A Computer-Controlled Sound Distribution System for the Performance of Electroacoustic Music

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1. Introduction

Since the advent of electroacoustic music, and even earlier, musicians have shown an interest in using the spatial distribution of sound as a significant consideration in composition. While there have been many attempts to realize the musical potential suggested by this possibility, certain problems have intervened. Significantly, these have been primarily in the technological rather than the musical domain. In electroacoustic music, the general problem can be considered to be the evocation of cues which cause an acoustic event to be perceived at a particular location in acoustic space. Such cues can be considered in terms of two categories: those affecting directional perception and those affecting distance perception. The current paper investigates the problems in the former category, the evocation of angular cues. [2]

Following a brief discussion outlining musical motivation, considerations pertinent to the control of directional cues are discussed. Included are both technological and psychological problems. After a survey of relevant approaches, a new system is presented within the context of the Structured Sound Synthesis Project (SSSP) at the University of Toronto. Finally, a working prototype of this new system is presented in order to validate certain of the basic concepts presented.

2. WHY LOCALIZATION? – Musical Motivation

The concept of music in relation to space has intrigued composers, in varying degrees, throughout musical history; however, it was only more recently – when technology seemed to be on the verge of providing tools to explore this aspect of music – that composers began to seriously consider its true potential. Not surprisingly, this rising awareness was strongly linked with the development of electroacoustic music. This is well illustrated by Stockhausen's essay in *Die Reihe*, "Music in Space," (Stockhausen, 1961).

In this essay, Stockhausen discusses historical precedents such as works by Gabrieli and Berlioz, in relation to their compositional exploitation of space. Following this theme, he then discusses the spatial aspects of his own compositions *Gesang der Jünglinge* (1956) and *Gruppen für drei Orchester* (1957). This discussion of precedents, however, does not in itself constitute musical motivation, which Stockhausen saves for the second half of his essay. Here, he presents the now familiar argument which can be summarized as follows: perceptually, music can be considered in terms of five parameters. These are:

1. pitch
2. time
3. loudness
4. timbre
5. location

Traditionally, music has emphasized those parameters at the top of the list, such as pitch and time (viz., melody). Webern's use of motives based on timbre, or "Klangfarbenmelodie," however, is an example of how this traditional hierarchy of parameters is gradually breaking down. Localization, for example, could just as easily function as a basis for motif development. This was, in fact, the premise under which both *Gesang der Jünglinge* and *Gruppen für drei Orchester* were written.

In essence, the justification for developing new tools for musical composition and performance is that such tools expand the realm of musical expression. The development of a

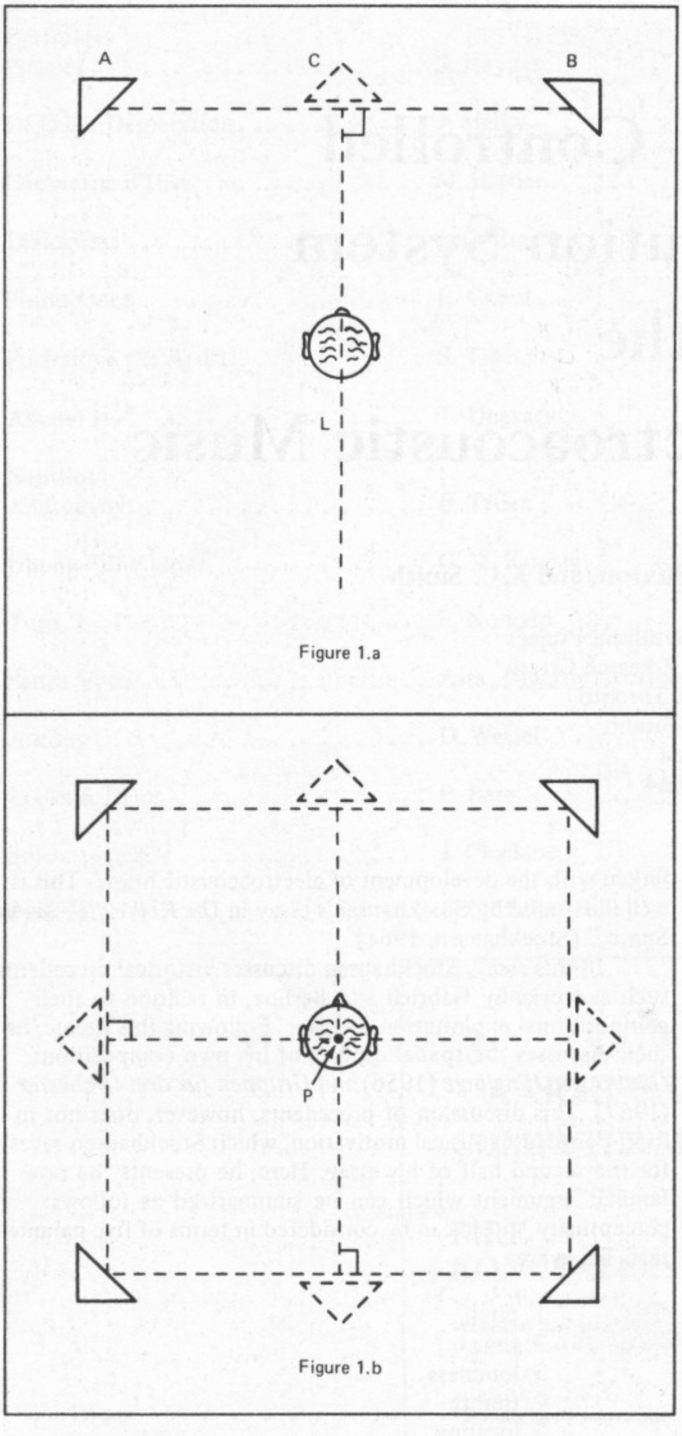

Figure 1.a

Figure 1.b

Figure 1. "Real" and "phantom" sound sources in stereo and quadraphonic sound systems.

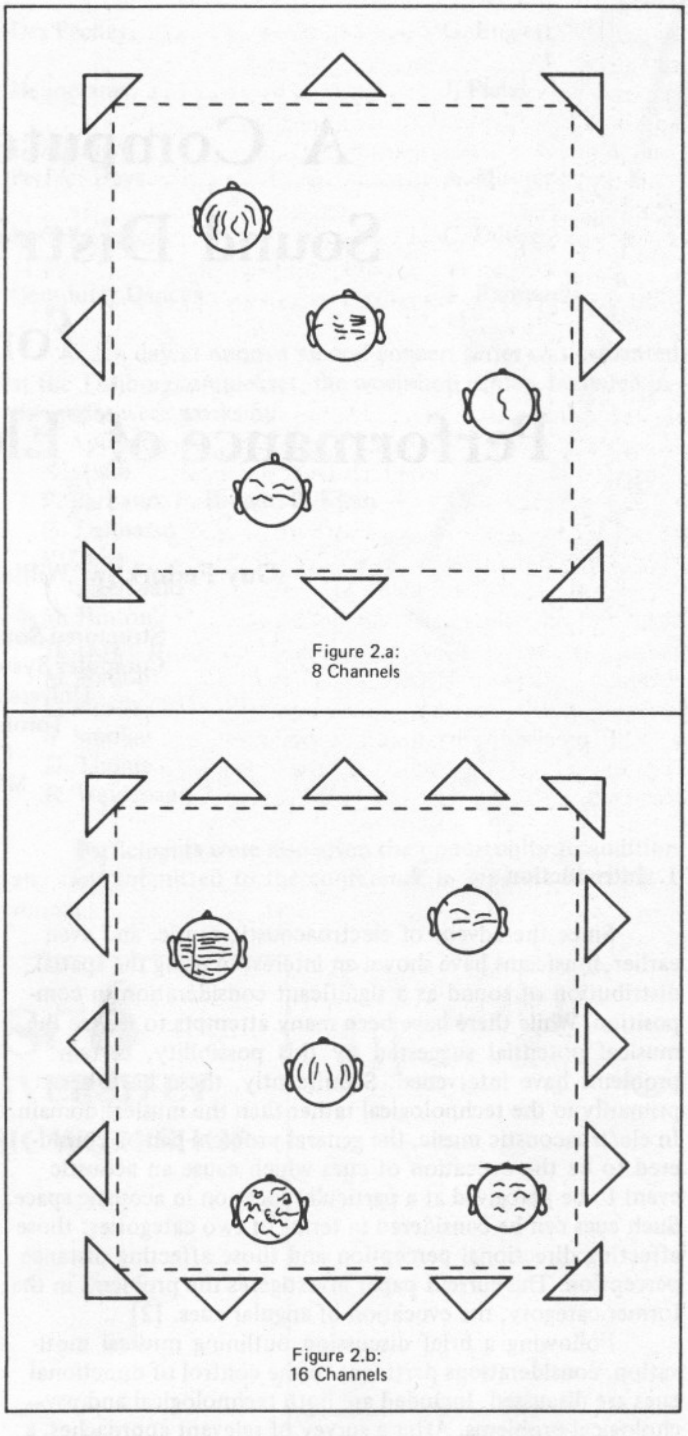
Figure 2.a: 8 Channels

Figure 2.b: 16 Channels

Figure 2. The "brute-force" approach.

tool to enable the efficient control of localization meets this criterion. Just as counterpoint, instrumentation and voicing have traditionally been used as means to separate musical parts, localization apparatus would enable space to be utilized for the same purpose. That the effect can be perceived and is of musical merit is obvious to anyone who has considered the distribution of instruments in the seating arrangement of a symphony orchestra. More formally, we have scientific evidence which supports the suggestion that it is easier to listen separately to two complex sounds if they occupy different position in space, rather than a common one (Koenig, 1950; Schubert and Schultz, 1962). We now proceed, therefore, to discuss the problems in developing a tool which will enable the composer to realize such control.

3. ARE FOUR SPEAKERS ENOUGH? – The Precedence/Haas Effect

If we concentrate for the moment on output transducers, we see that there are two main approaches to controlling directional cues. These are the stereo/quadraphonic approach, on the one hand, and the brute-force, many-loudspeaker approach, on the other. In the stereo/quadraphonic approach, the intention is to utilize a limited number of real loudspeakers to evoke the impression of a large number of "virtual" or "phantom" sound sources. A particular case of this effect, familiar to anyone who has used a pan pot, is shown in Figure 1.a.

Here we see that by applying the same signal, at the same amplitude, to both speakers "A" and"B", we cause the sound to be perceived as emanating from a phantom source "C" midway between "A"and "B". The important thing to realize, however, is that this is only true if the listener is equidistant (within 10 cm) from both speaker "A" and "B". Otherwise, he will perceive the sound as emanating from that source ("A" or "B") to which he is the closest. This is the precedence/Haas effect (Snow, 1954; Haas, 1972; Wallach et al., 1973). Figure 1.a deals with the simple case of stereophony. If we examine the case of quadraphony, as in Figure 1.b, we see that our problems increase. Whereas in Figure 1.a the listener needed only to be located on the *line* "L" in order to perceive the phantom source, in Figure 1.b he must be located at the *point* "P" (or within 10 cm of it) in order to perceive sounds originating from any of the phantom sources.

If precise localization is to be an integral part of the composition to be presented, the stereo/quadraphonic approach must be considered inadequate; however, if we retain for the moment the model of a set of sound sources which encircle the listener, as in Figure 1.b, an alternative approach suggests itself. This is the brute-force approach, where we simply replace each of the phantom sources with a real one (Figure 2.a).

By taking this approach, we see that angular resolution *for the entire audience* increases with the number of transducers, such as with the sixteen shown in Figure 2.b. In addition, we have great flexibility in our choice of speaker placement configurations. Besides the common approach of surrounding the audience on a single plane, layouts based on grids, cubic configuration (periphony), etc., can be easily realized. In any of these cases, the composer can be provided with complete control over the music's choreography in space.

In light of the above, one might wonder why the brute-force approach is not in common use. It is the case that we do not live in an ideal world; rather, our situation is one in which pragmatics must often impose compromise. Four-channel tape recorders and pan-pots are common technology; but sixteen or more channel devices are less common and more expensive. The problems, in fact, fall into two main categories. First what does one do about the physical bulk and expense of sixteen or more channels of amplification, especially in a portable system which must be easily installed and reliable? Second, even having such a system, how do we control with ease, for example, four audio signals which must be able to move independently over these sixteen channels of amplification? The remainder of this paper demonstrates how these problems can be overcome through the use of a computer-based system.

4. TWO PROBLEMS – Bulk and Control

4.1. Bulk

In terms of minimizing the bulk and expense imposed by the brute-force approach, two systems are of particular interest. These are the "Sal-Mar Construction" of the composer Salvator Martirano (Franco, 1974), and the "HYBRID IV" system of Ed Kobrin (Kobrin, 1975). What is interesting in these two systems is how they have exploited certain properties of psychoacoustics and electroacoustics in order to counteract the bulk/expense problems mentioned.

These two systems are similar in that both are used for live performance as well as composition. The Sal-Mar Construction is a large electronic instrument which, while not employing a computer, utilizes digital techniques. The system of Kobrin is a hybrid instrument which consists of several analogue signal-generating/processing modules controlled by a mini-computer. Both systems have multi-channel output: Martirano's 24, Kobrin's 16. While the control mechanism for each is quite different, the actual technique of sound distribution is not.

In order to fully appreciate the approach taken by these two systems, we need to consider certain key points. From the domain of electroacoustics, we should recognize that the problems of size and power requirement (hence expense) of the loudspeakers are primarily the result of having to reproduce the low frequency components of the music to be presented. Thus, if we could ignore the low frequency components we could obtain the same "loudness" from speakers which are smaller, lighter, less expensive, and which demand far less power.

Kobrin and Martirano take advantage of this fact by capitalizing upon certain properties of perception. The first of these is that of fusion. For our purposes, fusion means that under certain circumstances, the mind is capable of merging or fusing two signals in such a way that they are perceived as one. Specifically, both Martirano and Kobrin split their audio signals into two bands, one above and one below 200 Hz. For a given signal, only components above this threshold are distributed in space *via* the speaker network. Components below 200 Hz are transduced by a single high quality bass channel. Clearly, the transducers diffusing the mid-high and bass bands, respectively, will generally be located at different physical locations in the listening environment. Nevertheless, the effect to the listener is still that of a unified sound, emanating from a single source. Furthermore, the source to which this re-fused sound is localized is generally that transducer which is emitting the mid-high frequency band! [3] The implication of the above, as implemented in the systems of both Kobrin and Martirano, is that the location of the speaker transducing the low frequency band plays an insignificant part in the localization process. Thus, *perceptually* the same effect can be achieved using one bass and 16 (in the case of Kobrin) light, small, low power, inexpensive loudspeakers, as would otherwise be obtained by using 16 large, expensive, etc., full-range transducers!

In addition to the above, there is yet another benefit to the technique described. By splitting the signal into two bands, each with separate amplification, we are in effect implementing the technique known as "bi-amping." This technique has the advantage that intermodulation distortion is reduced since any clipping resulting from surges in the amplitude of the bass signal has no effect on the upper band. This is in direct contrast with single amplifier systems. That the effect results in a significant improvement in audio quality for signals having high amplitude bass signals (such as electronic music) is clearly demonstrated by sound systems for rock music, where the technique of bi-amping is common practice.

In summary, we see that the systems of Martirano and Kobrin demonstrate how good design can minimize the bulk/expense problems of the brute-force approach. In part six of this paper (The Channel Distributor), it will be seen that the use of multiplexing techniques will even further reduce the amount of wiring in the brute-force approach.

4.2 Control

There are two aspects to the control problem. The first of these is the amount of information that needs to be specified in order to define directional cues. We will refer to this as the "bandwidth" problem. Secondly, we have the problem of providing the musician/user with suitable transducers through which he can exercise his control. This we term the "man-machine communication" problem. Thus, we see that our problems reduce to the questions of "what" and the "amount of" information on the one hand, and "how" it is communicated, on the other.

4.2.1. Control Bandwidth

For the sake of discussion, let us assume a system which must distribute four channels of audio over sixteen discrete channels of amplification. Since each channel of audio can be transduced by any or all of the sixteen channels of output, it takes sixty-four (4 x 16) values to specify the state of the system at any given time. One question is, however, how much information is required for each of these sixty-four values? If the output of the audio channels to any transducer is controlled by switches (as in the case of the systems of Martirano and Kobrin), only one bit of information (on/off) is required; however, if the level of an output channel is to be continuously variable over each output transducer—as in the system to be described in this paper—sixty-four values of 6 to 8 bits of precision are required to describe a system state. Furthermore, if we are to allow controlled transitions from one distribution pattern to another during the real-time performance of a composition, we must provide for a complete system update—all 64 values—about every 20 ms (50 Hz.). In summary, in order to be of musical value, we must be able to provide the system with a large amount of data within severe time constraints (given that a human operator is the ultimate source of control information). We shall now proceed, therefore, to investigate methods of bringing such control within the bandwidth of a human operator.

A key point to consider is that while the bandwidth required to dynamically update the system is beyond the capabilities of a human operator, it is easily within the bounds of a small computer. The problem then reduces to producing a computer program which can generate the high bandwidth update data, given low bandwidth control information. A program which requires two input parameters for each of the four channels being distributed would meet this criterion. The production of such a computer program is rather straightforward [4]. This is especially true when we consider that we need never attempt to cause a sound to be localized to more than one point-source. This follows from the precedence/Haas effect, mentioned earlier. Since the audience will generally hear a sound as coming from a single source, this constraint does not seriously restrict the musical potential of the system. Note also that the restriction is to one source, not to one loudspeaker. We still want to produce "phantom" sources between two "real" transducers, for example when panning from one speaker to an adjacent one [5]. We see, therefore, that a single pair of continuously-varying (x, y) coordinates is sufficient to provide the spatialization-control information for each of the four channels to be distributed. [6] This reduction by a factor of eight of the control information bandwidth is made possible through the use of computer technology. We shall now proceed to discuss methods of obtaining this control information.

4.2.2. Man-Machine Communication

Given that each of the four audio channels to be distributed can be controlled by a stream of x, y coordinates, several methods of obtaining these control parameters come to mind. Each x, y pair could, for example, derive from a "joystick" (x, y) controller, the coordinates of a graphics tablet, or a pair of control voltages such as from an analogue sound synthesizer. The use of joystick controllers, for example, is already common technology in the audio industry, but up to now has been restricted to four-channel—or "quad"—systems. Light-pens and graphics tablets have also been used effectively in four-channel systems such as that of Chowning (1971). Each of these methods of control has demonstrated, therefore, that they can provide the mechanism for a congenial man-machine interface, especially when coupled with good aural and visual feedback response. A key design consideration, however, is that we want to go beyond merely providing a facility which must be controlled in real-time. Of far more interest is to enable the composer to pre-program the sound "choreography" of a composition so that he could limit his real-time activities to "fine-tuning" the performance. This approach—similar to the notion of "conductiing" as in the GROOVE system (Mathews and Moore, 1970) —then frees the composer/performer to consider other parameters (besides localization) during performance.

Luckily, the same computer which enabled us to reduce the control bandwidth also provides the potential for pre-defining the localization cues. Again, various protocols are possible. Kobrin, for example, uses an alpha-numeric command language to specify movement patterns and speeds. Martirano uses touch sensitive devices to define patterns to hard-wired logic controllers. Finally, Chowning provides powerful graphic techniques to program speed and path of motion, as well as doppler shift and reverberation. While the system of Chowning does serve as a very useful model, it has the restriction that it was not designed for real-time performance or for systems having more than four channels. It is clear, however, that much of the power of the Chowning system could be adapted to a portable system which had adequate graphics and computational ability. It is exactly this direction which is being pursued by the SSSP system. To date, two prototype systems have been implemented. The software for the first (Ladd, 1977) is an adaptation of the Chowning system which enables the composer to interactively define, edit, and mix localization functions; building up – piece by piece – the sound choreography of a composition. The second prototype is a scaled-down version of this sytem. It has been designed for the real-time performance situation. The software runs on an inexpensive portable microprocessor system, and is described later in this paper.

5. CONTEXT – The SSSP

Before progressing further with details of our approach, it would be worthwhile to give a brief overview of the research project which forms the context for the work presented. The Structured Sound Synthesis Project (SSSP) is an interdisciplinary project whose aim is to conduct research into problems and benefits arising from the use of computers in musical composition. This research can be considered in terms of two main problem areas: the investigation of new representations of musical data and processes, and secondly, the study of man-machine communication as it relates to music.

Central to this project, as a tool for both the observation of compositional activity and the verification of theoretical postulates, is the development of a medium whereby a musician can interactively define, audition, and modify both single events and complex groups of sound. Such a tool must exhibit high audio quality and an optimal man-machine interface. Additionally, the design should encompass:

- i. polyphonic synthesis: up to 16 voices possible.
- ii. spectrally complex and time-varying sounds: to enable the production of sounds of comparable complexity to those found in nature.
- iii. real-time or near real-time performance: to allow true interaction.
- iv. multi-channel: 16 discrete channels of output to enable directional localization.
- v. compact: to permit portability and use in performance.
- vi. practical: to make a system which is cost effective, reliable, and cognitively accessible.

Over the past twenty months the core of such a system has been developed at the University of Toronto. Buxton and Fedorkow (1977) presents an introduction to the system,

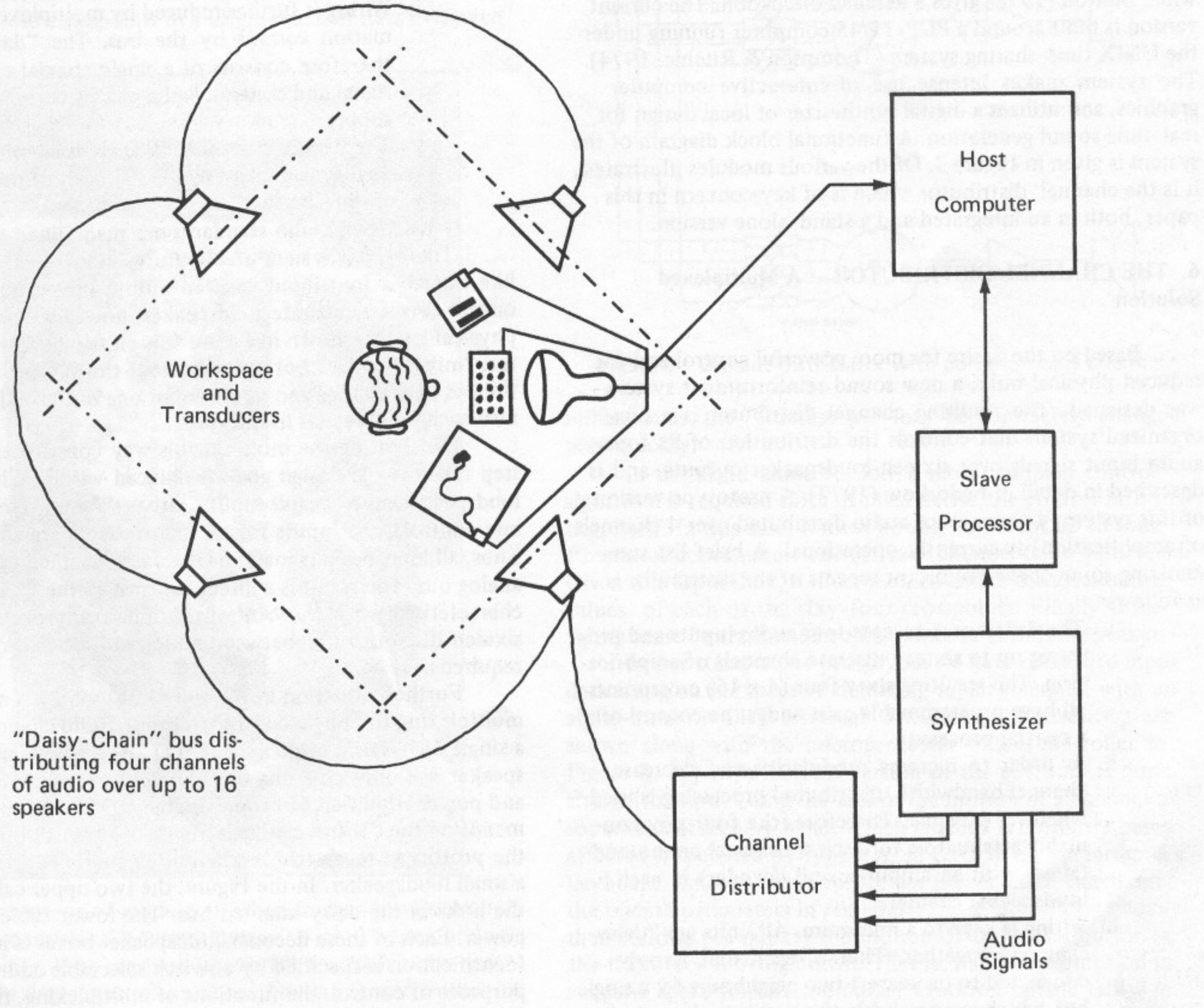

Figure 3. Functional block diagram of physical environment

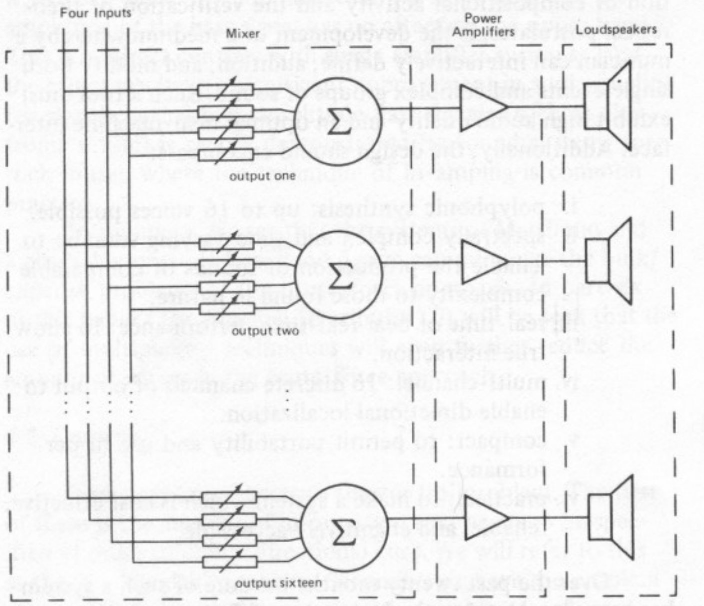

Figure 4. Output distributor, logical organization.

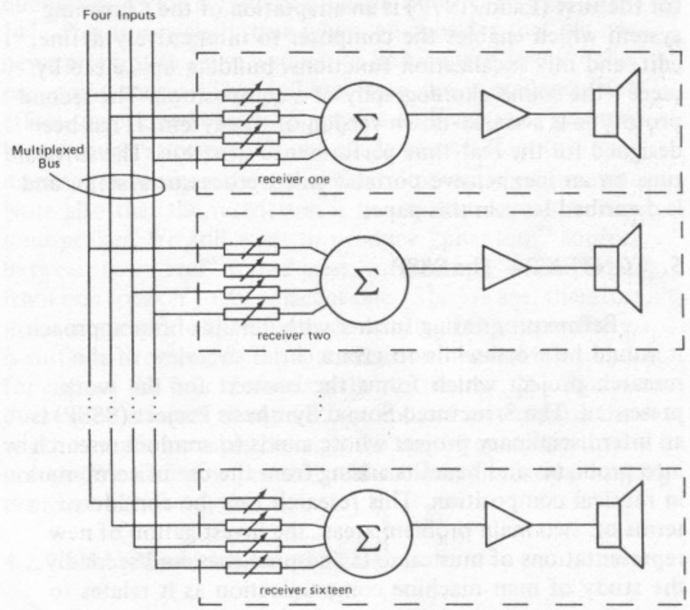

Figure 5. Output distributor, physical layout.

while Buxton (1978) gives a detailed discussion. The current version is built around a PDP-11/45 computer running under the UNIX time-sharing system (Thompson & Ritchie, 1974). The system makes intense use of interactive computer graphics, and utilizes a digital synthesizer of local design for real-time sound generation. A functional block diagram of the system is given in Figure 3. Of the various modules illustrated, it is the channel distributor which is of key concern in this paper, both in an integrated and a stand-alone version.

6. THE CHANNEL DISTRIBUTOR – A Multiplexed Solution

Based on the desire for more powerful control and for reduced physical bulk, a new sound reinforcement system was designed. The resulting channel distributor is a bus-organized system that controls the distribution of its four audio input signals over sixteen loudspeaker outputs, and is described in detail in Fedorkow (1977). A prototype version of this system (4 channels of audio distributed over 4 channels of amplification) is currently operational. A brief list summarizing some of the important aspects of the distributor is as follows:

i. The distributor accepts four audio inputs and produces up to sixteen discrete channels of amplification. The resulting sixty four (4 x 16) crosspoints all have programmable gain under the control of a central processor.
ii. In order to increase modularity and decrease channel bandwidth, distributed processing is used wherever possible. Therefore, the four programmable attenuators for each transducer are housed (along with an amplifier and decoder) in each loudspeaker cabinet.
iii. Wiring is kept to a minimum. All units are "daisy-chained" together. That is, each unit is only connected to its nearest two neighbours by a single bus which carries both the audio and control information.
iv. Wiring is further reduced by multiplexing the information carried by the bus. The "daisy-chain" therefore consists of a single coaxial cable for signal and control, and a pair of wires for power supply.
v. The overall system bulk is reduced while sound quality is improved, all as a result of the technique of "bi-amping."
vi. Good audio standards are maintained throughout.

The overall system architecture, shown in Figure 4, is like that of a four input, sixteen output mixer, where each output drives a separate loudspeaker; however, the system's physical layout, shown in Figure 5, is clearly different, in that the "mixer" is distributed throughout the system. That is, there is an amplifier and a four input one output mixer built into each loudspeaker in the system.

Distributing the mixer in this way constitutes a large step towards the design goal of reduced wiring. While each loudspeaker now has five inputs rather than one (four audio, one control), the inputs for all loudspeakers are the same. Thus, all loudspeakers may now be daisy-chained onto a single analog bus. The result is a direct contrast to the "spaghetti" characteristic with the centralized mixer approach (where sixteen discrete paths between mixer and loudspeakers are required).

Further reduction in the amount of wiring is made by multiplexing the single control and four audio channels onto a single high-speed coaxial cable. [7] As a result, each loudspeaker not only contains the remotely programmed mixer and power amplifier, but some digital logic to recognize commands on the control channel. Figure 6 shows a photograph of the prototype receiver/mixer/amplifier module, attached to a small loudspeaker. In the Figure, the two upper cables are the links in the daisy-chained bus. The lower cable is for power. Each of these decoder/loudspeaker boxes is identical. Identification is specified by a switch selectable address. For purposes of control, the functions of multiplexing, timing and command encoding are performed with the aid of a microprocessor in the centralized control unit.

As mentioned earlier, we have chosen to "bi-amp" our system, that is, to use one or two high power central woofers with a crossover network for each of four input signals. This results in a significant reduction in power and size for each of the sixteen "main" loudspeakers. Figure 7 gives details of connections when this crossover technique is used. It should be noted that crossovers on the inputs to the distributor restrict the use of the system; each of the input channels must have at least one distant location at all times, because the signal's low frequency components in the woofer cannot be turned off. However, in view of the fact that this system is a distributor of finished sounds, and not part of the synthesis process itself, we feel that the reduction in cost, weight and complexity more than compensates the minor restriction in use. Furthermore, at the expense of using full-range transducers throughout the system, one can omit the crossover and by-pass the problem.

In the larger integrated version, commands for the control unit come from the SSSP synthesizer in the form of an address specifying the loudspeaker and channel numbers, followed by the desired attenuation for that crosspoint. Attenuation is specified directly in decibels, in steps of .75 dB. Six bits of data yield over 45 dB of control range at each crosspoint. Because the control unit transmits attenuation values to all sixty-four crosspoints in a single block of data every ten

Figure 6. Prototype of module attached to each loudspeaker. The module consists of a demultiplexor, 4-to-1 mixer, and power amplifier. The modules (up to 16) are "daisy-chained" together by a single multiplexed bus. The bus is seen in the two wires (one in, one out) in the upper part of the photo. The lower wire is for AC power.

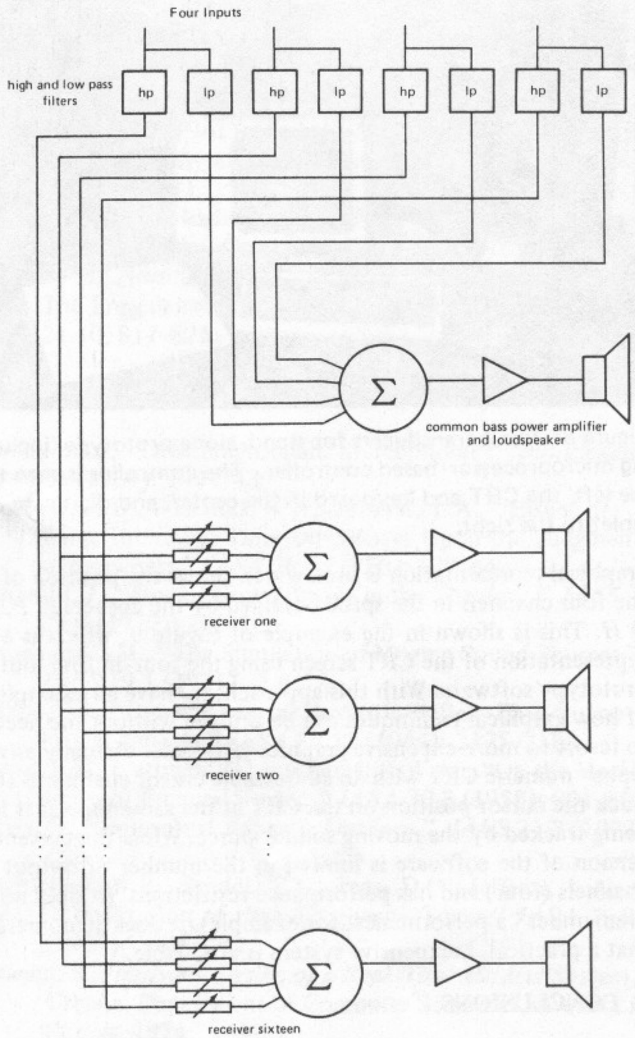

Figure 7. Output distributor with common bass channel.

milliseconds, the "soundscape" may be completely changed very quickly.

In the stand-alone version, a more complex control algorithm is required since it must derive the control information itself. In this case, it must be able to accept up to four streams of coordinates: one for each channel to be controlled. Given this input data, it must then calculate the attenuation values of each of the sixy-four crosspoints. Finally these values must be trasmitted to the mixer in each loudspeaker.

In our stand-alone prototype, we have chosen to input data using an inexpensive digitizing tablet, coupled with an alpha-numeric CRT terminal. These input transducers are shown along with the microprocessor based controller in Figure 8. [8] The current version of the software is quite straightforward. Using the tablet, the motion of a channel of sound is sketched by hand. Which channel is currently being affected is specified by typing *A, B, C,* or *D* on the terminal's keyboard. Since the microprocessor is capable of calculating the update parameters in real-time, the sound of that channel then follows (or *tracks*) the motion of the tablet cursor as the trajectory is being drawn. That is, the sound's motion in acoustic space follows the cursor's motion on the tablet *in real-time*. The CRT is used to give visual feed-back as to the current location of the four channels of sound. A pseudo-

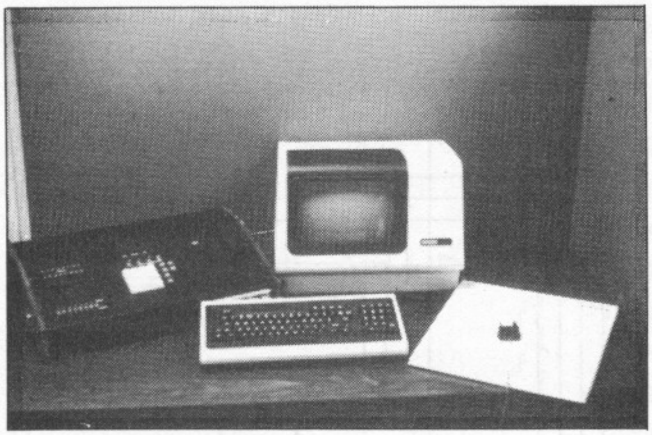

Figure 8. Input transducers for stand-alone prototype, including microprocessor-based controller. The controller is seen to the left, the CRT and keyboard in the center, and digitizing tablet to the right.

graphical representation is used which shows the position of the four channels in the space bounded by the corners *E, F, G, & H*. This is shown in the example of Figure 9, which is a representation of the CRT screen using the four-in four-out prototype software. With this approach, we have an example of how graphical techniques can be utilized without the need to resort to more expensive graphics terminals. Virtually any alpha-numeric CRT with an addressable cursor enables us to track the cursor position on the CRT at the same time as it is being tracked by the moving sound source. While the current version of the software is limited in the number of output channels (four) and has performance restrictions (it does not "remember" a performance, for example), it does demonstrate that a practical, inexpensive system is attainable.

7. CONCLUSIONS

In summary, it can be seen that this system attacks several problems familiar to the musician performing electroacoustic music. First, of course, it provides the hardware to produce directional cues for localization of sound in a concert setting. Furthermore, it does so with a minimum number of interconnections on a single, multiplexed bus. Decoder/loudspeaker units are all identical, and may be connected in any order, resulting in an easy to install and easy to maintain system. Finally, use of a microprocessor controller allows the distributor to be controlled by a wide variety of user-congenial sources, ranging from digital computers to analog control voltages.

8. ACKNOWLEDGEMENTS

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9. FOOTNOTES

1. An earlier version of this work was presented at the IVeme Journees d'Etude de Musique Electroacoustique, Bourges, France, June 1977.
2. The simulation of distance cues is not dealt with here; however, significant work has been carried out in this area by researchers at Stanford University (Chowning, 1971 and Moorer, 1977).
3. This effect is not in conflict with Gardner (1973) as might be at first suspected. Gardner also proposes splitting a monophonic signal into a low and high band. However, he uses the technique in order to produce a phantom source *between* the transducers of the two bands, rather than to have the fused image localized at one of the two transducers (which is our main concern). The effect described in this paper is achieved by maintaining the low (200 Hz.) cross-over frequency. Informal experiments which we have conducted show that for most broad spectrum sounds, as the cross-over frequency is raised up to about 400 Hz., the effect described by Gardner does, in fact, take over.
4. The algorithm used, however, is highly dependent on the configuration of the speakers in the listening area.
5. It is to enable these two cases (real and phantom sources) that we have chosen to provide variable gain at each of the sixty-four crosspoints in the system. Otherwise, single binary (on/off) switches – as used by Martirano and Kobrin – would have been adequate. While enabling panning and the creating of phantom sources may seem a contradiction to our earlier arguments against the stereo/quadraphonic approach to localization, this is not the case. Our main argument was against the worst-case resolution made possible by two and four transducer systems. By utilizing sixteen channels, the worst case situation of our distribution system is acceptable, and through the provision of variable-gain crosspoints, we

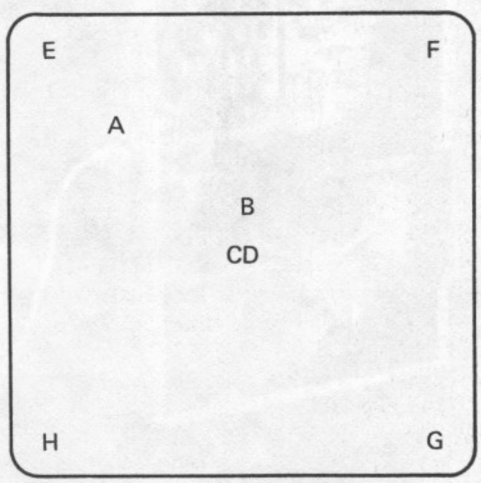

Figure 9. Representation of user's view of CRT when using the stand-alone prototype software. The four channels of sound (A, B, C, and D) are represented in the space bounded by E, F, G, and H. Channels B, C, and D are currently in the centre of this space. Channel A is being dragged off towards corner E. This is accomplished using the cursor of the digitizing tablet.

are able to exceed this worst-case situation for some of the audiences.

6. Clearly the use of more than two coordinates per channel (for example, x, y, z) could be used to musical advantage. For the purpose of this paper, however, we shall limit ourselves to a discussion of the simpler case.
7. A system using multiplexing techniques for the control of distributed audio and lighting systems has been previously described (Jacobson and Crowley, 1977). One main difference between the two approaches is that the system described in this paper not only multiplexes the low-bandwidth control information, but four channels of audio as well. Consequently, wiring is further reduced while greater physical distribution of the system is enabled.
8. The digitizing tablet used is a BIT-PAD, manufactured by Summagraphics, Fairfield, Conn. The controller is built around a Motorola M6800 microprocessor. The program resides in slightly less than 2K of EPROM, thereby avoiding the need for secondary storage. For the terminal, most any alphanumeric CRT with addressable cursor can be used.

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