Symco Biamp DSP Conferencing System Field Tuning Notes (Baseline recipes to "season to taste")
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- **Labeling & Presets:**
  - Labeling I/O, meters & matrix mixers is a minimum for DSP file serviceability
  - Presets should contain as few DSP blocks/attributes as possible. Fine grained presets, are recommended for matrix mixer & other blocks. Efficient preset management makes DSP files more serviceable.
- **Mic/AEC settings:**
  - Use the analog gain (6 dB steps) as much as possible for best signal to noise ratio.
  - Turn the gain down if the clip light comes on under loud/close speaking conditions
  - Set mic gain in the following sequence:
    - Peak meters: Loud speaking should get into the yellow, but not red.
    - AGC (automatic gain compensation) is often used for "close mics" or table mics to help adjust for soft/loud talkers. I generally recommend putting it in bypass for ceiling mics. Adjust the input gain or consider a -3dB or -6 dB AGC target to achieve the following
      - Closer/louder than normal should give gain reduction
      Further/softer than normal should give gain boost
    - Add gain to the output of the Auto Mixer to get back to unity gain or a 0 dB to +3 dB send
  - Taking the "edge" off of the high frequencies: Consider a High Shelf filter in the 4k Hz range -6dB setting or a parametric EQ in the 6kHz, 4 bandwidth, -6dB range. “Season to taste”
  - AEC references:
    - The primary signal to be sent to the AEC reference(s) is any far end conferencing input (ATC, VTC, USB conferencing input).
    - Program audio sounds best to the far end when routed directly & referenced out of the mics.
    - Never send a mic to its own reference. Use signal path identifier to check AEC & other routing.
    - Typically a single mic reference is sufficient, but for rooms with reinforced mics, separate AEC references for non-reinforced vs. reinforced mics gives a cleaner send of the reinforced mics to the far end (see Symco sample files for reference)
- **Processing:**
  - High Pass Filters for each mic are core to good conference system audio. A typical starting point is 125 Hz with a 36 dB/oct slope.
  - Compressors are recommended for "close mics" (gooseneck, wireless). Compressors should have minimal activation at normal distance/level & activate with close/loud bursts (for example a cough). If there is no compressor activation under “loud” conditions and a 0 dB threshold, increase the mic gain.
    A compression ratio of 3:1 controls the transient bursts while maintaining a natural level.
    A release time of 250 ms helps keep the sound from “pumping”
  - Set your amp gain in a consistent way. One technique that I recommend:
    - Turn the amp all of the way down
    - Send pink noise at 0dB to the speakers (one zone at a time)
    - Turn the amp channel up to read 75 dB SPL A weighted at the listening location (smart phone apps are typically accurate enough) YMMV (your mileage may vary, meaning if you find that 80 dB on your phone gives the needed level, make that your standard)
  - Volume controls give room audio stability when located between the source & matrix mixer. Dry tap offs or pre-volume feeds give flexibility for recording & conference crosspoints (-VTC to ATC) (-ATC to VTC)
  - Voice reinforcement: A typical starting point is -12 dB of the reinforced mic feed to speaker outputs.
- **Test calls:**
  - Call someone within your office using a land line as the far end. VoIP or VTC calls are best for tuning conference mic EQs for maximum intelligibility & most natural sound.
  - Adjust the telco send path as needed to meter at +3 dB when speaking in a normal to loud voice.
  - Calls thru a telephone bridging service typically sound softer than direct dialed calls. Setting the transmit level too high can cause audio clipping. Be sure that there is no clipping effect (particularly on inter-building calls which often have the highest levels, least signal loss)