Technical Notes Volume 3, Number 2:

# JBL's New LSR Mid-Field Monitors

## 1. Introduction and Basic System Description:

As the digital recording community contemplates higher sampling rates and greater resolution — and more playback channels — the need for continuing refinement in all aspects of monitoring takes on high priority at JBL.

LSR stands for Linear Spatial Reference, and it represents a change in the way we have traditionally measured and qualified monitor frequency response. Essentially, we have defined a nominal listening zone that is ± 30° horizontally and ± 15° vertically, and we have worked to make the response extremely flat within that entire solid angle, instead of limiting the design criteria to the more conventional approach of optimizing response along the primary (zero degree) axis. The need for this approach arises in the new era of surround sound mixing, where we expect there to be an accurate listening zone that will accommodate several important auditors: engineer, producer, and of course arranger/musician. JBL's first product embodying these design principles is the mid-field LSR32, a three-way system consisting of a 300 mm (12 in) LF transducer, a new 125 mm (5 in) MF transducer, and a new damped titanium 25 mm (1 in) dome HF transducer.

In order to achieve the degree of response smoothness necessary in this program, we had to pay closer attention than usual to details of transition frequency and slope, as well as transducer placement. For example, the LSR32 has its MF and HF transducers mounted on a rotatable sub-baffle so that their desired optimum orientation can be maintained for either vertical or horizontal positioning of the LF enclosure.

Two other aspects of monitor design have received added attention: reduction of distortion (by nearly an order of magnitude) and control of enclosure resonances (the entire baffle is a composite carbon fiber material that reduces resonances to a very low level).

The quest for lower distortion meant changes in diaphragm materials as well as fundamental changes in magnet/motor topology. JBL's Differential Drive™ technology, developed earlier for certain high-level music reinforcement applications, has been adapted to high-performance LF transducers for monitoring. Electromagnetic damping is used to control dynamic offset tendencies at high drive levels. The MF driver has the excursion capability to be crossed over at 250 Hz. Because of its extreme smoothness, the MF transducer can be crossed over to the HF unit at 2.2 kHz, providing nearly a decade-wide range of low distortion performance. The HF unit, a damped titanium 25 mm dome, is integrated with an Elliptical Oblate

Spheroidal waveguide with nominal 60° by 100° dispersion.

This technical note will cover all aspects of the system design, beginning with the transducers. We then move on to the details of enclosure/baffle and low-loss porting design, and dividing network considerations. Finally, we will present comprehensive measurements on all aspects of system performance:

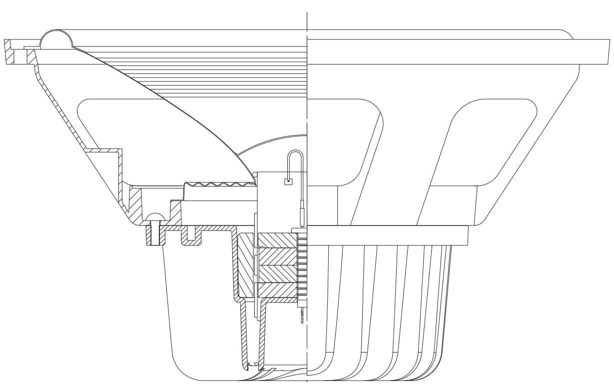
1. The LSR principle; response data and directivity curves
2. Harmonic distortion measurements
3. Port compression measurements
4. Power compression measurements
5. Impulse response measurements
6. Impedance measurements
7. Network high frequency adjustment.

## 2. Transducers:

## 2.1 Low frequency (LF) transducer:

Figure 1 shows a section view of the LF transducer. Note that there are two voice coils wound along the same axis and spaced approximately 36 mm, center to center. The neodymium magnet is located in the center of the structure. Note the shorted coil located midway between the two drive coils.

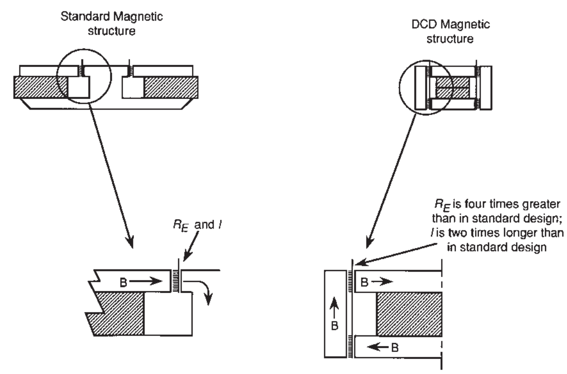
***Figure 1.*** **Woofer Cutaway**



We can get a clearer understanding of the advantages of Differential Drive by comparing it with an equivalent single coil structure, as shown in Figure 2. We use the term "equivalent" in that this comparison of both motor structures will have the same electromechanical coupling coefficient, $(Bl)^2/R_e$, and also the same moving mass. We will analyze the two structures in parallel, so to speak, in order to clarify the differences between them.

At 2A we show details of JBL's standard magnetic structure. Magnetic flux $B$ crosses the gap in which a coil of wire of length $l$ is placed. The voice coil has an electrical resistance $R_e$. These quantities establish the value of $(Bl)^2/R_e$.

***Figure 2.*** **Comparison of Standard and DCD technology**




Now, let us look at the Differential Drive topology shown at 2B. In this design there are two magnetic gaps with opposite flux. The two voice coils are connected in reverse so that the mechanical forces they produce will add (be in-phase). For the moving mass to remain the same, the two voice coils must have the <u>same height</u> and <u>half the thickness</u> as in the standard case. The value of B will remain the same.

When these changes are made, the total length (l) of the voice coil will be doubled and the resistance-per-unit length of wire will be doubled, since the cross-sectional area has been halved. The total resistance of both coils in series will then be four times what it was in the standard case shown at 2A. Since $(Bl)^2$ will have quadrupled (remember that $l$ has doubled), the new value of $(Bl)^2/R_e$ will be $(2Bl)^2/4R_e$. This is equal to $4(Bl)^2/4R_e$, which of course reduces directly to $(Bl)^2/R_e$.

In terms of electrical-to-mechanical coupling the two approaches are identical; but in other areas we have gained a great deal:

1. The new voice coil assembly now has <u>twice</u> the surface area of the old one. This means that it will have <u>twice</u> the heat dissipa-

tion of the old coil, which translates directly into twice the power (+3 dB) input capability for a given operating temperature and observed amount of dynamic compression.

2. The new voice coil structure will have less effective inductance than the standard one, since the reverse wound coils will have negative mutual inductance between them. This translates into a flatter impedance curve at higher frequencies, producing more acoustical output for a given drive signal.

3. The compact nature of the Differential Drive magnet structure requires much less iron in the magnetic return path. As a result, a Differential Drive transducer can weigh as little as one-third the equivalent standard design. The small overall magnet structure can be conveniently nested in a large heat sink for efficient dissipation of heat from the coils.

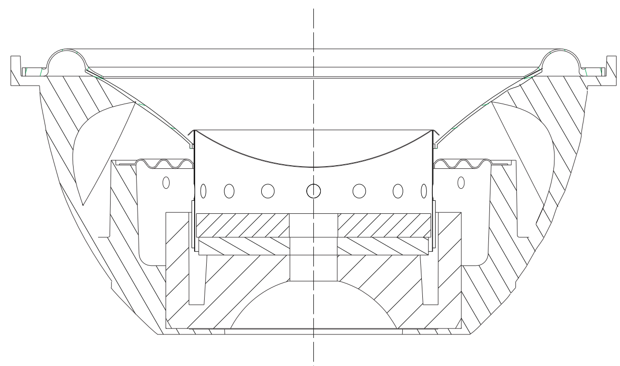
The LF cone is a carbon fiber composite that has an optimum combination of internal damping and stiffness. The outer suspension is a half-roll butyl rubber ring that provides a broadband termination for the cone, ensuring smooth response at upper frequencies and long term reliability.

The 252G also has a third shorted coil which operates as a dynamic brake. At normal operating levels, the shorted coil has no function except to provide the target moving mass of the transducer. However, at higher operating levels, the shorted coil will move alternately in and out of the two magnetic flux regions. The induced current will act in opposition to the force generated by the voice coil current, progressively damping, or braking, the peak excursions of the moving system. While this at first appears to be distortion generation, it is the opposite! The nonlinearity generated by the damping coil is in opposition to the dynamic instability of Bl product seen by the drive coils at their excursion extremes, and thus counteracts that loss, resulting in lower overall distortion at high cone displacements.

## 2.2 Midrange transducer:

A section view of the MF transducer is shown in Figure 3. It is a neodymium motor design and has a 50 mm (2 in) edge wound voice coil for high power handling capability. The cone is of woven Kevlar®, and the outer surround is a half roll of butyl rubber. The transducer has sufficient excursion range and linearity to enable it to be crossed over at 250 Hz, operating comfortably at rated input power. It is essentially free of the midrange distortion characteristic typical of similar drivers operating in the 250 Hz range at high levels.

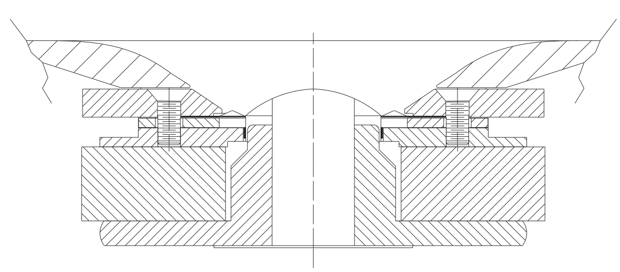
***Figure 3.*** **C500G Midrange Cutaway**



## 2.3 High frequency transducer:

The HF transducer is a 25 mm (1 in) smooth titanium dome that is damped by a thin coat of Aquaplas® as well as a unique low recovery foam in the rear cavity. Its frequency response extends smoothly beyond 20 kHz. Figure 4 shows a vertical section view of the elliptical oblate spheroidal waveguide that loads the diaphragm.

***Figure 4.*** **053ti HF Cutaway**

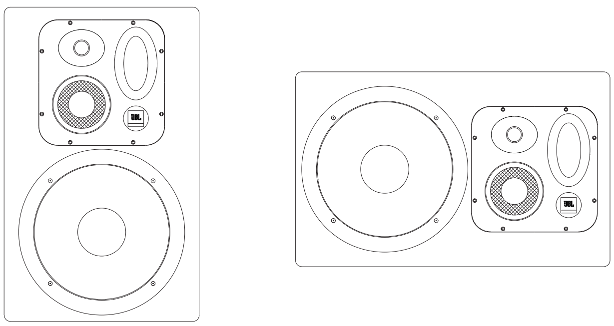


## 3. Enclosure and baffle details:

Figure 5 shows the two orientation options of the LSR32 system. The MF/HF section may be positioned inboard or outboard, depending on the width of the stereo soundstage desired. Note that all orientations permit the MF and HF elements to be positioned vertically, ensuring consistent maintenance of LSR coverage angles.

It is worth noting that system response is virtually the same, regardless of the positioning of the baffle plate.

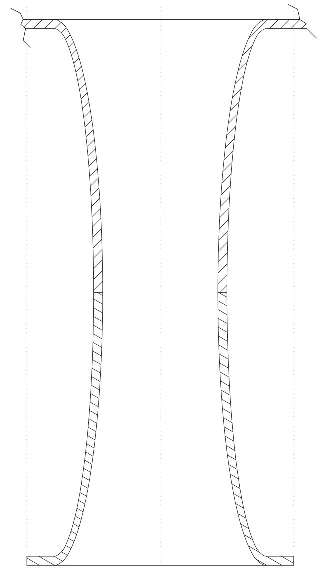
***Figure 5.*** **Horizontal and Vertical Orientation**



The baffle itself is made of a carbon fiber composite that has a thickness of 38 mm. It is virtually resonance free and is complemented by enclosure walls that are braced. As we engineer better transducers with lower distortion, we find that enclosure resonances, even those produced by conventionally good, sturdy design, may be heard as such and contribute to a distortion “signature”. It is the next step in refinement of system performance to reduce them.

Another source of distortion at high levels has been port turbulence, which occurs at enclosure resonance when cone excursion is at a minimum and air volume velocity through the port is at a maximum. Under that condition, air particle velocity will also be maximum, creating the “breathing” noises commonly noted. Contouring the port tube is one method of minimizing this effect. Figure 6 shows a section view of the enclosure through the port tube. Note the continuous flare through the tube.

***Figure 6.*** **Linear Dynamics Aperture**

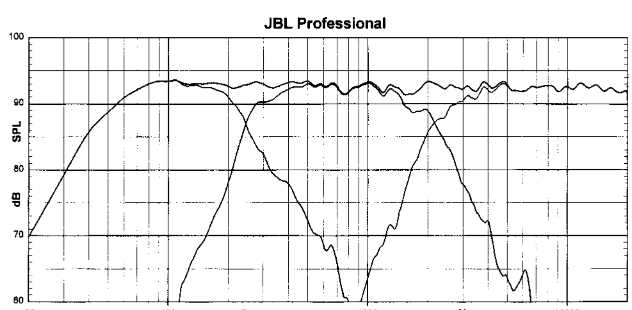


## 4. Dividing network design and performance:

The network makes use of fourth-order Linkwitz-Riley acoustic response filters. The high slopes have the advantage of minimizing lobing between adjacent elements in the regions of crossover. This is a crucial element in our efforts to assure smooth response throughout the target listening window.

All network elements in the primary signal paths are of the highest quality including low loss capacitors and high saturation current, low distortion industors. Figure 7 shows the on-axis acoustical contributions of each driver. Here, the -6 dB (in-phase) relationships at both crossover points can be seen.

***Figure 7.*** **Acoustic Contribution**




## 5. Performance measurements:

### 5.1 The LSR principle; axial measurements and directivity:

Harman International research activities in loudspeaker measurements have developed a family of response curves that convey much information for the user regarding the performance of a loudspeaker in a typical listening environment. The raw data for the analysis involves a set of anechoic axial frequency response measurements made at angular intervals of 5 degrees in both horizontal and vertical planes. That data is weighted and averaged to produce the following plots of level versus frequency:

1. On-axis response.
2. Averaged response over a nominal listening window which is ± 30° in the horizontal plane and ±15° in the vertical plane.
3. Averaged simulated early reflections arriving 5 to 10 msec after the direct sound and lasting for approximately 10 to 20 msec.
4. Total radiated sound power.
5. Directivity index of sound power.
6. Directivity index of early reflections.

Taken as a group, this family of six curves will to a very large extent, define a loudspeakers interaction with the listening space and give a clear indication of how listeners will respond to the system. The on-axis curve serves as a reference. The on-axis data and the averaged listening window data, presented in curve 2, will not vary greatly from each other in a well executed design.

The third curve is a calculation that simulates the ensemble of early reflections from walls, floor and ceiling. This curve normally tilts downward, showing the tendency of all loudspeakers to become more directional at higher frequencies. It is important that the general trend of this curve be free of any response irregularities, especially those that do not occur in the listening window curve.

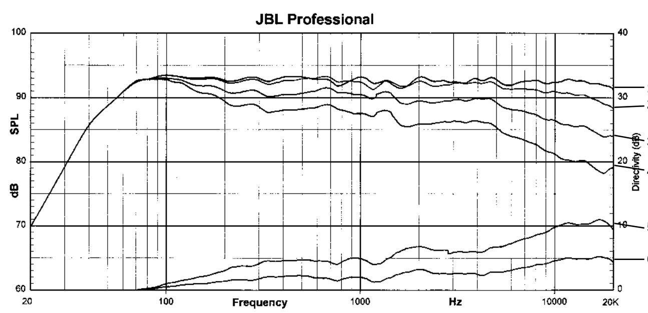
The fourth curve is a calculation of total radiated sound power from the loudspeaker. It is a measurement of the total acoustical energy radiated into the reverberant field of the listening room. This curve falls with rising frequency, indicating the normal tendency of all loudspeakers to become more directional with rising frequency. As with the previous curves, this one should also be uniform and free of irregularities.

Curve 5 shows the directivity index of the early reflections curve. Mathematically, this is the difference between the direct sound curve and the early reflections curve expressed in dB.

Curve 6 is the directivity index of the total radiated sound power. It is the typical directivity index curve seen on many loudspeaker specification sheets and represents the ratio in dB between sound radiated on-axis and the integration of sound radiated from the loudspeaker in all directions.

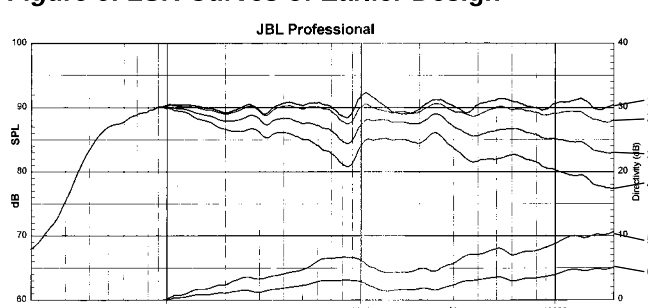
As an example, we show in Figure 8 this family of curves for the LSR32 system. The on-axis response is virtually flat (+1.0 dB, -1.5 dB); the listening window shows a slight divergence from on-axis at high frequencies, but is free of any abrupt variations. The averaged early reflections curve is likewise free of any abrupt variations, but shows slightly more roll-off at higher frequencies.

***Figure 8.*** **LSR32 Curves**




As an example of an earlier three-way loudspeaker system designed with attention primarily on attaining a flat on-axis response, we show the data of Figure 9. Note that the on-axis curve is fairly flat, but that the listening window and early reflections curves show clear irregularities.

***Figure 9.*** **LSR Curves of Earlier Design**



## 5.2 Second and third harmonic distortion data:

Plots of second and third harmonic distortion have traditionally been presented by JBL at various nominal operating levels. Typically, we will show the data for a level of 96 dB as measured at one meter. Those models intended for higher level applications may have their distortion at measured 102 dB as well.

Distortion data for 1 m on-axis SPL levels of 96 dB and 102 dB are presented in figures 10A and 10B, respectively. In these curves the distortion components have been raised 20 dB for convenience in reading.

***Figure 10A.*** **96dB Distortion Graph**

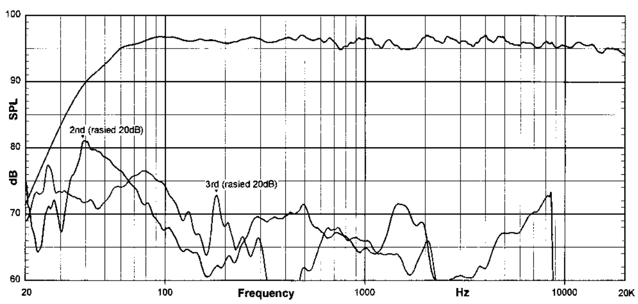
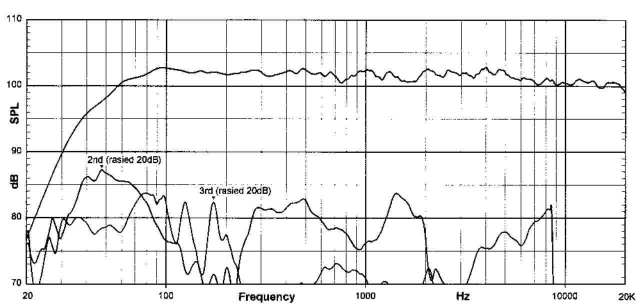



***Figure 10B.*** **102dB Distortion Graph**

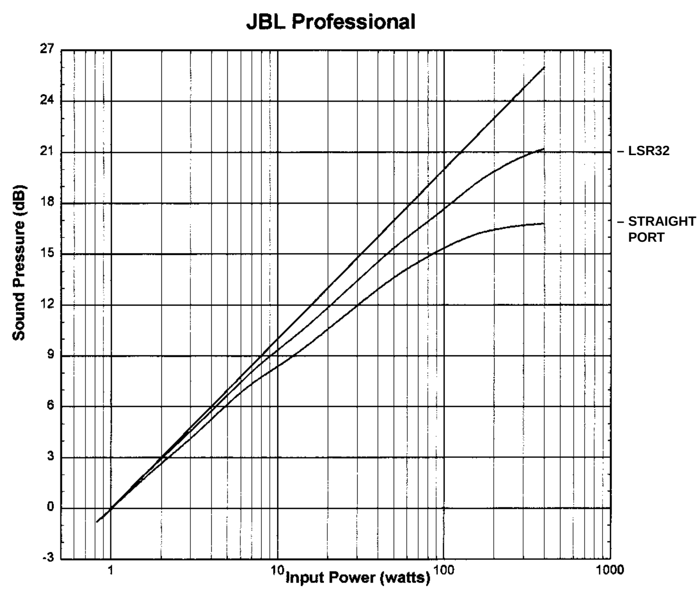



Note in Figure 10A that the level of both distortion components in the frequency range above 100 Hz is -40 dB or lower, relative to the fundamental. This corresponds to distortion of less than 1% and represents unusually fine performance at such a high acoustical output level. The data taken at 102 dB shows distortion components above 100 Hz lying - 35 dB or lower, corresponding to distortion in the range of 2% — again, remarkably fine performance.

## 5.3 Port Compression data:

Figure 11 shows data on port compression. The curves show the system output at the port tuning frequency as input power is progressively raised. The top reference line shows a linear relation between power input and sound pressure level. The difference between the LSR32 flared port and a straight port design can clearly be seen, reaching a difference of 4.4 dB at a power input of 400 watts. Note that for the LSR32 system there is about 0.5 dB compression at 10 watts input, gradually increasing to 2 dB for 100 watts input. For the straight port the compression at 10 watts input is 1.5 dB, increasing to 4.5 dB at 100 watts.

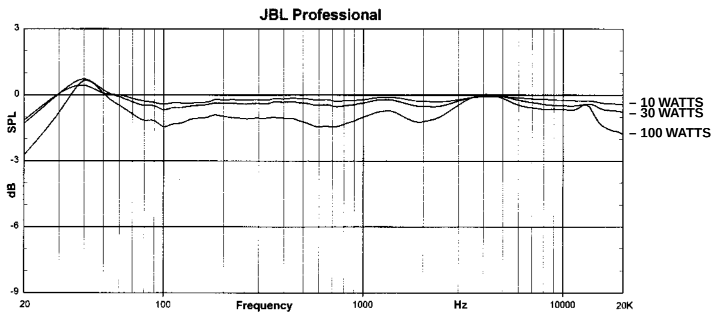
***Figure 11.*** **Port Compression Graphic**




## 5.4 Power compression data:

Power compression is normally shown by scaling and superimposing curves run at different power levels so that only the effects of non-linear response can be seen. Power compression is caused by voice coil temperature rise and the consequent loss of efficiency as the voice coil resistance increases. If there were no power compression at all, then the two response curves would lie directly one atop the other. Figure 12 shows the power compression of the LSR32 system at 10, 30 and 100 watts. The curves are normalized such that the flat curve at 0 dB represents the response measured at 1 watt. At 100 watts input, the power compression is seen to be less than 1.5 dB.

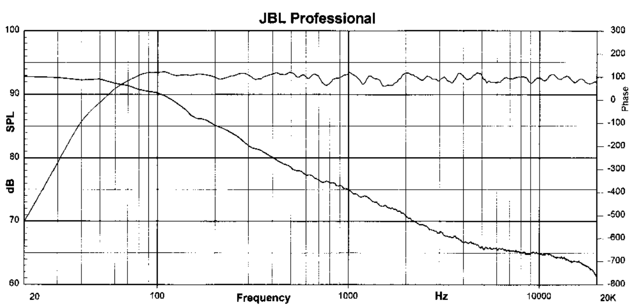
***Figure 12.*** **LSR32 Power Compression**




## 5.5 Time domain performance data:

Figure 13 shows plots of on-axis amplitude and phase response for the LSR32. Note the uniform system response and smooth phase curve.

***Figure 13.*** **Amplitude and Phase**




The phase curve shows a progressively larger negative shift with respect to frequency. For this system, the group delay (-df/dw) would then tend to be constant over most of the frequency range, since the phase response is of constant slope over most of the range.

***Figure 14.*** **Impulse Response**

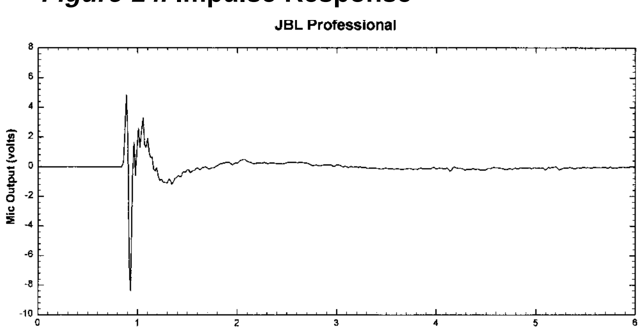
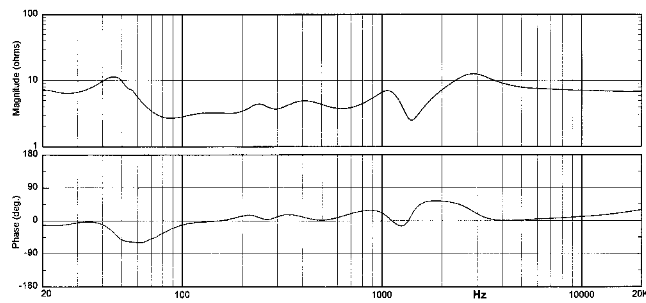



Figure 14 shows the impulse response of the LSR32 System. Note the smooth and rapid decay of the impulse response.

## 5.5 Impedance measurements:

Figure 15 shows the magnitude and phase of the input impedance of the LSR32 system as a function of frequency. By convention, we would refer to the system as having a nominal impedance of 4 ohms. Today, the solid state amplifiers normally used in high performance monitoring routinely handle impedances of 4 ohms and minimum values of 0.8 that value (3.2 ohms) in stride. Note that the phase angle does not exceed a value of ± 60 degrees, indicating that the loading on the driving amplifier will be stable at all times.

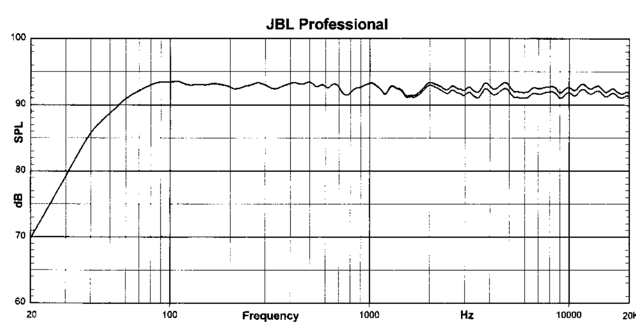
***Figure 15.*** **Impedance Amplitude and Phase**




## 5.6 Network response option:

Via a back panel jumper, the HF response of the LSR32 system can be reduced one dB between 2.5 and 20 kHz from nominal flat response, as shown in Figure 16. This is a user option to be made based on taste or prior practice. While there is a trend toward flat playback response in controlled environments, it may be some time before all users are ready to adopt it. Additionally, users have the option of using traditional monitor electronic equalization to achieve a given playback standard or “house curve”.

***Figure 15.*** **HF Adjustment**



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