

Sound in rooms

The behaviour of sound waves hitting a surface was touched on in the last article and WSDG's DIRK NOY now uses these surfaces to build a real room where a number of additional characteristics can be observed and studied. He picks on two fundamental aspects of room acoustics — reverberation and standing waves.



Figure 4. Control Room A, SAE Zurich, Switzerland.

The acoustic properties of room surfaces regarding their reflection, absorption and diffusion characteristics strongly influence reverberation. Reverberation is created when a sound (like a handclap) is produced in an at least partially enclosed space causing a substantial number of reflections slowly decaying in level as the sound is absorbed by the walls and the air. The duration of the decay, or reverberation time, receives special consideration in the architectural design of acoustical spaces such as concert halls and recording rooms, which need to have specific reverberation times to achieve optimum performance for their intended use.

Reverberation Time can be measured as follows (see Figure 3): a constant noise signal is reproduced in the room to be measured — for simplicity let's assume that the Sound Pressure Level is at 100dB. The sound source is switched off and simultaneously a stopwatch is started. The decay of sound level is closely watched on an SPL meter, and when the sound level reaches a value of 60dB below the initial value (in this example, at 100dB - 60dB = 40dB), the stopwatch is stopped. The measured time that was necessary for the 60dB level drop is defined as being the RT60 Reverberation Time. The RT60 value is a function of frequency: most rooms display longer reverberation times at low frequencies than at high frequencies.

Before the age of laptop computers and iPhone apps reverberation time could only be measured using a level recorder — a plotting device that graphs the noise level against time on a slip of moving paper. A loud, impulse noise was produced (using a balloon or a blank pistol) and as the sound died away the trace on the level recorder showed a distinct slope. Proper mathematical analysis of this slope revealed the measured reverberation time.

Currently, the measurement method used most commonly is by employing a two channel system. This method comprises of a sound being reproduced by an

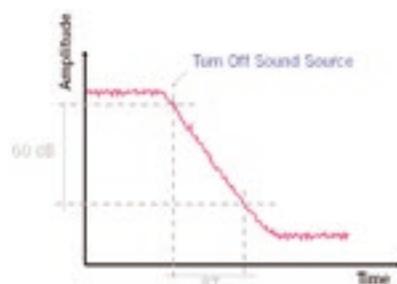


Figure 1. Measurement of reverberation time.



Figure 2. Plotting device (Bruel & Kjaer 2317 Level Recorder).



Figure 3. Omnidirectional measurement loudspeaker, KKL Luzern Concert Hall.

omnidirectional loudspeaker into a room and a recording of the sound in the room is made by use of an omnidirectional measurement microphone and that is mathematically compared to what was sent to the loudspeaker. These two channel measurement systems employ an algorithm named FFT (Fast Fourier Transformation) to derive the impulse response of the room. From the impulse response, the reverberation time can be calculated.

Two aspects of a room's reverberation time are studied. First there's the absolute duration of the reverberation time in seconds (for each individual frequency band or as a mean value). The ideal duration of the reverberation time is not an absolute number but a range of values (minimum and maximum) that is suited for the volume of the space and the purpose of the space. Spaces that are acoustically too lively are just as undesirable as acoustically dull spaces.

The second and often neglected aspect for assessing reverberation is the linearity of the reverberation time over frequency. A space with a very short reverberation time at high frequencies and a very long reverberation time at low frequencies is considered to sound very unbalanced, although the mean value might even be in the correct ballpark.

For control rooms and similar production environments the ITU/EBU standards body has specified Reverberation Time profiles, giving a maximum and minimum as a deviation from the mean value. The deviation may be greater at the high and low

end of the audio spectrum, as shown in Figure 5.

A standing wave in acoustics is formed from the following conditions: acoustical energy is present in the system; reflection takes place at the boundary surfaces; and the respective sound path equals an integer multiple of the half wavelength, meaning that the half wavelength must properly fit between the boundary surfaces.

This phenomenon is known as a standing wave, Eigentone or room mode. At given spatial dimensions (length, width or height), the associated fundamental frequency can be calculated as: $f[\text{Hz}] = c/2 \times \text{Dimension [m]}$ where c is the speed of sound (344m/s). Each fundamental frequency forces the inclusion of the integer multiples, the so-called harmonics, meaning that with any (already unwanted) fundamental standing mode frequency we get the double, triple, etc. frequencies for free. In terms of standing waves, rooms with an equal pair of dimensions (such as the same width and height) are likely to show a strong accumulation of modal effects. Even less ideal is a cubic room.

The introduction of a sound source, such as a loudspeaker, changes the playing field in that a number of additional questions arise: Where should the loudspeakers be located in relation to the boundaries? Should the loudspeakers generate frequencies in the modal range? How strongly can the loudspeakers excite the standing wave frequency? To answer these questions, specifically for rooms that are non-rectangular, a robust toolset is needed, which can be found in the Boundary Element (BEM) and Finite Element Methods (FEM), for numerically solving the wave equation.

Figure 7 shows the predicted pressure distribution for a room based on a BEM analysis. The plan view represents the pressure distribution at ear level as seen from above. The section view represents the pressure distribution on the symmetry-axis as seen from the side. You can clearly see a pressure node (blue) close to the listening position, the treatment suggested is a low frequency absorber at the antinode locations (red zones at ceiling and front wall).



Figure 5. Reverberation time Control Room A, SAE Zurich, Switzerland.

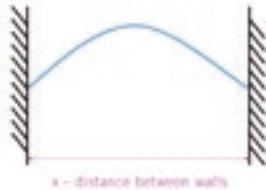


Figure 6. 1-dimensional standing wave (fundamental).

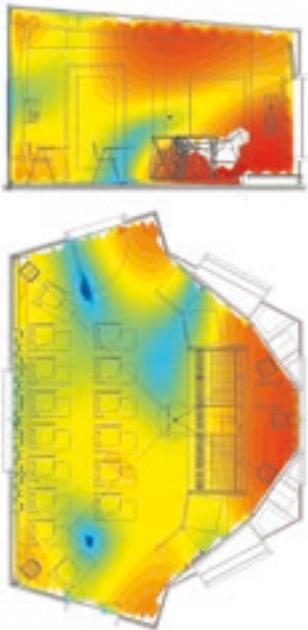


Figure 7. BEM solution for 72Hz pressure distribution: blue = node; red = antinode. Control Room A, The Art Institute of Michigan; ABEC, www.randteam.de

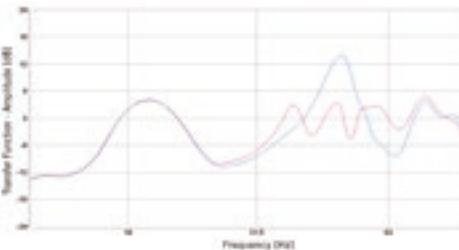


Figure 8. Corrective measure by active absorber: blue = before correction; pink = correction applied. Ovåsen Studios, New York, USA.

Ideally, the modal distribution of a room will be tailored so as to not have strong culminations of fundamentals and harmonics at one single frequency. A number of tools are available to optimize room ratio planning, mostly using a spreadsheet based approach (e.g. <http://www.wsdg.com/dynamic.asp?id=resources/technology/roomode>). Modal effects or standing waves can also be treated by active or passive low frequency absorption where the exact location and type of the absorber must be carefully analysed. Passive low frequency absorbers are likely to be of the membrane resonator type, employing a material (rubber, metal, wood, etc.) that vibrates in resonance with incoming acoustical energy, the vibration is then damped inside the unit and transformed to heat with the result that acoustical energy is removed from the room. A recent development is the availability of Active

Low Frequency Absorbers. In an attempt to properly define the term 'active absorption', we state that an absorber is active if its efficiency can be increased by external energy. An active low frequency absorber consists of a detector device (such as a microphone) that listens to incoming sound, and when an unwanted standing wave build-up is detected, a directly coupled subwoofer loudspeaker will produce phase-shifted energy that subsequently cancels out the unwanted Eigentone. An example of such a product is the BagEnd eTrap.

In addition to the Reverberation Time and the modal behaviour (probably the most important aspects) there are a number of other parameters that might be employed to describe room acoustics in a particular space. One popular measure is the STI (Speech Transmission Index) distribution for analysing speech intelligibility. Clarity (C80) and strength (G) are also relevant for speech intelligibility. Additional parameters such as Lateral

Fraction (LF) are important for classical music performance and recording spaces and even more parameters and effects are observed when bringing loudspeaker systems into our Control Rooms. These will be focused on in the next article in the series. ■

Walters-Storyk Design Group (www.wsdg.com), founded by John Storyk and Beth Walters, has designed more than 3000 media production facilities in the US, Europe, the Far East and Latin America. WSDG credits range from the original Jimi Hendrix Electric Lady Studio in Greenwich Village to New York City's Jazz At Lincoln Center performance complex, broadcast facilities for CBS and WNET, and corporate clients such as Hoffman-La Roche or T-Mobile. Recent credits include private studios for Timbaland's Tim Mosley, Tracy Chapman, Aerosmith, Green Day, Jay-Z, Goo Goo Dolls, Bruce Springsteen, R. Kelly and Alicia Keys. The firm maintains offices in New York, San Francisco, Argentina, Brazil, Beijing, Switzerland and Spain.