Loudspeaker and listener positions for optimal low-frequency spatial reproduction in listening rooms.

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Abstract:

This paper will briefly describe the physical and physiological mechanisms that enable low-frequency externalization and spatial reproduction in listening rooms. These mechanisms depend on reproducing a time-varying interaural time difference at the listening position through interference between symmetric and asymmetric room modes. The effect works successfully when the symmetric and asymmetric room modes are driven by independent portions of a multi-channel signal, typically the in-phase and the anti-phase component of a two-channel recording. The paper will describe how this overlap can be optimized by adjusting the loudspeaker and listener positions for a variety of playback rooms. The results show that certain common room shapes yield low spatiality regardless of loudspeaker and listener positions.

Introduction:

This note attempts to cut through some of the confusion and contradiction around the subject of loudspeaker placement in rooms, with an emphasis on reproduction of spatial information. This subject is inherently contradictory, as the placement that is optimal for spatial properties might be different than a speaker placement that is optimal for low frequency response smoothness. The advent of home theater, with the requirement of accommodating a large screen, has made the situation even more complicated.

Spatial perception of reverberation

Before discussing how room modes allow us to hear spatial properties in a recording, it is well to briefly discuss how spatial properties of a hall are perceived at low frequencies. This subject is extensively discussed in reference [1] and [2], and will be summarized here. When we are in a large reverberant space such as a concert hall, low frequency sounds – particularly moderately impulsive sounds such as a plucked string bass – are both easily localized, and seem to have a life of their own. When the bass is plucked, a direct sound wave is created that travels to the listener. The direction of the sound is determined from the arrival time difference between our two ears. There is plenty of time for this direction identification to occur, as usually the reverberant energy from the note arrives 50 to 100ms later. However, when the reverberation arrives, it arrives from all different directions, and at a complex variety of time delays. The perceived direction of

the sound becomes chaotic. The sound seems to swirl around the head. The sound is perceived as external, but coming from all around the listener.

Esthetically the hall is providing a living property to the rather dry sound of a pizzicato cello or bass. The effect is highly involving. Instead of perceiving the music as static and in front of the listener, only the attack of the note is in front. The body of the note seems to be all around the listener, drawing him or her into the performance. In short – the sound is beautiful.

Reproduced sound in a small room lacks this living quality. Often low frequencies are not perceived as external, but inside the head. This is because the low frequencies lack any pressure gradient; the interaural time delay is zero, and remains zero when the head is rotated. Not knowing how to localize these sounds, we perceive them as internal. Reproduced music with this property is so common we have come to accept it as inevitable, perhaps even desirable. But it is far from real, and it is not involving.

The question is: can we reproduce the experience of external, living bass in a small room? To do so, we must somehow cause the apparent direction of the reverberation to fluctuate chaotically around the listener, and to make the rate of change of direction mimic the reverberation in the original concert hall. We can if we are careful. The trick does not work in all rooms. But in many common room and loudspeaker arrangements the effect is easily heard.

Room Modes

When the room wall reflectivity is high – on the order of 0.8 or more – well known patterns of sound pressure are observed in the room. These patterns (room modes) consist of pressure maxima and pressure minima in predictable positions in the room. The modal pressure patterns are strongly dependent on frequency – enough so that one tends to think of the room as supporting only frequencies that correspond to the pressure maxima. This is not true, of course. A powerful subwoofer can reproduce frequencies well below the lowest room mode – and can fill in frequencies between the room modes.

Pressure maxima and minima also exist when the room is well damped, with wall reflectivities below 0.7. Pressure originating in one end of the room (perhaps supplied by a subwoofer in a corner) travels to the opposite walls, reflects, and interferes with the pressure traveling from the loudspeaker. The interference will produce a pressure minimum somewhere in the room if the room dimensions are greater than ½ the wavelength of the sound – regardless of the presence or absence of a so-called "room mode" at that frequency. Resonant room modes are the result when a pressure maximum in the reflected wave corresponds to a pressure maximum at the source position. A powerful loudspeaker can easily supply audible pressure to the room when this condition is not met, but pressure nulls will still exist.

It is a property of these pressure minima that although sound pressure can be low, the particle velocity (the pressure gradient) is at a maximum. It is the audibility of the pressure gradient – and its esthetic value – that is the topic of this paper.

Sound Velocity and pressure gradient

One of the common fallacies in this field is that the perceived sound quality depends only on the sound pressure at the listening position. A corollary of this assumption is that optimum bass in a room occurs when the pressure at low frequencies is maximally uniform both with frequency and listener position. In other words, sound is best when a listener hears all low frequencies equally loudly regardless of where he or she happens to be.

In practice, bass does sound better when it is free of gross frequency response anomalies. However the assumption that sound quality depends only on sound pressure is clearly untrue. We just described the case in a concert hall where randomly incident reverberation creates a sense of random direction of incidence.

However it is widely believed that the direction of sound incidence cannot be perceived in a small room. It is assumed that the presence of room modes makes localization impossible. The statement that low frequencies cannot be localized is easily shown to be true when a sine tone as used as a signal. However it is equally easy to show that an broadband – such as low-pass filtered noise or a low-pass filtered click – is easily localized in most rooms.

In free field we detect the azimuth of sounds at low frequencies by detecting the time differences between the zero-crossings of the pressure waveform at the two ears. Human perception is particularly sensitive to sounds with sharp onsets. We tend to localize the beginnings of sounds. We can localize an impulsive sound at low frequencies in part because it takes time for standing waves to establish themselves. But there is a second reason.

The second reason is that standing waves preserve the original sound direction information if you average over enough of them. Low frequency noise or a pulse will excite many room modes all at once, and the time differences between the ears will correspond to the source azimuth.

However to create spatial impression it is not sufficient to localize the sound source. We must use the sound source(s) to cause the time delay across the listener's head to vary chaotically – at the time rate of the original hall. To do this we require at least two independent low frequency drivers, and a recording made in such a way that the phase of the reverberation varies randomly at low frequencies.

Detecting time delay between the ears

It is easy to perceive sound velocity in the free-field. Human physiology has evolved to determine the direction of sounds. The primary direction cue at low frequencies is the time difference between the two ears. At very low frequencies we perceive these time differences by detecting the time difference between zero-crossings of the pressure wave.

The basilar membrane contains hair cells that open an ion channel when the pressure in the ear canal drops below the average pressure. When the pressure is higher than average, they do nothing. There is a burst of nerve firings whenever the pressure crosses zero in the downward direction. The time difference of this burst between the two ears is detected by a neural circuit exactly half-way between the two ears. The net result is that when the zero crossing occurs at a different time in the two ears, we hear the sound as coming from the right or left of the medial plane.

For the time difference to be perceived we require two conditions: first, that there be sufficient pressure at the two ears to cause significant nerve firings at the zero crossing, and second, that there be sufficient sound velocity (pressure gradient) across the head that the zero crossing occurs at a different time in the two ears.

When both conditions are met – as they are in a free-field (outdoor) condition – we can easily localize the sound. In an enclosed space – a listening room – the situation becomes more complicated, but detection of sound velocity is still possible whenever the above conditions are met.

It is very well known that low frequency sound propagation in a room is strongly influenced by "room modes". These modes have well known spatial variations of sound pressure. Note particularly that the spatial patterns of room modes are independent of the sharpness if the mode in frequency. A particularly nasty high Q room mode will have the exact same spatial pattern as a more gentle resonance at the same frequency.

An example – the large demo room at Harman Specialty Group

For example, lets look at one of the test rooms at HSG. This room has a length of 23', a width of 17' and a height of 10'. Actually there is a dropped ceiling composed of 80% absorptive tiles, with the actual ceiling height of 15'. Because of the very high vertical absorption, all modes in the vertical direction are suppressed in this room.

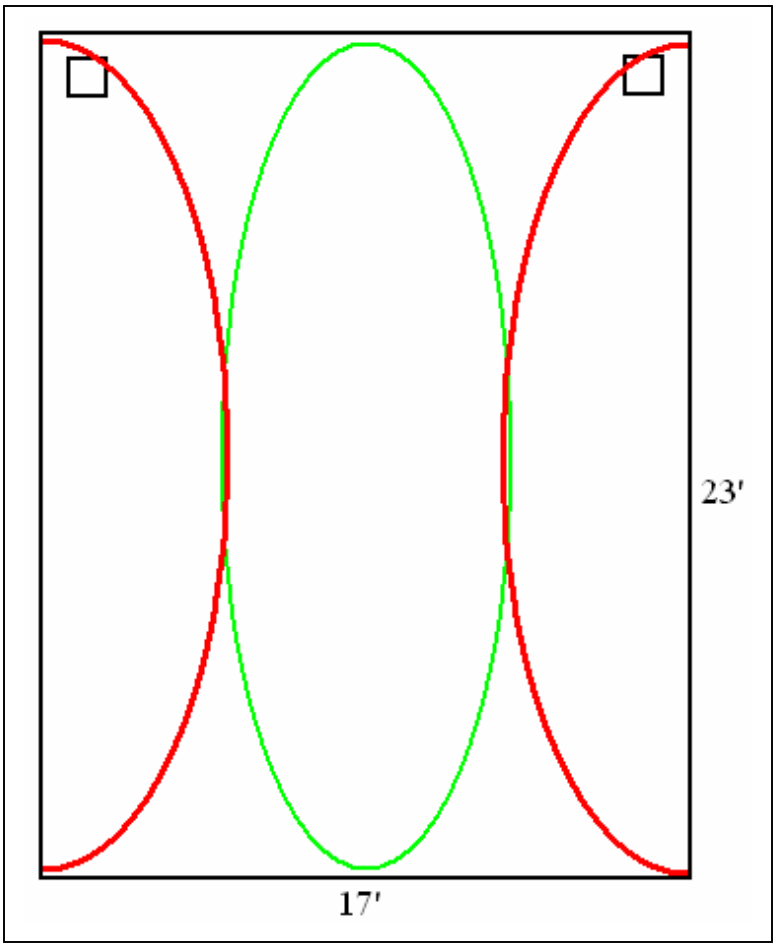

Figure 1: Regions of high pressure (red) and region of high pressure gradient (green) for the 32Hz lateral asymmetric mode in the large demo room at HSG.

Simple theory predicts the modal frequencies below 100Hz. The medial modes for a forward facing listener (normally the combination of the up/down and the front/back modes) are here only the front/back modes, and their frequencies are 24Hz (asymmetric), 48Hz(symmetric), 72Hz(asymmetric), and 95Hz(symmetric). The lateral modes (the left/right modes) have frequencies of 32Hz(asymmetric), 65(symmetric), and 97 Hz(asymmetric). The region of high pressure gradient for the 32Hz lateral mode is shown in green above, and the regions of high pressure for this mode are shown in red. Note that the spatial extent of these regions is independent of the Q (or sharpness of the mode) in frequency.

As can be seen above, the 32Hz lateral mode has a pressure zero in the center of the room, with pressure maxima on the two side walls. These pressure maxima have the opposite sign, so that when the left side has positive pressure the right side has negative pressure. The 65Hz lateral mode is symmetric, and thus has pressure maxima at the side walls, with two pressure minima half-way between the side walls and the center, and a pressure maximum (with the opposite sign) in the center.

The sound velocity in these modes is far from zero. Air responds to pressure differences, and moves from a pressure maximum to a pressure minimum. Thus the 32 Hz mode has a maximum sound velocity (or pressure gradient) in the center of the room, where the pressure difference is the greatest. The 65Hz mode has maximum sound velocity half-way between the center and the sides, and a minimum sound velocity in the center.

In the medial plane for a listener facing forward, the 24Hz mode has maximum sound velocity (in the front/back direction) in the center of the room, and a minimum pressure in the same position. The 48Hz and the 95Hz modes have maximum pressure in the center of the room, and a minimum sound velocity in the same position.

These facts are all well known. But let's add some new ideas. What happens in the center of the room when we excite both lateral and front/back modes at the same time? This is easy if we pick a frequency half-way between the center frequency of the modes, and the modes have relatively low Q. Let's pick a frequency half-way between the 32Hz lateral mode and the 48kHz front/back mode. A loudspeaker in the front left corner will excite both modes equally. Let's put a listener facing forward in the center of the room.

When the frequency is roughly half-way between 32Hz and 48Hz we will get lateral velocity from the 32Hz mode, and medial pressure from the 48Hz mode. The conditions for detecting sound velocity would then be met, and the apparent direction of the bass signal will shift to the left.

What happens if we put another loudspeaker in the front right corner, and excite the two with a stereo signal? In this case something very interesting happens. The 32Hz lateral mode is only excited by the difference signal driving the loudspeakers! If the loudspeakers move in phase, this mode is not excited. The 48Hz front/back mode is only excited by the sum signal.

Thus if the signal driving the loudspeakers fluctuates randomly in phase, the apparent direction of the sound at the listener position will also fluctuate randomly! We will have achieved our goal of reproducing the original envelopment, at least at this frequency.

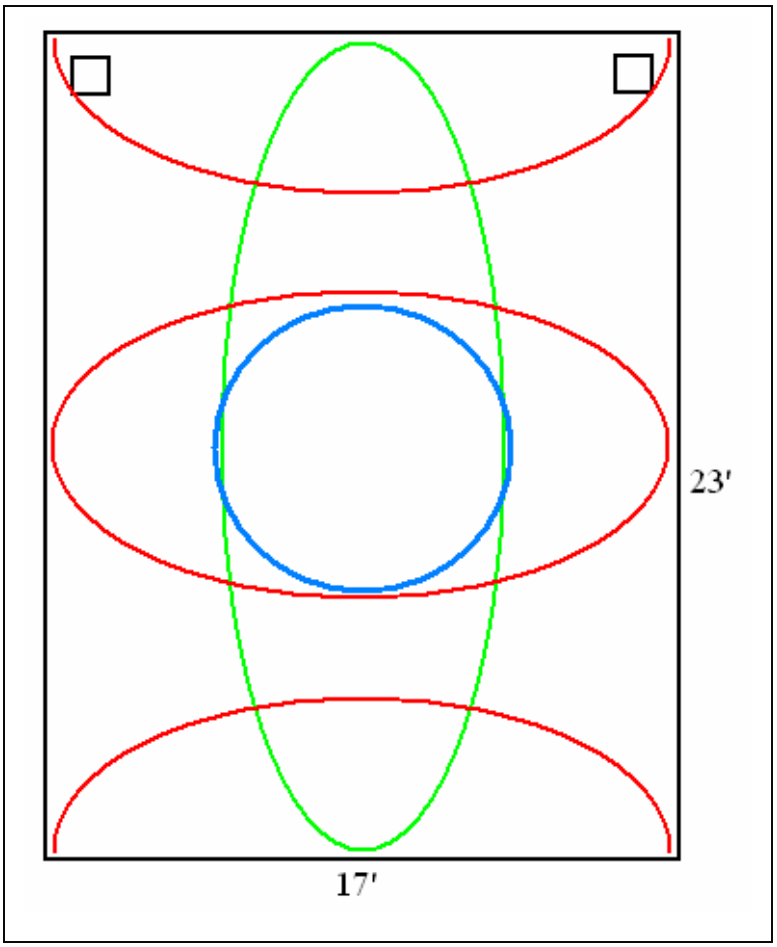

Figure 2: Regions of high pressure from the 48Hz front/back mode (red), with the region of high pressure gradient from the 32Hz asymmetric lateral mode (green). High envelopment will be heard at frequencies between 32Hz and 48Hz in the area where these two patterns overlap (blue).

There are other frequencies at which this trick works for a listener in the center of the room. We see that at 96Hz there is a maximum lateral velocity in the center of the room, since there is an asymmetric lateral mode at that frequency. At the same time there is a front/back mode with maximum medial pressure in the center of the room. At this position and at this frequency we will surely be able to detect envelopment, because the lateral mode is excited by the difference signal in the stereo recording, and the front/back mode will be excited by the sum signal.

In a similar way we can predict all the areas in the room where sound velocity is likely to be detectable for different frequencies – which is any area that has both a velocity component and a pressure component. There are many such areas – but only some of them produce esthetically pleasing spatial properties, because only some of them will have a pressure gradient that is excited from a different component of the stereo signal than the pressure.

The simplest way to excite different modes depending on the phase of two loudspeakers is if the in-phase component of the loudspeakers excite one set of modes, and the out of phase components excite another set of modes. [It is also possible to find loudspeaker arrangements where one loudspeaker excites one set of modes, and the other speaker excites another set – but this requires an asymmetric loudspeaker arrangement. This case will be considered later.]

Let's first consider symmetric loudspeaker arrangements. In this case we get envelopment when the asymmetric lateral modes (which are excited by the anti-phase component of the loudspeaker signals) mix with any medial mode (which are only excited by the in-phase component of the loudspeaker signals. In other words, to create good sound we need the combination of pressure from *medial* modes with velocity from the *assymetric* lateral modes.

The effect of these combinations on the sound in a room is better demonstrated than described. A simple test signal is a frequency sweep, say from 40Hz to 100Hz, where the left channel is about 2Hz different in frequency from the right channel. This signal has constant amplitude in the two loudspeakers but a cyclical phase relationship. Another signal is the sum and difference of the previous signals. For this signal the two speakers vary oppositely in amplitude, and switch phase relationship every half beat. I call this latter signal a wobble tone pair.

The wobble tones are a series of test signals with roughly 2Hz bandwidth, with an alternating phase relationship between the two channels. They are arranged into groups of five frequencies in each band on the CD, starting with 30Hz and going up in two Hz steps. Thus each band on the CD covers 10Hz. When we play this CD through two separate loudspeakers in the front of the room just described, the effect of the velocity and pressure mixing can be easily heard.

Lets look at what happens in the large demo room at HSG. The asymmetric lateral modes have frequencies of 32Hz and 97Hz. These modes both have velocity maxima along the center line of the room. Since the lateral mode at 67 Hz is symmetric, it will not contribute to envelopment for a listener along the center line.

If we are to hear envelopment at frequencies below 50Hz, the 32Hz lateral mode needs to overlap with the symmetric front/back mode, which has a resonant peak at 50Hz. If both these modes have low Q, there will be a range of frequencies over which the velocity of the lateral mode will be present at the same time as pressure from the front/back mode. Figure 2 shows that there are three large areas where the modes overlap in this way. (The region nearest the loudspeakers is not drawn.)

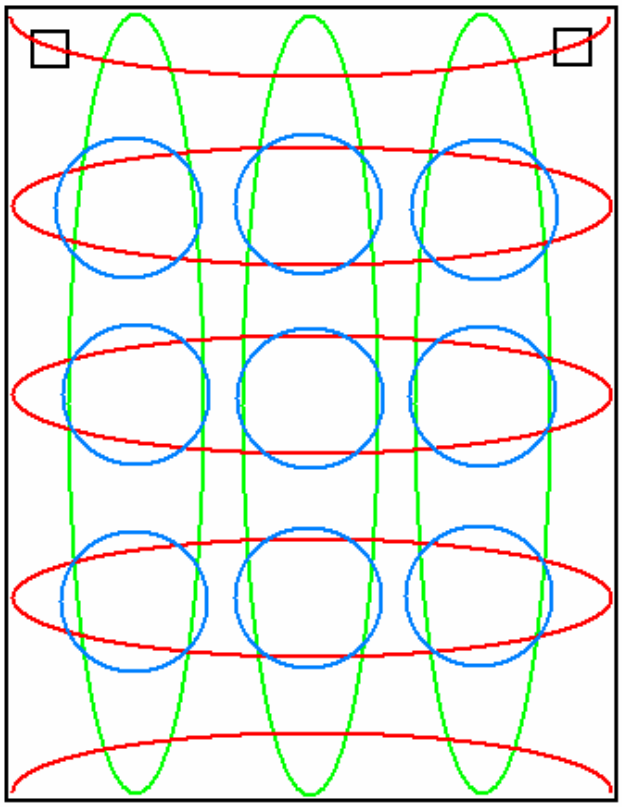

Figure 3: The zones of high pressure in the 96Hz front/back mode (red), with the regions of high pressure gradient from the 96Hz asymmetric lateral mode (green). High envelopment will be heard in the regions where the two overlap (blue). Because there is wide coverage at both 30-50Hz and in the 97Hz region, the room is perceived as spacious throughout.

In this particular room when we stand anywhere on the center line we predict the lateral velocity will be zero at 65Hz, and sure enough the wobble tones at this frequency localize inside the head. There is no rotation, and no external localization. We similarly predict that in three broad areas in the lateral center of the room there will be high spatial motion in the frequency range from 35-50Hz. This is easily observed. As we move laterally away from the center line of the room the motion decreases slowly, in proportion to the lateral velocity of the fundamental lateral mode.

The effect of minimum medial pressure on the sound motion is more subtle. We predict that at 72Hz there will be a minimum medial pressure in the center of the room, and thus low spatial motion. Sure enough, when we find the exact center of the room the motion ceases, and the tone goes back inside the head. This null is relatively sharp.

At HSG we have two rooms we regularly use for tests. The second room was purposely built for this purpose. It is trapezoidal in shape, and heavily damped at low frequencies. It has always sounded terrible. Simple measurements reveal the reason. The front/back

dimension is about 21' and the mean left-right dimension is 19'. This room is nearly square. It was originally set up completely symmetrically, with the listening position exactly in the center of the room. The fundamental frequencies of both the lateral and the medial modes is about 27Hz, with the first symmetric modes at 54Hz.

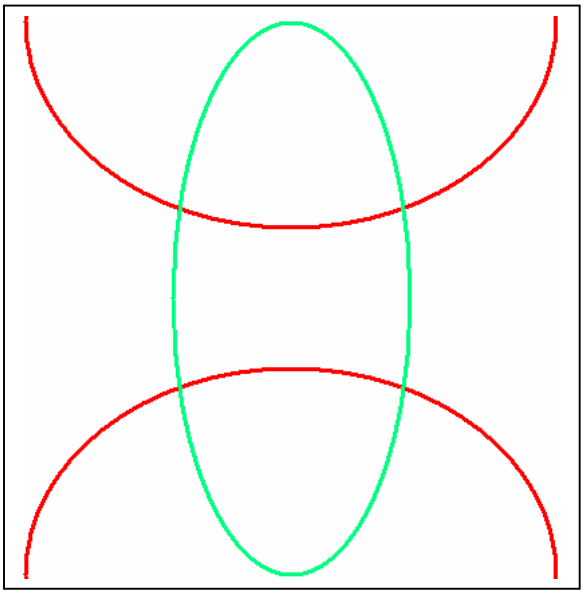

Figure 4: Demo room #2 driven by two speakers in the top corners. The speakers are reproducing a ~30Hz tone which wobbles in phase, so that all possible phase relationships are driven.

The red line shows the boundary of regions of high medial pressure, in this case it is the first front/back mode at about 27Hz. The green line shows the boundary of regions with high lateral particle velocity. In this case it is the first (asymmetric) lateral mode at 30Hz. We will hear envelopment where these two zones overlap. Note that no envelopment is heard at the center of the room, where the listening position was intended to be.

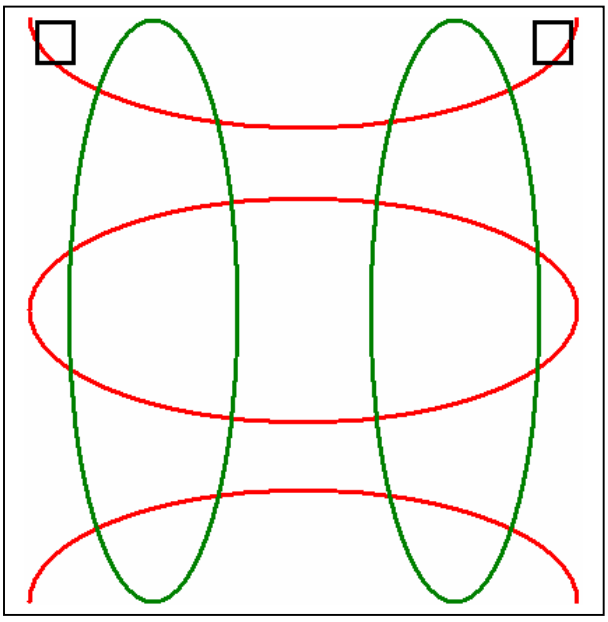

Figure 5: If we put loudspeakers in the corners of HSG demo room #2, an in-phase signal at 52Hz excites the red pressure pattern.

There is also a nearby lateral mode at about 56Hz, but this mode is symmetric and thus it is also driven by the in-phase component of the loudspeaker signals. The green line shows the boundaries of regions of high lateral particle velocity. Notice there is overlap with the medial pressure, but not in the center of the room. However, since both the medial pressure and the lateral velocity are driven by the same signal component, no envelopment is produced anywhere in the room by these two modes.

The marketing department actually sets up the room differently.

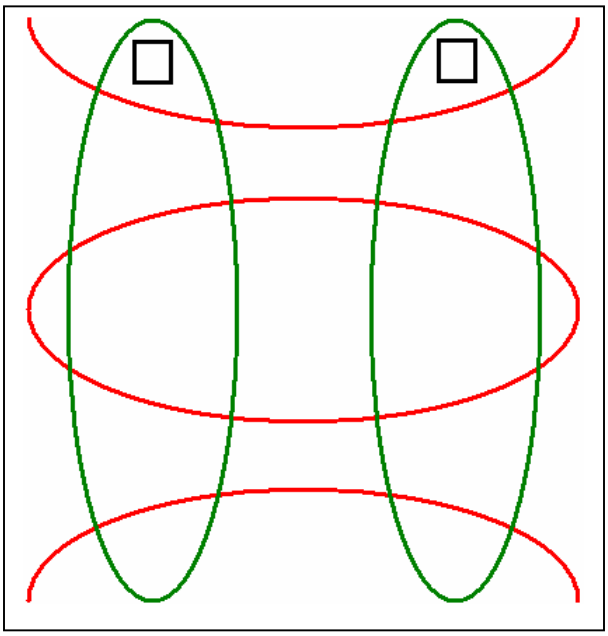

Figure 6: In the usual arrangement, the loudspeakers are placed at the *pressure nulls* of the 56Hz mode. This arrangement is more conventionally visually, but results in this lateral mode being impossible to excite. Even if envelopment would result if the 56Hz lateral mode were driven, it will not be heard with this speaker placement.

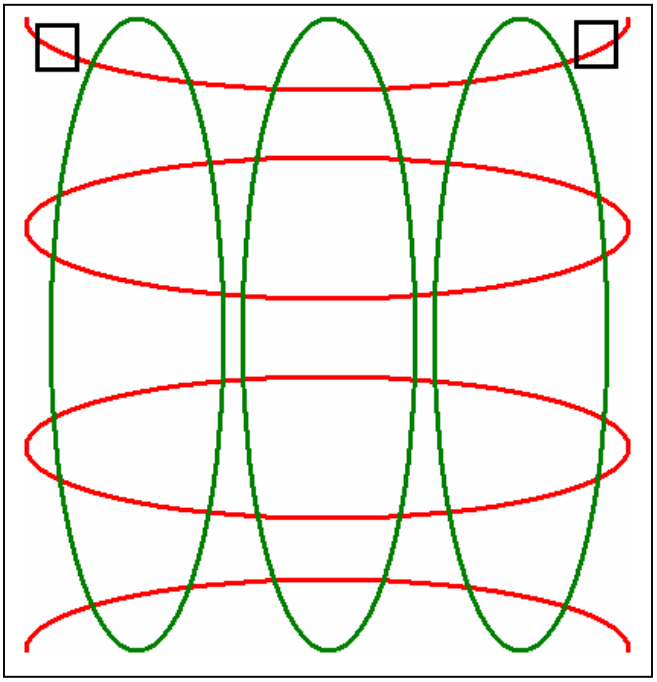

Figure 7: The next asymmetric lateral mode is at 87Hz. This mode is excited by the out-of-phase component of the loudspeaker signals, and produces the velocity patterns shown in green.

The pressure pattern shown in red is produced by the in-phase component of the loudspeaker signals. Envelopment will be heard where these two patterns overlap. Notice there is no envelopment produced in the center of the room, at the intended listening position.

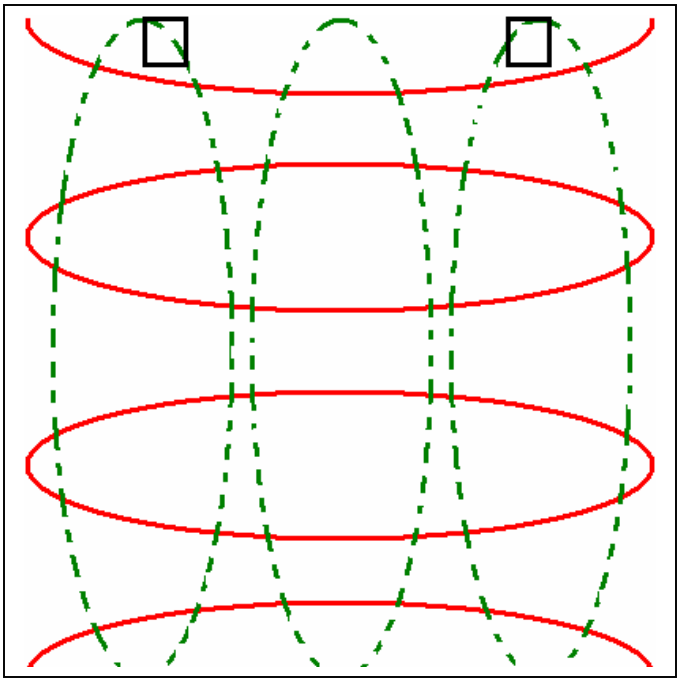

Figure 8: If we put the speakers in their actual positions in the room, the loudspeakers are close to the pressure nulls for the 87Hz asymmetric lateral mode.

As a consequence, this mode is not driven, and no envelopment is heard anywhere in the room!

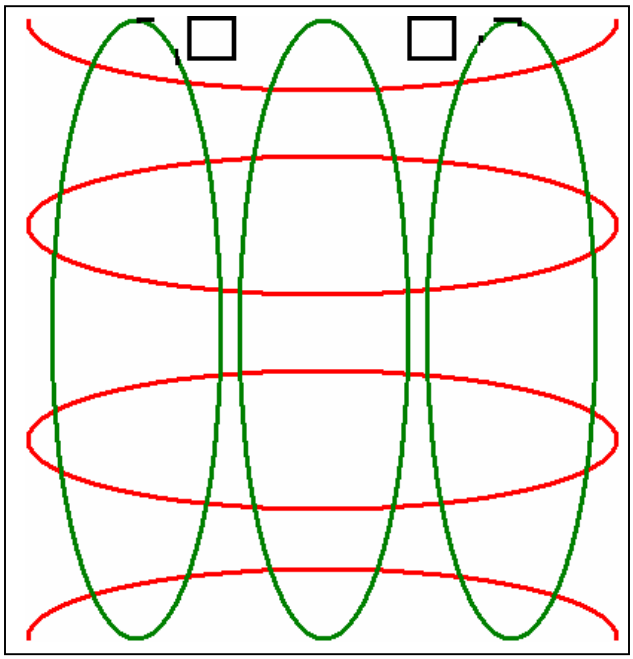

Figure 9: If we put the speakers a bit closer together, the 96Hz mode is strongly driven by the out-of-phase component of the program signals and envelopment will be strong where the green and red patterns overlap.

However in this case the 32Hz asymmetric lateral mode is not strongly driven, as the speakers are too close to the pressure null for this mode.

What if we arrange the loudspeakers asymmetrically?

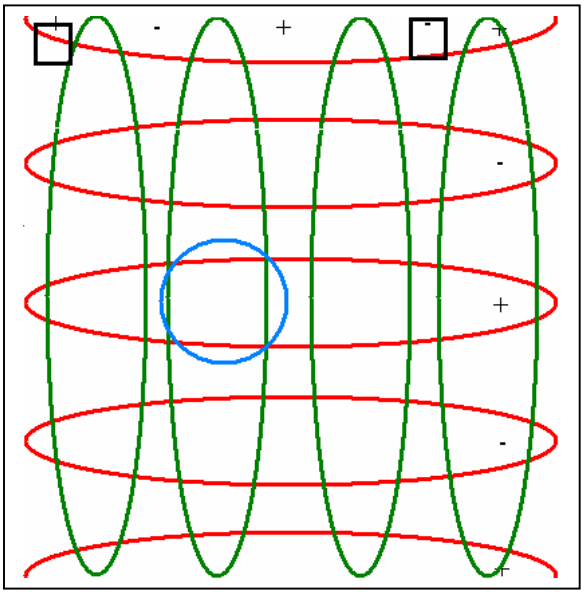

Figure 10: Put the left speaker in the corner of the room, and move the right speaker to the left just enough to place it in the pressure maximum of the symmetric lateral mode at 115Hz.

This mode will now be excited by the out-of-phase component of the loudspeaker signals, and will produce high velocity in the regions marked in green.

The blue circle marks a region where envelopment will be heard at frequencies around 115Hz. This is a possible listening position.

Although the arrangement in figure 9 is good at 115Hz, it is poor at 32Hz, since the listening position is too close to the front/back center of the room. See figure 3. The arrangement in figure 9 is also poor at 87Hz, since the lateral mode at that frequency would be excited by the in-phase component of the loudspeaker signals.

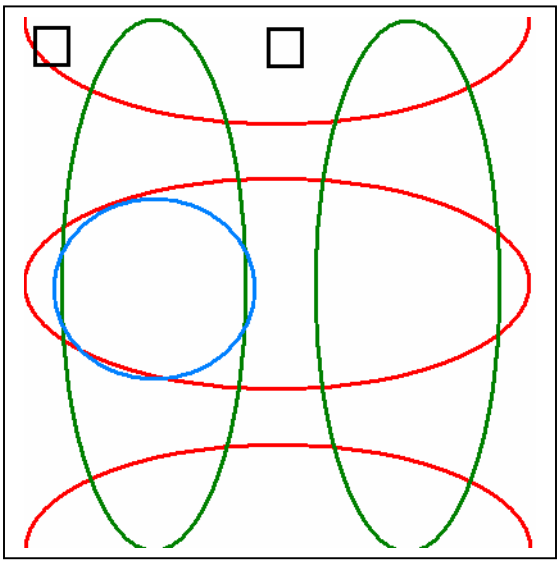

Figure 11: If we put the right speaker in the center of the room, the otherwise symmetric mode at 58Hz is driven by the out-of_phase component of the loudspeaker signals, and the in-phase component drives the front/back mode at 52Hz. There is high spaciousness in the listening position shown in turquoise.

Because many types of music can have strong energy in the 40-70Hz region, this loudspeaker should sound good. In fact, this is the most pleasant sounding speaker arrangement in this room.

Although the arrangement in figure 10 may be the best solution for this room, it is far from ideal. Envelopment is heard at the 40-70Hz frequency range. However there is very little envelopment at other frequencies. The 32Hz lateral asymmetric mode is excited by the left loudspeaker – but there is no overlapping pressure at the listening

position. The mode pair at 87Hz also produces little envelopment. Although the lateral mode at 87Hz will be excited by the left speaker alone, there is no overlapping pressure at the listening position, which is too close to the front/back center of the room.

Although the loudspeaker arrangement in figure 10 is probably the best spatially, it is quite difficult to make it work visually. In addition, the spatial properties would be best described as poor. This loudspeaker arrangement is still a lot better sounding that the original arrangement, where spatial properties were entirely absent. Moral: don't design a square room if you care about audio quality!

The loudspeaker arrangement shown in Figure 10 would probably be improved visually if the loudspeakers were arranged along the longer wall. This type of arrangement will be discussed in the next example.

The author's listening room at home is approximately 15' by 12'. Many attempts to set up the room with the loudspeakers along the 12' wall have not been successful. The sound is always best when the loudspeakers are along the 15' wall. Because this wall is not really 15' long, but is composed of a 12' wall surface with the addition of a 3' bay window, the left loudspeaker cannot be located in the left corner of the room, but must be located about 4' closer to the center. The listening position is thus asymmetric, about 2' to the right of room center.

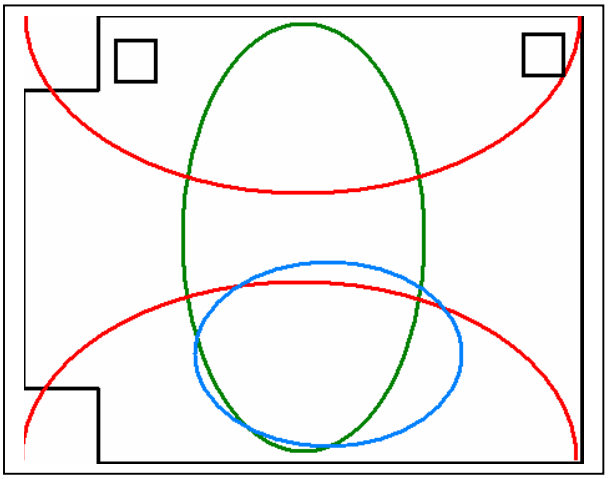

Figure 12: The velocity maximum of the 32Hz asymmetric lateral mode and the pressure maximum of the 46Hz front/back mode overlap in the listening position, shown in Turquoise. LF spaciousness is high in this position.

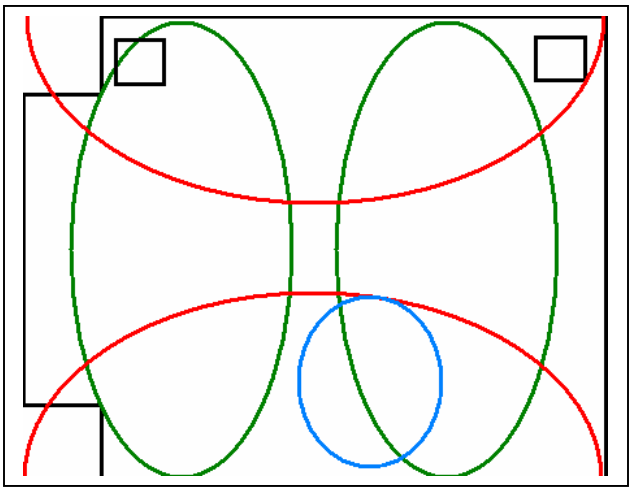

Figure 13: Overlap between the front/back mode 1 at 46Hz, and the second lateral mode at 64Hz. Although there is spatial overlap between the two modes at the listening position, at first it appears that both modes are excited by positive phase in the loudspeakers, so little envelopment is likely to result. In practice however, there is some envelopment, since the left loudspeaker in near the pressure null of the 64Hz mode. This means the mode will be chiefly driven by the right loudspeaker alone. There is some independence between right alone, and the left plus right component that drives the front/back mode.

The next modal frequency with significant velocity from negative phase is at 96Hz. This overlaps the front/back mode at 92Hz, and significant spaciousness is heard.

Because there is significant overlap between pressure and velocity at the listening position in this room, the low frequency spaciousness is high. It is a pleasure to work and to listen in this room.

Room feel and Statistics

It can be seen from the above discussion that it is never possible to achieve both velocity and pressure mixing at all frequencies and at all positions in a room. Fortunately, this is not necessary. A good sounding room will have some motion to the low frequencies in the region the listeners choose to occupy. It does not have to happen all the time to be convincing. The detection of room impression is statistical.

Room impression depends on the detection of sound motion in the reverberant component of a musical sound. We detect as room impression the beautiful shimmer to the reverberation once the direct sound has gone by.

We have spent a lot of time with the LARES system creating this impression in concert halls and opera houses that lack it. The artistic people we have worked with have been

very articulate about how the low frequency reverberant swirl adds a sense of life and emotion to the sound that is enormously important to the artistic impression. In the words of one of them – when the hall is too dry the sound dies. It starts in the pit, but dies there. When the hall is just right the sound leaps out of the pit and into the hall, where it takes a life of its own. It is this impression that we want to bring to sound reproduction.

In a concert hall the swirl is caused by the chaotic interaction between reflections from all parts of the hall. The rate of change of the instantaneous localization can be predicted from the reverberation time and the frequency properties of the original sound. One of the primary purposes of vibrato in wind instruments is to create this swirl in the concert hall, and the effect can be stunning. A good clarinet player does this by moving the bell of the instrument, giving the tone a life in the hall quite different than if he had been entirely still.

To reproduce the sound swirl we must first capture it. Fortunately this is easily done. Most standard recording techniques record the hall sound with widely spaced microphones, and the phase relationships between these microphones reflect the chaotic conditions in the original hall. If we play this signal through independent drivers in our playback room the chaotic phase will produce chaotic motion in the room, just as our wobble tones do.

It is not necessary for the room perception that all the frequencies should work equally well. Just that most of them should work pretty well. Once again, how well the impression works depends on the degree that medial pressure and lateral velocity should mix at the positions of the listener, and the percentage of frequencies that work well. Speaker positions that maximize the lateral velocity at the position of the listener are always helpful.

Spaciousness in the front/back direction:

We have examined speaker layouts and room dimensions for the case where the listener is facing forward. It is however very common that listeners face other directions, at least some of the time. We can extend the method shown here to other directions. To make this work we need at least one more degree of freedom in the recording, and at least one more low frequency driver. Both requirements are met by a professional discrete channel surround system. In our experience the additional speaker(s) should have low frequency capability to at least 60Hz.

The extra “independent” rear channels can be in practice derived from the front channels by adding a time delay. The time delay decorrelates the reverberant component of the low frequency sound, and can dramatically improve the subjective “feel” of a room. The “Logic 7” matrix system exploits this principle.

Room Layout

Consider the first of the rooms described above. The sound is very good. But the side walls are hard enough that the lateral modes propagate well from the front of the room to the listening position. If we add low frequency absorption to the side walls an interesting thing happens. Let's assume there are two loudspeakers in the front of the room. We can measure the sound pressure in the lateral modes as a function of the distance from the loudspeakers, and we find the pressure drops off pretty quickly as we move back in the room. By the time we get to the listening position the lateral modes might be 6dB less than they were closer to the speakers. At the listening position the bass will be dominated by the medial modes, and the spatial impression will be poor.

Placing a subwoofer pair closer to the listening position will solve the problem. In general, we want to put the subwoofers on either side of the listeners! If the back wall is closer to the listening position, put the subwoofers on the back wall. This will maximize both the bass uniformity and the spatial impression.

We have worked in a large room with relatively absorbing side walls. The difference in sound caused by moving the pair of stereo subwoofers to the sides of the listening position was enormous. With the subs up front, the room seemed artificial and dead. With the subs at the side the sound came alive. Quite a dramatic difference.

Both the rooms at HSG that are described here are large by the standards of many home listening rooms. What if the room is much smaller? It has been my fortune or misfortune to always have lived in homes with small rooms. The small studio discussed as the last example is roughly 12'x12' with a 9' ceiling. There is also a 3'x6' bay window at one end of the room. Visually the room appears to be roughly 12'x15'. I have tried many different speaker placements in this room. The only arrangement that gives satisfactory bass is a seven channel system that puts the front three loudspeakers along the 12' wall with the bay to the left. The speaker arrangement is symmetric with respect to the main walls of the room, but does not include the bay.

It is easy to see why this arrangement works, and why others do not. With the speakers essentially located along the long wall of the room the fundamental room mode – the one at 37Hz along the 15' axis – is a lateral mode, with maximum sound velocity at the listening position. The listener is not exactly centered at the pressure null of this mode, so the sound pressure is quite audible, and can be equalized to the match the overall sound pressure. This mode interacts with direct pressure from the loudspeakers, and with the beginning of the medial mode at 46Hz to create good spatial impression at low frequencies.

If we reconfigure the room so the speakers are facing the bay, the first lateral mode will have a frequency of 46Hz, but the listening position will be near the pressure null of the medial mode, which is now the one at 37Hz. With these conditions the spatial impression is poor.

Room Equalization

Although the physical shape of the modal patterns in a room does not depend on the sharpness of the resonances, the frequency variations of the sound pressure and the sound velocity does. The demo rooms at HSG fortunately were not very solidly built. The construction is a single layer of wallboard over studs. This type of construction is quite common because it is inexpensive. There is some flexing of the wallboard at low frequencies, and this flexing absorbs energy. The result is that the low frequency modes are broadened. Both rooms have also been acoustically treated, with fiberglass panels on all the walls, typically 2" thick.

As a result room modes above 100Hz are highly damped. The significant room modes in these rooms are well predicted by simple theory, and are relatively broad. However the damping provided by the flexing wallboard is frequency dependent. The damping is relatively good at most frequencies, resulting in modal Q values below 10. However for some reason the wall becomes quite stiff at 48Hz, and this mode has a Q of 16 or more.

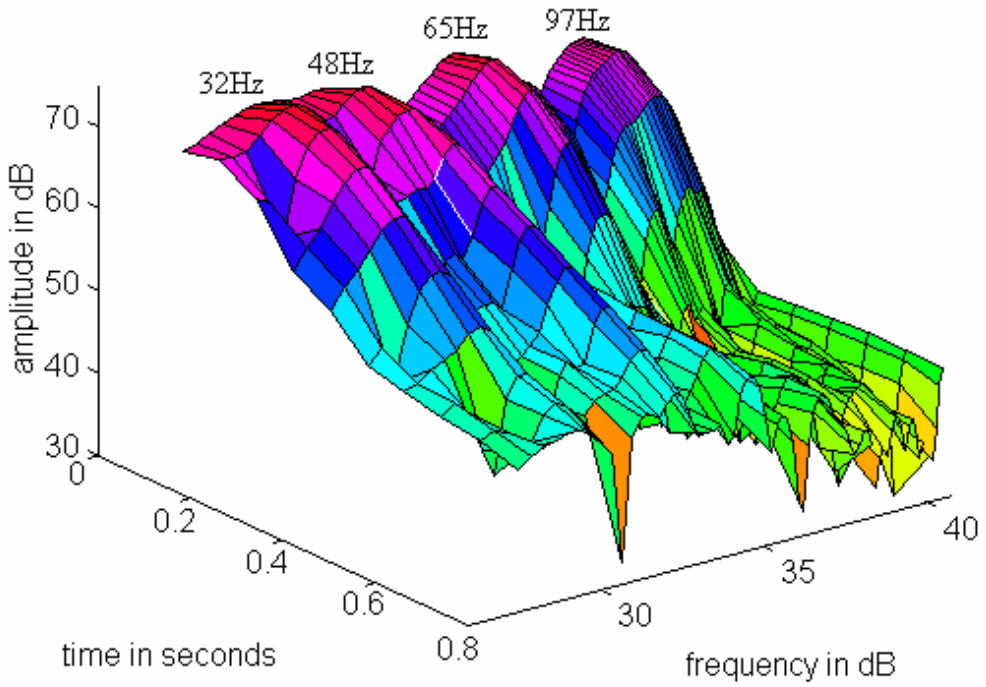

Figure 14: Waterfall plot of the sound decay in the large demo room at HSG. Note that only the first three modes have significant time dispersion, with the 48Hz mode having a particularly audible decay. This microphone position did not pick up the mode at 24Hz.

Q is a measure of the sharpness (on a logarithmic frequency scale) of a resonance, which is related to how quickly the mode decays. In fact, probably the best way to measure Q is to measure the decay rate.

Q = 24/(dB of decay each cycle)

Thus if a resonance decays one dB every cycle of oscillation, the Q is 24.

We can calculate the "reverberation time" of this resonance by finding the time it takes to decay 60dB.

RT = Q*60/(24*frequency)

In our example above, the 48Hz resonance with a Q of 16 has a RT of 0.83 seconds. Since the room as a whole has a reverberation time below 0.3 seconds, this resonance sticks out like a sore thumb. It gives a very audible buzzing sound to low frequency impulses.

The resonance at 48Hz is not the only one that sticks out. The 24Hz fundamental has a Q of about 6, with a corresponding RT of 0.65seconds. This resonance is also quite audible, as is the fundamental lateral mode at 32Hz.

As we said earlier, this room really sounds quite good with no treatment. But what if it was made with cinderblock instead of wall board? What if in and effort to "improve" the acoustic performance a double layer of wallboard had been used? We could easily see RT values at the resonant frequencies above 1 second.

Classically the most audible effect of high Q resonances at low frequencies is bass boom. When a bass guitar or other instrument with a steady pitch happens to excite one of these modes the sound pressure (in some parts of the room) can be very loud. Impulsive sounds also will excite these frequencies, which "boom out" their signature. When there are several resonances at enharmonic frequencies the resulting boom can be pretty unpleasant. As was explained in the first part of this note, high Q also reduces the frequency range over which modes can overlap, reducing the spatial properties of the room.

What to do? The traditional approach is to measure the power response of the room (with 1/3 octave pink noise for example) and to flatten the response with some form of electronic equalization. This can help, but it does not solve the fundamental problems.

Another approach recently advocated is to find speaker placements for the subwoofer(s) that minimize the coupling to troublesome modes. The result reduces the peak height of these modes, but does NOT broaden their width in frequency, or reduce their ringing time if they happen to get excited. A major disadvantage of this technique is that it completely eliminates the spatial response of the room.

Electronic damping

Fortunately, to the degree that the room modes can be described by a minimum-phase resonance, they can be damped electronically by introducing a dip in the speaker power

that has the identical frequency and Q. The precision of the frequency and Q that has to be found is beyond the scope of this note – but we want the frequency as accurate as possible – within 2Hz say. The Q can be much less accurate, but it is probable that accuracy to 10 or 20% is good enough.

Considerable experimentation with the level of the dip to be used in the damping circuit has shown that the dip should be proportional to the measured Q value.

Once frequency and Q values have been found for all the troublesome resonances (those with RT values above 0.5 seconds or so) we can install corresponding dips in the electronics and – presto! – the resonances exhibit much lower Q. The lower Q is evidenced in all respects. The power response becomes more uniform, and the annoying buzzes and rattles are eliminated in the time response to an impulse. Spatial properties also improve.

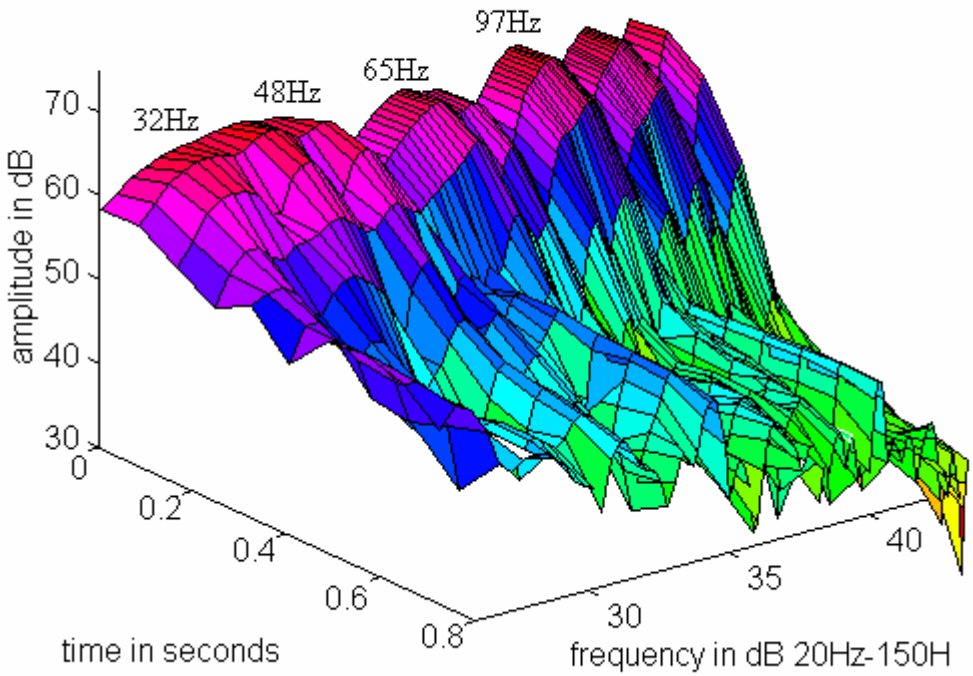

Figure 15: Waterfall plot of the large demo room at HSG after the modes at 32Hz, 48Hz, and 65Hz have been electronically damped. Notice that the time response has dramatically improved. The improvement is audible throughout the room. The bass no longer "boinks", and it is more uniform throughout the room. The power response at this measurement point has improved somewhat. Further equalization with low-Q filters might improve the power response at this microphone position, with the risk of making the response worse somewhere else.

Conclusions;

Although widely held to be unnecessary or impossible, reproduction of envelopment at low frequencies in small rooms can be achieved, particularly with a multi-channel sound system. Successful results depend on: 1. having an input recording that includes at least two channels where the reverberation is independently recorded, and thus uncorrelated with the other channels. 2. The presence of independently driven room modes that overlap in such a way that the lateral pressure gradient of one mode combines with the pressure of another. In the case of two channel stereo, the best results usually occur when an asymmetric lateral mode (driven by the difference signal between the loudspeakers) creates a pressure gradient at the listening position, and a medial mode (usually a front/back mode) supplies the pressure. Ideally both modal systems should be broad enough in frequency that there is a substantial frequency overlap, as well as a spatial overlap. Such spatial and frequency overlaps occur in rectangular rooms of various dimensions, but are rare in rooms that are close to square in dimension. Putting the front speakers along the long wall of a small room can be helpful, as can a somewhat asymmetric speaker layout. In many rooms it can be helpful to place low frequency drivers at the sides of the listening position rather than at the front of the room. Where high Q modes exist it is useful to damp the modes electronically by an inverse filter with precisely the same frequency and Q.

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