ECHO Project

P3H Anamorphic & P3H Pyramid Array

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Pin3Hot

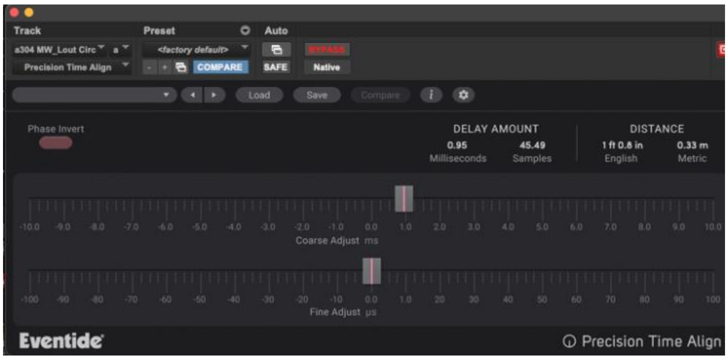
P3H Anamorphic Array (Traditional Orchestra Seating Layout)

I consider this array as four parts and the relative balance of those four parts varies depending on several factors from the room and the ensemble, to the music and the clients wishes. The four parts are: focus (MK2H X 5), dimension & depth (CO-100K X 4), vertical expansion (CUX-100K X 3), & glue (ES pair). [see array documentation]

In the provided ADM I have balanced this array as follows:

- The Decca Tree and Outriggers are at unity (often I will move the outriggers beyond LR, but in this case to stick with our agreed 7.1.4 configuration they are kept at the LR speaker position).
- The CO-100K are at -3dB relative to unity
- The CUX-100K are at -4dB relative to unity
- The ES is at +2.3dB with the E being +4.5dB relative to unity and the S at unity before decoding

- Due to the nature of our session we were unable to finesse microphone placement before recording. I found when listening that I needed to delay the CO-100Ks by 0.95ms to get my expected result.



- There was a slight Lo-Mid frequency build-up that I addressed with the following equalisation for the four CO-100K.

- I applied a 60Hz 18dB/octave high pass filter to the ES pair to minimize rumble

ES Pair Decoding:

- nb: the figure-8 is the "M" in the mid-side decoding, and the omni is the S. The resulting LR decode approximates two cardioid capsules back to back – the right one having the polarity reversed. This produces a relatively even balance of correlated and de-correlated signal.
- The figure-8 is routed to the left side of a stereo track and the omni to the right side.
- I trimmed the figure-8 +4.5 dB before decoding (this is subjective based on listening to the decode as it varies depending on the sensitivity of the mics used)

- For simplicity (and ease of applying equalization), I use a simple MS decoding/encoding plugin.

End P3H Anamorphic Settings

P3H Pyramid (Circular Orchestra Seating Layout)

This was an experiment and is not intended to be standalone. I would consider using this approach in a circular setting, alongside something to replace the purpose of the MK2H in my anamorphic array.

In the provided ADM I have balanced this as follows:

- CO-100K at unity
- CUX-100K at -1.5
- The ES is at +2.3dB with the E being +4.5dB before decoding
- There is no DSP or timing adjustment except a 60Hz 18dB/octave high pass filter on the ES pair.
- See the anamorphic array description for ES decoding

End P3H Pyramid Settings

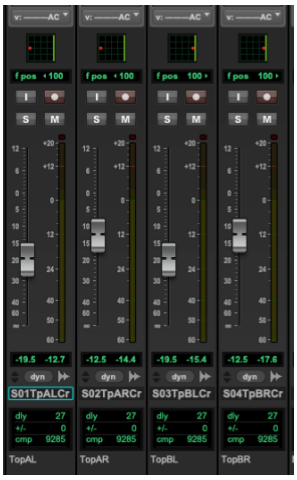
PRESENTATION MIX (Traditional Orchestra Seating Layout)

I have only provided a Presentation mix of the Traditional Layout. (Atmos BIN settings – mid for all except height which are far)

Anamorphic Array balance and processing is the same as detailed in explanation above.

Additional mics and processing.

- I found I quite liked the diffuse quality of the mics we had above the canopy, and that they worked well with my ES pair in adding some additional scale to the space. I used these mics as a LR height pair (as shown) with a 60Hz high pass filter. This was a spaced set of coincident omni and figure-8 capsules allowing the creation of a spaced stereo pair with flexible polar pattern (pointed straight up).

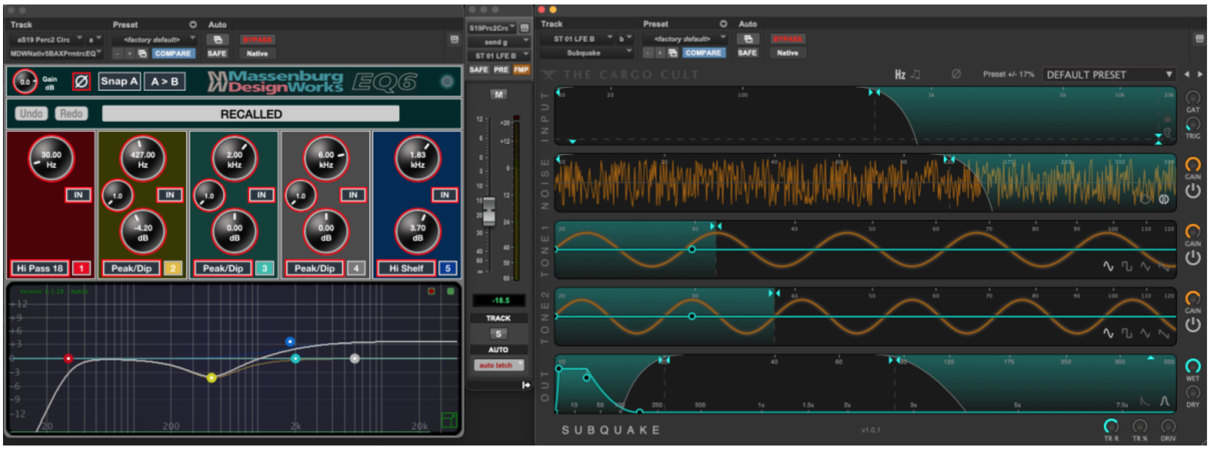


- I also found the sound of the four "floor" microphones (which by their placement approximate a low pass filtered pick-up) added a little more weight to the low end and was a good source to feed a subharmonic synthesizer with (which is how I almost exclusively generate LFE signals – this helps significantly in cinema playback to add scale and power when needed). These were equalized to avoid lo-mid build-up and loss of clarity. An aux send was used to send this signal to a subharmonic synthesizer which feeds the LFE channel. Both the 4.0 as shown below and the synthesized .1 are used in the mix.

- The contrabass spot mic was used to give additional weight to our single player, and also sent to the same subharmonic synthesizer as above. Some equalization was applied primarily to minimize leakage of spectrum that would outweigh the benefits of adding the mic in. The signal was also fed through an analogue circuit (McDSP APB) using a plugin (APB BOB) that allows controlled and targeted saturation of low frequencies – this helps to keep the bass strong regardless of playback equipment.

- Additional reverb was added for a slightly smoother and brighter tail. The reverb I find complements the hall well is unfortunately stereo. I run four instances of it with slightly varied settings. This is fed from the 5 MK2H. The returns of the four instances are shown below.

- The percussion spot mic was used to get a little more clarity on the percussion, and also a little LFE enhancement for impact. Rather than a subharmonic synthesizer, in this case I use a different tool for transient power rather than sustained weight.



Buss Processing:

Each layer (in my case just orchestra and re-amped piano) feeds a separate multichannel buss with its' own effects returns. These busses are then combined to create the final mix. This allows easy printing of stems, but is also very efficient in mixing as it is simple to make adjustments to individual layers/passes without affecting the whole, as well as separating the mixing of microphone feeds from the mixing of elements (ie: orchestra against re-amped layer).

Buss 1 – Orchestra.

- GML MDWDRC3 - while this is a dynamic range controller, I am really using it in this case, to provide a slightly enhanced sense of space. It is subtle but can be manipulated such that when the source of an impact causes a slight dynamic compression, the space is untouched which gives a sensation that the space itself actually expanded. Effectively providing dynamic control that expands the soundfield when compressing, rather than collapsing it.

- McDSP KD-1 (Analogue processing - APB)

This allows me to provide varied analogue saturation in a frequency range of my choosing. This is entirely about character, and to my ears provides a real tactile quality to the sound that can make me forget I am listening to a reproduction.

Buss 2 – Reamped PreRecorded Piano

- Same MDWDRC 3 processing as Buss 1
- McDSP Tape (Analogue processing – APB)
  This is tape emulation – however done with analogue circuits. In this instance I am using it to add a little more body and character to the sound of the piano.

Master Buss Processing:

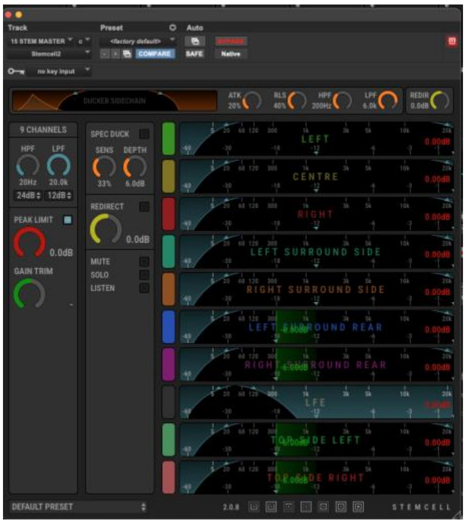
NB: Typically for film scores I will print two full mixes. One is just a summ of the various busses with no further processing. This is what gets delivered for the film and by definition is exactly the same as playing all the printed stems together. The second full mix is what I consider the soundtrack album version and will have additional processing targeted at consumer playback.

In this instance this is where our two busses (orchestra – Buss 1, and piano – Buss 2) are combined.

- McDSP Royal Mu Surround (Analogue Multichannel Limiter – APB)
  This is never actually doing any compression or limiting, but by driving it to the edge of compression it does impart a warmth to the sound that I find pleasing.

- StemCell2 (buss filtering, level adjustment etc…)

This is just to compensate for interchannel level differences introduced by how I use the Royal Mu Surround.



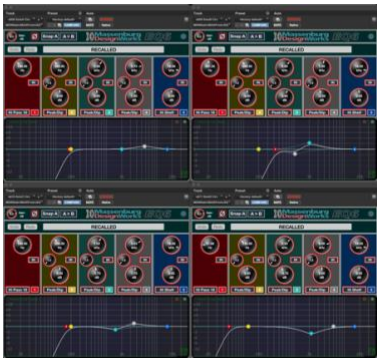
All of the above settles in the core sound I was aiming for.

At this point I get much more selective about what I'm doing (and whether it's worthwhile). I am happy with the overall sound, but there are still elements and moments that I want to improve if possible. All the below have varying degrees of automation in the different movements to keep them out of the way when not needed and appropriately balanced otherwise.

- Solo String mics. The balance in the room for the traditional seating made it hard to hear the solo lines clearly all the time. As they were not loud in the room their mics also had a lot of signal besides the solo strings themselves. Looking for both more volume and clarity in spots, I took a few approaches.

  1) I edited the mics so that they were only playing when there was content I actually wanted to hear in the mix.

2) I equalized them to work better in context both in terms of improving the solo clarity and balance, but also avoiding timbral/balance issues with other instruments.



3) In the case of the cello in places where it was pizzicato, and I wanted a crisper attack I also applied an envelope modifier.

4) I added the woodwind mics in (with level automation) for passages where (while the balance was good) I wanted more presence and clarity. These were heavily filtered.

5) Similar with the French horns.

6) In the last movement where the solo strings were almost inaudible in the array, and buried in their own spot mics I also added an additional "enhancement" buss for them which while I am not thrilled with the tone, I can now at least hear them. This buss was fed with the signals from the four solo spot mics. It was brutally filtered, the transients pushed, and then run through a modelled transformer to breathe a little life back into it.

- **Additional Reverb for these automated spot mics**

  Inevitably when these are processed as I have done and I am trying to make noticeable balance changes I need to make it sound like the player is playing louder or brighter in the room. In this case I add specific reverb for this. Sometimes I do this with algorithmic reverbs, sometimes with impulse responses taken in the space – however we now have the superb AIR Reverb which works brilliantly when working on projects recorded in the hall.

- **Re-Amped Piano**

  I was not happy with only my array mics (or any of the mics in fact) for this as I wanted something more focused that also would allow more flexibility in terms of spatial processing. In addition to my array mics (which are used sparingly as they provide some good dimension to the space), I also used the pre-recorded tracks Volker supplied that we played into the room.

  Similar to the spot mics in the orchestra layer I also used the Spitfire Air Studios Reverb. I would estimate about 80% of the sound of this material in my mix is the playback files with reverb, and 20% is the actual recorded sound with my array in the hall.

  Volker provided multiple stereo pick-ups for each layer of piano, as well as a stereo fx output for each. For the most part I chose a single pair for each piano with its' associated stereo fx.

1) Train Journey 1 (similar settings were used in 2)
Piano 1 (M49V) – positioned Front LR
Hi Pass, and High Shelf for clarity
Analogue Compressor (McDSP ChickenHead) for some subtle lo-mid saturation

Piano 1 FX – positioned Rear LR
Slight level trim (-1.2dB on right side)
40Hz High Pass Filter
Image Spreading (multiband mid-side adjustment)

Multiband Auto Panner

Piano 2 (M49V & C422) – positioned LssRss (both)
60Hz Hi Pass Filter & 8kHz Shelf Boost

Piano 2 FX
40Hz Hi Pass Filter
Nugen Halo Upmix (stereo to 7.1.2)

Piano 3 (M49V) (Positioned Rear RL)
60Hz Hi Pass Filter & 8kHz Shelf Boost
McDSP ChickenHead as Piano 1

Piano 3 FX (Positioned Front RL)

Piano 4 (C422) (Positioned Front LR)
Trim for center positioning - +1.5dB L & -1.5dB R
60Hz Hi Pass Filter
McDSP APB BOB (for targeted low freq saturation)

Sound Particles energy panner
This channel is a monophonic bass line. I wanted to give each note more clarity while still maintaining the space inherent in the provided recording. The settings applied push the panning mono on the transient but then fan out LR during the decay.