

Exploring Denoising Applications of One-Dimensional Fourier Transform in Microphone Filters

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Abstract. This paper presents a comprehensive exploration of the denoising application of one-dimensional Fourier transform in microphone filters. Beginning with an elucidation of the fundamental principles of one-dimensional Fourier transform, the study delves into its utility in denoising, encompassing short-time Fourier transform and frequency domain filtering. The analysis further scrutinizes sound signal characteristics, noise sources, and processing methodologies. Detailed discussions ensue regarding filter design methodologies grounded in one-dimensional Fourier transform. Through case analyses and experimental validations, the efficacy of one-dimensional Fourier transform in practical applications is demonstrated. Finally, the paper outlines prospective research directions. This research is instrumental in enhancing microphone signal quality by mitigating noise, thereby facilitating precise and effective signal processing. Keywords include One-dimensional Fourier transform, microphone filters, denoising, time-frequency analysis, frequency domain filtering, short-time Fourier transform, and signal-to-noise ratio in audio signal processing.

Keywords: One-dimensional Fourier transform; microphone filters; frequency domain filtering; short-time Fourier transform; signal-to-noise ratio.

1. Introduction

With the rapid development of science and technology, audio signal processing technology has become increasingly prominent in daily life and industrial applications, especially in the fields of speech recognition and communication transmission in noisy environments. As the primary acquisition device of sound signals, the output signal of microphone is often accompanied by complex background noise, which seriously affects the accuracy and effectiveness of subsequent signal processing. Consequently, enhancing microphone signal quality by effectively suppressing noise is currently one of the central subjects under investigation.

As the cornerstone of signal processing, the one-dimensional Fourier transform plays an irreplaceable role in noise suppression and signal recovery due to its powerful ability to convert signals from the time domain to the frequency domain. In the design and implementation of microphone filters, the Fourier transform enables us to intuitively and accurately observe and analyze the frequency components of audio signals, and then design and optimize filtering algorithms accordingly.

This paper focuses on the theoretical basis and practical effect of one-dimensional Fourier transform in the application of microphone filter denoising. First, we will introduce the basic principles of the one-dimensional Fourier transform and explain how it reveals the frequency structure of an audio signal to facilitate the distinction between noise and wanted signals. Subsequently, several typical filter design methods based on Fourier transform are analyzed in depth, and how these methods can achieve effective noise suppression in microphone filters are discussed [1].

In addition, this paper will demonstrate the application results of one-dimensional Fourier transform in practical microphone noise suppression projects through case analysis and experimental verification and evaluate its advantages in effectively reducing background noise while ensuring the integrity of speech signals. Finally, looking forward to the future research directions, this paper discusses how to further optimize and improve the microphone filter denoising technology based on one-dimensional Fourier transform with the help of the latest progress of modern signal processing technology, so as to meet the needs of more complex and changeable application scenarios.

2. Characteristics of Sound Signals

The sound signal serves as the conveyer of information pertaining to the wavelength and intensity inherent in mechanical wave phenomena. Based on the inherent properties of mechanical waves, these signals can be categorized into two types: periodic and aperiodic signals, with the former being a continuously varying analog signal that can aptly be modeled using a sinusoidal function. This sinusoidal representation encapsulates three critical attributes: the angular frequency denoted by ω (which is inversely related to the wavelength λ , where $\lambda = 2\pi v/\omega$), the amplitude A (with intensity directly proportional to the square of A), and the phase angle ϕ . These three parameters essentially govern the distinct characteristics of any given sine wave.

The sound signal is an irregular signal in a long-time interval, but in a short time interval in the millisecond range, the sound signal can be regarded as a regular signal that does not change from time to time.

In the process of sound signal analysis, an essential step involves initially extracting the pertinent attribute parameters from the speech signal. These fundamental features correspond to the key elements of a sinusoidal function: the angular frequency, ω (which is intrinsically linked to the wavelength λ where $\lambda = 2\pi v/\omega$), the amplitude A (where the intensity is directly proportional to the square of A), and the phase angle ϕ [1].

To rephrase, when dealing with sound signals, identifying and isolating the critical aspects of the speech component begins with pinpointing the sine wave's core specifications—its angular frequency that determines the period or wavelength, the amplitude affecting the signal's energy level, and the phase that reflects the timing offset within the cycle.

Sound is a sine wave, and its important parameters include frequency, which represents the pitch of the sound, and spectrum, which represents the magnitude of the sound. In the field of speech signal processing, effective analysis depends on the selection of appropriate feature parameters, and the differences in these parameters often determine the different types of research methods used.

The method of analyzing the amplitude variation characteristics on the time axis is called time domain analysis, which is characterized by the intuitive visualization of the waveform structure and relatively low computational complexity. On the other hand, if the frequency component of the signal and its energy distribution are concerned, frequency domain analysis is involved, which can accurately map the energy distribution of the speech signal to the frequency spectrum, which has more far-reaching significance for understanding and processing the signal [1].

3. Research on the Source and Treatment Method of Noise

3.1. The Source of the Noise

In signal processing, any interference that is not part of the target wanted signal band is considered noise.

The sources of noise can generally be summarized into three aspects: the first type is derived from the external environment and is called ambient noise, the second type is generated inside mechanical equipment and is called equipment noise, and the third type is generated by the circuit components themselves and is called component noise.

First of all, environmental noise cannot be completely avoided, such as the mobile phone microphone when collecting human voices and other sounds, although in the professional recording environment can be reduced by external interference through soundproof walls and active noise cancellation equipment, but in daily scenes, such as the location of the mobile phone user, there must be other environmental noise such as human voices and rain equipment sounds, which will inevitably cause interference to the signal acquisition process.

Second, device noise is passed through microphones in the signal chain, Introduced by hardware such as sound cards, when the sound signal is converted into an analog signal from the physical space through a microphone, and then converted into a digital signal through the analog-digital and

electrical conversion, and after a series of processing is carried out inside the computer, it is converted back to the sound signal and played out, and the equipment quality of each link in this process directly affects the final sound effect, and the high-quality equipment can effectively reduce the accumulation of equipment noise in the process by reducing the signal change in each link, or reducing the number of links.

Finally, component noise originates from the basic components in the circuit. For example, resistors, diodes, and transistors produce thermal noise, semiconductor P-N junctions produce scattering noise and flicker noise, and power supply noise generated by charging and discharging power supply capacitors after converting AC power to DC power, and the noise intensity generated by these components will be reduced when operating at constant temperatures.

The higher the signal-to-noise ratio, the better the purity of the signal, the smaller the ratio of noise to the total signal, the smaller the impact of noise, on the contrary, the lower the signal-to-noise ratio, the lower the purity of the signal, the greater the ratio of noise to the total signal, the greater the impact of noise, the following is the calculation formula of signal-to-noise ratio.

$$SNR = 10 \log_{10} \frac{\sum_{n=0}^{N-1} s^2(n)}{\sum_{n=0}^{N-1} d^2(n)} \quad (1)$$

$\sum_{n=0}^{N-1} s^2(n)$ represents the energy of the effective signal, and $\sum_{n=0}^{N-1} d^2(n)$ represents the energy of the noise signal [1].

3.2. The Treatment of the Noise Signal

For dealing with different noises, different methods are usually adopted, such as isolating noise sources to reduce noise, such as recording in a specific environment to avoid environmental noise, using low-noise, low-power DC power supply, separating the power supply and signal ground to reduce the power supply noise in the component noise, adding heat dissipation design such as heat pump heat sink coolant and other designs to reduce the thermal noise in the component noise.

As a general filtering technology, the filter can filter all forms of noise at different levels, and the sound filter can pass through the specific frequency components of the sine wave of the analog electrical signal through the one-dimensional Fourier transform, and greatly attenuate the other frequency components. This frequency selection of the filter allows interference noise to be filtered out or for spectrum analysis.

In the field of electronic signal processing, filters are subdivided into various types according to their functional characteristics and application scenarios. First of all, from the perspective of signal processing, filters can be divided into two categories: analog filters and digital filters. Analog filters are suitable for processing continuous analog signals, while digital filters are designed for discrete digital signals and are filtered by software algorithms or digital hardware.

Within the realm of digital filters, a more refined categorization results in the primary distinction between Infinite Impulse Response (IIR) filters and Finite Impulse Response (FIR) filters. IIR filters are notable for their dependency on both the present and historical inputs, meaning that their output is influenced not just by the immediate input state but also by a series of past inputs. This distinctive trait introduces a feedback mechanism, enabling higher gains and steeper transitions between passbands and stopbands, albeit potentially leading to phase non-linearity issues.

On the other hand, FIR filters exhibit a fundamentally different operational principle. Their output strictly depends upon the current and a finite number of preceding input samples, thereby eschewing any form of closed-loop feedback. This structural simplicity renders FIR filters inherently easier to configure for precise linear phase characteristics, rendering them particularly suitable for applications where phase linearity requisites are less stringent or must be meticulously maintained.

Filters can be bifurcated into categories of passive and active filters based on whether they incorporate an internal power source for operation. Passive filters are constructed exclusively from non-powered components like resistors, capacitors, and inductors, and they do not necessitate an

external power supply to function economically. However, their frequency selectivity and stability might be constrained. When considering the transmission behavior across various signal frequency bands, passive filters can be further classified into five main types: low-frequency transmit filter, high-frequency transmit filter, band-transmit filter, band-reject filter, and all-pass filter.

The primary role of a low-transmit filter lies in facilitating the transmission of low-frequency or DC components during signal processing while attenuating higher-frequency components, notably noise and interference. On the contrary, a high-transmit filter gives precedence to the transmission of signals emanating from the high-frequency spectrum, thus inhibiting low-frequency and DC components.

A band-transmit filter selectively permits the transmission of signals within a predefined frequency band, effectively suppressing peripheral signals, noise, and interference occurring either below or above this designated band. One could conceptualize it as a composite structure incorporating both a low-transmit and a high-transmit filter.

The band-stop filter, often referred to as a notch filter, “blocks” the transmission of signals within a specific frequency band while allowing unhindered passage of signals from other bands. It can be viewed as an inverse application of the band-transmit filter.

Regarding the all-pass filter, its defining characteristic is that the amplitude of transmitted signals remains constant across all frequencies, and it is predominantly employed for signal phase adjustments.

Moreover, apart from functional classifications, filters can also be categorized based on their placement and construction within a system, namely into board-mounted filters and panel-mounted filters. An on-board filter, embedded onto the circuit board, offers the advantage of cost-effectiveness. Nonetheless, its potential downside includes increased susceptibility to coupling due to the limited electrical isolation between its input and output terminals. Practically speaking, electromagnetic interference (EMI) within the enclosure can directly affect the signal line, and unfiltered EMI can propagate outward via cables, causing radiation and diminishing the effectiveness of the filter. Furthermore, external EMI might couple into other board circuits prior to interception by the on-board filter, impacting the system’s sensitivity. Lastly, because of the restricted ground impedance associated with the filter, its high-frequency bypass capability tends to decrease, collectively contributing to the likelihood that on-board filters may exhibit suboptimal high-frequency filtering performance.

In contrast, panel filters are usually fixed directly to the metal shell of the shielded chassis, and due to the good electrical isolation between the input and output, the electromagnetic interference on the cable can be effectively filtered out before entering the chassis, and the filter's superior grounding conditions make the panel filter have significant advantages in filtering effect compared to the on-board filter. Whether it is protection against external interference or internal signal quality, panel filters show higher efficiency.

However, in recent years, neural network technology has begun to have many applications in the field of filters, bringing new solutions to improve filtering performance.

As a new type of filter that integrates artificial intelligence and signal processing technology, the neural network filter makes full use of the powerful learning ability and adaptability of the neural network and can identify and suppress the noise and interference in the signal more finely and intelligently. By training the neural network model, the filter can automatically learn and extract the characteristics of the signal to achieve accurate filtering of complex signals, overcoming some inherent problems in the design and application of traditional filters, such as noise suppression of nonlinear systems that are difficult to accurately model, dynamic filtering with high real-time requirements, etc., not only that, the neural network filter can also self-adjust and optimize according to the changes in the actual application environment, and continuously improve the filtering effect, which is particularly critical in many high-end signal processing applications.

Neural network filters are designed to cover many more advanced mathematical and technical applications, which are not covered in detail in this article.

4. Application of One-Dimensional Fourier Transform in Band-Pass Filters and Band-Stop Filters

In the field of signal processing, the one-dimensional Fourier transform plays a key role in the design and implementation of band-pass and band-stop filters as a powerful mathematical tool. Band-pass filters and band-stop filters are two important types of filters classified according to their frequency response characteristics, which allow signals in a specific frequency range to pass through (bandpass filters) or prevent signals in a specific frequency range from passing through (band-stop filters), respectively.

Firstly, the application of one-dimensional Fourier transform in bandpass filter design is reflected in the fact that the signal can be converted from the time domain to the frequency domain by performing the Fourier transform on the input signal, and the distribution of the signal on each frequency component can be clearly displayed. The designer can set the corresponding gain characteristics in the frequency domain according to the frequency range that needs to be passed, that is, the signal amplitude in this frequency band is amplified, while the signal amplitude in other frequency bands is reduced or eliminated. The filter coefficients designed in this way can obtain the filter weight function in the time domain after the inverse Fourier transform, and then construct the actual band-pass filter structure.

In the design of band-stop filters, the one-dimensional Fourier transform also plays a crucial role. Similarly, the input signal is first transformed by a Fourier transform, and then the frequency range that needs to be blocked is determined in the frequency domain, and the signal amplitude in this range is given a gain close to zero, while the amplitude of the signal in the rest of the frequency bands remains basically unchanged or slightly attenuated. Then, the modified frequency domain signal is converted back to the time domain by the inverse Fourier transform, and the filter weight function for the band-stop characteristic is obtained.

Similarly, low-pass filters and high-pass filters, as a lower substitute for band-pass filters and band-stop filters, they share the same technical principles as band-pass filters and band-stop filters.

In actual signal processing, the design of band-pass filters usually involves digital signal processing, rather than directly expressing them through the formulas of one-dimensional Fourier transforms. However, the operation of a band-pass filter in the frequency domain can be described by the concept of a one-dimensional Fourier transform.

Let the input signal be $x[n]$ and its one-dimensional discrete Fourier transform (DFT) be $X[k]$, then:

$$x[k] = \sum_{n=-\infty}^{n=\infty} x[n] \cdot e^{-j2\pi k \frac{n}{N}} \quad (2)$$

where N is the number of points of DFT, j is the imaginary unit, and k is the frequency index.

In the case of band-pass filters, their operation in the frequency domain is to selectively vary the amplitude value in a specific frequency range in the spectrum $X[k]$. Suppose we want to design a band-pass filter with a center frequency of f_c and a pass band width of B , then in the frequency domain, the filter transfer function $H[k]$ should satisfy:

$$H[k] = A(k) \left(\text{when } |f_k - f_c| \leq \frac{B}{2} \right) = 0 \left(\text{when } |f_k - f_c| > \frac{B}{2} \right) \quad (3)$$

where $A(k)$ is the specified gain (usually be one or some constant value) within the passband, $f_k = k / N * F_s$ is the frequency point after DFT, and F_s is the sampling frequency.

In order to achieve bandpass filtering, we first DFT convert the input signal to obtain $X[k]$, and then multiply $X[k]$ by the filter transfer function $H[k]$ to obtain the filtered frequency-domain signal $Y[k]$:

$$Y[k] = X[k] \times H[k] \quad (4)$$

Finally, the filtered frequency-domain signal $Y[k]$ is converted back into the time domain by inverse discrete Fourier transform (IDFT), and the filtered time-domain signal $y[n]$ is obtained:

$$y[n] = \frac{1}{n} \sum_{k=-\infty}^{k=\infty} Y[k] \times e^{\frac{2\pi kn}{N}} \quad (5)$$

When designing a band-pass filter, the window function, filter design algorithm (such as window function method, frequency sampling method, etc.) or the coefficients of the digital filter are usually used for filtering. In the field of digital signal processing, the specific implementation of the design of band-pass filters usually uses the design theory of digital filters, such as Butterworth filters, Chebyshev filters, or elliptical filters.

In the actual implementation process, whether designing a band-pass filter or a band-stop filter, the application of one-dimensional Fourier transform simplifies the design process of the filter and makes it better meet the specific frequency screening needs in practical applications. In digital signal processing, fast Fourier transform (FFT) is often used for efficient operation, which greatly improves the efficiency and accuracy of filter design and implementation.

5. Application of One-dimensional Fourier Transform in FIR Digital Filters

In digital signal processing, the one-dimensional Fourier transform plays an indispensable role in the design and implementation of FIR (finite impulse response) digital filters. FIR filters are popular in many applications due to their stability and linear phase characteristics, and the design process takes full advantage of the properties of the one-dimensional Fourier transform [2].

In the design stage of FIR filters, the key role of one-dimensional Fourier transform is to convert the characteristics of the filter in the time domain to the frequency domain for analysis and design. First, the designer defines the desired frequency response characteristics, i.e., the gain curve of the filter at different frequencies, which is often the core element of the filter at the beginning of the design [2].

In the actual design process, the inverse transform of the one-dimensional Fourier transform is used to convert the characteristics of the designed filter in the frequency domain back to the time domain, so as to obtain the unit impulse response $h[n]$ of the filter. The designer can use the window function method or other optimization algorithms to determine the specific value of $h[n]$ on the basis of satisfying the design index. In the window function method, the designer first calculates the frequency response of the ideal filter, and then truncates and smooths it by selecting the appropriate window function to obtain a finite-length $h[n]$ sequence.

Through the one-dimensional Fourier transform, designers can also easily analyze the frequency characteristics of the filter, such as calculating important parameters such as pass band flatness, stop band attenuation, and transition band width. In addition, since the linear phase characteristics of FIR filters are critical for many applications, the Fourier transform makes it easy to ensure that the filter has a linear phase relationship at all frequencies, ensuring that the timing information of the signal is not distorted by filtering [3].

In the implementation stage, the Fast Fourier Transform (FFT) algorithm based on one-dimensional Fourier transform is widely used in the calculation process of FIR filters, which greatly improves the calculation speed and efficiency of filters in real-time systems. In practical applications, the FIR filter can update the output signal in real time through convolution operation every time it receives a new sampling point, and the convolution operation is equivalent to the point multiplication operation in the frequency domain, which undoubtedly benefits from the convenience provided by the one-dimensional Fourier transform.

In summary, the one-dimensional Fourier transform runs through the design and implementation of FIR digital filters, from conceptual design to specific implementation, from filter characteristics definition to efficient calculation, and plays a core supporting role, which greatly promotes the development and application of digital signal processing technology in many fields such as communication, audio processing, and image processing.

6. Application of One-dimensional Fourier Transform in Windowing Function

In the field of signal processing, the combination of one-dimensional Fourier transforms, and windowing function is widely used and critical, especially in signal analysis, filtering, and feature extraction. The windowing function is a special data processing technique that aims to localize the signal in the time domain before the Fourier transform to improve the converted spectral characteristics [4].

The one-dimensional Fourier transform converts the time-domain signal into a frequency-domain representation, revealing the frequency composition of the signal. However, for non-periodic or finite-length signals, the direct application of the Fourier transform results in spectral leakage and edge effects, as the Fourier transform defaults to an infinitely long and periodically repeating signal. To solve this problem, the windowing function comes in handy.

In practice, the windowing function weights the signal within its defined domain, so that the signal gradually approaches zero at both ends, thus forming a "window", and the signal in the window is treated as a local "complete" periodic signal for Fourier transform. Common windowing functions include rectangular window, Hamming window, Heming window, Blackman window, etc., each window function has its own unique spectral characteristics, and the appropriate window function type can be selected according to actual needs.

The one-dimensional Fourier transform of the windowed signal will show a smoother and more focused spectrum, which can not only effectively suppress the spectral leakage, but also provide more accurate frequency component information. For example, in signal analysis, windowing can be used to more accurately locate the dominant or harmonic frequency of the signal, and in filter design, windowing can optimize the frequency response of the filter, especially when designing finite impulse response (FIR) filters.

In summary, the combination of one-dimensional Fourier transforms, and windowing function is of great significance in signal processing, which effectively solves the non-ideal conditions encountered in practical applications and enhances the accuracy and reliability of signal analysis and processing. Through the reasonable selection and application of windowing functions, based on Fourier transform, it can better serve the purposes of signal identification, detection, and encoding [5].

7. Conclusion

The one-dimensional Fourier transform has proven its effectiveness and utility in denoising applications for microphone filters. By converting the signal from the time domain to the frequency domain, the Fourier transform makes the frequency component analysis of the signal intuitive and accurate, providing a theoretical basis and practical method for noise suppression and signal recovery. By combining modern signal processing techniques, such as neural network filters, the performance of denoising technology can be further improved to accommodate more complex application scenarios. The application of window function also significantly improves the accuracy and reliability of signal processing. In the future, with the continuous progress of signal processing technology, the microphone filter denoising technology based on one-dimensional Fourier transform will continue to be optimized to meet the growing demand for high-quality audio signal processing.

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