WHITE PAPER: THE JBL PROJECT EVEREST CONSTANT IMAGING SYSTEM

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Introduction:

Digital recording has refocused interest on microphone techniques in the studio, and today's critical listener is more aware than ever before of such qualities in a recording as imaging, depth, and space perspectives. These terms are largely subjective and probably cannot be precisely defined. However, they do refer to certain attributes in a recording which generally have to do with the subtleties of microphone placement in a natural acoustical environment.

The stability of the "stereo stage" has long been a concern. Almost all well designed loudspeakers can provide stable stereo images for a listener located precisely on the plane of symmetry. But even slight movements off that plane may result in a significant shifting of those phantom images.

Some loudspeakers which have a wide radiation pattern tend to produce a rather "spacey" sound, inasmuch as they may bring into play considerable local room reflections. Many listeners at non-center positions mistake this for cues in the recording itself, and such loudspeakers often give the impression of richness in ambience, but of course without precision of localization.

The chief design aim in this Project Everest system has been to widen the area over which the listener can perceive accurate stereo imaging. This has been accomplished through precise shaping of the system's polar response, largely by means of a unique defineded coverage high-frequency horn design.

Another design aim has been to provide wide dynamic range so that the high crest factors in modern digital recordings can be reproduced with low distortion. The system has a sensitivity of 100 dB SPL, referred to one watt at one meter, and this allows amplifiers of only moderate power ratings to drive the system to very high acoustical levels. Rated system program power input is a generous 300 watts, so each channel can produce peak acoustical levels of 115 dB SPL at a distance of three meters!

In this White Paper, we will discuss the various design aspects of the system, with particular emphasis on its unique polar characteristics.

Generation of Phantom Images:

In the frequency range below about 700 Hz, phase relationships at the ears are used to localize sound sources in the horizontal plane. Above about 2 kHz, the predominant cues are the amplitude relationships at the ears caused by shadowing of sound as it passes around the head.

Figure 1 shows these relationships. Here, we have combined both phase and amplitude cues in generating a set of phasors at the ears. The phasors are shown as arrows; their length indicates relative amplitude, and their angular orientation indicates a leading or lagging relationship between them. (Counterclockwise rotation indicates a leading relationship.)

For a sound source (S1) placed directly ahead of the listener, sounds arrive at the ears simultaneously with the same amplitude. Thus, the phasors are of equal length, and the phase angle between them is zero. For a source (S2) to the right, the phasor at the right ear is longer (indicating greater amplitude), and it leads the phasor at the left ear by some amount (indicating that sound reaches the right ear first).

For a listener seated along the median plane between a pair of stereo loudspeakers, as shown at Figure 2, we can create precise phasor relationships at the listener's ears for a center phantom image. In this case, the information fed to each loudspeaker is the same, simulating an image panned to the center position. While the individual phasors at each ear are different for the two loudspeakers, the summations of them, LT and RT, at each ear are identical. Thus, the listener will localize the sound as originating directly in front of him.

We can alter the amplitude relation between the loudspeakers, panning the signal slightly to the right. Again, when we reconstruct the phasors at the ears, we find a leading phasor at the right ear, indicating a phantom source displaced to the right of center. This is shown in Figure 3.

As long as the listener is exactly midway between the loudspeakers, the analysis is simple enough, and the phantom images are quite precise.

Off-Center Phantom Images:

For listeners displaced from the center, the phasor analysis becomes rather complicated. As shown in Figure 4, there will be different path lengths to each ear, and those differences will result in peaks and dips in frequency response (often referred to as"comb filter" response).

Furthermore, the pattern existing at each ear will be different, and the resulting phasor relationships between the ears will be quite complicated.

The situation shown in Figure 4 applies mainly to frequencies below about 700 Hz, where amplitude differences between the ears are not significant. The resulting localization cues in this frequency range are quite ambiguous, and little can be done to offset them.

At higher frequencies, the comb filtering at the ears becomes quite dense and less a factor in determining localization. In this frequency region, we take note of the fact that the nearer loudspeaker will be louder than the farther one, and this is something we can correct through the loudspeaker's directional characteristics.

Simply correcting the distance offset level in dB between the loudspeakers and the listener by altering the polar patters is not sufficient. Another factor, the precedence effect, comes into play. Transient (high frequency) information is localized at the earlier loudspeaker, even when the two loudspeakers are reproducing the sound at the same level at the listener. Thus, we designed into the system a degree of "over-correction" to counteract this. Figure 7 shows the general characteristics of the precedence effect in terms of level imbalance required to restore phantom center localization for a delay between loudspeakers. This data is derived from the studies of Haas, and there is a good bit of variability to it, depending on details of the listening atest setup and the precise nature of the program. In our studies, we found that approaching the Haas values of level imbalance, as a function of delay difference, was most useful above 4 kHz. This untimately resulted in our angling the UHF transducer sixty degrees from the normal straight-ahead position. Extensive listening tests in a variety of environments confirmed the validity of this.

Polar Patterns of the Project Everest DD 55000 Constant Imaging System:

In this system, all three drivers are angled inward, as shown in Figure 5. The low-frequency (LF) and high-frequency (HF) components are angled at 30 degrees relative to straight ahead, while the ultra-high-frequency (UHF) unit, as we have stated earlier, is angled at 60 degrees.

While the LF and UHF transducers each have symmetrical polar patterns about their normal axes, the HF horn is unique in that its coverage pattern is not symmetrical from side to side. This can be appreciated by looking at the front view of the left loudspeaker, as shown in Figure 6. That portion of the HF horn which points across the listening area has a higher directivity index than that portion which points directly ahead. This will become apparent when we examine the polar patterns for the left channel, as shown in Figure 8A through F.

Polar curves are shown at 500 Hz, 1, 2, 4, 8, and 16 kHz. What these curves show is that listeners moving off the center line between the stereo pair will be moving into a stronger polar zone of one loudspeaker as they move closer to the the other loudspeaker. The polar characteristics of the right channel loudspeaker are, of course, the mirror image of the ones shown here.

In normal operation, the loudspeakers would be arrayed as shown in Figure 9. This represents the standard orientation for which the system was optimized. Under this set of conditions, we have plotted the inverse square level differences between the loudspeakers and compared them with the polar pattern differences at each frequency for each of the three off-axis listening positions, A, B, and C. This information is shown in Table 1.

The overcorrection at high frequencies is shown in Figure 10. Note that the overcorrection is apparent only above 4 kHz. At 16 kHz, it only approximates the Haas data shown in Figure 7. The need for this correction was estimated at the outset of the design cycle, but the precise amount was determined empirically, as we stated earlier.

Use of the Everest System in Larger and Smaller Spaces:

While we recommend the basic ratios shown in Figure 9, the actual dimensions of the playback array can be scaled up or down, as required by the listening environment. Inverse square relationships will scale upward and downward, but time delay relationships will not. The larger the spacing is, the greater will be the need for the precedence effect overcompensation, and under these conditions, the user may turn the loudspeakers inward slightly. For use with smaller spacing, they may be turned outward slightly.

Even when the basic equilateral listening triangle, cannot be accommodated, slight inward or outward shifting of the loudspeakers will optimize off-axis listening positions. The user can experiment at length.

Additional System Acoustical Characteristics:

Figure 11 shows the ground plane (free-field) response of the Project Everest system as measured on the axis of the HF horn. The response is referred to one watt at one meter. In half space, the response would be augmented at low frequencies as indicated by the dashed line.

The voltage drive to the three transducers is shown in Figure 12. Note that all roll-off slopes are 12 dB/octave. The HF section has additional frequency contouring built in to correct for the natural power response roll-off of the HF driver.

Acoustical crossovers are at 850 Hz between the LF and HF sections and 7500 between the HF and UHF sections. The effects of three-position switches in the HF and UHF sections are shown, as are the effects of a mid-bass contouring circuit. We have found that in some acoustic settings, the user will want to reduce the response in the 125 to 250 Hz region.

System Efficiency considerations:

The relatively high system sensitivity of 100 dB SPL, one watt at one meter, provides effortless performance at normal listening levels. As mentioned earlier, our intention was to build a system which could do justice to the best digital material available today.

As a rule, a high sensitivity system sacrifices LF bandwidth. However, in the DD 55000 system, we are dealing with sufficient enclosure volume to yield smooth response down to 40 Hz. It is our opinion that 40 Hz represents an excellent trade-off, considering the advantages of such high sensitivity. The occasional listener who wants response which reaches down to, say, 25 Hz will need a sub-woofer to cover that range.

Another advantage to high sensitivity is that, for a given acoustical output level, the distortion will be lower than with a system with lower sensitivity, all else being equal.

Driver Complement:

The LF driver is the model 150-4H. It resembles the older 150-4 driver which was used in the original Paragon system. It has the same thick top plate structure as is used in the LE-15 driver.

The HF driver is the model 2425H with titanium diaphragm. It is used with the model 2342 defined coverage horn, which was originally designed for the JBL 4660 Professional system for speech reinforcement in rectangular spaces.

The UHF transducer is the model 2405H ring radiator with its radiation slot oriented vertically.

The on-axis directivity index of the system is about 10-11 dB over the range from 800 Hz to 20 kHz. This, along with the flat overall on-axis response, ensures very smooth power response. Thus, the ratio of direct-to-reflected sound in most rooms will be relatively constant over the frequency range.

Table 1

Freq.	Position 1 Inverse Square Δ	Position 1 Polar Δ	Position 1 Difference	Position 2 Inverse Square Δ	Position 2 Polar Δ	Position 2 Difference
500 Hz	0 dB	0 dB	0 dB	-1.5 dB	0 dB	-1.5 dB
1 kHz	0	0	0	"	0	-1.5 dB
2 kHz	0	0	0	"	+1	-.5
4 kHz	0	0	0	"	+2.5	+.5
8 kHz	0	0	0	"	+1	-.5
16 kHz	0	0	0	"	+4	+2.5

Freq.	Position 3 Inverse Square Δ	Position 3 Polar Δ	Position 3 Difference	Position 4 Inverse Square Δ	Position 4 Polar Δ	Position 4 Difference
500 Hz	-2.7 dB	+.5 dB	-2.2 dB	-3.7 dB	+1 dB	-2.7 dB
1 kHz	"	+1	-1.7	"	+2	-1.7
2 kHz	"	+4	+1.3	"	+5.5	+1.8
4 kHz	"	+5.5	+2.8	"	+6	+2.3
8 kHz	"	+4	+1.3	"	+7.5	+3.8
16 kHz	"	+9	+6.3	"	+14	+10.3

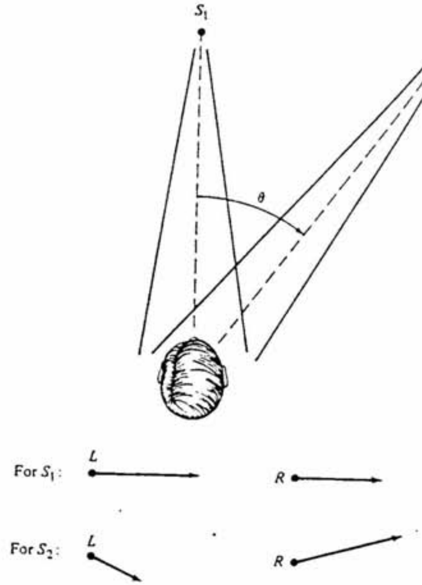

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Figure [illegible]. Phasors existing at the left and right ears for sounds in front of listener (S_1) and at front-right (S_2).

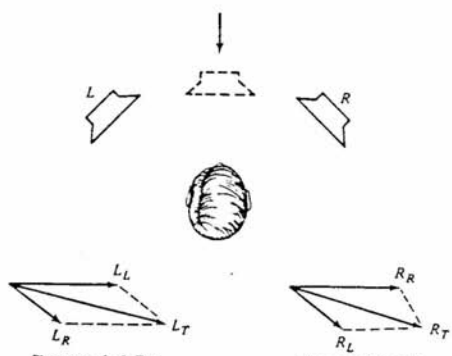

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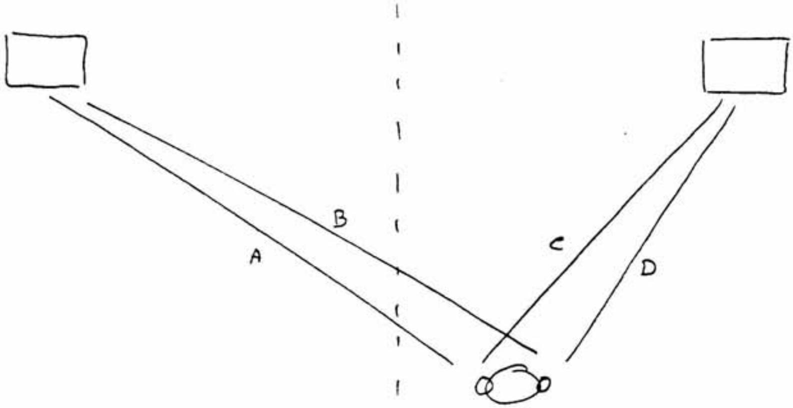
Figure [illegible]. Simulation of phasors for a sound source in front of listener using a left-right pair of loudspeakers.

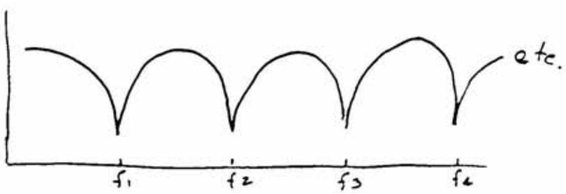
Figure 4. Phasors for Off-Center Locations



When the signal at L equals that at R (phantom center information), there will be unequal paths to each ear, producing cancellations and reinforcements. The lowest frequency at which cancellation will occur at the left ear is:

$$f_1 = 0.5\ c(A - C),$$

where c is the speed of sound. Thus:



$f_2 = 3f_1/2$

$f_3 = 5f_1/2$

$f_4 = 7f_1/2$, etc.

Another set of phasors will exist at the right ear, with f_1 determined as:

$$f_1 = 0.5\ c(B - D)$$

Figure 5. Everest System, Cutaway Top View

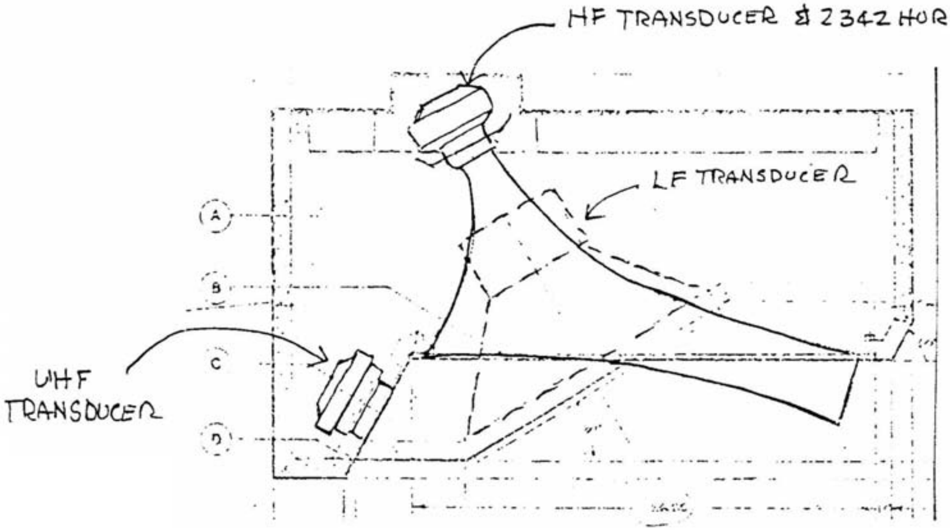


Figure 6. Everest System, Front View