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“VIRTUAL ARRAYS”, A NOVEL BROADBAND SOURCE LOCALIZATION TECHNIQUE

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ABSTRACT

Most source localization problems have been assessed so far using beamforming techniques on the data acquired with conventional microphone arrays. If the sound field can be assumed time-stationary, the number of transducers can be reduced dramatically by using data acquired with scanning techniques, such as “Scan&Paint”. This method is based on mixing tracking information with the signals recorded in order to characterize variations across a sound field. With one transducer, sound pressure or particle velocity can be assessed. Furthermore, relative phase of the sound field can be preserved by using an additional fixed reference sensor. Therefore, average magnitude and phase information of discrete spatial areas can be obtained, as if the field were measured with a conventional array. This paper introduces the underlying theory of virtual arrays and presents experimental evidence supporting its potential for assessing mid-low frequency localization problems.

1 INTRODUCTION

There are many applications which require using transducer arrays in order to localize sound sources across the space. Traditionally, this fact directly implies investing huge amounts of money in an acquisition system. Furthermore, the resolution of the measurements would depend on the number of transducers used and their positions. If the array is constituted by too many sensors, it becomes acoustically significant, biasing the sound field aimed to characterize.

A “Virtual Array” approach can be taken so as to avoid all constrains of conventional arrays if the sound field is assumed time stationary. Magnitude and phase of the sound field can be measured by using only two transducers, one situated in a fixed position and another moving across the measurable area [1]. Tracking information is acquired by processing a video recorded during the measurements. Then, time segments of a long sequence can be evaluated at different spatial areas. Moreover, the relative phase of the sound field can be acquired calculating the phase differences between fixed and moving transducers. This powerful novel technique can simplify many common problems due to its low time and cost requirements, but it also can improve the accuracy of traditional results due to its adaptable resolution and re-sizable measurement area.

The results found so far proved that the "virtual arrays" work remarkably well in laboratory conditions for mid-high frequencies [2]. However, there is no experimental evidence assessing their performance for lower frequencies. Typically, physical limitations relating the size of the array with their low frequency limit make unpractical using multichannel solutions for low frequency source localization problems. In contrast, "virtual arrays" have a great potential regarding this issue, due to the array size can be increased just by positioning the measurement plane further away from the camera which recorded the measurement.

This paper presents the new methodology required for using "virtual arrays" into real measurement scenarios, such as assessing the mid-low frequency noise produced by a gas plant. Furthermore, advantages and disadvantages of the measurement technique are discussed considering not only its theoretical and practical limitations but also the future developments required.

2 THEORY

As a first approach, the time stationary sound field produced by a monopole source is studied. This approximation will give a good understanding of how the phase can be acquired at multiple points even measuring at different time instants by using only two transducers. Then, a sum-and-delay beamforming algorithm is derived adapted to "virtual arrays".

2.1 Preserving relative phase

According to Kinsler [3], the complex pressure at a point n of a sound field produced by a monopole source at a distance r_n can be defined as

$$p_n(r,t) = \frac{A}{r_n}\, e^{j(\omega t - k r_n)} \tag{1}$$

where A is a term time and spatial independent related to the source features; and k is the wavenumber (ω/c_0). From Equation (1) can be inferred that, if A is known, the phase distribution across a sound field can be predicted at any point by defining its separation with the source.

Absolute time phase measurements implies acquiring information at discrete points of the space simultaneously. Since the number of transducer used in scanning techniques is reduced because they are moving during the measurement, absolute phase information cannot be characterized. Nevertheless, if the sound field can be assumed time stationary, relative phase variations can be measured at different time instances, allowing to use scanning techniques also for assessing phase information.

The relative phase differences between two points f and n can be obtained by calculating the product between the Fourier Transform of pressure at one point $P_n(\omega)$ and the complex conjugate Fourier Transform of the pressure at the other point $P_f^*(\omega)$, i.e.

$$P_n(\omega)P_f^*(\omega) = \frac{A^2}{r_f r_n}\, e^{jk(r_f - r_n)} \tag{2}$$

Equation (2) illustrates that the phase information now is only dependent on the wavenumber k and the distance difference between the two points $(r_f - r_n)$.

So far, arbitrary signals have been considered on the derivation but for real scenarios it would be necessary to deal with random signals of finite length [4]. One way to obtain relative phase information between two data segments of length T is by computing their cross-power spectral density (CPSD) [5]. The CPSD between two pressures at the points f and n can be defined as

$$S_{p_f p_n}(\omega) = \lim_{T\to\infty} \frac{E\left\{P_n(\omega)P_f^*(\omega)\right\}}{T} \tag{3}$$

where $E\{...\}$ denotes the expected value. As can be seen in Equation (3), phase information is again preserved similarly to Equation (2) after computing the CPSD. Therefore, the phase information will depend on the distance difference between the points and the source.

In order to apply Equation (3) to scanning techniques it is necessary to reformulate the problem regarding a fixed transducer at f and moving sensor at $\mathbf{n}$, where $\mathbf{n}$ is a matrix. Due to the position of the moving sensor varies along time, the measured sound field can be discretized into a finite number of spatial areas. Hence, beamforming techniques can be applied in the frequency domain once a general matrix with relative phase information of all measured positions $\mathbf{S}_{\mathbf{p_f p_n}}(\omega)$ is computed.

In conclusion, it has been proved analytically that relative phase changes of a sound field can be mapped by taking the cross-power spectral densities between two transducers: one at a fixed position and the other scanning an area of the sound field.

2.2 Source localization and DOA algorithms

One common application for sensor arrays is to determine the direction of arrival (DOA) of propagating wavefronts. In this section a conventional sum-and-delay beamforming is derived based on dealing with relative phase differences on the frequency domain.

Generally, an array receives spatially propagating signals and processes them to estimate their direction of arrival; it acts as a spatially discriminating filter [6]. This spatial filtering operation is known as beamforming. An conventional array processor steers a beam to a particular direction by computing a properly weighted sum of the individual sensor signals. Thus, this procedure results in the coherent addition of signals coming from the direction of focus, maximizing the energy in the beamformer output, whereas signals from other directions will be attenuated.

For short or intermediate distance away from a noise source, spherical propagation of the wavefront given in Equation (1) should be taken into account. If the sound field is produced by a simple source, waves arriving at the array can be characterized by

$$B(\omega) = \frac{1}{N}\sum_{n=1}^{N} S_{p_f p_n}(\omega) e^{-j\omega\tau_n} = \frac{1}{N}\sum_{n=1}^{N} S_{p_f p_n}(\omega) e^{-jkr_n} \tag{4}$$

where τ_n is the delay necessary to apply to the signal recorded at N positions for focusing the beam towards the source.

Assuming that the separation with the source is unknown then neither τ_n nor r_n can be calculated directly. If the array is sufficiently far from the source, the resulting wavefronts sampled by the array can be regarded as plane waves. In this case, according to Johnson and Dudgeon [7], the delays τ_n can be defined as

$$\tau_n = \frac{\vec{\zeta}\cdot\vec{x}}{c_0} \tag{5}$$

where $\vec{\zeta}$ denotes a unit vector which indicates the propagation direction; $\vec{x}$ corresponds to the measurement position; and c_0 is the speed of sound. Subsequently, evaluating Equation (4) for far field conditions leads to

$$B_{ff}(\omega) = \frac{1}{N}\sum_{n=1}^{N} S_{p_f p_n}(\omega) e^{-j\vec{\zeta}\cdot\vec{x}} \tag{6}$$

Beamforming maps can be obtained by evaluating B_{ff} for different directions of propagation $\vec{\zeta}$. Several coordinates system can be implemented [7, 8]. Polar coordinates have been found the most suitable for combining a background image and the beamformer output without using prior information of the of sound sources. Consequently, all source localization maps presented in this paper are evaluated across azimuth θ and elevation φ.

3 METHODOLOGY

The measurement procedure for acquiring the data is based on conventional “Scan & Paint” [9–11] . This novel method is a sound mapping technique based on mixing sound variations across a sound field with the relative position information of the probe extracted from a video. In the post-processing stage, the measurement plane recorded with the camera has been discretized into square regions with equal area. Additionally, a fixed reference pressure microphone has been used to preserve the relative phase information of the different grid positions according to Section 2.

4 INSTRUMENTATION AND EXPERIMENTAL SETUP

All measurements were carried out using a Microflown PU probe which contains a pressure microphone along with a particle velocity sensor. Furthermore, a Microflown microphone was used for measuring the reference pressure at a fixed position. In addition, a camera Logitech Webcam Pro 9000 was required for recording a video of the measurements.

Sweep measurement were performed along a total surface of 6 meters by 2 meters. The measurement time expended in the sweep shown were 4 minutes. A grid of 0.25 meters have been chosen for creating an array of 85 “virtual transducers”. Windcaps for reference and moving sensors were required due to the high speed wind conditions during the measurement.

A picture of the experimental setup during a test measurement can be seen on the left hand side of Figure 1. Furthermore, a satellite picture of the measurement location is presented on the right of the figure.

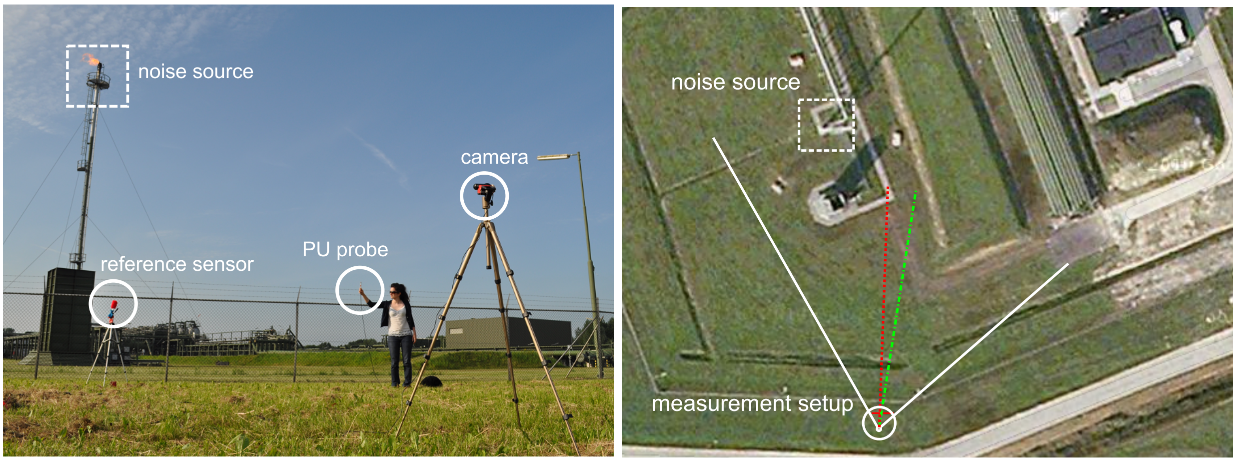

Figure 1: Experimental setup: performing a test measurement (left) and satellite picture of the measurement location (right). Green and red dotted lines indicate the camera central axis and a normal axis to the measurement plane, respectively

As can be seen in Figure 1 there was a misalignment between the center of the background picture (green dotted line) and a normal axis to the measurement virtual plane (red dotted line). Nonetheless, human errors were corrected during the post processing stage. The two axis were estimated by evaluating the pictures of the setup and the satellite images. The measurement plane was situated parallel to the surrounding fence whereas the center camera axis was estimated directly from the pictures took during the measurements.

5 RESULTS

One of the main assumptions made on the derivation given in Section 2 is regard the assessed sound field as time stationary. It allows using scanning measurement techniques for reducing the

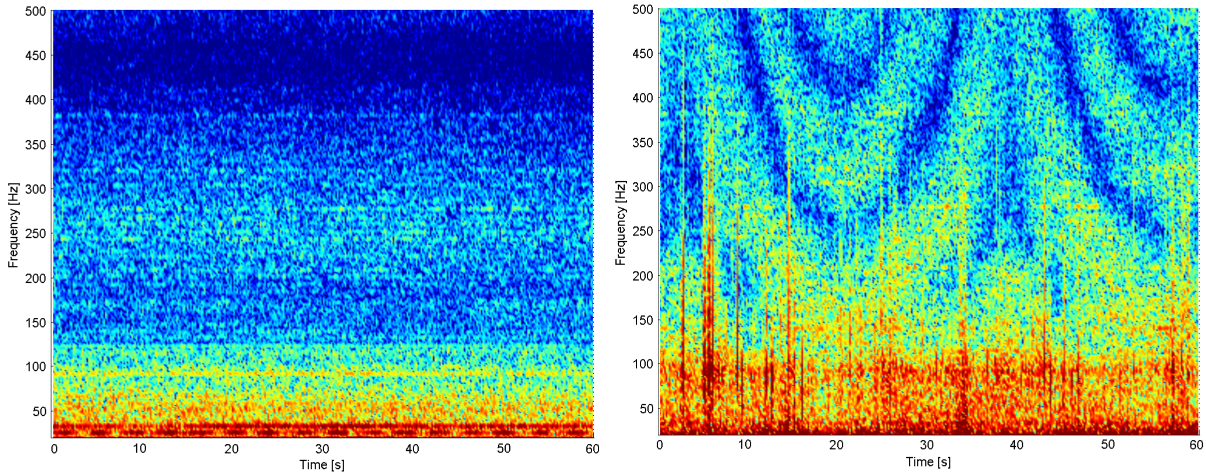

Figure 2. Spectrograms of the fixed microphone (left) and the particle velocity sensor (right)

number of transducers required to undertake the measurement. Figure 2 presents spectrograms of fixed and moving sensors during 1 minute of the measurement. On the left hand side, the degree of stationary of the sound field can be studied regarding the variations of frequency content along time. As can be seen, the spectra remains fairly constant across time above 40 Hz. Below that frequency, some noise source appears to be switching on and off for time periods of about 3 seconds. This non-stationary frequency band was not in the range of interest so it did not affect the localization maps presented later on. Besides, the spectrogram of the moving sensor is shown on the right hand side of Figure 2. In contrast with the first graph, now frequency changes across time and space are assessed simultaneously. This plot is useful for quickly detecting any manipulation noise during the measurement. Avoiding the noisy time intervals will lead to increase the accuracy of the sound localization maps.

Figure 3 shows a 360° localization map along with the error of the source localization estimations across frequency. The error was calculated assuming that the dominant noise source was located at the end of the gas pipe. The real position of the source was calculated using the pictures presented in Figure 1 along with the background image used in Figure 4.

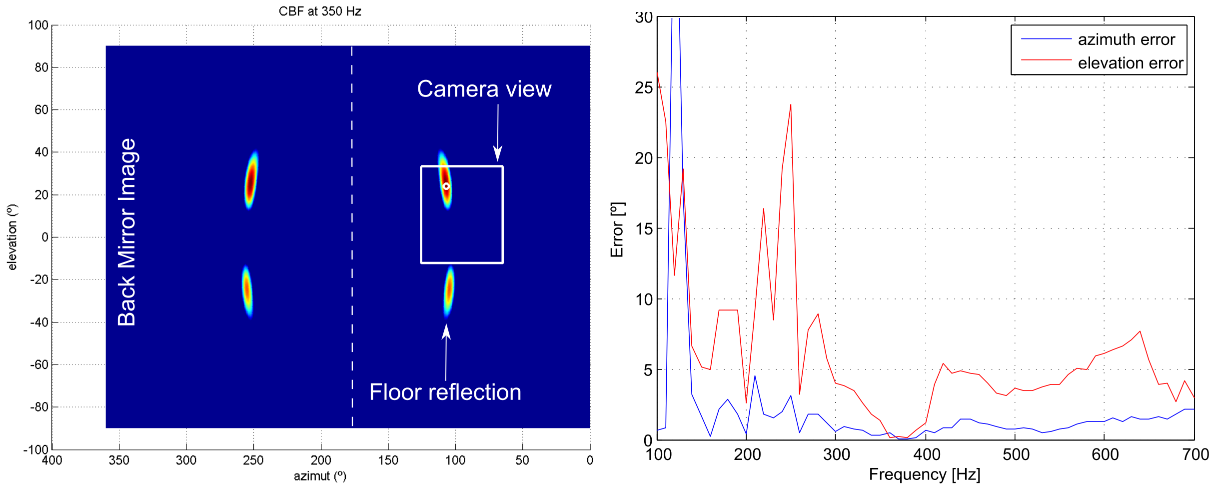

Figure 3. Error between estimated and real source location for azimuth (blue) and elevation (red)

As can be seen on the left hand side of Figure 3 the inherent back mirror effect of planar rectangular arrays appears when looking at the back of the array. Moreover, a secondary source below the ground level (0° of elevation) shows the floor reflection. On the other hand, azimuth localization is fairly stable an accurate across the whole frequency range evaluated. A slightly bigger error is obtained when assessing the elevation of the source. As have been mentioned above, the asymmetric array used has more virtual sensors along the X axis than the Y axis so it is reasonable to obtain different error curves for azimuth and elevation.

Next, Figure 4 presents source localization maps for several frequencies. A minor correction was applied (5 degree offset) in order solve the misalignment between the camera axis and the measurement plane normal axis shown in Figure 1. As can be seen, localization in the azimuth axis gives very good estimates even at 100 Hz. On the other hand, the elevation of the noise source was not as accurate mainly due to the dimensions of the rectangular virtual array used (6x2 meters). As it has been mentioned above, the number of transducers and the total effective length of the array are asymmetrical, leading to obtain better results for azimuth than for elevation .

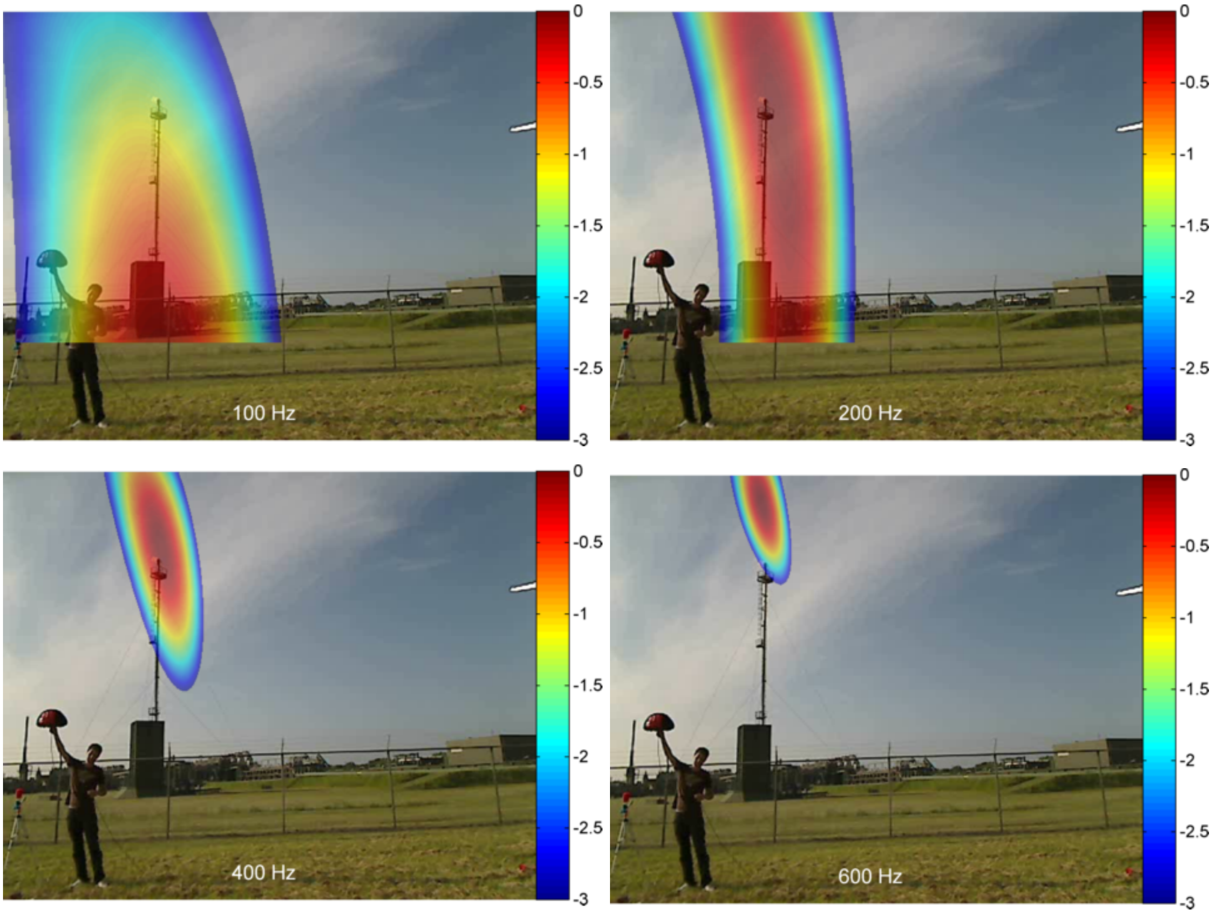

Figure 4: CBF localization maps at 100 Hz (top left), 200 Hz (top right), 400 Hz (bottom left) and 600 Hz (bottom right) [dB]

6 DISCUSSION

6.1 Multichannel solutions versus "virtual arrays"

One of the main problems of most conventional beamforming arrays is the cost of the measurement equipment. Not only the number of transducer required for performing the measurements but also the multichannel acquisition system rise the price strongly. Most of the current multichannel applications have far higher requirements against a one-probe solution.

Time required for setting up the instrumentation and performing the measurement is always a big issue. Manual sweeps of a single probe are a fast procedure for directly obtaining information about a sound field. The measurement presented in this paper was undertaken in less than 10 minutes, which can be seen as a reasonable amount of time for localizing the dominant noise sources down to 100 Hz. There are several commercial solutions which are also portable and easy to setup; however, their frequency range is very limited, specially in the low frequency region.

The measurement protocol and the post processing stage should be fairly intuitive. The use of a camera make sure that all the measurement process is filmed which proved to be helpful with trouble shooting. Color maps overlaid on pictures give a direct feedback that is easy to understand. This is a widespread feature in most array solution available on the market.

The flexibility of "virtual arrays" is one of its stronger advantages against multi-channel solutions. The novel method allows to scan from very small areas for evaluating high frequencies up to large spaces for assessing low frequency noise. On the other hand, multichannel arrays have several transducer distributed along a structure which normally is difficult to vary for changing their distribution to evaluate different frequency ranges depending on the problem.

The main outcome of a measurement technique is to ensure accurate results. The low error presented in the comparison between real and estimated source position along frequency demonstrate the great potential of combining scanning measurement techniques with beamforming algorithms. Besides, the fact that there are not physical fixed positions leads to minimize discretization errors. Choosing the transducer spacing and the array size will directly affect the frequency range assessed, whereas with "virtual arrays" the measurement grid is resizable in the post-processing stage.

6.2 Limitations

The measurement technique presented reduces the number of transducers, measurement time and cost, while maximizes the flexibility and simplicity. However, some limitations of the method have to be take into account,

- Possible misalignments between camera and measurement plane must be corrected for superposing localization maps to background pictures. This is feasible by assessing the pictures of the measurement setup or finding a better procedure to allocate the camera.
- One of the main problems of conventional scanning techniques is that time stationary conditions are required. Although some industrial applications are focus on transient or impulsive noise, many problems can be solved under stationary conditions.
- Human errors such as touching a surface or producing noise while the probe is moving are inherent to the measurement technique. Nevertheless, they can be detected and avoided during the post processing stage.
- Because the method does not measure the absolute probe position, there is only 2D information related to the background image. This fact could lead to have untraceable position errors along the Z axis.

6.3 Further development

By using scanning particle velocity sensors instead of pressure microphones should be possible to distinguish between front and back radiation. This feature should be exploded in future developments. Consequently, more complex environments with multiple sources around the array should be assessed to test the performance of the measurement technique.

So far, only simple sum-and-delay beamforming have been applied to the data. However, more localization algorithms should be implemented in order to enhance the resolution and accuracy of the results.

7 CONCLUSIONS

"Virtual Arrays" has been successfully validated as a novel broadband source localization technique for assessing environmental noise problems under stationary conditions.

The low error found between estimated and real noise source location provide clear evidence of the measurement success. It is important to highlight the good agreement even at lower frequencies, which commercial multichannel solutions are not able to assess due to size limitations of the arrays.

Assessing time stationary sound field the measurement technique introduced reduces the number of transducers, measurement time and cost of conventional microphone arrays. Moreover, the remarkable flexibility of "virtual arrays" make them a powerful tool for assessing broadband noise localization problems.

Further research will be undertaken for adapting different beamforming algorithms to the measurement technique developed. In addition, all potential of using particle velocity sensors will be investigated for creating accurate 360 degrees localization maps.

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