ScieniQ Publishers

Japan Journal of Medical Science

Research Article

Causes, Measurements, and Eliminations of Common Distortions of Hearing Aids

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Received: 09 Aug 2024
Accepted : 13 Aug 2024
Published: 19 Aug 2024

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Abstract
The common hearing aid (HA) distortions include nonlinear, frequency, and delay distortions. For the nonlinear distortion of an HA, we analyze the work characteristics of Class A, Class AB, and Class D circuits and the nonlinearity formations, and describe the principle of an advanced compressor with linear operation, i.e., floating-linear amplifier, and the ability to eliminate wide dynamic range compression (WDRC) distortion and to retain prescription formula. For the frequency distortion in an HA, we describe frequency response of a cardioid directional (Dir) mic and its 6 dB/octave up-slope unavoidably causes severe frequency distortion. Through viewing the Dir mic outputs with /voice/ input, we confirm that the frequency distortion exists, and the sound quality significantly drops. The multichannel operator with magnifiers can remove the frequency distortion. The resulting Dir mic has a balanced frequency response, and our simulation experiments verify that the sound quality is recovered well. For the delay distortion in an HA, we make several experiments on a simulated HA with open-dome and low gains and find the comb-filter effect with the pure tone composite and the sound coloration with the realistic speech sounds under the conditions of HA gains from 0 to 20 dB and delay time from 0 to 20 ms. Our experiments and discusses reveal the quantity relations of the comb-filter effect and sound coloration with HA gain and delay time. In fact, as long as the HA gain is less than 20 dB and delay are larger than zero ms, the delay distortion can be observed and perceived. We also suggest the approaches to eliminate the distortion such as raising sampling rate, multichannel of minimum/linear phase FIR filter bank, etc.

Keywords: Nonlinear distortion, frequency distortion, delay distortion, sound coloration, comb-filter effect, hearing aid

INTRODUCTION
The hearing aids (HAs) have good characteristics of amplification, frequency response, dynamic compression, etc. and the people with hearing loss significantly benefit from them on enhancing life quality and raising social engagement. An HA as an electronic device has unavoidably its troubles such as distortions, feedback squealing, etc. When designing an HA, we desire the non-distortion amplification, but the practical device always gives us various distortions. When the distortions are not eliminated well, an HA wearer may perceive an uncomfortable, misunderstood sound, thereby has to pay more cognition effort.

The common power amplifiers for HAs are classified as Class A, Class AB, and Class D, whose circuits all cause nonlinear distortions, i.e., harmonic distortion and intermodulation distortion. The results from [1], showed that under the condition of measured harmonic distortion from 1% to 35%, the average consonant identification scores decreased 15–29% therewith. The nonlinear distortion also results from WDRC (wide dynamic range compression) operation, which complies to the prescription gain formula, e.g., NAL-NL1 and achieves intelligibility for people with hearing loss [2,3]. Created a novel linear gain "compressor", called floating-linear (FL) operator, which is composed of gain-variable linear amplifiers controlled by a long-term monitor and a short-term monitor. When the fluctuation of long-term estimate of the input signal is within 9 dB, the FL amplifier maintains a constant gain while it complies to the prescription formula. In addition, an HA receiver also causes the nonlinear distortion [4], whose mechanism is similar to the amplifier nonlinearity.

Frequency distortion of a device in an HA occurs when a signal is getting in it and the corresponding frequency characteristic of its output is damaged. The frequency distortion is also referred to as the spectrum distortion. A traditional Dir mic is of cardioid polar plot and can reject the interference. However, through measurement, a cardioid mic causes frequency distortion since its frequency response presents a 6 dB/octave up-slope curve in the low-middle frequency region and a nearly horizontal curve in the high-frequency region. Comparing the waveforms of the input to output of a realistic speech [5], confirmed that the cardioid Dir mic significantly distorts the speech spectrum. Evidently, the frequency distortion of the Dir mic cannot be ignored since it damages the speech cues/naturalness and affects the speech intelligibility.

A digital HA usually needs several ms to tens ms to complete all the signal processing, depending on its circuits, algorithm, and speed of chip platform. By investigating, many audiologists reported that the delay behaviors can cause a kind of spectrum/waveform distortion in the HA mixed output under the open-dome fitting; the resulting conclusions indicate the relationships between speech perceptions and the delay times [6]. reported that the sound with a delay of 3 to 5 ms became noticeable and the sound with a delay of larger than 10 ms, objectionable [7]. Reported that the thresholds for hearing-normal participants able to discriminate col-

oration-pitches for delays were larger than 0.3 ms and the thresholds for hearing-impairing participants were 0.6 to 1 ms, depending on HA insert gai. [8]. Used a objective physics way to measure the delay times of five realistic HAs: hitting a mug with a spoon in front of the HAs and recording the time differences between output and input on an oscilloscope, and the resulting delay times are between 1.8 and 10 ms. The disturbing perception would be caused by HA delay larger than 14 ms; the detrimental effect on sound quality would be caused by the delay time 10 ms. In order to eliminate the delay distortions to make clearer sound quality, Wedix manufacturer first created ZeroDelay paradigm in the new MOMENTTM HA [9], which reduces the delay time to about 0.5 ms, so almost eliminates the comb-filter effect.

This paper intends to describe the causes, measurements, and eliminations of the above three types of HA distortions. Based on many data and figures from our researches, 1. the mechanism for the Class A, Class AB, and Claas D amplifiers used in HAs to cause nonlinear distortion is described and the float-gain linear amplifier to eliminate the WDRC nonlinearity but to retain the prescription formula is introduced; 2. the experiments on a traditional Dir mic with the realistic speech reveals the frequency distortion and the balanced Dir mic with multichannel FIR filter bank effectively removes the distortion; 3. open-dome and low gain are basic conditions for the HA to causes the delay distortion and the experimental results are discussed to show the relationships between the sound coloration, comb-filter effect, the delay time, and HA gains. The resulting information is very helpful for audiologists and hearing aid professionals to evaluate HAs, also for engineers to improve HAs on the distortions.

NONLINEAR DISTORTIONS OF A HEARING AID

Causes of the Nonlinear Distortions

The nonlinear distortion of an audio electronic device originates mainly from nonlinearity of its amplification circuits. Amplifiers in HA circuits have a work (output vs input) characteristic, which is somewhat nonlinear. Thus, even if the input signal is a pure tone, the output signal contains the fundamental tone (input tone) plus its harmonic tones. That is, the nonlinear distortion occurs. According to Fourier analysis of the output signal, its harmonic frequencies are two, three, … times the fundamental frequency. So the nonlinear distortion is also referred to as the harmonic distortion.

The HA amplifiers mainly contain the preamplifier and power amplifier. The former achieves the voltage amplification and its amplitude is often within a small range, so the caused nonlinear distortion can be ignored; the latter achieves the power amplification and is classified as Class A, Class AB, and Class D, whose amplifiers can reach the max battery voltage. The Class A has a low power efficiency (output/battery power rate) less than 50%, suitable for low-power amplification; the Class AB has a medium power efficiency less than 79%, suitable for the medium-power amplification; the Class D has high power efficiency less than 100% due to the circuit operation of off- or on-state, suitable for the high-power amplification. Fig. 1 shows work characteristic of a Class A amplifier and the red solid curve represents input-output function. We observe that the bottom bend O Bl results from CMOS (complementary metal oxide semiconductor) input characteristic and the top bend Cl Cn, from saturation of peak-clipping; the two bends cause the nonlinear output waveform (black) despite of the pure tone input (black). The center point A is called the work-point (bias setting), which ensures the most portion Bl Cl of the red curve is straight and the amplifier works in linear operation state with a pure tone input (red), so it can produce max undistorted output current waveform (red). However, this bias setting also makes the power dissipation of a Class A amplifier excessive, and it is only applied as the low-power amplifier.

A class AB amplifier is composed of a push-pull circuit of two Class A amplifiers. One Class A amplifier for positive waveforms uses the red solid curve in Figure 1 and the other amplifier for negative waveform uses another input-output function that is the red solid curve rotated by 180° (not drawn here) and symmetric with the point Bl. In order to have the maximum power efficiency, the work-points of two amplifiers are set as small currents as the point Bl. Plus, the Class AB amplifier works with a smooth transition at the overlapped work-point to reduces crossover distortion. In Fig. 1, the line Bl Cl is the work range of the positive waveform of Class AB amplifier; the work range for the negative waveform is the line Bl Cl of the symmetric red curve (not drawn here). Thus, the Class AB amplifier has three output ends for the positive, neutral, and negative signals. The distortion of class AB amplifier is caused by the incomplete linearities of the two input-output functions; however, due to the symmetric push-pull operation it cancels even-order harmonic components [10] and its nonlinear distortion is also low. The class AB is mainly applied to medium-power amplification; additionally, its receiver has three terminals.

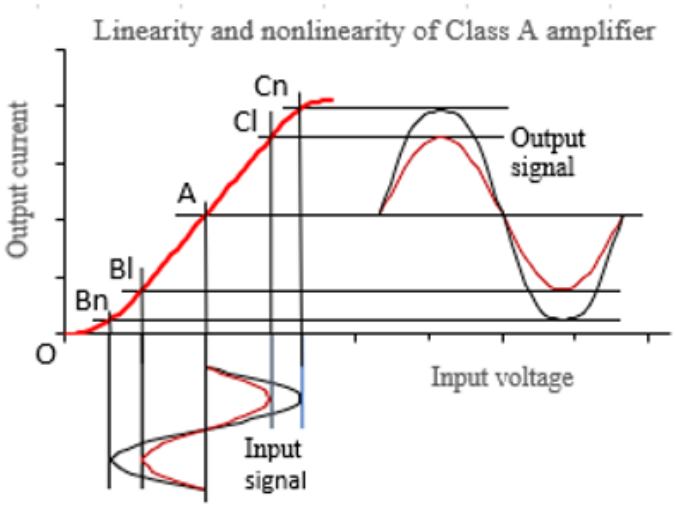

Fig. 1 Work characteristic of Class A amplifier and its nonlinear distortion

The class D amplifier for an HA is composed of a pulse width modulator (PWM) and a rectangular pulse power amplifier. The former utilizes triangular pulse string to encode an input audio signal into width-modulated rectangular pulse string. Fig. 2 shows the principle of PWM of a Class D amplifier for an HA; the triangular pulse string is regarded as the carrier and the comparator is used to encode the input signal. We observe that when the input level is low, the comparator produces a narrow rectangular pulse output and vice versa. The latter is bridge-configured inverters for amplifying the rectangular pulse string and forming a low-impedance voltage source. The bridge-configured inverters can be half-bridge or full-bridge (H-bridge) circuits, which requires double or single power supplies, respectively. In HA application, the half-bridge circuit and double power supplies are selected for simplicity [11] and the neutral voltage is 0.7 V. A Class D amplifier can dissipate as little as one-fourth the power of a Class AB amplifier for the same output. In practical device, the triangular pulse waveform is not that linear and causes nonlinear distortion; secondly the main cause of Class D nonlinearity is timing errors added by the dead time at the active driver, which is two-time intervals to ensure the MH and ML transistors off to avoid the shoot-through current, and achieved by manipulating the PWM [12]. The short dead time, e.g., less than tens ns, can control nonlinear distortion to be less than 3%. In the nowadays HA circuits, the power amplifier is not built with a transistor or a pair of transistors but rather a group of transistors such as an operational ampli-

fier of the differential mode in an IC chip. Thus, the amplifier nonlinear distortion is not controlled by HA manufacturers.

The HA receiver is composed of a coil, armature, diaphragm, etc. Because the back-and-forth movement of the diaphragm is not strict linear to the input current, that is, the output sound pressure level/input current is not a constant, especially under the large input current. The formation of the receiver nonlinear distortion is similar to that of Class A amplifier; once the armature reaches or touches either of the magnets, the saturation occurs, as at points Bn and Cn in Fig. 1. The output sound of the receiver contains the fundamental tone and harmonic tones when a tone signal is sent into the receiver. At the same input levels, the receiver distortion at low frequency is lower than that at high frequency [4].

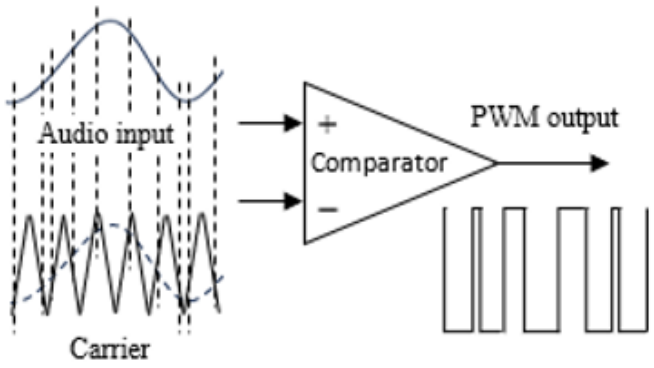

Fig. 2 Basic principle of PWM for a Class D amplifier

The dynamic range of human hearing is approximately from -10 to 130 dB SPL [13], the residual range of a person with hearing loss is lessened, e.g., from 30 to 130 dB SPL. Then an HA amplifier needs to amplify the -10 dB input sound to 30 dB and to maintain 130 dB input to fit external sounds. The most of HAs adopts popular WDRC amplification, which controls the gain to output an audible and comfortable level to an HA wearer. Thus, the input-output nonlinearity of the HA also contains the WDRC impact. In addition of the nonlinearities above, the WDRC makes the HA wearers perceive unnatural speech and somewhat exertive comprehension of speech although it ensures audibility and prevents the saturation distortion.

Perceptions and Measurements of the Nonlinear Distortions

During stage of the research and development, engineers can perceive the nonlinear distortions through viewing waveforms of the amplifier circuits on an oscilloscope. Generally, for the Class A amplifier the distortion waveform is the top clipping and bottom clipping; for Class AB amplifier the distortion is crossover disturbance around the neutral level; for Class D amplifier the distortion is downdip disturbance over the neutral level.

What do people perceive from listening to speech and music of HAs with nonlinear distortion? Different persons gave different answers such as raspy, tinny, raucous, buzzing, strident products, etc. [10], depending on HA brand, year built. The raspy and harsh distortion product may be caused by saturation nonlinearity, the buzzing product indicates the distortion from lack battery volume, and the strident product often comes from intermodulation (IM) distortion, more objectionable than harmonic distortion (HD). Even if the sound containing distortion products is still tolerable and intelligible for young people with normal hearing, such listening for a long time can lead to auditory fatigue. Many audiologists made some listening tests and verified that the distorted speech made the HA intelligibility going down in comparing to the undistorted speech. Reference [14] pointed out, the distorted information requires the brain to have to make more effort to learn, interpret, and comprehend the speech, and the brain must reassign more cognitive resources to untangle the distorted bio-electric auditory-neural signal.

IEC is an international electrotechnical organization, of which IEC 60118 is a group responsible for international standards on HA tests. ANSI is an American national standard organization, of which ANSI S3.22 is a group responsible for US standards on HA tests. In order to strictly identify the nonlinear distortion level of an HA, the two groups worked out the individual documents, in which the distortion specifications are defined and the tolerances are indicated for the HA manufacturers. Therein, the HD and total harmonic distortion (THD) are specified in clauses 7.12.1 of IEC 60118 [15] and 6.11 of ANSI S3.22 [16]. For example, the THD is defined as

$$\%\ \mathrm{THD}=100\sqrt{(P_2^2+P_3^2+P_4^2+\cdots)/P_1^2},$$

where P_1 = sound pressure of the fundamental tone in the earphone coupler and P_2, P_3, P_4, ... are sound pressures of the second, third, fourth, ...harmonics, respectively, in the coupler. When measuring the THD, the practical instruments always utilize only P_2 and P_3 for simplicity, which represent two the most intense harmonics. The tolerance of THD shall not exceed the value specified by a manufacturer plus 3% [16]. Meeting the specifications is an indication that the HA achieves the basic requirement. In fact the resulting THD from the above testing is, of course, a composition of the three nonlinear distortions from the amplifiers, receiver, and WDRC (if on) in an HA.

Another nonlinear distortion of HA, the IM distortion, is also defined by the two documents above. To facilitate the IM measurement, in clause 7.12.2 IM of IEC 60118 the test source gives two pure tones of frequencies F_1 and F_2, which have amplitudes within 1.5 dB of each other, F_2 is higher than F_1. The second order IM is the output of the test HA at $(F_2—F_1)$; the third order IM is the output of the HA at $(2F_1—F_2)$. Both IM distortions are expressed as a percentage or in dB referred to the output of F_2.

The nonlinear distortion of a receiver is measured by its manufacturer before shipping out and the specification is required to be Max 5% for most receivers [17].

According to the requirements of IEC 60118 and ANSI S3.22 standards, FRYE Electronics Inc. USA and Rohde & Schwarz, Inc. Germany have researched and developed their instruments for the HA tests. These instruments can measure almost all the specifications established by the two standard groups. For example, of the harmonic distortions, according to the standards, the specification HD is not measured. So the FONIX 7000 system by FRYE has a popular program that can measure the THD rather than HD [18]. However, if a manufacturer is interested in measuring HD, it can develop a self-made program that is incorporated into the FONIX 7000 system or Rohde & Schwarz instrument for the HD measurement.

Eliminations of the Nonlinear Distortions

The distortions from the amplifiers and receiver should be eliminated by the IC chip and receiver manufacturers, respectively, such as Texas Instruments, Analog Devices, and Knowles Electronics Inc., Sonion. The conventional WDRC projects the dynamics of the impaired hearing to the dynamics of the normal hearing, therefore it achieves the audibility and intelligibility of sounds in various listening environments; however, it does not perform well with preserving speech naturality, that is, it can ensure the hearing-impairing person with the HA to hear the various intensity

sounds but to perceive no naturality of speeches. Solution to ensure both the sound intelligibility and naturality is just "compression" of linear amplification, which has been researched for many years by audiologists.

In order to incorporate a prescription gain formula into the linear amplification, a novel compression operator was designed, called floating-linear (FL) amplifier, which is composed of gain-variable linear amplifiers in a multichannel operation controlled by a long-term monitor and a short-term monitor [3]. When fluctuation of the long-term estimate of the input signal is within 9 dB in a frequency channel, the FL amplifier retains the current gain; the input sound pressure range along the constant gain line is referred to as the linear window. When the input signal is an average speech, the monitor output (root of mean square) fluctuates within 10 dB. In a common conversation, the 9 dB FL window ensures the amplification to not cause nonlinear distortion. Figure 3 shows I/O work curve of the FL compressor in a frequency channel. The blue sold line represents the zero gain linear function; the black dotted line, the target gain of a prescription formula; the red solid line, FL window; the prescription gain is at the middle point, i.e., the cross point of the red and dotted lines; the dashed line, the transient impulse compression, nearly reaching zero gain; the dotted and dashed line, low sound level expansion, which compresses background noise, and its slope is the expansion ratio. When the fluctuation of the monitor estimate shows a big and slow shift along the current window, the input SPL is higher or lower out of the window, the FL amplifier gain immediately floats to the right-side or left side to form a lower or higher 9 dB linear window. Because the linear window is floating along the prescription gain formula, e.g., NAL-NL1, the FL amplifier ensures the output speech level is between audibility and comfortability. Meanwhile, the multichannel output integrates the linearly-amplified speech signals in all channels, acoustic naturalness is not lost and the HA wearer effortlessly understands the speech.

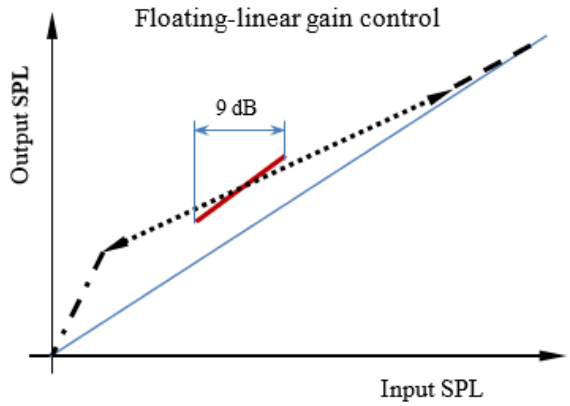

Fig. 3 I/O work curve of a FL amplifier in a frequency channel

FREQUENCY DISTORTION OF A HEARING AID

Cause of the Frequency Distortion

When we research HA microphone (mic), the essential specification of an Omni mic or a directional (Dir) mic is the sensitivity curve, whose unit is dB re 1V/Pa, where Pa is a unit of sound pressure. When their polar plot or frequency response is drawn, the units are a trouble. For expression simplicity, we use the concept of sensitivity (S) gain [5]. S-gain of a Dir mic=Dir mic sensitivity/Omni mic sensitivity; since its unit is dB only, it is convenient to express and draw a frequency response and polar plot of the Dir mic. That is, the S-gain of the frequency response and polar plot has a unit dB. Fig. 4 shows the frequency responses of an Omni mic and traditional (cardioid) Dir mic, denoted by the black and red, respectively. We observe that the Dir mic response is a 6 dB/octave up-slope curve while the Omni mic response is a horizontal line, this means that the spectrum components of the input sound are unequally operated by the Dir mic, that is, the Dir mic causes frequency distortion.

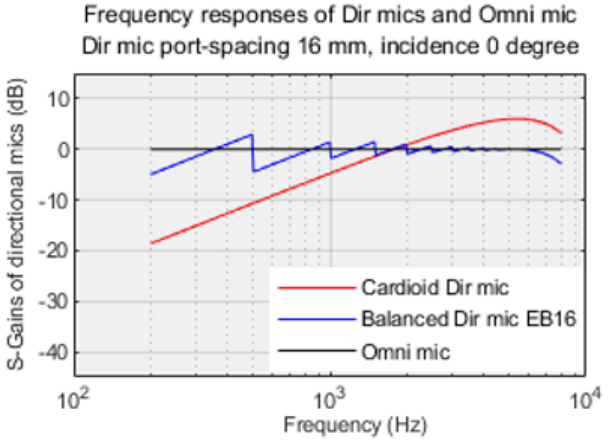

Fig. 4 Frequency responses of a cardioid Dir mic, Omni mic and balanced Dir mic

How does the frequency distortion of the Dir mic affect speech quality? To answer this question we need to make a test. A lady Amy's talk in a quiet room was recorded into a wave file; we abstracted a word "voices" waveform, which contains four phones /voi/, /c/, /e/, and /s/, and the entire /voices/ lasts 0.641 s. Here we use double slashes / / to represent the sound of a phone or word inside. We can view the waveforms by means of Time Scope of MATLAB. When the /voices/ enters the Dir mic, the input and output waveforms are shown in Fig. 5 (a) and (b), respectively, and through contrast, the high-frequency consonants /c/ and /s/ of the output are significantly enhanced and the low-middle frequency vowels /voi/ and /e/, significantly declined. This is a typical frequency distortion. Table 1 lists RMS (root-mean-square) data of the phones /voi/, /c/, /e/, and /s/

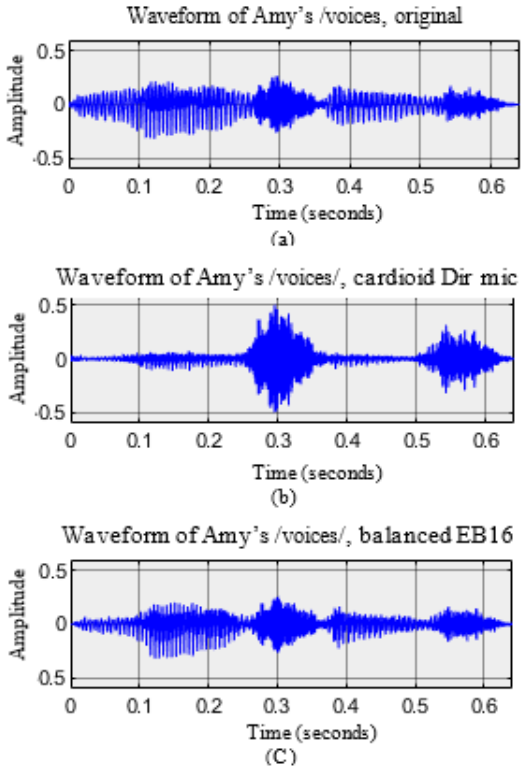

Fig. 5 Waveforms of /voices/, (a) original, (b) cardioid Dir mic output, (c) balanced Dir mic output with EB16

Table 1 RMS levels of phones of /voices/

phones	Original /voices/ RMS (mv)	Dir mic output RMS (mv)
/voi/	104	21.8
/c/	62.7	116
/e/	76.3	16.2
/s/	39	70.5

before and after the Dir mic operating, then we quantitatively know the changes of the spectral components.

Perception and Removal of the Frequency Distortion

Through viewing the input and output of the traditional Dir mic, the /voice/ waveform is distorted significantly due to wide power spectrum of the constituent phones. A practical way to perceive the change of a speech or music is the listening test. Any frequency distortion device must cause the change of speech pitch. Therefor we loaded long wave files of speech sentences from the Dir mic outputs into Adobe Soundbooth CS4, including a female Amy's and a male Brain's talks such as 11-word speech, "Hi, one of the available high-quality text-to-speech voices." One input sentence of the Dir mic and its output form one sentence pair for comparing. Then several sentence pairs are enough for our test material. When listening and comparing the sounds of each pair, we perceived that the overall pitch of the output sentence became obviously higher like rustling and noisy products due to the operation of the Dic mic; the tested speeches' naturalness is somewhat damaged, and the comprehension gets somewhat bad. When a word mainly contains high-frequency components, we perceive its pitch higher and when a word mainly contains low-frequency vowels, its pitch lower. The consequence is consistent with the frequency response of the Dir mic (red) in Fig. 4. Evidently, the frequency distortion cannot be ignored since it damages the speech cues/naturalness and affects the speech intelligibility.

Although the traditional Dir mic causes frequency distortion on speech, its strong suppression to an interference from the back side still attracts HA researchers. To remove the frequency distortion, balancing/flattening the frequency response of a Dir mic is an applicable and effective way. A conventional HA always contains multichannel/ band-split operation, which is used for noise reduction, feedback cancellation, etc. We can insert a magnifier into each band and the most magnifiers provide the mid-low frequency bands with compensated gains so that the sloped frequency response is modified into a balanced response, then the distortion is removed well. The effectiveness of the balanced frequency response depends on the number of multiple channels. The more the number is, the better the response is balanced. The Dir mic with the balanced response is called the balanced Dir mic [5]. Additionally, the multiple channels can have a few types of filter banks, e.g., equal bandwidth (BW) FIR filters, logarithmic (octave) BW FIR filters, and FFT plus IFFT operators. The logarithm BW FIR filter bank and the FFT plus IFFT (64 or 128 points) operators have better balanced response than the equal BW FIR filter bank has, due to their narrower BW of the low-frequency bands. But both the operations cause larger delay time than the equal BW FIR filter bank.

Sixteen Equal BW Filters' Bank

Currently Simulink of MATLAB provides extremely convenient function blocks to design various FIR, IIR, and FFT plus IFFT operators. For experiments of the balanced Dir mic, we designed three types of multiband filter banks at our Meadow SimuLink Lab, sixteen equal BW FIR filters, eight logarithmic BW FIR filters, and FFT plus IFFT processors. For example, of sixteen equal BW FIR filters, the filter bank meets the following basic requirements: 1. frequency coverage 100 to 8k Hz, 2. as short delay time as possible, 3. as low-ripple of frequency response as possible. Details of our design of sixteen equal BW FIR filter bank are presented as follows. 4. The Chebyshev II, linear phase FIR in DSP Toolbox of Simulink, which has shorter delay time than the logarithm BW filter bank. 5. In order to match the frequency responses of the balanced Dir mic with Omni mic, gains of the sixteen magnifiers are 6.1, 2.3, 1.4, 1.0, 0.82, 0.69, 0.6, 0.56, 0.53, 0.51, 0.5, 0.5, 0.5, 0.5, 0.5, 0.5. The magnifier gains are higher in the low-frequency region than those in the mid frequency region; the magnifier gains in the high-frequency region are 0.5 due to high-frequency gains 2 of the cardioid Dir mic. 6. Each filter of the bank has BW 500 Hz except for the 1st one 400 Hz and their center frequencies are 300, 750, 1.25k, …, 7.75k Hz. They characterize sixteen equal BWs, so we named the bank type EB16. 7. The frequency response of the EB16 has quite low ripples, within ±0.6 dB. 8. All the FIR filters have almost the same group delay of 5.8 ms. To confirm effectiveness of the filter bank EB16, we first measured frequency response of this balanced Dir mic and the result is shown with the blue curve in Fig. 4, which removes the 6 dB/octave upslope and is around the Omni mic response. Then, we measure the output waveform of this balanced Dir mic when inputting Amy's /voices/; the output is obtained, shown in Fig. 5 (c). We observe that the output waveform approximates to that of the input well; in comparing to Fig. 5 (b), the /voi/, /e/ are enlarged and /c/, /s/ are lessened, that is, the waveform distortion is basically removed. The further listening test with the Soundbooth CS4 verifies that we did perceive no difference between the two /voices/ in Fig. 5 (c) and (a). All the experiments verified that the output of the balanced Dir mic is almost the same as the input no matter viewing waveform or listening test. If the number of the bands is more, the waveform in Fig. 5 (c) will better approach that of in Fig. 5 (a), and the frequency response (blue) in Fig. 4 is smoother.

DELAY DISTORTION OF A HEARING AID

Causes of the Delay Distortion

In real-world, multiple sounds may arrive at our ears at their individual times and the listeners never perceive the sounds to be uncomfortable. This is a common time difference issue in our listening. In general, a listener is insensitive to the phases of sounds because the auditory system is a power (energy) detector rather than a pressure/amplitude detector [13]. Some mixed sounds in the ear channel, from the outside, eustachian canal, and head bones of the same source, arrive at our ear drums at different time, such delay times always are very short, so our ears' perception can not discriminate them, and we also ignore the delay effect. However, when the same speeches or music touch ears at different times, the listener may have an objectionable perception. The early analogue HA was composed of minimum time-delay components, having a negligible group delay [6], and the developers and customers hardly perceived the HA delay effect. Nowadays HA designs adopt DSP components and large delay of the output sounds is inevitable. On the other hand, open-dome fit to release air pressure in the ear channel is welcome by the HA wearers. Then the open-dome HA has two pathways to output sounds, one is the aid amplification pathway, and the other is the open-dome vented pathway. Sounds from the two pathways always present a large time difference and waveform and spectrum of the mixed sound may distort the HA output, that is, the delay distortion occurs. So many audiologists have investigated this effect, including objectively viewing waveforms and subjective perception on listening. Additionally, the delay distortion also occurs in multichannel operating of an HA, which splits the input signal into multiple bands; once the gain control or algorithm is completed, the split sub-signals are summed. These sub-signals may have different time-delay due to the different BWs and algorithm feature. After summing, the output envelopment may present delay distortion. Impact of the distortion depends on the delay differences of the band signals. For example, equal BW linear phase or minimum phase FIR filters cause less delay than logarithm BW

filters and FFT plus IFFT bins do. In a conventional multichannel operation, the former two need about a few ms and the latter two need more. They are allocated the large computation time while a feature algorithm, e.g., feedback cancellation, usually is allocated more operation time.

Experiments of the Delay Distortion

Many researchers have studied the delay distortion of HAs with open dome fit, including its occurrence, measurements, perceptions, and the conditions of impacting sound quality. Interaction of the HA output and the vented pathway output forms the mixed sound, which is delay-distorted and can be perceived when we listen to speech or view its waveform on an oscilloscope. The distorted sound is regarded as the sound coloration due to its changed pitch. When the input signal is a deterministic signal, e.g., the pure tone composite, the distorted output spectrum would present comb-like spectrum, which is referred to as comb-filter effect. We will verify the two characteristics in the following experiments.

We simulated a digital HA with constant frequency response and an open-dome vented pathway and focused on the total delay time of circuits and algorithms of the HA. When the sounds of both the HA and open-dome outputs are mixed, the mixed sound represents the final sound touching the ear drum of HA wearer. In terms of MATLAB Simulink platform, we achieved multiple experiments of six delay times and four aid gains with two sources: the deterministic broad-band signals and realistic talking speeches. Figure 6 depicts the block drawing of the delay distortion experiments. The input block Sound-input (file in) provides the mic output series; two output blocks Aid-output and Mixed-output (file out) record the HA output data and mixed output series, respectively. The simulated HA characterizes a multi-channel operator with feature algorithms, containing a bank of 16 equal BW FIR filters [19], an adjustable Gain block 0, 5, 10, 20 dB, and an adjustable Time-delay block 0, 1, 3, 5, 10, 20 ms. The delay time here includes the time-consumptions of both multiband filters and algorithms, exactly speaking, the delay time is from the HA mic input to receiver output. The mixed signal comes from the HA and vented pathway outputs. Particularly, the delay time is controlled by many samplings in series, one sampling interval is 0.0222676 ms, then 1 ms delay occupies 44 samplings in series. The deterministic Sound-input signal comes from a composite signal source, which combines 401 pure tones, frequency range from 0 to 8k Hz and intervals 20 Hz. The speech Sound-input signal comes from Amy's 11-word voice, "Hi, one of the available high-quality text-to-speech voices", of 167,579 samplings, 16 bits depth, and 3.8 s duration. Both the signal sources are of 44.1k Hz sampling rate. Aid-output and Mixed-output series will be transferred as wave files for us to view their waveforms and to listen to the sound's quality on the Soundbooth CS4.

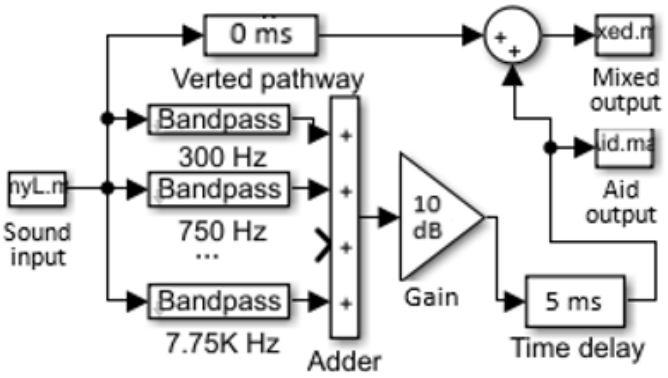

Fig. 6 Block drawing of delay distortion experiment by MATLAB DSP Tool

When the Sound-input series is deterministic composite source, comb-like spectra of the Mixed-output signals are observed by the MATLAB spectrum analyzer under the conditions: 1 the HA gain is less than 20 dB; 2. The delay time is in a range from 0 to 20 ms. 3. When the gain is close to 20 dB, the comb-filter effect is not observed; when the delay time is larger than 20 ms, the comb-filter effect should occur. Thus, the delay distortion by the HA with open dome occurs only with the low HA gain. Figure 7 shows spectral shapes of the two outputs, the red and blue curves represent the spectra of the HA, gain 10 dB, delay 5 ms, and mixed outputs, respectively. The HA output has a desired flat spectrum, and the mixed output has comb-filter effect verifying delay distortion. The further findings are 1. when the delay time is zero, the comb-like spectrum does not occur; the longer the delay time is, the denser the comb teeth are. 2. The lower the gain is, the longer the comb teeth heights are; when the gain is higher than 20 dB, the difference between spectra of HA and mixed outputs is invisible, that is, the comb-filter effect disappears. The more relevant figures are omitted here due to the space restriction. Analytically, the HA and vented pathway outputs have their deterministic phase responses; when one of them is delayed and added by the other, the spectral amplitude of the mixed output at some frequency is increased or decreased, depending on the in phase or out of phase of them, thus the spectrum of the mixed output shows the peaks and valleys, i.e. comb-like spectrum. The density of the comb teeth is related to the delay time and sampling rate. When the HA gain is 0 dB and delay is 0 ms, Amplitude of the mixed output will be double and comb-like spectrum disappears, presenting superposition of the two same signals; however, the HA gain 0 dB is not a practical situation. When the HA gain is increased, the vented pathway output performs a relatively weak effect; the spectral peak-valley (comb teeth) heights are reduced and they will disappear with the gain higher than 20 dB.

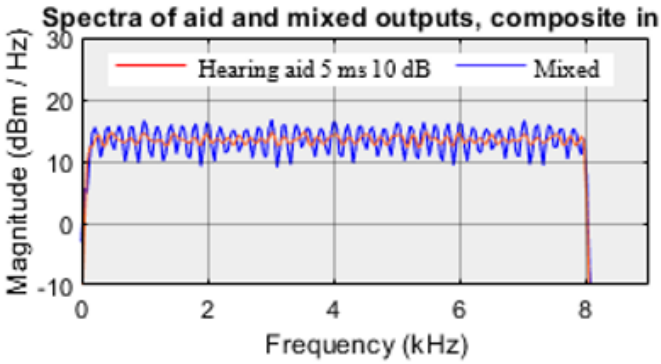

Fig. 7 Flat spectrum of HA output and comb-filter-effect spectrum of the mixed output under composite input

When the Sound-input series is a speech signal source, the comb-filter effect of the Mixed-output signal is hardly observed by the MATLAB spectrum analyzer no matter how HA gain and delay time are selected. Statistically, the speech signal is a kind of noise-like signal and the HA and vented outputs have their random phase spectra. When the HA output is delayed and added by the vented output, the amplitude spectrum of the mixed output randomly fluctuates, we have to use spectrum mean to express it. The spectrum mean of the HA output always is stable no matter the delay time is selected. Due to the same reason, the comb-filter effect is also randomly fluctuating; after averaging, the spectrum mean of the mixed output is indeterminate, related to delay time, thus, the spectrum would be colored if the delay occurs. Using the same conditions as the comb-filter effect test, we tested the impact of the delay distortion to coming speech. Figure 8 shows speech spectra of the two-pathway outputs, the red and blue curves represent the spectra of the HA, gain 5 dB, delay 1 ms, and mixed outputs, respectively. We observed that the HA output has a undisturbed spectrum and the mixed output spectrum is distorted or disturbed by the delay, i.e., colored. The details are 1. when the delay time is zero, the spectral disturbance does not occur; when the delay time is larger than zero, the spectral disturbance starts occurring but the longer the delay time is, the weaker

the spectral disturbance is. 2. When the gain is zero, the mixed output spectrum is lower than double of the HA spectrum; when the gain is increased, the disturbance gets weak; when the gain is higher than 20 dB, the difference between spectra of HA and mixed outputs is invisible, that is, the sound coloration disappears. The more relevant figures are omitted here due to the space restriction.

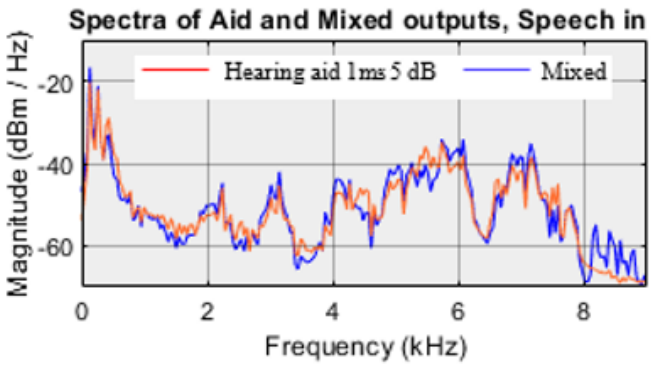

Fig. 8 Spectra of a hearing aid and the mixed pathway outputs under speech input

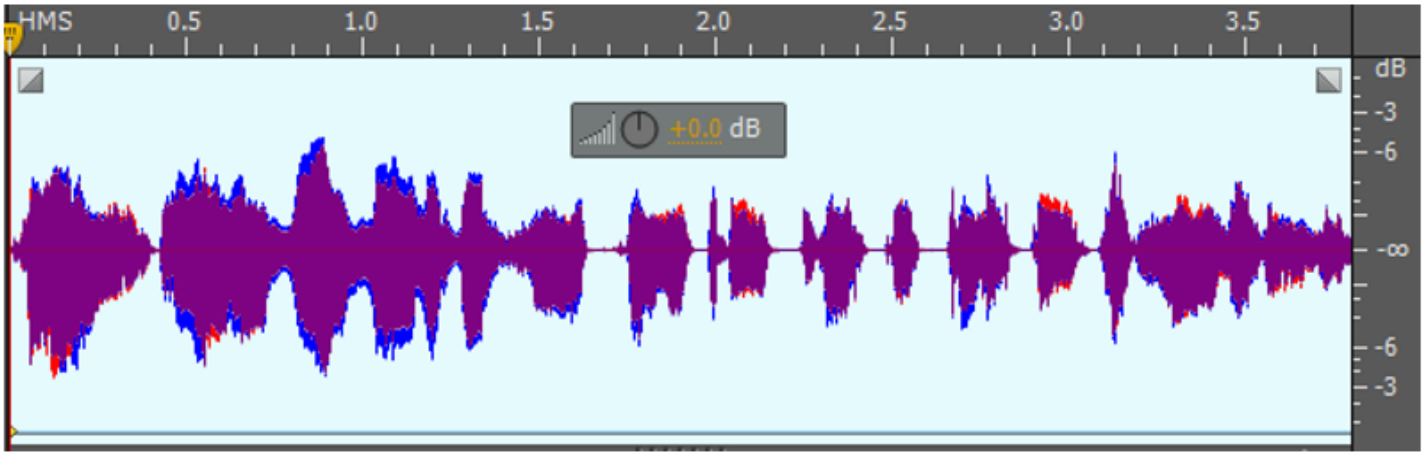

Fig. 9 Waveforms of HA (red), gain 10 dB, delay 5 ms, and the mixed (blue) outputs; purple waveform for the common part

In order to further verify impact of the delay distortion to speech quality, we switched to the measurement of waveforms and adopted the test conditions of the same HA gains and delay times as in the spectrum measurement above. The Sound-input speech signal is Amy's 11-word voice, see the last but two paragraphs, and the oscilloscope is Adobe Soundbooth CS4. Before experimenting, we need to transfer Mixed-output mat files into wave files in terms of the Simulink program [5]. Through viewing the Mixed-output waveforms and listening to sounds on the Soundbooth, we find that the delay distortion of the Mixed-output also presents on the waveforms. Figure 9 shows the waveforms of the HA and mixed outputs under the conditions of HA gain 10 dB and delay time 5 ms, and they are denoted by the red and blue, respectively, the purple waveform is the common part of both outputs. Our findings are 1. when the gain is lower than 10 dB and 5 ms delay is retained, the difference between the both outputs is larger; when the gain is higher than 10 dB, the difference gradually disappears; when going to 20 dB, the difference is invisible, that is, only the purple waveform exists. 2. When the delay time is larger than 5 ms and 10 dB gain is retained, the difference decreases a little; when less than 5 ms, the difference increases gradually; when reaching 0 ms, both the waveforms are the same but the mixed out is 2.39 dB larger than the HA output. 3. However, the differences between the Mixed-output and Aid-output waveforms are not as significant as the comb-filter effect with the input deterministic signal.

Perception and Elimination of the Delay Distortion

Many audiologists have investigated and measured the delay distortions of HAs since the birth of digital HA. The delay distortions have also been perceived by us through viewing the spectra and waveforms of the mixed sound, as well as listening test. Such sound distortions bring about a change in pitch and timbre, i.e., coloration of the mixed sound. The degraded sound quality was rated as poor quality, tube-sound, seashell-sound, objectionable or hollow [20]. The sound coloration can be perceived by normal-hearing and mild hearing-loos listeners wearing HAs, especially when listening to a certain music. There are situations where this "coloration" phenomenon may occur. For example, when processing several music tracks that include some of the same sounds, significant pitch and timbre change between the tracks may be perceived. Reference [19] studied the delay times in multichannel operators, one of which is composed of the logarithm BW FIR filters with quite different group delays, from 0.374 to 9.25 ms. The narrower the filter BW is, the larger the number of filter taps is and the longer the filtering delay is. Such large delay difference can cause distortion of the synthetical output envelopment. When the output waveform was viewed, its envelopment was noticeably different from the input envelopment of the operator. However, his listening test by the Soundbooth CS4 confirmed that the change of speech quality was not perceived due almost no spectrum change. This consequence is consistent with the other report [7]: the detectability of spectral distortions due to HA delay are possibly extremely low.

Generally human ability to locate a sound source in space is based on the auditory apparatus of two ears, including elaborate-shape pinnae. They can sense the orientation of a sound source, including azimuth and elevation,

as well as distance. The auditory location mainly relies on two cues from two ears of a listener. One is the interaural level difference and the other is interaural time difference (ITD) [13]. The sound arriving at two ears will presents a certain ITD or delay time when its location is away from this listener. Auditory evoked potentials measure the millisecond-by-millisecond activity of populations of neurons as they form an auditory percept. If a listener wears an HA with certain time delay and locates a sound source, the perceived distance is always more away from the side of the HA. Monaural location perception is based on the acoustics of the pinna, depending on the timing, intensity, and frequency of the sound arriving at the single ear in spite of the HA time delay.

Our above test results show that the delay distortion really occurs in the mixed output of an HA with delay time > 0 ms and mainly affects listening effectiveness of the wearers with soft and mild hearing loss. Theoretically the delay distortion is a type of disturbance that the auditory perception transmits from the ear to the brain; as a result, the brain has to work hard to understand and comprehend the disturbed sounds, thereby the cognitive load is increased. To reduce delay time of HA operating, considering the time allocation of DSP operation, the solutions contain the aspects: shortening sampling interval, adopting minimum or linear phase FIR filters, designing high-efficient algorithms, etc. Specifically, 1. The standard audio data sampling rate is 44.1k Hz. Assume that an HA processor adopts this rate and has a delay time 10 ms. In a simple analysis, when the other sampling rate 10 times this rate is used, the HA delay time will drop to 1 ms. An HA IC chip platform adopting sampling rate 441k Hz is not impossible in the nowadays microelectronic industry. The Velox™ platform of Oticon has the sampling rate 3M Hz [21]. 2. In digital HAs, the conventional multichannel operation and algorithm incorporation are indispensable. The multichannel structure can adopt equal BW FIR filter bank, logarithm BW FIR filter bank, or FFT plus IFFT operators; therein, FIR filter banks have much less delay time than FFT plus IFFT operators and the elaborate selection is critical. 3. The linear phase FIR filter has linear phase response, so group delays of the filters in different frequency bands are equal and no delay difference occurs when the band signals are summed, referring to Fig. 6. Considering the lowest delay effectiveness, the minimum phase FIR filters should be selected. The minimum phase FIR filter has the shortest group delay and the concentrated energy of toward time 0 of its impulse response [22], but the group delays of different band signals are a little unequal and the slight delay distortion may occurs when the signals are summed. So the minimum-phase filter has shortest group delay but the linear phase filter has equal group delay. 4. Additionally, the filter length is selected as short as possible. For example, when a 512-tap FIR filter has 5.8 ms delay, a 128-tap filter reduces the delay to 1.45 ms. Each bin of FFT plus IFFT operators can be regarded as a FIR filter, which has neither linear phase nor minimum phase, rather mixed-phase; the group delay of FFT plus IFFT operators is longer than that of minimum or linear phase filter bank but its advantages are large number of frequency channels, design simplicity, and less computation load.

SUMMARY

This paper describes the nonlinear, frequency, and delay distortions of hearing aids, including their causes, measurements, and eliminations. On the main body it is our research result, and its minority is our review for readers to benefit from the comprehensive knowledge of distortions.

1. The nonlinear distortion is a kind of often-heard HA distortions, it occurs in early Class A, Class AB amplifiers, and later Class D amplifier, also in the receiver and WDRC operation. We elaborately analyze and illustrate the distortion causes of the three-type amplifiers. These nonlinear distortions of HD, THD, and IM can be measured according to the standards by the IEC and ANSI organizations, and easily be perceived by viewing waveforms and listening test. These objectionable distortions are eliminated well by HA IC chip manufacturers and HA design engineers. In addition, we describe the operation of an efficient linear "WDRC", FL-amplifier, which performs with both linear amplification and prescription gain, Thus, the resulting HA possesses intelligibility, comfortableness, and naturalness together.
2. The frequency distortion of an HA has occurred since the Dir mic was born. It has a severe symptom: the frequency response of a cardioid Dir mic presents an up-slope of 6 dB/octave in low-middle frequency region. When we view the output waveform of the Dir mic with an input of sound /voice/, the phones with low-frequency tones will drop the amplitude significantly and the phones with high-frequency tones will raise the amplitude significantly. When we listen to the output sound of the Dir mic with a realistic speech, we perceive remarkable higher pitch. The frequency distortion product is a noisy sound, objectionable, but easy to remove. In the multichannel operator, a group of magnifiers are individually inserted into each channel; when the magnitudes' gains are properly designed, the frequency response of the Dir mic is balanced, i.e., the 6 dB/octave up-slope is removed. By viewing the output waveforms and listening to the sounds, we verify that the frequency distortion is removed.
3. The delay distortion is a type of confusing HA distortion, not easy to be perceived and measured. It occurs when the HA has a open-dome vented pathway and low gain. We made some delay distortion experiments and found that the comb-filter effect of the mixed output spectrum of the HA exists under the condition of the deterministic pure tone composite, HA gains less than 20 dB, and delay times larger than 0 ms. In the other experiment, we viewed the spectral and waveform differences, respectively, between the mixed and HA outputs with the 11-words speech inputs and under the delay distortion conditions: HA gains less than 20 dB and delay times larger than 0 ms. Such spectral differences indicate the speech sound coloration caused by the delay distortion. The waveform differences between the mixed and HA outputs bring about objectionable perception. According to the DSP delay time allocation, we suggest several approaches to eliminate the delay distortion, e.g., high sampling rate, multichannel operation of FIR filter bank, and adopting minimum/linear phase filters.

All the distortions occurring in an HA all are objectionable to HA wearers, affect the audibility, comfortableness, naturalness, and comprehension, and the solutions to eliminate them are effective.

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Cite this article: Xubao Zhang. (2024) Causes, Measurements, and Eliminations of Common Distortions of Hearing Aids. Japan Journal of Medical Science 5 (1): 165-173.

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