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A HYBRID PREPROCESSING METHOD TO IMPROVE CLARITY OF DIRECTIONAL AUDIO

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Abstract. *Acoustic fidelity of an audio system is decided by the component sound signals at different frequencies. For a directional audio system, the audible sounds are realized through demodulation with nonlinear effect of the acoustic parametric array, so the harmonics are to occur, in which the difference-frequency second harmonics bear low frequency characteristics affecting clarity of the audio system. A DSB_AM-SSB hybrid weighted preprocessing method is proposed to suppress the harmonics in audible sounds for enhanced clarity. The simulation results indicate that with the proposed method, the performance of difference-frequency harmonic suppression is promoted by 4 times, and acoustic fidelity of the directional audio system is improved obviously.*

Keywords: Difference-frequency second harmonic, Hybrid weighted preprocessing, Directional audio, Clarity

1. **Introduction.** Since Westervelt et al. put forward the acoustic parametric array in the 1960s [1-3], the directional audio systems on ultrasonic had developed substantially. According to their theories, the preprocessing modulating technique was key to realize the directional audio and decided the performance and volume of the audio system. Consequently, the preprocessing methods were studied extensively and deeply. Dekun et al. proposed the bandwidth efficient recursive equalization transactions [4,5], Yoneyama et al. proposed the DSB method [6]; Gan et al. gave the square root method and SSB method, proving their effectiveness to increase sound pressure [7]; Ji et al. introduced the phononic crystals into the acoustic parametric array [8]; Ji et al. presented the adaptive Volterra series filter method to predict harmonics [9,10]. The above work contributed a lot to improving the performance of directional audio systems.

The audible sounds comprise multiple components at different frequencies with different expressive forces. For example, high frequencies decide representability, medium frequencies decide tone colour and low frequencies decide spaciousness [11,12], but if the low-frequency components are overabundant, the resulting sounds will be vague with low clarity. In realization of directional audio by means of acoustic parametric array, the audible sounds are obtained through self-demodulation with the air nonlinear effect, during which lots of high-order as well as low-order harmonics will be produced. Especially, the difference-frequency second harmonics affect clarity of sounds seriously. All the above-mentioned preprocessing methods were intended to lower the holistic harmonic distortion, but were not specifically aimed at the control of low-order harmonics by difference-frequency second harmonic distortion. Moreover, although theoretically the square root preprocessing algorithm was not accompanied by harmonic output, that perfect state was realized on condition of infinite band width of the ultrasonic emitter, and the algorithm was rather complicated for operation.

In this paper, the generation mechanisms for low-order difference-frequency second harmonic produced during the nonlinear demodulation in the popular DSB and SSB preprocessing algorithms are analyzed. To suppress the low-order harmonics and promote clarity of audible sounds in a convenient engineering way, a DSB_AM-SSB hybrid weighted preprocessing method is proposed. The simulation results show that, the superposition of different weights of the DSB and SSB preprocessing algorithms can effectively attenuate the low-order harmonics by difference-frequency second harmonics produced in the acoustic parametric array, and is a good example for enhancement of sound clarity.

2. **Hybrid Preprocessing Method.** According to the Berktay far-field solution and derivation of the KZK equations [1,2], the amplitude relation between the audible sounds and the original FM signals through air self-demodulation is:

$$p_d = \frac{\beta p_0^2 S^2}{8\pi\rho_0 \alpha c_0^4 z}\frac{\partial^2}{\partial\tau^2}E^2(\tau) \tag{1}$$

where β is the nonlinear factor, p_0 is the amplitude of formation of carrier wave acoustic source, $E(\tau)$ is the envelope function, ρ_0 is the air density, and z is the hearing distance.

To analyze the harmonics produced in this nonlinear process, the general procedure is applied: two uniform-amplitude audio signals are used for the input, and the harmonic components in the output are observed to evaluate the performance of the system. $f(t) = \cos(\omega_1 t) + \cos(\omega_2 t)$ is for the input, where ω_1 = 1kHz, and ω_2 = 3kHz. The DSB_AM preprocessing method is chosen for analysis. Letting the modulation depth be m and the envelope function $E(t) = [1 + mf(t)]$, from Formula (1) the audible sound through self-demodulation is obtained:

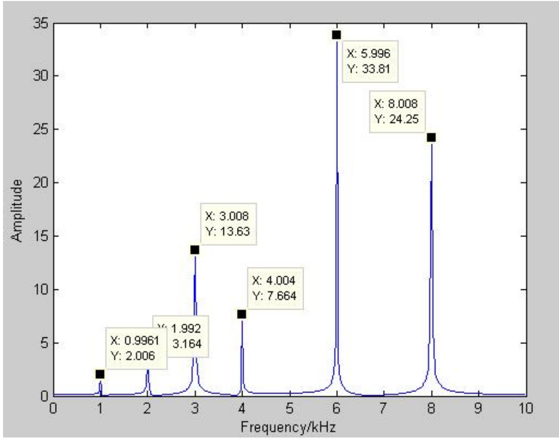
$$\begin{aligned}
p_d(t) &= \frac{\beta p_0^2 S^2}{8\pi\rho_0 a_0 c_0^4 z}\frac{d^2}{dt^2}[1 + m\cos(\omega_1 t) + m\cos(\omega_2 t)]^2 \\
&= -\frac{\beta p_0^2 S^2}{8\pi\rho_0 a_0 c_0^4 z}\left\{2m\left[\omega_1^2\cos(\omega_1 t) + \omega_2^2\cos(\omega_2 t)\right] + m^2\left[\omega_1^2\cos(2\omega_1 t) + \omega_2^2\cos(2\omega_2 t)\right]\right. \\
&\quad \left. + m^2\left[(\omega_1+\omega_2)^2\cos(\omega_1+\omega_2)t + (\omega_1-\omega_2)^2\cos(\omega_1-\omega_2)t\right]\right\}
\end{aligned} \tag{2}$$

From the above equation, demodulation of the input dual audio signals with the acoustic parametric array gives the audible sound comprising various frequency components: the fundamental frequency $2m\left[\omega_1^2\cos(\omega_1 t) + \omega_2^2\cos(\omega_2 t)\right]$, the difference frequency $m^2(\omega_1 - \omega_2)^2\cos(\omega_1 - \omega_2)t$, the sum frequency $m^2(\omega_1+\omega_2)^2\cos(\omega_1+\omega_2)t$ and the second harmonic $2m^2\left[\omega_1^2\cos(2\omega_1 t) + \omega_2^2\cos(2\omega_2 t)\right]$, and the sound amplitude is related to the modulation factor. Figure 1 shows that, with the increase of the modulation depth, the harmonics will dominate the output (when $m = 1$, the harmonics account for over 70.1% of the output [13]) and the sound effect can be hardly realized; on the other hand, as the modulation depth decreases, the harmonics will be effectively suppressed, but the system volume will fall down evidently and the emitting ultrasonic wave occupies a large portion of the power amplifier output energy.

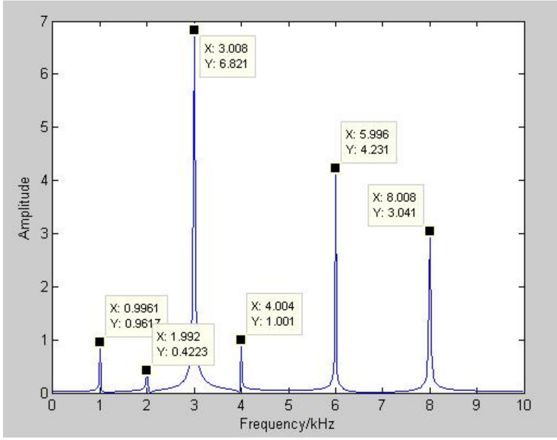
In short, in the case of the double side band amplitude modulation, the sound effect will be satisfactory only when the modulation factor is very small, and the volume and clarity of the audio system can hardly be simultaneously promoted if the same power amplifier is employed.

When the SSB premodulation technique is applied, the corresponding envelope square of the modulated signal is given [7]:

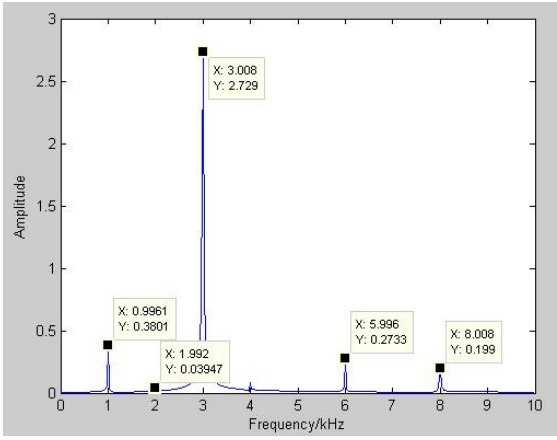
$$E^2(t) = f^2(t) + \hat{f}^2(t) = \left[1 + m^2(\cos(\omega_1 t) + \cos(\omega_2 t))\right]^2 + \left[m^2(\sin(\omega_1 t) + \sin(\omega_2 t))\right]^2 \tag{3}$$



(a) $m = 1$



(b) $m = 0.5$



(c) $m = 0.2$

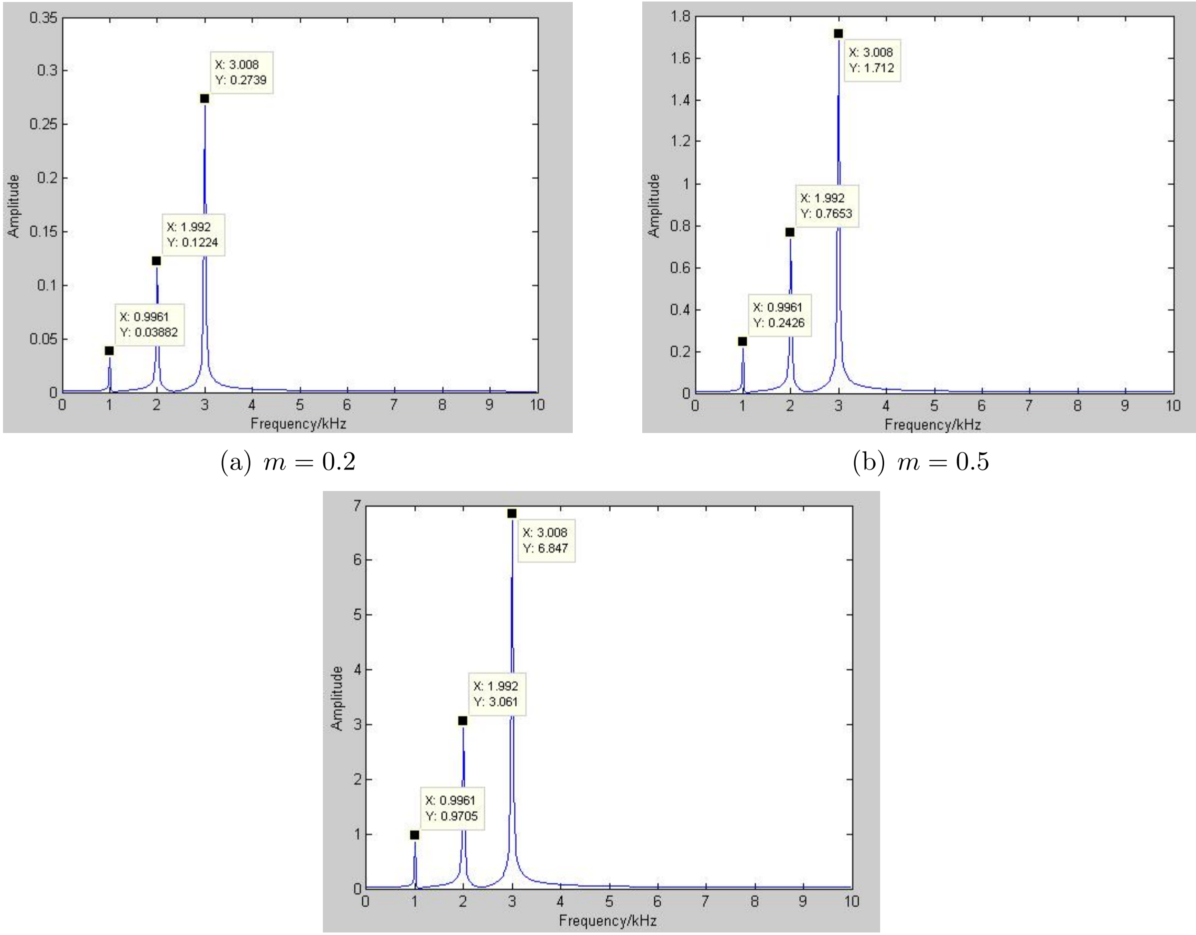
FIGURE 1. Harmonics produced through cross-modulation with the DSB_AM preprocessing method at different modulation depths

Substitute (3) into (1), where $\omega_1 = 1\text{kHz}$, $\omega_2 = 3\text{kHz}$, the audible sound pressure produced in the self-demodulation for the SSB algorithm is written [10]:

$$p_d = -\frac{\beta p_0^2 S^2 m^2}{8\pi\rho_0 c_0^4 z a_0}\left[\omega_1^2\cos(\omega_1 t) + \omega_2^2\cos(\omega_2 t) + (\omega_1 - \omega_2)^2\cos(\omega_1 - \omega_2)t\right] \tag{4}$$

The frequency spectra of the output harmonics at different modulation factors m are shown in Figure 2. Obviously, the harmonic signal $\omega_2 - \omega_1 = 2\text{kHz}$ remains existent and the ratios of its amplitude to the amplitudes of the output dual sound signals keep constant except that the amplitudes are different at different modulation depths, the said ratios keep at about A (3kHz)/A (2kHz) = 2.237, A (1kHz)/A (2kHz) = 0.322. The SSB modulation preprocessing method has the advantages of power saving, band width saving, small side band distortion and low signal-to-noise ratio. And harmonics can hardly be found besides the second difference frequency as depicted in Figure 2, which is broadly used for the premodulating method to realize directional audio. However, for the harmonic diagram, the ratio of amplitude of the second difference frequency to that of the effective audio frequency keeps constant. So if the SSB preprocessing method is chosen, to enlarge the attenuation from the second difference frequency to the audible sound and improve the sound clarity, some new way is needed.

Given high efficiency of the SSB method, it is promising in development of convenient and economic directional audio systems while the influence of the second harmonic on



(a) $m = 0.2$ (b) $m = 0.5$

(c) $m = 1$

FIGURE 2. Harmonics produced through cross-modulation with the SSB preprocessing method at different modulation depths

the sound clarity should be effectively controlled. Analysis of Formulae (2) and (4) indicates that, at a specific amplifier power, superpositions of different weights and different modulation factors for the two algorithms of DSB and SSB could form or weaken the term $(\omega_1 - \omega_2)$ in Formulae (2) and (4) but keep the outputs of (ω_1) and (ω_2) almost unchanged. Therefore, at the same power level, the sound clarity could be substantially promoted and audition effect distinctly improved.

As is stated above, at a given amplifier power, the SSB preprocessing method is employed as the baseline to simulate and verify the performance of the second harmonic difference frequency at different weight values and modulation factors. The hybrid weighted method was formulated with different weight values and different modulation depths to output the audible sound:

$$Aout(f) = k_1 * DSB\_AM(m_1) + k_2 * SSB(m_2) \tag{5}$$

where k_1 and k_2 are the weight values and the condition $|k_1|+|k_2| = 1$ is required to attain the same amplifier power, and m_1 and m_2 are the modulation factors for the DSB_AM method and SSB method, respectively. According to the requirements of different engineering applications, such as the highest clarity or highest sound pressure or the tow compromise, select a different $k_{1,2}$ and $m_{1,2}$ values. The SSB method needs linear power amplifier, the DSB_AM method is not, so SSB costs higher than DSB_AM. The hybrid method is carried out by the following two ways: one is getting finest $m_{1,2}$ with assigned weight $k_{1,2}$ (specified the cost method); the other is determining $k_{1,2}$ and $m_{1,2}$ by the optimizing (cost free method). The details list subsequently.

Method one: given weight method (specified the cost method)

1) load a given weight, initialization parameters.

2) cycle optimization

2.1) according to the accuracy set step, DSM_AM modulation coefficient from the 0-1 cycle,

2.2) according to the accuracy set step, SSB modulation coefficient from the 0-1 cycle,

2.3) the Fourier transform is used to obtain the power of the two difference frequency signal by Formula (5),

2.4) contrast with the last difference signal power, better than the last value, recorded,

2.5) execution cycle.

3) consistent with the results of the optimization, obtain the system's two difference frequency signal and harmonic power etc.

4) output results.

Method two: optimum clarity (cost free method)

1) initialize the relevant parameters, set the optimum clarity goal such as two difference frequency harmonic suppression parameters, the output sound pressure and the maximum harmonic.

2) optimizing

2.1) given the accuracy to the set step, the weight of k_1 from 0 to $|k_1| <= 1$.

2.2) given the accuracy to the set step, weight k_2 from 0 to $|k_2| <= 1 - |k_1|$.

2.3) DSM_AM modulation coefficient from the 0-1 cycle.

2.4) SSB modulation coefficient from the 0-1 cycle.

2.5) the Fourier transform and Formula (5) are used to obtain the power of the two difference frequency signal, the output sound pressure value, and the harmonic power.

2.6) compare with the previous data; update the record with the best data.

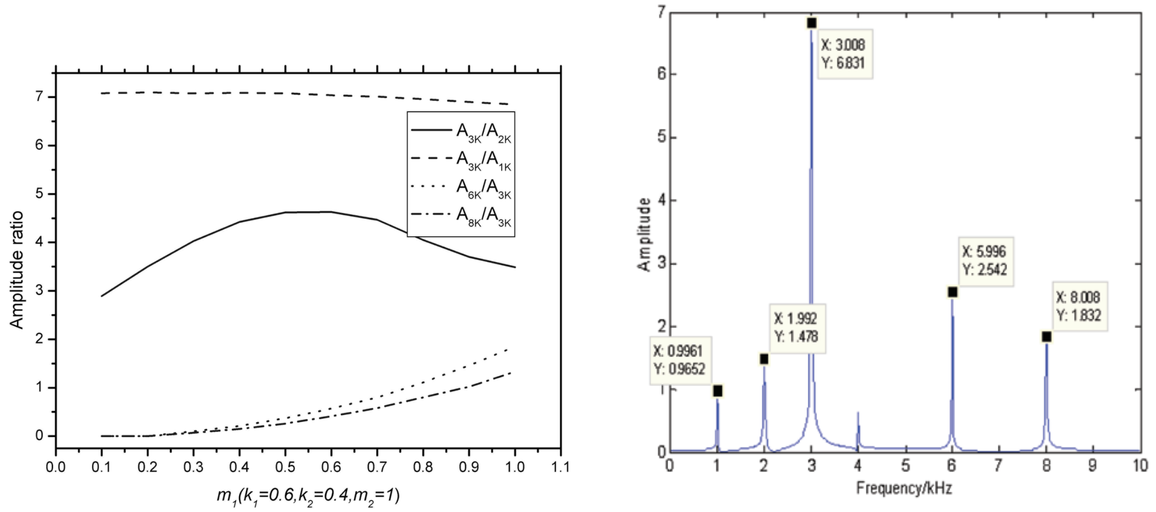
2.7) the implementation of loop optimization.

3) according to the final record give the final results and accuracy.

4) output the results.

3. **Simulation Results.** Along with the actual needs of the project, the first method is used in the experiment. The simulation results are shown in Figure 3 and Figure 4. It is clear that, for the given weight values $k_1 = 0.6$ and $k_2 = 0.4$ (from project cost budget), when the modulation factor m_2 for the SSB method equals 1, there exists an optimum modulation factor $m_1 = 0.5$ for the DSB_AM method to suppress the second harmonics, which corresponds to a ratio value A 3k/A 2k = 4.63, about 2.1 times of that out of the only SSB modulation. Similarly, let the modulation factor m_2 for the SSB method be 0.5, an optimum modulation factor $m_1 = 0.4$ for the DSB_AM method corresponds to the peak ratio value A 3k/A 2k = 8.95, about 4.06 times of that out of the only SSB modulation. Evidently, application of this hybrid weighted method could well improve the suppression effects on the second difference harmonic frequency while at a certain price.

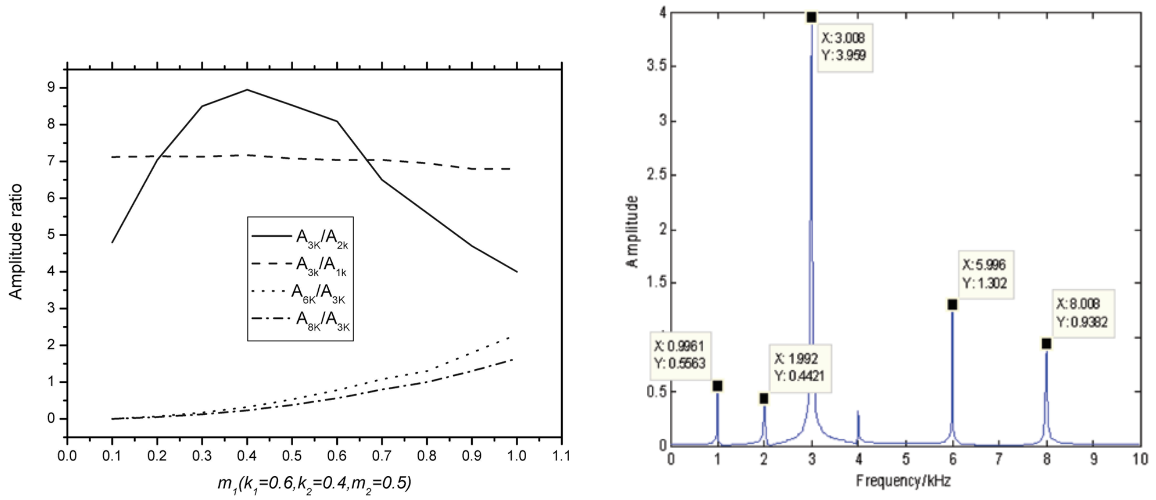
Moreover, from Figure 3 and Figure 4, with the increase of modulation factor m_1, the corresponding second harmonic will be boosted. Furthermore, according to the second harmonic suppression ratios shown in Figure 3 and Figure 4, with the fall of modulation factor m_1, better performance of second harmonic suppression will be achieved while the sound amplitude after self-demodulation will go down correspondingly. So the application of the proposed weighted method will substantially suppress the second harmonics and



(a) Output harmonics with the m_1 change

(b) Output at the $m_1 = 0.5$, $m_2 = 1$, $k_1 = 0.6$, $k_2 = 0.4$

FIGURE 3. The output harmonics vs. modulation factor m_1 ($m_2 = 1$, $k_1 = 0.6$, $k_2 = 0.4$)



(a) Output harmonics with the m_1 change

(b) Output at the $m_1 = 0.5$, $m_2 = 0.5$, $k_1 = 0.6$, $k_2 = 0.4$

FIGURE 4. The output harmonics vs. modulation factor m_1 ($m_2 = 0.5$, $k_1 = 0.6$, $k_2 = 0.4$)

promote the sound clarity while bringing about sound pressure loss to some extent, which should be synthetically considered in practice.

4. **Conclusions.** The sound effect of a directional audio system is dependent on the difference-frequency component sound signals, among which the low-frequency components decide the sound clarity. A DSB_AM-SSB hybrid weighted method is proposed to enhance the second difference frequency harmonic suppression. The simulation test shows that the presented method could promote the suppression capacity by over 4 times for the second difference frequency harmonic, and better second difference frequency harmonic suppression performance could be gained if the weight values and modulation factors are synthetically adjusted. Moreover, if the hybrid weighted method combines with the RMS

and SSB method or RMS, SSB and DSB_AM etc., although it increases the system's complexity and hardware cost of implementation, it maybe gets better performance on the second difference frequency harmonic suppression and the sound clarity.

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