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Nonlinear Distortion Identification based on Intra-wave Frequency Modulation

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Abstract: Recently, most of the methods used to measure and analysis signal property are based on the linear transform theory, such as FFT, STFT, wavelet, etc. Unfortunately, these methods usually cause meaningless results when it is used to analysis nonlinear signal. In this paper, we use Hilbert-Huang transform (HHT) to review the nonlinear distortion and define a novel nonlinear parameter named Nonlinear Distortion Degree (NDD) which is based on intra-wave frequency modulation measurement. The loudspeaker model simulations are used to illustrate the intra-wave frequency modulation caused by nonlinear distortion. The results agree that NDD can reveal more accurate and physical meaningful nonlinear distortion characteristic.

Keywords: Nonlinear distortion, HHT, intra-wave frequency modulation, loudspeaker.

1. Introduction

In the signal analysis field, Fourier based analysis methods such as FFT and STFT take the dominated position. However, there are some crucial restrictions of the Fourier spectral analysis: the system must be linear and the data must be strictly periodic or stationary; otherwise, the resulting spectrum will make little physical sense. By far, some time-frequency analysis methods, including wavelet analysis, Wigner-Ville distribution, etc. have been used for non-stationary signal analysis and achieve excellent results, but they have their respective limitations [1]. Because of limited length of the basic wavelet function, wavelet may cause energy leakage. Beside this, wavelet analysis has the difficulty for its non-adaptive nature when it encounters nonlinear problem in which deformed wave-profile may cause spurious harmonics. The difficulty with WVD is the severe cross terms which will make some frequency ranges have negative power. On the other hand, limited by signal analysis principle, researches on the nonlinear distortion phenomenon did not get significant progress. Recently, nonlinear distortion is measured in term of total harmonic distortion (THD) which is the ratio of the harmonic wave energy to total signal energy. Obviously, it is too rough for the interesting details.

Huang et. al. raised a novel signal processing method named Hilbert-Huang Transform (HHT) based on EMD (Empirical Mode Decomposition), which is suitable for analyzing nonlinear and non-stationary signal [2]. Different from the traditional signal processing methods, the HHT is an adaptive decomposition method and can yield more physical results. EMD is a complete, approximately orthogonal and self-adaptive method which has the ability to decompose signal by time scale. Some numerical experiments show that EMD behaves as a dyadic filter bank [3, 4].

As a strongly nonlinear system, loudspeaker is associated with several nonlinear effects such as electronic, magnetic, mechanical and sound. In the early stage, the displacement of the diaphragm in the low-frequency range was found can be described by Duffing equation [5]. Loudspeaker is considered as a mass/spring device driven by a Lorentz force and loaded by acoustical impedance. Such a model was presented in term of equivalent parameters by Thiele and Small [6].

In this paper, the HHT method and its improvements was studied firstly. Then, a novel nonlinear distortion parameter called nonlinear distortion degree (NDD) is presented based on intra-wave frequency modulation which is a unique definition in HHT. Finally a Duffing-like loud-

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speaker model is learned and simulated under different voltages. The simulation results demonstrate that the wave profile deformation caused by nonlinear distortion should be represented as intra-wave frequency modulation. It also verifies NDD is more accuracy and physical for nonlinear measurement.

2. Hilbert-Huang Transform Method

To analysis the nonlinear and non-stationary signal, the definition of instantaneous frequency and energy is needed, which must be functions of time. The principle of instantaneous energy and envelop is accepted widely, but the opinion of instantaneous frequency was under arguments for a long time until Hilbert Transform was introduced to define analytic signals by Gabor [7].

2.1. Instantaneous Frequency and Empirical Mode Decomposition

Gabor's theory is given as follow: for an arbitrary real-valued signal, $X(t)$, the Hilbert transform is defined as

$$Y(t) = \frac{1}{\pi} P \int_{-\infty}^{+\infty} \frac{X(\tau)}{t-\tau} d\tau \tag{1}$$

where P denotes the Cauchy principal value. $X(t)$ and $Y(t)$ form the complex conjugate pair, then we can have the analytic signal, $Z(t)$.

$$Z(t) = X(t) + jY(t) = a(t)e^{j\theta(t)} \tag{2}$$

in which

$$a(t) = \sqrt{X^2(t) + Y^2(t)} \tag{3}$$

$$\theta = \arctan \frac{Y(t)}{X(t)} \tag{4}$$

Here a is the instantaneous amplitude, and θ is the instantaneous phase function. The instantaneous frequency is simply defined as

$$\omega(t) = \frac{d\theta(t)}{dt} \tag{5}$$

With both amplitude and frequency being a function of time, we can express the amplitude in terms of a function of time and frequency. However, Cohen point out that only the 'monocomponent signal' can be transformed to instantaneous frequency by Hilbert transform[8]. The monocomponent signal means that at any given time, there is only one frequency value. The definition of 'monocomponent' is not clear before N. E. Huang proposed the definition of intrinsic mode function (IMF). An IMF is a function that satisfies two conditions[2]: (1) in the whole data set, the number of extrema and the number of zero crossings must either equal or differ at most by one; and (2) at any point, the mean value of the envelope defined by the local maxima and the envelope defined by the local minima is zero. Unfortunately, most of the practical data are not IMFs. Therefore, a method called Empirical Mode Decomposition (EMD) was proposed to decompose a complicated signal into a group of IMFs. Given a signal $x(t)$, the effective algorithm of EMD can be summarized as follows:

1. Find out all local extrema of $x(t)$, obtain the upper and lower envelope which are the cubic spline interpolated of its local maxima and minima.
2. Calculate the mean function $m_1(t)$ of the envelopes.
3. Compute the residue value $h_1(t) = x(t) - m_1(t)$. If $h_1(t)$ satisfies the two conditions of IMF, it should be an IMF. Otherwise, treat $h_1(t)$ as the signal, repeat the shifting steps 1-3 on it until the shifting result is an IMF, we denoted it as $c_1(t)$.
4. Obtain the residue $r_1(t) = x(t) - c_1(t)$. Apply the above procedure on it to extract another IMFs.
5. The process is repeated until the last residue $r_n(t)$ is a monotonic function or has at most one local extremum.

The shift result is highly determined by the stop criterion. Different stop criterions make results vary. Although there are several kinds of stop criterion, Wu *et. al.* suggest fixed sifting time criterion[9]. In separate research, Flandrin and Wu point out that EMD is in fact a dyadic filter bank[3,4]. In the research by Wu and Huang[10], the dyadic property is available only shift times is about 10. Too many or too few iterator numbers would decrease the dyadic property.

2.2. Time-Frequency Distribution Spectrum

In order to obtain the time-frequency-energy distribution of given signal, a naturally step is to apply the Hilbert transform to each IMF component and calculate the instantaneous frequency by means of Equation (5). Unfortunately, some theoretic condition limit this presumable application. A major problem is caused by Bedrosian's theory[11].

From Equation (5) we can conclude that the instantaneous frequency is only determinate by phase function. For any IMF $x(t)$, we can express it in terms of

$$x(t) = a(t) \cos\theta(t) \tag{6}$$

The physically meaningful instantaneous frequency require the signal to satisfy

$$H[a(t)\cos\theta(t)] = a(t)H[\cos\theta(t)] \tag{7}$$

where $H[\cdot]$ denotes the Hilbert transform. The Bedrosian theory point out that only if the Fourier spectrum of $a(t)$ and $\cos\theta(t)$ are absolutely disjoint in frequency space and the the frequency range of the $\cos\theta(t)$ is higher than $a(t)$, the Equation (7) is valid. Unfortunately, practical data seldom satisfy this condition. To overcome this difficulty, a normalization scheme is proposed by Huang *et. al.*[12]. The algorithm is given as follows: for arbitrary data $x(t)$ given as Equation (6),

1.Find all of the maximum points of the absolute value of $x(t)$.
2.Get the upper envelop $e_1(t)$ by spline interpolation of these maximum points
3.Conduct the normalization scheme on the $x(t)$:

$$f_1(t) = \frac{x(t)}{e_1(t)} \tag{8}$$

Ideally, $f_1(t)$ should be identical to $\cos\theta(t)$, but the envelop $e_1(t)$ often cut $x(t)$ when $a(t)$ changes sharply. Under this circumstance, $f_1(t)$ may have values higher than 1.
4.To obtain useful FM part, the normalization is used as an iterative process

$$f_n(t) = \frac{f_{n-1}(t)}{e_n(t)} \quad n = 2, 3, \ldots \tag{9}$$

until all the maxium values are 1.
5.Then the FM and AM components are defined as

$$F(t) = f_n(t) \tag{10}$$

$$A(t) = \frac{x(t)}{F(t)} \tag{11}$$

The combination of the normalization scheme and the application of Hilbert transform to the FM component are called Normalized Hilbert transform (NHT). Then the time-frequency distribution can be defined as follows:

$$H(\omega, t) = \mathrm{Re}\left[\sum_{i=1}^{n} A_i(t) e^{j\int F_i(t)dt}\right] \tag{12}$$

3. Intra-wave Frequency Modulation and Nonlinear Distortion Degree

In the HHT principle, the fluctuating of instantaneous frequency in one oscillation is regarded as intra-wave frequency modulation. The intra-wave frequency modulation is a unique definition of HHT and based on a common phenomenon: if the frequency changes from time to time within a wave, its profile can no longer be a pure sine or cosine function. Therefore, any wave-profile deformation from the simple sinusoidal implies the intra-wave frequency modulation [2]. However, this important nonlinear information is totally lost in Fourier spectral analysis and wavelet analysis. In the past, the wave profile deformation was treated as harmonic distortion. Intra-wave frequency modulation offers new understanding on nonlinear oscillation systems in more details.

With the power of HHT, we can discuss the definition of nonlinear distortion. To quantify the nonlinear distortion, an index is needed to give a quantitative measure of how far the final outputs of nonlinear system deviates from original signal.

To define the nonlinear distortion degree, NDD, the first step is to obtain the marginal spectrum $M(\omega)$ of signal. Marginal spectrum is defined as the integral of HHT time-frequency spectrum

$$M(\omega) = \int_0^\infty H(\omega, t)dt \tag{13}$$

which offers a measure of total amplitude contribution from each frequency value.

Consequently, the degree of nonlinear distortion is defined as the max deviation from the carrier frequency.

$$NDD = \frac{\max|M(\omega) - f_c|}{f_c} \tag{14}$$

Obviously, for a linear system, the output signal's marginal spectrum will concentrate to the carrier signal frequency. Then, the NDD will be identically zero. Under this condition, the Fourier spectrum can equal to the marginal spectrum and make physical sense. If the marginal spectrum spreads into a frequency range, this index will no longer be zero, then the physical sense of Fourier spectrum will decrease. The higher the NDD value, the more nonlinear is the signal.

The degree of nonlinear can also be a function of time implicitly, because the definition depends on the time length of integration of marginal spectrum as shown in Equation (13). As a result, a signal can be piecewise linear. Therefore, the nonlinear distortion degree can be modified to vary with time. The $NDD(t)$ is defined as

$$NDD(t) = \frac{\max|\overline{M(\omega)} - f_c|}{f_c} \tag{15}$$

in which the overline indicates average in a short time span at time t. For based on intra-wave frequency modulation, the NDD have the potential to reveal more details in nonlinear distortion.

4. Loudspeaker Experimental Results

4.1. The Loudspeaker Model

The large signal behavior in low frequency rang of loudspeaker can be modeled by a group of simplified nonlinear differential equations[13], such as Equation (16) and Equation (17).

$$E(t) = R_{eb} i_c + \frac{d(L(x)i_c)}{dt} + \Phi(x)\dot{x} \tag{16}$$

$$\Phi(x) i_c = M_{ms}\ddot{x} + R_{ms}\dot{x} + k(x)x - \frac{1}{2}\frac{dL(x)}{dx} i_c^2 \tag{17}$$

The Equation (16) is the electrical part of the model, where $E(t)$ is the voltage, i_c is the current and x is the loudspeaker diaphragm displacement. $\dot{x}$ and $\ddot{x}$ denote the loudspeaker diaphragm velocity and acceleration respectively, where the overdots represent first and second derivatives of x. The main electrical parameters are the voice-coil electrical resistance R_{eb} and the voice-coil nonlinear self inductance $L(x)$. The Equation (17) describes the mechanical part, which is modeled as a damped mass-spring mechanical system, in which the M_{ms} is moving mass (including air mass), R_{ms} is mechanical damping and $k(x)$ is nonlinear stiffness. Beside them, there is an added reluctance force $-\frac{1}{2}\frac{dL(x)}{dx}i_c^2(t)$ caused by the nonlinear self-inductance. The two equations are connected by the loudspeaker motor interdependence between them. Additionally, the nonlinear force factor $\Phi(x)$ of Equation (16) denotes the ratio between the force produced by the motor and the current.

The nonlinear differential equation that describes the nonlinear vibration can be derived if we approximate $\Phi(x)$, $L(x)$ and $k(x)$ by a truncated power series:

$$\Phi(x) = \phi_0 + \phi_1 x + \phi_2 x^2 + \cdots + \phi_n x^n \qquad (18)$$

$$L(x) = l_0 + l_1 x + l_2 x^2 + \cdots + l_n x^n \qquad (19)$$

$$k(x) = k_0 + k_1 x + k_2 x^2 + \cdots + k_n x^n \qquad (20)$$

The nonlinear coefficients ϕ_i, l_i and k_i can be estimated from experimental results, using the procedure developed by Park and Hong[14]. Using such a model, the harmonic distortion of the diaphragm velocity at low frequency range can be predicted with good agreement of real measurements.

The model simulation and real loudspeaker measurement are studied later. The loudspeaker to be analyzed is TianAi PW-11C, with the parameters:$R_{eb} = 5.87(\Omega)$, $L_{eb} = 207.321(\mu H)$, $M_{ms} = 2.0144(Ns/m)$, $k_0 = 2282(N/m)$, $\Phi_0 = 4.762(N/A)$ and $E(t) = \sqrt{2}A\sin(\omega t)$. The same parameters are also used in the model.

4.2. Model Simulation

The distortion analysis for the loudspeaker was carried out by means of simulation tests with 1v, 5v, 10v and 15v at 125Hz, 250Hz and 500Hz stimulate signal respectively.The instantaneous frequency is calculated from the diaphragm displacement with different driven voltage and shown in terms of overlapping the diaphragm displacement diagram with the same time axis.

At the beginning of our simulation, we carried out the test under small stimulate signal at low frequency. In the Figure (1), with the 125Hz sinusoidal driven signal at 1v, the vibration of the diaphragm is almost similar with pure sinusoidal wave and the instantaneous frequency is fairly close to a horizontal line. We can learn form the figure that deformation appears at both ends of the instantaneous frequency curve. In the EMD algorithm, the cubic splines interpolation creates top and bottom envelopes which are implemented in the first step of the shifting process. It is difficult to interpolate data near the beginnings or ends, where the cubic splines can have swings. Estimation of top and bottom envelopes is difficult as there are not enough data [15]. From the diagram, we can reach the conclusion: with small stimulate voltage, loudspeaker behaves like linear system.

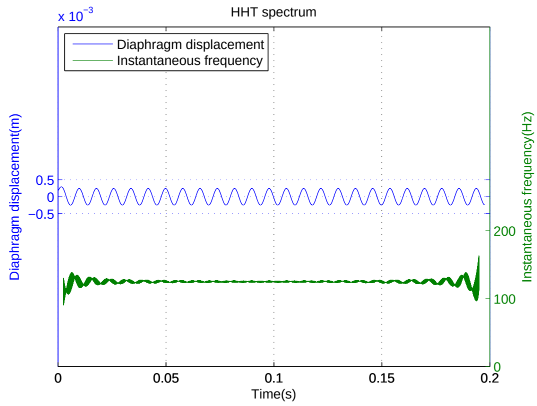

Figure 1 loudspeaker model simulate result with 1v and 125Hz sine signal stimulate

With the increasing of the input signal voltage from 1v to 15v, the nonlinear effect is enhanced notably. In the traditional Fourier transform analysis, this nonlinear phenomenon is recognized as harmonics in infinite frequency range. Figure (2) demonstrate that the instantaneous frequency fluctuates in the whole time length. The degree of the oscillation reflects the nonlinear distortion.

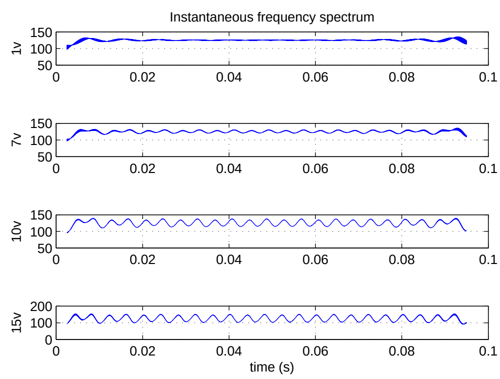

Figure 2 Instantaneous frequencies of model test results with 1v, 7v, 10v, 15v stimulate at 125Hz

Under different driven signal voltage, there is a significant phenomenon: following the increasing of stimulate voltage, the nonlinear effects grow correspondingly. It is due to the nonlinear of electrodynamic coil-magnet system and the large displacements of the rim[16]. In the process of oscillation, the nonlinear effects make the diaphragm displacement no longer a pure sinusoidal function, which is a typical intra-wave frequency modulation phenomenon. By contrast, due to the nonlinear deformation of diaphragm vibration, in linear analysis methods, it leads to various phenomena: harmonic, subharmonic and superharmonic frequency entrainment and chaotic behavior in small range of control parameters[17–19]. These frequency structures reveal that the output of loudspeaker have the nonlinear distortion which is expressed as intra-wave frequency modulation in HHT principle uniquely. The intra-wave frequency modulation means the frequency changes within a wavelength whose wave profile is no longer a pure sinusoidal function. Therefor, the the wave profile deformation which is known as nonlinear distortion implies the intra-wave frequency modulation. The nonlinear distortion usually illustrate in terms of harmonics in Fourier analysis. However, these harmonics generally do not have physical meaning because FFT is not suitable for nonlinear signal. The intra-wave frequency modulation is a better definition of these nonlinear phenomena.Furthermore, based on the principle of HHT, the distortion components do not spread energy over whole frequency range. It just concentrate into a narrow frequency band which match the practical perception better.

4.3. Experiment Results

The measurements for the loudspeaker was carried out with 137.5Hz, 275Hz, 550Hz, 1100Hz, 2200Hz, 4400Hz and 8800Hz respectively.

The test result of the stimulate signal from 137.5Hz to 4400Hz is shown in Figure (3), in which the instantaneous frequency of the loudspeaker output fluctuates 2 times in one wave cycle. Obviously, the structure of the loudspeaker output instantaneous frequency is illustrated as intra-wave frequency modulation. Furthermore, the frequency deviation varies under different stimulate frequency. From 137.5Hz to 1100Hz, the IF fluctuate range decrease to almost zero. On the other hand, from 1100Hz to 4400Hz, the range raise significantly. This means the smallest nonlinear distortion appears at 1100Hz and increases when stimulates become both higher and lower.

When stimulate frequency increases to 8800Hz, the instantaneous frequency of the loudspeaker output disunite into 3 IMF. The joint distribution is shown in Figure (4). From the diagram, we can see that the highest frequency oscillation mode take the dominate percentage of the energy. Nevertheless, there are two extra component which indicate that the nonlinear distortion under 8800Hz stimulate is very high and have the crosstalk distortion with lower frequency components.

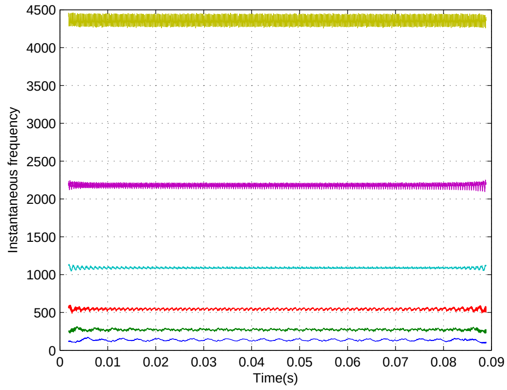

Figure 3 Instantaneous frequency of loudspeaker output with 137.5Hz, 275Hz, 550Hz, 1100Hz, 2200Hz and 4400Hz stimulate.

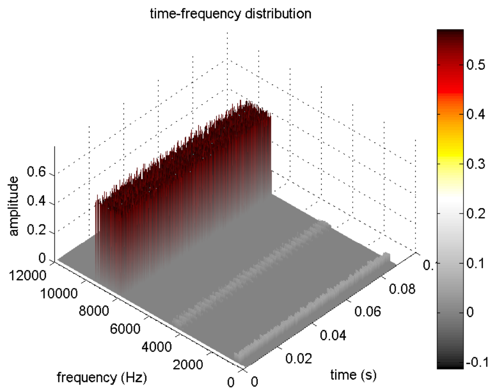

Figure 4 The time-frequency-amplitude distribution of 8800Hz stimulate.

4.4. NDD and THD Comparison

Total Harmonic Distortion (THD) is a common parameter to characterize nonlinear distortion, which is defined as

$$THD = \frac{\sqrt{\sum_{i=2}^{\infty} H_i^2}}{H_1} \tag{21}$$

where H_1 denotes the level of fundamental frequency and H_i denotes the level of i^{th} harmonic. The difficulty of THD is the Fourier spectrum lost some important nonlinear information which represents the characteristic of system. Therefore, the THD offers only an approximation for the distortion.

The NDD and the THD are measurement for all the experiment data. The results under different voltages are shown in Figure (5).

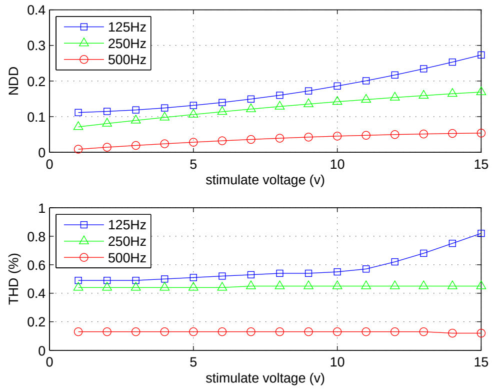

Figure 5 NDD and THD measurement results for the simulation data.

Both of NDD and THD reveal the nonlinear characteristic from different aspect. From the results of NDD and THD, we can extract the general character: with the increasing of stimulate signal voltage, the distortion of loudspeaker output decrease notably. On the other hand, the NDD parameter clearly demonstrate the increasing trend of distortion with the raise of stimulate voltage. By contrast, the THD results fail to show this important characteristic. The difference between the THD and NDD agrees that NDD can represent more detail of nonlinear system and have more physical sense.

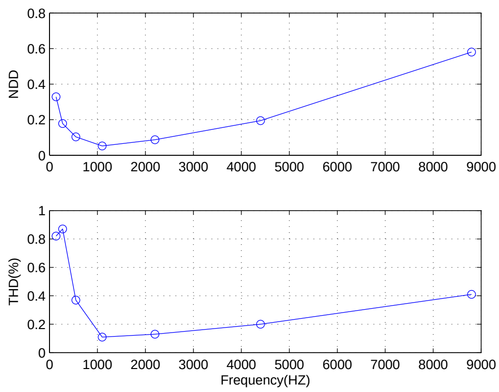

Figure 6 NDD and THD measurement results for the experiment data.

Figure (6) compares the NDD and THD on the real experiment measurements with different stimulate frequency. Both of them agree that the distortion have low level in middle frequency range. The NDD curve line is more smooth than THD. The lowest point appears at 1100Hz in both THD and NDD curve. However, in THD measurement, the distortion in high frequency range do not have significant increase. Part of the reason is the THD measurement do not consider the sub-harmonic energy. On the other hand, in NDD diagram, the distortion in high frequency range increases notably. From Figure (3) we know that in high frequency range, the intra-wave frequency modulation degree increases with the rising of frequency. Therefore, the NDD value render the real trend of loudspeaker distortion.

The Fourier spectrum uses linear superposition of trigonometric functions, therefore, it needs additional harmonics to simulate the deformed wave shape caused by nonlinear effects. As a result, the harmonic explanation for nonlinear phenomena just fulfil the mathematical requirements for fitting the data but not have physical mean. Therefore, the THD which is based on harmonics level loss some important information about the nonlinear distortion. In this way, the NDD is an better alternative to character nonlinear distortion.

5. Conclusion

In this paper a novel nonlinear distortion identification parameter called nonlinear distortion degree (NDD) is proposed, which is based on Hilbert-Huang transform. The HHT method absolutely lies on the character of the signal and have local-adaptive property. Consequently, HHT can avoid the shortcomings of linear transform methods and is very applicable to analyze nonlinear and non-stationary signal. The loudspeaker model was simulated and the results in terms of diaphragm displacement were obtained. By applying HHT on the output of the loudspeaker model, the instantaneous frequency spectrum shows that the essential of loudspeaker distortion is intra-wave frequency modulation which relates the deformation of diaphragm displacement from simple sinusoidal. By way of analyzing the structure of time-frequency spectrum of model simulation output, NDD demonstrate better performance than classic THD parameter.

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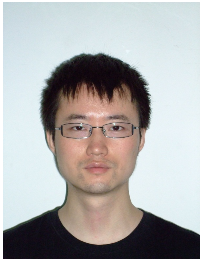

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