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CONGARD CODE MANUAL

Addressing the problems inherent to implementing a multichannel sound system in a small room, and a practical guide for performing and optimising such an implementation

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1.1 The #1 Home Theater problem

- Shrinking a commercial cinema sound system to install it in a small room
- Small room acoustic issues: These are mainly as follows

1.2 The LF problem

- Presence of room modes for f<100 Hz up to f<200 Hz according to room size
- The smaller the room, the more severe the issues. Ex: 1st mode @ 20 Hz for 8.5m length, 40 Hz for 4.25m long

1.3 Early reflections

- Reflections reaching the listener with a delay <20 ms are perceived individually, and provide some signal blurring due to interference with the first wavefront. The smaller the room, the nearer the walls, and the shorter the reflections

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1.4 Loudspeakers

- There is only one proper location for the most essential front (L,C,R) loudspeakers: Behind the screen. (please refer to other .ppt documents about this)
- Typical commercial theater (cinema) loudspeakers have a bandwidth of 40 Hz to 20 kHz
- Having 3 sound sources reproducing frequencies 40 Hz $< f <$ 100 Hz located behind the screen is generally not practical:
    - The axis of the room is not the ideal position for a sound source operating below 100Hz, as it is exciting major room modes
    - The position of the speakers is dictated by the screen position, whether or not it is practical
    - The required size of speakers to provide a LF extension down to 40 Hz is not convenient
    - No stereophonic information exists below about 150 Hz, so 2 sound sources are redundant

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2) Common practices

2.1 Bass management

- Consists in cutting off the LF signals at 80 Hz from the information channels, mixing them with the LFE channel and feeding the resulting signal to the subwoofer
- This allows the use of reduced sized main loudspeakers, that can be easily placed behind the screen
- The LFE and L,C,R signals do not have coherence, phase cancellation can randomly occur
- The phase coupling of the L,C,R speakers with the subwoofer depends on their respective location, on the cutoff functions (generally Linkwitz-Riley 4th order), on the frequency and on the phase response of the main speakers. Cancellations often occur.
- Finally, AV processor manufacturers are often not disclosing what they are actually doing in bass management. Some propose an adjustment of the cutoff frequency (separate for each channel), others offer a level of adjustment in the mix.

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2.2 Subwoofers at the rear

Some extensive work and practice has been made about « filling the dips » in the frequency response, dips that are due to acoustic cancellations created by standing waves , or « modes ».

A typical approach is to increase the number of subwoofers, placing some in the rear corners of the room.

The effect of this is a much more linear measured frequency response.

However, the measurements are performed in the frequency domain, whereas the sound is a variation of air pressure level with time.

The problem is, if we perceive two or more successive waves of low frequency sound with impact (blasts, cannon firing, etc.) we lose the physical sense of impact as we are not reached by a single wavefront.

Of course, the rear subwoofers can be time-delayed so as their wavefront reaches the listener at the same time as the front wave. However, this works only for a « sweet spot », and not for the rest of the audience where the delay settings should be different.

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2.3 Room equalisation

Room equalisation is about making the frequency response as near as possible to a predefined « target » curve.

Again this induces issues related to making a difference between the time domain and the amplitude domain.

- If an early reflection (<20 ms) of sufficient amplitude reaches a listener after the 1st wavefront generated by a loudspeaker, it will create an artefact in the amplitude frequency response, either a dip or a bump, or both
- The ear-brain perception system deciphers a difference between what it considers as essential, i.e. the first wavefront and what is a reflection, the importance of which is minimised.
- Equalisation to compensate for the frequency response defects only affects the first wavefront, the most perceived one.
- Modifying the first wavefront frequency response, if it was initially correct, is likely to induce severe colouration, masking, and all the unwanted effects of an uneven response
- Averaging the measurements does not resolve this problem

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The Congard Code solution

1) Setup

1.1 Front loudspeakers management

- Place 3 x limited-range (100 Hz-20 kHz) speakers behind the screen, where it is optimum for the sound-image coherence
- Use a DSP crossover to separate the bass frequencies (<100 Hz) from the limited-range ones. It should be provided with a comprehensive input-output routing and a possibility to sum the channels
- Set all channels of the AV processor as « large »
- Adjust the lip-sync delay of the processor on the centre channel with the lip-sync tool
- Allot a subwoofer to reproduce the L,C,R channel signals from 40Hz to 100 Hz via the crossover. Place it below the front speakers, preferably offset from the main room axis
- Alternatively, use an array of smaller subwoofers with a step < 3.4 m

Basic 5.1 configuration

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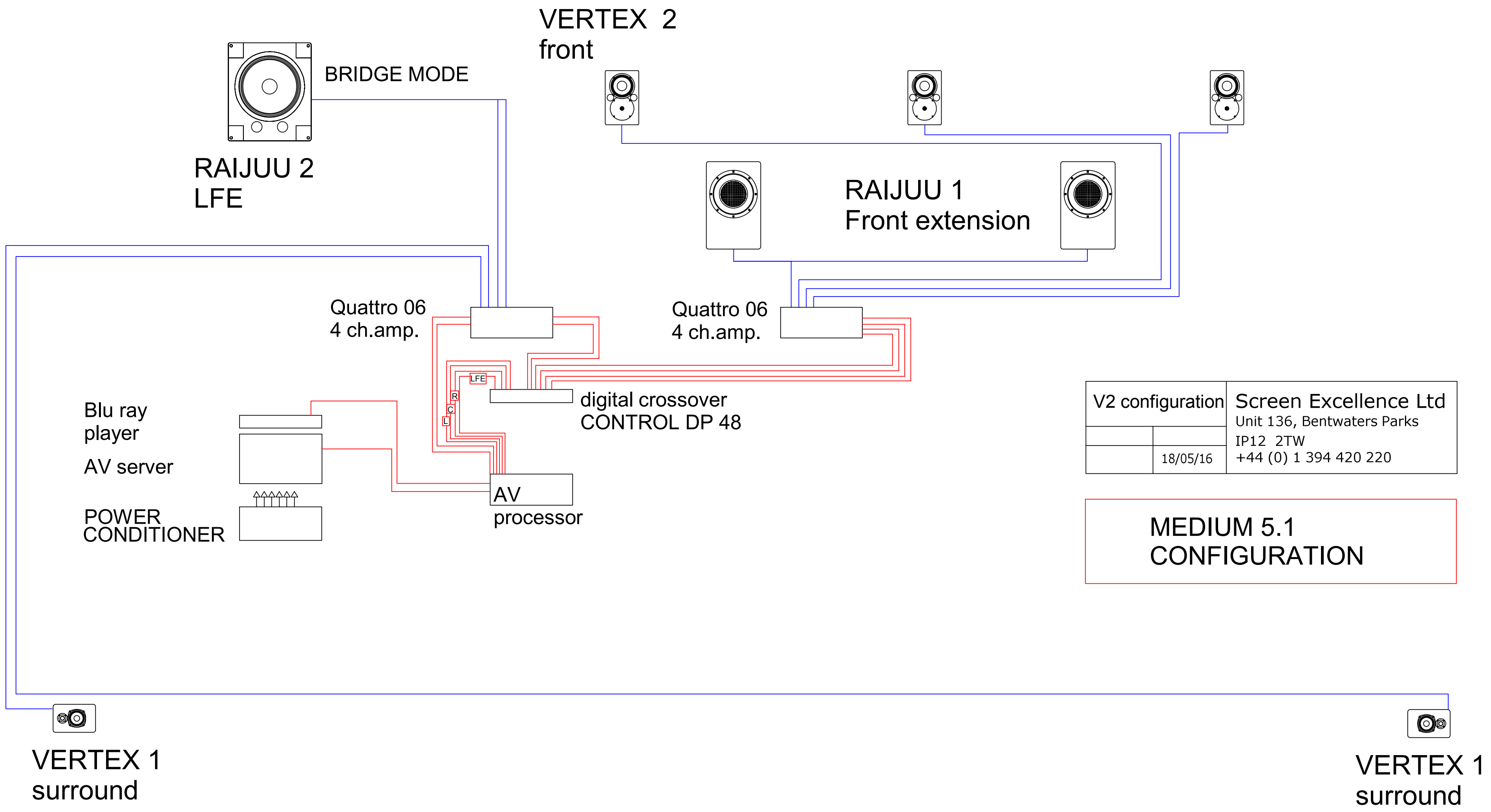

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1.2 LFE & subwoofer

- The subwoofer(s) must receive only the LFE signal. Check that all other channels are set as « large », even if the speakers are actually small
- When possible, use 2 subwoofers, and place them in the two corners of the room that are on the same side as the screen
- Alternatively, use an array of smaller subwoofers, with a step < 3.4m. This should be as near as possible to the screen and parallel to it
- There is no need to go through a crossover, but if you do so, use a frequency >120 Hz and a gentle slope, preferably a Bessel function.
- If you use a DSP crossover, adjust for the position using the DSP delay function instead of the distance adjustment provided by the AV processor

1.3 Room modes and acoustics

This subject matter is presented in chapter 3)

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2) Measurements and EQ.

Use MLSSA (see https://en.wikipedia.org/wiki/Maximum_length_sequence)

2.1 verify each sound source with MLSSA measurement

- Check each channel for semi-near field response and polarity. Discard reflections, although you can use the measurement to identify reflecting objects

2.2 Verify the coupling between the centre limited range speaker and L,C,R subwoofer

- This is done by placing the microphone half-way between these two sources, adding 50 cm rearwards offset. There should be no dip around the cutoff frequency

2.3 Parametric EQ of each source

- Use preferably a DSP controller. Dirac EQ is to be investigated

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3) Room Acoustics

3.1 Room modes should be reduced by absorption:

- Using bass traps in the rear corners. These can be passive or active
- Using hollow cavities, like seats risers, false ceilings, etc.
- Leaky walls (depending on sound insulation requirements)

3.2 Early reflections < 20 ms delay should be avoided:

- Using room geometry
- Using absorption (especially in the first half of the room)
- Using oriented reflectors (RFZ concept)

3.3 Reverberation >20 ms delay should be scattered:

- Using diffractors, like Schroeder diffusors, at the rear of the room

3.4 Leather armchairs should be avoided:

- Early reflections from leather armchairs cannot be controlled. Although we made ourselves a compromise on this in our lab-room (!)

4) Headroom

4.1 formulae

- Power level: W=10 Log W/W0
- Sound Pressure Level: SPL= 20 Log P/P0

4.2 Implications:

- When you double the source, you increase the level by +6 dB
- When you double the power, you increase the level by +3 dB
- When you double the distance, you decrease the level by -6 dB
- The intensity is proportional to the power, not to the SPL

4.3 Figures:

- If the sensitivity is 90 dB /1W/1m, you'll need 1 kW to reach a SPL of 120 dB @ 1m, and 108 dB @ 4m. With 200 W, you'll only reach 103 dB
- If the sensitivity is 95 dB/1W/1m, you'll only need 320 W for the same SPL

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4.4 Thermal compression

- When the voice coil temperature rises, the real and the virtual resistance increases, inducing a loss of loudspeaker sensitivity
- At maximum power, this can be a loss comprised between 4 and 6 dB
- When computing the SPL figures, you should take this loss in account
- This can have odd effects on the loudspeaker response when passive parallel crossovers are involved
- It is always preferable to use loudspeakers that can handle more power than what is needed: This will minimise thermal compression

4.5 Two easy rules of thumb

- Use the continuous max. SPL available from a loudspeaker as its peak max.SPL as this one will be reduced by thermal compression
- The worst case for sound quality and loudspeaker safety is a clipping power amplifier. Always use over-rated power amplifiers (preferably as much as twice the rated power acceptance of the loudspeaker).