Automotive Audio Design (A Tutorial)

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Abstract: The automotive space is a harsh and challenging listening environment. This paper will cover the various spectral and spatial aspects of the vehicle environment and how system designers approach the challenges provided by the environment: Amplifier and Speaker design, Speaker placement, and System tuning and control.

INTRODUCTION

This paper is intended as an overview of a few fundamental elements in automotive audio system design. The first that is discussed is the vehicle listening environment itself, because it is that environment that imposes the boundary conditions on the remaining elements of system design. The use of additional amplification after the radio head unit for additional power output and sound field control follows with emphasis on how different sound field characteristics are approached in the design of an amplifier. Following the environment and amplifier design is the close relationship between loudspeaker design and loudspeaker placement in a vehicle. The type and methods for obtaining objective measurements, such as frequency response, low frequency capability, distortion, and dynamic characterization, is then discussed. Then the final element of system design comes in, the subjective evaluation of the audio system's performance. Listening test methods and listener training are described and the varying degrees of test methods' usefulness are compared.

VEHICLE LISTENING ENVIRONMENT

As described in House [1], the automobile is not an ideal listening environment. The one redeeming fact is that the seat position for each listener is known; otherwise, the automobile is a small noisy environment with several negative influences on the spectral, spatial, and temporal attributes of a reproduced sound field. A short summary of those negative influences follows.

Spectral Characteristics

Many factors can contribute to a vehicle's real and perceived frequency response: The interior volume of a vehicle, the size and shape of interior boundary surfaces, absorption characteristics of all the boundary surfaces, and the location of speakers in the vehicle relative to the listener and relative to nearby boundaries.

For instance, an average mid-sized automobile has a volume of approximately 3.5 m^3. In that volume there will be a large uniform coupling of acoustic modes between 80 and 300Hz. There will typically be some major resonances due to transverse modes between 120 and 150 Hz. [Illus. 1] The amplitude of these modal resonances can be as high as 12 dB SPL. Above 300 Hz, and up to approximately 1kHz, the large coupling of modes is reduced and modal resonances with Q values of less than 3 are numerous. In this region there is a great deal of spectral coloration due to the broader spacing of the modes, and variations in amplitude as high as 8 dB SPL can be observed. The volumes of the automobile's trunk and doors where speakers are mounted can contribute resonances as well between 150 to 500 Hz, with amplitudes anywhere from 4 to 8 dB SPL. [Graph 1]

Mechanical resonances can occur due to the vibration of the roof (which is the case for vans and minivans) or the trunk lid due to road motion or engine vibration. These surfaces can radiate acoustic energy at Q values higher than 5 between 300 to 1k Hz with amplitudes from 4 to 6 dB SPL.

Vehicle interiors consist of glass, plastic, carpet, cloth or leather on surfaces that are at various compound angles which respect to each other, the listener and the loudspeaker. Reflections off of these many and varied boundaries result in complex interference and diffraction effects which cause frequency response aberrations and sound coloration at frequencies above 300 to 500 Hz. Interference occurs between direct and reflected waves. Diffraction will occur through speaker mounting holes and grille coverings, and from mounting locations that are drastically off-axis from the listener.

The vehicle interior also has inherently high ambient noise conditions when the vehicle is on the road. Road, wind, and engine noise can reduce an audio system's dynamic range and mask low frequencies. For the typical mid-sized automobile, the noise is most dominant below 500Hz. The average noise SPL is greater than 80dB for frequencies below 100 Hz and velocities above 35 mph.

Some of these problems can be controlled and overcome. Noise masking can be combated with dynamic loudness curves that are tuned to the specific vehicle. Some mechanical resonances can be reduced by reducing the acoustic resonances. Some cannot. The vehicle's structural integrity is the main source of such problems. Some frequency response aberrations can be successfully equalized by an audio system designer. The quality of that equalization work depends on the skill of the designer and how severe the cause of the aberration(s) might be. In general non-minimum phase aberrations cannot be successfully equalized. And, when using analog electronics, added equalization peaks or dips of Q values higher than 5 can cause phase shifts that can become as audible as other aberrations. The acoustic radiation from a mechanical vibration may be reduced by equalization, but the audio sound may still have a poor damping characteristic, which causes it to sound unnatural.

Amplifier Design (Spectral)

A typical block diagram of an automotive amplifier is shown in [Illus. 2]. Whether the automotive amplifier is integrated or not, the pre-amp section of the amplifier that is designed specifically for control of the spectral aspects of the automotive environment will usually have some form of parametric equalization capability. Parametric equalization provides the ability to add filters with control parameters of center frequency, gain, and Q. The equalization should be available, at least, for each quadrant of the vehicle (left front, right front, left rear, and right rear), or for best results, for each channel of the amplifier. This will allow a wide range of adjustment in the amplitude versus frequency response to correct for any sound field anomalies for all listening positions. In some cases, shelving filters will be used (primarily in the rear seat) to maintain a balanced sound field for both the front seat and the rear. The pre-amp will have split-band high pass and low pass active filters. The order of the filters are determined by the needs of the specific application, but a practical nominal type is a 2nd order, 12 dB/Octave, Butterworth filter. The high pass and low pass filters are used to optimize the use of each loudspeaker within its most efficient power bandwidth with minimal harmonic and modulation distortion and spectrally balance the voicing of channel of the complete system. Most quality automotive audio amplifiers provide some form of distortion limiting in the power amp section. Split-band voltage sensing limiting is useful in dynamically, and rapidly, adjusting the output gain for each channel to reduce distortion at high volume levels. The front and rear channels should be adjusted independently. As useful as distortion limiting is, as with any aspect of the audio chain, proper adjustment is critical. Overuse of limiting can cause a pumping of sound as the limiters kick in and out too much. One other feature which could be added here would be a speed sensing loudness or gain control on all or specific channels, depending on what channels were being masked by wind and road noise.

The outputs of the amp can be used, as described, for multi-channel supplies or as bridged, mono, supplies. Either way, an automotive amplifier needs to provide full protection against all the adverse conditions that can exist in the automotive environment. Signal overload is nothing new to audio amplifiers, and protection against it in the automotive field is still required. Thermal overload is a large concern in automotive audio, much as it is in pro audio. Automotive amplifiers, especially in O.E.M. applications, are mounted in tight quarters where little or no airflow is available. Amplifiers need to be designed to optimize heat convection and heat dissipation to reduce the chance of overheating. Therefore, the output section of the amp will also have a way of detecting thermal overload and then scaling back output levels or simply shutting down to avoid damage to itself and the area surrounding the amplifier. Automotive amplifiers are also packaged in locations where humidity, condensation, or direct contact to moisture occurs periodically or constantly. The amplifier needs to be protected from the moisture as much as possible without also compromising its thermal capacity. It also needs to be capable of handling a short on the input power line due to the moisture. It should either be able to handle the overload without sending it directly to the loudspeaker or it should simply shut down. Direct voltage from an automotive battery to a loudspeaker usually means serious damage to the loudspeaker and potentially serious damage to area where it is mounted.

Spatial and Temporal

Because of the small dimensions of the vehicle interior, the reverberation time is virtually zero. In this size environment a reverberant field cannot form. The sound field in a vehicle consists entirely of direct energy and early reflections, which are quickly absorbed or dissipated. For the typical vehicle, the decay time for a 60dB signal reduction is 30 to 50ms. This provides a very dead space for sound reproduction. The large amount of early reflections produce a type of diffuse sound field where 90% of the energy occurs within the first 10ms of the direct wave arrival. [Graph 2a & 2b.] The listener's brain integrates that information and uses it to formulate a perception of the sound field. Effectively the listener is experiencing a diffuse and frontally incident sound field. Looking at impulse response curves or energy time curves [Graph 3.], it can be seen that in this diffuse and frontally incident sound field, those first 10ms of soundwave incidence are somewhat busy. In the driver's position (with the window on the left side of the driver) the time curves show that the sound from the opposite side speaker to the ear (e.g., right speaker to left ear) arrives approximately 1 to 2ms later than the same side speaker to the ear. The arrival from the opposite side speaker has a greater amplitude than the same side arrival. And, whereas the same side arrival decays naturally, the opposite side speaker has a reflection off a nearby surface that arrives approximately 4 to 5ms later before it begins to decay. The amplitude of that reflection is also nearly as large as the initial opposite side arrival. The listener integrates this subtle information and creates an impression of the width and breadth of the sound stage. This is not an experience common with a typical listening room. There are no nearby boundaries to smear the time signature of the sound field and confuse the lateral staging or clarity. With the advent of practical digital signal processors for automotive amplifiers, subtle time corrections, 3D algorithms, and surround sound processing can be used to overcome some of the spatial limitations of the automotive listening environment. This too is at the mercy of the designer's skill and can cause even worse problems if done incorrectly.

Amplifier Design (Spatial & Temporal)

Another typical block diagram of an automotive amplifier is shown in [Illus. 3]. This one differs from the earlier one mentioned in that it has broken the multi-channel architecture into more channels for better coverage and control. There are added benefits for the spectral aspects of the sound field in the automobile in terms of a better frequency balance and a cleaner sound. What makes the spectral aspects better can also greatly improve the spatial quality of the audio reproduction. The greater detail in this amplifier design is a step closer to a more accurate presentation of the sound field, and therefore, a more accurate spectral and spatial reproduction. The power amp section is pretty much the same as before with limiting and overload protection. The pre-amp section has a similar set of parametric equalization and split-band high pass and low pass filtering as before. A center channel has been added for the front listening space, and the rear channels have been split for bi-amplification between low and high frequencies (very similar to the way the fronts were for the previous example). More so than in the spectral example, the use of the high pass and low pass filters are used both to maintain a good spectral balance and to voice each channel of the complete audio system so that the spatial quality of the system (sound stage and acoustic imaging) are enhanced. This level of amplifier design can be realized in analog but is usually more practical and elegant as a digital design. Many types of additional control can be added. One example is the capability of adding time delay to any channel in order to align all the arrival times to a point in the cabin interior for each amplifier channel. Time smearing and phase distortion can be reduced through time alignment. Other examples could include the addition of three-dimensional and/or surround sound steering algorithms. Here again speed dependent channel gain or loudness becomes important, due to the effects of masking on the steering of spatial information in the vehicle.

LOUDSPEAKERS

In some cases, quality spectral and temporal electronic control can make up for some short-comings in loudspeaker design limitations and loudspeaker locations in a vehicle. The corollary is also true: A good custom loudspeaker design placed in the best possible location in a vehicle can make up for some short-comings in electronic capability.

Loudspeaker Design

All the parameters that any consumer or pro product designer would be interested in are what an automotive audio designer would be interested in. A loudspeaker's resonance (Fo), Q, efficiency and DC resistance (DCR), power handling, and dispersion are a few of those loudspeaker characteristics. Generic loudspeaker designs and carryover of existing, or commercially available, designs can have economical advantages in system design, unless they start to compromise the sound quality of the audio system. Ideally, an automotive loudspeaker is designed to exploit its location in a vehicle. Its Fo and Q are matched to its mounting location. Whether it is a door, an instrument panel, or truck cavity, the Fo and Q are specified so that once the loudspeaker is mounted in that location it is properly damped and has the desired frequency response near Fo. The efficiency of the loudspeaker is specified so that its gain relative to other speakers and the listening location(s) is well balanced. Its DCR is specified so that will not be putting undue stress on the power amp section of the amplifier, but yet be as efficient as desired or possible. The loudspeaker's off-axis characteristics are specified to best serve the system's overall frequency balance, staging and imaging for everyone listening in the vehicle. Some loudspeaker locations that are closely coupled to the listener will require less off-axis information than other more distant locations that are meant to cover a broader area.

Loudspeaker Placement

Loudspeakers are located to take advantage of their design and to reduce the amount of electronic control necessary to create the best spectral and spatial sound quality

Woofers and Subwoofers

> Bass drivers are located so they can best couple with the structural and acoustical modes of a vehicle and provide that information to both the front and rear passengers. This location is typically in the front or rear corners of the vehicle. At these locations, the low frequency drivers can couple well with the longitudinal modes of a rectangular cavity. This works well for most sedans, coupes, and SUV's. It becomes difficult with truck cabins. A truck's cabin is very close to being a square volume, and the lack of modal reinforcement and high level of modal cancellation doesn't allow for much low frequency reinforcement and actually causes some cancellation in the low frequency.

All loudspeakers require some sort of baffling to avoid cancellation of the front and rear waves from the loudspeaker diaphragm. For bass drivers, the required baffle in order to reproduce most of their piston band acoustic radiation would need to be able avoid canceling a wavelength of approximately 3 – 17 meters (for 100 – 20 Hz). It would need to greater than ½ the wavelength to be considered an infinite baffle. An enclosure can be substantially smaller than the dimensions required for a good baffle. With some attention to leakage paths and structural integrity some automotive door locations can prove to be good loudspeaker enclosures. With the same attention to detail, rear package shelves with the trunk cavity behind can also be good locations. Drawbacks to using existing door volumes for the truck space for bass enclosures occur when leakage and structural integrity cannot be controlled. Loudspeaker parameters can also change readily due to direct exposure to moisture and temperature. Alternatives are separate enclosures designed to fit into the door volumes and under rear package shelves. These enclosures provide better control over some of the problems. They are also more expensive and add weight to the vehicle.

Midranges

Midrange drivers should be placed at elevations above the knees, closer to ear level to raise stage and image. They should not be placed too distant from woofers. This improves the smooth transition between the bandwidths of the woofer and midrange. If midranges are placed in the instrument panel, usually the most optimal location is one that is within a distance between the loudspeaker to the apex of the windshield and instrument panel equal to one loudspeaker diameter. [2] Illustration 4 and Graph 4 & 5 help illustrate these points in the smoothness in the frequency and time domains. Center speakers can be used for a complete, seamless stereo balance and multi-channel applications.

Tweeter and Upper Midranges

Tweeter and upper midrange drivers are placed depending on the goal of the system. If broad coverage is desired, the drivers are not directly aimed at the listening positions, they are crossfired (aimed at the opposing listening position). [Illus. 5] If close-coupling is desired (i.e. personalized sound fields), drivers are placed as close to the listening position as possible, aimed directly at a single listening position. Locations for drivers could be the headliner, headrest, etc. depending on the ability of the electronic control available to redirect or steer the sound stage and image to the normal listening position.

Front and Rear Effects

The placement of midrange and tweeter drivers is tempered by the effects they have not only on the near seat positions, but on the effect they have on other seating positions. For better or worse, front seat speakers contribute to the rear seat sound, and *vice versa.* The rear speaker locations provide rear fill for the front passengers, which helps provide them with a sense of spacious. Likewise the front drivers help elevate and create a frontal stage for the rear passengers. However, for example, midranges or tweeters placed too high in the rear upper door or overhead will pull the stage and image too far back behind the front passengers. Use of time delay on these locations can alleviate some of these problems, but cannot completely fix poor positioning of a loudspeaker.

OBJECTIVE MEASUREMENTS

To help quantify the effects of the loudspeakers' design and placement and the effects of the electronic controls overlaid on them, a set of objective measurements in an automobile is needed. There are several aspects of the system's audio performance that need to be evaluated. By the end of the design cycle, all of the following will have been covered. At any given time in the design process a subset of them will be of interest. The measurements should be made at any listening position that is considered a primary or secondary listening position. In most vehicles the primary is the driver and the secondary is the right rear passenger.

Frequency Response & Total SPL

Using a CD source and/or FM transmission into the radio, the frequency response of the audio system is measured. For complete documentation, measurements of each individual speaker location should be made, as well as measurements of the sound balance for a full front fade, then a rear fade, and then a full system balance (FSB) measurement. The source that can be used to easily document sound pressure amplitude levels is pink noise. (Pink noise also proves to be a good for a trained listener to pick out high Q resonances.) A combination of uncorrelated and correlated pink noise can be used. Uncorrelated pink noise is more representative of stereo music, and correlated is more like mono voice or music. Depending on the type of music, the bass will behave more or less like one of the types of pink noise. Bass for pop music behaves more like correlated pink noise, and bass for orchestral music behaves more like uncorrelated pink noise. A consistent measurement level for each measurement was determined empirically by asking a large group of listeners to sit in a vehicle and set the volume to a comfortable listening level for all sources. The resulting SPL level was 85 dBA FSB, and that is the reference for all the other SPL measurements.

A set of six omnidirectional microphones is recommended, with a spacing of approximately 14cm between each microphone. Omnidirectional microphone capsules are manufactured today with elements as small as 2 mm, which makes them omnidirectional up to 20,000 Hz. This microphone array should be placed at the approximate intended location of the listeners head, at a slight angle, approximately 30°. The purpose of the 6 microphones is to ensure that the measurement is an average over the space of the listener's head. A microphone in a single location could be sitting in a node or anti-node of the sound field and be providing information that the brain and ears are not really focusing on. [3] The output of each microphone is multiplexed together into one averaged measurement and recorded using a spectrum analyzer. Binaural dummy heads are also available for making measurements. There are several models available. Each has been proven to have some angles of incident which are less accurate than others. [4]

Traditionally, 1/3rd octave bands have been used for making audio measurements, but research has shown [3] that Q's as high as 50 can be heard in controlled circumstances if their amplitude is high enough. Measuring or making equalization adjustments with 1/3rd octave bands will provide an understanding of only the gross overall nature of the frequency response and could be hiding the true flaws that are being perceived by a listener. At least 1/6th octave would be better to have a better understanding of the frequency response. Octave bands of 1/20th of an octave are available on most PC-based, or otherwise portable, analyzers today and can be useful for seeing the detail behind a measured resonance and determining if it will be audible or not.

Dynamic Capability

Before any distortion testing is done, some idea of how loud the system can get before it distorts should be determined. A measurement should be made of what SPL can be achieved across the audio spectrum at a volume level that is just before clipping is audible. This can be done with a 1 octave correlated pink noise source with a 4:1 crest factor. Each octave band is played and the volume turned up until clipping is heard or max volume has been reached. At that point the SPL is recorded.

Low Frequency Performance

A percent total harmonic distortion (%THD) measurement should be made using a sinewave input from 20 – 500 Hz at least. The measurement should be made on the amplifier to determine its electrical %THD, and an acoustical measurement should be made in the vehicle. In some cases, 2^{nd} and 3^{rd} harmonic measurements, which are very useful in loudspeaker design, can also be useful for understanding a source of harmonic distortion. If it can not be seen with a high-resolution THD measurement, then 2^{nd} and 3^{rd} harmonic measurements will provide the next step in better resolution. The harmonic distortion measure can be made at any level. It can be useful to see the harmonic distortion at several voltage levels (including what has been determined to be maximum) to see how the system is changing with volume levels.

Intermodulation Distortion

Intermodulation distortion should be measured at various voltage levels (including the maximum) to determine if any coloration could exist due to intermodulation. Frequencies of 60 Hz / 700 Hz and 60 Hz / 7000 Hz are a good examples of sets of frequency tones that will help identify any low frequency and midrange and high frequency interplay.

System Linearity (w/o Loudness)

How a system sounds at varying input levels is of interest because the goal of the system design is to have it remain balanced at all levels. Because most radios have a variable loudness contour attached to the volume control which boosts the bass and in some cases boosts the highs a little depending where the volume is set, to get a true understanding of system linearity, it needs to be measured without the loudness being used. The way to do this is to bypass the volume control by creating sources which have varying output levels. So the radio is first set to maximum volume (where the effect of any loudness curve will be 0dB) and a reference frequency response measurement is made there. A CD source which has - 6, -10, -20, -30, -40, and -50dB recorded levels of pink noise on it is used next and SPL measurements made at each reduced output level. The resulting SPL curves are normalized and compared to the reference curve. Any deviation from the reference curve is a non-linearity. Once a level of linearity has been established, then the effect of the loudness contours can be considered and modified if it is degrading the sound quality.

Impulse responses

Impulse responses run at each listening position for each speaker location and FSB provide a variety of information to the designer. It is, first, time domain information about the early reflections of the car and where delay could be used. It is also information about the transient behavior of the system or individual speakers. An understanding of how well the system is damped is gained. And, after a convolution into the frequency domain, amplitude and phase information is available. The system level for running the impulse response was determine to be 85dBC.

Radio Frequency Response and Distortion

The frequency response for the radio/head-unit should be measured for varying volume control levels for all sources that are available: e.g., AM/FM, Cassette, CD. This documents any non-linearities in the radio and what (if any) loudness contouring is taking place. Distortion measurements (%THD) should also be run on the radio at varying vloume control levels.

Dynamic Measurements

> Road noise measurements should always be done or the results known in order to understand how the road noise, or engine noise effects the low frequency response in a vehicle. For example frequency response curves should be made at 0 mph (with the engine running), 10, 30, 50, and 70 mph.

Next, a good way to evaluate and record what our perceptions are of what we hear is needed. A good subjective measurement to compare to the objective measurement is needed so the design of the audio system can be optimized for listening.

VEHICLE LISTENING TEST METHODS

The effects of this “listening room”, which is the interior volume of an automobile, on the loudspeakers in that “room” are rather dramatic and severe. The automotive space produces a sound field that is much more complex than the typically controlled listening environment where speaker systems might be evaluated. To make listening tests valuable -- with repeatable, statistically significant, quantitative results which will be useful for a system designer, a design team, as well as a marketing team -- very careful training of the listeners involved in automotive sound evaluations is therefore required. Then, a simple unbiased listening test method is required.

Listener Training

The benefits of trained listeners have been well established for listening in rooms and at computer workstations. [6],[7],[8] Recent studies have illustrated the use of a self-administered PC-based training program to improve listeners’ ability to reliably identify and rate different types of spectral peaks and dips which have been added to a variety of programs (i.e., resonance detection). [9],[10] Similar investigations have indicated that critical listening in automobiles might require a more detailed training regimen and additional repeats within a trial.

The method of using a self-administered PC-based training program that requires the listener to identify spectral peaks and dips in a set of source material reinforces the listeners’ ability to relate a perceived spectral aberration to a common frequency scale. However, experience has shown that because a subject consistently scores high on such a resonance detection test, it does not mean that they are well trained for listening evaluation purposes. Training sessions in the use of the preference ranking and/or high fidelity rating and timbre balance scales are also necessary. The objective is to verify that the listener is applying the resonance detection training properly in the context of a listening evaluation or experiment. The training should focus on frequency-related problems since these are the problems most untrained listeners find difficult to describe. Spatial quality rating and distortion detection training should also be pursued for a well-rounded training.

All the training, resonance detection, preference ranking/high fidelity rating, timbre balance, spatial quality, distortion detection, should be performed until the subject has demonstrated consistent behavior within the error guidelines of the listening test. Guidelines of 95% confidence levels are acceptable for producing results that can considered statistically significant and repeatable. The training should be repeated on each subject to maintain their skills at an acceptable level and to keep them focused on the task. For regular listeners, a refresh every six months proves useful. For less regular listeners (listening less than once a week) monthly refreshes, or a refresh before any listening session is advised.

Listening Methods

In-situ sighted listening tests are commonly used within the automotive industry to evaluate the sound quality of automotive sound systems. However, studies on consumer loudspeakers indicate that sighted judgements of sound quality are strongly biased by non-auditory related biases that include size, price, and brand name [11]. It would be logical to suspect that most listening tests done in automobiles may be influenced by these biases as well. Daily work in automotive sound design and evaluations show this to be true, and past experimental results have shown the same indication.

Traditionally, blind listening tests in automobiles have not been done for technical and logistical reasons. In-situ blind comparisons between automobiles driven under road conditions are out of the question for obvious reasons. In-situ blind tests with the automobiles stationary are possible but rapid A/B blind comparisons (i.e. paired comparisons) are difficult without the listener having some knowledge of the devices and variables under test. As such, the test can no longer be considered blind. Furthermore, for blind or sighted in-situ tests randomization of variables known to influence judgment of sound, like program material and the automobiles themselves, is also impractical. Yet, without rapid A/B comparisons the discrimination and reliability of subject responses can be negatively impacted due to our limited acoustic memory.

With that in mind, to test just the different methodologies, a method for performing double-blind in-situ listening tests has been developed and is used as a benchmark for single stimulus comparison of listening methodologies. In this method a strong effort has been made to remove the non-acoustic feedback in the automobile interior. The interior is scented with a very strong scent disk to mask any new-car, old-car, diesel truck, smell that might influence the listener's opinion. The automobile seat is covered with a mat that removes any seat qualities from the listener's opinion. The listener is asked not to touch any part of the interior, but there is always a chance that that request will be violated during the time when a blinded listener is first situated inside the automobile. Therefore, the center console where a stick shift might be is covered with thick sheet of non-reflective material. The foot pedals are covered to remove evidence of a manual or automatic drive. And, the steering wheel is covered to change its feel. The listener is blinded in another room away from the vehicle. A complete blindfold is used to remove any presence of light and visual feedback. Care is taken not to cover the ears. The listener is then given a pair of headphones to wear. Pink noise is played over the headphones to remove any acoustic feedback while the listener is led to the vehicle, helped into the driver's seat, and the door closed. There is an intercom communication fed into the headphones. (Later in the test, the intercom is made available to the listener through a small portable speaker.) After the listener is in the automobile, the pink noise is interrupted and the listener is told to remove the headphones.

Trained listeners with verified good audiometric performance are always used for the lowest possible variability and highest reliability. The listeners use an interval scale to rate the sound quality of the system in terms of overall preference, timbre balance, spatial fidelity, and absence of distortion. The listeners use the PrefTest software, as illustrated before. In the blind in-situ tests, the administrator runs the software outside of the automobile and communicates with the listener through the intercom. The radio controls for the vehicle are also made remote to the administrator, who maintains a constant system volume and changes source material as instructed by the PrefTest software, which randomizes the playback. In listening test comparisons where it is not possible to remove or remote the radio controls outside of the vehicle, the administrator occupies the passenger seat and communicates directly with the listener. As a last test in the blind evaluation, the administrator varies the playback volume, and the listener is asked to rate the overall dynamic response of the system from low to high volume levels.

This method is a very reliable single-stimulus benchmark for static systems, and it is being used on a regular basis to perform competitive analyses of stationary automotive sound systems. The method is very consistent, repeatable with meaningful results, and is capable of discerning some subtle differences in a large set of very closely matched sound systems in vehicles considered to be in the same class. Yet a vast improvement in that ability to discern subtle differences could be made if rapid A/B comparisons were possible. And, too, making on-road evaluations is a must for a complete assessment of any automotive sound system performance.

Another method, which does allow rapid A/B/C/D double-blind comparisons to be done in an efficient and cost-effective way, is one that uses a high quality binaural record/playback system. This method is highly repeatable and allows excellent systematic control of nuisance variables known to influence listening tests.

A binaural dummy head (plus torso and legs) is placed in the exact listening position as a listener would be. Binaural recordings of the source material are made on DAT and edited on a PC as *.wav files. The *.wav files are played back through a digital audio card on the PC, processed through an external D-to-A Converter, and amplified through a headphone amp where the output is maintained at a constant level consistent with the listening level in the vehicle (85dBA). The analog conversion is done outside of the PC to reduce the noise that still exists on most PCs during *.wav file playback. The playback is listened to using Etymotic ER4s earphones, which, with a good seal in the ear canal, provide extremely linear full-band reproduction, at least 15dB of external sound isolation, and very high repeatability from one session to the next. PrefTest is used to evaluate the *.wav files. The source recordings are randomized and four recordings of a source made in four different cars are presented to the listener, who can rapidly switch between them by clicking on the appropriate GUI button. Recordings can also be made under road conditions and with varying volume settings. The recordings can then be synchronized and presented to the listener who then evaluates the sources under all the different conditions.

This method has proven to produce excellent results when evaluating the spectral aspects of a sound system, whether it is a home system or an automotive system. [6, 12, 13] However, there is a lack of bone conduction with the binaural playback method, which effects the absolute agreement of blind in-situ and binaural bass results. They do, however, agree in relative terms. If the bass is ranked higher for one vehicle over the other for a blind in-situ test, it will also be ranked higher for a binaural test, just not on the same scale. The most difficult point when using the binaural method is that binaural playback of recordings made with a stationary dummy head can prove to be less than accurate in reproducing the spatial aspects of the sound field. Lack of head movement and mismatches between the pinna of the listener and those of the dummy head can cause the results of sound stage and sound image evaluations to be confused when compared to the same results from a blind in-situ test. If recordings using pinna that are similar to the listeners' and recordings related to head movement could be made and synchronized to the listener's head movement, then the playback of the binaural recordings could more accurately reproduce the live experience and become more useful for spatial evaluations.

Another method, which some pilot experiments have shown has very good agreement between it and the blind in-situ method in all aspects of the sound field, is what is called the "placebo" method [14]. The placebo method could be an even more efficient, cost-effective, and controlled method of sound evaluation than the binaural. In the pilot experiments for the placebo method, as before, qualified listeners used PrefTest to evaluate a single automotive sound system in-situ, blind and sighted. A digital amplifier system was used in the vehicle's sound system. Three equalizations (EQs) were stored in the amplifier's memory. One was the target EQ for the system, and the other two had spectral aberrations added to them. The listener evaluated several sound sources with all three EQs, first blinded and then sighted. The sources and the EQs were randomized, and the listener performed the evaluation five different times for both the blinded and sighted. [12] However, only the data for the target EQ was used as an assessment of the sound quality and analyzed in comparison to the blind in-situ results – with good agreement. Part of the analysis was also to see if the listener was giving different scores to each of the EQs and being consistent in the rankings, and not just ranking them all the same or trying to scale the ranking in order to force an opinion onto the data.

The qualities that the placebo method has over other methods are that it factors out biases and opinions based non-acoustic aspects of the sound. Since the listener doesn't know which EQ is the target EQ, all of the EQs must be given an honest appraisal. And, the listener can do the evaluation in the actual vehicle and while driving on the road. In future experiments the assumption will be made that all systems for evaluation will have a CD or cassette player. The aberrations that were made with the digital amplifier will be made with a re-recording of the source material onto CD or cassette. So, no modifications would need to be made to the original equipment. There will also be more types of aberrations. Not just spectral differences, but small time delays and phase shifts will be added. The intention would be to create a set of sonic aberrations that could be expanded into a larger standardized set of aberrations which could keep growing and changing randomly. With a truly random set of sonic aberrations and enough repetitions to maintain listener reliability, a much better double-blind single stimulus evaluation should be achieved. The next thing wanting would be the ability to do a rapid paired comparison.

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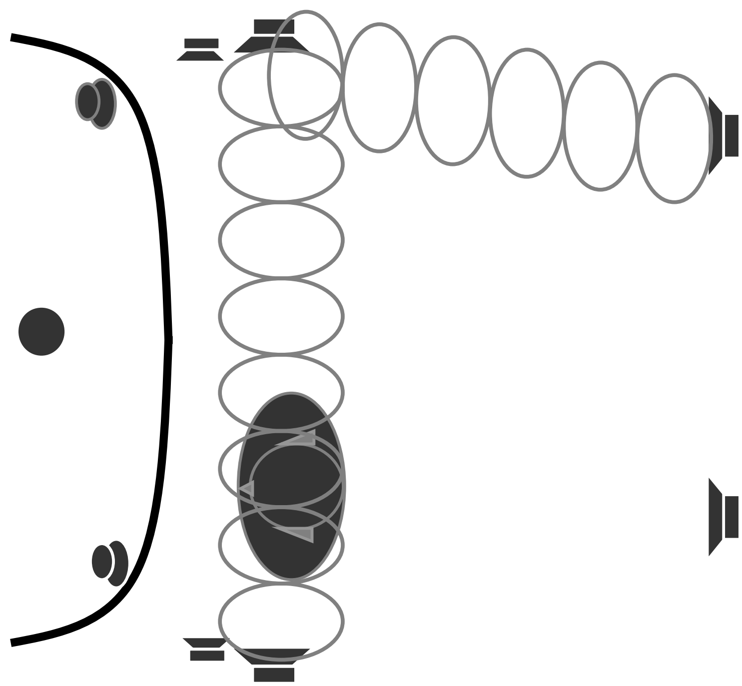
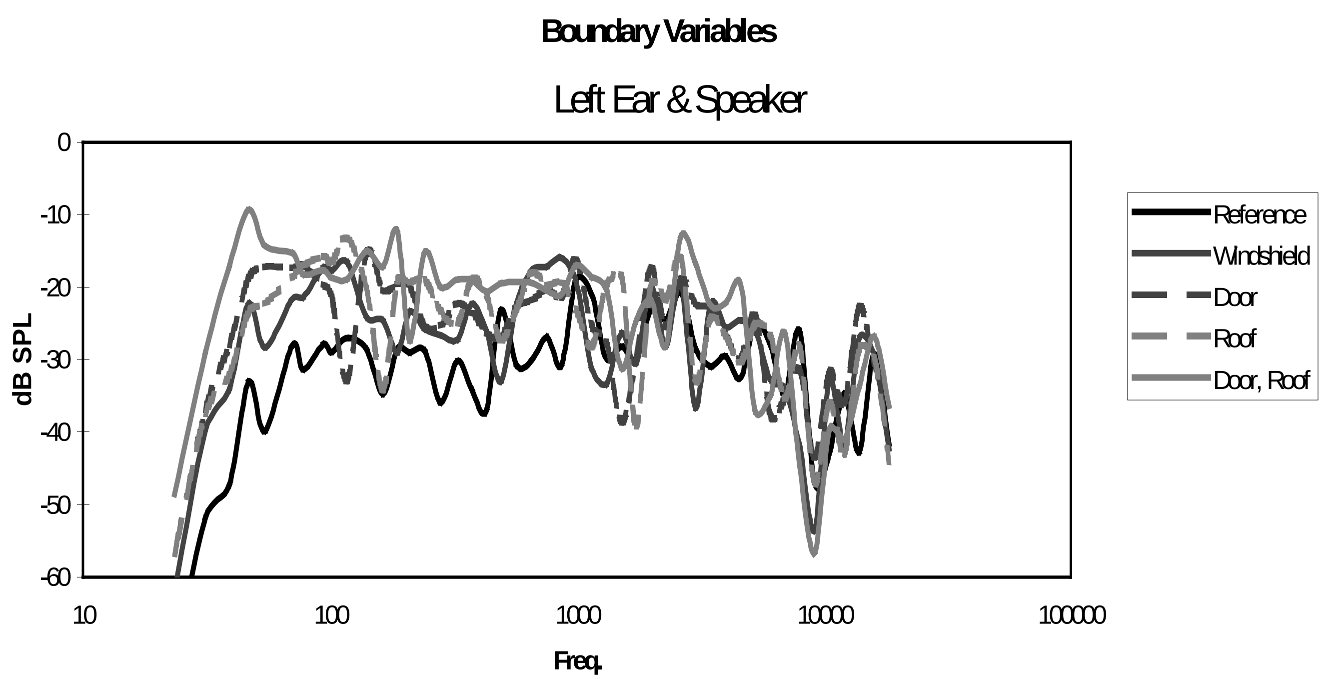


Illustration 1. Standing Waves



Graph 1. Cabin Resonances

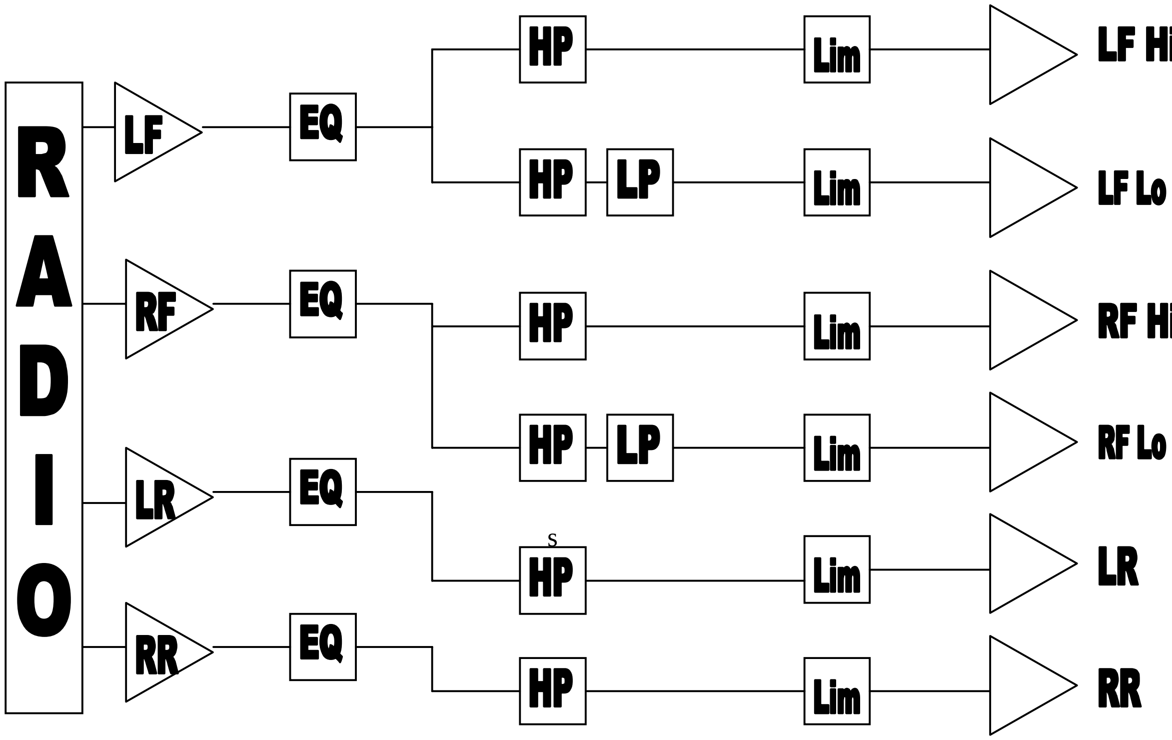
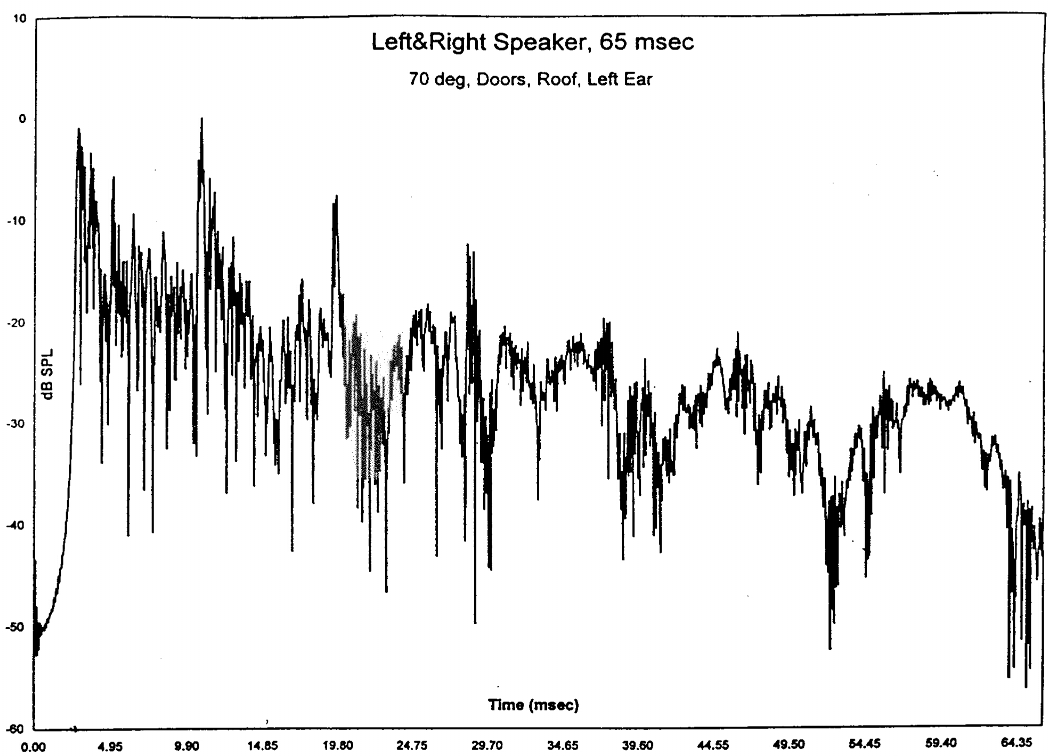
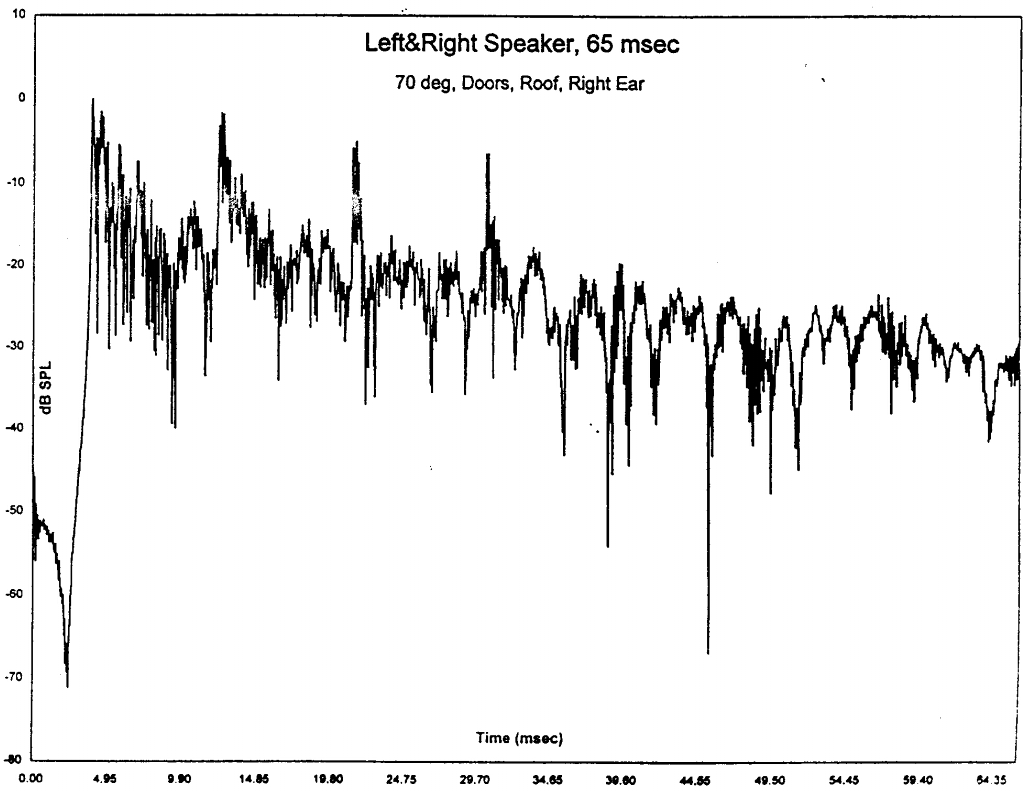


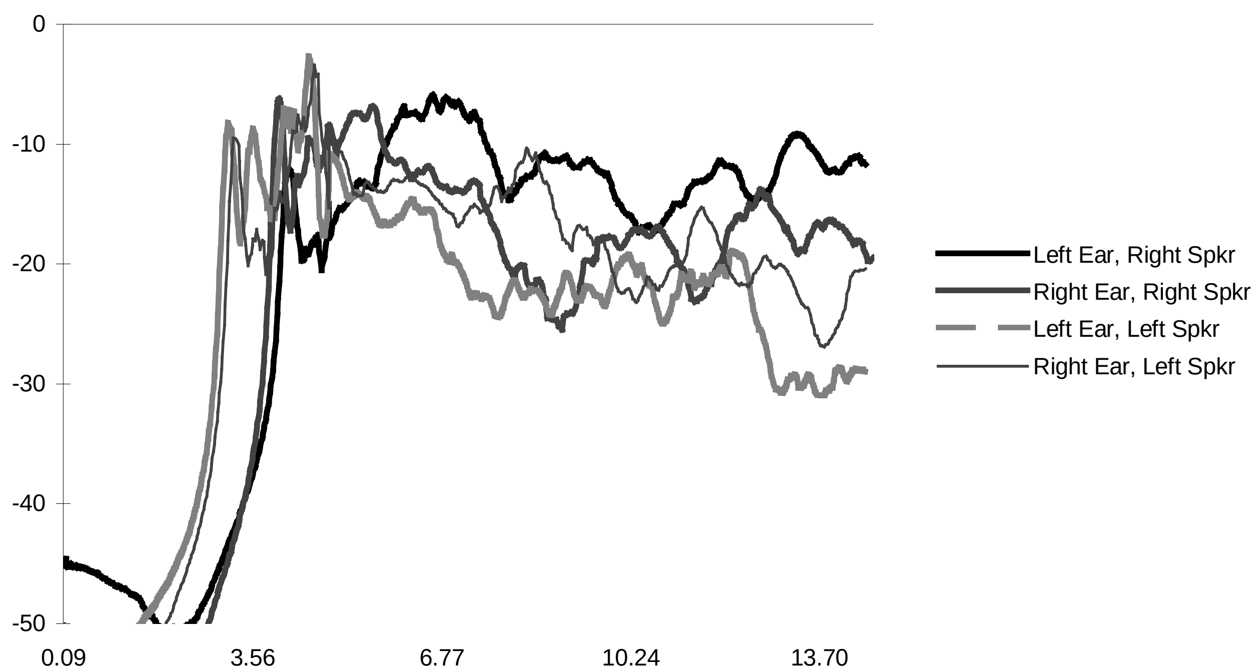
Illustration 2. Amplifier Block Diagram (Spectral)



Graph 2a. Time Domain: Left & Right Speaker, Left Ear



Graph 2b. Time Domain: Left & Right Speaker, Right Ear.



Graph 3. Time Domain: Crosstalk.

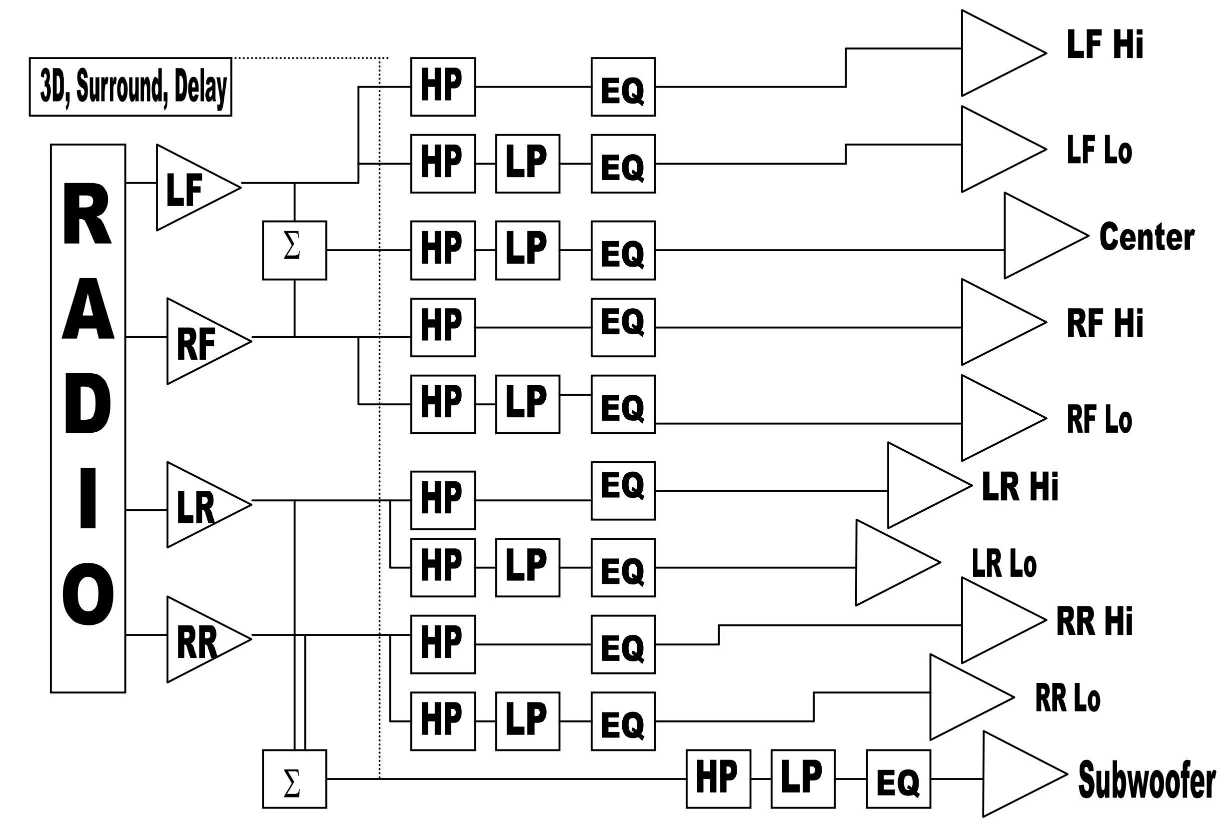


Illustration 3. Amplifier Block Diagram (Spatial & Temporal)

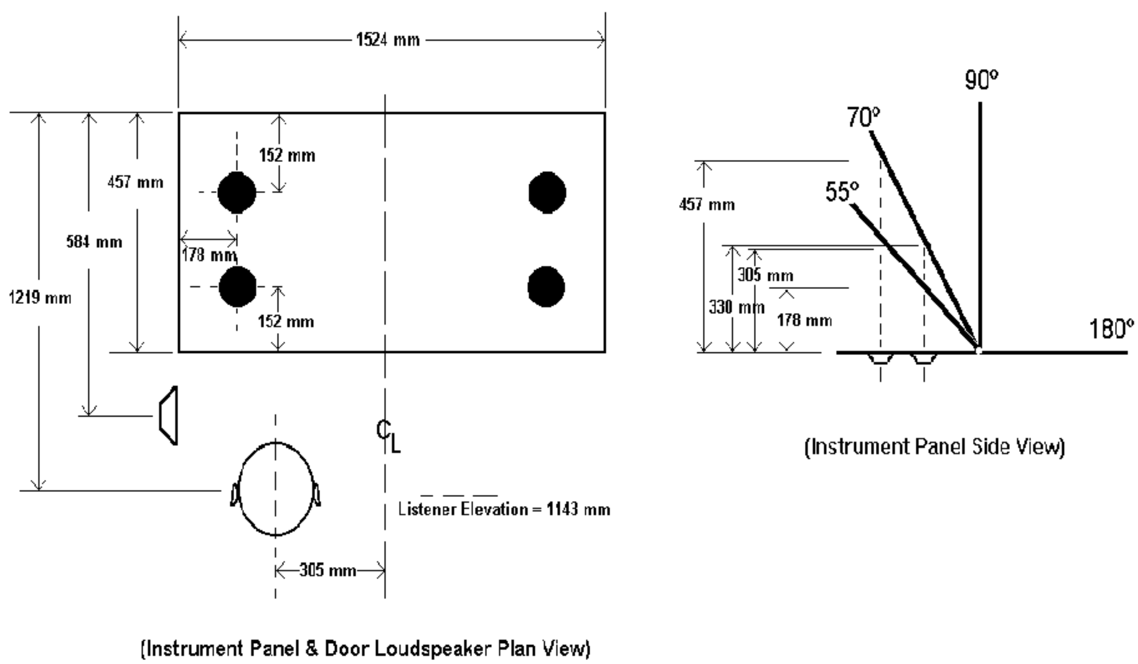
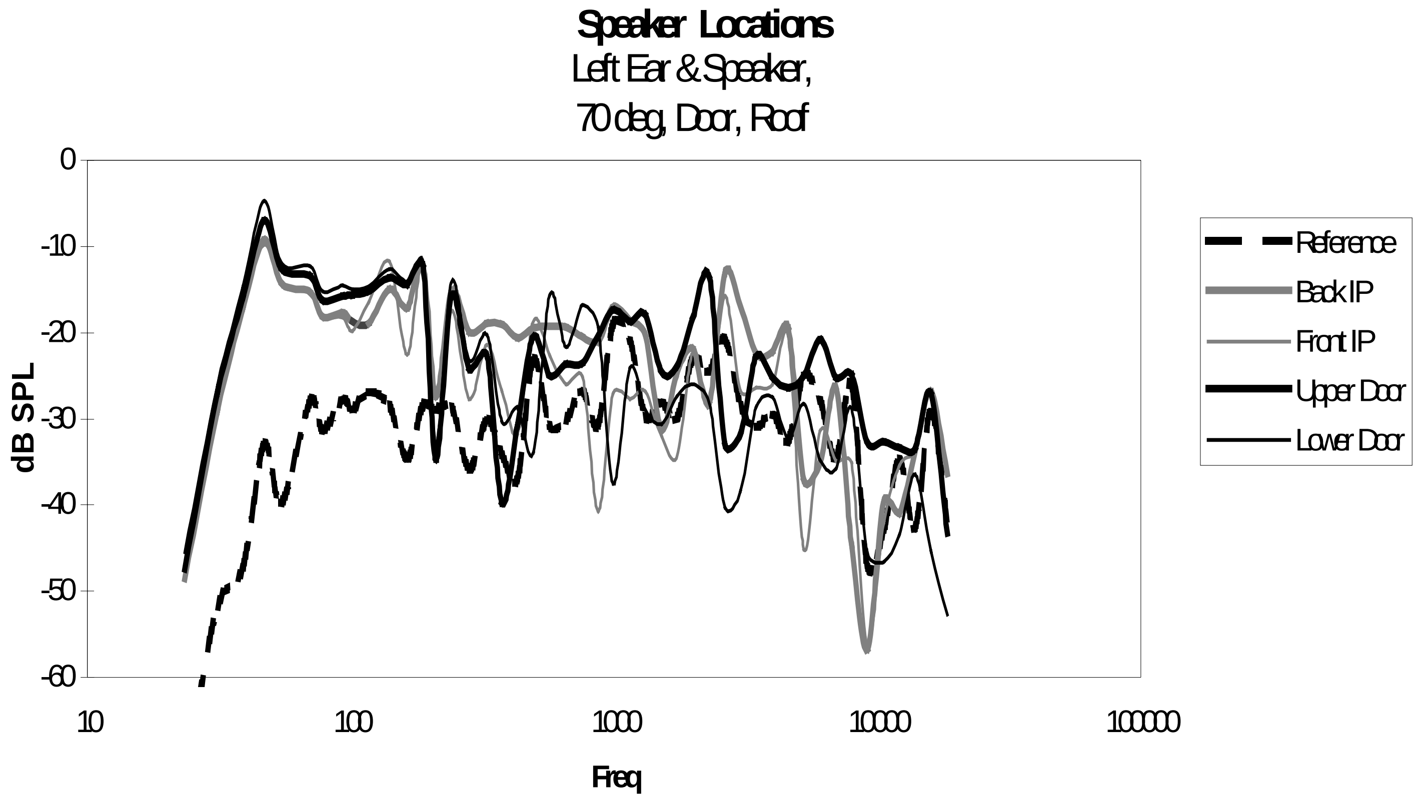
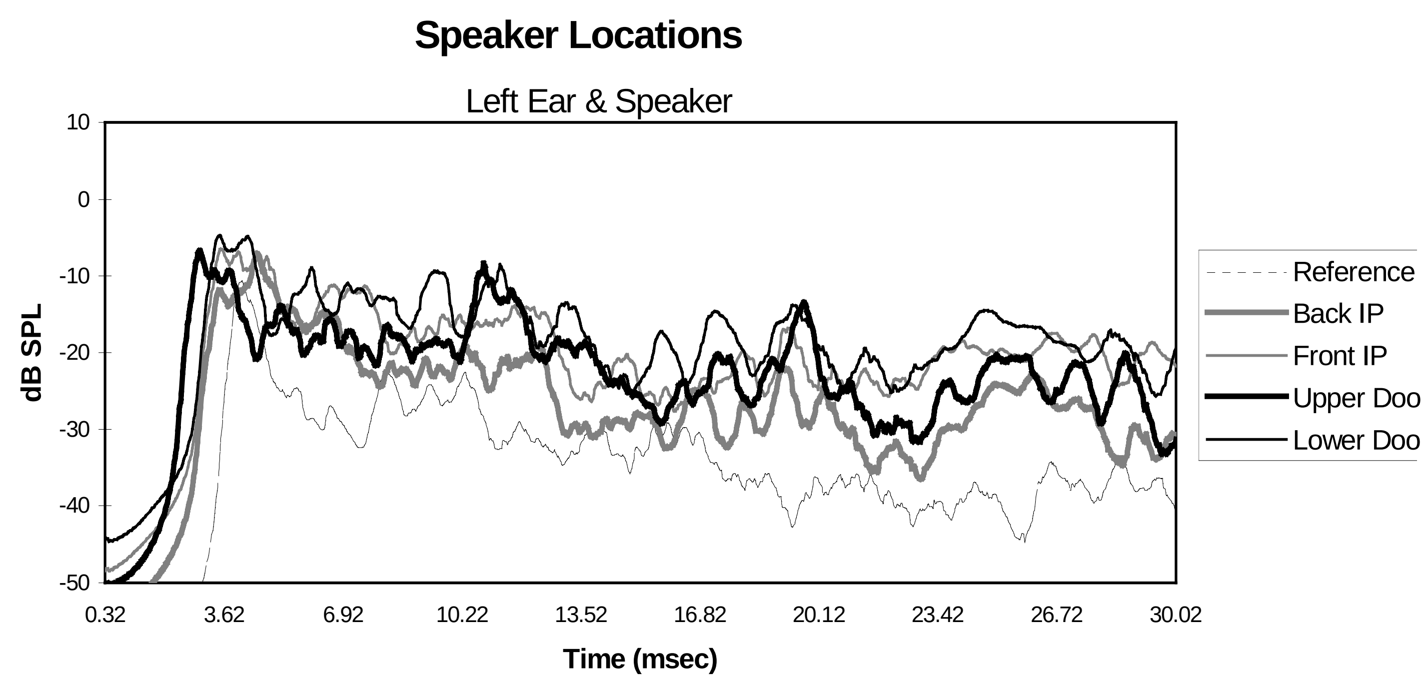


Illustration 4. Loudspeaker Placement Midranges



Graph 4. Midrange Placement. Frequency Domain.



Graph 5. Midrange Placement. Time Domain

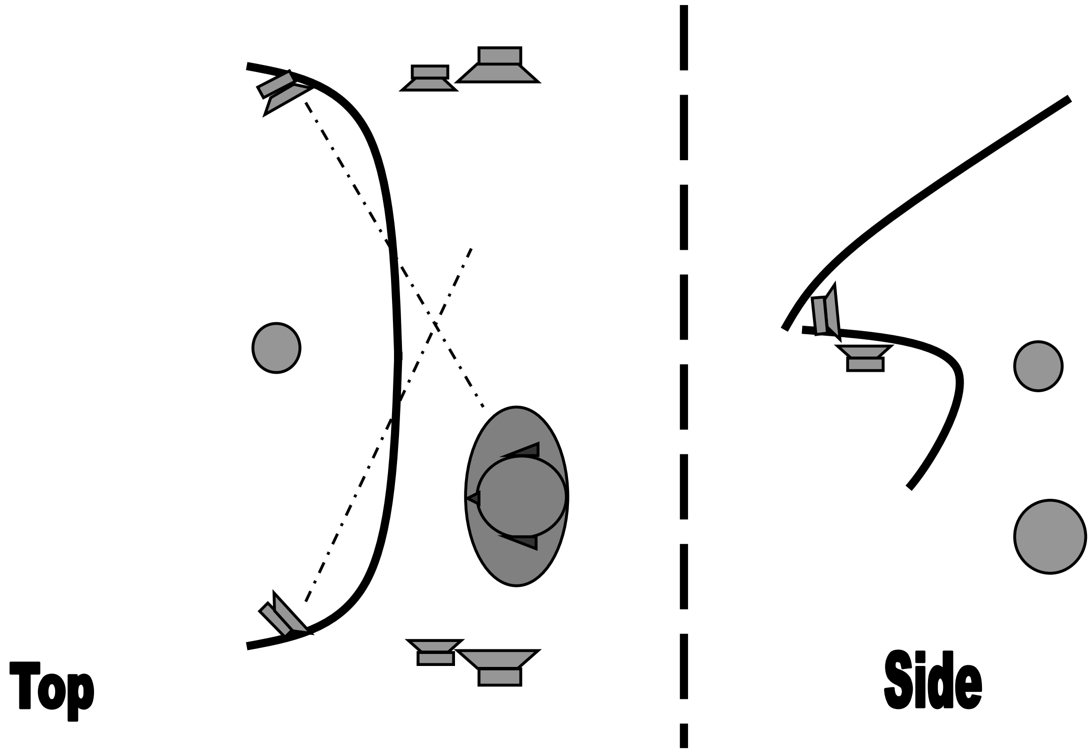


Illustration 5. Loudspeaker Placement: Tweeter and Upper Midranges.