PREPRINT NO. 763 (D-2)

THE EFFECT OF MICROPHONE AND LOUDSPEAKER DIRECTIONAL CHARACTERISTICS UPON RECREATING ACOUSTIC FIELDS

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PRESENTED AT THE
39th CONVENTION
OCTOBER 12-15, 1970

AN AUDIO ENGINEERING SOCIETY PREPRINT

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THE EFFECT OF MICROPHONE AND LOUDSPEAKER DIRECTIONAL CHARACTERISTICS UPON RECREATING ACOUSTIC FIELDS

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The importance of microphone and loudspeaker directionality is discussed and graphical data are presented. Both the acoustics of the initiating and reproducing environments, and the number of transmission channels dictate the choice of optimum microphone and loudspeaker directional characteristics and placement. Apparent discrepancies between theory and practice can often be traced to practical limitations of the transducers and/or environments.

I. INTRODUCTION

Various systems have been proposed during the past decade which attempt to add depth information to present two channel reproducing systems.[1-8] Systems have been proposed which transmit the direct sound and the reflected sound over the two channels in such a manner as to put "transmission of room information" or "ambiophony" ahead of lateral localization.[1,2] Others have proposed multichannel systems which transmit both lateral and depth information.[3-6] Much effort has recently been spent in attempts to achieve the transmission of both lateral and depth information while maintaining two channels.[7,8] Some, if not most of this effort has been put forth in the interests of achieving compatibility with the present two channel, tapes, discs, and broadcasting formats.

It would appear, however, that certain theories, and practical systems have proven unsatisfactory due, to limitations, not necessarily in the realm of theory, but rather in practical details of the recording and reproducing process. While it is true, that much of the difficulty in achieving satisfactory multichannel reproduction lies in the area of practical microphone and loudspeaker deviations from ideal, much is also due to the way in which they are utilized. In the case of one method of recording, currently being proposed, most of the problems of the recording process are eliminated by using direct pickup of individual sources and positioning them in space according to a very well defined computer program.[9] This type of processing still leaves the program material at the mercy of the loudspeaker-listener relationships and the acoustics of the reproducing environment.

Also, the general feeling concerning some of the attempts at producing a compatible four channel to two channel system, via matrixing, has been somewhat less than enthusiastic. Perhaps some new insights can be gained if a careful analysis is made of the recording and reproducing process.

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[1] J.G.McKnight, "Why Stereo? The Philosophy of Multichannel Recording of Music", J. Audio Eng. Soc., Vol. 8, No. 2, P. 87 (April 1960)

[2] Klaus Wendt, "The Transmission of Room Information", J. Audio Eng. Soc., Vol. 9, No. 4, P. 282 (Oct. 1961)

[3] John W. Hogan, "Four-Channel Stereo Adds Depth To Tapes", Electronics World, May 1961 (Preprint available from Nortronics Co. Inc., Minneapolis, Minn.)

[4] M. Camras, "Approach to Recreating A Sound Field", J. Acous. Soc. Am., Vol. 43, No. 6, PP. 1425-1431 (June 1968)

[5] J. Cunningham, "Tetraphonic Sound", db, Vol. 3, No. 12, PP. 21-23, (Dec. 1969)

[6] Granville Cooper, "Tetrahedral Ambiophony", Studio Sound, Vol. 12, No. 6, P. 233 (June 1970)

[7] E.R. Madsen, "Extraction of Ambiance Information From Ordinary Recordings", J. Audio Eng. Soc., Vol. 18, No. 5 (Oct. 1970).

[8] David Hafler, "A New Quadraphonic System", Audio, Vol. 54, No. 7 P. 24, (July 1970)

[9] John M. Chowning, "The Simulation of Moving Sound Sources", J. Audio Eng. Soc. Preprint No. 726 (presented at 38th Convention May 1970)

II. SAMPLING THE ACOUSTIC FIELD

A. Free Field vs. Diffuse Field

In order to isolate the possible problem areas in the recording and reproducing process, it is best to define the conditions under which the originating time-space continuum will be sampled. Since it has been noted for many years that a highly reverberant environment tends to reduce the accuracy of localization [10], it would probably be better for purposes of analysis to specify free field conditions in the originating time-space continuum. This will allow a study of the source-microphone relationship without the interaction of environmental acoustics effects. Before the study can be completed for practical systems, the effects of a diffuse field must be taken into consideration, but it also must be remembered that a human listener has the advantage of the precedence effect of binaural hearing. [11-13] For a listener with binaural hearing, there remains, even in a fairly uniform diffuse field, some ability to sense the direction of the source when it emits sounds of a short, transient nature. Most listening is done under only slightly reverberant conditions as compared to the diffuse field conditions necessary to measure total loudspeaker radiated power or microphone random energy efficiency. It has been noted that even in the relatively reverberant conditions which are found in good concert halls, much directional information is present and can be measured.[14] To say that directionality is of no consequence in a diffuse field is true only in Plato's renowned "World of Ideas". As the diffusivity of the field increases, the number and direction of relevant reflections increases in magnitude. As mentioned previously, this will tend to impair the ability of a listener to determine source location. As the number of transmission channels is increased, to more accurately define the acoustic field parameters of the originating time-space continuum, the deleterious effects of the diffusivity of the reproduce time-space continuum become more apparent.

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[10] J.C. Steinberg and W.B. Snow, "Symposium On Auditory Perspective - Physical Factors", Elec. Engrg., Vol 53, PP. 12-17, (Jan. 1934)

[11] H. Wallach, E.B. Newman, and M.R. Rosenzweig, "The Precedence Effect in Sound Localization", Am. J. Psychol., Vol. 52, PP. 315-336, (1949)

[12] Helmut Haas, "Uber den Einfluss eines Einfachechas auf die Horsamkeit von Sprache (on the Influence of a Single Echo on the Intelligibility of Speech)", Acustica, Vol. 1, PP 49-58,(1951)

[13] Mark B. Gardener, "Historical Background of the Haas and/or Precedence Effect", J. Acous. Soc. Am., Vol. 43, No. 6, PP. 1243-1248, (June 1968)

[14] Von R. Thiele, "Richtungsverteilung und Zeitfolge Der Schallruckweirfe in Raumen" Acustica, Vol. 3, P. 291, (1953)

B. The Omnidirectional Microphone

Figure 1. shows the frequency vs. amplitude response curve of a dynamic omnidirectional microphone. The frequency response tends to drop off, at the higher frequencies, as the angle of incidence approaches 180°. This is a common characteristic of all omnidirectional microphones of finite dimensions. The dashed curve represents the random incidence response of the microphone in a diffuse field. If the microphone is oriented vertically, the frequency vs. amplitude response will be uniform for all angles around it in the horizontal plane.

C. Single Point Sampling and Reproducing (Monophonic)

Figure 2. shows a single omnidirectional microphone placed in a free field time-space continuum at (T1). With vertical orientation, as noted, the response will be uniform at all angles in the horizontal plane. If a source moves around the microphone at a constant radius, the sound pressure at the microphone will remain constant. A listener at position (A), listening to the sound reproduced by a loudspeaker at (T1) in a second, free field, time-space continuum would sense no movement of the source. If, however, the source moves to position (X) or (Y), the intensity of the sound will decrease or increase and the listener will assume that the source has moved away from or toward him.[15]

D. Two Point Sampling and Reproducing (Lineal-phonic)

For many years, two channel sampling and reproducing systems have been designated as being either binaural or stereophonic.[16] From an analytical viewpoint such systems should probably be referred to as lineal-phonic since the maximum information that can be gained from such two point sampling and reproduction would, at most, define a line between them. The microphones do not have the binaural hearing capability of a human listener. They merely sample the acoustic field at two points in space. For example, the apparent effects of a receding or approaching source, under free field conditions, may be simulated, with a stationary source, by merely decreasing or increasing the intensity of the sound emitted by the source.

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[15] John M. Eargle, "Stereophonic Localization: An Analysis of Listener Reactions To Current Techniques", I.R.E. Trans On Audio, Vol. AU-8, No. 5, PP. 174-178, (Sept.-Oct. 1960)

[16] R.J. Tinkham, "Binaural or Stereophonic?" Audio Engineering, Vol. 37, No. 1, P.22 (Jan. 1953)

When the interaction of a source and two omnidirectional microphones under free field conditions is considered, some interesting insights can be gained. Figure 3. shows two microphones, T1 and T2, located 12 feet apart in a free field environment. This environment is the originating time-space continuum. The sources at S1, S2 and S3 all lie on a circle with a radius of 8 feet. This circle describes a condition whereby the relative intensities at microphone T1 and T2 differ by a constant 6dB ratio. The sound pressure at T1, produced by a source on this circle, is always 6dB higher than that at T2.

T1 and T2 also designate two loudspeakers, in a second free field time-space continuum, with a listener equidistant from both. When a source moves on this circle, a listener will judge movement away from or toward him at a certain angle between the loudspeakers. This is represented by the dashed line of Figure 3. The arrows indicate the apparent motion of the source. When the source is between the two microphones, it will appear to be closest to the listener. When the source is located at the farthest point from the two microphones, it will appear to be farthest away from the listener. The dB levels of Figure 3. show the relative sound pressures for a source which produces 100dB at 1 foot.

When the time delays between the source and the two microphones are considered, another most important factor becomes apparent. At the closest distance from the two microphones, the time delay differential (TDD) is equal to: [17]

$$TDD_1 = 7.1 \text{ msec} - 3.6 \text{ msec} = 3.5 \text{ msec.}$$

At the farthest distance, this becomes:

$$TDD_2 = 21.3 \text{ msec} - 11.7 \text{ msec} = 9.6 \text{ msec.}$$

Therefore, the change in time delay differential between closest and farthest distances from the two microphones is:

$$\Delta TTD = TDD_2 - TDD_1 = 9.6 - 3.5 = 6.1 \text{ msec.}$$

This variation in time delay will have a definite effect upon localization. As the distance between the microphones is increased, the 6dB circle will increase in radius. This means that the change in time delay differential will become greater. An exact plot was not made, but the apparent shift in source localization is shown by the dotted line of Figure 3.

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[17] The Transit Time for a Sound Wave is Based Upon The Velocity of Sound of 1129 ft/sec at 20°C and 760mm.(1/1129=.886 msec/ft.)

E. Three Point Sampling and Reproducing (Planar-Phonic)

When three microphones are used to sample an acoustic field, they may be used in two different ways. The first way allows a more precise delineation of the direct sound radiation by a group of sources and is intended for reproduction via two channels. Although more than two microphones are used, the end result can still be considered as lineal-phonic.

The second method includes the use of three transmission channels and three loudspeakers. In the past, such a three channel system has been used basically to obtain better delineation of the direct sound from a group of sources.[18] The two microphones are augmented by a centrally placed microphone. Generally all three microphones are on a line parallel to the group of sound sources.

Consider a tri-array microphone arrangement. T1, T2, and T3 of Figure 4, represent such an array. Such a three point arrangement will allow a planar sound field to be sampled. The microphones shown in Figure 4. are 12 feet from each other in a free field time-spacecontinuum. A line passing through T1 and T2 is at 90° to a line passing between T1 and T3. A source at S2 is on the same 8 foot radius circle as shown in Figure 3. and is shown in a position which causes the sound pressure level at T2 and T3 to be 6dB less than that at T1.

The distance between S2 and the microphones is shown as well as the respective time delays. It is interesting to observe the effect of loudspeakers at T2 and T3 upon the listener at Ⓐ . While the direct sound in the originating time-space continuum would arrive at the listener from the left-front, the sound reaching the listener in the reproduce time-space continuum, arrives from the left-front and also from the left-rear and right-front. This is unnatural and tends to be disconcerting, however the fact that the level of the sound emitted by T2 and T3 is 6dB down from that of T1 and the fact that there is an 8.8 msec time delay for the sound from T2 and T3, tends to allow the listener to locate the sound as coming from the area of T1. With the reproduce arrangement as shown there are, of course, no time delays caused by the speaker interrelationships with the listener.

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[18] Harvey Fletcher, "Symposium on Auditory Perspective, Basic Requirements", <u>Elect. Engrg.</u>, Vol. 53, PP. 9-11 (Jan. 1934)

As the source moves on the circle as shown in Figure 4, the localization that obtained with the two microphone , two loudspeaker arrangement of Figure 3 is no longer valid. As the source moves toward T3, it will shift its apparent location in that direction. If the source moves off the circle toward Ⓧ, it will not only appear to move away but because the sound pressure differential at T2 and T3 relative to T1 will be less than 6dB and the time delay differential will be reduced below 8.8 msec., the ability of listener to locate the source will be diminished. The sound will appear to come from an area toward the left-front. As the source moves to position Ⓨ , the localization of the sound is made much easier. The sound pressure at microphone T1 is 17.1dB greater than that at T2 and T3 and 19.5dB greater than that at T4. Also the time delay of a sound emitted from the loudspeakers at T2 and T3 is 12.6 msec behind that emitted from T1 while the sound emitted from T4 is 14.8 msec behind the emitted from T1.

F. Four Point Sampling and Reproducing (Planar-Phonic)

When the fourth microphone T4, is added to the tri-array as shown in Figure 4, the localization process is made even more difficult. With the source at S2 again, the sound pressure at T4 is 8.3dB less than that at T1 but only 2.3dB less than that at T2 and T3. Also, the time delay of the sound emitted from a loudspeaker at T4 is 14.4 msec later than the sound emitted from a loudspeaker at T1 but only 5.6 msec. later than the sound emitted from loudspeaker at T2 and T3. The fact that the listener in the reproduce time-space continuum receives signals from all four quadrants with relatively little sound pressure and time differential, while if he were in the originating time-space continuum he would hear sound from the left front direction only, is a meaningful clue as to why certain matrixing schemes may cause unenthusiastic listener reactions.

Figure 5 shows another relationship between a source, S3, and the four microphones. In this case, the sound pressure at T2 and T4 is 6dB less than that at T1 and T3. The time delay of sounds emitted from loudspeakers at T2 and T4 relative to the sounds emitted from loudspeakers at T1 and T3 is 9.5 msec. If the listener maintains the head orientation as shown, the source will be localized with much less confusion than the location of S2 in Figure 4. This is because binaural hearing is most effective for sounds arriving perpendicular to the ear.[19]

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[19] L.J. Sivian and S.D. White, "Minimum Audible Sound Fields", J. Acous. Soc. Amer.,(April 1933)

Figure 6 represents a worst case condition with respect to sound localization. The source at ⓧ has moved to a position which still maintains the 6dB ratio of sound pressure between T1-T3 and T2-T4 but the time delay between the sound emitted by loudspeakers at T2 and T4 will be only 4.8 msec. behind the sound emitted by T1 and T3. Whereas the real source should appear to be only a little over four feet from the listener and directly to the left, the sound arriving from all four quadrants via the loudspeakers, will cause severe disorientation.

Again, we can see the relationship of the four microphone, four loudspeaker arrangement, to the matrixing of four electrical channels. The matrixing systems can cause an even more severe disorientation since there is no time delay differential between the sounds emitted from the four loudspeakers and intensity differences alone must be relied upon to provide localization clues.

Figure 7. shows the development of a 6dB differential path. This is the path along which a source must travel if a 6dB minimum sound pressure differential is to be maintained between any one microphone or pairs of microphones and the remaining microphones. Of course, the time delay differentials will vary somewhat. If the listener maintains the head orientation shown, the sound will tend to go in and out of focus, so to speak, as it travels on the prescribed line. This is due to both the varying time delay differential and the variations in acuteness of the sound localization ability of the listener with respect to angle of sound incidence.[20]

G. The Cardioid Microphone

The microphone specified up to this point in the discussion, being omnidirectional, has responded equally to sound incidences 360° in a horizontal plane. The sound emitted from a loudspeaker directly opposite the position of the real source as shown in Figure 4. has caused disorientation. If the level of the sound emitted from T4 of Figure 4. could be reduced, this disorientation could be reduced. Figure 8. shows the amplitude vs. frequency response of a cardioid condenser microphone. The relative smoothness of the response of this microphone at various angles off axis is apparent. This characteristic of performance allows a cardioid microphone to be utilized as a directional acoustic attenuator. The dashed line shows the random incidence response of the microphone in a diffuse sound field.

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[20] Ibid.

Figure 9. shows the increase in ratios between the sound pressures emitted by the four loudspeakers in the reproduce time-space continuum, due to the use of the cardioid microphone. The actual sound pressures in the originating time-space continuum are the same as those of Figure 3. however, the response of the cardioid microphone discriminates as to the angle of incidence whereas the omnidirectional microphone did not. The sound pressure from loudspeakers at T2 and T3 will be -18dB with reference to the sound pressure at loudspeaker at T1. This is a 12dB increase over that obtained with the omnidirectional microphone.

The loudspeaker at T4 will produce a sound pressure which is at least 28dB less than that from T1. This represents a 20dB increase. Thus, a considerable enhancement in sound localization can be achieved when the cardioid microphone is employed.

ANALYSIS OF AN ORIGINATING TIME-SPACE CONTINUUM

A. Source, Microphone, and Listener Locations

Figure 10. represents a plan view of a room which is 100 feet in length and 60 feet wide. A group of sources is located in an area measuring 20 feet by 10 feet as shown. A source at Ⓧ is centrally located 10 feet from the front wall. The sources at Ⓨ and Ⓩ are located 20 feet from the front wall 20 feet from the respective side walls. Since there will be reflections from the side walls, care must be taken in the placement of microphones.[21] Microphones T1 and T2 are located 10 feet from and directly in front of Ⓨ and Ⓩ respectively. T1 and T2 are also 20 feet from the side walls. Microphones T3 and T4 are located 30 feet from T1 and T2 and 16 feet from the side walls. A listener at Ⓐ is centrally located 30 feet from the source group area.

B. Direct Listening Conditions

If it is assumed, for analytical purposes, that sources Ⓧ, Ⓨ and Ⓩ each produce a sound pressure level of 100dB at one foot, the sound pressure at the listener can be calculated. The sound pressure levels produced at the listener position by the direct sound from Ⓧ and Ⓩ are 68dB and 70dB respectively. A first order reflection from the side wall produced by a sound emitted by source Ⓧ is also shown. Whereas, the direct sound traveled 40 feet, this first order reflection must travel a distance of 72 feet. If the absorption of the air and the boundary surface is ignored, the sound pressure level of this first order reflection would be 62.8dB or 5.2dB less than the direct sound pressure. The time delay between the direct sound and this first order reflection is 28.6 msec. Even though the level difference is only 5.2dB, the 28.6 msec delay allows the listener at Ⓐ to determine the position of the source at Ⓧ quite easily.

It is also worth noting that a first order reflection from the rear wall will have to travel 140 feet. This reflection can produce a sound pressure at the listener of 57dB, again neglecting the effects of absorption. This level is 11dB less than that produced by the direct sound but the time delay will be on the order of 88.6 msec.

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[21] R.V. Waterhouse, "Interference Patterns in Reverberant Sound Fields", J. Acoust. Soc. Amer., Vol. 27, No. 2., 254 (1955)

C. Relationships Between Sources, Reflections, and Microphones

A great deal of information may be extracted by a careful analysis of Figure 10. Only a few main points will be touched upon. Source Ⓧ will produce sound pressure levels at microphones T1 and T3 of 73dB and 65.5 dB respectively. If T3 is a cardioid microphone, and is used as a directional acoustic attenuator, the difference in sound level emitted by the loudspeakers in the reproduce time-space continuum can approach 27.5dB rather than 7.5dB. The time delay between the sound arriving at T3 with respect to the sound arriving at T1 is 26.7 msec.

The first order reflection from the side wall to microphone T3 will travel 67.3 feet and will arrive at T3 13.5 msec later than the direct sound. This compares with the time delay differential of 28.2 msec for the direct vs. reflected sound with the listener at Ⓐ . The difference in sound pressure level can be 14.1dB as opposed to only 5.2dB for the listener at Ⓐ described previously. The lower sound pressure level of this wall reflection during the reproducing process can help overcome the effect of the shorter delay.

Comparing the reflection from the rear wall, as sampled and reproduced, with that heard by the listener at Ⓐ it will be seen that the time delay will be 124 msec vs. 116 msec, a reduction of only 8 msec. The difference in level will be only 6dB. It is apparent then, that with the microphone placement as shown, the reflections from the rear of the room can be reproduced with a very close approximation to those heard by a listener at Ⓐ . The side wall reflections are not as accurately reproduced with respect to time delay and level, but it would appear that a decrease in the sound pressure level, caused by using a cardioid microphone as a directional acoustic attenuator, can compensate for the shorter time delay due to fact that the microphones are in different physical locations than the listener at Ⓐ .

IV. ANALYSIS OF SOME REPRODUCE TIME-SPACE CONTINUUMS

A. Standardizing Loudspeaker-Listener Positions

The ideal situation would be one in which the optimum loudspeaker and listener positions are specified and adhered to. The sound pressure and time delay differentials could be specified for various size rooms. The time delay function is directly proportional to distance, being .886 feet per msec. The sound pressure function is proportional to the square of the distance. This difference in proportionality could be a problem. However, the sound pressure level produced by the various loudspeakers may be varied at will by manipulation of the volume controls of the associated amplifiers. It would be quite easy to determine the volume control settings necessary to compensate for differences in listener distance from the various loudspeakers. Once the volume control settings are made, the listener would remain in the specified position. This is, of course, completely unsatisfactory for most listening situations.

The answer to the problem lies, not in standardizing the loudspeaker listener positional relationship, but in making certain that the time delay differential between the various microphones is great enough to compensate for the negative time delay differential caused by a listener being too close to a loudspeaker which should be emitting a delayed sound with respect to one or more of the other loudspeakers. This increase in time delay, obtained by wide spacing of the microphones in the originating time-space continuum will allow a satisfactory reproduction, of the source location and environmental acoustics of the originating time-space continuum, over a large listening area of the reproduce time-space continuum.

B. Quadrant Loudspeaker-Listener Positions

Figure 11. shows a room 30 x 22 feet with the listener toward the middle of the room and the loudspeakers near the four corners. If the sound from the group of sources shown in Figure 10, are reproduced via this arrangement, (ie: the loudspeaker at T1 of Figure 11. reproduces the sound picked up by T1 of Figure 10.), the time delay differential at the listener will be 7.2 msec less than that existing in the electrical signals. This is because the listener at Ⓐ in Figure 11. is 8.1 feet closer to loudspeakers T3 and T4 than he is to loudspeakers T1 and T2. This means that the electrical signals to loudspeakers T3 and T4 must be delayed by a minimum of 7.1 msec if the sound is to appear to come from the original area between T1 and T2. This delay may be accomplished

by microphone positioning in the originating time-space continuum or by other electrical or mechanical time delay schemes. The microphone placement of Figure 10. provides enough time delay (26.7 msec, T1 to T3) to allow a listener to approach loudspeaker T3 very closely and still locate the sound as originating from near loudspeaker T1. The spacing between the loudspeakers T1 and T3 of Figure 11. is 26 feet which corresponds to a time delay of 23 msec. To a listener near T3 loudspeaker then, the sound will arrive from loudspeaker T1, 26.7 - 23.0 or 3.7 msec earlier than the sound emitted from loudspeaker T3. Of course, the higher sound pressure level from T3 would have to be reduced for a listening position near T3 if the time delay advantage of T1 is not to be overcome.

C. Front-Side Loudspeaker-Listener Positions

Figure 12 a. and 12b. show two possible arrangements of the loudspeakers and the listener in a room which measures 20 x 16 feet. The close proximity of the listener at A) to loudspeakers T3 and T4 in Figure 12a. causes a time delay differential of 9.9 msec. which must be overcome if the sound is to appear to come from the area between T1 and T2 as it should. Again, the microphone spacing shown in Figure 10., which provides a minimum time delay differential of 26.7 msec, easily provides the necessary margin, (26.7 - 9.9 = 16.8 msec) The listener can approach loudspeaker T3 or T4 very closely and still locate a sound as originating from between loudspeakers T1 and T2. Of course, the foregoing analysis assumes that the volume controls of the amplifiers supplying electrical signals to loudspeakers T3 and T4 have been adjusted to reduce the signals by at least the 10.2dB level difference shown in Figure 12a.

If the listener approaches T3 or T4 more closely, then the level must be reduced even more. The sound pressure and time delay differentials shown in Figure 12b. indicate that this arrangement of loudspeakers and listener is to be preferred to that of Figure 12a. This is because both the sound pressure and time delay differentials are reduced from those of Figure 12a.

Figure 13. shows a room of 26 x 20 feet and the loudspeaker-listener relationship. It will be seen that the wider spacing has reduced both the sound pressure and time delay differentials. This would tend to indicate that wider loudspeaker spacing is desireable. This would be true if it were not for two facts. The wider loudspeaker spacing can cause a "hole" in the reproduced sound field if the listener moves between any two loudspeakers and they do not have wide angle dispersion. Also, the greater distance of the listener from the loudspeakers will allow the direct to reverberant sound pressure ratio to decrease. As noted previously this will

reduce the accuracy of localization.[22] If the sound pressure vs. frequency relationships which existed at the microphones in the originating time-space continuum are to be reproduced accurately throughout a wide listening area in the reproduce time-space continuum, the loudspeakers must exhibit a uniform frequency response over a certain horizontal angle.[23] For most listening situations this angle should be at least ±45° from the horizontal axis of the loudspeaker. Radiation angles greater than ±45° will only cause more acousticalenergy to be available for reflections from the walls which tendsto reduce the ability of the listener to locate the relative positions of the original sources. The dispersion at angles off the vertical axis can be less. In fact, a narrower vertical dispersion can reduce the amount of acoustical radiation available for floor and ceiling reflections which tend to confuse localization.

D. Frontal Area Loudspeaker Arrays

Figure 14. shows another arrangement of four loudspeakers and a listener.[24] At first, this arrangement might seem to be non-complimentary to the arrangement of Figure 10. If it is remembered that a four point sample was made of the acoustical field in the originating time-space continuum, then there appears to be no reason why the recreated acoustical field cannot be heard by a listener outside its imaginary boundries. Referring to Figure 10. once again, it can be seen that a listener at position B will be in an acoustical field created by sound waves passing through the area bounded by imaginary lines joining the four microphones. The variations in sound pressures and time delays have been continuously sampled by the four microphones. When the listener at A in Figure 14. hears the sounds emitted by the four loudspeakers, he is listening, essentially to the acoustical field as it existed around the four microphones of Figure 10 but from a position outside the field. The negative time delay differential of 7.5 msec is easily overcome by the positive time delay differential of 26.7 msec caused by the microphone arrangement of Figure 10. The difference in sound pressure level of 4.0dB is the smallest of any of the five arrangements shown. The radiation pattern for loudspeakers T1 and T2 should be similar to that described earlier. Loudspeakers T3 and T4 may have a broader radiation pattern since they are not operating from a corner position. A broader radiation pattern may not be desireable however, since it will allow more indirect sound which can decrease the ability of the listener to locate the original source positions.

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[22] J.C. Steinberg and W.B. Snow, Op. Cit., P. 16.

[23] E.C. Wente and A.L. Thuras, "Auditory Perspective - Loudspeakers and Microphones", Electrical Engineering, P. 17, (Jan. 1934)

[24] J. Cunningham, Op. Cit.

V. CONCLUSIONS

A method has been presented which will allow acceptable recreation of the directional aspects of the sounds emitted by sources in an originating time-space continuum. This method is based upon analysis of the relationships between the direct and reflected sounds, their relative intensities and time delay differentials with respect to the number, placement, and directional response characteristics of the microphones. In most cases, the originating time-space continuum will be larger than the reproduce time-space continuum. This will allow the advantage of wide microphone spacing. This will mean that the time delay differential will be greater in the electrical signals than that caused by the loudspeaker spacing encountered in most time-space reproduce continuums. As the number of channels is increased, both the directional response of the microphones and the radiation angle of the loudspeakers should be narrower. Even in a reverberant acoustic field, meaningful directional clues exist and can be sampled by using directional microphones. These directional clues, consisting of both intensity and time delay differentials can be injected into the reproduce time-space continuum in a manner that will allow a listener to sense both the location of the source and the characteristics of the originating time-space continuum.

Comparisons between microphone and loudspeaker spacial relationships point up the difficulties encountered when attempting to recreate acoustic fields after passing them through simple electrical matrixing systems.

The important effects of time delay differentials between microphones and loudspeakers cannot be neglected if satisfactory results are to be achieved.

The difference in intensity and time delay as functions of distance, cause acoustic field distortions which can make the positional accuracy of a moving source extremely difficult to recreate.

EXPERIMENTAL VERIFICATION

Recordings were made, using a four channel tape recorder in order to verify the conclusions arrived at during the foregoing analysis. Both omnidirectional and cardioid microphones were employed alone and in various combinations. An "up close" recording of a large musical ensemble via two microphones plus two microphones out in the auditorium as shown in Figure 10, provided very satisfactory results when reproduced by systems discussed in Section IV B., C. and D. and similar to those shown in Figures 11, through 14. Recordings were made of moving sounds and of persons talking while moving in and around various microphone arrays, both in and out-of-doors. These recordings tended to prove that when a source moves to the center of an area bounded by an imaginary line joining the four microphones, confusing directional information is presented by the loudspeakers. The greatest accuracy is produced when a moving source is outside the boundaries and at a great enough distance to allow a relatively smooth change in intensity and time delay differentials between the various microphones.

Pseudo-acoustic fields were created which tended to verify the importance of intensity and time delay differentials. Both single channel and two channel programs were re-recorded on the front two channels of a four channel recorder. The single channel program material was first processed through comb filters to derive the two channel information. The rear two channels were supplied with the same material but after a certain delay. Frequency shaping the spectral distribution of the rear channel signals would have enhanced the effect, but for the purposes of this study this was not felt to be necessary. In most cases even trained observers had difficulty in telling the true four channel from the pseudo four channel programs. All agreed however, that both types of four channel programs were superior to simple two channel programs in their ability to create a realistic acoustic field.

Acknowledgements

The author wishes to thank Messrs. Charles McConnell, James Wellwood and Charles Mattas of Ampex Corporation for their assistance in making the experimental recordings as well as Messrs Robert Schulein and James Kogen of Shure Bros. and Mr. Andrew Braken of AKG for the use of various microphones. The cooperation of Mr. Daniel Queen, Manager of Acoustics. Amplivox Div., Perma Power Corp. in measuring the diffuse field response of microphones is gratefully acknowledged. Thanks are also due to Mr. James Cunningham, Manager of Eight Track Recording Company for the many useful and informative discussions.

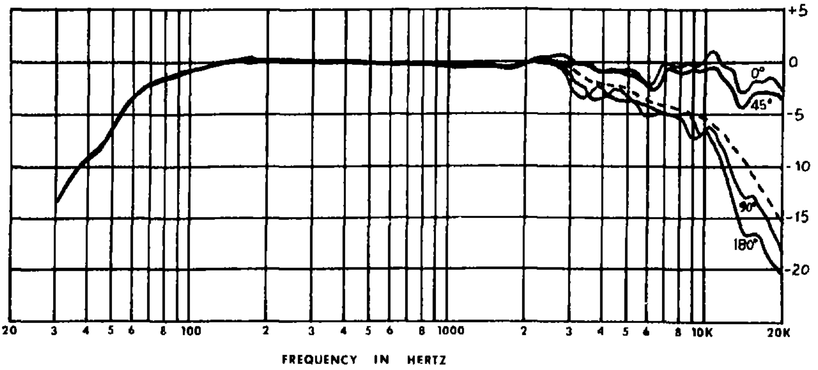


FIGURE 1. Response of Shure 578 omnidirectional dynamic microphone. Free Field response at various angles (solid curves) and random incidence, diffuse field response (dashed curve) are shown.

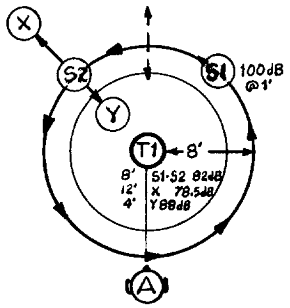


FIGURE 2. Monophonic conditions. Inner circle is 6' reference radius used in all following Figures. Source on 8' radius will not appear to move, while source movement to Ⓧ or Ⓨ will be discernable during single channel playback.

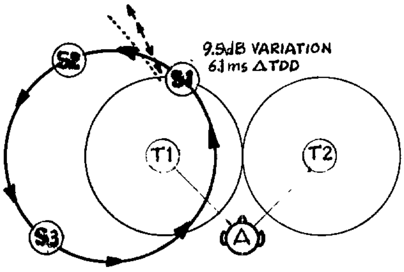


FIGURE 3. Lineal-phonic conditions. Source moving on 8' radius will be heard as movement shown by arrows due to constant 6dB SPL differential. Dashed line shows position due to Time Delay Differential. Microphones 12' apart as used in following Figures.

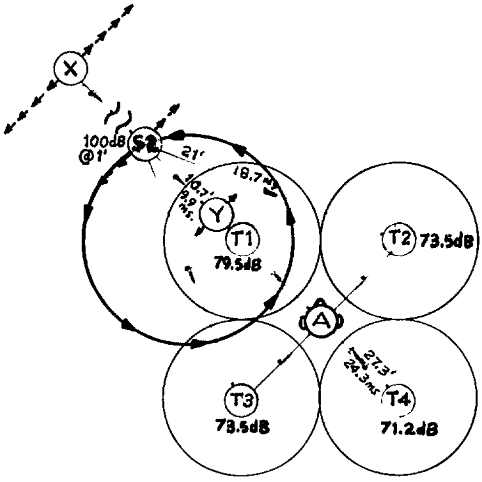


FIGURE 4. Planar-Phonic conditions. Tri-array T1, T2, & T3 provide more directional information which can cause confusion as to exact source location. T4 adds information from the opposite direction compounding the confusion. Source at Ⓧ is heard in a general area, while source at Ⓨ is more exactly placed during playback.

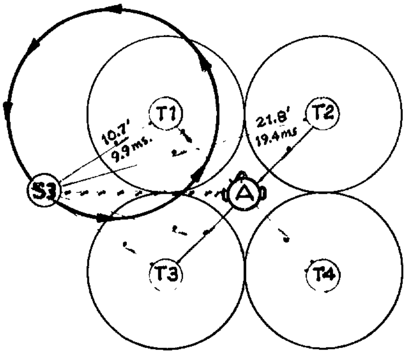


FIGURE 5. Planar-Phonic conditions. SPL from T2 and T4 will be only 6dB less than SPL from T1 and T3. Time Delay Differential is 9.5 milliseconds. Information from T2 and T4 causes confusion as to location of source.

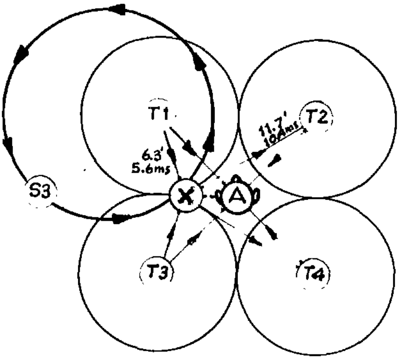


FIGURE 6. Planar-Phonic Conditions. Same as Figure 5, except source moves to Ⓧ . SPL differential remains 6dB but Time Delay Differential is only 4.8 milliseconds. Severe disorientation results.

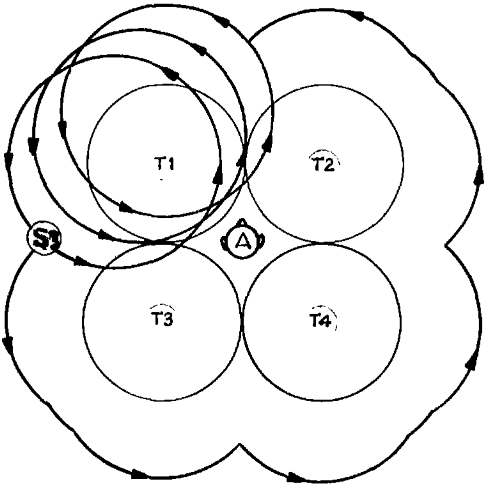


FIGURE 7. 6dB SPL differential path for Source and 4 microphones.

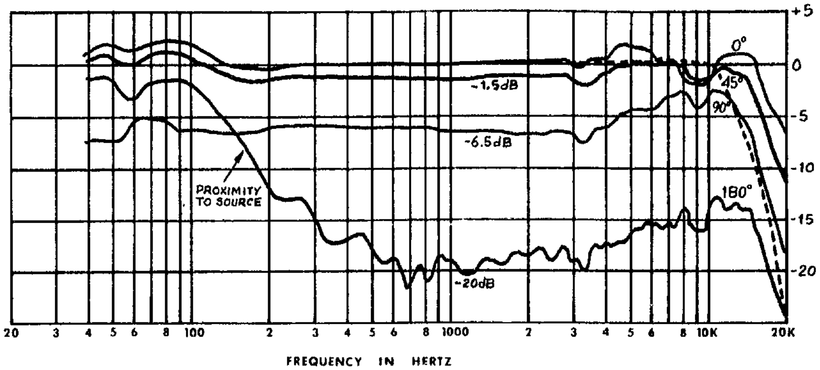


FIGURE 8. Response of AKG 451E cardiold condenser microphone. Free field response at various angles (solid curves) and random incidence, diffuse field response (dashed curve) are shown.

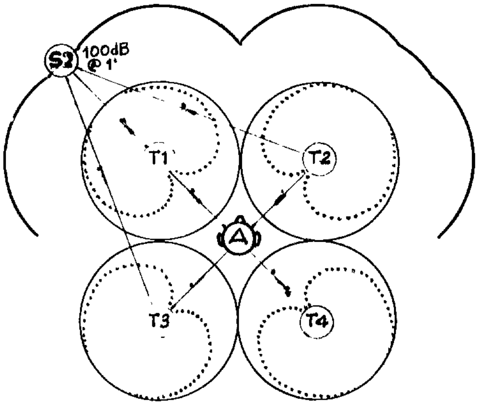


FIGURE 9. Same as Figure 4. except cardioid microphones are used to increase SPL differentials.
S2-T1: Same as Figure 4
S2-T2,T3: -6dB + (-12dB) = -18dB (9.2 msec re: T1)
S2-T4: -8.3dB + (-20dB)*≅ 28dB (16.6msec re: T1)
*-20dB@ 180° for practical cardioid microphone.

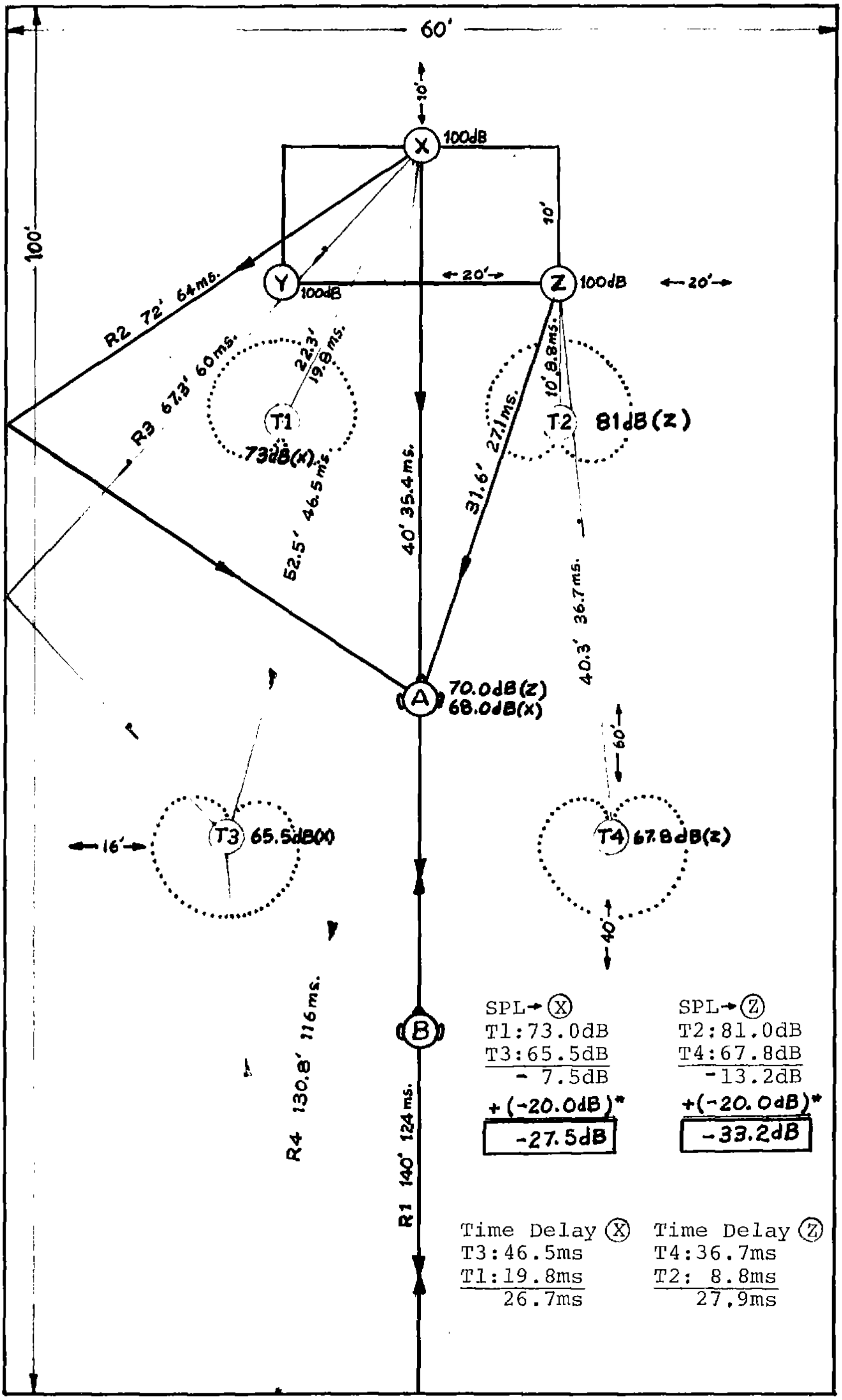


FIGURE 10. Source, microphone relationships in a real environment. 4 channels are shown. Same scale as previous Figures. SPL @ 1' from sources (X), (Y), and (Z) is 100dB.

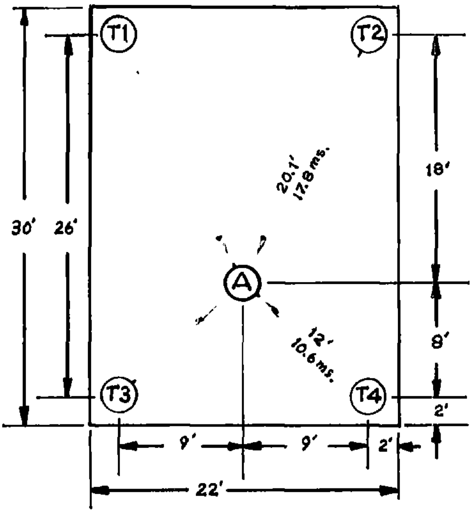


FIGURE 11. Quadrant loudspeaker arrangement. SPL re: 90dB @1'.

T4: 69.2dB SPL
T2: 64.4dB SPL
4.8dB SPL

T2: 17.8ms.
T4: 10.6ms.
7.2ms.

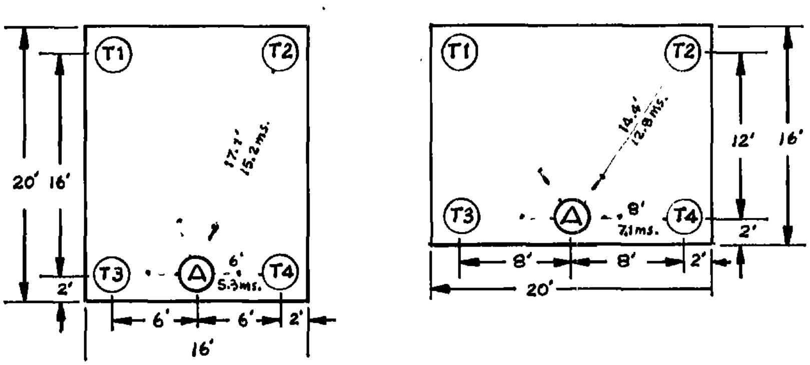


FIGURE 12. Front-Side loudspeaker arrangment SPL re: 90dB @1'.

a) T4: 76.0dB
T2: 65.8dB
10.2dB SPL

T2: 15.2ms
T4: 5.3ms
9.9ms

b) T4: 73.2dB
T2: 67.5dB
5.7dB

T2: 12.8ms.
T4: 7.1ms
5.7ms

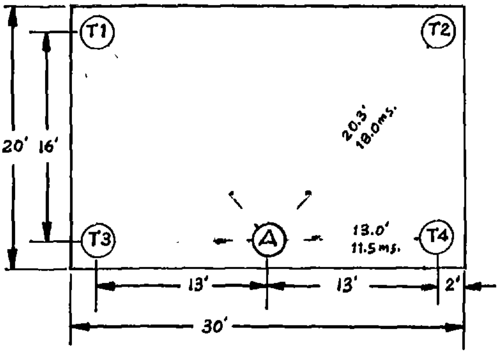


FIGURE 13. Front-Side loudspeaker arrangment. SPL re: 90dB @1'.
T4: 68.5dB T2: 18.0ms
T2: 64.3dB T4: 11.5ms
4.2dB 5.5ms

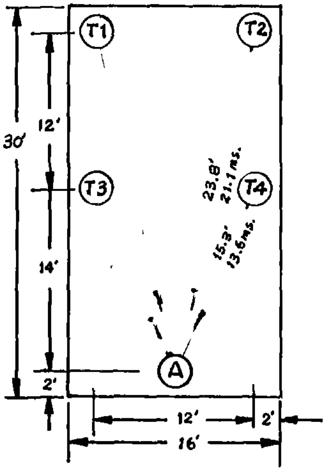
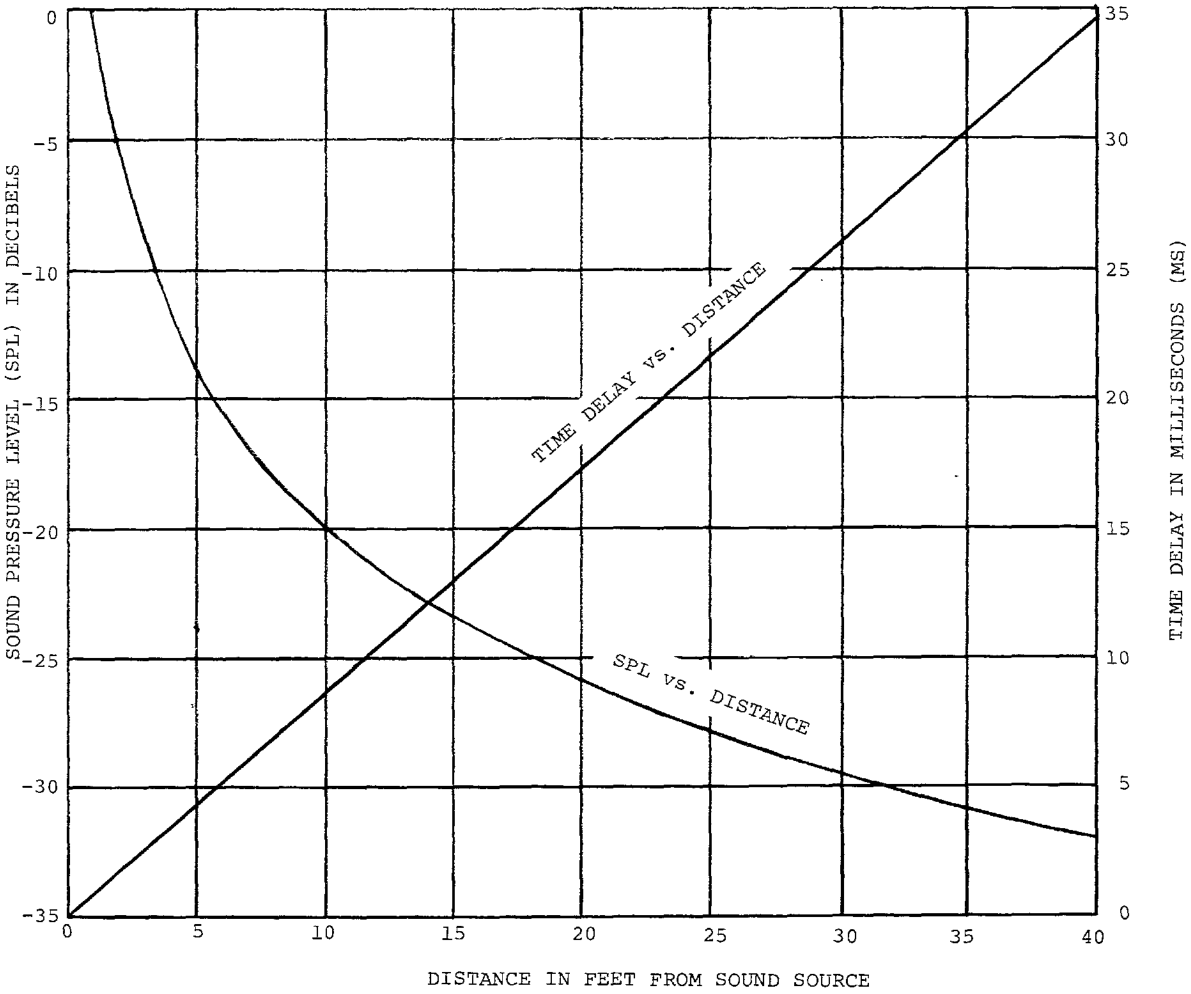


FIGURE 14. Front-front loudspeaker arrangement SPL re: 90dB @1'.
T4: 66.9dB T2: 21.1ms
T2: 62.9dB T4: 13.6ms
4.0dB 7.5ms


SOUND PRESSURE LEVEL (SPL) IN DECIBELS
0
-5
-10
-15
-20
-25
-30
-35
TIME DELAY IN MILLISECONDS (MS)
35
30
25
20
15
10
5
0
0
5
10
15
20
25
30
35
40
DISTANCE IN FEET FROM SOUND SOURCE
TIME DELAY vs. DISTANCE
SPL vs. DISTANCE