Reverberation, modal frequencies and the small room

Room modes are important because they affect the quality of sound in the room by reinforcing the reverberation at some frequencies and not others. The frequency concentrations caused by room modes are known as *eigentones*.

In real rooms, reverberation is not the uniform decay of a diffuse sound-field: it is instead, the non-uniform decay of sound energy concentrated into narrow bands centred on the resonant room-mode frequencies.

As direct sound energy ceases, the decaying sound energy concentrates at the modal frequencies, each mode decaying at a rate appropriate to the absorption that is effective in that mode. Even if the energy in the room is evenly distributed at the commencement of the decay, the energy will rapidly redistribute itself into these well defined spatial patterns. These favoured sound patterns can be thought of as a room's bad habits, to which it will effortlessly return unless prevented from so doing. In an analogy with light, the room is said to, "colour" the sound, because coloured light has some frequencies reinforced with respect to others. These eigentones therefore colour the reverberation response of a room; leaving a characteristic acoustic signature on everything recorded or reproduced in it.

The eigentone frequencies of any room may be calculated using Raleigh's equation,

$$\mathbf{f_n = C/2 \times (\, a^2/L^2 + b^{\,2}/W^{\,2} + d^{\,2}/H^{\,2})}$$

where $\boldsymbol{L}$ = room length (the longest dimension)
$\boldsymbol{W}$ = room width
$\boldsymbol{H}$ = room height (the shortest dimension)
$\boldsymbol{C}$ = velocity of sound

and ***a*, *b*** and ***d*** are integers 0,1,2,3,4,....etc., substituted in turn. Thus, the lowest room mode frequency is obtained by setting ***a* = 1**, ***b* = 0** and ***d* = 0** giving,

$$\boldsymbol{f = C/2 \times 1/L} \text{ and so on….}$$

Taking the dimensions of a room, we can generate a table as shown below.

Axial (first-order) eigentones for a small room

Small room

Axial Mode	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
L	25	45	90	135	181	226	271	316	361	406	451	496	542	587	632	677
W	37	70	139	209	278	348	418	487	557	626	696	765	835	905	974	1044
H	42	73	145	218	290	363	436	508	581	653	726	799	871	944	1017	1089

Large room

Axial Mode	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
L	2.3	4	8	12	16	20	24	28	32	36	40	44	48	52	56	60
W	2.4	5	9	14	18	23	27	32	36	41	45	50	54	59	63	68
H	3.7	7	13	20	27	33	40	47	53	60	67	73	80	87	94	100

Frequencies in Hz, dimensions in metres

The data in the upper table (and Figure 16 long bars, which illustrates the same data graphically) illustrate a vital point: that the first few room modes are relatively widely separated, whereas, at higher, harmonic modes, the frequencies merge together until they become a virtual continuum.

Now compare this information with the data in the lower table (Fig. 16, short bars); in which the modes have been computed for a large theatre. Once again, the same pattern emerges: a few, widely spaced, low-order modes, gradually coalescing into a continuum of higher order modes. In each case the higher modes are not so troublesome because they are relatively constant across the entire frequency band. It is the first few, low-order modes which do the damage because they are distinct and separate.

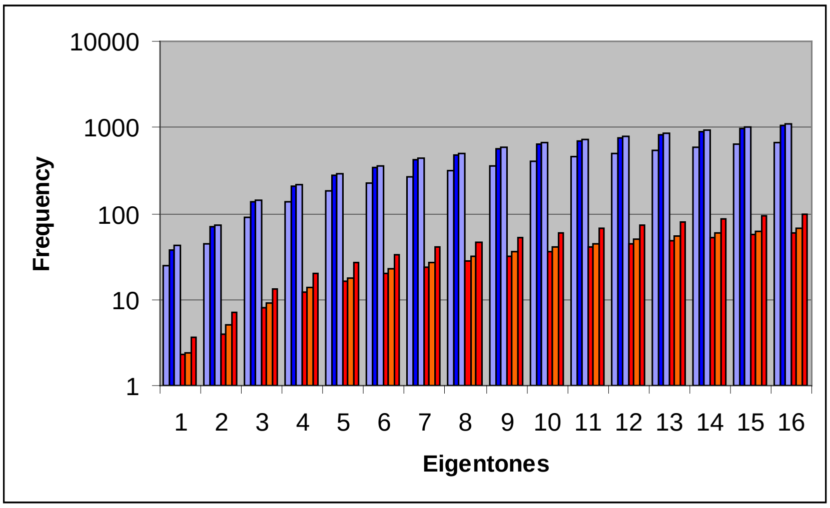

Figure 16 - First sixteen eigentone frequencies for a small room (long bars) and a large room (short bars)

Figure 16 illustrates why there is a major difference between a large room (like a concert hall) and a small room (like a small studio) because, in the former case, the first few, low-order, widely-spaced room modes are sub-sonic. This is very important because it means that these room modes are both rarely excited (because music and speech rarely contains energy at these frequencies) and are in any case inaudible. In a small room, conditions are very different and eigentones are well within the frequency range of the musical gamut. Herein lies the real difference between small and large rooms and explains why the treatment for a small room is actually much more difficult than that for a large room. The effect of low-order room modes is especially relevant and troublesome for amateur, project studios and broadcast speech studios which are often confined to a small space in which the wavelengths of normal instrument frequencies approximate to the dimensions of the room. Their treatment in the context of a small control room or a small talk or music studio is covered below.

Measurement of reverberation

Reverberation measurements should be made using 1/3rd octave noise which should be reproduced for long enough to establish an equilibrium condition. For a small room this is not long and 1 second should easily be enough. This is fortunate because the stimulus needs to be at a loud level to be effective. This stimulus should be abruptly terminated and the response of the room measured as the sound decays; remembering that ***Rt*** equals the time the sound takes to decay by 60dB. Ideally, a measuring microphone should be used but a modern omnidirectional condenser microphone of good quality will do.

In practice, it can be quite difficult to achieve these measurements satisfactorily; the first problem encountered is often lack of dynamic range. Take a room with a background noise level of 55dB. To be able to measure a -60dB point in the decay, the noise source acoustical output level would have to be more than 120dB (to give a measurable margin of 5dB above background noise level). This is a requirement very hard to meet for practical amplifiers and loudspeakers. To avoid having to resort to impractically high volume levels for the stimulus, instead of measuring the complete decay, it is normal to measure the 15, 20, 30 or 40dB decay point which afterwards is extrapolated to 60dB. The validity of this methodology relies on two assumptions a) that an ideal sound decay will form a straight line when amplitude is plotted on a logarithmic axis; and b) that the part of the decay that we used is representative for the entire decay. Whilst the first of these assumptions is safe enough, the second isn't always. To avoid confusion it is common practise to specify the decay point used to derive the measurement as a ***T20*** test (for a decay time measured at -20dB), a ***T30*** test (for a decay time measured at -30dB) and so on. Decay may be inspected by recording the room response to the DAW and inspecting the amplitude waveform on the computer display.

An alternative technique is to excite the loudspeakers with wideband pink-noise, or even a wideband impulse signal, and filter the microphone signal into 1/3 octave bands for measurement. Or, even better, perform a Fourier transform on the decay signal in order to detect any eigentone signals present in the general decay. Once only the province of specialist test houses, the wide availability of software Fourier transform analysers on standard personal computers has opened this technique to most musician-engineers. The results of this technique are given for a small (untreated) studio below. Figure 17 illustrates the reverb tail (with the time-base and amplitude varied after the first 50mS to reveal the structure of the end part of the reverb trail.

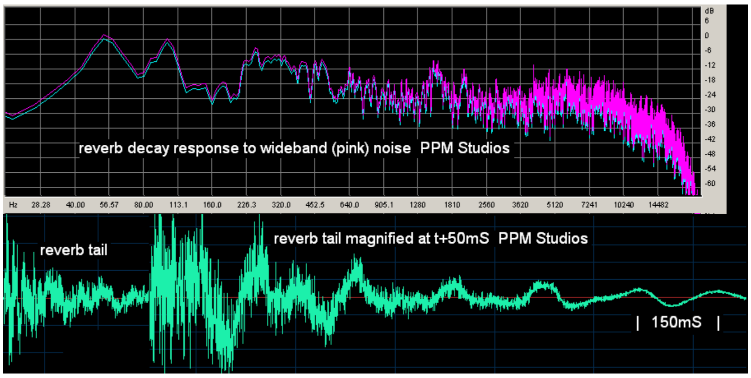

Figure 17 - Using the DAW as an acoustic measurement system

The upper part of the diagram illustrates the frequency response; derived by FFT transform of the lower signal. Clear room modes are evident at just below 50Hz and just below 100Hz. These tally with the first two computed axial modes for the small room calculated above (which were based on the dimensions of the room measured here.) Interestingly, even in the time-domain, it is possible to perceive the virtually sinusoidal evidence of the lower of these eigentones as the decay dies away. The frequency response curve also illustrates a complex of higher-order room-modes in the 200 - 250Hz range.

In a further analysis, a Fourier analysis was performed over 50mS intervals during the reverb tail. Each of these analyses is illustrated in Figure 18, so that the consecutive series of transforms appear down the page.

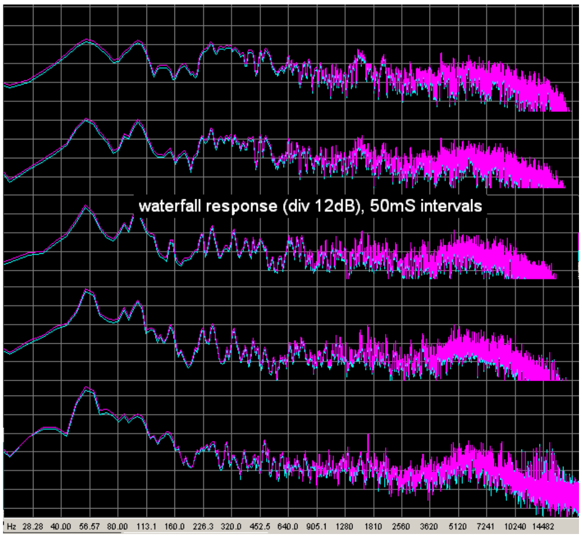

Figure 18 - Fourier analysis performed over 50mS intervals during the reverb tail

If this type of analysis is presented in three dimensions so that time is the z-axis which is nearly perpendicular to the page, it is possible to create what is called a *waterfall response*, which gives a unique insight into the reverberant energy as it decays in the room as shown in Figure 19. Notice how the peak caused by the first-order axial mode remains dominant as the rest of the reverberation dies away. This type of analysis is not available in most standard audio editing and recording software but low-cost software, or even freeware, is available to analyse recorded sound-files and create waterfall responses.

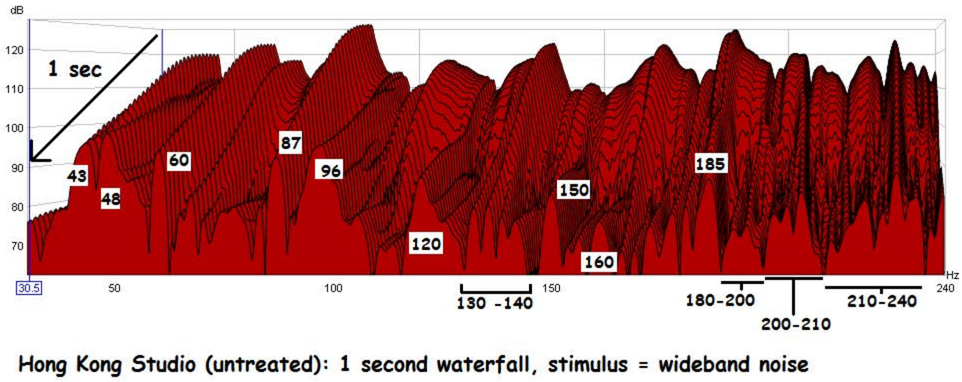

Figure 19

The BBC issue advice on the acceptable reverberation limits versus frequency for music and speech control rooms and this is summarised in Figure 20: the shaded area indicates the acceptable limits. Because of the nature of the standing waves, these measurements should be made at several different microphone positions and the results averaged.