THE USE OF MICROPHONE LINE ARRAY FOR ACOUSTICAL SURVEILLANCE SYSTEM; THEORY

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SUMMARY

The possibility of using microphone line array for acoustical surveillance system has been conveyed through theoretical study. The range of application of the existing methods based on correlation matrix without spatial smoothing technique to identify complex noise sources such as monopole, dipole, tripole, or their combinations has been investigated. The paper also introduces simplified dipole and tripole source models to model more practical noise sources. Their characteristics in practical situations have been studied in terms of the dipole angles and the dipole moments in the dipole model, and the strength ratio between the monopole and dipole in the tripole model.

I - INTRODUCTION

When machine experiences certain malfunction due to certain defects in any part of its components, it expresses its condition by demonstrating changes in vibration, sound, contents of lubricant, or others. Therefore, machine fault diagnosis or condition monitoring of machines can be translated as a mean of finding unique finger prints; each of them must be associated with specific malfunction or condition of machine of interests. Normally these finger prints can be sufficiently characterized by transforming the time signal of interests into frequency. However, if one looks at the machine which has many components; i.e. generators, compressors, engines, and transmission systems, then one can recognize very immediately that there is not only signal with respect to time but also one with regard to space; information concerning the location where certain signal is originated. This can be envisaged as listening the fluctuating signals as well as seeing the signal's origin.

Identifying the origins of sound can be achieved either by constructing acoustic holographic picture of sound or using noise source identification methods if one handles only sound generated by machines. Noise source identification methods such as beam forming, MUSIC or ESPRIT which basically utilizes eigenstructure of microphone array signals have very strong advantage compared with holographic method which normally requires lot of measurements and complex instrumentations, because of its simplicity of instrumentation; it only requires line array of microphones.

There are rich literatures associated with sound source identification; mainly bearing angle detection in radar technology and under water acoustics [1-8]. The methodologies can be classified into the beamforming method [2] in which the phase matching between measured signals and assumed signals by an assumed plane wave source is used and the eigenstructure methods such as MUSIC(MUltiple SIgnal Classification) [3] to which the orthogonality between noise subspace and the aforementioned assumed signal is applied.

The difficulties associated with the application of these methods to sound field which is not far from sources; this is the situation of most of industrial noise case, have lead to the development of alternative methodologies. The method must consider the attenuation and the curvature with respect to the distance from the sources [9,10]. Other things are to be related with coherency between sources, and associated acoustic fields; effect of reverberant characteristics upon the source localization method. These issues have been addressed in the works of Choi and Kim [9-12], and Alvardo [13]. Recent work [14] examines the properties of correlation matrix which expresses the correlation between microphones in detail.

The possibility of using microphone array for acoustical surveillance system is obviously related with the characteristics of correlation matrix of array microphone, part of the works by authors [14] are reprinted for the purpose of acoustical surveillance applications.

II - PROPERTIES OF CORRELATION MATRIX WITH REGARD TO THE CHARACTERISTICS OF SOUND SOURCES [14]

This section examines the properties of correlation matrix which expresses not only the frequency characteristics between the sources but also the total number of incoherent sources; the rank of the matrix is the number of incoherent sources.

2.1 - Approaches in Time Domain

Let us use the complex notation for brevity to consider the effects of interference. The complex sound pressure signal $p_m(t)$ received at the m-th microphone by K noise sources, $s_1(t), s_2(t), \cdots, s_K(t)$, can be expressed as

$$p_m(t) = s_{1m}(t) + s_{2m}(t) + \cdots + s_{Km}(t) + n_m(t) \\ = \int_{\omega}\left[\sum_{k=1}^{K}(H_{km}(\omega)s_k(\omega,t))\right] d\omega + n_m(t), \quad (1)$$

where $s_{im}(t)$ is the sound pressure at the m-th microphone position due to the i-th source. $n_m(t)$ is the white and additive noise, and $H_{km}(\omega)$ is the impulse response function from the k-th source to the m-th microphone position. Here the $H_{km}(\omega)$ could be expressed as a Green function such as $e^{j\omega r_{km}/c}/r_{km}$ for a point monopole source. Then a correlation matrix is generally defined as

$$\mathbf{R} = E\left[\mathbf{P}(t)\mathbf{P}(t)^H\right], \quad (2)$$

where $\mathbf{P}(t) = \{p_1(t), p_2(t), \cdots, p_M(t)\}^T$. E stands for expectation, M is the total number of microphones. H is the Hermitian, and T is the transpose operator. To investigate the behaviours of the elements of the correlation matrix, let us choose an element of that:

$$R_{ij} = E\left[p_i(t)p_j^*(t)\right] \\ = E\left[\left(s_{1i}s_{1j}^* + s_{2i}s_{2j}^* + \cdots + s_{Ki}s_{Kj}^*\right) + \left(s_{1i}s_{2j}^* + s_{1i}s_{3j}^* + \cdots + s_{1i}s_{Kj}^*\right) + \cdots\right], \quad (3)$$

where * stands for complex conjugate and each element of the first and second parenthesis within bracket presents the relations between the noise sources at the different microphones. The effects of these can be expressed by the following formulas,

$$E\left[s_{ki}(t)\, s_{kj}^*(t)\right] = \int_{\omega} H_{ki}(\omega)H_{kj}^*(\omega)E\left[s_k(\omega,t)s_k^*(\omega,t)\right]\, d\omega \quad \text{: finite value}, \quad (4)$$

$$E\left[s_{ki}(t)\, s_{lj}^*(t)\right] = \int_{\omega} H_{ki}(\omega)H_{lj}^*(\omega)E\left[s_k(\omega,t)s_l^*(\omega,t)\right]\, d\omega \\ = \begin{cases} \int_{\omega} H_{ki}(\omega)H_{lj}^*(\omega)E\left[s_k(t)\, s_k^*(t-\tau_{kli})\right] d\omega & \text{:Case A and B} \\ 0 & \text{:Case C} \end{cases}, \quad (5)$$

where τ_{kli} is the time delay between the k-th and the l-th coherent sources with respect to the i-th microphone. *Case A* is for coherent noise sources with pure tone, *Case B* is for coherent noise sources which are white within finite band of frequency in the case of $\tau_{kli} \neq 0$, and *Case C* is for incoherent sources. Here the *Case A* and *B* can be rearranged as follow

$$\begin{cases} \int_{\omega} H_{ki}(\omega)H_{lj}^*(\omega)\; \varphi(\tau_{kli})\; E\left[s_k(t)\, s_k^*(t)\right] d\omega & \text{:Case A} \\ \varepsilon & \text{:Case B} \end{cases}, \quad (6)$$

where $\varphi_k(\tau_{kli})$ is the pressure factor with respect to the k-th source, and ε, which may be interpreted as a measure of the possibility of identification of noise sources or an interference index, goes to zero fast when the frequency band becomes wider. Therefore, the correlation matrix for *Case B* and *C* can be expressed as

$$\mathbf{R} \cong \mathbf{R}_{source\,1} + \mathbf{R}_{source\,2} + \cdots + \mathbf{R}_{source\,K}, \quad (7)$$

$$\mathbf{R}_{source\,i} = E\left[\mathbf{P}_{source\,i}(t)\mathbf{P}^{H}_{source\,i}(t)\right], \tag{8}$$

where the rank of this matrix becomes K since the total correlation matrix consists of each independent correlation matrix without interference terms. Henceforth, the conventional methods which give effective results for the identification of incoherent noise sources are suitable in these cases. The equality sign in equation (7) means the cases of incoherent noise sources (*Case C*). However in the *Case C* with pure tone the rank of the correlation matrix becomes fewer than the number of sources, and therefore source identification methods based on eigenstructure analysis such as MUSIC cannot applied to.

2.2 - Approaches in Frequency Domain

Since the characteristics of the correlation matrix in time domain consist of each frequency component in frequency domain, one can consider that correlation matrix in both domains could have the same characteristics.
The general array model for array outputs and correlation matrix in frequency domain can be expressed as

$$P_m(f,T) = \frac{1}{T}\int_0^T \left(s_{1m}(t) + s_{2m}(t) + \cdots + s_{Km}(t)\right) e^{j2\pi ft}\, dt, \tag{9}$$

$$\mathbf{R}(f) = E_T\left[\mathbf{P}(f,T)\mathbf{P}^H(f,T)\right], \tag{10}$$

$$\mathbf{P}(f) = \left\{P_1(f,T),\, P_2(f,T),\, \cdots,\, P_M(f,T)\right\}^T, \tag{11}$$

where E_T stands for the expectation based on time period T. In order to examine the properties of the correlation matrix, let us consider an element of that:

$$\begin{aligned} R_{ij}(f) &= E_T\left[P_i(f,T)P_j^*(f,T)\right] \\ &= \frac{1}{T^2}E_T\Big[\int_0^T\int_0^T \left(s_{1i}(t)s_{1j}(u) + \cdots + s_{Ki}(t)s_{Kj}(u)\right) e^{j2\pi f(t-u)}\, dt\, du \\ &\quad + \int_0^T\int_0^T \left(s_{1i}(t)s_{2j}(u) + \cdots + s_{1i}(t)s_{Kj}(u)\right) e^{j2\pi f(t-u)}\, dt\, du + \cdots \Big], \end{aligned} \tag{12}$$

Here, one can rewrite the first and second term within the bracket.

$$\begin{aligned} &E_T\left[\int_0^T\int_0^T s_{ki}(t)s_{kj}^*(u)\, e^{j2\pi f(t-u)}\, dt\, du\right] \\ &= \int_0^T\int_0^T\int_{f'}\int_{f'} H_{ki}(f')\, H_{kj}^*(f')\, E_T\left[s_k(f',t)s_k^*(f',u)\right] df'\; df'\, e^{j2\pi f(t-u)}\, dt\, du \quad : \textit{finite value}, \end{aligned} \tag{13}$$

$$\begin{aligned} &E_T\left[\int_0^T\int_0^T s_{ki}(t)\, s_{lj}^*(u)\, e^{j2\pi f(t-u)}\, dt\, du\right] \\ &= \int_0^T\int_0^T\int_{f'}\int_{f'} H_{ki}(f')\, H_{lj}^*(f')\, E_T\left[s_k(f',t)s_l^*(f',u)\right] df'\; df'\, e^{j2\pi f(t-u)}\, dt\, du \\ &= \begin{cases} \int_0^T\int_0^T\int_{f'}\int_{f'} H_{ki}(f')\, H_{lj}^*(f')\, E_T\left[s_k(f',t)s_k^*(f',u-\tau_{kl})\right] df'\; df'\, e^{j2\pi f(t-u)}\, dt\, du & : \textit{Case A and B} \\ 0 & : \textit{Case C} \end{cases} \end{aligned} \tag{14}$$

where k is not equal to l. Each case is the same as those in equation (5) and the *Case A* and *B* in equation (14) can be rearranged as

$$\begin{cases} \int_0^T\int_0^T\int_{f'}\int_{f'} H_{ki}(f')\, H_{lj}^*(f')\, \varphi(\tau_{kl})E_T\left[s_k(f',t)s_k^*(f',t)\right] df'\; df'\, e^{j2\pi f(t-u)}\, dt\, du & : \textit{Case A} \\ \varepsilon & : \textit{Case B} \end{cases} \tag{15}$$

These results are the same as those in time domain approaches, and therefore the correlation matrix for *Case B* and *C* can be obtained as

$$\mathbf{R}(f) \cong \mathbf{R}_{source\,1}(f) + \mathbf{R}_{source\,2}(f) + \cdots + \mathbf{R}_{source\,K}(f). \quad (16)$$

$$\mathbf{R}_{source\,i}(f) = E_T\left[\mathbf{P}_{source\,i}(f,T)\,\mathbf{P}^H_{source\,i}(f,T)\right]. \quad (17)$$

These relations present the applicability of the conventional methods to the identification of noise sources of which frequency characteristics are white within a finite band of frequency, while the identification of noise sources with a finite band of frequency can be accomplished by the successive applying the conventional methods at each frequency. However this approach requires the same procedures each time. To alleviate this rather cumbersome process, one can use the CSM(Coherent Signal subspace Method) [6-8, 11, 12] that is a useful method to modify correlation matrix at any frequency component into that at reference frequency.
Now let us consider the frequency domain approach for practical purpose. If one can get microphone array outputs by frequency domain averaging, the outputs of the array can be expressed by a column vector at each frequency. Then the correlation matrix can be expressed as multiplication of this vector and its Hermitian vector. Henceforth the rank of the correlation matrix becomes one and this property make many estimators which are very suitable for the identification of uncorrelated sources unsuitable.
The CSM introduced by Choi and Kim [11, 12] for the spherical source model can be a reasonable tool to give the same number of rank as the number of sources. Their validity can be proved as the following.

Type I
An equivalent correlation matrix can be obtained by the superposition of the correlation matrix at each frequency of which elements are modified by the CSM. That is,

$$\widetilde{\mathbf{R}} = \frac{1}{N}\sum_{n=1}^{N}\widetilde{\mathbf{R}}(f_n). \quad (18)$$

$$\widetilde{\mathbf{R}}(f_n) = \left[\widetilde{\mathbf{P}}(f_n)\,\widetilde{\mathbf{P}}^H(f_n)\right]. \quad (19)$$

where the tilde sign stands for the modification using CSM-FC(CSM by using Frequency Correction) or CSM-FM(CSM by using Focusing Matrix) [11, 12] for planar wave sources and CSM-FCS(CSM-FC for Spherical wave sources) or CSM-FMS(CSM-FM for Spherical wave sources) [11, 12] for spherical wave sources. One can easily find that the rank of the correlation matrix has the same as those of sources. This proves the validity of the source identification methods based on the CSM [11,12].

Type II
Another type of array outputs can also be taken, that is,

$$\widetilde{\mathbf{P}} = \frac{1}{\sqrt{N}}\begin{bmatrix} \widetilde{P}_1(f_1) & \widetilde{P}_1(f_2) & \cdots & \widetilde{P}_1(f_N) \\ \widetilde{P}_2(f_1) & \widetilde{P}_2(f_2) & \cdots & \widetilde{P}_2(f_N) \\ \vdots & \vdots & \ddots & \vdots \\ \widetilde{P}_M(f_1) & \widetilde{P}_M(f_2) & \cdots & \widetilde{P}_M(f_N) \end{bmatrix}. \quad (20)$$

where tilde sign means the complex sound pressure signal modified by CSM. The correlation matrix by this becomes the same as the results of Type I by the property of matrix and vector operations.
Based on these observations, one could find that the rank of the correlation matrix has the same as those of sources when the differences of the source strengths between frequency components are recognised by mathematical expressions.

2.3 - IDENTIFICATION OF PURE TONE SOURCES

The previous sections describe the applicability of the source identification based on the correlation matrix for the noise source identification with finite band of frequency. Now, we are going to try to identify the noise sources with pure tone by classifying into the following three categories.

2.3.1 - Perfectly coherent sources

The interference of coherent sources affects on the array outputs and therefore the rank of the correlation matrix becomes fewer than the number of coherent sources. Henceforth the methods based on the eigenstructure analysis which requires the same number of rank as the number of sources, are not suitable. Although this condition is not necessary for the methods based on phase matching techniques such as beamforming, the peaks in beamforming power are severely distorted from the locations of true sources. Moreover, a way to identify perfectly coherent noise sources based on the spatial smoothing techniques [1, 3, 4] for the planar wave model does not work out due to the sphericity of wavefronts. Henceforth the researches on this problem will be followed in the future.

2.3.2 - Coherent sources with the different source strengths between the sources at each OPerating Condition(OPC)

It is noteworthy that the rank of the correlation matrix is equal to that of the sources of which the characteristics in the frequency domain are white within the finite band of frequency. In other words, based on the results of the section 2.2, these could be interpreted by the superposition of the array outputs which have different source strengths between sources at different frequency, keeping the same locations of noise sources. Under these observations one could intuitively consider the case that the noise sources have the pure tone, but the relative source strengths between the noise sources change according to the each OPC, keeping the same locations of noise sources. That is, let us take the correlation matrix, which consists of the superposition of array outputs at each OPC. The construction procedures of the correlation matrix can be obtained by the same procedures as those of Types I and II in section 2.2 except that the array outputs at the each spectral line change into those at the each OPC. The following are those procedures at frequency f_n.

Type III: Modification of Type I

$$\mathbf{R}(f_n)=\frac{1}{Q}\sum_{q=1}^{Q}\mathbf{R}(f_n,T_q), \tag{21}$$

$$\mathbf{R}(f_n,T_q)=\left[\begin{bmatrix}P_1(f_n,T_q)\\P_2(f_n,T_q)\\\vdots\\P_M(f_n,T_q)\end{bmatrix}\left\{P_1(f_n,T_q)\quad P_2(f_n,T_q)\quad\cdots\quad P_M(f_n,T_q)\right\}^*\right]. \tag{22}$$

Type IV: Modification of Type II

$$\mathbf{R}(f_n)=\frac{1}{Q}\mathbf{P}(f_n)\,\mathbf{P}(f_n)^H, \tag{23}$$

$$\mathbf{P}(f_n)=\begin{bmatrix}P_1(f_n,T_1) & P_1(f_n,T_2) & \cdots & P_1(f_n,T_Q)\\P_2(f_n,T_1) & P_2(f_n,T_2) & \cdots & P_2(f_n,T_Q)\\\vdots & \vdots & \ddots & \vdots\\P_M(f_n,T_1) & P_M(f_n,T_2) & \cdots & P_M(f_n,T_Q)\end{bmatrix}, \tag{24}$$

where T_q means the time duration for the q-th OPC, and Q denotes the total number of OPCs. The validates of these procedures can be proved by the same procedures as those in section 2.2 by changing the frequency components into OPC components and the rank of the correlation matrix K, which is the same result as those of section 2.2. Note that the condition in this section could be also realised in the case of that the small strength differences between the sources at the different OPCs can be recognisable in mathematical expressions. Therefore the noise source identification could be successively applied to the coherent scene where the relative source strengths are changing in a very small or very large amount with time, keeping the same source locations.

2.3.3 - Coherent sources with independent white noise

This condition can easily be handled by the reinterpretation of the conditions of the above section; the variation of the relative source strengths at each OPC in the above section can be described as the effects of the independent white

noises, which are located at each source implicitly.
Let us assume that the $s_1(t), s_2(t), \cdots, s_K(t)$ are the signals of noise sources, and $n_1(t), n_2(t), \cdots, n_K(t)$ are the signals of independent white noise at each source. Then the sound pressure signal at the m-th microphone can be expressed like this,

$$p_m(t) = \int_{\omega} \left[\sum_{k=1}^{K} (H_{km}(\omega) s_k(\omega, t)) \right] d\omega + \int_{\omega} \left[\sum_{k=1}^{K} (H_{km}(\omega) n_m(\omega, t)) \right] d\omega . \quad (25)$$

Here $H_{km}(\omega)$ is the complex propagation factor from the k-th source to the m-th microphone. The first part of the left hand side of equation (25) means that the sound pressures from the true sources and the second part stands for the sound pressures due to the independent white noise sources, which are included by the true sources.
Finally noise source identification in this category could be redefined as the identification of independent white noise sources contaminated by coherent noise sources. By this redefinition one could easily apply the conventional methods to the identification of incoherent noise sources since the conventional ones have been effectively used to identify incoherent noise sources.

III - IDENTIFICATION OF NOISE SOURCES ACCORDING TO THE PROPAGATION CHARACTERISTICS OF SOURCES [14]

In this section we try to briefly review planar and spherical source model and methods identifying noise sources at given frequency. The dipole and tripole source models are developed in this section to give a possibility to localize more complex sources.

3.1 - PLANAR WAVE MODEL

Conventional methods for the identification of noise sources have used this model, which is expressed by the source strengths and bearing angles, for example,

$$s_{planar} = e^{j\omega x\cos\theta/c}, \quad (26)$$

where θ is the k-th bearing angle, and x is the distance of the line array from the reference position.
Conventional beamforming [2, 4-5] and eigenstructure analysis such as MUSIC [3, 4, 5] have been used for the identification of uncorrelated noise sources at given frequency. The methods based on the CSM-FC(Coherent Signal subspace Method by using Frequency Correction) or CSM-FM(Coherent Signal subspace Method by using Focusing Matrix) [11, 12] can be used in the case of the identification of noise sources with a finite band of frequency.

3.2 - SPHERICAL WAVE MODEL(MONOPOLE MODEL)

Since the planar wave model is not valid for the identification of noise sources located in the near range, the source model with spherical wave fronts, which has the sphericity of the wavefronts, was suggested and proved its effectiveness for the source identification in near field [4, 8, 10]. The typical spherical source model can be expressed as

$$s_{monopole} = \frac{1}{r} e^{j\omega r/c}, \quad (27)$$

where r stands for the distance between the source and the microphone position.
Spherical beamforming and spherical MUSIC [9-12] are suitable for the identification of uncorrelated noise sources at given frequency. The CSM-FCS(CSM-FC in Spherical wave sources) and CSM-FMS(CSM-FM in Spherical wave sources) [11, 12], which were obtained by the extension of the CSM-FC and CSM-FM respectively, were proved for their effectiveness by the identification of spherical noise sources with a finite band of frequency.

3.3 - DIPOLE MODEL

The acoustic dipole, which is generated by a fluctuating force such as an oscillating wall, can be obtained as the superposition of two monopoles with 180^0 out of phase. Figure 1(a) shows a simple dipole model based on this interpretation and the sound pressure signals in (r, θ) co-ordinate can be expressed as

$$p(r,\theta,t) \cong \frac{j2B}{r}\sin\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)e^{j(\omega r/c-\omega t)} \tag{28}$$

where B is the amplitude of each monopole to consider and the distance h between the monopole source is small enough to satisfy $r >> h$.

From this equation one can realize that the dipole model has the two variables of r and ϑ in the sine argument while the conventional source identification methods have been used as a simplified model with one variable: i.e., bearing angle θ in planar wave model, and the distance r between the source and the microphone position in the spherical wave model.

The scan vector, which consists of true source model, can be obtained by the equation (28) in terms of $\omega h/c$, r, ϑ except the amplitude factor B. But the scanning procedures based on the three unknown variables require considerable time. So this paper introduces the power series expansion of the sine function to reduce the unknown variables. Considering these observations, equation (28) can be rewritten as

$$p(r,\theta) \cong \frac{j2B}{r}e^{j\omega r/c}\left[\frac{\omega h/c}{2}\sin(\vartheta-\theta)-\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)^3+\cdots\right]. \tag{29}$$

Then, the scan vector of the dipole model can be found in the following way.

$$s_a(r,\theta)=\frac{1}{r}e^{j\omega r/c}\left[\tilde{q}_1\sin\theta-\tilde{q}_2\cos\theta+\tilde{q}_3\sin 3\theta-\tilde{q}_4\cos 3\theta+\cdots\right]. \tag{30}$$

$$\tilde{q}_1=\cos\vartheta\left(\left(\frac{\omega h/c}{2}\right)-\frac{3}{4}\left(\frac{\omega h/c}{2}\right)^3+\frac{5}{8}\left(\frac{\omega h/c}{2}\right)^5-\cdots\right), \tag{30a}$$

$$\tilde{q}_2=\sin\vartheta\left(\left(\frac{\omega h/c}{2}\right)-\frac{3}{4}\left(\frac{\omega h/c}{2}\right)^3+\frac{5}{8}\left(\frac{\omega h/c}{2}\right)^5-\cdots\right), \tag{30b}$$

$$\tilde{q}_3=\cos\vartheta\left(\frac{1}{4}\left(\frac{\omega h/c}{2}\right)^3-\frac{5}{16}\left(\frac{\omega h/c}{2}\right)^5+\cdots\right), \tag{30c}$$

....

where $\tilde{q}_i$ is an unknown factor which consists of the dipole angle ϑ and dipole moment $\omega h/c$. The number of expansion terms of sine function makes the two times of the unknown variables.

By considering the orthogonality between signal subspace and noise subspace and using $\frac{\omega h/c}{2} << 1$, we can obtain the scan vector as the following way.

$$s_b(r,\theta)=\frac{1}{r}e^{j\omega r/c}\left(\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)-\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)^3+\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)^5-\cdots\right). \tag{31}$$

where the number of expansion terms can be selected. With these scan vector s_a or s_b, the existing method such as beamforming and MUSIC including the signal processing technique described in section 2.2 can be applied to. In sum, the equations (30) or (31) can be used as dipole scan vector approximately.

3.4 - TRIPOLE MODEL

A tripole model could be understood as a superposition of a monopole and a dipole source models (Figure 1(b)), which can be expressed as

$$p(r,\theta) \cong \left[\frac{A}{r}+\frac{j2B}{r}\sin\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)\right]e^{j\omega r/c}, \tag{32}$$

where $h/\lambda << 1$. So, by considering monopole and dipole scan vectors, a simple scan vector can easily be expressed as

$$s_c(r,\theta) \equiv \frac{1}{r} e^{j\omega r/c}\left[1 + (q_1 \sin\theta - q_2 \cos\theta + q_3 \sin 3\theta - q_4 \cos 3\theta + \cdots)\right],$$
$$q_i = -j2B\bar{q}_i / A. \tag{33}$$

Using the value of coefficients q_i, we can make another scan vector s_d as follows (34)

$$s_d(r,\theta) = \frac{e^{j\omega r/c}}{r}\left[1 + j\frac{2B}{A}\left(\left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right) - \left(\frac{\omega h/c}{2}\sin(\vartheta-\theta)\right)^3 + \cdots\right)\right] \tag{35}$$

IV - FIELD EFFECTS

This can be classified by free, reverberant, and diffuse fields. For the free field, all the noise source identification methods have been developed under this assumption. Therefore, there are no problems when one wants to apply noise source identification methods to this sound field. Applicability to the reverberant field could be justified by the idealisation of that field, that is, an enclosure which consists of many reflectors surrounding some volume might be modelled by a number of sources which can replace those reflectors. However many image sources make an ill-posed problem; the total numbers of microphones are less than those of sources including image sources. Although this is a very serious problem, the effects may be reduced by the increase of the distance from the enclosure. Moreover the enclosure could be replaced by the image sources at distant, and the effects of image sources are negligibly small compared to those of direct sources. These make well-posed problem. Finally noise source identification in a reverberant field might be redefined: identification of multiple coherent sources. If an enclosure can be regarded as a combination of standing waves, one may find the standing wave pattern in a room by using bearing angle estimation techniques based on a planar wave model.

The aforementioned methods are not valid any more for the diffuse field in which the difference of sound pressure over array microphones is not be recognised.

V - EXPERIMENT

For the incoherent/coherent source identification, ideal experiment was performed in anechoic room. Figure 2(a) shows the experimental set-up and the source and array position. Three speakers were used for noise sources. All three speakers were excited with coherent signal of 492Hz pure tone and the incoherent signal of 512Hz pure tone was added to the third speaker. Figures 2(b) and 2(c) show the MUSIC power spectrum. This result tells the possibility of source localization with incoherent/coherent sources and the applicability to acoustical surveillance system using microphone line array.

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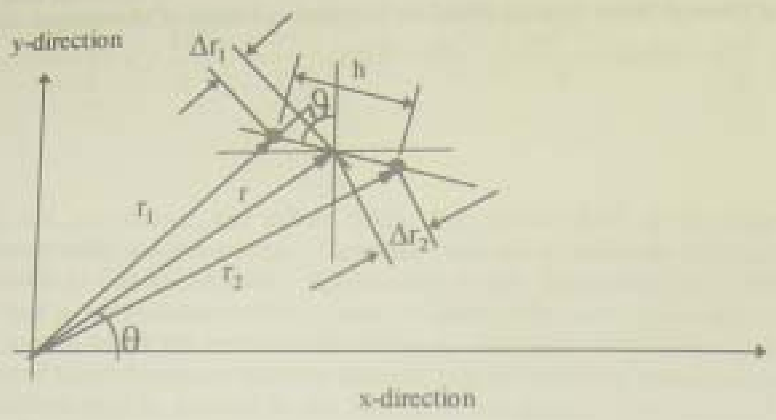

Figure 1(a). Dipole source model consisting of two monopole sources.

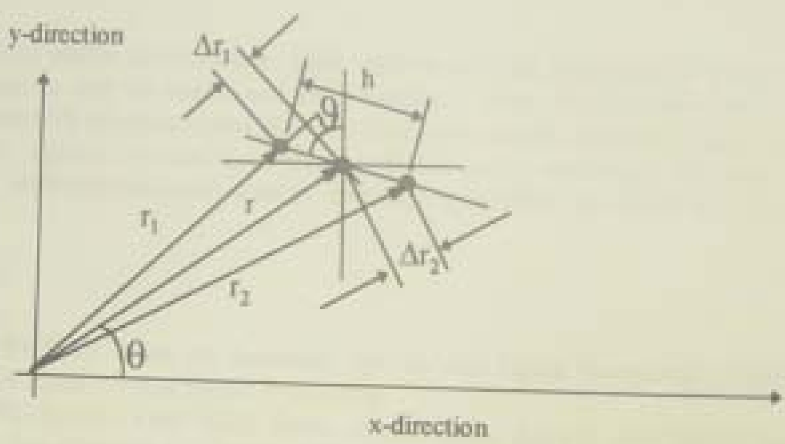

Figure 1(b). Tripole source model consisting of three monopole sources.

Figure 1. Typical models for dipole and tripole source.

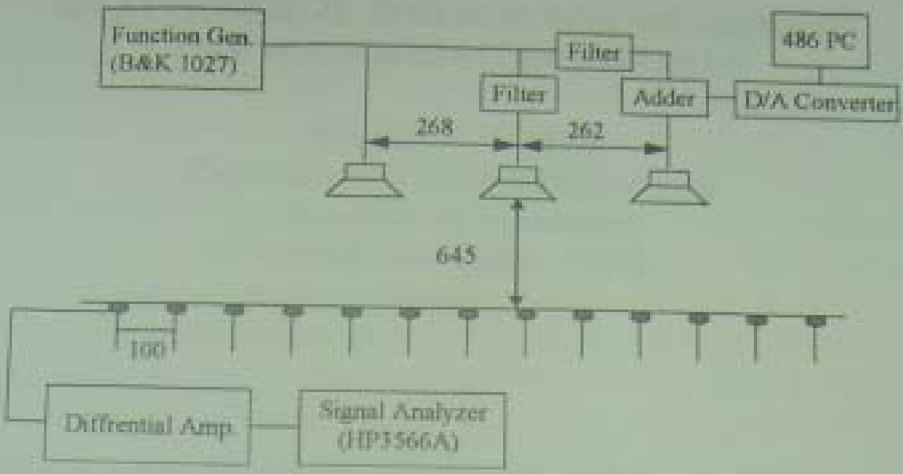

Figure 2(a). Experimental set-up - source and array position (unit : mm).

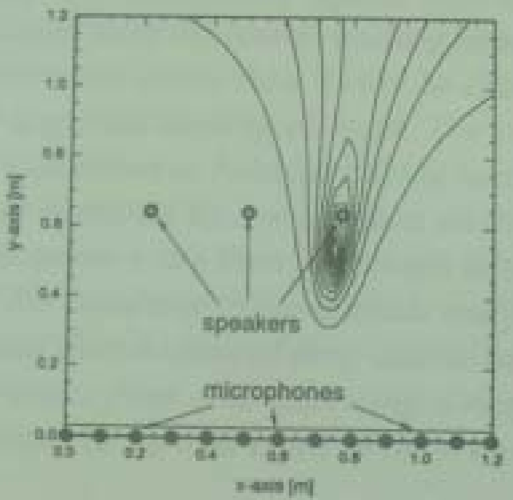

Figure 2(b). MUSIC power specturm for incoherent case (512Hz).

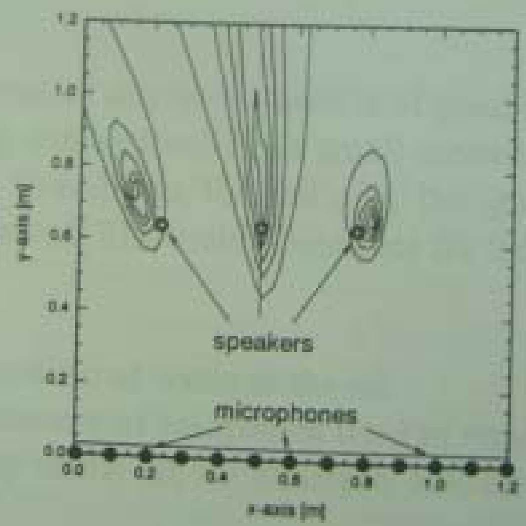

Figure 2(c). MUSIC power spectrum for coherent case (492Hz).

Figure 2. Typical experiment and result for source localization.