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Application of Adaptive Finite Focus Beamforming Method for Localization of Low-frequency Transformer Sound

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Summary **— In this paper, the developed adaptive linearly constrained minimum variance finite focus beamforming method is used to increase the accuracy and resolution of low-frequency transformer sound source localization. In the first step, simulations on virtual arrays are conducted to define the array size and compare the influence of three different beamforming algorithms on sound localization accuracy and resolution. In the second step, measurements of the regulating transformer as a noise source are conducted in the semi-anechoic chamber. A large rectangular microphone array, with 90 microphone positions spaced 0.44 meters apart, is used for that purpose. The developed algorithm is compared with infinite and finite focus beamforming for the localization of total noise and noise at specific noise harmonics.**

Keywords **— finite focus beamforming, adaptive beamforming, rectangular microphone array, regulating transformer.**

I. Introduction

The conventional beamforming technique for sound source localization in acoustic cameras is applicable in far-field and at higher frequencies, mainly above 500 Hz. However, dominant frequency components of transformer noise are often below that frequency. The problem is highlighted in the literature [1], [2], and [3], where an acoustic camera is used for the localization of low-frequency transformer sound. The hotspot localization results show low spatial resolution in the frequency range below 300 Hz. That makes the method inapplicable for localizing sound sources in a transformer. Instead, knowing the exact distance from the source to the beamforming array, a finite-focus method [4] can be applied. Additionally, adaptive beamformers [5] can be used by choosing weights based on the statistics of the received data.

Besides beamforming, there are three significant sound source localization techniques: nearfield acoustic holography (NAH), statistically optimized nearfield acoustic holography (SONAH), and direct acoustic intensity measurements. Another noise source identification technology that combines the far-field beamforming method with the near-site acoustic holography can be applied to the transformer for a higher resolution [6]. The application of direct acoustic intensity measurements to localize noise on 16 MVA and 75 MVA transformers is explained in [7] and [8], respectively.

This study pointed out the possibility of the beamforming met hod application for localizing low-frequency transformer sound. The motivation for the work is to develop an easily applicable and accurate method with high resolution at low frequencies at which the transformer operates. A simple application of the method is achieved with only one microphone and the possibility of measurement automatization without requiring measurements at very short distances from the surface, as in [6–8]. Sound localization accuracy is achieved using an unconventional finite-focus method, and increased resolution is achieved using an adaptive beamformer, thus avoiding the conventional acoustic camera problems [1-3]. Compared to the previous research, this work developed a new, improved method that needs only one microphone and combines finite-focus and adaptive beamforming with a known distance from the source to localize low-frequency sound.

In the paper, a brief overview of the beamforming theory is given. The three different beamforming algorithms are simulated on the specified microphone array. A wooden frame for positioning microphones along the beamforming array is presented, and the measurement procedure is described. Analysis of the measurement in the semi-anechoic room is carried out. In the end, conclusions and suggestions for future research are given.

II. Theoretical Background

In the beamforming technique, the amplitude and the phase of the sound pressure over an array of microphones are measured. The technique maximizes the total summed output for the sound coming from the specified direction while minimizing the sound coming from another direction. It is used in the far field of the sound source and is more suitable for higher frequencies. The frequency and the dynamic range of beamforming measurements depend on the array type and size. The beamforming has the disadvantage of low spatial resolution in localizing a noise source, especially at low frequencies. The advantage is that it can image distant and moving sources. The two types of beamforming are infinite-focus distance and finite-focus distance beamforming.

A. Infinite focus beamforming

The plane waves are assumed for the infinite focus beamforming, and the spherical waves are assumed for the finite focus beamforming. Beamforming theory is explained in more detail in

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[4], [9]. If the planar array consists of L microphones at locations $(x_l, y_l, l = 1, ..., L)$ in the x-y plane, and the measured signals p_l are individually delayed and then summed, the output of the array is:

$$p(\boldsymbol{n}, t) = \sum_{l=1}^{L} w_l p_l(t - \Delta_l(\boldsymbol{n})) \tag{1}$$

where w_l is the weighting coefficient, which reduces the importance of the signals coming from the array edges. The quantity $\boldsymbol{n}$ is the unit vector in the direction of the maximum sensitivity of the array, and the time delays Δl are chosen to maximize the array sensitivity in the direction $\boldsymbol{n}$, and it is defined as:

$$\Delta l = \frac{\boldsymbol{n} \cdot \boldsymbol{r}_l}{c} \tag{2}$$

where the vector $\boldsymbol{r}_l = (x_l, y_l)$ and the c is the speed of sound. The output of the beamformer in the frequency domain analysis at angular frequency ω is:

$$P(\boldsymbol{n}, \omega) = \sum_{l=1}^{L} w_l P_l(\omega) e^{-j\omega\Delta_l(n)} = \sum_{l=1}^{L} w_l P_l(\omega) e^{-j\boldsymbol{k}\cdot\boldsymbol{r}_l} \tag{3}$$

where $\boldsymbol{k} = -k\boldsymbol{n}$ is the wave number vector of a plane incident from the direction $\boldsymbol{n}$. The beamforming in the frequency domain provides the possibility to process independent frequencies. It is less computationally intensive and more suitable for steady-state situations. The time domain beamforming is more suitable for rapidly changing environments.

B. Finite focus beamforming

Finite-focus beamforming, assuming a spherical wave, follows a similar analysis. The various microphone delays should align for the array to focus on a point source at a finite distance. Equation (3) still applies, but the delay Δl is defined as:

$$\Delta l = \frac{|\boldsymbol{r}| - |\boldsymbol{r} - \boldsymbol{r}_l|}{c} \tag{4}$$

where $\boldsymbol{r}$ is the vector location of the source from an origin point in the same plane as the array, $\boldsymbol{r}_l$ is the vector location of microphone l in the array with respect to the exact origin, and $|\boldsymbol{r} - \boldsymbol{r}_l|$ is the scalar distance of microphone l from the source. An example of the beamforming localization principle is shown in Figure 1.

C. Adaptive linearly constrained minimum variance beamforming

A principle of adaptive beamforming is that the weights are chosen based on the statistics of the received data so that the signal-to-noise ratio (SNR) can be improved. That approach enables distinguishment between desired and interference signals. A linearly constrained minimum variance (LCMV) beamforming is first introduced in [10]. The algorithm is based on the minimum mean square error used to determine array weights. The optimal weights of the LCMV beamformer are calculated as follows [9]:

$$\boldsymbol{w}_{opt} = S^{-1}C(C^H S^{-1} C)^{-1}\boldsymbol{R} \tag{5}$$

where S is the sum of the noise and interference covariance matrix, C is the constraint matrix, H is the transpose conjugate of a vector, and $\boldsymbol{R}$ is the signal gains due to the constraint.

The advantages of the LCMV beamforming are that it has higher spatial resolution than the conventional beamformer, it is nulling in the direction of interference sources, sidelobes are smaller than in the conventional beamforming, and it only requires the direction of arrival to maximize SNR. Disadvantages are low convergence rate and susceptibility to self-nulling.

III. Methodology And Measurement

Three different algorithms are tested in this work:

- Infinite focus (IF) beamforming (conventional)
- Finite focus (FF) beamforming
- Finite focus (FF) adaptive beamforming (developed)

A. Testing Algorithms on a Virtual Array

A virtual array of 9×10 microphones is created in MATLAB, as shown in Figure 2. The distance between microphones is set to 0.44 m. The total size of an array is 3.56 m in height and 4 m in length. The size is selected so that the wooden frame with holes for microphone positions can be placed in a semi-anechoic chamber for experimental verification.

Simulated point sound sources are located at a finite distance of 2 m. The first source is at 100 Hz at the spherical coordinates [30°, 20°]. The second source is at 300 Hz at coordinates [-20°, 0°]. Spherical coordinates are marked as [azimuth, elevation]. The amplitude of simulated sound pressure is set to 0.1 Pa, in decibels equal to 70.97 dB RMS.

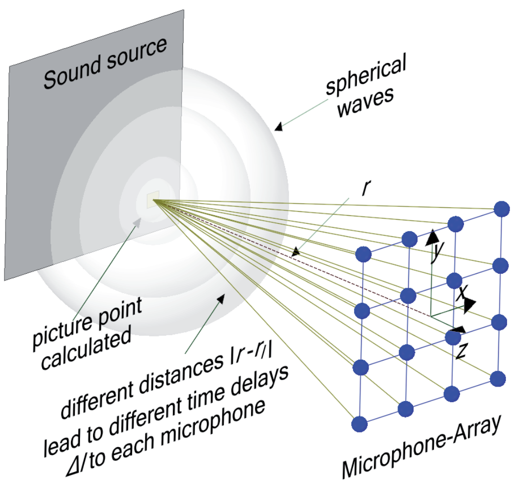

Fig. 1. Finite-focus beamforming localization principle

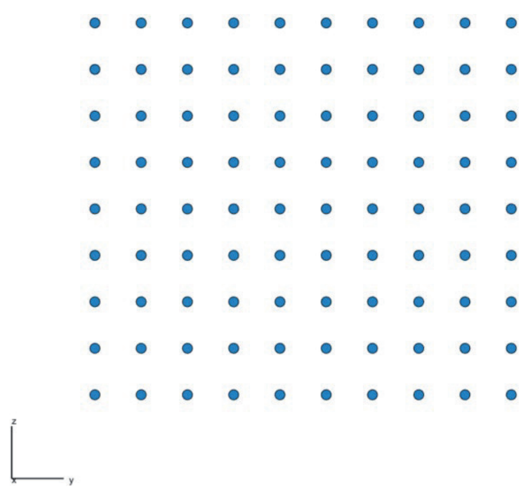

Fig. 2. Virtual array of 90 microphones in MATLAB

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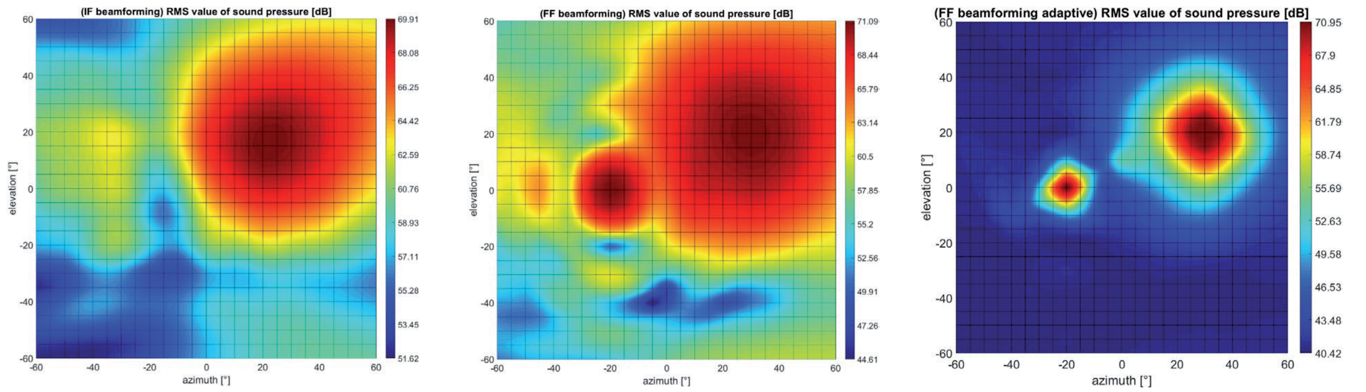


a) b) c)

Fig. 3. Localization of two simulated sound sources at a distance of 2 meters using a) infinite focus beamforming, b) finite focus beamforming, c) finite focus adaptive beamforming

Localization of two simulated sound sources from specified angles using IF beamforming and FF beamforming is shown in Figure 3a and 3b. Application of IF beamforming for the finite distance point source decreases localization accuracy. The 300 Hz source is not localized, and the coordinates of the 100 Hz source are localized with an error. The center of the source is located at approximately [22°, 17°] instead of assigned [30°, 20°].

FF beamforming does not increase the spatial resolution. It is roughly the same for both algorithms by comparing localizations of the 100 Hz source. On the other hand, localization accuracy compared to the IF beamforming is higher. The sources are localized at the exact coordinates.

A higher resolution is achieved by using a developed adaptive FF beamforming algorithm. Compared to the conventional FF beamforming algorithm, spatial resolution is increased, as shown in Figure 3c.

B. Measurement Procedure in the Semi-Anechoic Chamber

Measurements are conducted in the semi-anechoic chamber on the regulating transformer as a noise source. The dimension of the transformer is 1.3 m in width, 1.3 m in length, and 1.6 m in height.

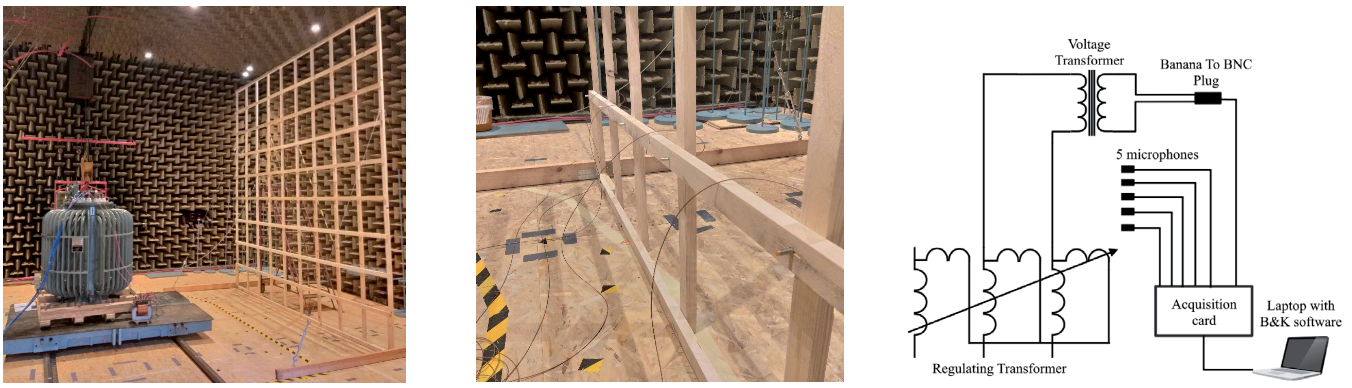
Table I.

Nominal Data Of The Regulating Transformer

Rated power	160 kVA
Rated output voltage	0-500 V
Rated output current	185 A
Frequency	50 Hz
Number of phases	3
Mass	1150 kg

The measurement procedure consisted of a total of 18 measurements. Five sound pressure and voltage signals are recorded in each measurement using an acquisition card and laptop with B&K software. A voltage transformer is used to lower the voltage in order to measure it using an acquisition card. The measurement setup diagram is shown in Figure 4c.

A voltage signal aligns pressure signals as if measured at the exact moment. This technique enables the application of the beamforming algorithm without having 90 microphones for each measurement position. The regulating transformer is measured at 2 m from the source and 2.5 m from the source. Two different distances are used as the criteria to validate the method. The sound source should be localized at the exact coordinates regardless of the distance.



a) b) c)

Fig. 4. a) Regulating transformer and a wooden frame for a beamforming array in a semi-anechoic chamber b) placement of microphones used for measurements of sound pressure c) measurement setup diagram

Nominal data of the regulating transformer are given in Table I. The measurement setup with the regulating transformer and the wooden frame is shown in Figure 4a. Construction for a beamforming array consists of a total of 90 holes. Five microphones used for sound pressure measurements are shown in Figure 4b.

IV. Results

Voltage signals of all 18 measurements and sound pressure signals of 18 microphone measurements at the first position before alignment are shown in Figures 5a and 5c, respectively. The same signals after the alignment are shown in Figures 5b and 5d. The figure shows that the voltage signal is used as a reference signal,

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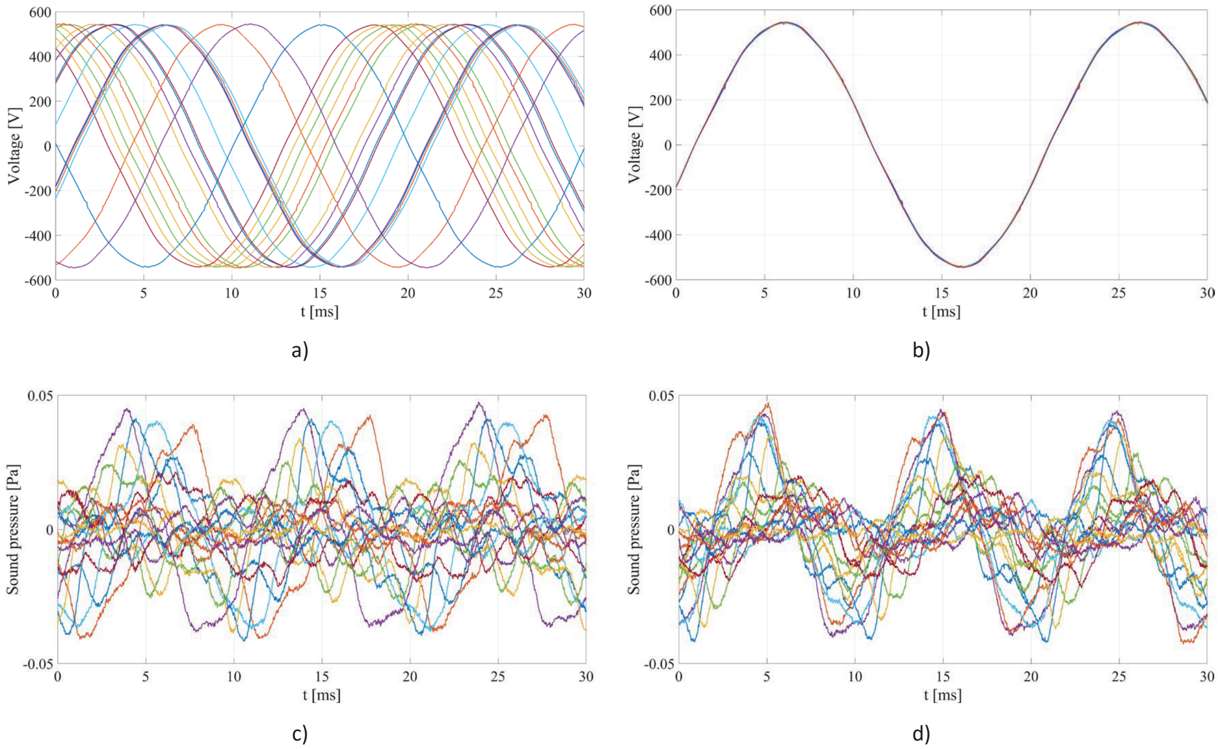

Fig. 5. a) Voltage signals of all 18 measurements before alignment b) after alignment c) sound pressure signals of microphone 1 of all 18 measurements before alignment d) after alignment

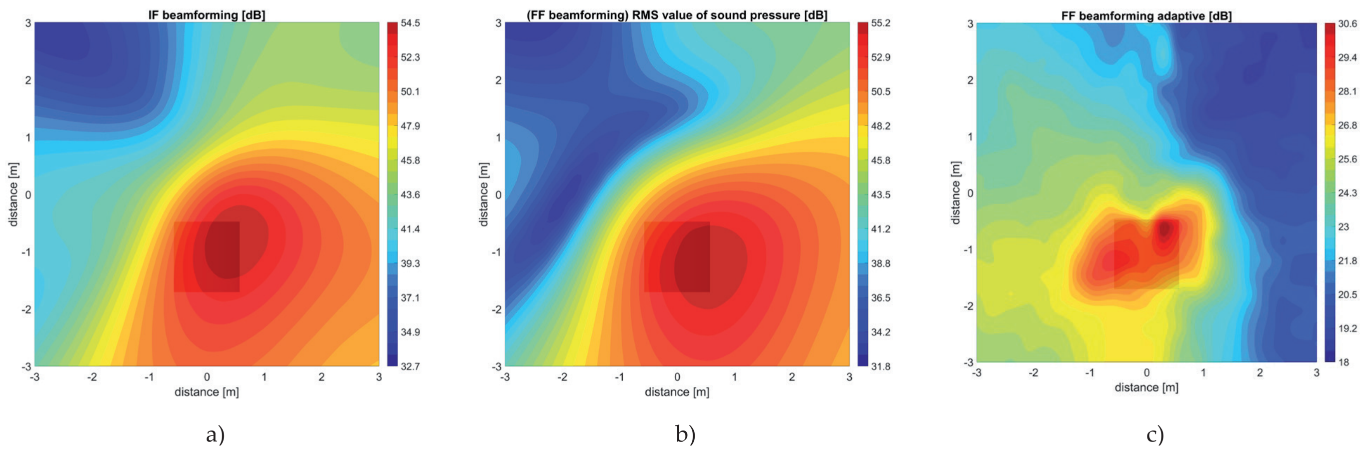

Fig. 6. Localization of regulating transformer noise at a distance of 2 m from the source using a) conventional infinite focus beamforming, b) finite focus beamforming, c) finite focus adaptive beamforming

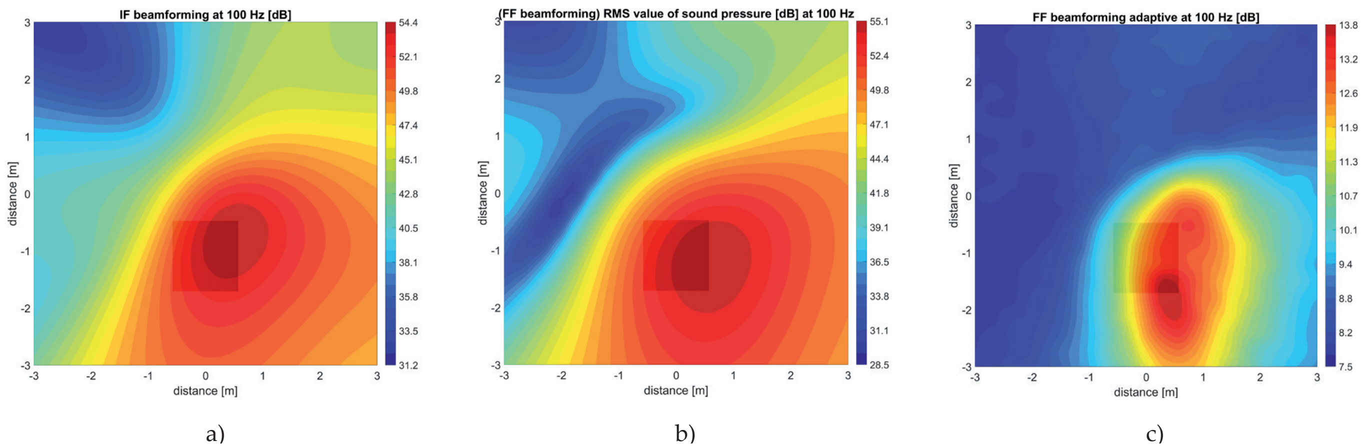

Fig. 7. Localization of regulating transformer noise at a distance of 2 m from the source at 100 Hz frequency using a) conventional infinite focus beamforming, b) finite focus beamforming, c) finite focus adaptive beamforming

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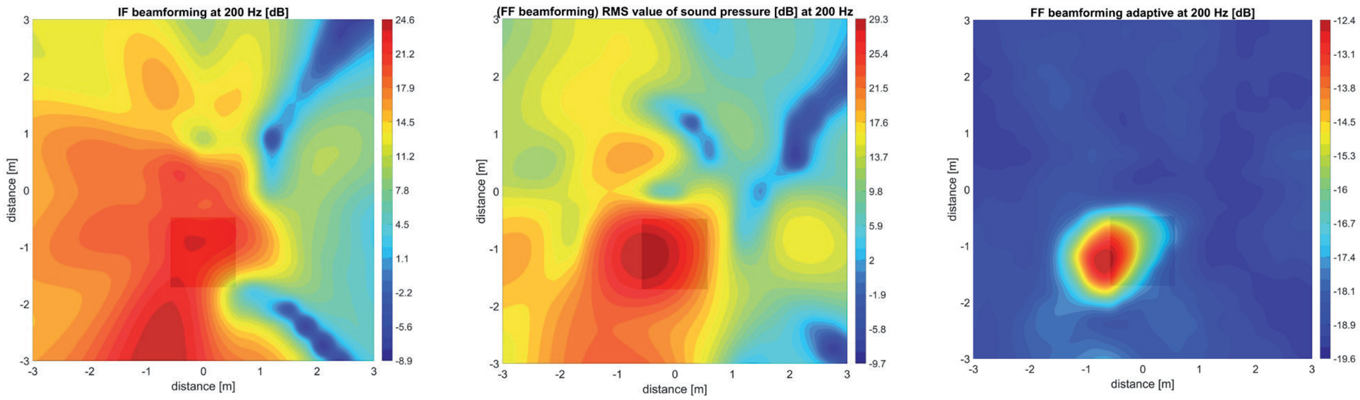
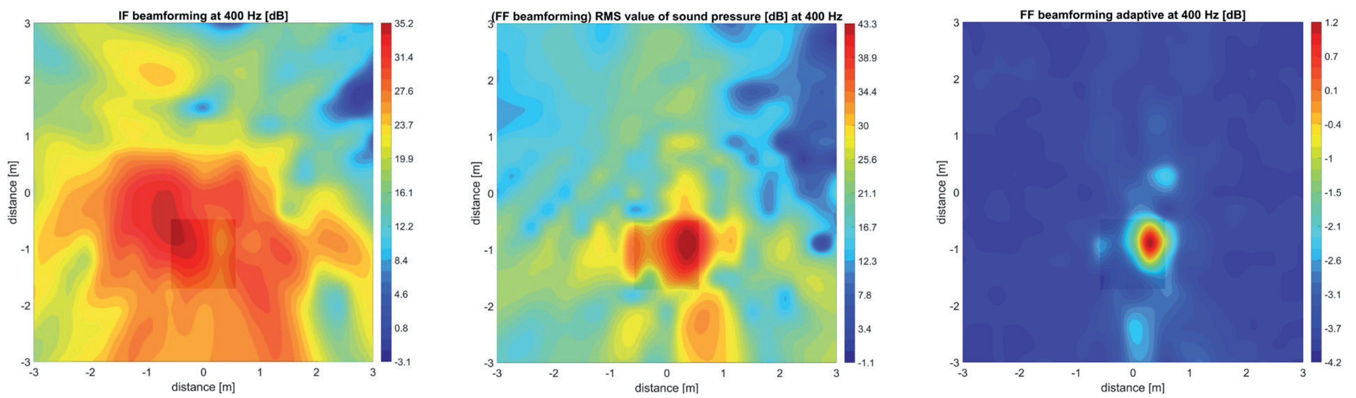


a) b) c)

Fig. 8. Localization of regulating transformer noise at a distance of 2 m from the source at 200 Hz frequency using a) conventional infinite focus beamforming, b) finite focus beamforming, c) finite focus adaptive beamforming

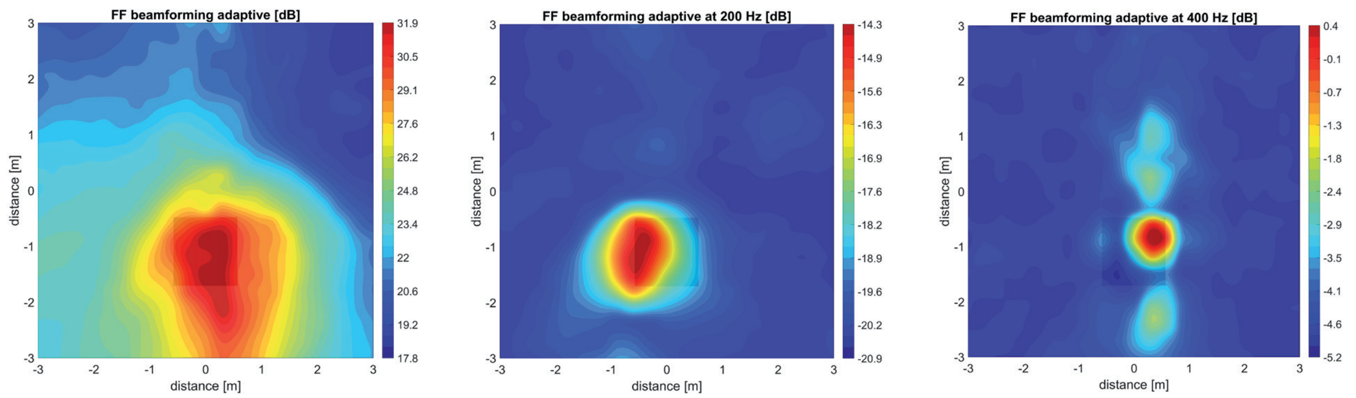
and sound pressure signals are aligned accordingly. The localization of regulating transformer total noise for three beamforming algorithms at a distance from 2 m is shown in Figure 6. The primary noise source is localized at the right side of the regulating transformer using IF beamforming and FF beamforming. As the simulation results from Figure 3 show, the FF beamforming algorithm results in higher accuracy. FF adaptive algorithm localized sound dominantly at the upper right corner and at the regulating transformer's left side. Resolution is increased compared to non-adaptive algorithms.

The localization of regulating transformer noise for three beamforming algorithms at significant transformer noise harmonics (100 Hz, 200 Hz, etc.) is also conducted. Figure 7 shows the localization at 100 Hz component at 2 m from the source. Using FF algorithms (conventional and adaptive), the sound source is localized at the bottom right side of the regulating transformer, and by using the IF algorithm, the sound is localized at the upper right side.The localization of regulating transformer noise at a distance from 2 m at a frequency component of 200 Hz is shown in Figure 8. The sound source is localized at the left side of the regulating transformer using FF algorithms (conventional and adaptive). Localization of the sound source for IF beamforming is more in the middle and at the bottom of the regulating transformerThe localization of re-



a) b) c)

Fig. 9. Localization of regulating transformer noise at a distance of 2 m from the source at 400 Hz frequency using a) conventional infinite focus beamforming, b) finite focus beamforming, c) finite focus adaptive beamforming



a) b) c)

Fig. 10. Finite focus adaptive beamforming at a distance of 2.5 meters from the source for localization of regulating transformer a) total noise b) noise at 200 Hz frequency c) noise at 400 Hz frequency

gulating transformer noise at a distance from 2 m at a frequency component of 400 Hz is shown in Figure 9. The sound source is localized at the upper right side of the regulating transformer using FF algorithms (conventional and adaptive). Localization of the sound source for IF beamforming is at the upper left side of the regulating transformer.

The highest resolution is obtained using the FF adaptive beamforming algorithm. Results from measurements at 2 m are compared with measurements at 2.5 m. Total noise from Figure 6c, 200 Hz frequency from Figure 8c, and 400 Hz from Figure 9c are compared with localizations from Figure 10.

A comparison of total noise at two distances shows that adaptive beamforming algorithms are unsuitable for total noise signals. On the contrary, applying the developed method (FF adaptive beamforming) to the filtered single-frequency components (noise harmonics) increased the spatial resolution. Thus, the filtering in combination with an adaptive algorithm in the developed method increases the resolution. Also, the sound source is more accurately localized compared to the conventional infinite focus beamforming.

V. Conclusion

In this paper, three different beamforming algorithms are tested on simulated sources and measurements on the regulating transformer used as a noise source in a semi-anechoic chamber. FF beamforming method with the known distance of the source compared to the conventional IF beamforming method does not increase the resolution. On the other hand, localization accuracy is increased. Except for those two methods, the adaptive FF beamforming method is developed and investigated. This method increases spatial resolution and localization accuracy for low-frequency sounds. A disadvantage of this method is that it is unsuitable for total noise signals. However, transformer noise is at 100 Hz and its harmonics, so it is not a problem to localize specific frequencies at which higher noise levels will occur.

In future work, the automatization of measurements in the form of a robot following the microphone position path can be used. The presented method allows the automatization because a voltage signal aligns pressure signals as if measured at the exact moment. Surface vibration measurement using a laser Doppler vibrometer can also be added to the automatized robot setup to obtain a complete vibroacoustic picture of the transformer, as a continuation of the authors' previous work [11]the vibrations on the surfaces of the tank wall, stiffeners, and the cover of a 5 MVA transformer experimental model were measured during open-circuit and short-circuit transformer tests. Vibration measurements of a transformer tank side were conducted at discrete points using two different voltage sources in no-load test. Using interpolation functions, the RMS values of acceleration and vibration velocity are visualized and compared for each considered measurement configuration (no-load and load tests and two different excitation sources.

Theoretically, even a higher array size can be achieved, which results in increasing spatial resolution. There is only a limit to the space size in which the sound pressure for beamforming is measured. The following step is to test the method in echoic rooms, where applying beamforming and especially adaptive algorithms can be challenging.

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