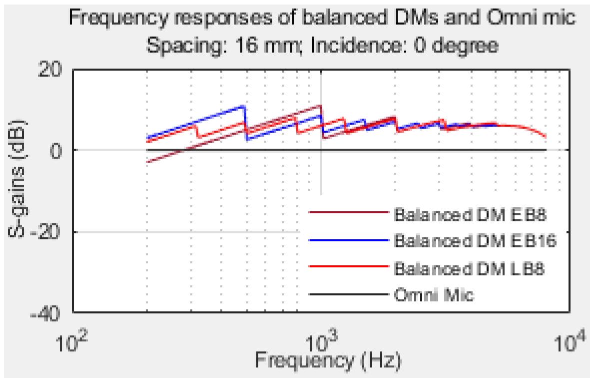


Fig.16 S-Gain frequency responses of balanced DMs at incidence 0°

We intercepted voices in a quiet room from Amy's wave file[7], which was an 11-word phrase[9], about 167,579 samples, sampling rate 44.1k Hz, word length 16 Bits and duration 3.8 s, and represented average speech/conversation. We acquired a traffic noise from a wave file[12], lasting 4 s, and a talking interference of Brian's voice from the wave file[7], lasting 4 s. The sound pressures of all the acquired time series were calibrated to be 60 dB SPL as a competing level. When an aid wearer enters a traffic spot, the noise surrounds the hearing aid from all orientations. We selected eight incident angles 0°, 45°, 90°, ..., 315° to represent the surrounding intrusion of traffic noise, but three of them, 225°, 270° and 315°, were ignored because of the shade effect of the wearer's head. Moreover, a hearing aid can be controlled by its wearer's head to back onto the interference, so we selected the individual talking interference at five angles 135°,157.5°, ..., 225° to represent the back intrusion, but two of them, 202.5° and 225, were ignored because of the same shade effect. Table 3 lists outputs and S/Ns of an Omni mic and three balanced DMs in the competing noises and interference. These data indicated that ① in the traffic noise, the three balanced DMs' outputs were 0.041, 0.0406 and 0.0492, respectively. When plus the speech, the S/Ns of the three DMs were 3.32, 5.45 and 6.24 dB, respectively, and the S/N of the Omni mic, 1.3 dB. ② In the talking interference, the balanced DMs' outputs were 0.00108, 0.0138 and 0.017, respectively. When plus the speech, the S/Ns of the balanced DMs were 17.1, 17 and 17.2 dB, respectively, almost without a difference, but the S/N of the Omni mic, 1.83 dB. These measuring results show us that the balanced DMs suppress the surrounding noise a few dB better than Omni mic does; and in the talking interference, over 15 dB better than the Omni does.

Table 3. Outputs and S/Ns of the three DMs and Omni mic in competing noise and interference fields

Types of sounds	Omni mic (RMS)	Balanced DM EB8 (RMS)	Balanced DM LB8 (RMS)	Balanced DM EB16 (RMS)
Amy's speech	0.0475	0.0601	0.0761	0.101
Traffic noise, S/N (dB)	0.0409 1.30	0.0410 3.32	0.0406 5.45	0.0492 6.24
Talking inter-ference S/N (dB)	0.0475 1.83	0.0108 17.1	0.0138 17.0	0.0170 17.7

6. Conclusions

Based on these data and curves from our measurements with balanced DMs of three types, we conclude that

1) Although we cannot achieve a quantity definition of the spectrum distortion of a directional DM with speech, the waveforms and spectrum measurements on the cardioid DM have explicitly shown spectrum distortion of the DM output, compared to those of the DM input signals.
2) To remove the distortion, the balanced DM which incorporates a multiband filter bank(band splitter), multi-magnifiers and an adder can perform a balanced frequency response, and reconstructs the original speech signal. The band splitter can have these types: equal BW filter bank, logarithmic BW filter bank and FFT plus IFFT processor.
3) We designed three balanced DMs, whose types were EB8, LB8 and EB16. When speech signals entered the DMs, the waveforms and spectrum of their outputs approximated to the original those of the DMs' inputs to a great extent. The effectiveness of the DM EB16 is better than that of the DMs EB8 and LB8, delay time is longer, about 5.9 ms, and the DM EB8 has shorter delay time of 2.9 ms. The waveform construct of the DM LB8 output changed slightly much; the cause was the quite different delay times of the filters of the bank LB8, from 0.374 to 9.25 ms. The delay time is related to the bandwidth, so the logarithmic BW filters can cause time-delay distortion although the logarithmic BW filter bank has advantages when implementing other signal processing.
4) When the three balanced DMs were moved in competing traffic noise and talking interference fields, they all behaved with reasonable suppression: in the surrounding noise, the three DMs performed a few dB S/N improvement than Omni mic did; in the back interference, the three DMs performed over 15 dB S/N improvement than Omni mic did.

References

[1] G. Keidser, K. Rohrseitz, H. Dillon, V. Hamacher, L.Carter, U. Rass, and E. Convery, *The effect of multi-channel wide dynamic range compression, noise reduction, and the directional microphone on horizontal localization performance in hearing aid wearers*, Int. J. Audiol. vol.45, 563-579, 2006.

[2] MC. Flynn, *Maximizing the voice-to-noise ratio (VNR) via voice priority processing*, The Hearing Review. vol.11, pp.54-59, 2004 (2008, republished).

[3] K. Moeller and C. Jespersen, *The effect of band split directionality on speech recognition and noise perception,* The Hearing Review, vol.5, pp.17-24, 2013.

[4] Phonak Insight paper, *Binaural directionality*, Stafa, Switzerland, pp.1-4, 2010.

[5] X. Zhang, *Polar plots of a directional microphone with real-world sounds and its spectrum distortion*, EPH-International Journal of Science and Engineering, vol.3, pp.1-10, 2017.

[6] Sonion, *Directional Microphones*, https://www.sonion.com/hearing/microphones/directional-microphones/.

[7] Voice Samples-LAMP Words for Life, https://www.aacapps.com/lamp/voices.

[8] Julius O. Smith, *Audio FFT filter banks*, Proc. of 12th Int. Conference on Digital Audio Effects (DAFs-09), Como. Italy, Sept. pp.1-4, 2009.

[9] X. Zhang, *Benefits and limitations of common directional microphones in real-world sounds*, Clinical Medicine Research, vol.7, pp.103-118, 2018. DOI: 10.11648/j.cmr.20180705.12.

[10] MathWorks, Bandpass/Filtering/DSP System Toolbox/ Simulink/Matlab. R2017b, Sept. 2017.

[11] JM. Alexander 20Q: *The highs and lows of frequency lowering amplification*, www.audiologyonline.com/ 20Q, Article#11772, Apr. 2013:1-13 .

[12] Traffic Wav Mp3 Sound Effects, http://www.pachd.com/traffic-mbience.html.

Author's Biography

Xubao Zhang *received his doctorate in electronics from Xi'an Electronic Science and Technology University in China and was a postdoctoral fellow at McMaster University in Canada. He worked as an EA and EMC engineer with Unitron Sonova, also with Oticon Canada. He has been interested in hearing aid technology strategies and performance evaluation, also in radar digital signal processing*