Audio in Houses of Worship

- Common Signal Processing Blocks
- Specific Examples

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Introduction

Audio is an essential element in any modern-day religious service. What is heard by the congregation is a combination of the acoustic qualities of the room and the performance of the audio system. Some of the desirable acoustic qualities in a house of worship are:

Reverberance – when well controlled with early decay, the effect is perceived as a beautiful sound that enhances the quality of the audio. See the Rane Pro Audio Reference for a definition of "reverberation."

Clarity – is the ratio of the energy in the early sound compared to that in the reverberant sound. Early sound is what is heard in the first 50 - 80 milliseconds after the arrival of the direct sound. It is a measure of the degree to which the individual sounds stand apart from one another.

Articulation – is determined from the direct-to-total arriving sound energy ratio. When this ratio is small, the character of consonants is obscured resulting in a loss of understanding the spoken word.

Listener envelopment – results from the energy of the room coming from the sides of the listener. The effect is to draw the listener into the sound.

Where a conference room would be optimized for articulation and clarity, a symphony hall is optimized for reverberance and listener envelopment. A good house of worship is optimized as a compromise between the somewhat conflicting requirements of music performance and the spoken word. Articulation must be excellent but sufficient reverb is required to complement music performances. All reflections must be well controlled to achieve this balance and ensure the best possible listener experience.

An Example of Good Sound

There are other possible examples but the author really likes this one. In some mosques, cathedrals and tabernacles there are wonderful low-domed ceilings that have marvelous natural acoustic properties. The acoustic coupling from performers to the congregation grouped under the dome makes for a very (dare I say) "spiritual" experience. For the purpose of this article, this level of performance is a "gold standard" to which other acoustic spaces will be compared in the search for improvements and recommendations.

The U.S.A. Pavilion at Florida's Epcot® Center makes for an interesting case study. There is a dome ceiling in the pavilion. Under the dome an eight-part acappella group called the "Voices of Liberty" performs. For those under the dome listening to the group, the sound is beautiful and inspiring. Moving out from under the dome, the "magic" is gone.

This level of performance is not feasible in a typical house of worship but it does establish an icon as to what could be if there was sufficient skill (and budget) applied to the acoustic and audio system design.

And Now The Ugly World in Which We Live

Contrast this to a typical public address system squawking bad sound to the congregation. That which was good is replaced with misery. You reach for a bottle of aspirin to calm the headache induced by a pair of blaring powered speakers.

Some of the problems encountered by audio designers/consultants include:

Excessive Reverberation – such that articulation and clarity is poor.

Echo – where a discrete sound reflection returns to a listener more then 50 milliseconds from the direct sound and is significantly louder then the reverberation sound.

Flutter echo – repeated echoes that are experienced in rapid succession that occur between two hard parallel surfaces. All echoes ruin the acoustic properties of a room and a flutter echo is particularly damaging.

Coloration due to reflections – when a reflection destructively recombines with the direct sound modifying the frequency response in the process. These are non-minimum-phase colorations as correction with equalization is not possible.

Delayed Sound – from coupled volumes (contamination from adjacent rooms storing sound energy and then returning the energy to the main room).

Psychological preconditioning – It is a common problem for the clergy and congregation to be so preconditioned by bad sound that they become resistant to change and find it difficult to (at first) recognize good sound. This can also work in the audio consultants favor when the customers are preconditioned by good sound and are willing to invest the required resources toward good audio design.

For those of us designing audio for houses of worship with a rectangular room, flat walls and probably a vaulted ceiling, some form of sound reinforcement is required. Through attention to detail and careful design of the audio system, the experience of the congregation can be non-aspirin inducing and the system simple to use.

Common Signal Processing Blocks

Let's begin by looking at the universal signal processing chain common to all audio systems. In the simplest systems these functions are accomplished in an audio mixer that feeds a pair of powered speakers. More sophisticated systems include equalization, compression, limiting, automation, feedback suppression, electronic crossovers and other tools of the trade. These days it is possible to include all of these functions in a DSP (Digital Signal Processor). One example of the signal chain from the minister's microphone to the power amplifiers is shown in Figure 1.

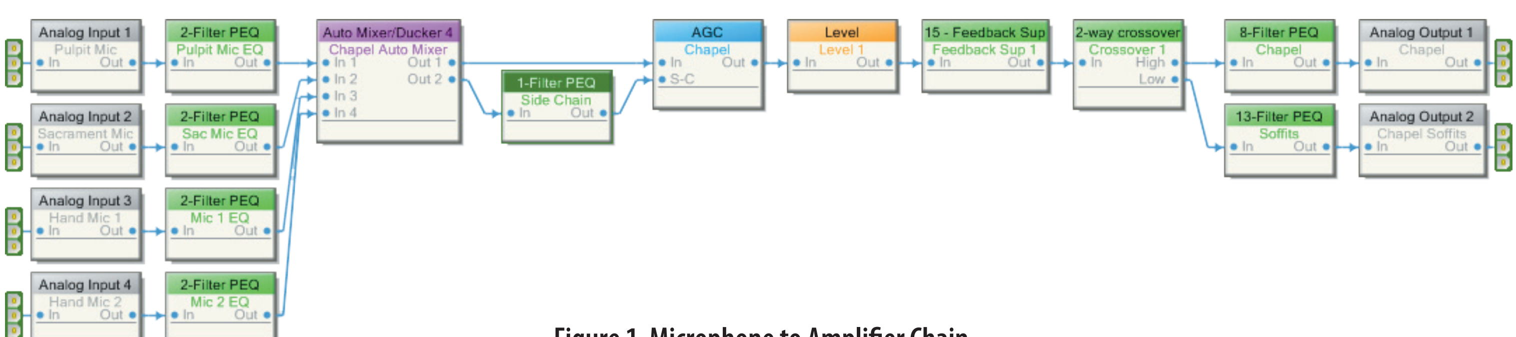

Figure 1. Microphone to Amplifier Chain

The signal processing flow starts at the Analog Input. A 2-band Parametric Equalizer filters out-of-band low frequencies. The microphone signals are summed together in an Automatic Mixer. An AGC (Automatic Gain Control) reduces the dynamic range and a High-Pass Filter in the side chain improves the performance of the AGC. The Level control can be tied to a pot on the wall or a smart remote. There is a Feedback Suppressor for good measure. A 2-way Crossover supports a biamplified system. The 10-band Parametric Equalizers are utilized for both wide- and narrow-band corrections. Generally, wide-band filters correct minimum-phase frequency response irregularities in the speaker drivers and in the room response. Narrow-band filters are useful to partially correct non-minimum-phase related problems such as energy stored in room modes (reverberant energy). A Limiter could also have been added to protect the system from clipping if that feature is not included in the power amplifier.

Now let's take a look at some of these signal processing blocks in greater detail.

Analog Input / Microphone Preamp

It is surprising how often even experienced audio consultants will configure an audio input incorrectly. It is important that as much gain as possible is accomplished at the front end of the system in the Analog Gain stage. Any additional gain from Digital Trim after the input stage degrades optimum signal-to-noise performance.

As an example, let's set the input gain to a value of +40 dB. One way is where the analog gain is set to a value of +45 dB and the digital trim is set to -5 dB (as in Figure 2), the measured input referred noise is -127 dBu. A common (but incorrect) way would have the analog gain set to a value of +30 dB and the digital trim set to +10 dB (the author has seen this repeatedly), to give the same Mic gain of 40 dB — but now the input-referred noise is degraded to -114 dBu. That is an increase of 13 dB for the noise floor, or a change (in the bad direction) of 8 dB in the maximum SNR (Signal to Noise Ratio). Your exercise is to determine why the SNR was only degraded by 8 dB rather then the intuitively obvious value of 13 dB.

Answer: *The noise floor does drop by 13 dB, but this combination of settings causes the analog input stage to clip at an input level that is 5 dB lower. Hence, the change in system SNR is 8 dB.*

Applying attenuation after the input stage (rather then gain) reduces overload performance and so should be used with skill and discretion. It is the proper technique to maximize noise performance.

For more detailed technical information please see the RaneNote "Selecting Mic Preamps."

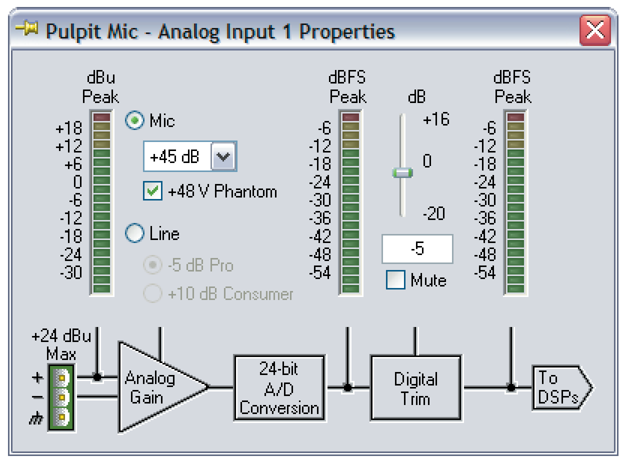

Figure 2. Drag Net Input Block

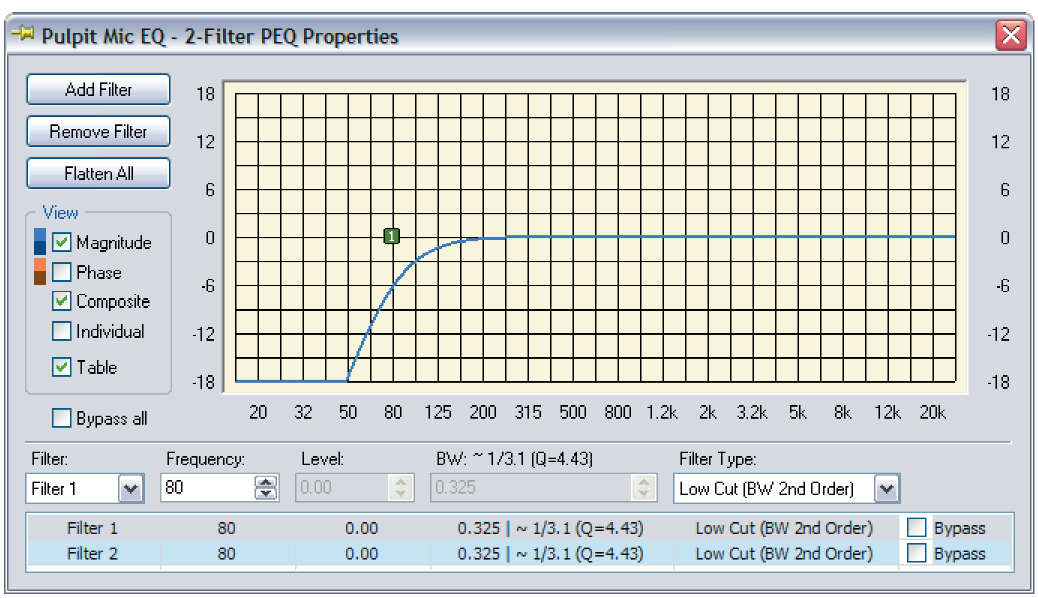

Figure 3. Drag Net Parametric EQ for Input Low Cut

Input Low-Cut Filter

A very good idea is to add a low-cut filter set to ~80 Hz after the input stage to minimize the effects of undesirable low-frequency noises such as bumps and thumps that come from handling the mic and also wind blasts and pops from speaking too closely into the microphone. In Figure 3, both 2nd-order filters are set to the same frequency to produce a 4th-order filter.

There should also be a low-cut filter in line with the SC (Side Chain) input of the AGC (Automatic Gain Control). This filter can be set to a higher corner frequency (such as 120 Hz in Figure 4) to improve the performance of the AGC by rejecting the effects of low frequency noises.

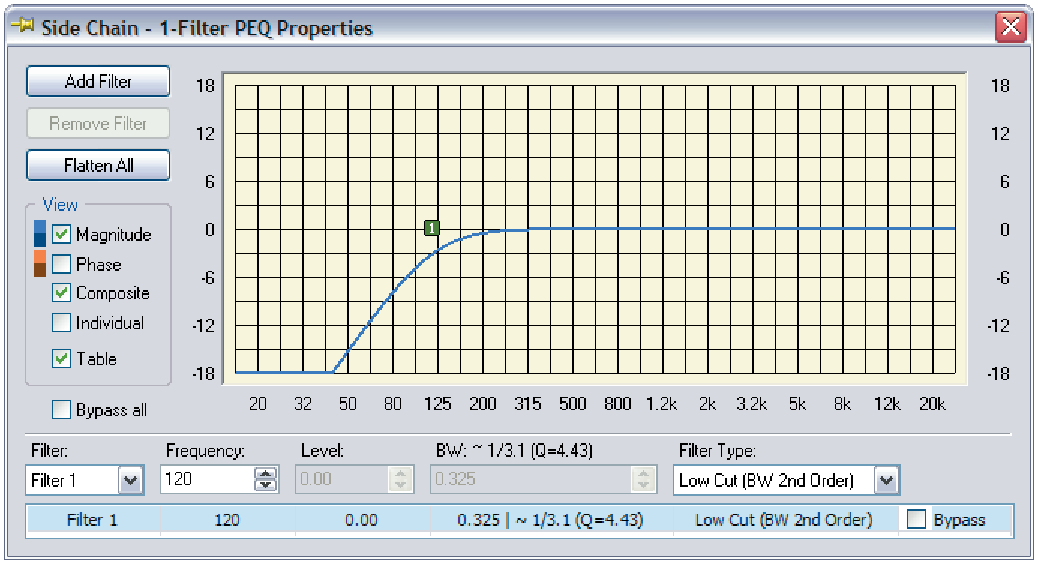

Figure 4. Drag Net Parametric for AGC Side Chain

The Auto Mixer — A Little Automation Buddy

An Auto Mixer (shown in Figure 5) is a good idea when there is more then a single open microphone. Auto Mixers combine the signals from multiple microphones and automatically correct for the changing gain requirements as the NOM (Number of Open Microphones) changes.

Threshold with Last On is a useful setting for all microphones used in a worship service (Figure 6). Unused microphones (input levels are below threshold) are gated. When the input of a microphone is above threshold then other inputs with a lower assigned priority level are ducked.

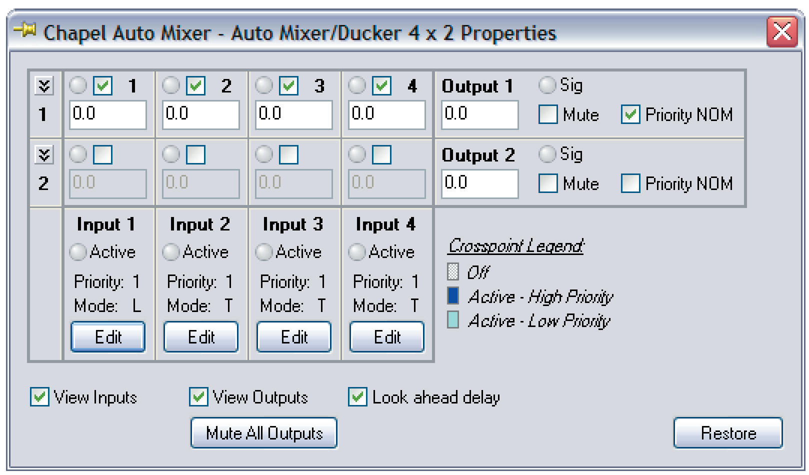

Figure 5. Drag Net Auto Mixer Block

Automatic Gain Control

A Compressor is the correct processing block in this link of the audio chain. Something is needed here to prevent exuberant preaching from melting down the congregation. Surprisingly, an AGC can be very useful in this position but configured to behave more like a specialized compressor by using the settings shown in Figure 7.

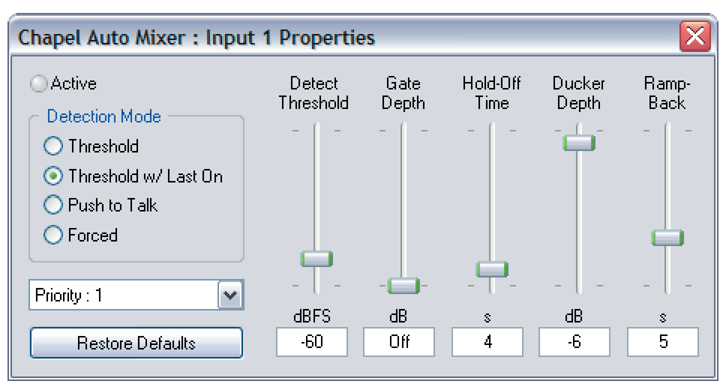

Figure 6. Auto Mixer Input Edit Block

The value of "Threshold re: Target" is set to have an offset of 0 dBr so that "Threshold" has the same value as the "Target." "Maximum Gain" becomes 0 dB and the gain curve starts to look like a compressor but there are additional controls in an AGC for Hold and Release that are useful when the input level is below threshold. These settings avoid the problems of compressor "pumping" when that exuberant speaker is at the microphone as attenuation levels are held between spoken phrases. Then, when transitioning to a more reserved speaker, the hold time (below threshold) is short enough to expire so that the gain returns to a normal level.

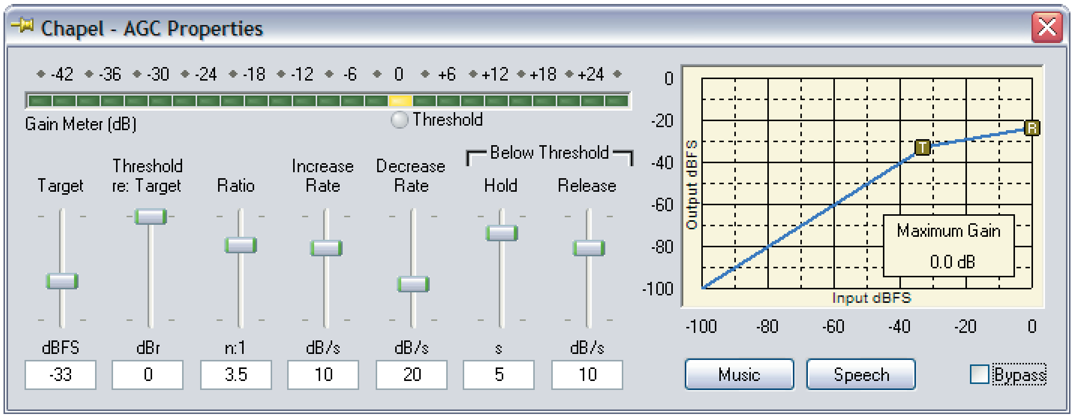

Figure 7. Drag Net AGC Block

An Exciting Labor-Saving Tip — Put a Control On the Wall

A level control can provide attenuation as needed under the control of a pot on the wall or a smart remote. This is handy in systems where a minister needs to run a system alone without the assistance of an audio specialist who is running a mixing board. The remote can be located on or close to a pulpit which places control of the audio system at the fingertips of the minister. The DSP control is shown in Figure 8.

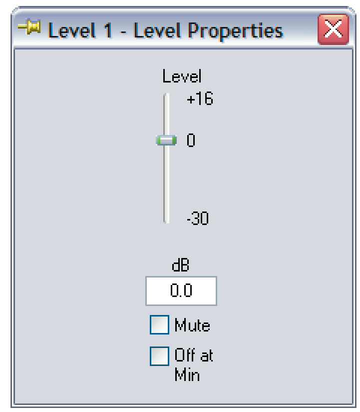

Figure 8a. Drag Net Level Block Mapped to a Remote Level Control

Figure 8b. You can mount a 20 kΩ pot anywhere, or Rane makes a remote that fits in any standard U.S. electrical box and can be covered with a Decora™ plate cover.

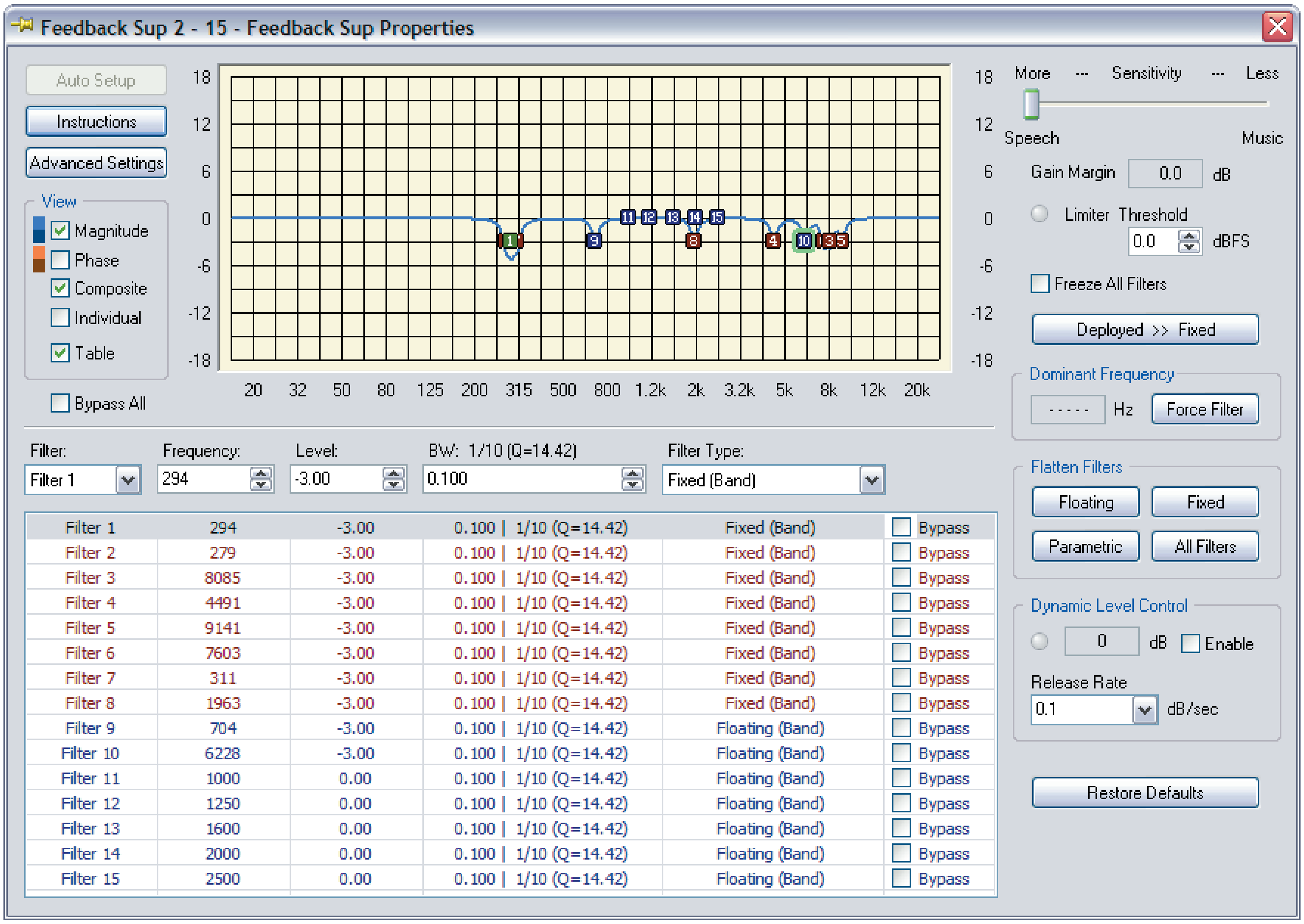

Figure 9. Drag Net Feedback Suppressor

Feedback Suppression — A Gift From Above?

The next item in this processing chain is somewhat controversial. It is a Feedback Suppressor. To some audio consultants a Feedback Suppressor is heresy! The argument is that a properly calibrated system has no need of such a Band-Aid®. This is generally true, but there is one case when it is wise for an audio consultant to suffer the ignominy of using a Feedback Suppressor — a lay clergy where the person speaking is untrained and/or unfamiliar with proper use of a microphone. The author has witnessed such a person cup their hands (in the attitude of prayer) directly around the microphone capsule. The hands form a resonant chamber that results in squealing feedback. A good Feedback Suppressor would have locked on to the offending tone and notched it out posthaste.

Feedback Sup Setup Instructions

Using Auto Setup to ring out a system

1. Setup the system's gain structure.
2. Umute the mic(s).
3. Talk into the mic(s) and adjust the system gain until it is on the verge of feedback.
4. Click **Auto Setup** to automatically deploy *Fixed* filters as feedback occurs.

Auto Setup deploys unused (flat) *Fixed* filters. Once Auto Setup is complete, *Floating* filters are deployed should feedback occur.

Parametric Equalization: Now We're Having Real Fun

Parametric equalizers are used for both wide and narrow band corrections. Generally, wide-band and shelf filters can correct for minimum-phase frequency response irregularities.

One interesting detail of Figure 10 is Hi-Shelf Filter 1. This filter was added after achieving flat in-room response. Since the system was calibrated in an empty room, this extra high-frequency energy is intended to compensate for the high-frequency absorption of the congregation when the room is full of people. There is also a noise-masking effect in some congregations that will tend to obscure the intelligibility of the spoken word. In practice this approach of adding a bit of extra high-frequency energy into the room works well.

Narrow-band filters (Figure 11) are useful to partially correct non-minimum-phase related problems such as energy stored in room modes. At low frequencies this energy causes bass to sound indistinct, and in midrange to lower treble this energy is perceived as reverberation. These filters attenuate the frequencies bouncing about the room. In an acoustically live room, room resonances can propagate for a surprisingly long time causing these frequencies to "build up." Narrow-band filters are just a partial solution. Greatest effectiveness is achieved when filters are used in conjunction with acoustic room treatments such as diffusers, high/mid frequency absorbers and bass traps. This topic is beyond the scope of this RaneNote but an important part of the audio consultant's craft.

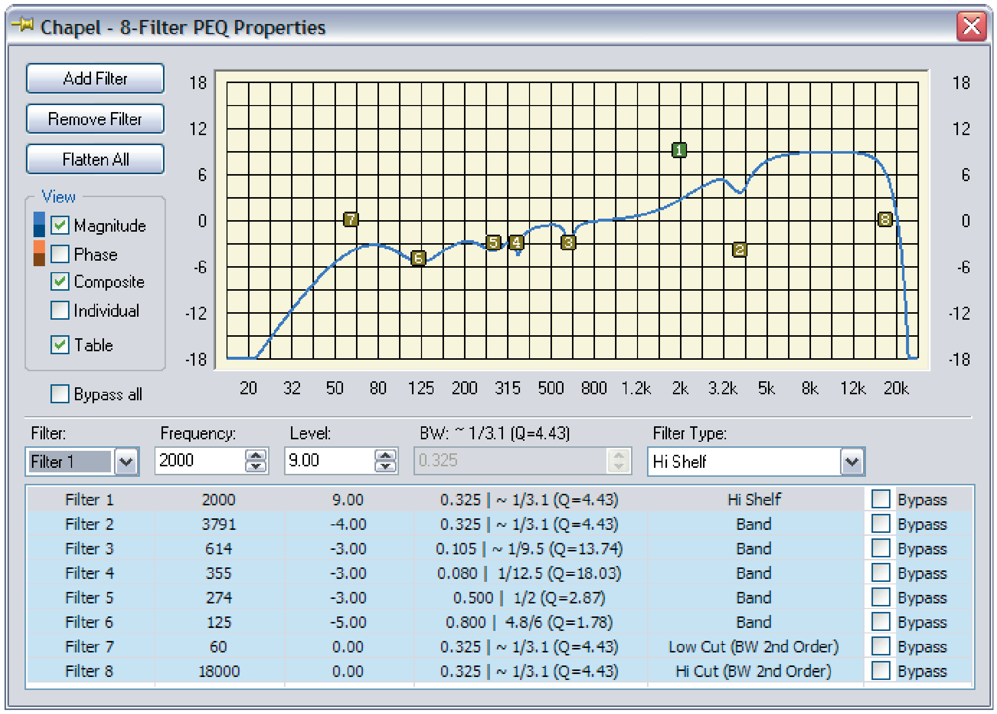

Figure 10. Drag Net Parametric Block (May Have up to 15 Bands per Block)

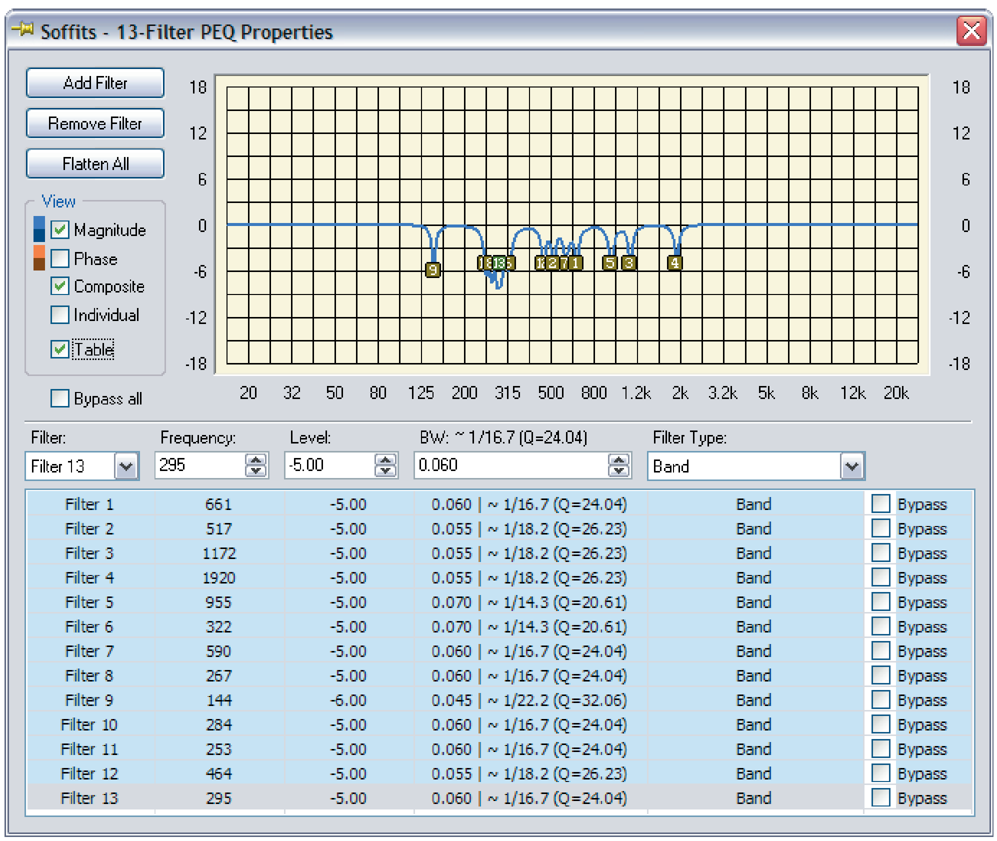

Figure 11. Parametric with Narrow-Band Filters

Specific Examples
Example #1: A Small Church

Description

The ceiling is low suspended acoustic tile over an open space covered with thin carpet. The RT60 (the time it takes the reverberant sound to decrease by 60 dB) is short, so controlling reverberation is not a problem. In fact, the room is a touch "dry" for music, and content of the worship service includes live music performances. Audio sources are the minister's wireless microphone, the band, a DVD/CD player and other devices as needed. Control is via a 24-channel mixer with all inputs used. Output is to a pair of powered speakers mounted high in the room corners in a stereo configuration. This installation was done by members of the congregation without professional audio consultation.

Problems

- The quality of the audio is poor with numerous problems including uneven frequency response.
- An experienced sound person is required to run the mixer for all audio system use.
- There is poor congregation coverage from the stereo speaker pair. People sitting in the hot spots just in front of the speakers are blasted with excessive level, and the rest of the congregation is exposed to a strong interference pattern between the two speakers. The system is uncompensated for room modes, room response and speaker response irregularities. There is a small "sweet spot" in the center of the room where the two speakers combine coherently but there is an isle down the center. Since there are no chairs, no one is seated in the "sweet spot".

So does this audio system work the way it is? Yes, but even the pastor knows the congregation may not be receiving the best possible audio experience.

Recommendations

Improvements to this system are accomplished in a number of ways. A DSP can be used for equalization, other processing and to add automation to the minister's microphone. The entire worship band could be run through a mixer with each individual input processed by an AGC. There are admittedly downsides to automating the audio mixing of a large group, as the automation is not as intelligent as an experienced sound person, but is possible in some cases.

The speaker system is examined for options providing more even coverage of the congregation. Improvements can be introduced in phases.

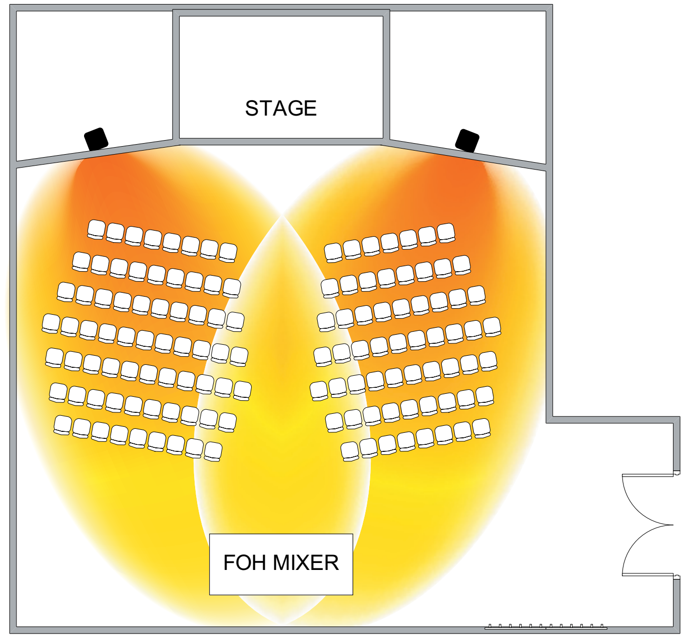

Figure 12. Stereo Speaker Pair Coverage

Phase 1

Add a DSP box between the mixer output and main speakers and on-stage monitors. Features added could be:

- **Parametric Wide-Band Equalization.** This alone would greatly improve this system.
- **Parametric Narrow-Band Equalization.** A short RT60 makes this unnecessary at this time. However, remodeling could increase RT60 to where narrow-band equalization would be needed. (This room could use bass absorbers).
- **High-Pass Filtering**. If not already in the mixer.
- **Compression**. Always a good idea with microphones because of the inverse square law relationship between the preacher's mouth and the location of the microphone. See the *Rane Pro Audio Reference* entry for "Inverse Square Law."
- **Feedback Suppression**. If needed.

Phase 2

Automation is incorporated with automixers and remote controls. There are many exciting ways to add these features depending on the congregation needs. The most obvious upgrade is to add the ability for a minister to turn on and control the main microphones from a simple control panel in easy reach at the stage.

Phase 3

The very uneven coverage of the congregation by the stereo speaker pair needs to be addressed as shown in Figure 12. The seats directly in front of the speakers have enough level to kill small animals.

If the audio system were perfect then each seat in

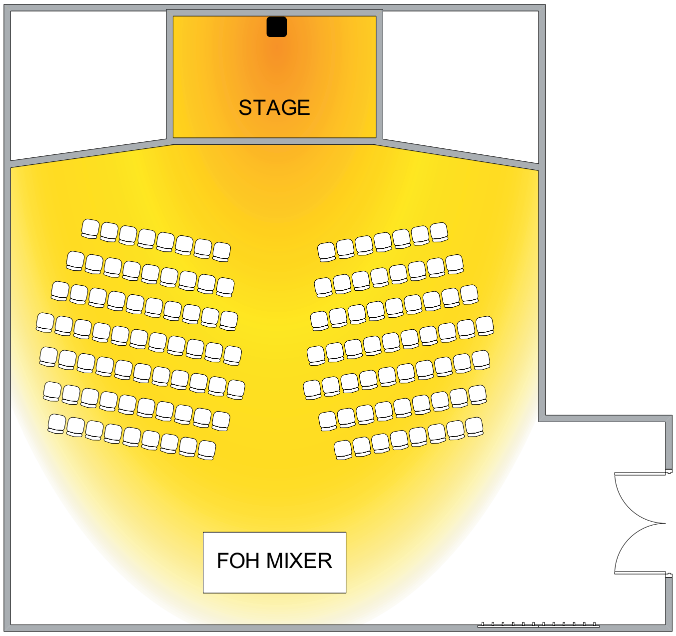

Figure 13. Line Array Speaker Coverage

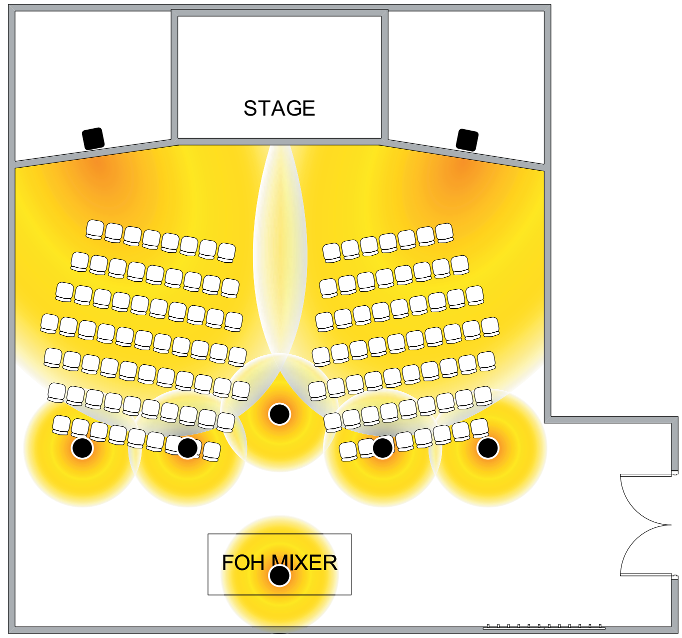

Figure 14. Distributed Array Speaker Coverage

the congregation would have the same audio level. In the author's experience, similar rooms have been controlled within a couple of dB. In this example, the seat closest to each loudspeaker is about 15 dB louder then the worst seat on the floor, and interference between the two speakers adds to a very lumpy and unpleasant frequency response. The FOH (Front Of House) Mixer is placed in a location for good sound, causing the levels at the ends of the front rows to be way too loud.

Line Array Speakers

One improvement is to remove the stereo pair of point-source loudspeakers and install a line array located in the center of the back wall as shown in Figure 13. Coverage of the congregation is more even, and the level at the FOH Mixer location is very similar to the coverage level over the whole floor of the congregation. The level of the stage monitors is greatly reduced and may no longer be needed by the musicians. Within the near field of the line array there is a range were the audio level will decrease by only 3 dB for each doubling of distance which greatly helps even the coverage across the entire floor. The audio is distributed across the whole line so that even if a microphone is right next to the array, there is little tendency to feedback.

In this example, there is a low suspended acoustic tile ceiling that shortens the length of a line array speaker. This limits the mounting options and the maximum length of a line array so this might not be the best solution. If the room were remodeled so there was a high ceiling, then a line array would fit. This is especially true if the newly remodeled ceiling was acoustically reflective causing the RT60 of the room to be much greater. The high directivity of a long line array greatly helps to project the audio out to the floor rather then have the audio directed toward the ceiling where it contributes to the reverberant energy and echoes in the room.

Supplemental Distributed Array Speakers

Because of the dropped ceiling, another option is a distributed array of supplemental ceiling speakers in the back of the room as shown in Figure 14. The loudness level of the main stereo pair could be reduced by at least 12 dB. This would greatly diminish the hot spots in the front, but would leave the level at the back way too low. Ceiling speakers can be added in the locations shown to fill in the audio in the back of the room.

It is important to include a speaker over the mixer location so the audio at that location matches the level in the congregation to acheive an accurate mix.

Why The Delay?

The ceiling loudspeaker signals should be time delayed so their output combines coherently with the point-source pair in the front of the room. If the rear loudspeakers are not correctly delayed then the loudspeakers in the room will not combine correctly.

This room is too small for audio from the front of the room to be perceived as a distinct echo. Applying delay to the ceiling speakers can minimize the problem of localization confusion occuring if the first arrival sound is coming from the overhead loudspeakers and not the front of the room.

Example #2: A Mid-Sized Contemporary House of Worship

Description

This second example is a medium sized house of worship. The vaulted ceiling is high and the floor in the congregational seating area is covered with hard vinyl. The RT60 is approximately 1.5 seconds so reverberation is a problem in an empty room. The sources of audio are ministers on a microphone and a worship band. Control is via a 32-channel mixer. The speaker system is an array of three large boxes mounted as a central cluster high in the peak of the ceiling. A professional audio company did the installation and calibration.

The quality of the audio in this church is much better than in the first example. An interesting question is: how good is "good enough"? When interviewed, members of this congregation can usually hear. Rarely is the audio painful to listen to, so some say that the audio quality is fully acceptable. Reflect back on the example in the introduction where domed ceilings were held up as an icon of natural acoustic wonderfulness. Let's see how this audio system installation stacks up.

Problems

- Reverberance is not controlled and is dependent on the configuration and occupancy of the room. Low-mid frequencies are a particular problem as the energy builds up and is never trapped.
- Clarity is fairly good, meeting a minimum standard.
- Articulation is acceptable but not outstanding. The ALCONs (Articulation Loss of Consonants) rating of this room is fairly low but in the acceptable range. However, there is room for improvement.
- Listener envelopment is nonexistent and pales in comparison to the example of a domed ceiling.
- As in the first example, an experienced sound person is required to run the mixer for any use of the audio system, as there is no system automation.
- There is good coverage of the congregation from the central cluster, but people sitting in the area where the coverage patterns between two of the speakers overlap experience uneven frequency response due to the comb filtering caused by the interference between these two speakers.
- Bass response is particularly poor. The poor bass response leads to the impression that the system lacks sufficient power.

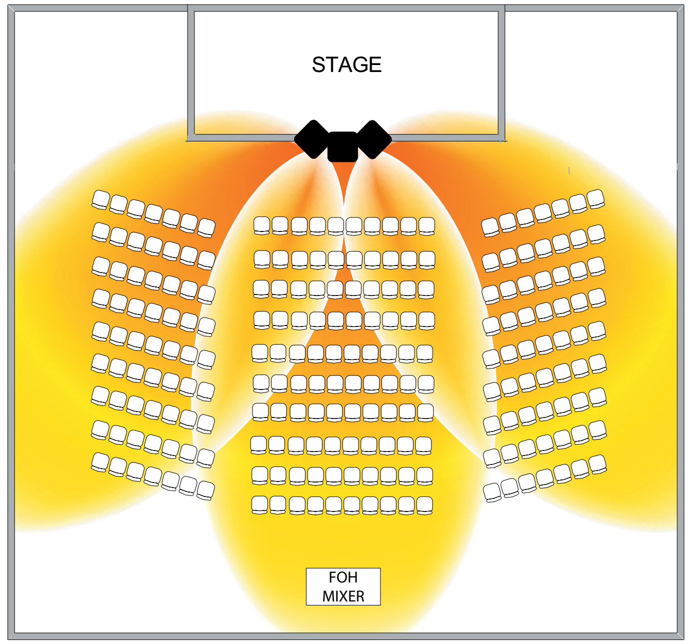

Figure 15. Distributed Array Speaker Coverage

Recommendations

A DSP unit is already in the system and can be used for additional equalization and other tasks. The same recommendation applies to add enough automation so that a simple service can be done without bringing in a sound person.

The speaker system may already be fully adequate. The first temptation may be to add a subwoofer, but it is probable that the buildup of mid-bass energy makes the bass quality so poor that adding more will only make matters worse. To fix the room, the ceiling and walls could be covered in bass absorptive panels, but this is not practical. A compromise is to add bass traps to the room corners and the ceiling ridge.

If it is not possible to tame the room with traps, narrow-band filtering techniques could solve things. The room is evaluated for the modes that build up room energy and these frequencies are notched out with a very narrow filter. A combination of some absorptive panels and narrow-band filters might be the best compromise.

There are regions (as shown in Figure 15) where the coverage from the individual speakers in the cluster interfere with each other rather than combine cooperatively. This interference is frequency-dependent. The solution is to reduce the contribution of some of the speakers of those problem frequencies so that interference is minimized.

The system would then require re-calibration to complement the above changes. That should do it.