Sound Systems in Three Reverberant Ice Arenas at the1988 Winter Olympics

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Among the professional sound systems used for the 1988 Winter Olympics were designs created for three highly reverberant indoor ice arenas. The systems were configured with medium-Q speakers in hybrid central-cluster and delayed-distributed speaker layout strategies. Advanced computer simulation programs were used to predict the performance of the systems in advance, both with respect to acoustics and array architecture. Room descriptions, design approach, performance predictions, array design, adjustment procedures, and measurement techniques are presented.

O. INTRODUCTION

For the first time, a single official supplier was chosen to provide the professional sound systems for the Olympics. The Olympic games in Calgary, Canada, gave the supplier an opportunity to devise a unified approach to systems installed in nearly twenty venues. The goal in every case was to provide excellent speech intelligibility and, where applicable, high fidelity music reproduction in every seat. In three indoor ice arenas, reverberation times equaled to or exceeded four seconds, making the acoustic conditions difficult for speech and music.

An approach was chosen which used only all-cone wide dispersion (medium-Q) speakers instead of clusters of narrow dispersion (high-Q) constant directivity horns. In this approach, supplementary speakers were distributed and delayed whenever necessary to provide an adequate ratio of early-to-late sound energy at the listener, a ratio known to correlate with speech intelligibility and music clarity [1-4, 17]. A purely distributed system was not possible since localization of music to the center of the arenas was an absolute requirement for figure skating competitions. Furthermore, a sound source localized to the action was preferable for the audience.

Systems were designed and communicated to contractors using commercially available computer programs. One program [5] was used to predict acoustic performance and is capable of advanced simulation of sound systems in rooms. The program uses the three dimensional dispersion characteristics, physical orientation, power distribution, and electrical time delay of speakers, the specific type and distribution of absorptive materials, and an arbitrarily complex planar room model to predict the direct as well as reflected sound energy. The other program [6] allows the designer to accurately simulate the three dimensional characteristic of speakers and speaker arrays, and the physical environments in which they are to be placed. The program was used extensively to create architecturally accurate drawings of speaker arrays, and was extremely useful in communicating the designs to installers and other contractors.

1. THE VENUES

The three venues varied in size seating capacity, and function. Table-1 shows each room's basic characteristics. In each room, the only significant absorption provided was by the audience; other surfaces were very reflective.

Table-1
Basic Room Characteristics

Room	Volume		T_{60}	Seats	Event(s)
Father David Bauer	1,510,000 (ft^3)	43,000 (m^3)	4.0	1700	Hockey
Max Bell Arena	1,312,000	37,000	4.5	3000	Curling, Speed Skating
The Stampede Corral	2,950,000	85,000	5.5	6000	Hockey, Figure Skating

Reverberation time measurements were taken using a starter's pistol in empty rooms. Recordings were made and later fed to a digital signal analyzer. These quasi impulse responses were squared, reverse integrated, and converted to decibels in order to display a classic room decay curve. A best straight-line fit was made to the resulting curve. Thus the reverberation times quoted here are worst case; the reverberation time would decrease in the presence of an audience. However, the sound systems were required to operate under these conditions, both during practice sessions and sparsely attended events.

2. DESIGN PHILOSOPHY

The single most important design goal was to provide intelligible speech. To accomplish this, the factors known to affect speech intelligibility were controlled including distortion, frequency response, sound pressure level, ambient noise, and the ratio of early-to-late sound energy arriving at the listeners. Of these, the early-to-late ratio is the most critical since it is so difficult to control after the speakers are installed, and because a high ratio is difficult to achieve in excessively reverberant rooms. It is also the most difficult to predict. The formula defining the ratio of early sound to late sound used here is shown in equation-1. This ratio shall be denoted *dB-S/N*. Fig.-10.1 shows a graph of speech intelligibility versus the early-to-late ratio for ten rooms used in previous intelligibility experiments [7], [8]

$$\text{dB-S/N} = 10 \, \text{Log} \frac{\int_{t=0}^{80ms} p^2(t)\, dt}{\int_{t=80ms}^{\infty} p^2(t)\, dt} \qquad (1)$$

Where, dB-S/N is the early-to-late ratio
p(t) is the impulse response

The sound system simulation computer program was used to indicate an adequate early-to-late ratio. In addition, a formula may be used to predict the ratio based upon simple statistical acoustics parameters (equation-2). The formula is derived by assuming diffuse conditions in the room, and a theoretical relationship between critical distance, speaker directivity, room volume, and reverberation time. (Note: the formula assumes that any time delay speakers have been set.)

$$\text{dB-S/N} = 10\,\text{Log}\left\{e^{kb}\left[\left(\frac{d_c}{d}\right)^2 + 1\right] -1\right\} \qquad (2)$$

Where, dB-S/N is the early-to-late ratio
k is a constant = [0.080 / 10log(e)] = 0.01842
b is the decay rate = 60 / T_{60}
d_c is the critical distance of the closest speech loudspeaker
d is the distance from listener to closest speech loudspeaker

Other factors were taken into account such as level, coverage, and physical considerations. Overall sound pressure level at the listeners was predicted by calculating the sum of direct and reverberant energy. Direct field was mapped onto the audience using the simulation program, and steady state reverberant field was predicted using a modification of a classical formula to account for the specific distribution of absorption, orientation of the sound sources, and geometry of the room [5]. Sound intensity coverage was displayed graphically by using a distribution curve and its corresponding statistical parameters such as standard deviation, maximum deviation, and mean.

In each system, the decision was made to use separate equalization for music and speech. The primary reason was to allow for enhancement of intelligibility by adjustment of speech frequencies, and elimination of low frequency masking due to the proximity effect of microphones.

3. THE SYSTEMS

The basic systems created for the three venues are shown in Fig.-10.2. In all cases, only three different speakers were used: one bass speaker and two array speakers. The bass speaker uses a unique waveguide enclosure to reproduce frequencies from 25 Hz to 125 Hz. Tube-like geometry and all-plastic construction make this system extremely lightweight (60 lbs). The two array speakers contain four and eight identical 4.5" (11.4 cm) drivers respectively. The four driver array operates from 90 Hz to 16 kHz. The eight driver array speaker operates from 125 Hz to 16 kHz when used with the bass speaker and 55 Hz to 16 kHz when used alone.

3.1 Father David Bauer Arena

The system created for the Father David Bauer Arena consisted of four eight driver arrays hung below the scoreboard over center ice, and sixteen four driver arrays distributed and delayed around the perimeter of the ice surface and aimed at the bleacher seats. Fig.-10.2.1 shows the floor plan with cluster locations and system spreadsheet.

3.2. Max Bell Arena

The system installed in the Max Bell Arena consisted of four stacks of two eight-driver arrays, and four bass systems all flown in a single cluster above center ice. Sound to the end-of-arena bleachers was provided by eight four-driver speakers distributed and delayed in time. Fig.-10.2.2 shows the floor plan with cluster locations and system spreadsheet and Fig.-10.3.1 shows an architectural view of the center-ice cluster.

3.3. The Stampede Corral

The system created for the Stampede Corral consisted of a large central cluster, and distributed/delayed speakers providing sound to the end-of-arena bleachers. The central cluster consists of two columns of four eight-driver speakers aimed at the two ends of the arena, two sets of two stacks of eight-driver speakers aimed at the sides of the arena, and two single eight-driver speakers aimed straight down at the ice. Fig.-10.2.3 shows the floor plan with cluster locations and system spreadsheet. Fig.-10.3.2 shows an architectural view of the center-ice cluster.

4 PREDICTED RESULTS

Performance prediction for the three venues has been condensed for clarity and includes: 1) early-to-late ratio as predicted by equation-2, presented as the maximum allowable distance from the nearest speech speaker to the listener, 2) maximum level, calculated by summing the mean direct field to the predicted steady state reverberant field, and 3) coverage mapped on the audience area and distilled to a distribution curve. Items 1 and 2 are presented in Table-2, and item 3 in Fig.-10.4.

Table-2
Showing: a) maximum allowable distance between listeners and nearest speech speaker to ensure an early-to-late ratio greater than 0-dB (very good intelligibility) and b) maximum sound pressure level (average) in room, calculated as the sum of direct and reverberant field. (Note: peaks approximately 6 dB higher.)

Room	D_{max}		$dB\text{-}SPL_{max}$
Father David Bauer Arena	74 (ft)	23 (m)	109
Max Bell Arena	61	19	109
Stampede Corral	79	25	107

5 SYSTEM ADJUSTMENTS AND MEASUREMENTS

Because of tight security and continuous use of the venues, the sound systems had to be installed, wired, and adjusted quickly and in the late evening and early morning hours. Therefore, a simple step-by-step procedure was developed and implemented in an effort to optimize the system performance in very little time. Unfortunately, much of the measurement data which would illuminate this presentation could not be taken in the time allowed.

5.1 Equalization

Two target listener power response curves were used: one for music and the other for speech. Fig.-10.5 shows the two curves. These curves were realized by equalizing to the target curve at a representative listener location with pink noise injected into the system, and measuring on a one-third octave spectrum analyzer. The stability of this curve was checked in another representative listener location and a compromise generated if the discrepancy was significant. While some deviations were found from position to position, they were not considered serious as judged by program material listening tests.

5.2 Music Listening Tests

After adjusting the music equalizer to result in the listener power response curve of Fig.-10.5, listening tests were performed. One compact disc selection in four categories was chosen in advance and was used subsequently in all venues. The categories were: male vocal, female vocal, classical string instruments, and popular music. Each selection was judged by a group of three in each of the two listening positions where spectrum measurements were made. If an objection was expressed by two or more listeners, the spectrum was adjusted. This process was repeated as many times as necessary. The next step was to walk around the entire listening area and listen for any other spectral anomalies. If any were found (this was rare) the analyzer was again employed in that area in an attempt to quantify the objection. Adjustments to equalization were made in only one venue as a result of this secondary listening. The conclusion was that if coverage was good, then equalization in one representative location was sufficient for the entire arena.

5.3 Intelligibility Tests

After adjusting the speech equalizer to result in the listener power response curve of Fig.-10.5, intelligibility tests were performed. An abbreviated version of the USA Standard Method for Measurement of Monosyllabic Word Intelligibility [10] was conducted in a variety of seats. Briefly, the method consists of reading monosyllabic English words embedded in a carrier sentence over the system while subjects write down the words. In all cases, scores of 90% or better (considered very good to excellent intelligibility) were predetermined to be the acceptable range.

In the tests conducted in the three Olympic venues, the words were read over a system microphone by one of the three adjusters. The other two adjusters took the test. It should be noted that these tests were not conducted rigorously because of time constraints. Instead, the entire audience area was first scanned by the two listeners for any suspect regions, and then these regions were investigated in detail. In the Stampede Corral, intelligibility was found at first to be unacceptable at the far ends of the arena, representing about 10% of the seats. Excessive reverberation was responsible, and no amount of equalization made the intelligibility scores improve appreciably. Delay speakers were added and had a profoundly positive effect on these regions. In the Max Bell Arena, an unanticipated development degraded intelligibility in the end-of-arena seats in the form of absorptive banners hung from the rafters after the initial system was installed. Again the solution was to add delay speakers.

Because comprehensive measurements were not possible, and because intelligibility testing could not be conducted under actual event conditions, a compromise was proposed. A technician was given an all-area access and was allowed to make binaural recording during actual Olympic events. The recordings were made using in-the-ear binaural microphones and a small professional-grade cassette recorder. Thus a compact system was used to make recordings in a variety of locations, and in a variety of ambient noise environments [11].

5.4 Maximum Level Measurements

As a measure of maximum undistorted level without feedback, a rather simple technique was used. The criteria set for all three systems were speech peaks of 110 dB-SPL without distortion or feedback. These targets were met with some remaining headroom in all cases.

5.5 Coverage Measurements

Pink noise was injected into the system upon completion of all other adjustments and measurements. A sound pressure level meter set on A-weighting and slow metering was used to give a rough indication of coverage. (Note that this measure was not intended as an exact verification of coverage predicted by the computer program since a broadband signal was used to excite the room whereas the computer program predicts coverage in a user-defined octave band only, usually 1 or 2 kHz.) The measurements should agree in general to the predictions. These measurements are shown in Fig.-10.6.

6 DISCUSSION

With regard to music reproduction, the comments of the adjusters (including this author) and others may be summarized as; " Well, it's the best music system in a hockey arena I've ever heard, but I wouldn't want my stereo at home to sound like this." When asked to be more specific, the most important objection was that the bass was undefined, or muddy. Vocal clarity and overall balance was very good. The bass problem is difficult to solve, because of very slow sound decay at low frequencies, and because the bass speakers are nominally omnidirectional and were located far from the listeners.

In the somewhat artificial conditions of empty arenas late at night, intelligibility was very good to excellent in every area of each of the three venues. However, the binaural recordings made during actual Olympic events were far more revealing to the same people who adjusted the systems. Overall, the three primary adjusters were delighted at the results. In the vast majority of situations the speech was not only intelligible but natural sounding. One engineer who heard the tapes said that if he ever met the announcer he had just heard, he would know who she was! However, some problems are audible on the tapes, dominated by cases where the system was not loud enough, either because the announcer was not speaking loudly enough, or because the audience ambient noise level increased.

Gain before feedback was more than adequate when the systems were adjusted, but on one of the binaural recordings, at one point in the program, a slight ringing can be heard. This apparent reduction in the system's gain before feedback could be explained by one of many factors, including multiple open microphones, excessive talker-to-microphone distance, a defective microphone, an untried orientation of a microphone and so on.

7 CONCLUSION

The systems designed, installed, adjusted and operated in these venues proved to the audio professionals involved that wide dispersion (medium-Q) loudspeakers could be used as basic building blocks to create superb and intelligible sound even in highly reverberant rooms. In the area of speech intelligibility, these systems offered dramatic evidence that the early-to-late ratio is an accurate predictor of intelligibility.

8 ACKNOWLEDGMENTS

The author gratefully acknowledges Bose Limited, who managed the entire professional sound system effort at the Games, Tom Clelland and his staff who installed the systems, Bruce Myers who was the principal designer of the systems, and Tom Birkle, who is the author of the software programs, and who was one of the three adjusters.

9 REFERENCES AND FOOTNOTES

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[2] M. Barron, "The Subjective Effects of First Reflections in Concert Halls -- The Need for Lateral Reflections," *J. Sound Vibration*, vol. 15, pp 475-493 (1971).

[3] T. Houtgast, H.J.M. Steeneken and R. Plomp, "Predicting Speech Intelligibility in Rooms from the Modulation Transfer Function I. General Room Acoustics," *Acustica*, vol. 46, No. 1, pp 60-72 (1980). See in particular page 65.

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[5] T. K. Birkle, and K. D. Jacob, "The Latest Advances in Predicting sound System Performance in Real Spaces: Combining Intuitive User Interface with Acoustically Relevant Output," presented at the Audio Engineering Society 6th International Conference on Sound Reinforcement, Session 4, Nashville, Tennessee, May 5-8, 1988. The name of the computer program is Modeler® Design Program.

[6] The program title is SpeakerCAD™ Graphics Program.

[7] K. Jacob, "Subjective and Predictive Measures of Speech Intelligibility -- The Role of Loudspeaker Directivity," *Journal of the Audio Engineering Society*, vol. 33, No. 12, pp. 950-955, December 1985.

[8] K. Jacob, "Development of a New Algorithm for Predicting the Speech Intelligibility of Sound Systems," Presented at the 83rd Convention of the Audio Engineering Society, New York, 1987.

[10] ANSI #S3.2-1960 (R1971), "USA Standard Method for Measurement of Monosyllabic Word Intelligibility."

[11] Edited cassette copies of these binaural recordings are available from the author on request.

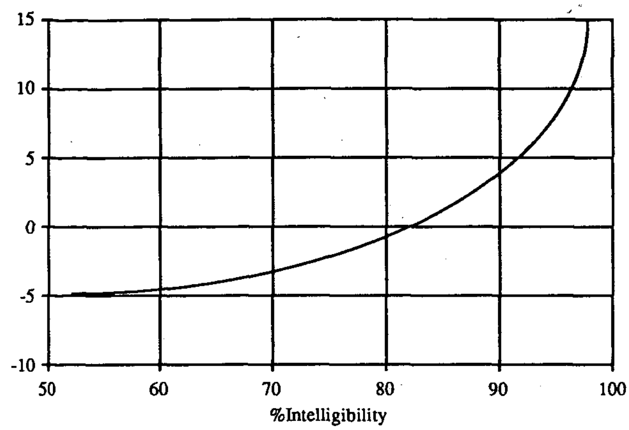
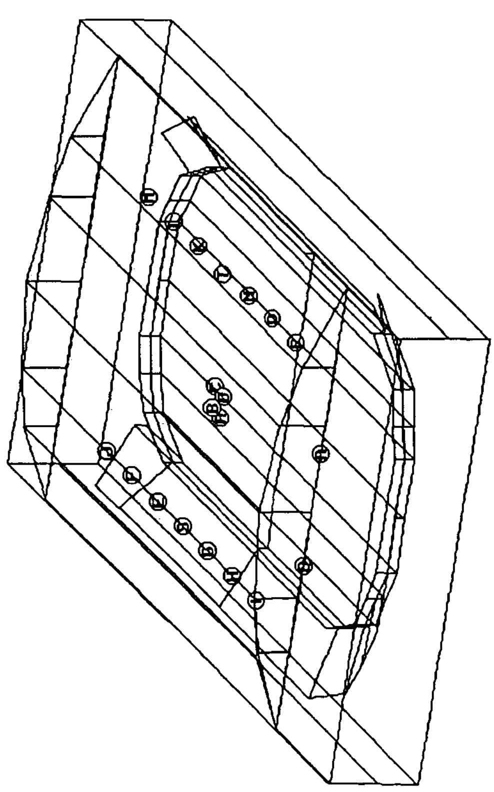
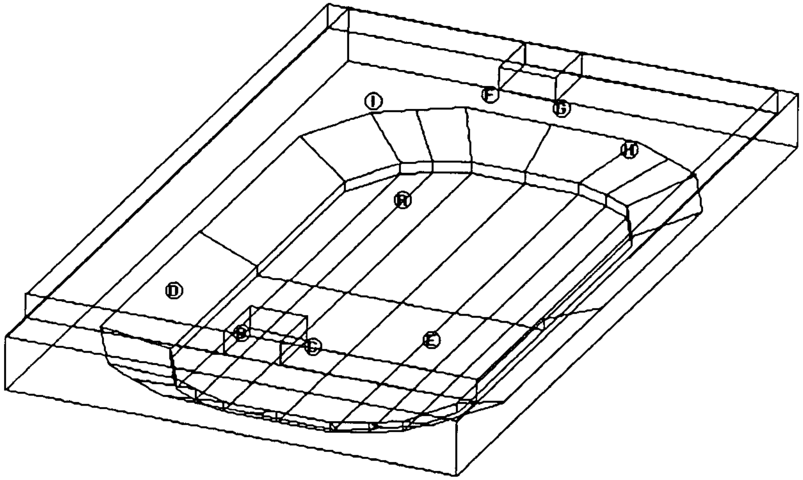


Fig.-1 Best curve fit of mearured early-to-late ratio vs. %intelligibility from two experiments [7, 8]. Notice that no improvement in intelligibility occurs for S/N ratios above 5 dB and that very good to excellent intelligibility results for S/N ratios greater than 2 dB-S/N.



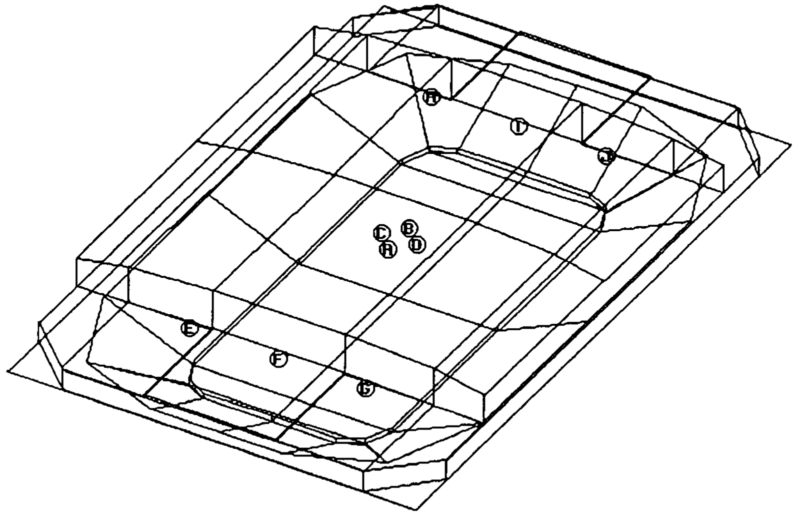
Speaker	Cluster	Height	Roll°	Pitch°	Yaw°	Power(W)	Time(mS)
802	A	25.0	0.0	20.0	150.0	240.0	0
802	B	25.0	0.0	20.0	30.0	240.0	0
802	C	25.0	0.0	20.0	-30.0	240.0	0
802	D	25.0	0.0	20.0	-150.0	240.0	0
402	E	30.0	0.0	50.0	80.0	80.0	45
402	F	30.0	0.0	50.0	60.0	100.0	70
402	G	30.0	0.0	50.0	100.0	80.0	45
402	H	30.0	0.0	50.0	120.0	80.0	70
402	I	30.0	0.0	50.0	135.0	100.0	70
402	J	30.0	0.0	50.0	45.0	125.0	70
402	K	30.0	0.0	50.0	-80.0	80.0	45
402	L	30.0	0.0	50.0	-60.0	100.0	70
402	M	30.0	0.0	50.0	-45.0	125.0	70
402	N	30.0	0.0	50.0	-100.0	80.0	50
402	O	30.0	0.0	50.0	-120.0	81.3	70
402	P	30.0	0.0	50.0	-135.0	128.8	70
402	Q	30.0	0.0	50.0	180.0	102.3	100
402	R	30.0	0.0	50.0	180.0	102.3	100
402	S	30.0	0.0	50.0	90.0	81.3	45
402	T	30.0	0.0	50.0	-90.0	81.3	45

Fig.-10.2.1 Wire frame model of Father David Bauer Arena showing cluster locations (top), and system spreadsheet showing cluster height, speaker types, orientations, power, and delay (bottom).



Speaker	Cluster	Height	Roll°	Pitch°	Yaw°	Power(W)	Time(mS)
2 X 802	A	35.0	0.0	18.0	145.0	251.2	0
2 X 802	A	35.0	0.0	18.0	-145.0	251.2	0
2 X 802	A	35.0	0.0	18.0	35.0	251.2	0
2 X 802	A	35.0	0.0	18.0	-35.0	251.2	0
402	B	28.0	0.0	30.0	140.0	125.9	90
402	C	28.0	0.0	30.0	-140.0	125.9	90
402	D	28.0	0.0	30.0	158.0	100.0	90
402	E	28.0	0.0	30.0	-158.0	100.0	90
402	F	28.0	0.0	30.0	40.0	125.9	90
402	G	28.0	0.0	30.0	-40.0	125.9	90
402	H	28.0	0.0	30.0	-22.0	100.0	90
402	I	28.0	0.0	30.0	22.0	100.0	90

Fig.-10.2.2 Wire frame model of Max Bell Arena showing cluster locations (top), and system spreadsheet showing cluster height, speaker types, orientations, power, and delay (bottom). (Note: 4-bass systems not shown.)



Speaker	Cluster	Height	Roll°	Pitch°	Yaw°	Power(W)	Time(mS)
4 X 802	A	34.0	0.0	10.0	-180.0	120.2	0
4 X 802	B	34.0	0.0	10.0	0.0	120.2	0
2 X 802	C	34.0	0.0	15.0	135.0	120.2	0
2 X 802	C	34.0	0.0	15.0	45.0	120.2	0
802	C	34.0	0.0	90.0	90.0	25.1	0
2 X 802	D	34.0	0.0	15.0	-135.0	120.2	0
2 X 802	D	34.0	0 0	15.0	-45.0	120.2	0
802	D	34.0	0.0	90.0	90.0	25.1	0
402	E	32.0	0.0	15.0	135.0	120.2	150
402	F	32.0	0.0	15.0	180.0	120.2	150
402	G	32.0	0.0	15.0	-135.0	120.2	150
402	H	32.0	0.0	15.0	45.0	120.2	150
402	I	32.0	0.0	15.0	0.0	120.2	150
402	J	32.0	0.0	15.0	-45.0	120.2	150

Fig.-10.2.3 Wire frame model of The Stampede Corral showing cluster locations (top), and system spreadsheet showing cluster height, speaker types, orientations, power, and delay (bottom). (Note: 4-bass systems not shown.)

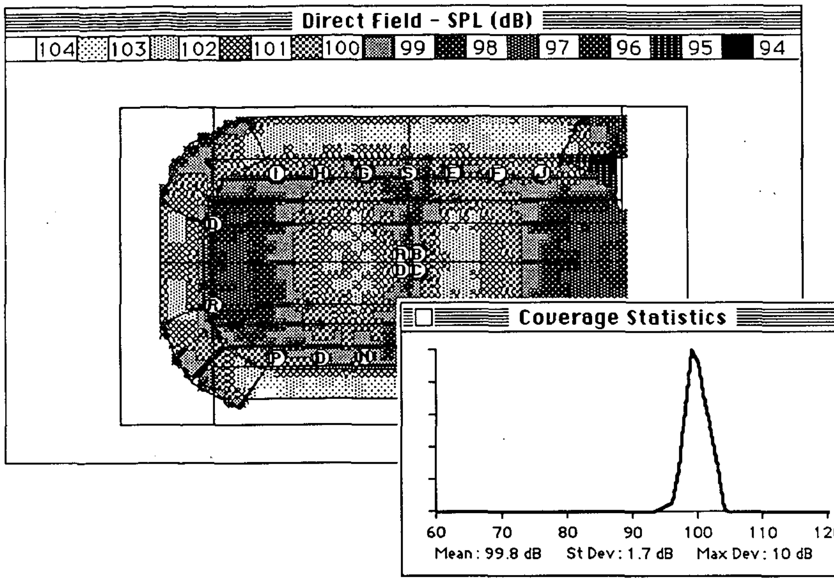


Fig.-10.4.1 Direct field sound pressure level map and coverage distribution curve for the Father David Bauer Arena sound system. Horizontal axis of curve is SPL, and vertical axis is frequency of occurrance. A standard deviation of less than 3 dB is considered very good coverage.

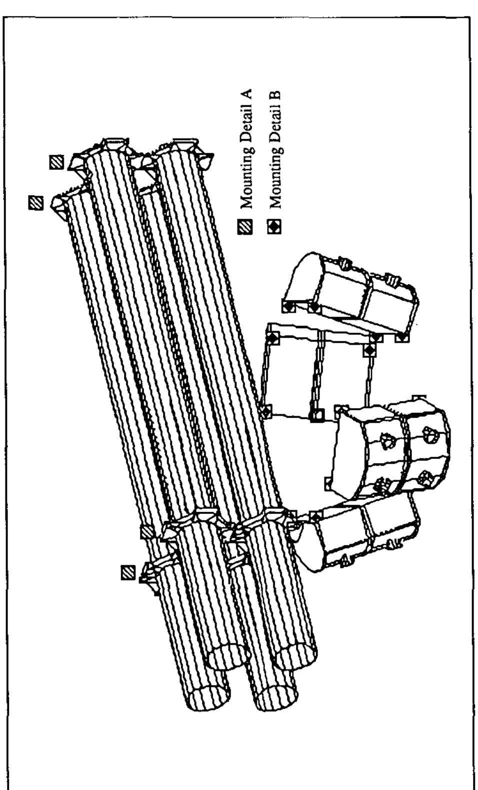


Fig.-10.3.1 Oblique view of above the ice cluster in the Max Bell Arena (scale: 0.5" = 1').

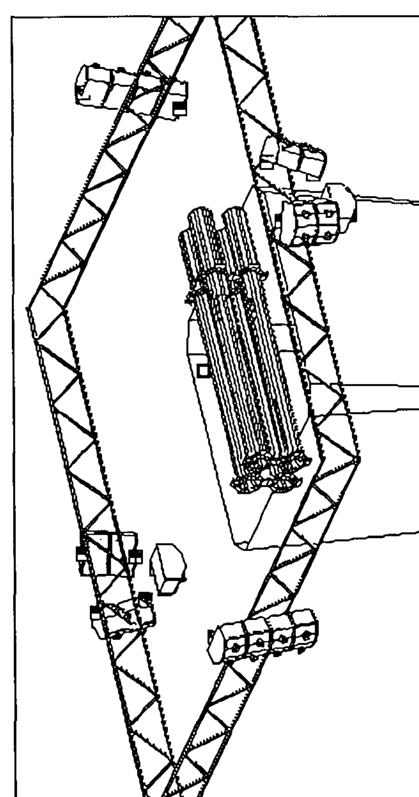


Fig.-10.3.2 Oblique view of above the ice cluster in The Stampede Corral (scale: 0.25" = 1').

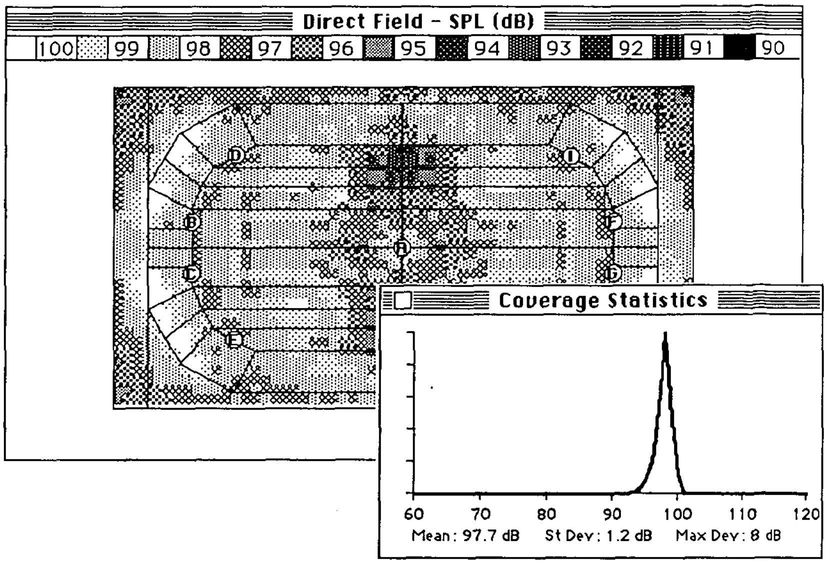


Fig.-10.4.2 Direct field sound pressure level map and coverage distribution curve for the Max Bell Arena sound system. Horizontal axis of curve is SPL, and vertical axis is frequency of occurrance. A standard deviation of less than 3 dB is considered very good coverage.

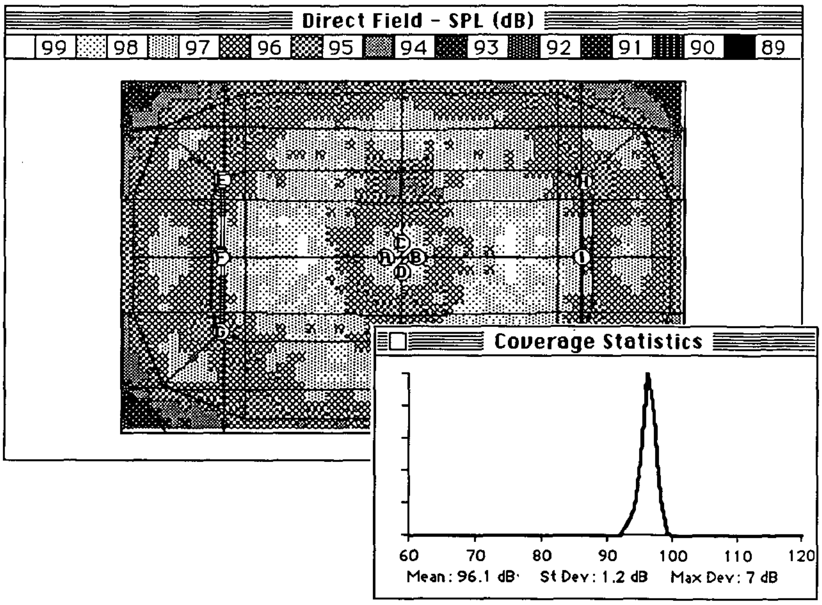


Fig.-10.4.3 Direct field sound pressure level map and coverage distribution curve for the Stampede Corral sound system. Horizontal axis of curve is SPL, and vertical axis is frequency of occurrance. A standard deviation of less than 3 dB is considered very good coverage.

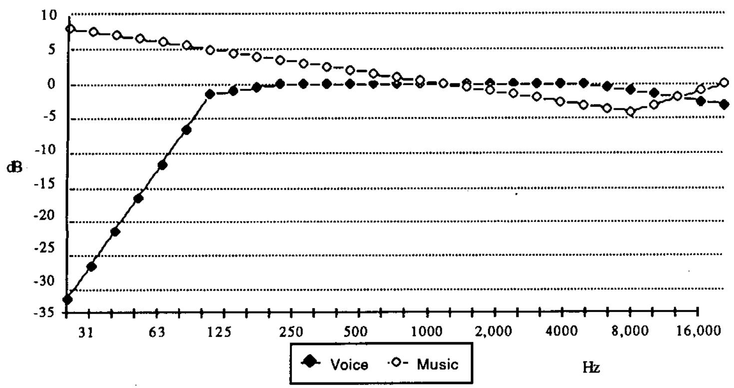


Fig.-10.5 Target listener response spectra for voice and music. Curves correspond to curve as measured on a one-third octave analyzer at representative listener locations.

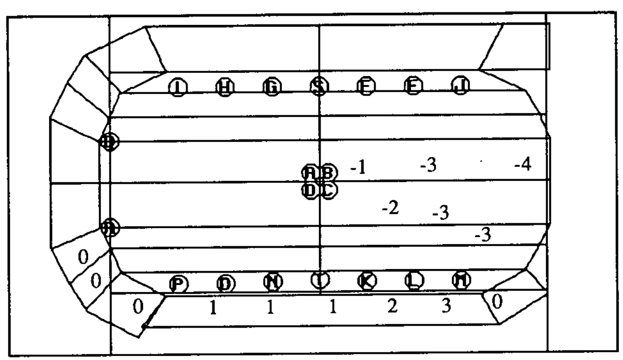


Fig.-10.6.1 Actual coverage measurements from the Father David Bauer Arena. Numbers correspond to the difference between reading taken on an SPL meter and the average reading. Measurements were made using an SPL meter set on A-weighting and slow metering, with pink noise injected into the system.

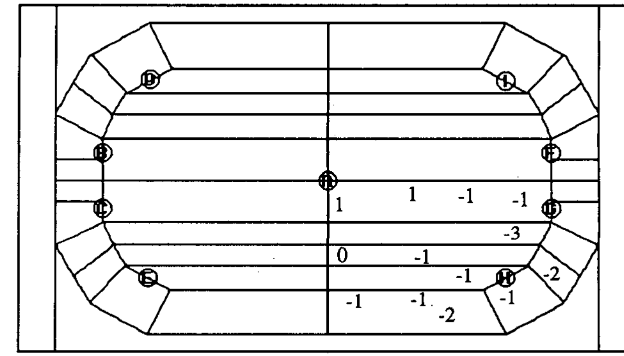


Fig.-10.6.2 Actual coverage measurements from Max Bell Arena. Numbers correspond to the difference between reading taken on an SPL meter and the average reading. Measurements were made using an SPL meter set on A-weighting and slow metering, with pink noise injected. into the system.

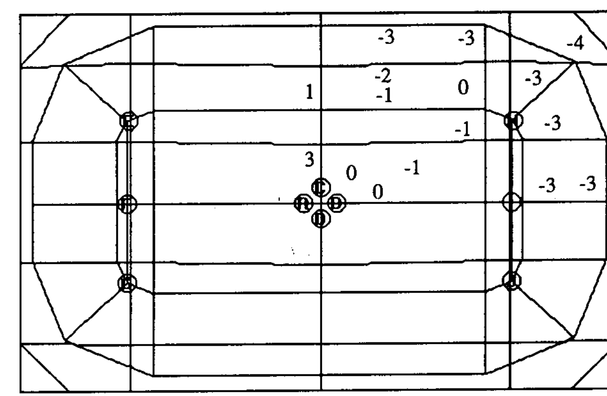


Fig.-10.6.3 Actual coverage measurements from the Stampede Corral. Numbers correspond to the difference between reading taken on an SPL meter and the average reading. Measurements were made using an SPL meter set on A-weighting and slow metering, with pink noise injected into the system.