The Acoustics of Sound Systems For Baseball

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ABSTRACT

Acoustic considerations impact the design of sound systems for baseball stadiums in several important ways. Intrusion by the sound system into adjacent residential communities must often be minimized. Excess attenuation greatly increases the required high frequency component of acoustic power for sound traveling long distances, while wind and temperature gradients modify the effective coverage patterns of loudspeaker systems and exacerbate community noise problems. These atmospheric effects can vary radically from moment to moment.

Reverberation is a major concern in both enclosed and partially enclosed stadiums, as are long delayed reflections. Loudspeaker systems must be carefully designed with these factors in mind. Where a given loudspeaker's sound goes in addition to the audience it is intended to cover is the most important aspect of the design.

These considerations tend to favor use of distributed systems, at the same time architectural considerations make them difficult to implement. And especially with distributed systems, the timing of sound arrivals at each listener's ear is often more important than the overall amplitude of the sound field.

In modern stadiums, other auxiliary systems are required. Crowd mics fed to distributed loudspeakers must bring the "crack of the bat and the small of the crowd" into private suites which are often enclosed. That crowd noise, along with the public address announcer, must also be fed to broadcast media for their use. Timing is also an issue with these systems. This paper addresses both design challenges and practical solutions.

SOUND SYSTEM USES AND NEEDS

Modern stadiums and arenas have evolved over the years to the extent that many are now multipurpose facilities. While some are designed exclusively for baseball, many can serve other sports, and even non-sports functions. Some host events as widely varied as public meetings, religious events, pop music concerts, exhibitions, and tractor pulls.

Most modern sports facilities have chosen to install very powerful sound systems with the capability to excite the crowd with full range music and speech. Those facilities with multiple uses may have multiple systems, or reconfigurable systems. Virtually all sports facilities need these functions:

- A main sound system covering all public spectator areas. This system must support public address functions, sound reinforcement for a small band and one or more vocalists, and playback of music and other programs. In all facilities, this system should cover spectator areas with very good uniformity of direct sound pressure level and frequency response, with very good speech intelligibility, and with wide dynamic range.
- Crowd ambience fed to enclosed boxes and suites (in addition to the main park sound system)
- Public Address systems which feed an official scorer to areas used by the working press.
- Distribution to the electronic press of feeds from the main sound system

(with and without the park announcer), crowd ambience, and wireless mics worn by umpires or referees.

- A distributed sound system which can selectively relay the main sound system, play-by-play audio from a local TV or radio station, and full range music playback to concession areas, concourses, and public walkways.
- A radio transmission system for the hearing impaired.
- The capability to selectively feed any or all of the sound systems from the facility's main security office.
- Permanently installed wiring to route microphones and line level audio around the facility for use with the main sound system and by broadcast media.
- A production intercom system, with permanently installed tie lines extending throughout the facility.

PERFORMANCE REQUIREMENTS

The main sound system should satisfy the following requirements in every patron seat. Requirements are listed in order of their relative importance, with the most critical ones listed first.

- All arrivals within 15 dB of the direct sound should be within 30 ms of the direct sound.
- direct sound levels should be equal within ±3 dB.
- The frequency response of the sound in each listener seat should be uniform within ±3 dB from 250 Hz to 8 KHz, integrated over no more than 30 ms. In the case of point source systems, a high frequency tolerance limit greater than about 5 KHz may be impractical. Where a reverberant field exists, or where there are multiple arrivals at low frequencies, a somewhat longer integration time for low frequencies may be appropriate.
- Ideally, the system should be capable of providing average direct sound pressure levels over the audience which is comparable to or even greater than the level of a loudly cheering crowd. In practice, such levels can be difficult or impractical to achieve, either because loudspeakers would have to be too large, or the system would be too costly.
- If community noise is a concern, program dynamics should be carefully controlled. It is almost always good practice to use good compression and peak limiting on all program material, and especially on the main park announcer.
- Automatic control of system level should be available so that, if desired, the level of voice announcements and other program material is adjusted to follow (but be at least 10 dB louder than) the overall background loudness of the crowd.

ATMOSPHERIC EFFECTS

Leo Beranek said, "Sound waves travel from source to receiver outdoors through an atmosphere that is in constant motion. Turbulence, temperature and wind gradients, viscous and molecular absorption, and reflections from the earth's surface all affect the amplitude and create fluctuations in the sound received. The longer the transmission path through the atmosphere, the less certain the average amplitude and the greater the fluctuations in the received sound."

Sound traveling through the air is affected by the trip in two principal ways. First, there is excess attenuation, which is defined as attenuation beyond inverse square law. It's effect is like that of a low pass filter whose slope is proportional to distance (i.e., a variable pole filter) and whose slope and attenuation varies in a complex way with both temperature and humidity. When the path is only a few tens of meters in length the loss is usually insignificant, in that it can be easily equalized and is relatively constant. At 50 meters, significant rolloff can occur, depending on temperature and humidity, and can begin to limit maximum sound pressure levels above about 6 KHz. At 100 - 200 meters, again depending on temperature and humidity, this attenuation can seriously limit the sound pressure levels that can be achieved at frequencies as low as 2 KHz.

Excess attenuation is approximated by:

$$A_e\,(T) = \frac{A_e(68^{\circ}\,F, 50\%\ \mathrm{RH})}{1 + 2\,\mathrm{x}\,10^{-3}\,\Delta T\ f}\ \mathrm{dB}$$

$$\text{For: } \ \mathrm{f} = \text{Frequency in KHz}$$

$$f = 50\%\ \text{Rel Humidity}$$

$$\mathrm{T} = 50^{\circ}\ \text{-}\ 86^{\circ}\ \mathrm{F}$$

Figures 1, 2, and 3 are plots of excess attenuation over a path length of 200 ft for various combinations of temperature and humidity. Note that around room temperature, excess attenuation is more a function of humidity than temperature, and that the greatest attenuation occurs at very low relative humidity. Figure 3 shows that the 50%, 70%, and 90% data, averaged over 70-90 degrees Fahrenheit, are essentially the same. This data is particularly useful, because it represents both a middle ground and a likely "worst case" condition for most non-desert environments. For this condition, excess attenuation will cause a 3 dB rolloff at 9 KHz. At 600 ft the rolloff will be much more severe -- 3 dB at 2 KHz, 6 dB at 5 KHz, and 9 dB at 9 KHz!

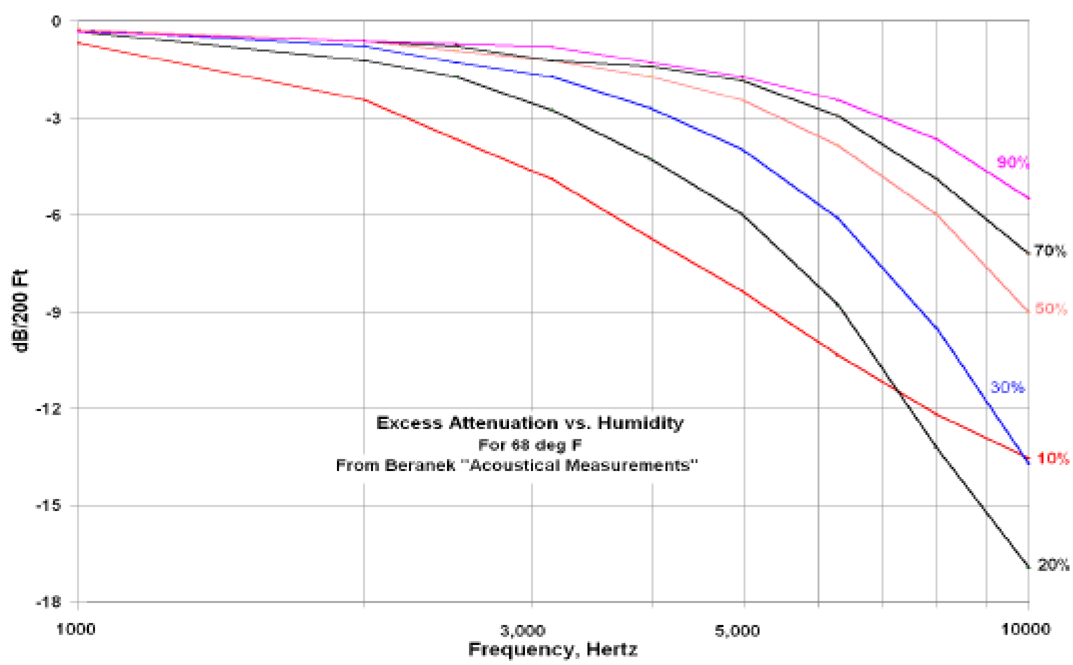

Figure 1

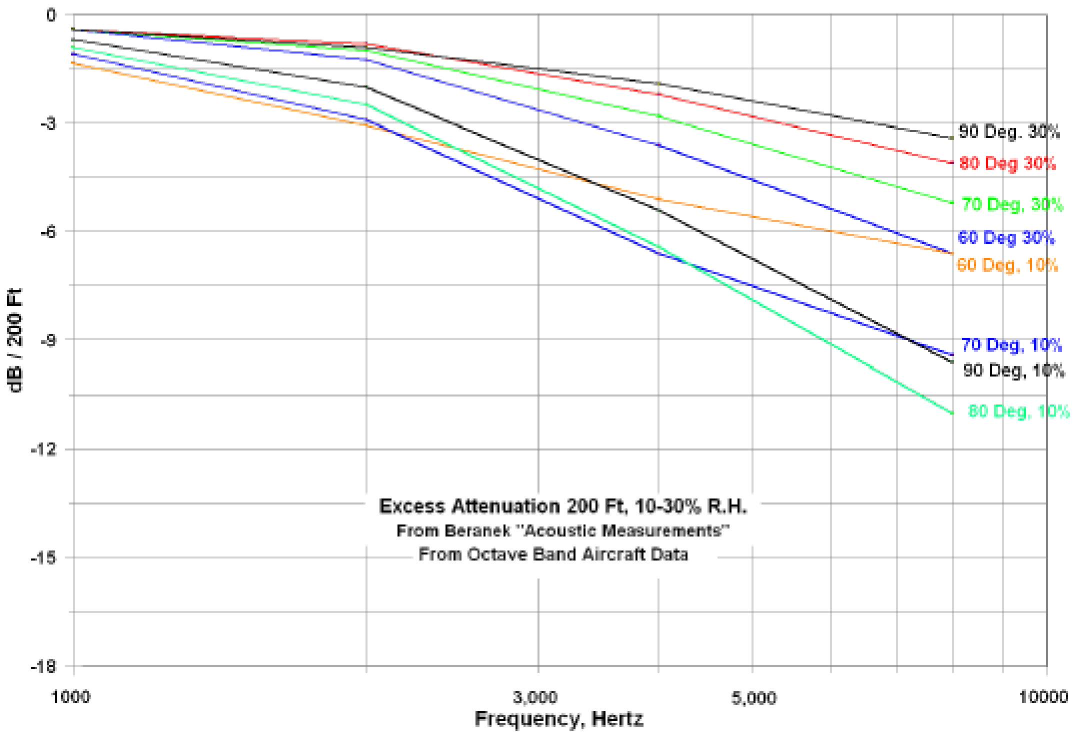

Figure 2

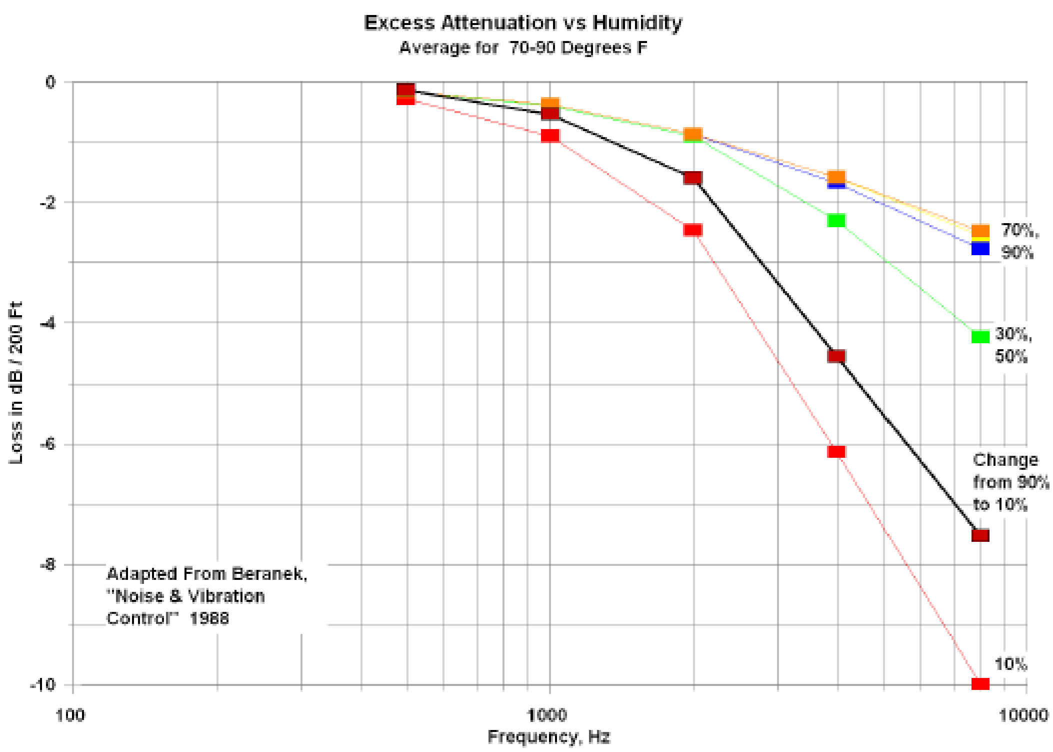

Figure 3

The second major effect of the atmosphere does is refraction -- i.e., change of the direction of the sound waves. This happens because the speed of sound is proportional to the square root of temperature.

Consider a wavefront moving parallel to the surface of the earth, and the air temperature dropping with increasing elevation. The part of the wavefront which is at higher elevations will be moving more slowly than at closer to the earth, and the wavefront will bend upward. Often, however, weather conditions will exist where the air will be coolest at the surface of the earth and be warmer at higher elevations. Under this condition, known as a temperature inversion, the wavefront will be bent downward.

Likewise, the direction of travel is bent by wind gradients. Consider again the sound source radiating parallel to the earth, but into the wind. Since the boundary conditions set wind speed equal to zero at the earth, the top of the wavefront will move more slowly than the bottom of the wavefront, and the wave will bend (refract) downward. Conversely, with the wind at its back, sound will refract upwards.

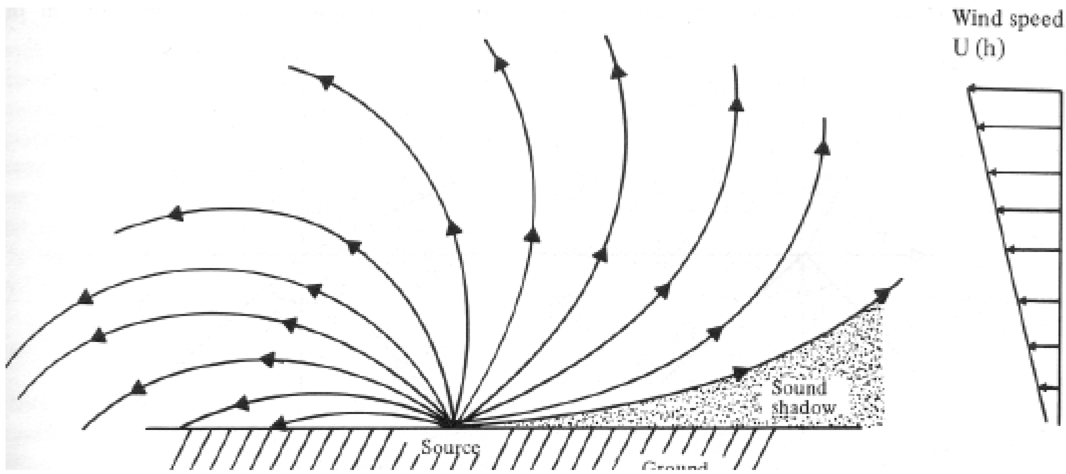

Figure 4

As Beranek so eloquently reminds us, these simplified conditions rarely exist -- the air is, indeed, in constant motion. So on a typical day, the more distant the sound source, the more variable will be the sound in any given seat. On some days the sound will be clear, distinct, and rock solid, while the next day (or even a few moments later, it will become indistinct or vary in both intensity and character.

These effects can be quite powerful. Variations in mid-range and high frequency levels due to refraction can easily exceed 20 dB over paths as short as a few hunded meters. The sound from directional horn loudspeakers aimed at a section of grandstand across the playing field can often be refracted upward so that it is carried over the grandstand, then be refracted back down to earth by differing weather conditions a mile or more away. The fans don't hear the park announcer clearly, but the neighbors in a lakefront highrise do! And in gusty conditions, the sound can literally fade in and out between words, or from one musical measure to another.

There are two possible approaches to this problem. One is to use a large, centrally located loudspeaker system whose directivity can be controlled, changing the aiming of the loudspeaker system to compensate for temperature and wind gradients as they are sensed. Eastern Acoustic Works (EAW) has developed a loudspeaker system optimized for very long throw applications which is a beginning step in this direction. It's control system was designed with this in mind, and although it has limited ability to adjust to existing conditions in this manner/ here is much room for improvement in this area, and the system is probably not suitable for use with atmospheric compensation without very well trained operators. It is, however, a very excellent system.

The second approach to system design is to minimize the distance that sound has to travel in the air to reach listeners. This solution uses a lot of loudspeakers, spread around the stadium, and all relatively close to listeners. The atmosphere is still attenuating and refracting the sound, but because the travel paths are much shorter, the effects are much less severe. And because the sound doesn't have to travel as far, the loudspeakers can be much smaller than those which can produce enough sound to carry longer distances.

So to summarize, these atmospheric effects require large, central loudspeaker systems (often called clusters) to be much larger and more powerful to compensate for loss in the air, the sound quality from them can often be wildly variable, and the sound from them can easily spill excessively into the surrounding neighborhood.

REVERBERATION, REFLECTIONS, AND TIMING

The second major design factor is the interaction of the sound system with the acoustic environment. Enclosed stadiums may have considerable reverberation or the geometry to establish long delayed echoes. Even outdoor stadiums and partially enclosed arenas can have at least some reverberation and surfaces capable of producing very strong echoes. At Wrigley Field, for example, the seating in the upper half of the main grandstand (i.e., far under the upper deck) is relatively reverberant. Even though it is not an enclosed space, there are enough hard surfaces (and enough diffusion) to produce a statistically valid reverberant field in much of that seating.

Reverberation degrades intelligibility. The audience is largely sound absorptive, while most other surfaces in stadiums are not. Thus sound that hits the audience will mostly be absorbed, while sound that hits reflecting surfaces (floors, ceilings, walls behind seating,) will be returned to contribute to a reverberant field. This random sound will generally act as background noise, masking the direct sound. In general, the ratio of direct sound to reverberant sound should be as great as possible. In most reverberant spaces this ratio will be less than unity over most of the audience-- i.e., the direct sound will be lower in level than the reverberant field. But when the ratio drops such that the level of reverberant sound is more than about 6 dB greater than that of the direct sound, intelligibility will be significantly degraded.

Timing is a very critical factor in system design. Many listener seats will typically receive sound from multiple loudspeakers. If the system is to provide good intelligibility, sound from all of these loudspeakers must arrive within a period of about 30 ms. Sound travels at about 1160 ft/second (about 0.8 ms per foot), so acoustic travel times have to be equal within about 35 ft.

Loudspeakers should have enough pattern control (i.e., directivity) throughout their frequency range to limit their coverage to a well defined area, with minimum spill into other seats. Likewise, loudspeaker must be carefully located, and they must be well aimed. Care must be taken that loudspeakers covering an upper deck don't spill down onto main level seating 75 ms later.

If the spill cannot be avoided, it may be necessary to delay main level loudspeakers so that spill from the upper deck is within the 30 ms window required for intelligibility. But such a solution is not without peril, since it increases the total system delay from a live microphone to any given spot in the park where that microphone might be in use, and where a master of ceremonies or singer might be severely distracted by the long delay. More about that later.

Echoes are bad because they degrade intelligibility -- i.e., they make it difficult for the brain to understand speech. (An echo is a reflection which arrives at a listener's ear long enough after the direct sound that a listener's brain has difficulty "processing the data.") Thus, reflections from rear walls become echoes when they hit the wall at an angle close to 90 degrees and hit audience members at a great distance from the wall. Echoes from rear walls can be avoided by tilting walls downward (or upward), or turned into reflections with much shorter delays by placing loudspeakers higher and closer to the walls. The reflections still occur, but if they return to the parts of the audience very close to the reflecting wall so their travel times are within the 30 ms window, they are no longer problematic.

Many modern facilities have concave geometries. Loudspeaker locations must be carefully chosen to avoid reflections in seating areas, especially the strong focused reflections which can come from these curved surfaces. Even a plane surface behind multiple seating areas can produce a long delayed reflection that is strong enough to cause intelligibility problems in some seats (especially seats where direct sound coverage may be marginal).

It is also quite important that sound levels be approximately equal in all seats, but this can sometimes be difficult. The system at Northwestern University's football stadium illustrates this principle. A row of distributed loudspeakers just under the face of the upper deck and directed towards the playing field covers field boxes which are nearly 120 ft away, but these loudspeakers are only 10 ft over the heads of patrons directly below them. And to further complicate things, the loudspeakers had to be physically short so that they would not block sight lines for a half dozen or so rows of seats behind them. This very difficult throw ratio of 22 dB (based on inverse square law) sounds impossible to accommodate, but two interesting techniques were used to successfully do so.

First, a custom loudspeaker was developed using a compact horn which, by virtue of its mouth size, maintains pattern control down to 1 KHz. Then, a vertical bass array of 6-inch drivers was placed in the box next to the horn, spaced so that they produced a downward null due to their half-wavelength spacing at about 500 Hz. This spacing placed the drivers quite close together, and the resulting null is quite broad and only a few dB deep (because the driver is too large to be a point source in that range). But since the drivers are in phase in the far field, they their coherent addition on axis contributes 6 dB of directivity. The combined directivity of the box is on the order of 9 dB down to at least 400 Hz, and higher in the range of the horn.

The second technique that helped here was to use enough loudspeakers that they functioned as a horizontal line array in the far field (i.e., at the most distant boxes nearest the playing field). This meant more coherent addition, particularly in the midrange and below the frequency range of the horns. Measurements on the completed system showed that direct sound levels varied by only 5 dB at 400 Hz from directly below the loudspeakers to the most distant boxes. This particular system was designed for speech and vocal music only.

One of the most important aspects of system design for large spaces is where the sound goes after it hits (or passes) the audience it is intended to cover.

CONTROL OF DYNAMICS

One of the most overlooked aspects of system design is the control of system dynamics. That control falls into four basic categories -- system protection, optimizing system performance, compensating for dynamics of the program material, and keeping up with crowd noise. All four are important, and the manner in which they are addressed are different, but may also overlap.

System protection primarily falls to peak limiters at the input of power amplifiers, which are typically set to prevent over-excursion of the transducers. In many systems, these limiters will rarely operate, or will operate only on the loudest transients on the loudest program material. In systems which have limited headroom, however, these peak limiters can allow average levels to be significantly increased with minimal degradation of sound quality. In other words, they can optimize a smaller system and allow it to perform as if it were much larger. The latter is an example of dynamics control being used to optimize system performance.

The loudness of many park announcers will vary considerably as the game gets more interesting, while others remain calm and collected no matter what happens. It's good to allow some of the extra excitement to get through, but often there's much more variation than is desired. A good integrated dynamics process which combines automatic level control, compression, and peak limiting can be a useful addition to most systems for sports facilities.

The overall loudness of a crowd at a sporting event will vary over a very wide range (greater than 20 dB). If the system is set to sound right when the crowd is quiet, it won't be heard with two out and the bases loaded. And if it's set to be loud enough for an exciting part of the action, it will be blasting the fans out of their seats when the crowd is quiet. Complete compensation is not necessary, nor is it even desirable. Like most dynamics processing, moderation is always in order, and compensation for about one half of the variation is usually a good starting point.

Crowd compensation systems work by sensing the loudness of the crowd, using either a microphone suspended over the crowd or using a loudspeaker as a microphone. Most systems sample the noise during a short time period where there is no amplified sound (i.e., between announcements). This system is called "gap sensing." Others are able to sense continuously, using signal processing to subtract the amplified sound.

All dynamics processing requires careful adjustment while the system is in operation with program material which activates the processing being adjusted, and careful listening throughout the process. The only instrumentation required is accurate indication in real time of the gain reduction provided by each processor, accurate indication of the clipping of system electronics, and a fast, accurate voltmeter (or, even better, a fast real time analyzer) connected across loudspeaker lines.

Adjustment of system dynamics processing takes time, often twice as long as equalization. Once the system adjustments are achieved, they should be locked down so that they cannot be disturbed. Protection, optimization, and crowd noise compensation fall into this category.

Program dynamics control should be treated differently, especially if there will be skilled operators. In general, these adjustments should be optimized during system setup, but left accessible to system operators so that they may adjust to different announcers and different program material.

SKY BOXES AND AMBIENCE

Sky boxes and private suites are often enclosed, and the resulting acoustic isolation from the rest of the crowd can be disconcerting. There are several possible solutions. The most satisfactory is to pick up the sound of the crowd in stereo with a spaced pair of microphones and feed them to the enclosed areas through some sort of stereo or alternating playback system. The same playback system can also carry an electrical feed from the main sound system (carefully delayed to match any direct sound the crowd mics pick up from the sound system), and can be switched to carry audio from a radio or TV broadcast.

It is generally necessary to switch between broadcast audio and crowd ambience rather than mixing the two signals. Broadcast audio of the event will almost always have crowd noise and there is likely to be considerable difference in the delay time between the broadcast's crowd sound and the crowd sound from the ambience mics.

A very successful implementation of ambience in sky boxes is at Northwestern University's Ryan Field, where a stereo pair of PZM mics (spaced at about 30 feet) on the face of the upper deck overlooking the pep band is fed to an array of compact loudspeakers just above the skybox window. PZM mics, properly installed on a boundary, provide very good rear rejection and very natural sound in the forward direction. In this case, the loudspeakers were only about 15 feet away, but were behind the plane of the mics so the leakage is quite manageable. It helps that the sound system is high pass filtered at 250 Hz.

The loudspeakers alternate left and right channels for the entire length of the 120 ft long box, and the system is equalized for flat response between 50 Hz and 15 KHz. Interestingly, no low frequency boost was required to get this system flat on the low end -- although no single loudspeaker had a lot of low end, the combined acoustic output of many (a total of 40 were used in two rows at a spacing of about 6 ft) provided plenty of good sounding low end.

WRIGLEY FIELD -- A DISTRIBUTED SYSTEM

Wrigley Field is one of the last of the "old time" ballparks. Like Boston's Fenway Park, it maintains a level of intimacy, both visually and acoustically, which is rarely achieved in most larger facilities (although the success of newer parks of moderate scale like those in Baltimore and Cleveland has led a trend in that direction). And while Wrigley is one of the smallest parks in terms of seating capacity, it's not a lot smaller -- only about 10% less than most new parks which have opened in the past decade or are currently under construction. And it has more seats than the new park in Pittsburgh and one being planned for Montreal!

Intimacy and charm apparently has its appeal with fans. Over the past five years, attendance at Wrigley Field is 60% greater than at the 14% larger Comiskey Park. Those more intimate parks in Baltimore and Cleveland have also been a hit with fans, consistently outdrawing most larger parks in larger cities.

Blessed with this fine park, Chicago Cubs management has long subscribed to the concept of providing a traditional, low key family-oriented baseball experience to their fans. It was clearly understood that the sound system was to be simply a speech announcement system, and was not intended to excite the crowd with loud music. Loudspeakers should not be so large that they interfere with sight lines.

The existing system was installed in 1986 and expanded to cover newly constructed skyboxes when they were added in 1989. Since then, it has had only routine maintenance.

The system is entirely a distributed one. Four rows of loudspeakers cover the entire grandstand -- one row facing forward and one facing rearward over the main grandstand and field boxes, one row facing forward and one facing rearward over the upper deck. Three large horns located on top of the scoreboard cover the bleacher seating. All three are carefully aimed to minimize their spill to the grandstand, where they could produce long delayed echoes.

The loudspeakers used throughout the grandstand were designed in the 40's, have been a workhorse of the industry, and are still being manufactured today. But we didn't want to use them in 1986, because they are neither as powerful or as full range as we would like for a modern stadium.

When this system was being designed in 1985 for installation in 1986, the author approached a number of loudspeaker manufacturers asking them to develop a newer, improved replacement for the older model. The product needed to be weatherproof, and it needed good efficiency, good power handling, wide (and uniform) angular dispersion, and frequency response good enough for voice and some music. In 1986 (and in 1989 when the system was expanded), new products had not been developed. Finally, in 1998 one manufacturer, Community Light and Sound, responded with a very nice series of wide range, multiway horns designed for use in outdoor distributed systems.

MIXED POINT SOURCE AND DISTRIBUTED SYSTEMS

Real world architecture often dictates complex solutions. The 45,000 seat stadium at Northwestern University, which is situated in a residential neighborhood, is a good example. There is a two level grandstand, the lower level of which can be effectively covered by a distributed system located just below the lip of the upper deck. The upper deck, interrupted by a large section of enclosed suites, requires special treatment. Visitor side bleachers can only be covered by long throw horns on the home side of the field -- there's no other place to put loudspeakers!

Community noise is also a potential for problems -- seating in one end zone is only about 150 feet from a residential apartment building that is just across the street from the stadium. On the other hand, there's a large stretch of University-owned land behind the visiting side bleachers to act as a buffer for sound leaking to neighboring houses. This is far from an ideal buffer -- as noted, atmospheric conditions can easily carry sound over the bleachers and into the neighborhood -- but it was the best solution available at the time.

Two large stair towers flank the main grandstand, and the one nearest the end zone seating was used to house a point source system to cover the end zone and visiting side bleachers. The choice was driven purely by geometry -- the horn covering the end zone could be directed to minimize spill to the apartment building behind the end zone, and leakage from the long throw horns covering the visiting side bleachers would mostly spill into the University's parking lots to the north and east of the bleachers. One point source was used rather than two, since the delay differences between horns mounted on both stair towers would have resulted in serious echoes in a large number of bleacher seats.

The asymmetrical location of this point source greatly complicated the design of the distributed systems, however, because the long throw horns put considerable sound in many lower grandstand field boxes (i.e,, the first 10-20 rows of seats from the field). This made it necessary to delay the distributed system to keep it in time with the long throw system.

Another potential problem is that the distributed system on the lip of the upper deck facing the field might put enough sound in the visiting bleachers to result in a destructive echo. The fact that distributed systems acts as a line array in the far field, with levels falling off at 3 dB/doubling of distance increases the likelihood of that occuring. A careful study of the EASE model's prediction of energy-time curves (ETC's) in the bleachers ensued. It showed that the multiple steps of delay being applied to the distributed loudspeakers resulted in their spill to the bleachers being a diffuse set of arrivals rather than a single, well defined echo, and a judgement was made that it would not be a problem. The completed system confirmed the correctness of that judgement.

CONCLUSIONS

The successful design of sound systems for baseball and other sports facilities demands a thorough understanding of many phases of acoustics, and draws on a wide range of techniques for problem solving. Each ballpark brings with it some of the same challenges, and often its share of new ones. And each facility has its own personality, imposed upon it by a combination of its architecture and its management. The wise designer will be sensitive to these factors, and tailor his or her design to them, realizing that the sport is the star of this game, not the sound system.

REFERENCES

"Acoustical Measurements, Revised Edition" by Leo Beranek, American Institute of Physics, 1988

Acoustical Designing in Architecture, Revised Edition" by Vern O. Knudsen and Cyril M. Harris, American Institute of Physics, 1978

"Noise and Vibration Control, Revised Edition," Edited by Leo Beranek, Institute of Noise Control Engineering, 1988

"Physics of the Air, Third Edition" by W. J. Humphreys, Dover Press, 1964

"Noise and Noise Control, Volume 1," by Malcolm J. Crocker and A. John Price, CRC Press, 1975