Acoustical Optimization of Control Room 'A' at the McGill University Recording Studios

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ABSTRACT

The loudspeaker system and the room interface are the two main components in any listening environment. Research will be conducted focusing on the room component using Control Room 'A' of the McGill University Recording Studio in an attempt to optimize the monitoring situation. The sound field of the room will be broken down and analyzed in both time and frequency domains. The problem areas of the room will be identified and the surfaces altered by means of absorption, reflection and diffusion.

Résumé

Le systéme de haut-parleur et la salle sont les deux éléments principaux dans l'environnement d'ecoute. La mise au point de la recherche sera conduit sur la salle elle même en utilisant la salle de régie 'A' du studio d'enregistrement de l'Universitie McGill en essayant d'améliorer l'état du contrôle en studio. Le cycle de vie de son de la salle sera séparé et analysé en deux catégories: temps et fréquence. Les espaces problématiques de la salle seront identifiés et les surfaces modifiés par moyen d'absorption, réflexion et diffusion.

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1. INTRODUCTION

Any audio listening environment can be broken down into many components; the input (microphone feed or playback device), the preamplifier (mixing console), the power amplifiers, the loudspeakers and finally, the room interface.

An improvement in any of these components carries with it the consequence of further exposing any deficiencies that exist in other parts of the listening chain. Most often, attention is given to the electronic/electro-acoustic components rather than the acoustical component of the room.

It is certain that in the field of professional audio, the ways in which sound is recorded/reproduced will be ever-evolving as the technology is in a constant state of flux. However, the laws of physics cannot be changed so that the room acoustics will always be a constant problem that the engineer of the future will have to deal with, regardless of whatever "current" technology is employed.

This paper will address the problems inherent in the acoustical (room) component of the chain and will use Control Room 'A' of the McGill University Recording Studios as a type of case study in acoustical optimization. (Although it is realized that many of the ideas presented can also be applied to any control room as well as domestic listening rooms).

Control Room 'A' will be analyzed in accordance with the many acoustical variables that will be discussed beforehand. Specific ideas for improvement will be gleaned from this study and then implemented.

2. MONITORING PRACTICE

2.1 Monitor Placement

Before looking at the alteration of room surfaces as a means of acoustical optimization, a basic understanding of two aspects must first be given their due attention:

I: the behaviour of loudspeakers in a room

II: how the listener's position affects the perceived tonal balance, imaging, etc.

Perhaps the primary determinant in how any given loudspeaker will perform in a room enclosure is its placement within the room with respect to its main surfaces. Here, the focus is on the soffit-mounted, main monitor systems as opposed to free-standing, near-field monitors (which will be dealt with later in this paper) although certain principles may still apply.

When mounting the monitors in wall soffits, they must be as rigid as possible to discourage any mechanical vibrations transferred through the soffit by the loudspeaker. These vibrations can play havoc with imaging as the structure moves in response to low frequencies. This slight movement will modulate the high frequencies resulting in a type of "doppler" effect which blurs the transient detail and stereo imaging especially since the left and right side speakers move in different relation to each other. This excessive movement can also dull the "punchiness" of the bass as the structure decays at its own rate after being excited into motion.

Another danger is inherent in the fact that sound travels faster through solids (such as wood) than it does through air. If any mechanical vibrations are not damped, sound may be transmitted structurally via the ceiling, walls, and floor and actually arrive at the listener ahead of the direct sound. Taking precautionary measures to avoid such structure-borne sound is actually part of the specification for having a certified LEDE[1] control room design [1], [2].

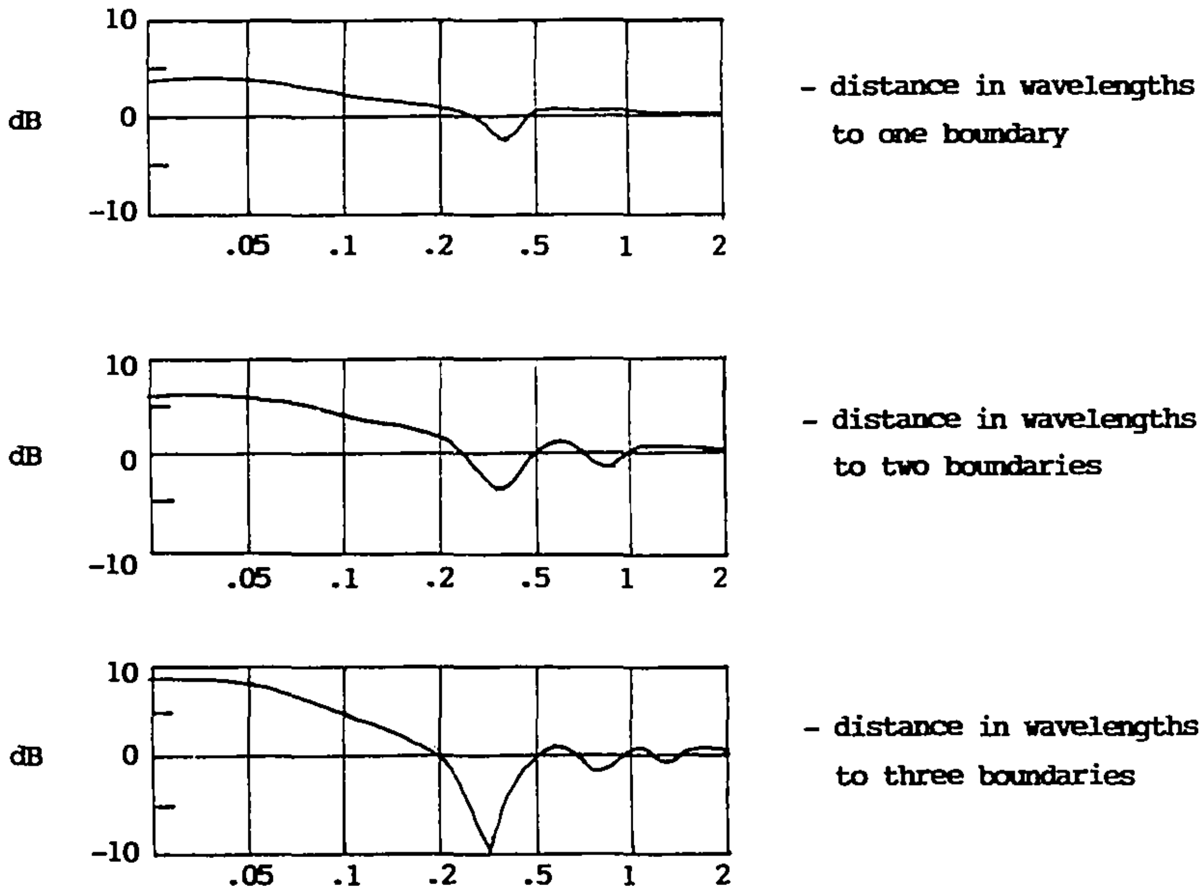
How the monitors are mounted affects the low frequency response in another important way. The most common means of installing main monitors is to have them flush-mounted. This is in an attempt to produce a folded "infinite baffle" effect. Other more appropriate acoustical terms might be "half-space" or "2π" mounting.

Installing loudspeakers in such a manner will exhibit an increase in the low frequency response by a theoretical 3dB over its free-field response. This is a doubling of the power output as the radiation pattern is hemispherical.

As any low frequency source (such as a loudspeaker) is moved closer to a rigid boundary, there will result a series of output reinforcements and cancellations. This is called the (SBIR) Speaker/Boundary Interference response [3]. Reinforcement occurs when the boundary to speaker distance

1) LEDE is a registered trademark of Syn-Aud-Con.

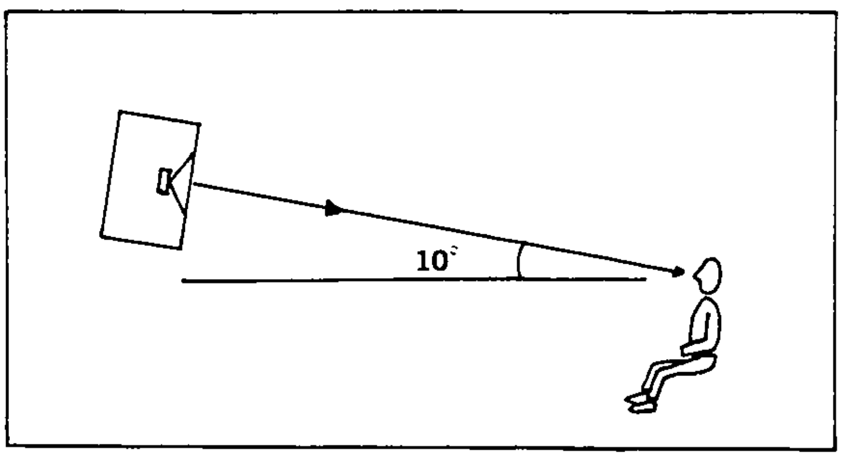
is less than a ¼ wavelength of a frequency. A degree of cancellation occurs when this distance is between ¼ and ½ wavelength (Fig. 2-1).



There appears to be some advantages in using surface boundaries to modify the (low end) frequency response of the system. By placing the monitors as close to the trihedral corners as possible, the output doubling effect would yield a 9dB increase. The advantages are twofold; the first being that all surfaces would be close enough to the loudspeakers to avoid any serious out-of-phase reflections at long wavelengths. The second advantage is that most of the resonant room modes can be excited into oscillation in such a corner. At any other place along a given surface, a fewer number of modes are excited yielding a less even frequency response. (A type of "all or nothing" philosophy applies here).

But perhaps what dictates the placement of monitors within a room is not so much the resultant low frequency response but the stereophonic imaging. Although conventional stereo reproduction (through two separate loudspeakers) can only convey a soundstage in the horizontal plane, there are certain accepted guidelines to follow for "vertical" imaging through monitor height placement.

Many studio designers dictate the monitor height by prescribing the optimal angle at which the loudspeaker is aimed at the listener [4], [5]. This optimum angle is roughly 10° (Fig 2-2).

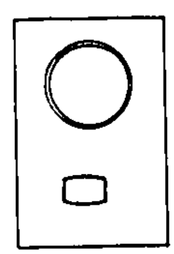


More often however, the monitor height is determined by the front wall configuration where a large control room window position can force a higher placement than is optimum. The only solution in such a case is to tilt the baffle angle downward to offset the extreme height.

Sometimes monitors that are situated too low can be obstructed by console-top speakers. Although such a low position may be advantageous in that it minimizes the amount of sound reflecting from the console surface.

Wrightson takes a different approach as he states that optimum monitor height is "concerned with visual, not sonic acceptability" [6]. There is some validity to the idea that the listener expects a certain sound event to be reproduced at a normal height. A solo cello being reproduced 4 feet overhead can be quite disconcerting.

Wrightson also points out that some studios in an effort to alleviate an abnormally high speaker placement will turn them upside down placing the high frequency drivers below the woofers and closer to ear level (Fig. 2-3). This may lead to some confusion as high frequency (overtones) above 7KHz are usually localized as being higher in elevation than low frequencies (even when reproduced by a single driver) [7].

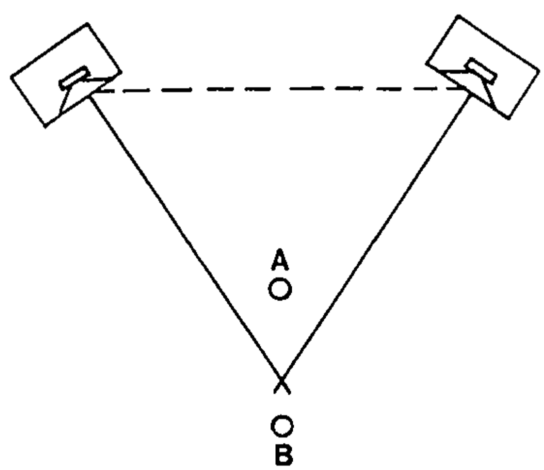


Regarding horizontal spacing, this may be limited again by the geometry of the control room. However, an average spacing is about 2.1 metres apart from the center of each monitor [8].

Ideally, such spacing should be determined by proper stereophonic reproduction although "true" stereo is rarely encountered in most commercial recording studios as they rely on multi-track, mono-panned sources to create a soundstage over two loudspeakers. (True stereo recordings employ stereophonic microphone techniques which encode a more complex ambient sound field which the listener uses to discern the direction, distance and size of a sound).

Another way to look at the issue of monitor spacing is to introduce the listener into the picture realizing that a given spacing may seem too high and wide if the listener is situated in close enough proximity.

Some designers [9], [10] recommend the ideal listening position (or so-called "sweet-spot") to be at the junction of the monitor axes when forming an equilateral triangle (Fig. 2-4).



Augspurger [11] states that the optimum listening spot lies within the triangle (point A in Fig. 2-4) about one metre in front of the junction. This is to avoid wasting any of the monitor coverage area to allow other engineers, producers, and musicians working near the console to still be within the triangle.

Toole [12] advocates having the listener placed behind the crossing (point B in Fig. 2-4) where doing so would help to stabilize the center image. In the event that the listener moves sideways, the sound from the loudspeaker opposite the direction of movement would increase in level as the listener becomes more on-axis. Sitting <u>within</u> the triangle would cause the center image to shift towards the direction of movement.

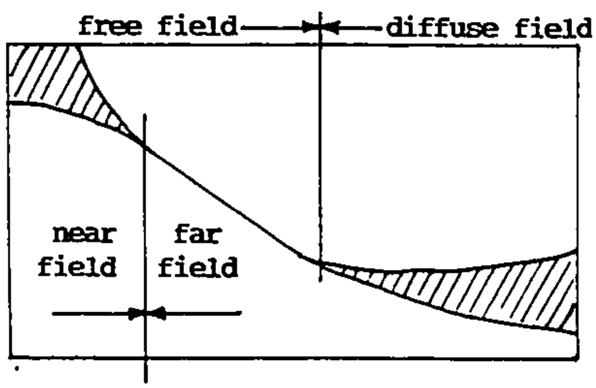
Of course, all this is dependent upon the directivity (polar response) of the chosen monitors. Position 'B' would not only bring the listener further out into the reverberant field of the room, but would also subject the listener to more of the off-axis sound of the monitor in the direction of movement. These situations could be made worse with loudspeakers that have a poor (uneven) off-axis response.

As well, a good, solid stereo image requires a high ratio of direct-to-reflected sound since it is the direct sound that carries the basic information needed for the auditory system to localize any image [13]. So by being further outside this direct sound zone (critical radius), the crucial directional information becomes more masked by the reflections.

2.2 Near-Field Monitors

An alternative to using large, soffit-mounted monitors is to listen on so-called "near-field" speakers (usually) mounted freely upon the mixing console meter bridge.

The term, "near-field" monitors, attributed to Ed Long of Calibration Standard Instruments, has become popular within the recording industry even though it is a misnomer. This is because the term has a precise and different meaning in the field of acoustics. It refers to one of two regions within the Free Field [14] (Fig. 2-5).



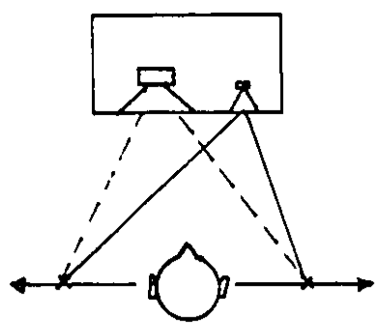
In this Free Field, the path between the sound source and receiver is devoid of any reflections. In the Far Field, the sound propagation follows the Inverse Square Law of attenuation whereby the level drops 6dB for each doubling of distance. However, once close enough to the origin of sound, this simple relationship breaks down and becomes unpredictable as the slope of the curve (in Fig. 2-5) no longer follows a straight line. The transition point that divides the two regions varies directly with the size of the sound radiator. In the case of a typical loudspeaker, the distance of this transition point is equal to the diameter of the largest transducer of the system. So in common practice, the engineer is listening well within the Far Field. The sound would otherwise be quite erratic if the loudspeaker was listened to within the actual Near Field. So perhaps a more appropriate term would be "Free Field" or "Close Field".

The main advantage in using near field monitors aside from minimizing the effect of the room, is that they can be a portable "reference" that the engineer can bring along to any unfamiliar recording studio. Of recent, small, self-powered monitors have begun to gain popularity as this setup can further avoid any "reference" doubts by being able to bring the power amplifier along with the loudspeaker all in one package.

The small size of nea field monitors also allows the engineer control over placement for the optimum or preferred stereo imaging.

This freedom can also lead to the common error of placing monitors on their sides upon the console meter bridge. Albeit, this is usually done to avoid obstructing the view over the console or casting an acoustic "shadow" over the main monitors.

However, most small monitors are designed with a vertical array placement of the individual drivers on the baffle. This yields the widest listening angle in the horizontal plane. When placed on their sides, the speakers exhibit lobing patterns. Any slight lateral movement of the listener will cause image shifting and apparent changes in timbre as the distance relative to each driver changes (Fig. 2-6). These effects are especially noticeable around the crossover frequency (typically around 2KHz). This will restrict lateral movement of the operator if a consistent reference is to be maintained.



Since the dispersion characteristics in the median plane is usually quite narrow, it is beneficial to maintain the proper vertical placement in order to minimize unwanted reflections off of the console surface. Improper placement on the side not only allows the wider of the dispersion angles to favour the direction of the reflective console top. The high frequency driver would also be closer to the console resulting in a stronger reflection/comb-filter pattern.

Another potential situation of frequency coloration may occur when positioning loudspeakers atop a console. In such a placement, the speaker would now be radiating into a (2π) or half-space environment resulting in a (theoretical) 3dB rise in the bass response due to the reflection from the solid angle. With small monitors, this bass-loading effect is most evident in the 100-200Hz octave band.

2.3 Electronic Equalization

As a last attempt to cure the room anomalies and achieve a smoother frequency response, studio owners may resort to inserting some type of electronic equalization into the monitoring signal path. This was once a very common procedure which has since been found to be in error for many reasons [15].

Firstly, any boosts that were employed in the EQ would in effect, reduce the headroom of the system in a non-linear fashion. As well, it would increase the risk of overloading the power amplifiers into clipping or even blow out a driver.

Any cuts used in the EQ would often be an attempt to reduce the effects of standing-wave room resonances. The misuse inherent here is that an equalizer can only alter the level of the direct sound at the resonating frequency whereas the offending resonance has an accompanying longer decay time which would remain despite the EQ changes. In addition, these peaks and nulls as caused by the standing waves are location dependent so that any overall equalization would be an overkill solution.

Furthermore, the equalization would alter the all-important direct sound from the loudspeakers towards the characteristics of the EQ curve. It is part of the "Precedence Effect" that the auditory system puts more emphasis on this direct sound for localization, source recognition and intelligibility. Therefore, it is important (and possible) to choose a loudspeaker that has a relatively smooth anechoic frequency response so that given a poor-sounding control room, the direct sound could at least be trusted and left intact. But by using electronic equalization, a distorted sound would be fed into the same unevenly resonating room thus compounding the problem altogether.

3. SOUND FIELD IN AN ENCLOSED SPACE

In attempting to understand so complex a situation as the propagation of sound in an enclosed space, it is best to divide it into 2 different categories:

I: temporal domain
II: spectral domain

3.1 Temporal Domain

The temporal domain can be further subdivided into 3 categories:

direct sound ___ early sound ___ reverberant sound

The direct sound is the first arrival at the listener's ears as it follows the "direct" and shortest route thereby not being subject to any modification by room boundaries. Very early reflections (of around 1ms.) caused by diffraction effects from any cabinet surface protuberances (and edges) should be considered part of this direct sound category as they radiate from a part of the sound source itself and are independent of the surrounding environment [16].

In terms of the perceived sound field, this direct sound is all-important due to the nature of the hearing mechanism's dependence upon the first sound arrival [17] (precedence effect) in localization and timbral recognition. Therefore, it is important to select a loudspeaker system that has a flat pressure amplitude (free-field, anechoic) response, negligible time domain and harmonic distortions. Accomplishing that, one of the variables in the complex listening environment can be considered eliminated as it would act as a constant with which to work around.

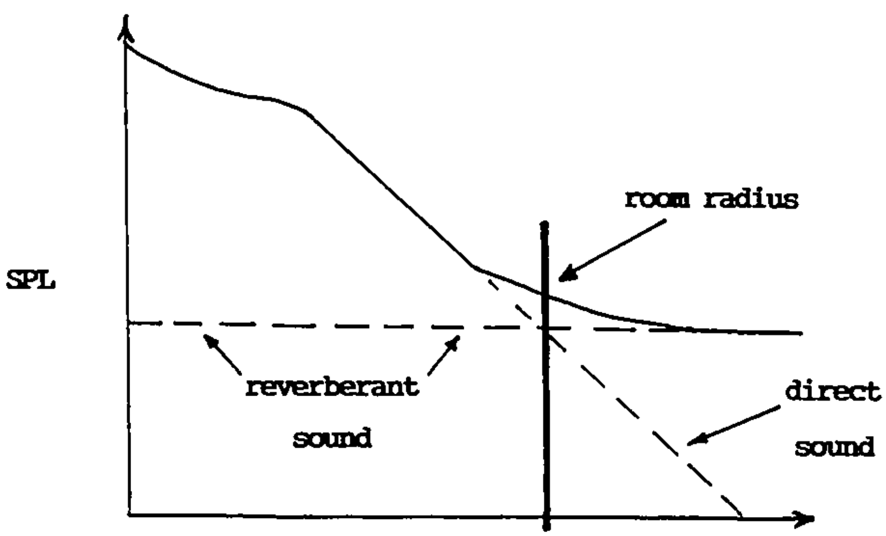
Following the direct sound in the range of about 2 - 40ms are the early reflections generated by the adjacent boundaries and console surface. These reflections are significant in that they have an effect on the sense of spaciousness and can distort the spectral balance. Many studio design concepts revolve around the control of this part of the soundfield [18], [19], [20], [21].

The audibility of these reflections depends upon their point of origin. Reflections coming from the same general direction as the direct sound will be masked unless they measure 5 - 10dB louder than the direct sound. They are more noticeable if they originate far enough away from the loudspeaker. Of course, program type can also have an effect on their audibility where impulsive music (ex. pizzicato, staccato) will expose these reflections to a greater degree.

Finally, a reverberant soundfield is set up after the original sound has undergone many reflections. It can be confusing to think of a "reverberant" soundfield here as this term is usually associated with larger rooms than the typical control room. The difference is that the reflections that comprise the reverberant field are too quickly absorbed with each boundary impact to become audible as a reverberant decay. (Typical RT60 values of control rooms range from 0.2 to 0.4 seconds). However, this sound field affects the total perceived quality and clarity as does a true reverb field. The only difference is that it does not become increasingly diffuse as is often thought. Instead, the sound becomes increasingly ordered into a well-defined spatial pattern of acoustic energy distributed across the room. (This effect is most evident at low frequencies typically below 400Hz). This concentration of sound energy within narrow frequency bands with independent decay times is due to the normal modes of oscillation in the room.

The relative importance of any 3 of the sound field categories to the perceived quality of sound is dependent upon the position of the listener relative to the direct sound source and its room boundaries.

A useful concept in acoustics is the idea of the "room radius". This is the radius/distance surrounding the sound source where the level of the direct sound is equal to the level of the reverberant sound (Fig. 3-1).



The room radius is a function of the RT60, absorption characteristics, and volume of a room as derived by the equation [22];

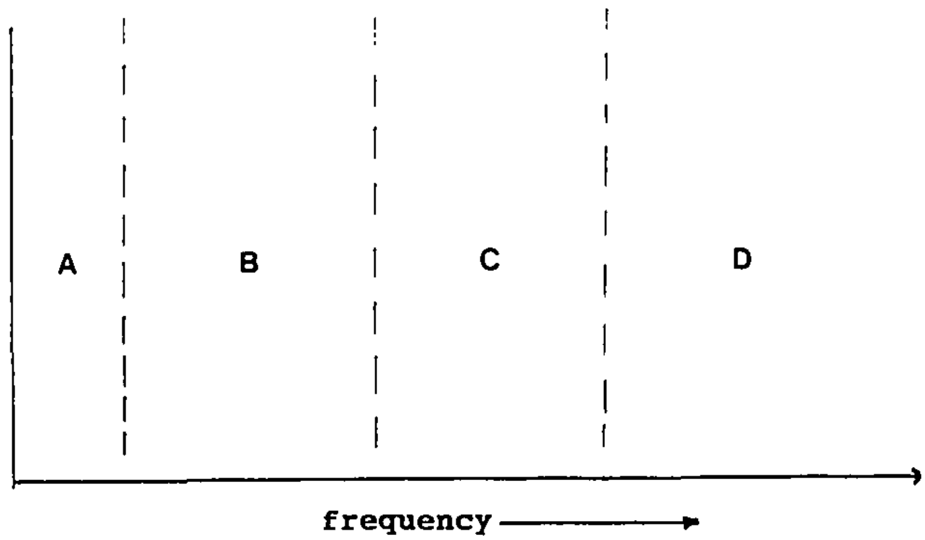
$$\text{room radius (m)} = 0.057 \frac{\text{room volume (m}^3\text{)}}{\text{reverb time (sec.)}}$$

It should be understood that this equation assumes a perfectly omnidirectional sound source and receiver as well as a homogeneous room in terms of its reverberation characteristics. In practice, it should really take into account the variable directivity of the sound source and receiver with respect to frequency. This is because any complex radiator such as a loudspeaker or musical instrument will usually have a gradually narrower radiation angle at high frequencies thereby feeding less sound into the room relative to its axial direction. This would translate into a larger room radius value than for lower frequencies; in other words, the higher frequencies would appear to have more "reach".

A typical value for the room radius in a control room is about 1 metre for 1KHz, and longer for higher frequencies. This means that when listening to the main monitors, the engineer is easily situated within the reverberant field.

3.2 Spectral Domain

Another way of simplifying the problem of room acoustics is to break it down into spectral zones. Everest [23] suggests the division be into 4 wavelength dependent areas (Fig. 3-2).



The first division occurs at a point dependent upon the longest dimension of the room (which may be taken as the diagonal). It is derived from the formula;

$$\frac{c}{2L} = \frac{344}{2L} = \frac{172}{L} = f \text{ (Hz)}$$

c = speed of sound

L = longest dimension (m)

Below this point, there can be no modal reinforcement of the low frequencies in this region and therefore there is inefficient acoustic coupling.

The next region (B) is governed by Wave acoustical theory; region C is characterized by mid-frequency diffusion and diffraction effects; region D follows Ray acoustical theory.

The main difference in approach is between Wave acoustics and Ray acoustics.

Ray or Geometrical acoustics follow the assumption that sound travels in a manner analogous to light rays. Here, the conditions are simple in that sound travels in rays until it strikes a boundary where part of it is absorbed, and part of it reflected at an angle equal to the angle of incident arrival. This assumes that the surfaces are large compared to the wavelength of impinging sound.

In contrast, Wave (or Physical) acoustics deal with sound propagation as spherical wave fronts. In a room enclosure, this region is characterized by standing waves.

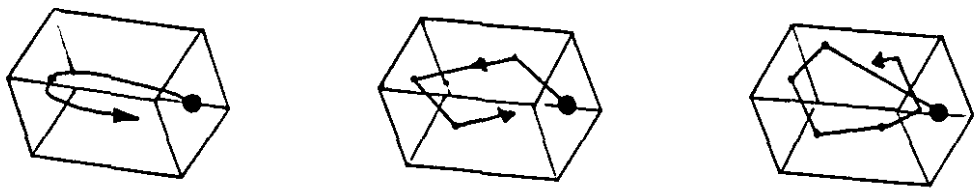
The crossover point from wave to geometrical frequencies can be derived from [24];

$$f_c = \frac{3c}{d} \qquad \begin{array}{l} c = \text{velocity of sound} \\ d = \text{smallest room dimension} \end{array}$$

Below this frequency (f_c), "the air in a room should be treated as an assemblage of resonators, normal modes of vibration which are selectively excited, maintained, and damped out under the influence of a number of controlling factors" [25].

There are theoretically an infinite number of these modes, but as frequency is increased above f_c, these modes become so closely spaced that individual resonances are undetectable as the modes form a continuum. (Of course, this transformation takes place only gradually above and below the cutoff frequency).

The 3 types of modes are depicted in Figure 3-3.



Axial Tangential Oblique

Axial modes are the simplest having 2 waves travelling in opposite directions as reflected between 2 parallel surfaces. So there are 3 different sets of axial modes corresponding to the 3 main axes of a rectangular room.

Tangential modes consist of 4 travelling waves reflected between 4 surfaces resulting in 2 fundamental modes. These have ½ the energy of axial modes since they travel a further distance and are subject to more absorption losses at each additional surface.

Oblique modes are the least prominent having only ¼ the energy of axial modes as they involve 8 traveling waves reflected between all 6 surfaces.

The 3 main variables that determine the behaviour of standing waves in a room are:

1) Dimensions
2) Proportions
3) Shape

1) The dimensions of a room determine the frequency of resonance between each pair of the 6 surfaces. The smaller the dimension, the higher its fundamental mode. The overall volume of the room defines the number of modes per frequency band. Therefore, small rooms suffer from wide and irregular frequency spacing of modes resulting in a ragged low-end room response. As well, the smaller the room, the larger will be the portion of the spectrum dominated by these resonances.

2) The proportions (or dimension ratio) determine the spatial distribution of the modes. A poorly chosen ratio would result in a "piling up" of resonances in certain areas of the room.

There have been many suggestions for the ideal room proportions but they may be overestimating the influence of proportions and therefore, oversimplifying the problem of poor room response.

All in all, it is not certain whether there are any "optimal" dimension ratios, but only that there exist "bad" ratios. For example, a room of cubical shape should be avoided as well as any ratios that are even multiples such as a 1:2:4 ratio.

Bonello [26] suggests that the optimum dimension ratio also depends on room volume. Gilford [27] simplifies the whole situation by pointing out how the axial modes are the most dominant and so any choice of proportions should be made by predicting their behaviour within the chosen dimension.

3) Adjusting the shape of a room by sloping its boundaries is another option that is often exercised in an attempt to improve the diffusion of the normal modes [25]. Avoiding any parallelism has a considerable effect on the modal structure [28]. Although the magnitude of the pressure variations remains about the same, the frequencies at which the standing waves are set up are changed as well as the nodal line patterns. However, the results of these changes are difficult to predict.

Such splaying of the walls certainly improves geometrical frequency behaviour by discouraging flutter echoes as well as redirecting early reflections if need be. Regardless, the modes remain, yet they now form an asymmetrical pattern across the room which may work against any efforts to attain a uniform sound field with stable low frequency imaging.

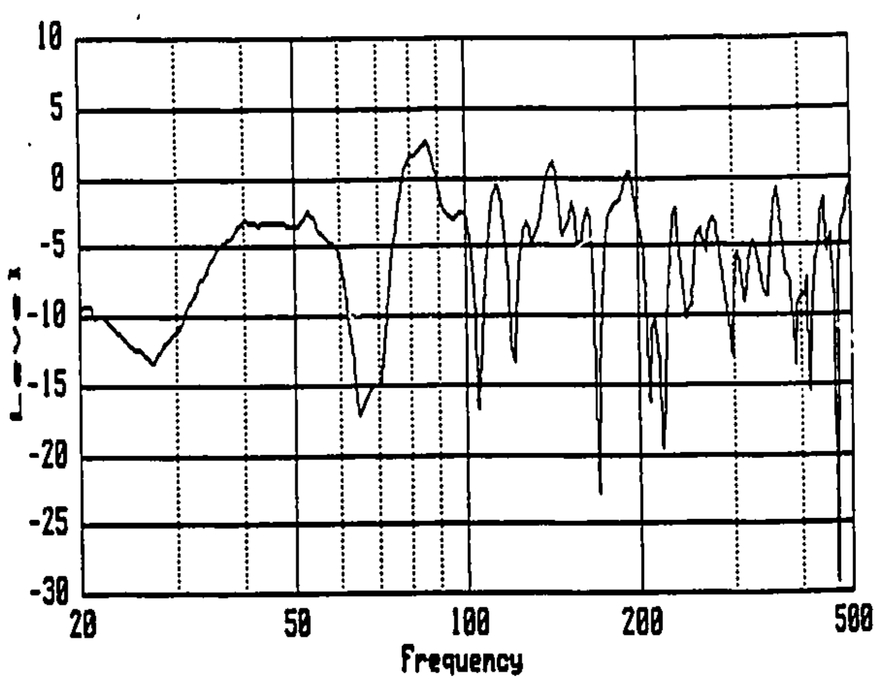
It should be noted that these standing waves cannot be eliminated without removing the reverberation itself. They are part of the acoustical signature of a room, good or bad. Without them, the room would sound lifeless, dry, not unlike listening to music outdoors or in an anechoic chamber.

Instead, attempts should be made to improve the diffusion and dampen the resonance energy through changes in room geometry and application of absorption and/or diffusion materials.

Gilford [29] lists 5 factors that determine whether a normal mode will contribute to audible colorations:

1) bandwidth of the mode
2) degree of excitation of the mode
3) its separation from strongly-excited neighbouring modes
4) the position of the sound source and receiver with respect to standing waves
5) spectral content of sound source

Looking at an example (Fig. 3-4) of a swept-sine tone frequency response of an average sized room, it is surprising that the room doesn't sound as bad as it looks on paper. This is because most of the resonances have a high Q and a small bandwidth (usually about 5Hz).



The degree of excitation of any mode should rise linearly with the output level of the sound source but it seems that the auditory system is more sensitive to the effect of the modes when excited by a louder sound. This is because the longer decay attached to the original sound by the resonance becomes more apparent.

Modes have a greater tendency to be driven into oscillation by a steady state sound source input. However, this is an uncommon situation as music and speech signals are usually more or less transient and discontinuous in nature. Although it has been argued [30], [31] that this is the reason that standing waves aren't so menacing as they appear to be in measurements, the decay portion still lingers after the mode excitation, contributing to a reduction in bass "tightness" and "punchiness".

Room mode induced colorations may also become audible owing to poorly chosen room proportions which result in coincident or near-coincident buildups at certain frequencies.

The position of the loudspeakers can also have an effect on the number of modes put into action with corner placement exciting the greatest number of modes.

The chosen listening area can also have an influence on the degree of audibility. Care should be taken to avoid a position such as the exact center of a room where the greatest number of nulls in the standing wave pattern can be found.

Finally, it should be realized that the spectral content of the program source itself can make the modes more apparent especially when there is an abundance of energy below 300Hz. However, this is a variable outside the control of acoustic design.

4. TOOLS OF ACOUSTIC DESIGN

Whenever a sound comes into contact with any boundary, one or any combination of three events can occur;

1) part of the energy is absorbed (a)
2) part of the energy is reflected (r)
3) part of the energy is transmitted (t) through the boundary

This event can be expressed by a simple equation;

$$1 = a + r + t$$

This paper will not concern the 3rd classification of transmission which has little to do with sound quality within a room. Also, the second classification of reflection can be subdivided into specular reflection and diffusion.

Altogether, the 3 above conditions comprise the 3 main acoustical "tools" with which to repair the listening environment; absorption, reflection and diffusion.

4.1 Absorption

The reduction in sound pressure level of an incident sound wave when in contact with an absorptive material is given by the general equation; [32]

$$\text{SPL reduction (dB)} = 10\log(1 - a)\text{dB}$$

a = absorption coefficient

So for a boundary with 90% absorptivity, a reduction of only 10dB will occur. Further reduction values are illustrated in the table below.

Absorption coefficient	Attenuation (-dB)
0.5	3.0
0.6	4.0
0.7	5.0
0.8	7.0
0.9	10.0

The effectiveness of any absorber is highly frequency dependent with no universal absorber existing which is equally effective across the whole audible spectrum.

High frequency absorbers are principally of the porous type. These are probably the most common type of absorbing material and easily the most effective. Curtains, carpets, fiberglass and foams are commonly used.

Absorption occurs as sound enters the material causing the fibers to vibrate yielding frictional losses as the sound energy is converted into heat.

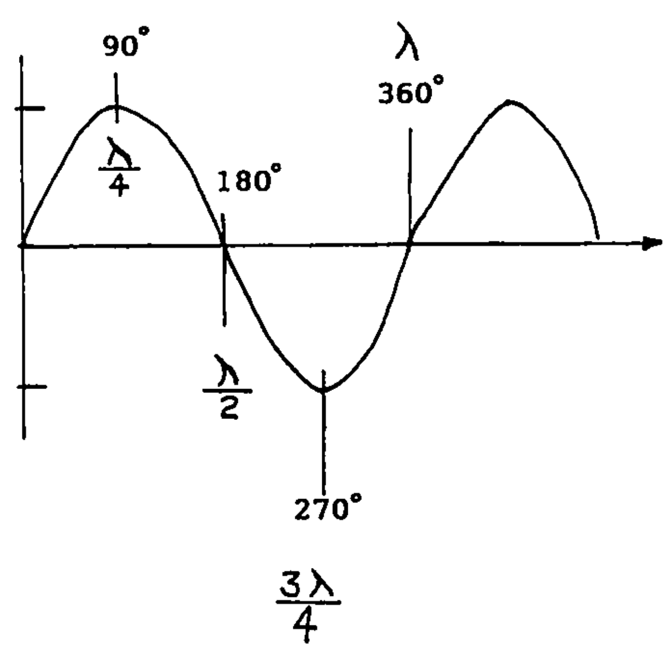
The 2 properties of density and thickness are the chief determinants in any porous material's effectiveness in absorbing sound energy.

Density is the measure of how tightly packed the fibers are in a porous material. The effect is relatively minimal for a substantial increase in density. For semirigid fiberglass, a doubling of density only results in a slight improvement in the absorption characteristic below 250Hz.

Extremely low-density materials will have the fibers spaced so widely that absorption will be poor. In converse, high-density materials make poor absorbers as penetration is discouraged where the material will actually become quite reflective at high frequencies.

Increasing the thickness of a porous material has a substantial effect as the absorption coefficient increases and its effective range extends downward to a lower frequency. However, this increase becomes proportionately smaller in relation to the increase in thickness.

For any given frequency, there is an optimum thickness for which any further increase beyond this will not result in a significant improvement. This optimum thickness is rated at ¼ wavelength of a chosen frequency. This corresponds to 90° on the cycle of the sound wave where particle velocity is at a maximum (Fig. 4-1).



Probably the most common pre-fabricated porous absorber (besides carpet) is SONEX, manufactured by Illbruck and marketed by Alpha Audio Acoustics in the United States. It is made of open-cell urethane foam and comes in thicknesses of 5, 7.5 and 11 centimetres. The sheets have wedge-shaped contours which provide a type of graduated flow resistance to sound impinging upon it. The manufacturer also claims that such a shape presents an effective surface area 450% greater than a flat surface absorber of the same dimensions. The Azonic company also produce a similar foam absorber using pyramid-shaped contours instead.

The advantage of using this type of pre-fabricated foam absorber is that they are fire-resistant, and their performance variables are known whereas self-made absorbers may not be so predictable or dependable.

In practice, the absorption of mid to high frequencies is often unavoidable as common furnishings such as carpets and curtains will behave as effective aborbers in this frequency range. Achieving adequate low frequency absorption to supplement or balance this existing high frequency loss is a difficult task because of the longer wavelengths involved. However, it can be accomplished by means of a totally different system.

This basic "system" used in recording studios and performance venues is built upon the principle of a diaphragm.

They are often called "panel" or "membrane" absorbers and consist of a flat panel mounted at a solid wall separated by an airspace. The mass of the flexible panel and the springiness of the enclosed air behind constitute a simple, mechanical resonant system whose frequency is given by [33];

$$f\ (Hz) = \frac{63}{\sqrt{MD}}$$

D = enclosed air space depth (cm.)

M = surface density of panel (Kg/m^2)

When a sound waves comes into contact with the panel, it (the panel) is excited into resonance absorbing sound energy through internal dissipation of the panel itself. If the panel is underdamped, the resonant vibration may become audible as a "hangover" at, and around its natural frequency of resonance.

Common building materials such as wood panels, gypsum boards, suspended plaster ceilings, plastic, window glass, and hollow doors all could behave in some way as diaphragmatic absorbers.

To increase the effectiveness of the panel absorbers, a variation of the Helmholtz resonator principle can be used [34], [35].

There are 2 basic types of these cavity absorbers; the perforated panel, and the slat (or slot) absorber. Both variations feature a small cavity (hole or slot) connected to a larger one (the air space behind).

Lowering the effective absorbing frequency is accomplished by using a thicker panel or a deeper air space. But probably the most practical and economical way is to decrease the "open-area ratio" by making the cavity (holes/slots) smaller or by simply using less holes.

Broadening the bandwidth of absorption may be achieved by following any of the 4 methods below;

1) using holes (or slots) of varying sizes across the panel

2) irregularly-spaced holes (slots)

3) varying the depth of the air space by mounting the panel at sloping angle to the rigid back wall

4) placing a porous absorber material in air space behind panel

Fiberglass sheets are the best absorbent to apply for the last method (#4). Applying it directly to the back of the panel is the most effective placement since sound particle velocity is greatest at or near the exit of the cavities.

One of the most common pre-fabricated LF absorber units on the market is called the TUBE TRAP marketed by ASC. Basically, it is a cylindrical broadband absorber that comes in full-round, ½-round, and ¼-round shapes. The manufacturer recommends corner placement where most high pressure zones for room modes are sure to be found. The convex shape also promotes mid-frequency sound scattering (diffusion) by rotating the side with the plastic reflector built in.

Another manufacturer of portable modular LF absorbers is Black Box Acoustic Conditioning Systems from the U.K. [36]. These are simply membrane absorbers that are custom-designed for any studio needs.

Another commercial LF absorber is based on the Helmholtz Resonator principle. It is called "SOUNDBLOX", and is manufactured by the Proudfoot company of Greenwich, Conn.. It is simply a hollow concrete block with 2 narrow slits leading to a fiberglass-lined interior cavity. The absorber operates only in a one octave range of 120 - 240Hz.

The most exotic of LF absorbers stems from one of the many inventions that came out of the RCA laboratories by H.F. Olson [37]. This is an electronic absorber called the "SHADOW ACTIVE LF ACOUSTIC CONTROLLER" manufactured by Phantom Acoustics [38].

It resembles the TUBE TRAP in appearance but is quite a different system altogether. It is an electronic transducer system consisting of a microphone, loudspeaker and amplifier using a negative feedback system to reduce the sound pressure in the vicinity of the microphone. The "Shadow" operates in the range of 30 - 200Hz as claimed by the manufacturer.

4.2 Diffusion

Another useful means of manipulating sound in a room is through diffusion.

A sound field is said to be perfectly diffuse if there is a uniform distribution of sound energy throughout, and the directions of propagation are random [39]. Of course, this is an ideal situation which is virtually unattainable.

A certain amount of diffusion can be attained by the simple presence of various objects, equipment and people in a room. This diffusion occurs through the diffraction phenomenon whereby the normal propagation pattern of a sound ray is "bent" resulting in more random directions.

Surfaces that promote diffusion over specular reflections can improve the perceived quality of sound in a room. There are 2 basic types of diffusion; temporal and spatial.

A diffusing surface can return or reflect incident sound into the room at different times producing a rich, dense, nonuniform comb-filter pattern which is perceived as a pleasant ambience. A planar surface produces a deep, regular comb-filter pattern corresponding to multiples of the lowest frequency of interference.

A diffusing surface also returns or scatters sound spatially by breaking up the incident sound and spreading it over an area both vertically and horizontally depending upon the shape and orientation of the diffusing object.

In comparison, a specular reflection from a planar surface would only redirect sound in one direction following ray acoustics theory. This single, strong reflection would be more likely to be perceived than the spatially spread out reflections (with much lower individual energy) as produced by a diffusive surface.

From this it can be seen that the overall advantage of sound diffusion is that it does not lessen the total energy in the room but rather, it increases the number of reflections per unit time thereby lessening the intensity of the individual reflections . In effect, lessening the interference potency between the direct sound and its associated reflections. This efficient redistribution of sound energy by diffusion is an important factor in a studio monitoring environment. The reliance on too much absorption on room surfaces would "drain" the useful sound energy emitted from the loudspeaker system. But if used properly, diffusion would recycle this sound energy into the room allowing for lower monitoring levels.

Another effective means of introducing diffusion to a room is really a by-product of the distribution of absorption materials.

It has been shown [39], [40] that patches of absorbent material contrasted with open reflective areas in between breaks up the wave fronts through diffraction at the edges of the absorption materials causing scattering. Also, by alternating absorption units of different design frequencies and/or different potencies, a certain amount of diffusion can be obtained.

In the 1940's, J.E. Volkmann from the RCA company, proposed the use of convex wood panels as effective sound diffusers [41]. These so-called "poly-cylindrical" diffusers became ubiquitous in RCA recording studios throughout the world and are still found today.

This was probably the first purpose-built device for sound diffusion. Volkmann relied on the inherent sound dispersion properties of convex shapes. The dispersion of sound not only occurred by virtue of its curved shape, but by the diaphragmatic action of the thin wood panels which would scatter any sound it did not absorb.

It was recommended that these "polys" be mounted with their axes perpendicular to each other on opposing surfaces thus providing maximum diffusion across 3 orthogonal planes. Also, by using polys of differing cord dimensions, a wider bandwidth of effective diffusion would result.

Probably the most interesting means of achieving diffusion thus far stems from the research of Dr. Manfred Schroeder at AT&T Bell Laboratories [42], [43], [44]. There, he developed an acoustical grating system analogous to the diffraction gratings used to scatter light rays.

This grating system is arranged into periods containing a series of grooves of varying depth organized svmmetrically about the center. The various depths are derived from Number Theory using "residue sequences" based on prime numbers. Taking the prime number 17 for example; The numbers from 1 to 16 are squared, then divided by 17 (the modulus) and the remainder (or residue) produces the following sequence:

1, 4, 9, 16, 8, 2, 15, 13, 13, 15, 2, 8, 16, 9, 4, 1

Deriving the grating depths from the sequence involves multiplying by the wavelength value of the design frequency (the lowest frequency desired for diffusion), then dividing by a factor corresponding to its numerical position within the sequence.

The lowest frequency where the diffuser is effective is directly related to the depth of the wells. The highest frequency of diffusion is inversely related to the (constant) width of the wells. The degree (or amount) of diffusion is directly proportional to the overall length of the diffuser module. The optimum bandwidth of diffusion is achieved with a unit having a large number of deep and narrow wells.

The scattering pattern is a result of wave interference between the incident (incoming) and reflected sound due to the different phase relationships as delayed by the varying well depths.

It is easy to see that the depth of the wells also determine the length of the diffuse "tail" as deeper wells will return the sound over a longer period of time resulting in less "punch" or "tightness" and more ambience.

Jeffrey Borish [45] warns that the use of such diffusers could produce a type of spectral distortion. A simple planar surface would be unbiased in its treatment of an incident sound by reflecting it in total according to Borish. But a Schroeder diffuser would have an unequal treatment of the spectrum usually with more diffusion taking place at higher frequencies. This could cause different notes or overtones of a reproduced instrument to have different spatial characteristics distorted by the diffuser. Yet, it should be realized that no surface is a perfect reflector, absorber or diffuser. Some type of spectral transformation of the original sound would occur under any of these 3 conditions. Our auditory system becomes conditioned to expect such processing of sound by the room boundaries. This is what characterizes sound indoors as opposed to outdoors.

Despite this one minor criticism, diffusers have become a very successful tool in acoustic design not only in recording studios but performance halls and places of worship.

This success is mainly due to the aggresive research, development and promotion of diffusers by RPG Systems of Maryland [46], [47], [48]. They have introduced many diffuser products that have different characteristics, some which also employ absorption into the unit.

5. ANALYSIS OF CONTROL ROOM 'A'

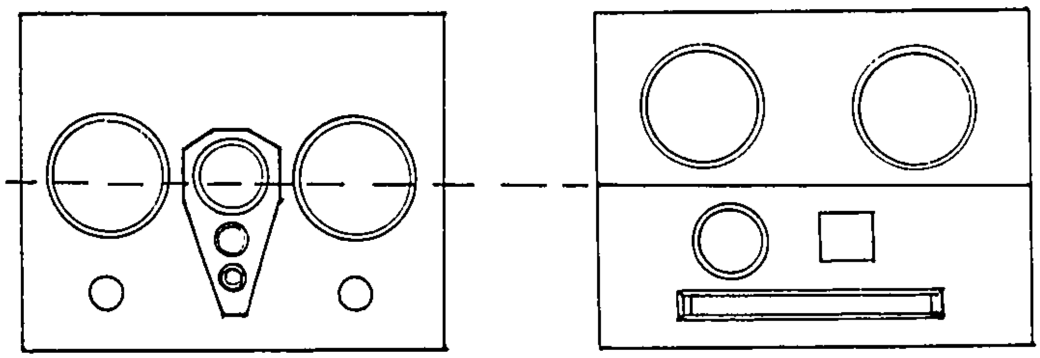
Having outlined the important variables that constitute any control room monitoring environment, the next step is to analyze the control room under study according to these variables.

The McGill University Recording Studio A and its accompanying control room was designed by Dr. Floyd Toole of the Acoustics Division of the National Research Council of Canada in 1979 and construction was completed by the end of that year.

Overall, the design reflects the protocol of that time period in terms of style and control room acoustic theory, many ideas of which are still in use today. An important point to emphasize here is that no changes have been made to the architecture of the studio since its construction over 10 years ago from this writing. Most (commercial) recording studios make significant changes at least every 5 years whether in an attempt to update the cosmetic/style or improve the ergonomics and "performance" of the room.

The only alteration significant to this project is the changeover of loudspeaker system. Up until then, the monitor system was a 4-way, tri-amp system with midrange horn designed by Dr. Toole. The current speakers were installed in the summer of 1988 into the original cabinet structure with some modifications to the front baffle surface and interior.

The original cabinet was split into 2 sections separating the woofers from the mid and high frequency components. The slope of the baffle was changed from the dual section wedge shape to a single, flat, downward-angled surface at 15°(Figure 5-1).



current system previous system

The monitors themselves are a custom designed STATE OF THE ART ELECTRONIK system with JBL and Dynaudio components arranged as in Figure 5-1. The design is loosely based on the SOTA CF-2000 series monitor except for the cabinet.

Referring to the floor plan layout of Figure 5-2, the loudspeakers are 3.25 metres apart (from their center axis). The axes of the equilateral triangle cross over the console about 60 cm. in front of the engineer's position. This geometry is advocated by Dr. Toole as he believes it provides a more stable stereo image [49].

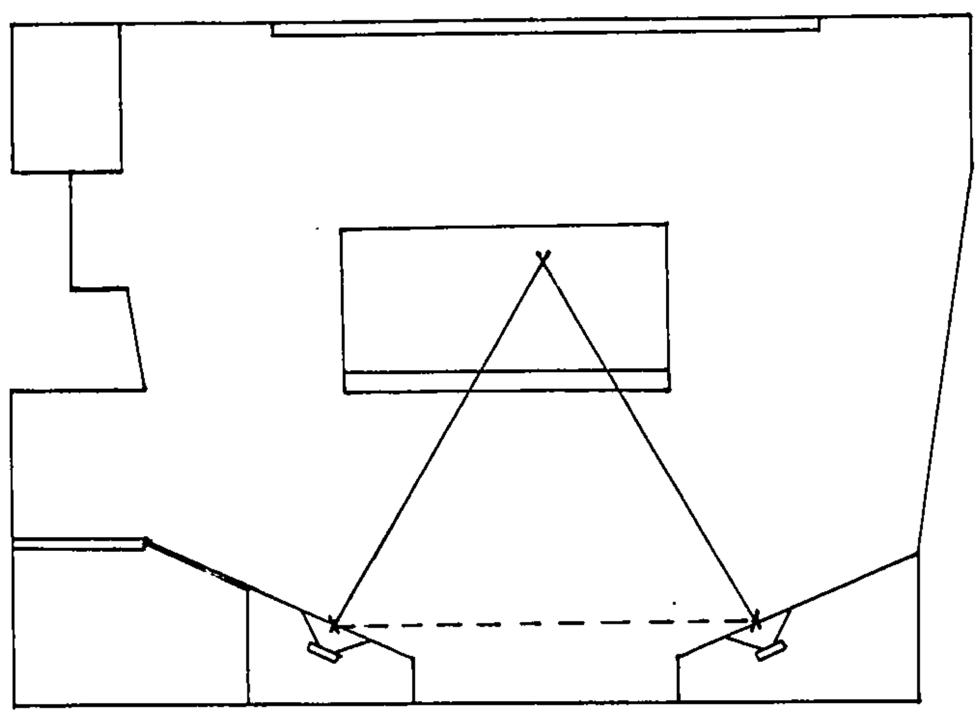


Figure 5-2

The center point of the loudspeaker array is situated 2.15 metres above the floor and about 0.5 metres from the ceiling. This is a common placement in most control rooms which is an inverted situation from most home listening rooms where the loudspeakers are closer to the floor instead.

Such a close proximity to the ceiling surface can usually be problematic by causing strong early reflections of less than 5 milliseconds which when fused with the direct sound, can produce audible comb-filter colorations. But the effect of these reflections is somewhat lessened in control room A since the ceiling is covered with 5cm. thick semi-rigid fiberglass (under a thin acoustically transparent cloth) which would help absorb some of this astray sound energy.

Below both monitor cabinets are similar wooden cabinets which serve as storage space for tapes, microphones, etc.. Their outline forms a continuation from the monitor surface straight down to the floor. In effect, this serves to extend the baffle surface into a larger one.

The wooden surfaces are of course reflective. This live front-end design was commonplace in the late 1970's advocated by such studio designers as Michael Rettinger [50]. The idea behind this was to increase the efficiency of the monitor system. Any such increase would result in reduced loudspeaker cone excursion which in turn yields lower distortion and superior linearity at bass frequencies also reducing listener fatigue. In addition, the power amplifiers can be run at a lower level resulting in less amplifier noise for a given SPL.

Since these cabinets provide a smooth transition from the baffle surface, there are no significant protrusions or sharp discontinuities to cause unwanted early reflections and diffraction effects.

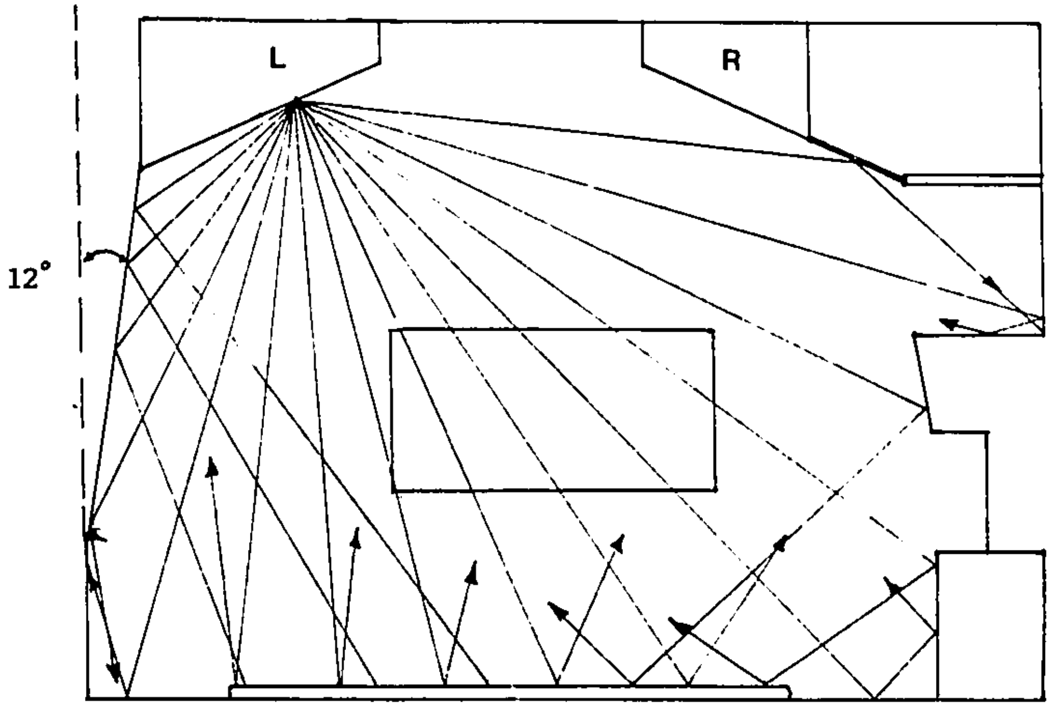
This live front end is opposite to the practice of LEDE type rooms pioneered by Don Davis and Chips Davis [51]. Basically, LEDE rooms demanded that the frontal area around the loudspeakers absorb mid-high frequencies as much as possible to eliminate any early reflections from entering the engineers's position. Other design theories loosely followed this idea with such terms as RFZ[1] (Reflection Free Zone) [52], [53], and CTP (Controlled Travel Path) [54] from Westlake. The difference being that they would use reflection to redirect the sound energy away from the engineer's position by splaying the side walls. Dr. Toole made use of this simple idea in control room A by splaying the walls about 12° outward which is an often-used angle in control room design.

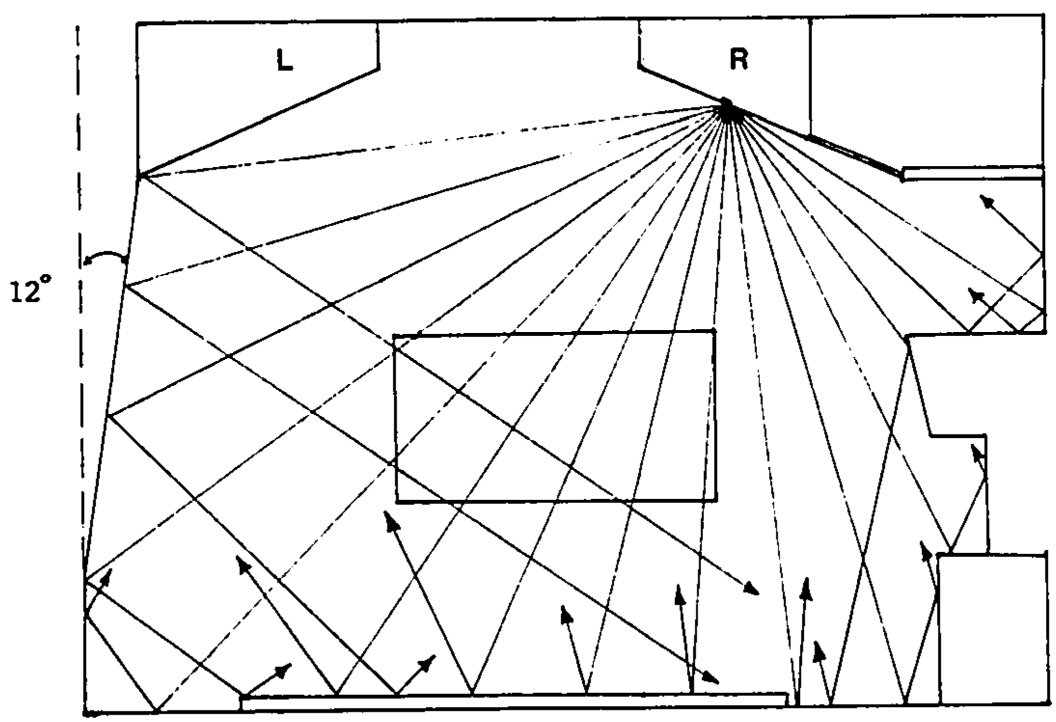
The best method of determining the sound propagation from the loudspeakers is drawing a geometric analysis or "ray trace" of the room. Such a map is only an idealised situation assuming perfect hemispherical radiation from the loudspeakers.

Looking at Figure 5-3, it can be seen that the horizontal reflection pattern is quite favorable with the wide dimensions and splayed side walls mangaging to reflect the first-order reflections away from the listening position. However, the

1) RFZ is a registered trademark of RPG Diffuser Systems Inc..

<u>Figure 5-3</u>





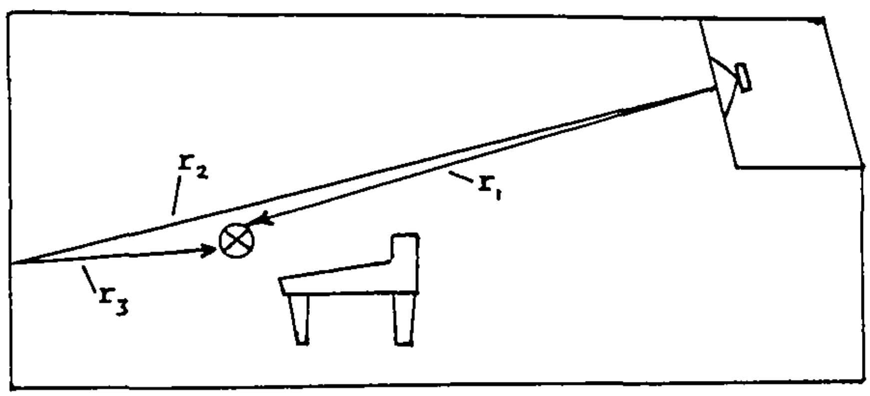
back wall is parallel to the front resulting in a strong first-order reflection back towards the listening area. The wall is about 1.2 metres behind the listening position making a total distance of (1.2 x 2) = 2.4 metres before it arrives at this position. The delayed time of arrival can be calculated by the formula [55];

$$\frac{r_3 + r_2 - r_1}{c} \times 1000 = R \text{ (ms.)}$$

in metres $\left\{ \begin{array}{l} r_1 = \text{direct sound travel path} \\ r_2 = \text{secondary travel path} \\ r_3 = \text{reflection of secondary travel path} \end{array} \right.$

c = velocity of sound (344m/s @ 20 C)

$$\frac{(1.2) + (1.2) - (3.4)}{344} = 6.9\text{ms.}$$



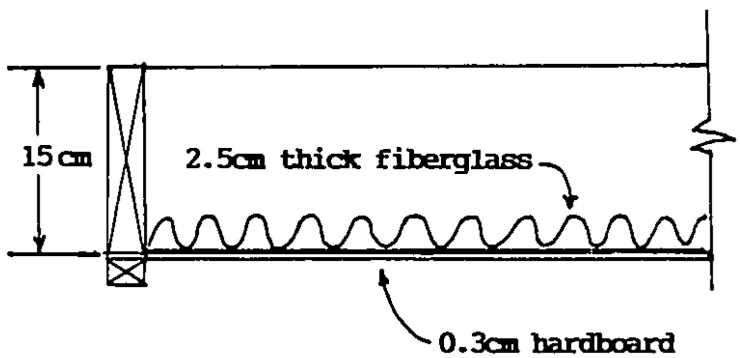
Any reflection arriving less than 20ms after the direct sound can produce audible coloration as it is well below the so-called Haas Zone [56] of 20 - 40ms. Since it is a first-order reflection, it will return its incident direct sound fairly well intact. (The heavy carpet-like fabric does little to absorb frequencies above 1KHz). A strong, delayed reflection that is so well correlated with the original direct signal can corrupt the overall perceived sound quality.

Furthermore, a reflection that originates from the same general direction as the direct sound can be masked by that same direct sound quite effectively [57]. But a reflection from behind or side of the listener will be far more easily perceived.

Another point of analysis is discerning whether the room is bi-laterally symmetrical. Any significant deviation from symmetry can corrupt stereo imaging by biasing the the sound towards the side that produces stronger and/or earlier reflections. This can occur by a number of situations by one side wall being closer, more reflective or splayed at a different angle than its opposite wall.

By examining Figure 5-3, it can be seen that the general angle of both side walls are equal with the north wall surface being broken up by the doorway and machine soffit areas. This only introduces more diffusion than that of its relatively smooth-surfaced opposite wall. As well, there is no absorption whatsoever on the north wall. The south wall has a large window which can be slightly effective (although unintentional here) at absorbing low frequencies.

Below this window is a low frequency membrane absorber constructed as in Figure 5-4.



5.1 Room Dimensions

Much can be learned about the potential sound quality of a room simply by knowing its dimensions. The 3 axial dimensions of control room A are;

Length = 4.8m

Width = 6.8m

Height = 2.5m

total room volume = $81.6m^3$

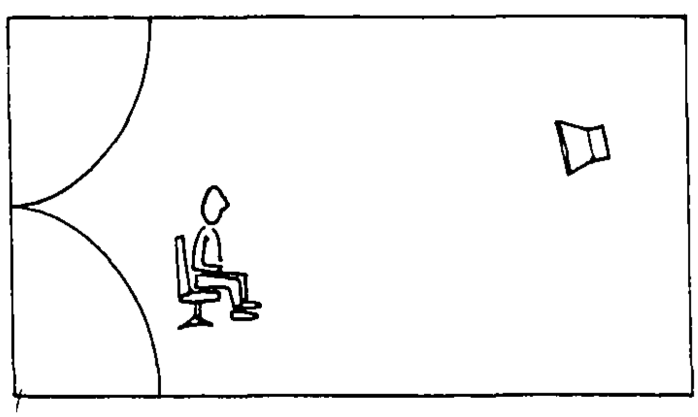
The crossover point (f_c) at which room modes gradually begin to form a continuum can be calculated from the room's smallest dimension; the height.

$$\frac{3c}{D} = f_c$$

$$3\,\frac{344}{2.5} = 412.8\text{Hz}$$

The smallest dimension of 2.5 metres results in an aberration common to many home listening rooms as well as recording control rooms. The fundamental axial mode across the vertical dimension will have its null throughout the room at about a seated listener's ear level which is midway between the ceiling and floor (Fig. 5-5).

mode (0,0,1)



This produces a huge dip at around 69Hz as can be seen at a number of different locations within control room A (Fig. 5-6).

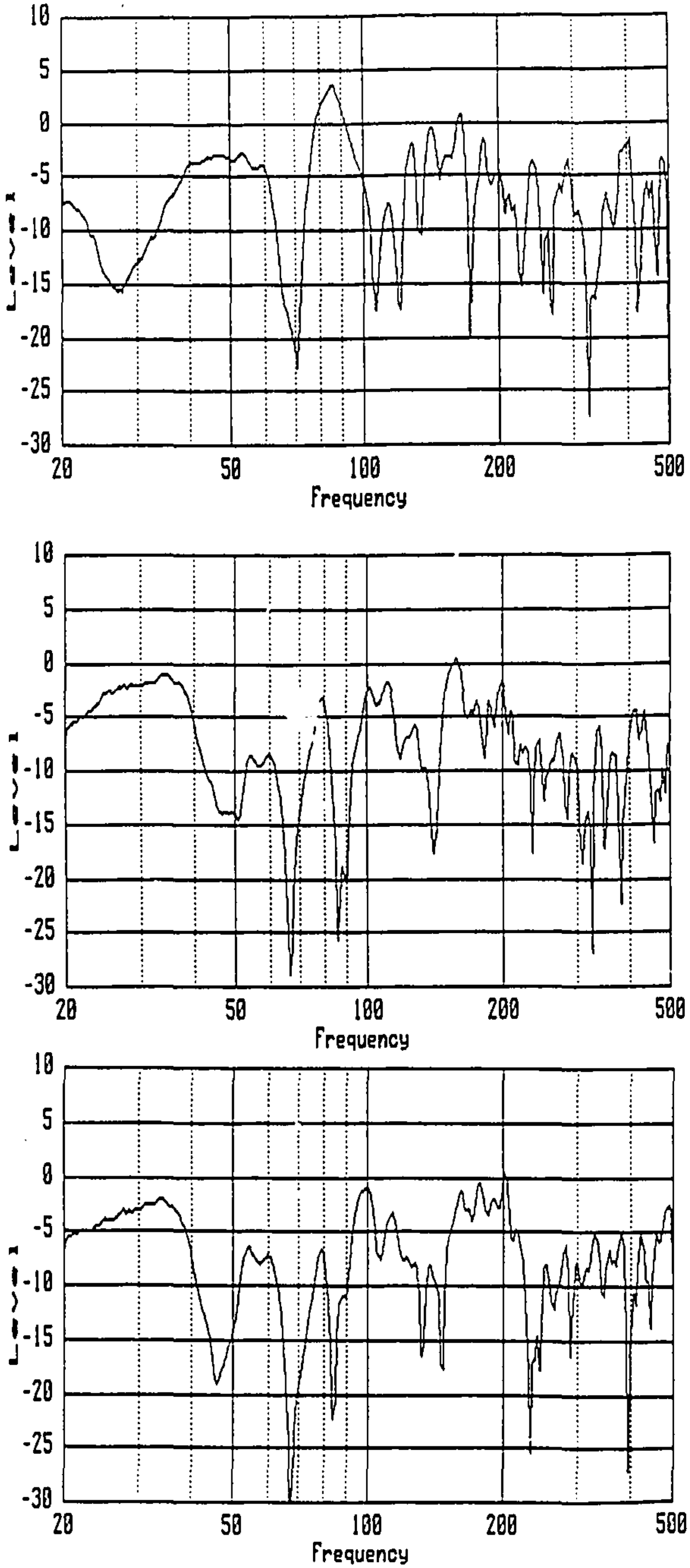

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Figure 5-6

The best way around this is to have a higher ceiling which would move this frequency null lower and out of the range of most music program spectral content. For instance, a ceiling height of 3.5 metres will produce a lower notch at about 49Hz.

One possible advantage that can be derived from this is that most home listening rooms have a ceiling height of 2.5 metres, so it is at least one dimension/variable that is consistent with the room the recording was mixed in.

The longest room dimension in control room A is the width of 6.8 metres. This relatively large width is beneficial in keeping early side wall reflections from reaching the engineer's position. (The 12° splaying also aids and abets this).

Just as the smallest dimension can be used to find the crossover frequency, the longest dimension can be used to derive the bottom frequency limit of the room from the formula;

$$\frac{172}{D} = F_L \text{ (Hz)} \qquad D = \text{longest dimension (m)}$$

$$\frac{172}{6.8} = 25.2\text{Hz}$$

This is not to say that lower frequencies (than F_L) reproduced by loudspeakers cannot exist in a room, they will simply not have the modal support as F_L and above has.

From all 3 dimensions, the normal modes of a room can be calculated. It is these dimensions that determine the fundamental resonance of each axial direction and its harmonically related upper resonances.

A modal analysis of control room A will be restricted to the axial modes since they are the most dominant and are responsible for the majority of room colorations.

By the formula:

$$n \frac{172}{D} = f_n$$

D = dimension

n = mode number

Axial Mode Resonances (Hz)

W	L	H
25.3	35.8	68.8
50.6	71.6	137.6
75.9	107.4	206.4
101.2	143.2	275.2
126.5	179.0	344.0
151.8	214.8	412.8
177.1	250.6	481.6
202.4	286.4	
227.7	322.2	
253.0	358.0	
278.3	393.8	
303.6	429.6	
328.9	465.4	
354.2	501.2	
379.5		
404.8		
430.1		
455.4		
480.7		
506.0		

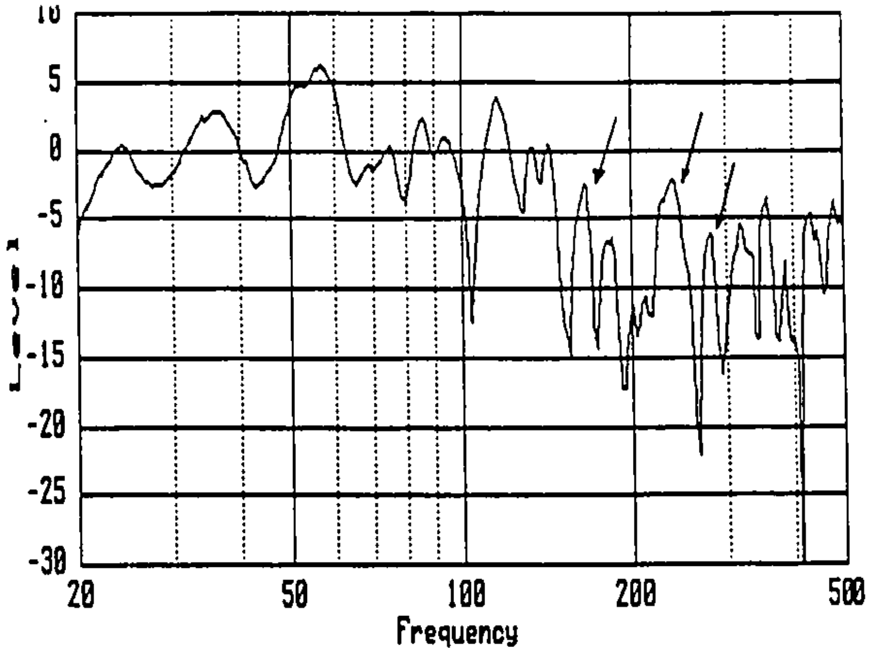
Ascending order	Mode spacing (Hz)
25.3	
35.8	10.5
50.6	14.8
68.9	18.3
71.6	2.7
75.9	4.3
101.2	25.3
107.4	6.2
126.5	19.1
137.6	11.1
143.2	5.6
151.8	8.6
177.1	25.3
179.0	1.9
202.4	23.4
206.4	4.0
214.8	8.4
227.7	12.9
250.6	22.9
253.0	2.4
275.2	22.2
278.3	3.1
286.4	8.1
303.6	17.2
322.2	18.6
328.9	6.7
344.0	15.1
354.2	10.2
358.0	3.8
379.5	21.5
393.8	14.3
404.8	11.0
412.8	8.0
429.6	16.8
430.1	0.5
455.4	25.3
465.4	10.0
480.7	15.3
481.6	0.9
501.2	19.6
506.0	4.8

(It should be noted here that the list of modes is only a close approximation of the resonant points. Since control room A is not perfectly rectangular, the irregular shape might produce a slight error margin especially in the width dimension figures).

The cause of most audible colorations is a bunching up of resonances (coincident frequencies) further separated by a larger gap from the other points [58]. By inspecting the table on the previous page, there are no coincident axial modes although there are some areas where resonances are within a few Hertz of each other, most notably; 177-179Hz, 250-253Hz and 275-278Hz.

Figure 5-7 illustrates some positions within control room A where the troublesome frequency areas show up as maximas. However, it should be understood that any of these mode resonances are not constant throughout the room. They follow a nodal pattern of ridges and peaks varying from zero to maximum level depending on the position in the room. So although they may appear potentially troublesome on paper, their pattern position relative to the listener may render them harmless.

Figure 5-7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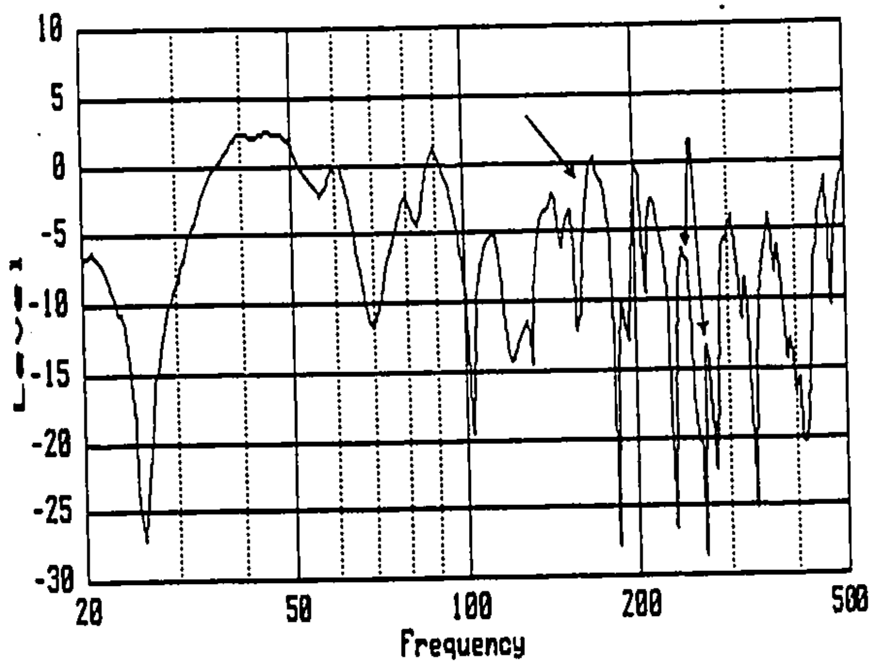
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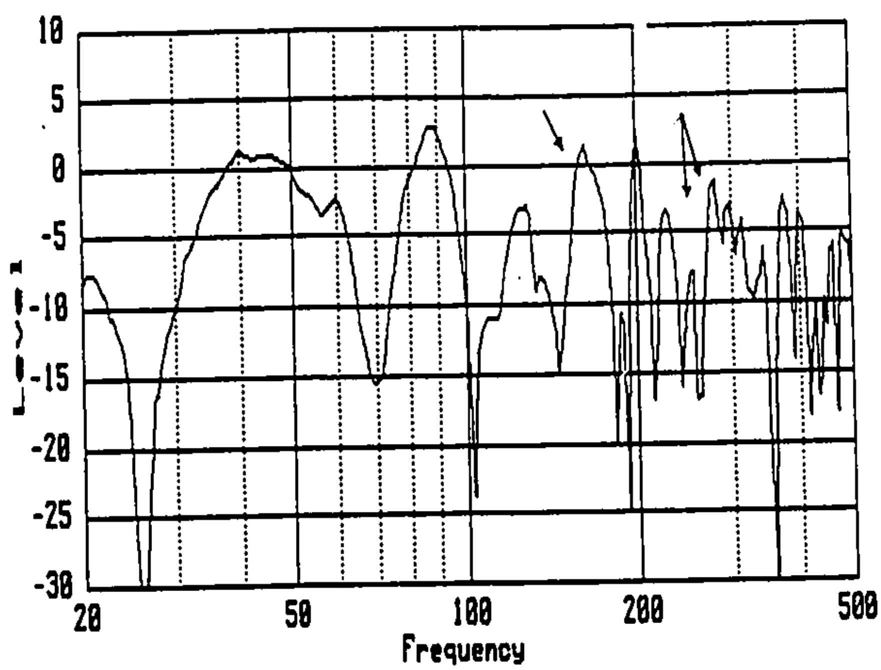
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6. IMPLEMENTATION

6.1 Repositioning of Console

After a thorough analysis of the many variables involved in acoustic design and optimization, some proposals were put forth and implemented. The first phase of implementation involved a repositioning of the mixing console (Sony MXP-3036) and in effect, repositioning the listening spot along with it.

The console was moved ahead towards the loudspeakers by 30 centimetres. Any further movement was physically prohibited by the fixed length of multi-cable connected to the console from beneath the floor.

By moving the listening spot closer to the monitors, it increased the angle causing the sound image to be raised. The original angle of roughly 10° is considered optimal. The repositioning ahead by 30 cm. resulted in a slightly steeper angle (on average) but still within an acceptable range (10° - 20°).

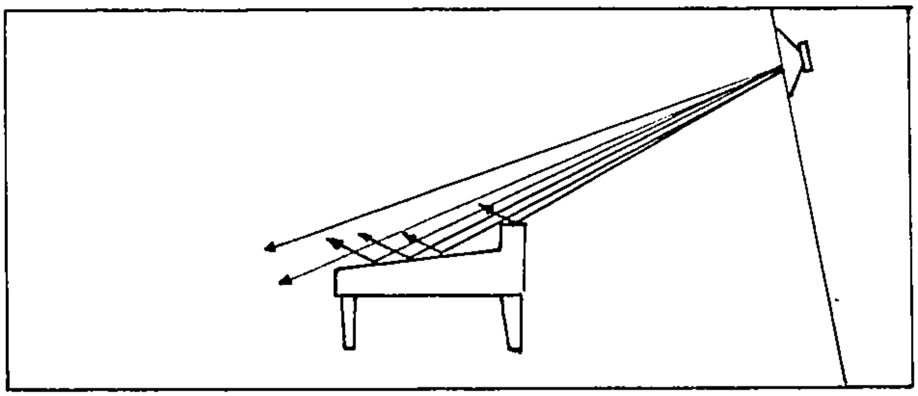
The original placement of the console forced the engineer's position to be behind the apex of the crossing monitor axes by about 30 cm.. This happened to be the distance moved forward causing the (average) listening spot to be right at the point of crossing. This new position is an even compromise of the opposing ideas discussed in section 2.1.

Another consequence of the new position is a slightly wider stereo image. The distance between the speakers does not change, but the angle is widened from the reference point of the listening spot. This broadening of the stereo width is favourable by bringing more left-right separation and the effect of pan-pot settings can be more clearly perceived.

Stereo localization is also improved by increasing the direct-to-reflected sound ratio when moved nearer to the loudspeakers. In this way, there is a heightened sense of the stereo sound field through the higher proportion (than previous) of direct sound from the loudspeakers. In addition, the engineer can hear more clearly what the microphones are actually picking up including the early reflection pattern surrounding the microphone by virtue of the control room monitoring environment imposing a lesser degree of its reflected sound.

By being closer to the monitors, fewer first-order side-wall reflections enter the listening area. As well, the distance between the rear wall and the listener is increased. Any reflections that reach the listener from behind are slightly lower in absolute level, as well as being lower in relation to the direct sound.

Another potential problem area for early reflections is the console surface. By moving the console forward, the loudspeakers "see" less of the surface and only a very narrow range of radiation angles reach and reflect off of the console (Fig. 6-1).



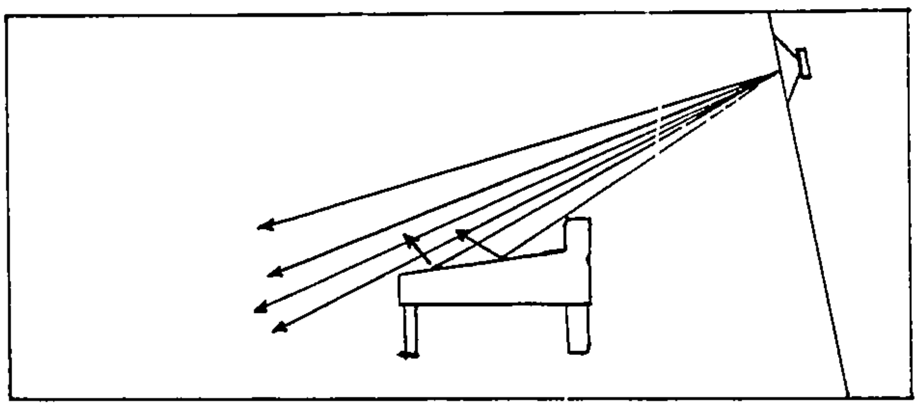


Figure 6-1

Among other more practical benefits, is the increased area behind the console. This extra room is especially needed for classes where between 4 and 6 students must share this space behind the console to make judgements and analysis of recordings. In effect, this extra space was "stolen" from the area in front of the console which was abundant and wasteful.

It should be noted that in 1989, a Dolby Stereo Surround decoding system (with 4 surround and 1 center loudspeaker) was installed in control room A. Mounting the two rear surround speakers on the back wall proved to be quite close to the mixing position as the sound emitted from surrounds should not be heard as separate entities, but as an enveloping ambience. Moving the listening position forward helped to somewhat rectify this situation.

A final practical advantage is that the listening is now moved further away from distracting noise as generated by the multi-track recorder, hard disk drives and cooling fans.

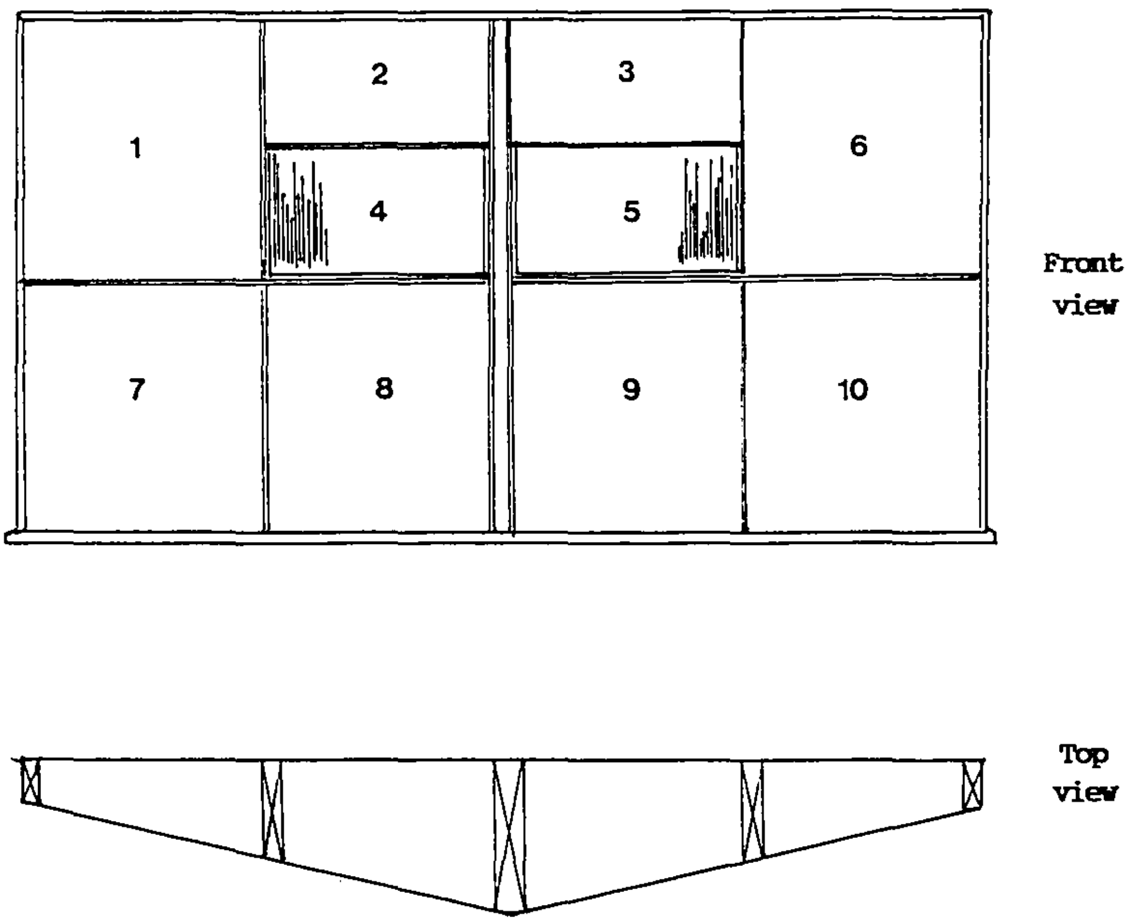
6.2 Rear Wall

The next target for optimization was the rear wall. The existing rear wall was completely removed and replaced by a triangular-shaped wall structure. This structure featured a modular approach where the wall was divided into 10 different modules independent of each other (Fig. 6-2).

The wall encompassed low frequency absorption (modules 1, 6, 7, 8, 9, & 10), high frequency absorption (modules 2 & 3), mid - high frequency diffusion (modules 4 & 5), and shelving space to accommodate (and hide) the 2 rear surround speakers.

This modular approach facilitates the disassembly of any or all of the sections for replacement with different acoustical treatments if desired.

Figure 6-2



KEY:

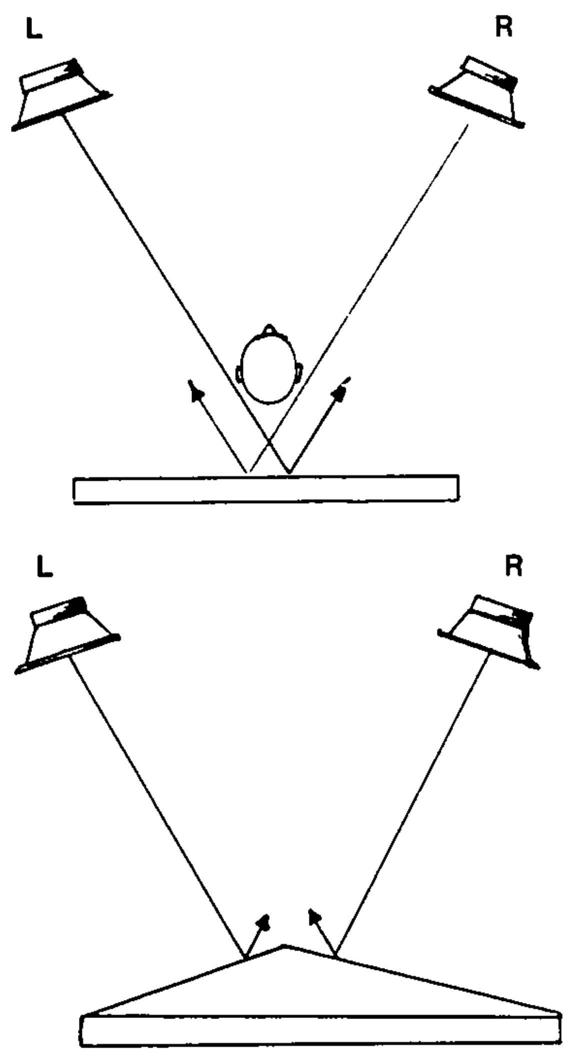
1, 6, 7, 8, 9, 10 —— L.F. Absorbers

2, 3 —— H.F. Absorbers

4, 5 —— Diffusors

The overall triangular/convex shape of the wall was chosen primarily for its ability to redirect most relflections away from the listening area. The effect of this is to reduce any interference with the direct sound which would otherwise cause spectral coloration.

As well, it increases the left-right separation by directing reflections from the left (or right) loudspeaker further towards the left (right). Otherwise, reflections from a flat wall surface (behind) would combine to form a mixed left-right sound field returning towards the mix position and increasing channel crosstalk (Fig. 6-3).



6.2.1 Low Frequency Absorber

The purpose of installing low frequency absorbers was to hopefully dampen some of the room modes and broaden their bandwidth. (A broader bandwidth means a lower 'Q' value resulting in a shorter decay time).

Usually the most efficient place to put a bass absorber is in the corners of a room. The 2 front corners in control room A were out of the question since they were too close to the loudspeakers. Any bass absorption there would drain some of the sound as soon as it was output. Bass absorbers could not be physically placed in the 2 rear corners since they were occupied by equipment and a microphone cable storage area.

The obvious place for the bass absorbers was against most of the rear wall where it would drain some of the bass energy before it resulted in a pressure buildup from a reflection.

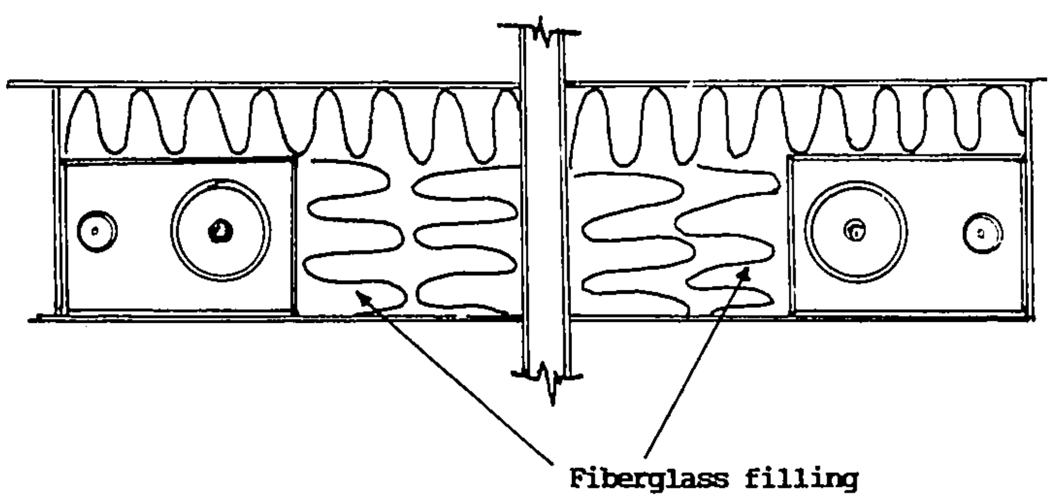
Four of the six LF absorber modules were situated along the lower half of the wall where any type of high frequency acoustical treatment would be wasteful.

The overall gradual wedge shape has 2 advantages for LF absorption. First, it provides more surface area than if it were a flat wall. This is similar to the thought behind Sonex wedge absorbers (discussed in section 3.2) where there is more useful absorbing surface for the same amount of wall space. The second advantage is that its slanted shape relative to the rigid back wall allowed for the absorber to be effective over a broader bandwidth. Since depth is an important quantity in LF absorption, the varied depth included a wide range from 14.5 to 42cm.. The target range of effective absorption was 90 - 300Hz; this is where most audible and offensive colorations fall. The combination of wide-angle slanting and fiberglass insert helped to provide this wide bandwidth of absorption.

A combination membrane/Helmholtz resonator type absorber was built using 0.3cm. thick particle board with 0.8cm. diameter holes spaced about 10cm. apart yielding a perforation percentage of 0.25%. Each perforated particle board was tightly sealed to the frame and covered with a robust, loose-knit fabric. Into each module, a fiberglass (10cm.) sheet was applied against the back of the perforated board to further broaden the effective bandwidth.

6.2.2 High Frequency Absorber

The two compartments featuring high frequency absorption also share the space with the rear surround speakers (Fig. 6-4).



The HF aborption only takes up a small percentage of the wall but its strategic placement is the most potent and practical. Here, it is at the highest portion of the structure where it can be the most effective at absorbing some of the high frequency energy that isn't diffused or redirected by other parts of the wall.

It is comprised of a cavity area of varying depth stuffed tightly with (10cm. thick) fiberglass. The cavity is covered with an acoustically transparent fabric wrapped around an open frame.

6.2.3 Diffuser

Diffusion was chosen for its ability to reduce the sound energy incident upon it and backscatter it uniformly over a wide area. The overall reduction in sound energy is helpful in minimizing any interference with the direct sound which may cause sound coloration and image shifting. The broad area of backscattered sound would result in a wider, more uniform listening area.

Its placement within the arrangement of modules was optimal in that it was the most direct path of mid - high frequencies beamed from the loudspeakers. There was no need to build a floor-to-ceiling diffuser module as the bottom half would be wasteful since most direct mid/high frequency sound would not reach that area. (The top quarter section was needed to house the rear surround speakers).

Most incident sound energy to the sides of the diffuser modules would be directed away from the prime listening area by virtue of the slanted shape.

Two identical diffuser modules were built and placed adjacent to each other forming 2 periods of a quadratic residue sequence based on the number 53.

The effective bandwidth of the unit is defined by the design (lowest) frequency and maximum frequency. Both points are obtained using the deepest well depth and the (constant) well width within the formula:

$$f = \frac{c}{2D\ (m.)}$$

- The deepest well depth = .31 metres;

$$f = \frac{344}{2(.31)}$$

$$f_L = 554\text{Hz}$$

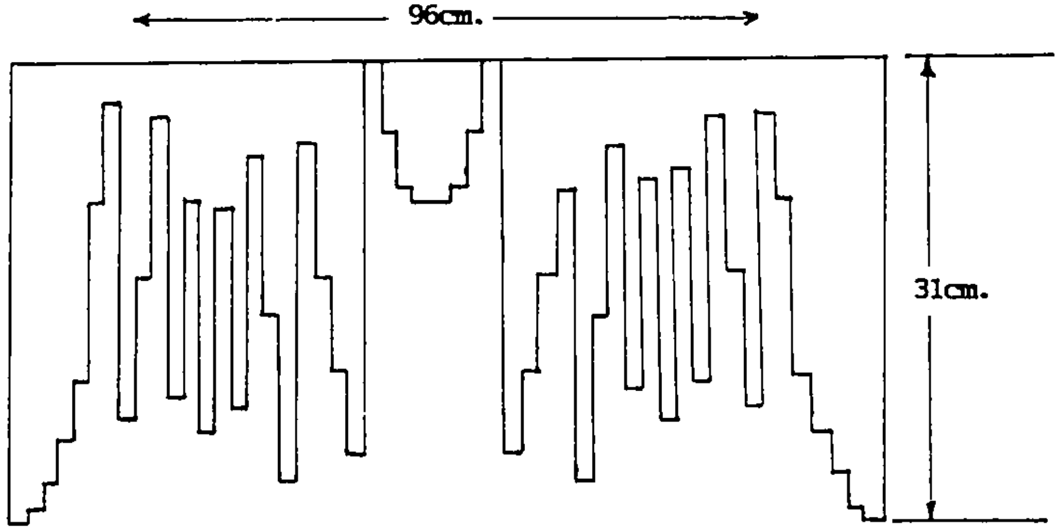
- The well width = .016 metres;

$$f = \frac{344}{2(.016)}$$

$$f_M = 10750\text{Hz}$$

This yields an effective range of between 554 and 10750Hz. Behaviour outside this range (above and below) tends to be specular rather than diffusive.

There are 53 wells for each unit as shown in Fig. 6-5.



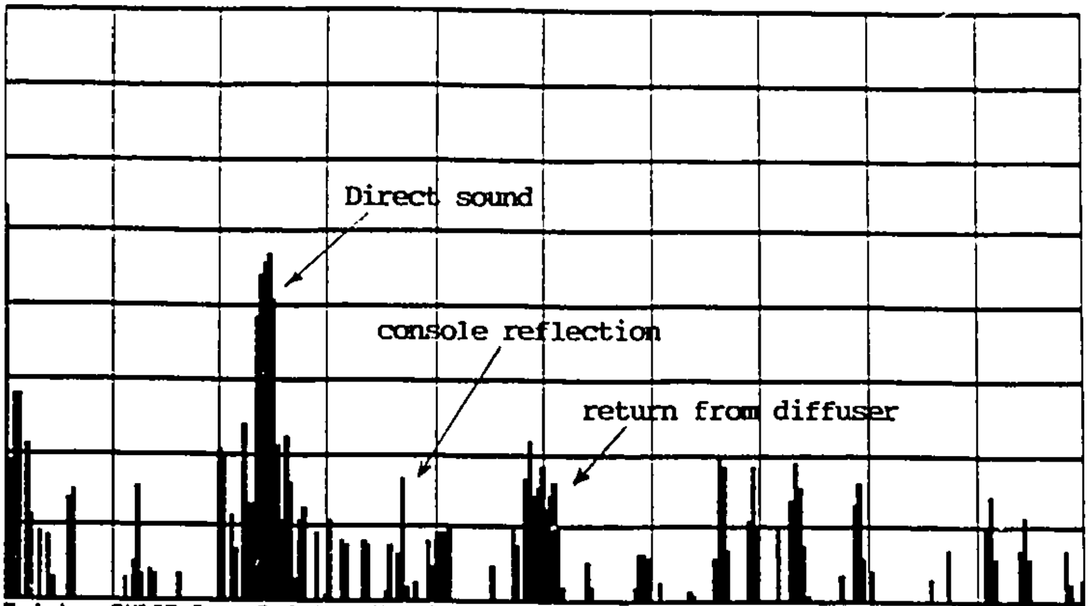
(6-5) Top view of diffuser module showing grating sequence pattern.

Figure 6-6 shows the Energy Time Curves of 3 different locations comparing the back wall with and without the diffuser modules.

They exhibit a significant reduction in the level of the return from the back wall by 5dB when compared to a diffuser-less wall. However, there is no significant tail of diffuse reflections as was expected and hoped for. This tail of reflections is what constitutes the diffusion of sound over time (temporally). Instead, the ETC show the returns from the diffuser to be concentrated over time.

Figure
6–6a

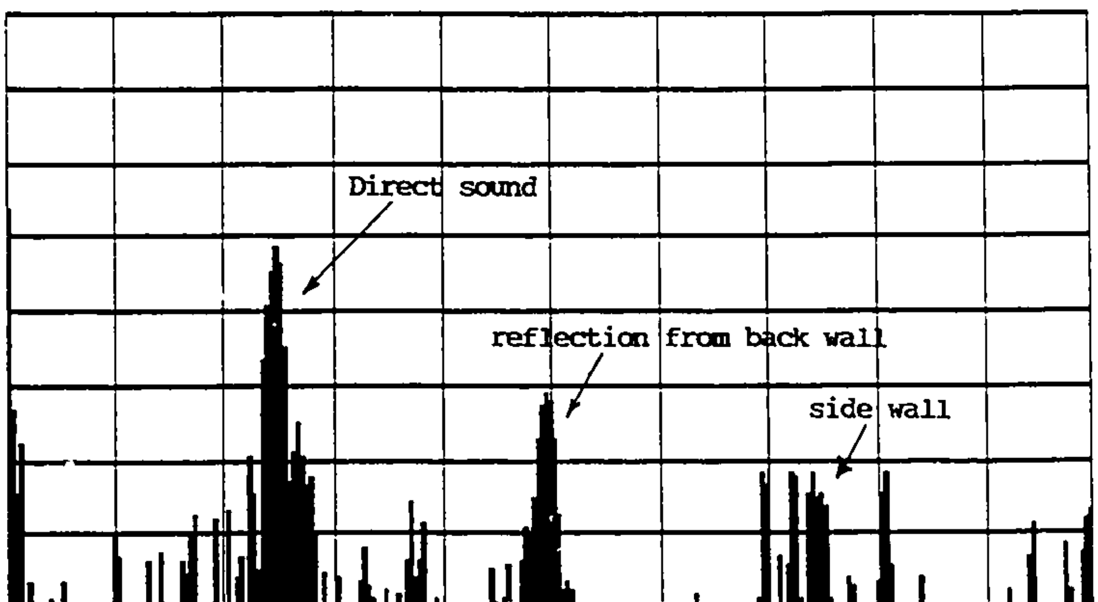
with
diffuser



F data: PULSE.2 P data: PULSE.3 F ref: FTEST.REF P ref: PTEST.REF
Path: Save file: TEMP
Format: ETC Eq: FREQ Coupling: DC Freq step: 25.00 Hz
Filter: LP Order: 1 Form: WERTER Cut-off: 10.00 KHz
Scale: 5.000 dBV/div Reference: 35.00 dBV
Time: 3.999 mSec/div Start: 0.00 Sec Samples: 1024 FFT
FINISHED

<u>POSITION 1</u>

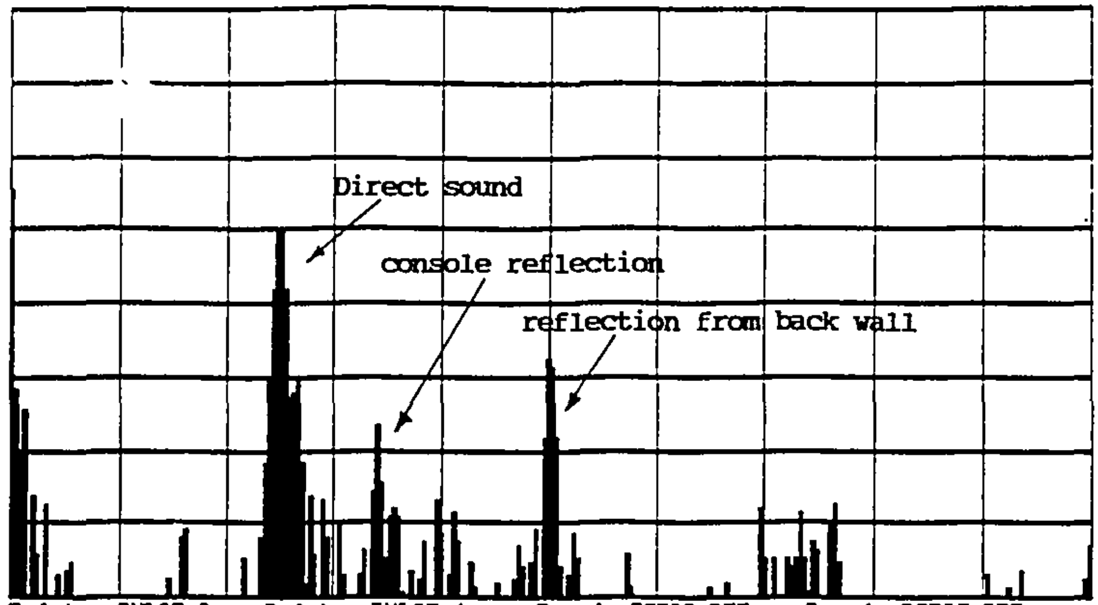
without
diffuser



F data: PULSE.38 P data: PULSE.39 F ref: FTEST.REF P ref: PTEST.REF
Path: Save file: TEMP
Format: ETC Eq: FREQ Coupling: DC Freq step: 25.00 Hz
Filter: LP Order: 1 Form: WERTER Cut-off: 10.00 KHz
Scale: 5.000 dBV/div Reference: 35.00 dBV
Time: 3.999 mSec/div Start: 0.00 Sec Samples: 1024 FFT
FINISHED

Figure
6-6b

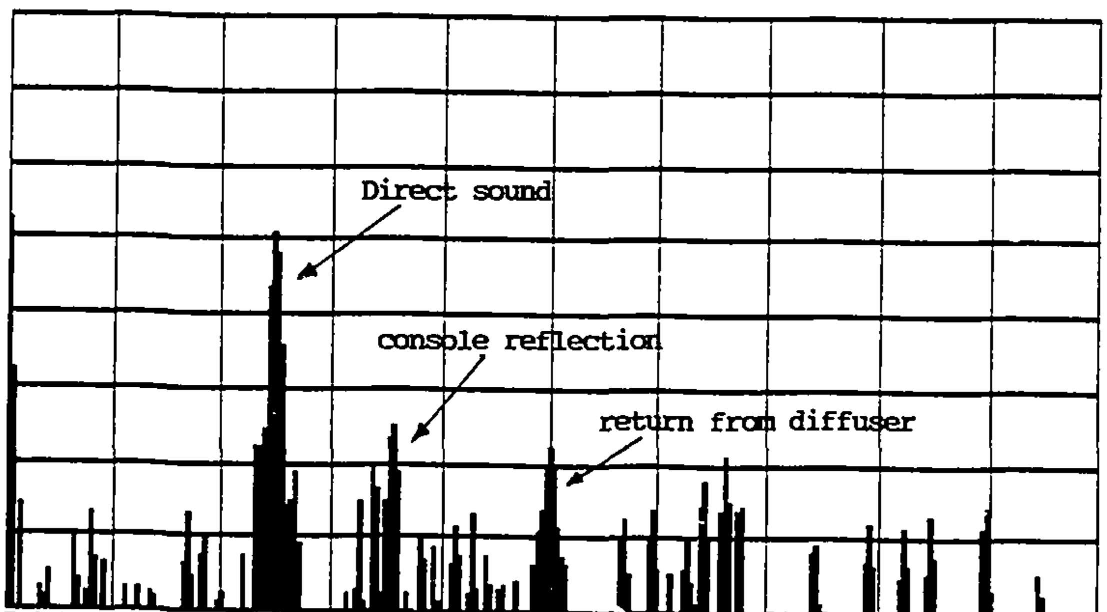
without
diffuser



F data: PULSE.0   P data: PULSE.1   F ref: FTEST.REF   P ref: PTEST.REF
Path: Save file: TEMP
Format: ETC   Eq: FREQ   Coupling: DC   Freq step: 25.00 Hz
Filter: LP   Order: 1   Form: WERTER   Cut-off: 10.00 KHz
Scale: 5.000 dBV/div   Reference: 35.00 dBV
Time: 3.999 mSec/div   Start: 0.00 Sec   Samples: 1024   FFT
FINISHED

POSITION 2

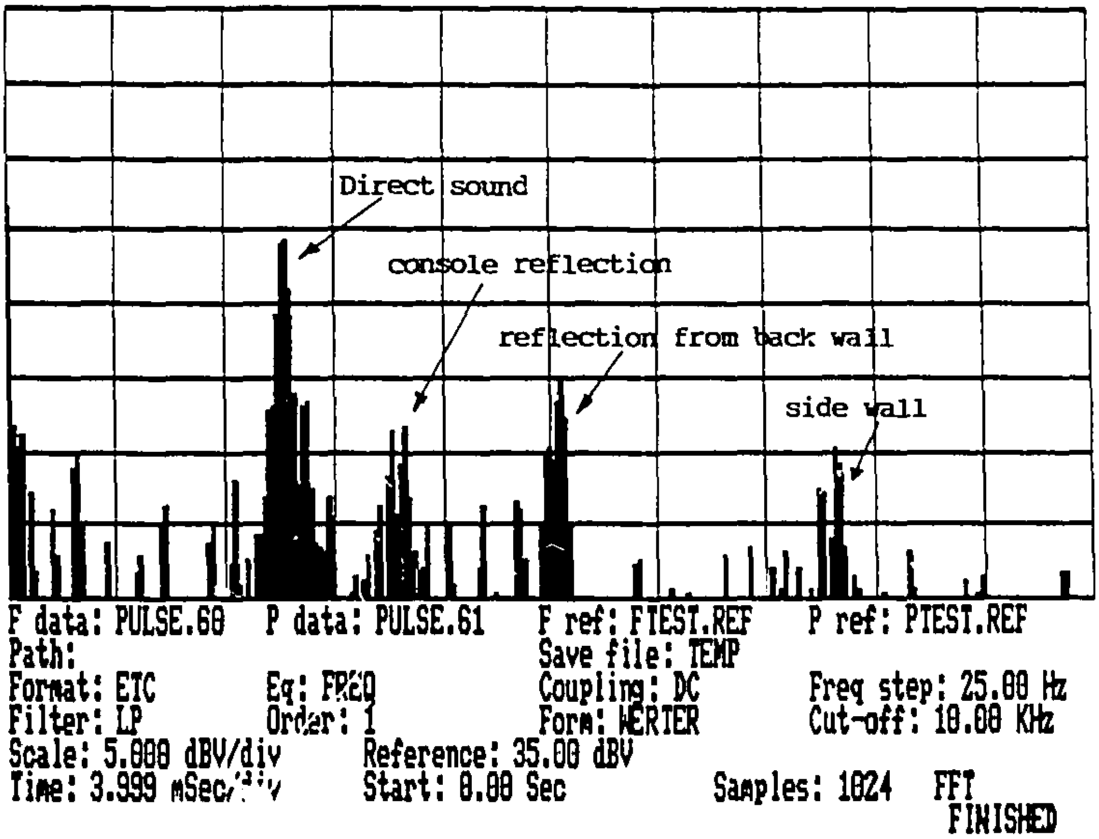
with
diffuser



F data: PULSE.48   P data: PULSE.49   F ref: FTEST.REF   P ref: PTEST.REF
Path: Save file: TEMP
Format: ETC   Eq: FREQ   Coupling: DC   Freq step: 25.00 Hz
Filter: LP   Order: 1   Form: WERTER   Cut-off: 10.00 KHz
Scale: 5.000 dBV/div   Reference: 35.00 dBV
Time: 3.999 mSec/div   Start: 0.00 Sec   Samples: 1024   FFT
FINISHED

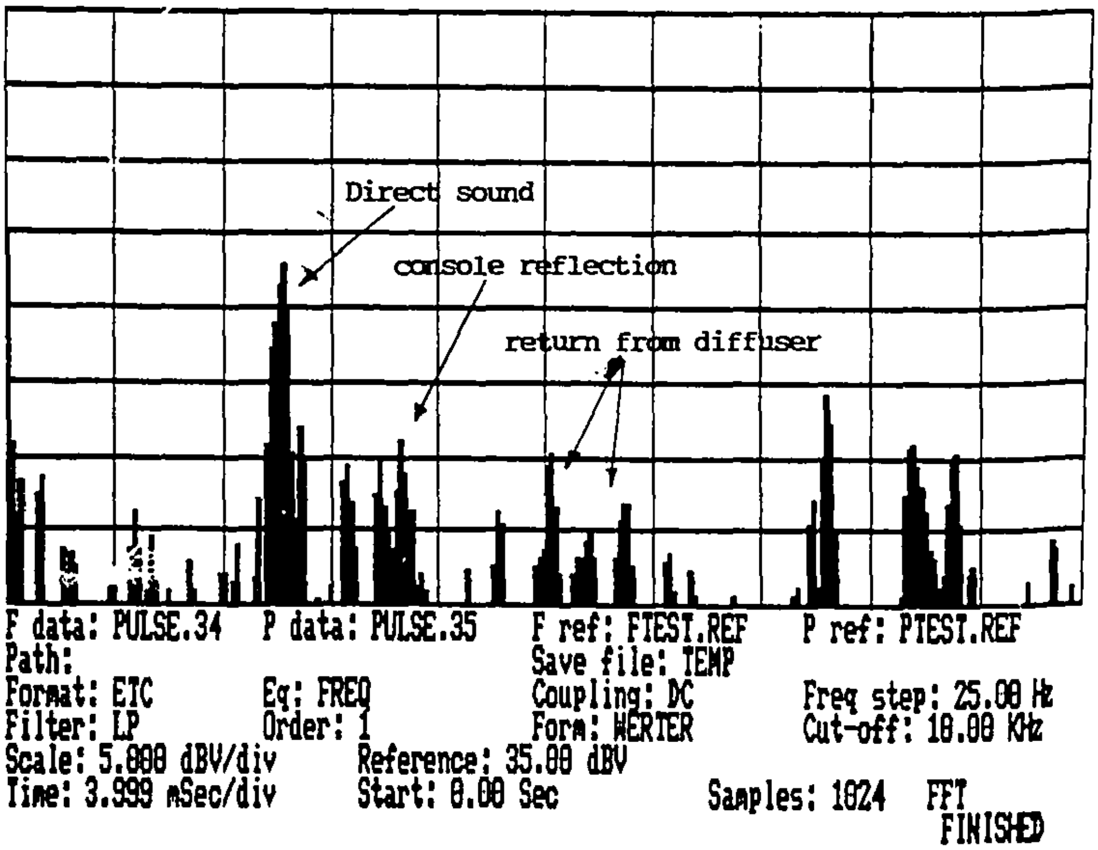
Figure
6-6c

without
diffuser



POSITION 3

with
diffuser



6.3 Subjective Effects

Overall, the students who use control room A on a regular basis noticed a slight improvement in imaging after the project implementation was completed. They reported that the images became more clearly laid out and defined across the stereo soundstage.

This is probably due to a combination of the repositioned console, the overall back wall shape, and the diffusers.

The back wall also "appears" to be further away than it is since its effect is partially removed by the splayed surface and diffusion/absorption modules. This allows the engineer to focus more on the direct sound from the loudspeakers without much interference from behind.
In effect, the impression of a larger room is produced.

7. CONCLUSION

It should be noted that improving the neutrality of a listening environment does not guarantee improved sound quality as perceived by the listener. Often what occurs instead is that other inferior parts of the sound reproduction chain become more apparent as the room's influence is partially removed.

All in all, there were 2 separate areas of optimization that were implemented: a repositioning forward of the console/listening area; and replacement of the back wall with a modular structure.

These ideas were born out of developing an understanding of the behaviour of sound in an enclosed space and in relation to different surface treatments. The changes made do not fulfill the ideal acoustical environment for listening, but rather they are steps towards optimization and neutrality of the room. More specifically, a greater understanding of control room A was achieved, opening it up for new improvement ideas in the future.

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