ACTIVELY CREATED QUIET ZONES USING DIRECTIONAL ARRAY OF LOUDSPEAKERS

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Previous research have done on the quiet zones created by multiple control sources and error microphones. But the size of the quiet zones are restricted by calculating ability of DSP as the number of control source increase greatly when a big quiet zone is needed in practice. In this paper a new control strategy is presented using a directional array of loudspeakers as a control unit to create a specific quiet zone. When different units work together, a big quiet zone can be created. Using directional array can greatly decrease the unwanted interference between units, and when the arrangement is optimized, the largest area of quiet zone can be created with the least increase of total power output.

Keywords: Active noise control, Directional array, Local control

1. Introduction

Using active control technique to alter the acoustic field to a desired one is the main purpose of local control. Previous work on local active control in free space have shown that the performance of the control system depends on the size of the desired quite zone, the number of the secondary loudspeakers and the physical structure of the system. When a large quiet zone is needed, numbers of secondary loudspeakers will be used which means it's difficult to realize since the calculating ability of the DSP is limited. And when the quiet zone is achieved, there always comes a problem that sound pressure level in areas outside the quiet zone will increase.

Jiannan Guo and Jie Pan have thoroughly examined the global active noise control system using linear loudspeaker array [1] and surface loudspeaker array [2].It is found that the performance of the control system is greatly dependent on the distance between the primary sources, secondary sources and the selected quiet areas. The relationship between the effective range of the broadband noise and the physical structure is also discussed. When the physical structure of the system is optimized, a big quiet zone is achieved in both cases. But this comes with the price of large amplification in the sound field, which is not wanted.

Spherical loudspeakers are also used as multiple secondary sources for local active control [3].Formulation of optimal source velocity has been developed using spherical harmonics representation, showing that the natural cancellation region is a shell around the spherical source. Boaz Rafaely have done analytical simulations and shown a potential for significantly larger quiet zone compared to a monopole source. Tomer Peleg and Boaz Rafaely have done a further investigation and shown that the use of two spherical sources achieved a relatively large extent for the quiet zones, in particular when designed around a rigid sphere [4].But this also comes with the condition number and amplification levels increasing in the areas outside the quiet area.

In order to achieve a quiet zone while having little amplification level augment in the areas outside the quiet area, steerable parametric array loudspeakers is used as secondary loudspeakers. The parametric array sound is first theoretically addressed by Westervelt [5], the array is dubbed as an audio spotlight to generate two ultrasounds [6] which will self-demodulate to form a main lobe and two extraneous side lobes. In 2005, Tan and Tanaka studied the applicability of PAL as a control sound source for ANC [7].And in 2008, Komatsuzaki studied experimentally the possibility of suppressing the sound pressure [8].But the quiet area achieved is very narrow due to the main lobe’s tiny breadth.

Previous papers showed that the problem of increase in the areas outside the desired quiet area is difficult to solve and large quiet area requires great amount of computation. The aim of this paper is to present a local control system using the constrained least square method and distributed control strategy to achieve a big quiet area while having little increase in the areas outside the quiet area while greatly release the calculation burden of the DSP. The constrained least square method is used in 3D sound reproduction by Yankai Zhang to reproduce a planar sound wave in a specific area, but similar methods are not used in active noise control. In this paper, numerical calculations are done showing that a large quiet zone can be achieved with little increase in the areas outside the quiet area using the previous method. And when the physical structure is optimized, large quiet zones in the far field rather than the error microphone place can be achieved, which is a very interesting phenomena and experiments are expected.

2. General Theory and System Description

The system is consist of K units, each unit consists N control sources, M error microphones and L constrained points. Control sources and primary sources are spaced equally in two parallel lines, and the constrained points are spaced in two semi-circle in the microphone plane in order to decrease the unwanted spillover in the areas outside the interested area, which is shown in Fig.1.

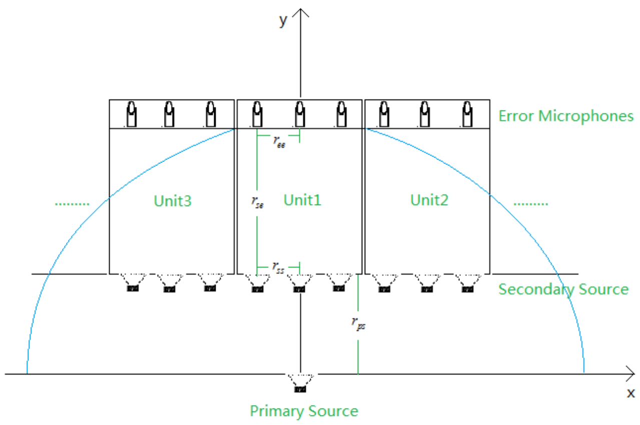

Fig.1: A sketch map of the control system, the blue circle is the constrained points of unit 1, the constrained points of other units are omitted.

2.1 Only one unit work

When only the corresponding source strengths are, respectively,

$$q_p = [q_{p1}, q_{p2}, ..., q_{pM}]^T , \quad (1)$$

$$q_s = [q_{s1}, q_{s2}, ..., q_{sM}]^T , \quad (2)$$

The sound pressure at the observation point in the system is,

$$P = P_p + P_s = H_p q_p + H_s q_s , \quad (3)$$

where P_p denotes the sound pressure caused by the primary source and P_s denotes the sound pressure caused by the second sources. H_p and H_s are acoustic transfer impedance vectors from the primary and secondary source to the observation point, respectively. The sound pressure at the error microphone point is,

$$P_e = H_{pe} q_p + H_{se} q_s . \quad (4)$$

If the constrained point is not considered, it's a simple local control problem since only one unit is working. The sum of the squared sound pressures at microphone positions is selected as the cost function,

$$J = P_e^H P_e = (H_{pe} q_p + H_{se} q_s)^H (H_{pe} q_p + H_{se} q_s) , \quad (5)$$

where H_{pe} is an $L \times M$ matrix of acoustic transfer impedance from the primary source to the error microphones, H_{se} is an $L \times N$ matrix of acoustic transfer impedance from the secondary sources to the error sensors. When the regularization parameter is considered, the solve of the equation is,

$$q_s = -(H_{se}^H H_{se} + \lambda I) H_{se}^H H_{pe} , \quad (6)$$

where, λ denotes the regularization parameter. The shape of the quite zones of this system is a wide-mouth shape, which will be shown in Fig.2 (a).

If the constrained point is considered, the equation and cost function will be changed. The equation will be changed to,

$$\begin{bmatrix} H_{pe} \\ H_{pc} \end{bmatrix} q_p + \begin{bmatrix} H_{se} \\ H_{sc} \end{bmatrix} q_s = \begin{bmatrix} P_e \\ P_c \end{bmatrix} , \quad (7)$$

where, H_{pe} is an $K \times M$ matrix of acoustic transfer impedance from the primary source to the constrained points, H_{se} is an $K \times N$ matrix of acoustic transfer impedance from the secondary source to the constrained points, P_c denotes the sound pressure at the constrained point which will be zero at all constrained points. The cost function will be changed to,

$$J = \left[pe\ pc \right]^H \begin{bmatrix} pe \\ pc \end{bmatrix} . \quad (8)$$

When the regularization parameter is considered, the solve of the equation will be,

$$q_s = -(\left[Hse\, Hsc \right]^H \begin{bmatrix} Hse \\ Hsc \end{bmatrix} + \lambda I) \cdot \left[Hse\, Hsc \right]^H \cdot \begin{bmatrix} Hpe \\ Hpc \end{bmatrix} . \quad (9)$$

Because of the constrained points, the shape of the quiet zones will be changed, there will be little increase in the area outside the quiet zone and the sound energy of second sources will be focus in the direction will the error microphone is placed, which is shown in Fig.2 (b).

2.2 All the units work together

When all the units work together, each of the units is aimed to control the sound pressure in the corresponding error microphones. Because each unit is designed only to control a part of the whole error microphones, there will be crosstalk between adjacent units, and owing to the constrained points, the interferences won't be too critical. The sound pressure at the observation point will be,

$$P = H_p q_p + H_{s1} q_{s1} + H_{s2} q_{s2} + ... + H_{sn} q_{sn} ,$$

where, $H_{s1}...H_{sn}$ denotes the matrix of acoustic transfer impedance from the secondary sources to the constrained points, $q_{s1}...q_{sn}$ denotes the source strengths of secondary sources in each units. The quiet zones created by each units will overlap in the far field and complex quiet zones will be created, which is shown in Fig.2 (c).

When the structure of the system is optimized, a big quiet zone will be created in the far field. And when the distance between the error microphone and the secondary sources R_{se} is increased, the quiet zone will move towards far field.

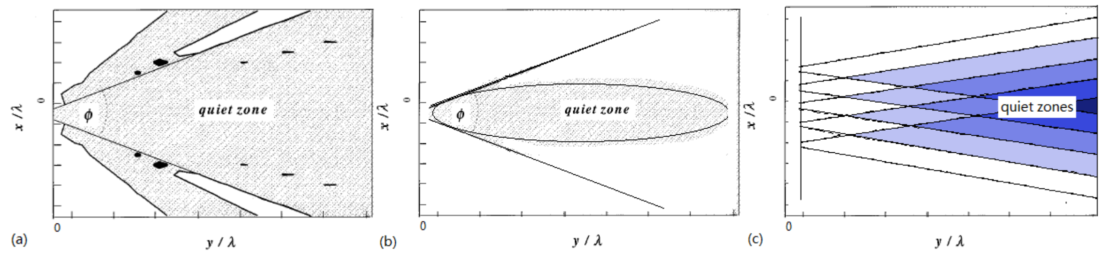

Fig.2: (a) Shape of quiet zone when one unit works without constrained points, (b) shape of quiet zone when one unit works with constrained points, (c) shape of quiet zones when all the units work, the dark colour means the area is influenced by a lot of units and the shape of the quiet zones will be complex.

3. Numerical simulations

3.1 The set up used in simulations

Simulation is divided into two parts, the first part compares the quiet zones generated by one unit, three units and five units with and without the constrained points. The second part discuss the influence of R_{se} to the quiet zone when five units work together.

Firstly the system was simulated in the frequency domain. The primary source is a pure tone noise of 500Hz placed 1000m away from the centre of the secondary sources array, which can be considered as plane wave in the near field of the concerned area. The unit is consist of 19 secondary sources, 270 error microphones and 80 constrained points. The distance between the secondary sources r_{ss} is 0.15m and the distance between the error sensors r_{ee} is 0.01m, and the distance between the secondary source array and the error microphones is R_{se}, which is altered as an important parameter in the simulation. The constrained points is placed in two quarter circle in both sides of the units with a radius of the distance between the edge of the error microphone array and the centre of the secondary microphone array. All the units are placed side by side in a line, the distance between the units is r_{ss}, which means each unit is responsible for an error microphone area of 2.7m. Since the number of error microphone and constrained points are much more than the number of secondary sources, the system is an underdetermined system, so the regularization parameter is needed and the singularity of the system can be represented by the condition number.

3.2 The quiet zones with and without constrained points

When the error microphones is placed in the near field of the space, the quiet zones generated with and without constrained points is compared in the simulation, where systems consist of one unit, three units and five units are simulated separately. All the simulations in this part is done when the distance between the error microphones and the secondary sources R_{se} focused at 4m. The results are shown in Fig.3.

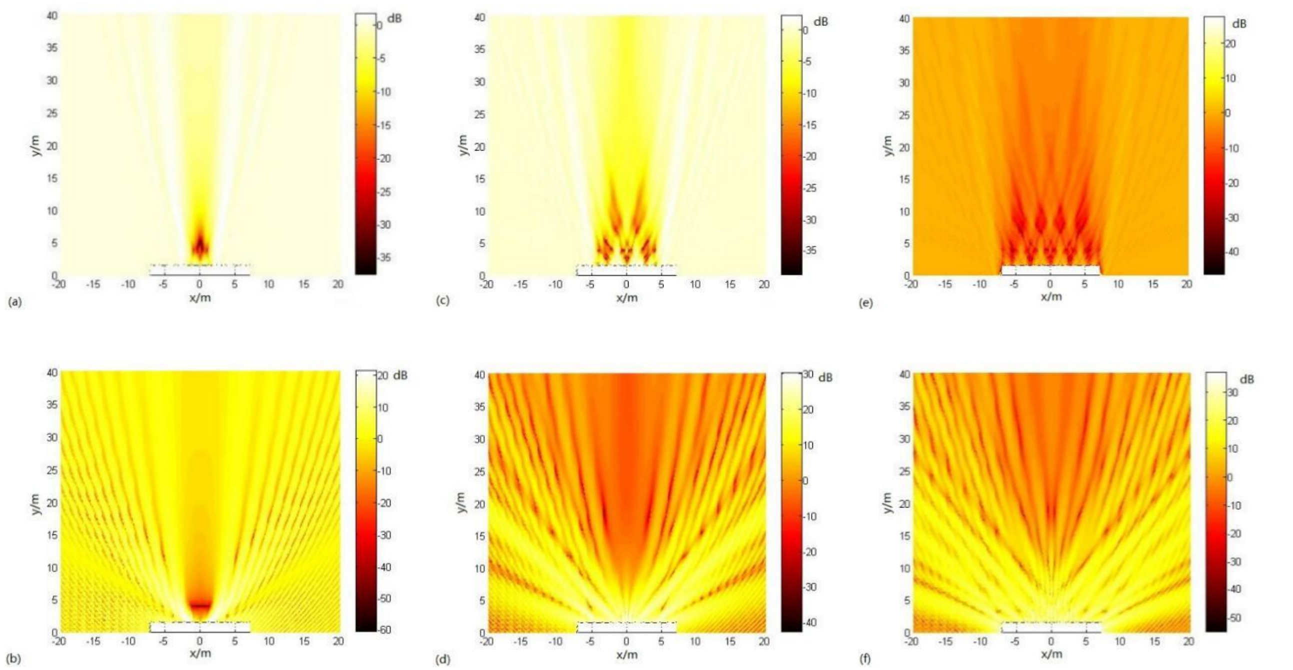

Fig.3: The shape of the quiet zone when (a) one unit works with 80 constrained points, (b) one unit works without constrained points, (c) three units work with 240 constrained points, (d) three unit work without constrained points, (e) five units work with 400 constrained points, (f) five units work without constrained points, the blank in each figure is the secondary source area where the sound pressure is omitted in order to enhance the contrast ratio of the whole figure.

Fig.3 (a) shows the shape of the quiet zone when one unit works with 80 constrained points. A quiet zone with more than 10dB reduction is created near the error microphone area, and reduction area can also be found in the far field which is decreasing with the distance. And in other areas the increase of sound pressure is not significant, this is owing to the effect of constrained points which can focus the sound power in the concerned direction. Fig.3 (b) shows the shape of quiet zone when one unit works without constrained points. A larger quiet zone with more than 10dB reduction is created near the error microphone area and a similar reduction area can be found in the far field, but in other areas sound pressure increase periodically.

Fig.3 (c) shows the shape of the quiet zone when three unit works with 240 constrained points. Three quiet zones with more than 10dB reduction is created near the error microphone area, and two similar quiet zones in a farer place which is owing to the influence between three units. Reduction areas can also be found in the far field, and in other areas the increase of sound pressure is not significant. Fig.3 (d) shows the shape of the quiet zone when three unit works with no constrained points. Sound pressure in the area near the error microphones increase significantly, and reduction areas and increase areas appear periodically in the far field. This is because of the influence between the units will be more significant when there are no constrained points.

Fig.3 (e) shows the shape of the quiet zone when three unit works with 400 constrained points. Two lines of quiet zones with more than 10dB reduction is created near the error microphone area. Reduction areas can also be found in the far field, and in other areas the increase of sound pressure is not significant. The shape of each quiet area near the error microphone area is like the shape of a diamond, which is owing to the overlapping of the quiet zones created by five units separately, which can also be shown in Fig.2 (c). Fig.3 (f) shows the shape of the quiet zone when three unit works with no constrained points. Similar to the situation when three units work without constrained points, sound pressure is increased near the error microphone area and reduction areas and increase areas appear periodically in the far field. And the increase in the near field is more significant than the situation of three units, this is because the influence between the units will be more critical when the number of the units increase.

3.3. The quiet zones when error microphone placed in the far field

This part analysis the quiet zones created when error microphones are placed in the far field. The system used in this simulation is consist of five units and 400 constrained points, the distance between the error microphone and secondary sources R_{se} is 10m, 30m, 50m and 70m. The quiet zones is shown in Fig.4.

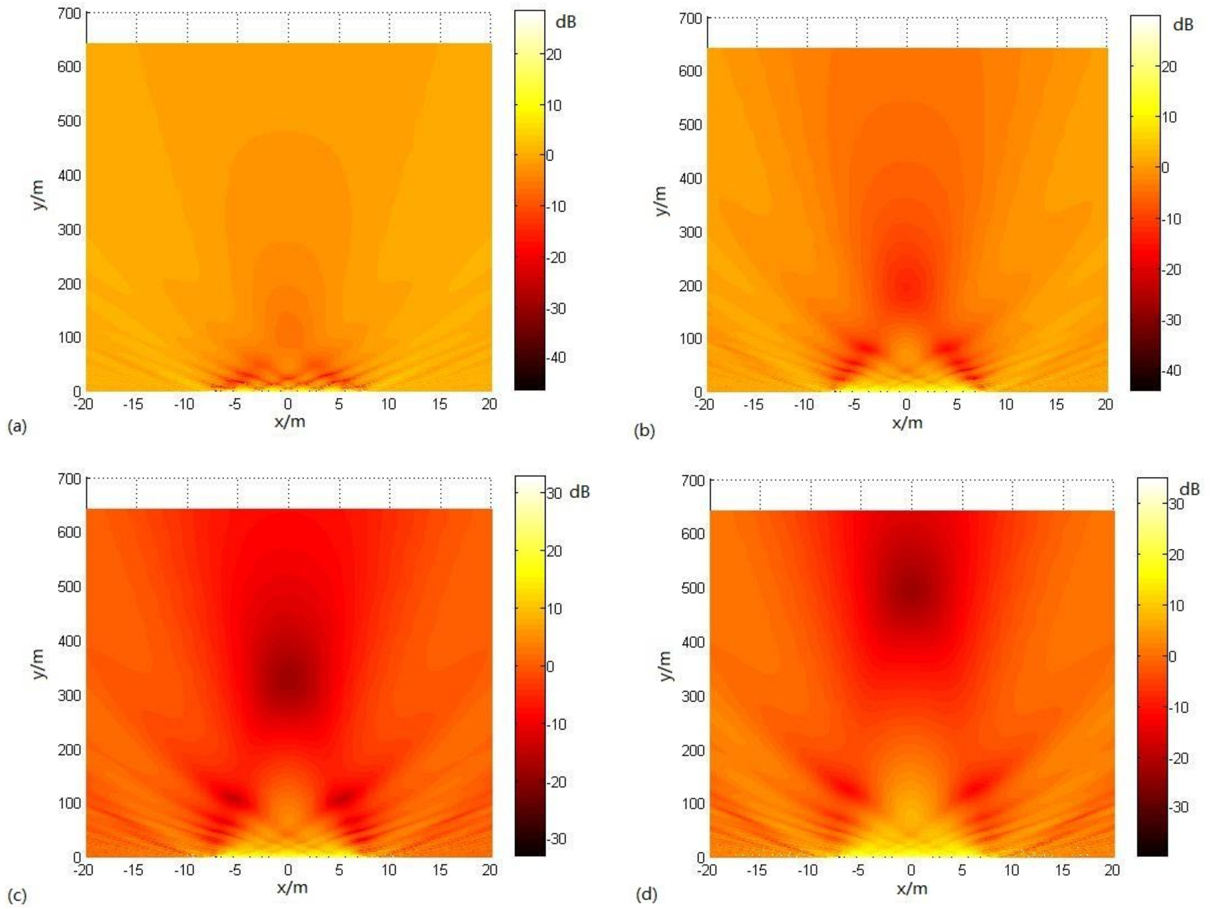

Fig.4: The quiet zones created when the distance between the error microphones and secondary sources is (a) 10m (b) 30m (c) 50m (d) 70m.

Fig.4 (a) shows the shape of the quiet zones when R_{se} is 10m. Quiet zones with more than 10dB deduction appear periodically in the area near the error microphone plane, and reduction areas extended to the far field. Even in the area where the distance between the centre of the secondary sources and the observation point is more than 400m, about 2dB reduction can be found. Fig.4 (b) shows the shape of the quiet zones when R_{se} is 30m. Quiet zones are created in both sides of the error microphone area and are aimed towards the central axis in the far field, but reduction areas didn't appear in the centre of the error microphone areas. In the far field, quiet zones converged and a big quiet zone with more than 10dB reduction is formed in the area where the distance between the centre of the secondary sources and the observation point is 150m-270m. Similar results happened when the error microphones are placed in a farer place.Fig.4 (c) shows the shape of the quiet zones when R_{se} is 50m. Quiet zones converged and a big quiet zone with more than 10dB reduction is formed in the area where the distance between the centre of the secondary sources and the observation point is

225m-500m. Fig.4 (d) shows the shape of the quiet zones when R_{se} is 70m. Quiet zones converged and a big quiet zone with more than 10dB reduction is formed in the area where the distance between the centre of the secondary sources and the observation point is 350m-600m.

It can be shown that when the distance between the error microphone area and the secondary area increase, the main quiet zone with more than 10dB reduction will be larger and will appear in a farer place, but the sound pressure in the centre of the error microphone area will increase as price. If the target area is in the far field and the target area, the system above will be a way of solution.

4. Conclusion

A new control system is invented in this paper. Directional loudspeaker array, microphone array and constrained points are used as a unit, when different units work together, large quiet zones can be achieved while much less computation is needed. In the simulation, when the distance between the error microphones and secondary loudspeakers are large enough, very large quiet zones can be achieved in the far field, while only a little increase in the other areas occurs. Experiments are expected using this control system.

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