A (very) brief tutorial on

Sound Reinforcement System Design: Principles and Practice

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1.0 Introduction

There are many ways in which a sound reinforcement system can be designed and implemented. These range from traditional "single source" systems to various types of distributed/delayed systems. While certain approaches are generally preferred, the "best match" of sound reinforcement system to acoustic space should be determined based on a number of performance criteria. What these criteria are and how a particular sound reinforcement system implementation can be "evaluated" on the basis of these criteria will be addressed in Section 2.0.

Closely related to these performance criteria are the factors which can make a particular space difficult to reinforce. This discussion will appear in Section 3.0.

There are various, proven ways to implement sound reinforcement systems, as well as some "unproven" (i.e., having little or no precedent) techniques which advances in digital electronics have recently made possible. A discussion of several different techniques which could potentially be applied, along with a discussion of the relative merits of each approach, will be covered in Section 4.0.

2.0 Sound Reinforcement System Design Goals

Various approaches to sound reinforcement system implementation can be analyzed and compared on the basis of a number of **design criteria**. The manifestations and relative importance of these criteria will be detailed in this section.

Perhaps the foremost design goal in a sound reinforcement system is **evenness of coverage,** i.e., the uniformity with which each person in the seating area can hear the reinforced program material. The way in which uniformity of coverage is typically expressed is as "no more than $\pm$ x dB (decibel) variation" over the seating area. As a point of reference, $\pm$ 1 dB is viewed as the threshold of perceptibility while ± 3 dB is "just noticeable" (and therefore perfectly acceptable). A sound pressure level ("SPL") which is "6 dB down" (or -6 dB) is "half as loud" relative to its "0 dB" (or "reference level") counterpart. In general, a variation of 5 dB or less over the seating area is considered to be acceptable. Ways in which evenness of coverage can be attained will be described in Section 4.0.

Another design criteria is **frequency range** (or **frequency response**) along with the **smoothness** of the **response curve**. Here, two quite different sets of criteria can be applied, depending on the ultimate objective of the sound reinforcement system. If the system is being designed for reinforcement of **speech** only, the frequency response (i.e., the range of frequencies which can be reproduced) need not go lower than 100 Hz nor higher than 4000 Hz. If **music** is to be reproduced, however, this range should be at least 50 Hz to about 10,000 Hz. For each case, the **smoothness** of response (again, measured in dB), affects the **naturalness** of the reinforced sound. Typically, the response curve (for either speech or music) should vary no more than ±3 dB over entire frequency spectrum. Both horns as well as cone-type drivers can have "peaks" and "dips" in their response curves which can have deleterious effects on the naturalness of the reinforced sound. Most modern reinforcement systems therefore utilize **equalizers** (an equalizer is essentially a "tone control" for each third-octave of the entire frequency range) to "smooth out" the system response to the ±3 dB specification.

An absolutely critical design criteria for sound reinforcement systems used for amplification of either speech and/or music is **intelligibility**. This criterion, typically specified as the **articulation loss of consonants** (% ALCONS), is primarily a function of the ratio of the direct sound field energy to the reverberant sound field energy. A typical design goal is for the ALCONS not to exceed 10%. The **direct sound** field, as its name implies, is the sound arriving at the listener’s ear directly from the live talker and/or loudspeaker. The **reverberant sound** field, which is basically uniform throughout an enclosed (reverberant) space, results from **reflections** of the live talker’s voice and/or the amplified signal emanating from the loudspeakers echoed among the walls, ceiling, and floor. To achieve acceptable intelligibility (i.e., ≤10% ALCONS), the direct sound field must exceed the reverberant sound field by **at least 20 dB** at each seating location. This is normally accomplished using **controlled directivity radiators** (in practice, usually "constant directivity" high frequency horns) with their main lobes aimed at the seating area, in order to minimize the energy which might bounce off the walls and/or ceiling and ultimately contribute to the reverberant sound field. So in addition to providing "even coverage" over the seating area, care must be taken to minimize the energy contributed to the reverberant sound field in order to maximize intelligibility.

A design criterion which is more "aesthetic" in nature than those previously discussed is **locality of reference.** In other words, for reinforced sound to be as natural as possible, it should be **perceived** as emanating from the live source. Here it is important to understand the basics of how human hearing determines directionality. The basic point of interest is that human hearing is far more sensitive to horizontal (side-to-side) directionality than it is to vertical (up-down) directionality. Specifically, a loudspeaker mounted even a considerable distance above (up to 40 feet) the live talker will preserve the perceived locality of reference, while loudspeakers mounted to the left and/or right of the live talker will severely distort the locality of reference. One of the main reasons central cluster reinforcement systems are generally preferred over other approaches is due to their superior ability to preserve locality of reference.

Another important factor, closely linked to intelligibility, is a sound reinforcement system's **gain before feedback.** **Feedback** is the annoying "howl" a sound system produces as the result of the direct sound field impinging on the pickup pattern of an open microphone. Without sufficient gain before feedback, the direct sound field level necessary to ensure adequate intelligibility may not be produced. To achieve adequate gain before feedback (just as to achieve a high degree of intelligibility), controlled directivity transducers — in particular, constant directivity high frequency horns — are aimed such that **minimum direct field amplified sound** impinges on the pickup pattern of any open microphone. An important side note is that microphone pickup pattern type (e.g., omnidirectional vs. cardiod) has little to do with feedback control — this is contrary to "popular belief." Rather, the pickup pattern primarily affects the "reach" of the microphone (i.e., the distance one can be away from a given microphone and still generate an adequate voltage level for a good signal-to-noise ratio) — cardiod microphones are more directive and hence have better "reach", while omnidirectional microphones have less reach but typically have a smoother frequency response (and hence are usually best for music pickup). The main culprit limiting gain before feedback, then, is off-axis sound "leaks" from the loudspeakers. It should be noted that distributed overhead systems, particularly for cases in which there is a high ceiling, usually suffer from poor gain before feedback.

One final factor which should be considered in the design of a sound reinforcement system is **headroom**. Usually specified using dB, it indicates the amount of "amplification reserve" the **system** (i.e., the total reinforcement chain — preamp, signal processing, power amp, loudspeakers) has available. Adequate headroom helps ensure that speech/music peaks will be reproduced faithfully without distortion or "crackling" (or, worse yet, destroying components). There are no "hard and fast" guidelines on headroom, but one appropriate "rule of thumb" is to provide 6 dB of headroom for loudspeaker components (e.g., apply no more than 50 watts RMS to a driver rated to handle 100 watts), and at least 3 dB for power amplifiers (e.g., use a 150 watt amplifier where 100 watts are required). The preamp/signal processing chain should have at least 10 dB of headroom.

In summary, the design criteria of primary significance for sound reinforcement systems are **evenness of coverage, intelligibility,** and **gain before feedback**. Also significant, although not as critical, are **frequency range, smoothness of response, locality of reference,** and **headroom**. These criteria will be applied in Section 4 to compare various possibilities for meeting sound reinforcement needs.

3.0 Factors Which Complicate Sound Reinforcement System Design

As if meeting the six design criteria outlined in the previous section weren't difficult enough, a number of architectural factors can further complicate sound reinforcement system design. In this section, these factors and their consequences will be addressed.

Perhaps the biggest problem is dealing with a **highly reverberant space,** i.e., a room in which the reverberation time exceeds 3 seconds (a consequence of highly reflective walls/ceiling). Here, the reverberant field is so easy to excite (and subsequently overwhelm the direct sound field) that extreme care must be taken in choosing and aiming drivers. For spaces which possess moderate reverberation time (1.5 - 2.5 seconds), it is normally possible to design a reinforcement system which will provide high intelligibility.

The next architectural factor impacting sound system design is that of a **long, narrow room.** The best way to envision the complications involved is to imagine trying to "light" the seating area — with uniform intensity — using a collection of spotlights mounted to the ceiling towards the front of the auditorium. One would quickly find that **lenses of different focal lengths** would be necessary to "focus" the spotlights (i.e., adjust their spot sizes) on various areas of the seating space. In particular, "long" lenses would be needed to light areas that were far away, while "short" lenses would be needed to light areas that were closer. Further, dimmer controls would probably be necessary in order to "equalize" the illumination cast on the closer areas versus those further away.

This "light analogy" is very apropos, since sound propagates through space in a manner quite similar to light. The "long" and "short" lenses correspond to different types of high frequency transducers: "long throw" (high Q) horns and "short throw" (low Q) horns, where Q is proportional to the **directivity index** of the transducer. "Uniform lighting" of the space is directly analogous to "uniform sound field coverage."

The complication imposed by the "long, narrow room" is the need to combine various long throw (high Q), medium throw (medium Q), and short throw (low Q) high frequency radiators — all aimed towards different portions of the seating area — into a "cluster." Further, large boxes containing the low frequency transducers must be combined with the high frequency array for full-range frequency response. Not only is the design fairly complex, but the resulting menagerie of components usually lacks aesthetic appeal (a common characteristic of most "central cluster" systems).

The final architecturally-related complication worth mentioning concerns variable room fill. If a "filled seat" has different acoustical properties than an empty seat (e.g., wooden pew, metal chairs, etc.), then the acoustical properties of the room will be a function of the room fill. In particular, the reverberation time will increase and the amount of energy contributed to the reverberant field will likewise increase as the room fill decreases (i.e., sound which would have been absorbed by a human is instead reflected by the hard surface), thus causing the system intelligibility to decrease. Architects aware of this sound system engineering nightmare will help alleviate this problem by specifying that the seats be **padded** with a material possessing spectral absorption characteristics similar to that of a human.

In summary, the architectural complication of particular significance here is that of the "long, narrow room." As will be seen in the section which follows, this factor complicates virtually every approach to sound system implementation.

4.0 Appropriate, Proven Ways to Implement Sound Reinforcement Systems

As one might guess, there are only "so many ways" in which to successfully implement a sound reinforcement system (just as there are only "so many ways" in which to light a room). In this section, all appropriate, proven approaches which could potentially be applied will be discussed. First, though, a little more background must be established.

Throughout this discussion, it is important to remember the "light analogy" described in the previous section — one of the primary goals is to "light the seating space" as uniformly as possible. Unfortunately, however, a major complication to the "light analogy" is that light and sound **do not travel at the same speed;** rather, sound travels at a (rather slow) 328 m/sec or — looking at it another way — traverses approximately one foot in one millisecond.

To understand the potential consequences related to the speed of sound, attention must again be directed toward the human hearing system. Sounds which arrive close together in time are **integrated** (i.e., "connected" together): **early arrivals** are the source of "warmth" and timbre, while **later arrivals** cause the perception of **reverberation**. Sounds which arrive greater than 50 milliseconds apart, however, are perceived as **echoes**. Note that this **50 millisecond time differential** corresponds to (approximately) a **50 foot distance**.

With this in mind, consider someone speaking into a microphone with the amplified signal fed into a loudspeaker which is 50 feet or more **in front of** the live talker. Any listener hearing the live speech augmented by its earlier-arriving amplification will be severely distracted, as a very clear echo will be heard. This "50 foot rule" is extremely critical, and will be considered in detail in the discussion which follows.

4.1 Central Cluster Systems

By far the most commonly used approach to sound reinforcement implementation is the **central cluster** system. As its name implies, all the transducers used to cover the entire seating area are grouped together into a single "cluster." Further, it is normally located along the central axis (midpoint, from left to right) of the room, typically 30-40 feet above the audience seating area.

As indicated previously in the context of the "light analogy" discussion, one of the major challenges in designing a central cluster system is to provide **uniform coverage** across the entire seating space (a desirable goal is to have no more than ±5 dB over the entire seating area, while at the same time minimizing the amount of energy reaching the side walls or ceiling). This is normally accomplished by using a combination of long-throw, medium-throw, and short-throw loudspeaker components, all aimed at different portions of the seating area. If an extremely high Q radiator is required, transducers can be **stacked** to achieve this result.

Many techniques for cluster design have been developed over time, and the results obtainable are quite reliable. A properly designed central cluster system will have excellent coverage, high intelligibility, high gain before feedback, a frequency response which is both wide and smooth, and the best possible (of all available approaches) locality of reference. Further, there is no problem with unwanted echoes often associated with (non-delayed) distributed reinforcement systems, due to the "point source" nature of the cluster.

Because of this long list of advantages relative to other potential approaches, the central cluster is the "first choice" of most audio engineers and acoustical consultants. One major drawback, however, is that most architects do not design large rooms (*inherently* in need of sound reinforcement) with an architecturally pleasing/acceptable place for installation of sound reinforcement hardware, in particular, loudspeaker clusters (most sound system engineers find this both baffling as well as frustrating!). Unfortunately, it is quite difficult to "hide" a central cluster suitable for a large room at its optimum mounting location; further the "naked" cluster is almost always very unappealing from an aesthetic point of view.

Despite all the advantages associated with a central cluster system, then, it is often necessary to consider other potential approaches purely for aesthetic reasons. These will be discussed in the sub-sections which follow.

4.2 Split Source Systems

If architectural considerations obviate some variation of the central cluster approach, perhaps the "next best" choice is some form of **split source** reinforcement system. As its name implies, loudspeaker systems (usually **line source arrays**) are located at the left-front and right-front of the auditorium, slightly forward of the speaking position(s).

The main difficulty with split source systems is with locality of reference. The *preferred* way to design a split source system is to use a two-channel ("stereo") approach, with *each* line source array possessing the capability to cover the *entire* seating area. To preserve locality of reference, a live talker speaking into a microphone on the left side of the room would be reinforced through the left channel, on the right side through the right channel, and in the middle through both channels (an obvious complication, however, is a live talker with a wireless microphone pacing back and forth — here, a manual **pan** by the operator would be required in order to preserve locality of reference). Often, however, those who install split source systems often forgo the multi-channel paradigm, in the interest of cost savings as well as simplicity of operation, and operate the system monaurally. Here, each line source array need only cover one half of the seating area, easing the directivity requirements of each array somewhat.

Split source systems can have a variety of shortcomings: most problems stem largely from the type of line source arrays used, their mounting locations, and their orientation. Good split source systems can provide fairly good uniformity (although normally not quite as good as that attainable with a central cluster system), good intelligibility (line source arrays generally have excellent vertical directivity control), excellent gain before feedback, and fairly good frequency response (some type of multi-way line source is required for "high fidelity" music reproduction however; speech reproduction, though, is usually very good). Further, the headroom and distortion levels associated with line source arrays are generally *superior* to those which can be achieved with a central cluster system. As stated previously, the primary shortcoming of this approach concerns poor locality of reference.

In summary, properly designed split source systems — even those which are operated monaurally — represent a reasonable (and much more aesthetically pleasing) solution to reinforcing "difficult spaces."

4.3 Distributed and Distributed/Delayed Systems

A number of sound reinforcement system variants fall under the categories of **distributed** and **distributed/delayed**. As the name implies, in these types of reinforcement systems, loudspeakers are distributed (in a variety of different possible ways) throughout the room. When a large space is involved (i.e., when any dimension of the room exceeds 50-60 feet), the amplified signal sent to each group of loudspeakers (or "zone") must be **delayed** using **digital time delay processors**) to ensure the amplified sound not arrive at a listener's ear before the sound emanating from the "live talker".

Overhead distributed systems are very popular for providing background music/paging in large spaces (ballrooms, department stores, open concept offices, etc.) — they are unobtrusive, low cost, trivial to design, and can provide fairly even coverage. For these applications, no delay is required since typically they are not used to reinforce a "live talker."

A variety of complications arise when an overhead distributed system is used for sound *reinforcement* rather than merely for providing background music/paging. One fairly serious problem is that if the **difference in distance** between the "live talker" and the nearest overhead loudspeaker (referred to as the **path length difference**) exceeds 50 feet, the early overhead arrival of sound will not only confuse the locality of reference, but also cause the "live" sound (arriving approximately 50 milliseconds later) to be perceived as an echo. Besides being confusing, this phenomenon seriously degrades intelligibility (i.e., %ALCONS discussed previously). To help solve these problems, groups of loudspeakers which are equidistant from the primary live talker location(s) — referred to as **zones** — are fed a signal which is **delayed** an amount of time commensurate with the **path length difference** for each corresponding group of seating locations. In order to implement such a **distributed/delayed** reinforcement system, a **digital time delay processor** with multiple (programmable) taps is necessary along with a **separate power amplifier** for each zone of loudspeakers.

While path length difference problems associated with large distributed overhead systems can be resolved in a fairly straightforward (albeit expensive) manner, there are several other problems inherent to this type of design which are not so readily resolved. As will be seen in the paragraphs which follow, these difficulties stem largely from architectural factors.

Overhead distributed systems are best suited for rooms with relatively low ceilings (i.e., 10 feet or less) which are acoustically "dead" (i.e., sound absorbing). Once again, the reason for this can be ascertained using the "spotlight" analogy. The cone-type loudspeakers usually used for overhead distributed systems have a relatively "soft focus" (i.e., they act as a light source with a fairly "short" lens). In other words, the light "spreads out" rapidly with distance and, consequently, must be fairly close to the "target" to provide adequate "illumination." (An additional complication is associated with cone-type loudspeakers: the "illuminated" area actually varies as a function of the reproduced spectrum, with high frequencies resulting in "small spots" and low frequencies resulting in "large spots". Note that constant directivity horns, typically used for central cluster systems, do not exhibit this anomaly. Finally, note that line source arrays have directivity control in *one dimension* only.)

Use of an overhead distributed system in a room with a relatively high ceiling has several deleterious effects. First, considerably more energy will reflect off the side walls (due to the wide horizontal directivity of either a single loudspeaker or of a line source array), thus contributing to the energy in the reverberant sound field and, consequently, decreasing **intelligibility**. Note that a narrow room further exasperates the problem. Second, if overhead loudspeakers are located near an open microphone, the potential **gain before feedback** of the system is greatly reduced.

For the various reasons cited above — need for delay electronics, need for extra power amplifiers, reduced intelligibility, poor locality of reference, and reduced gain before feedback — distributed overhead systems are typically *not* used for sound reinforcement systems, particularly in long, narrow rooms with high ceilings. There are, however, some *legitimate* uses for distributed/delayed systems in large room sound reinforcement. One is to augment coverage provided to balcony areas, particularly where the balcony ceiling is relatively low. Another is to use distributed, *side-wall mounted* line source arrays rather than overhead mounted loudspeakers. Finally, for particularly difficult spaces, distributed "pew back" speakers (i.e., 4 to 5 small loudspeakers distributed along the back of each row of seats) have been used. This obviously results in an extremely expensive reinforcement system, and should be considered only if all other alternatives are untenable.

In summary, distributed loudspeakers for sound reinforcement systems almost always require digital delay processors along with a separate power amplifier for each delay zone. This adds expense and complexity to the design, but not extraordinarily so. Overhead distributed systems only work well in rooms with low, acoustically dead ceilings. In rooms with high ceilings, overhead distributed systems suffer from reduced intelligibility, reduced gain before feedback and poor locality of reference. Legitimate distributed/delayed reinforcement systems include those which utilize side-wall mounted line source arrays, and those which utilize a large number of "pew back" loudspeakers.

4.4 Recent Advances in Loudspeaker Technology

Recent advances in loudspeaker technology have made possible drivers that possess nonconventional directivity patterns better suited for even coverage of difficult spaces. One example is the ElectroVoice EVI ("Vari Intense") series speaker systems, which throw a rectangular pattern that is proportional to the mounting height and vertical aim angle. "Vari Intense" is a patented technology that throws a 6-10 dB hotter signal to the back of the room. This compensates for the drop in SPL over the longer distance to the back of the room. A single "Vari Intense" horn can replace a short-throw/long-throw horn combination (described previously), reducing the material and labor costs of an installation (and significantly improve its aesthetics).

For overhead distributed systems, new "pattern control" ceiling loudspeakers are now available. An example is the ElectroVoice EVID "HC" (high ceiling) series, which is very effective for reinforcing reverberant "problem" rooms.

Line arrays have become very popular for "touring" (live concert) applications. Electro-Voice, JBL, and other manufacturers of commercial audio equipment market stackable line array components. Optimal aim, placement, and splay of these arrays (as well as design of the array components themselves) is a current research topic.

A "futuristic" sound reinforcement system design is one which utilizes electronically steered loudspeaker arrays. While the idea of delay steering is not particularly new (it was first demonstrated over 30 years ago), it has not become practical until recently. Consequently, there is very little precedence for this type of design methodology.

The basic idea is as follows. Rather than *physically* aiming a large collection of radiators directly toward the intended listening area (as done in the central cluster approach), they are instead mounted *on the ceiling* (facing straight down). The main lobe of the sound field is then **electronically steered** towards the central portion of the seating area by utilizing signal delay and shading (successive amounts of delay are applied to the signal sent to each array element, while amplitude shading is used to suppress sidelobe generation). While this is, of yet, still a relatively expensive solution (due to delay electronics and amplification requirements), it possesses all the desirable properties of a central cluster system — good locality of reference, good uniformity, good intelligibility, high gain before feedback — all without the "ugly hanging mess."

In summary, delay steered arrays represent an exciting alternative to conventional central cluster systems, especially where aesthetics are of major concern. Unfortunately, there is currently very little precedence for this type of design or experience with it. Integrating all the signal processing (EQ and delay), networking, and power amplification electronics with the physical loudspeaker — creating an "intelligent" loudspeaker element — will help make this next advance in sound reinforcement technology a reality.