Optimistic locating method of transformer voiceprint based on beam and delay algorithm

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Abstract. In view of the on-line inspection mode of transformer voiceprint information during operation, an optimistic location method is proposed to assist the acoustic print diagnosis for fault identification, location and further evaluation work. Firstly, according to the directional location algorithm of antenna array (Beamforming), traditional directional location is extended to general spatial location. Then on the basis of TDOA(Time Difference Of Arrival) algorithm of time domain signal characteristic analysis, integrated weighted processing and data centralization, on the one hand, reduce the edge value, on the other hand, correct the trend of bad data, and improve the existing cross-correlation operation. The simulation results show that the optimized beam-based and weighted TDOA algorithm has excellent performance in 2D localization of transformer voiceprint. These research results can provide a basis for transformer online monitoring and voice print diagnosis.

1. Introduction

Transformer early fault identification is very important for transformer state maintenance and has important guiding significance for transformer maintenance. How to carry out power transformer fault live detection is a hot and difficult research topic in the industry. Common transformer detection and diagnosis methods including short-circuit impedance method, frequency response analysis method, low-voltage pulse method and frequency response method are difficult to meet the requirements of accurate online monitoring. At present, the substation is mainly detected by manual or machine inspection, including power transformer appearance, oil level meter, infrared temperature measurement, noise monitoring and other data to determine whether the power equipment is in a healthy state of operation. The vibration characteristics of winding can be used to judge whether the running state is normal or abnormal.

Currently, there is no unified or excellent standard for transformer sound print diagnosis system. The voiceprint of transformer in substation is mainly at the monitoring and subjective level and without very reliable system calculation example or data support, still in the stage of continuous development and improvement. Literatures and researches about transformer fault diagnosis based on the feature of sound pattern are relatively few. In the existing researches, a set of sound pattern monitoring method, device, equipment and storage medium scheme being designed comprehensively in transformer sound pattern monitoring and diagnosis. The field calibration method based on transformer voiceprint monitoring device is put forward, which is helpful to verify and analyze the monitoring results accurately and reliably. However, most of the researches focus on the feature extraction, analysis and later diagnosis and monitoring of acoustic signals, seldom considering specific sound source location methods.
On account of the above problems in transformer voiceprint detection diagnosis, in this paper about voice print diagnosis of transformer, how to implement using multiple microphone receives the signal positioning problem of transformer noise is studied and optimized. In order to safely, quickly and accurately identify the noise source location, forming the sound source location and sound intensity cloud image at the same time. It can intuitively display the power equipment noise intensity, location and other characteristic quantities and improve the efficiency and accuracy of transformer inspection [1].

2. Materials and Methods

2.1 Principle and algorithm of transformer sound source location

The ability of sound source location is to find the most possible position of sound source point in space, including the judgment of three-dimensional data composed of three vectors: horizontal, vertical and distance of sound source. From a biological point of view, sound source discrimination can be realized mainly based on: direction estimation according to the time delay of sound to the ears, the frequency spectrum effect of bilateral auricle and life experience model. In this paper, the location method is ground on the basic principle to realize sound source identification and location.

2.1.1 Beamforming algorithm

Beamforming originally refers to the classical antenna technology used for directional signal transmission or reception, in which the signals received by a multi-antenna array are weighted together to form the desired signal. The later developed algorithm that uses weighting to form a beam for positioning is based on the following formula [2].

$$B(t, \theta) = \frac{1}{N} \sum_{n=1}^{N} k_n P_n (t + \tau)$$

Upper formula: $B(t, \theta)$ is the position of vibration noise sound pressure signal corresponding to the maximum frequency band of sound pressure level. $P_n$ is the vibration noise sound pressure signal corresponding to the maximum frequency band of sound pressure level. $\theta$ is the focusing direction of sound source. $k_n$ is the weight vector of the vibration noise sound pressure signal corresponding to the maximum frequency band of sound pressure level. $N$ is the number of microphones. $\tau$ is the delay compensation.

Common beam algorithm uses a group of sensors arranged in a straight line. The signal information between two acoustic sensors will give a fuzzy direction information. On this basis, other sensors will clear and optimize the fuzzy information and its precision is determined by the total number of sensors. According to the requirements, in order to identify the space position rather than just the direction of the positioning, firstly we need to add the number of microphones, which should be greater than or equal to the total four. Furthermore, we need to change the direction of the original parameters modified into three-dimensional parameters and then change the way the microphone array like center quadrangle, spiral, concentric circles or etc. Making the differences between the signal quantity is able to response the exact location of the sound source.

2.1.2 Time delay difference method

Commonly positioning technology such as TDOA, its principle is relatively simple, usually through the sound to the sensor time difference analysis calculation, namely the time difference estimation. Firstly, the phase difference or delay is calculated by the signal data obtained by the acoustic sensor. Based on the obtained time difference, known location of microphone array and the sensitivity of different directions, the algorithm can be further designed to locate the most likely position of sound source [3].

In practical system design, by comparing the time difference between signals arriving at each microphone, a hyperboloid with two microphones as the focus and the distance difference as the long
axis can be made. Through increasing the total number of microphones, the intersection line can be further compressed to the point. This method has a small amount of calculation and is easy to process, but its accuracy and anti-interference ability are relatively weak.

2.2 TDOA Localization algorithm

2.2.1 Correlation analysis calculation

The most important step of TDOA is to calculate the time delay. The basic calculation method of the time delay using cross-correlation is defined as: infinite integral of the first function multiplied by the second function after complex conjugation and translation [4].

\[ R_{hh}(x) = \int_{-\infty}^{+\infty} f^*(x^-x)h(x')dx' \]
\[ R_{hh}(x) = \int_{-\infty}^{+\infty} f^*(x')h(x'+x)dx' \]

(2)

With the help of fast Fourier transform and inverse transform, the cross-power spectrum \( X_{12} \) and cross-correlation function \( R_{12} \) can be obtained.

\[ X_{12}(\omega) = X_1(\omega)X_2^*(\omega) \]
\[ R_{12}(\omega) = \int_{-\infty}^{+\infty} X_{12}(\omega)e^{j\omega}d\omega \]

(3) (4)

In formula, \( X(\omega) \) is the FFT result of one set of signals, multiplied by the conjugate of the FFT result of another set of signals.

2.2.2 Weighted correlation analysis and centralized processing

From the point of view of discrete signal analysis, two sets of signals with sampling points (whose amount is M) are extracted from the set constituted by total sensors and written in discrete basic form. The established cross-correlation index is equivalent to the operation of the following formula [5].

\[ U(\mu) = \sum_{i=0}^{N} x_i(i-\mu) \cdot x_2(i) = \sum_{i=0}^{N} x_1(i) \cdot x_2(i) \]

(5)

In order to reduce the blank error generated by the edge values in the calculation of the cross-power spectrum, the weight function was used to appropriately reduce the weight of the values at both ends of the original discrete signal in the cross-correlation calculation. To optimize the calculation process, we can simply adopt cubic interpolation function with adjustable parameters (or other functions according to the situation).

\[ X_{12}(\omega) = G(\omega)X_1(\omega)X_2^*(\omega) \]

(6)

Besides, when desiring to obtain a planar representation of the region based on sound source localization, in addition to the foundation anchor point source, each position around the location for centralized processing also needed. Owing to the TDOA operation mode on the geometric similar to hyperbola, the extension of hyperbolic acoustic source points should be close to the orientation [6].
3. Results and Discussion

3.1 Analysis of measured data

Based on this principle, the actual transformer noise is simulated and run. In Figure 1, the overall layout design of this paper assumes that the distance between the transformer sound source plane and the plane where the microphone is located is 2m. The sound source of each frequency band is monophonic source and directly superimposed. Keeping whole positioning space in a vertical plane of 5m*5m. After full filtering, picking signal sequences with signal-to-noise ratio no less than 2:1 and noise time domain characteristics not completely overlapping. Through cross-correlation analysis and positioning, the original cross-correlation analysis diagram is shown in Figure 2.

![Figure 2. Primitive Cross-correlation Result](image1)

Weighting signals and reducing the edge weight of contrast signal group by introducing weighted function-g(x). Improved correlation function is shown as follows. It can be seen in Figure 3, that the low correlation area is compressed toward the center while the relative size is weakened.

![Figure 3. Improved Cross-correlation Result](image2)

Furthermore, introducing a centralized method to construct a complete positioning image. The correlation degree value of each point is used as the probability of sound source. As what shows in Figure 4, hyperbolic branch is more close to the location point.

![Figure 4. Visual Plane Location](image3)
4. Conclusions
This paper studies the principle of traditional beamforming and the location algorithm of time difference. Based on the correlation analysis of TDOA, the weighted improvement on the edge of the signal sequence is carried out. And the complete location is visualized simply by using the centralized method. It can be seen from the simulation results that compared with the original correlation function, the weighted analysis method can obviously improve the resolution and contrast. Especially when the actual delay does not reach the edge value, it can further improve the accuracy of the correlation function, which has a certain value in practical application.

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Reference
[1] Rhodri Bevan, L.T., Jie, Z., Nicolas, B. (2019) Experimental Quantification of Noise in Linear Ultrasonic Imaging[J]. IEEE TRANSACTIONS ON ULTRASONICS, FERROELECTRICS, AND FREQUENCY CONTROL, 66(1): 79-90.
[2] Charalampos Dimoulas, A. (2016) Audiovisual Spatial-Audio Analysis by Means of Sound Localization and Imaging[J]. IEEE TRANSACTIONS ON MULTIMEDIA, 57: 1969-1976.
[3] Knapp, C., Carter, G. (1976) The generalized correlation method for estimation of time delay. IEEE Transactions on Acoustics, Speech, and Signal Processing, 24(4): 320 – 327.
[4] Duoli, Z., Xiulei, S., Yukun, S. (2016) The Implementation of Large FFT Convolution on Heterogeneous Multicore Programmable System[J]. IEEE International Conference on Integrated Circuits and Microsystems, 69: 349-353.
[5] Velasco, J., Martín-Arguedas, C.J., Macias-Guarasa, J. (2016) Proposal and validation of an analytical generative model of SRP-PHAT power maps in reverberant scenarios. Signal Processing, 119: 209-228.
[6] Salvati, D., Drioli, C., Foresti, G.L. (2018) Sensitivity-based region selection in the steered response power algorithm. Signal Processing, 153: 1-10.