HORIZON AUDIO SERVICES LTD.

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Resource Stage
Equalization Seminar

I'm sure we've all heard the word. We have equalization payments from the federal to the provincial provinces. We have equal opportunity employers. We are supposed to have equality of the sexes. We have kids saying 'that's not fair; he or she got to do this or go there, why can't I?' I used to tell my kids that they would be treated with equality, not equally. Whatever that means!

Equal; of the same measure or quantity, value, quality, number, degree or status as another.
Equalize; to make equal, uniform or constant.

We are going to talk about a specific kind of equality tonight; audio equality. Properly executed equalization of a sound system will help avoid this! (*show slide of Davis quote.*)

What is equalization?

- the process or action of making equal the frequency response of the sound system.
- smoothing out the system response to make it reasonably consistent, level, or equal across the frequency spectrum of the speaker or system.

Why do we equalize?

- To make the system sound more pleasing to the listener (tonal adjusting; use the equalizer as you do the tone controls on your stereo at home)
- To get rid of acoustic feedback
- To increase system acoustic gain
- To compensate for acoustic characteristics of the room the system is installed in

Typically, three of these are processes are done only when the system is initially set up or when a significant change is made to the system or the room. The other (first one above) is a process you as an operator will do often.

History; Back in the early days of the telephone when very long cable runs were used to transmit the voice, a great deal of attenuation (signal loss) occurred due to the resistance of the cable. Technicians could work around this loss by amplifying the signal along the way. However, cable also has capacitance, and they realized that there was also frequency dependant signal loss occurring due to this capacitance. Electronic circuitry was developed to differentially boost the frequencies that suffered greater attenuation. Since these circuits made all frequencies more equal in level, they were termed ***'equalizers'***.

Another type of circuit developed was the filter, which did not use amplification, and could attenuate a certain portion of the spectrum but allow the remainder of the spectrum to pass unaffected. Eventually equipment was developed that combined both these actions – frequency selective amplification and filtering. It was called the equalizer.

In 1967 the first 1/3 octave equalizer was introduced by Don Davis while at Altec Lansing Corporation, followed a year later by the 1/3 octave real time analizer, made by Hewlett-Packard.

Between then and today engineers refined the equalizer, developing various versions, including

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the parametric equalizer. Today we are using the 5th or 6th generation of digital signal processing equalizers, which offer excellent flexibility, control, value and economy. Here ends the history lesson.

Go to cable run slide and relate to video projection today.

There are a couple of terminology issues I should explain. First, I tend to use the word 'filter' when talking about the equalization process. This is partly because I usually want to 'filter out' a feedback or resonance point.

The second terminology issue is that of equalization and tone control. Whether they are old manual 1/3 octave models or digital processing models, what we do with equalizers for eliminating feedback, increasing acoustic gain, and compensating for room acoustics is ***'equalization'***. We are altering the electric signal going to the speaker to change its direct sound field in some manner. It is ***not*** 'tuning the room' or 'Eqing the room'. We can't change the room, unless we bring in the demolition crew or add sound absorbing and or diffusing panels.

What we do on the mixer input channel controls is ***tonal adjustment*** to obtain a pleasing tonal quality; it is ***not*** equalization, even though the manufacturers and most professionals call these controls 'the EQ'. I suppose labelling them as 'bass, midrange and treble' may not convey the professional image we all want.

One other point; I'm sure you've heard the word 'phase'. Phase is part and parcel with frequency. When we adjust a frequency response we are at the same time adjusting the phase response. Let me quote John Murray, a 30 year industry veteran who has worked for Electro Voice, MediaMatrix and Toa. "Any time you change the amplitude in a generally minimum-phase device like a loudspeaker, you will also affect the phase response and delay of the signal. Hi-fi home theatre purist types will tell you that equalizers are bad because they affect phase as well as amplitude. Well, if you have an amplitude problem in a speaker, with its accompanying phase distortion, and you fix the amplitude problem with the opposite EQ filter, guess what happens to the phase? It gets fixed too. As for hi-fi home theatre loudspeakers, the EQ filtering is done at the loudspeaker level and the components needed for this equalization are incorporated into the passive crossovers. In this way the purists are blissfully ignorant that any equalization has been done to their 'pristine' speakers."

One thing I have learned over the past 25 years is that every time the light bulb goes on and I suddenly understand something new, I realize that I still have much to learn. I'm certainly not the last word on equalization, frequency and phase, but hopefully tonight you can walk away with something new from our seminar.

What We Can and Cannot Equalize;

- We ***can*** EQ the response of a speaker. The EQ alters the frequency response of the electrical signal that is feeding the speaker, thereby altering the frequency response of the speaker.
- We ***cannot*** EQ a bump or hole in the response of the speaker or a system that is caused by a reflection off a surface combining with the direct sound of the speaker.
- We ***cannot*** EQ out problems caused by destructive interference. This is irregularities caused by the sound fields of two speakers combining acoustically to produce a series of additions and cancellations. This is an acoustic problem and evidence of poor system design and/or installation. This is one reason why we are only using one speaker here tonight.
- We ***cannot*** use EQ to fix the poor dispersion characteristic of a speaker.
- We ***cannot*** efficiently EQ out problems caused by the poor microphone use techniques of the system user.

The equalizer is a tool used to enhance a properly designed and installed system, not a permanent fix for a poor product or system.

Let's focus on one point above for a moment; that of the poor dispersion characteristic of a speaker. When the engineers design a speaker they decide what they want the horizontal and vertical dispersion of the model to be. A common dispersion is 90 degrees horizontal by 45 degrees vertical. If the engineers want the model to have this specification through the entire frequency range of the model including the bass region, this model will be a very sophisticated, expensive and likely rather physically large speaker. However for most of us common folks they use a high frequency horn that exhibits the dispersion spec, stick a woofer in the box and give it a model number. It's not quite that simple and for the serious manufacturers there's still quite a bit of engineering that goes into the typical two or three way box. The important thing to understand though is that it will only be the horn that is 90 by 45, and not the entire box. The size of the horn dictates how low in frequency it will hold its 90 by 45 angles, and also the crossover frequency. Small boxes will have a smaller horn, and therefore a higher crossover frequency. It follows then that the small cabinet will not exhibit any significant dispersion control until the higher frequencies. This may be fine for a background music speaker, but not for, example, a monitor, or a speaker such as the one we are using tonight.

But now for my main point concerning dispersion control. How does the speaker do off axis? We've got this speaker here covering an audience. Some of you are sitting directly on axis of the speaker. You will hear a response like this. But, what do those sitting off axis hear? Ideally it should be the same as those on axis, but in reality it will not be. Here's an example. I measured this speaker on axis, then at 20, 30, 40 and 50 degrees off axis. Keep in mind that the 40 degree curve is still within the 90 degree spec of the unit, and the 50 degree curve is just outside the spec. You can see quite a difference here as both the higher frequencies and volume generally fall off. With cheap speakers this curve will look much worse. With more expensive speakers, it will usually look even better.

This speaker is reasonably good, given it's price tag. Actually these curves are of this very speaker you are hearing tonight. These changes are inherent in the speaker itself; in the architecture of the horn and its phasing plug. We cannot do anything to change this; we cannot equalize this. Suppose we set our measurement mic at the 40 degree off axis point and set our equalizer accordingly. We can flatten that curve out, but what will that do to the on axis curve?

The lesson here is you should purchase the best quality speaker you can afford, even if it means you have to delay the purchase for a year or even two. You will have a better result and be much more satisfied in the end.

I add here that inexpensive monitors are notorious for having poor dispersion control. By inexpensive I mean the $500.00 area. I have measured many monitors that exhibit significant differences in their response curves even when only moving the measurement microphone about two feet horizontally. This is the same as taking about one step sideways, or when you have two people using the same monitor. You do not see reasonable dispersion control happening until you get close to the $1000.00 mark for a floor monitor. The first example here happens to be an Electro Voice Force model that sold for between $400.00 & $500.00 in its day. The second happens to be an Electro Voice Eliminator model that sold for about $800.00 or so. You can see how the dispersion control begins to look reasonably good.

Driver polarity problem slide *brief comment.*

Let's look quick at the last point above; we cannot efficiently equalize for poor microphone use techniques.

Use handheld mic at a distance to demonstrate system gain limitation from distance use compared to close up use. Use Audix to demonstrate the superior feedback rejection characteristics of it over others.

In many situations you can fix a feedback problem quickly and without spending money! Simply talk louder and /or closer to the microphone! The sound system can only work with what you put into it. This fix will work every time, guaranteed. You will gain 6 dB improvement in system performance simply by halving the distance between the microphone and the talker! Halving that again gives you a total of 12 dB improvement.

Analizers To Help Us Set The EQ;

I haven't got to the point yet where I can simply listen to a speaker and adjust the EQ to optimize the system, although given enough time I could come reasonably close. It is difficult to listen to a group of frequencies or a feedback ring and pick out exactly what band or frequency it is. So we use fancy analizers to quickly tell us these things. In the past it was the 1/3 octave real time analizer. This unit essentially takes the spectrum and displays the relative volume of each 1/3 octave band in a row. We use pink noise as our excitation signal. If we put the pink noise directly into the analizer we would see a straight line display. So if we pick up the noise reproduced through the sound system with an expensive instrumentation microphone we see what the sound system is doing to the spectrum. We adjust the EQ accordingly.

But wait! The 1/3 octave analizer is time blind. It doesn't know if what it is hearing is coming directly from the speaker or if there is a surface reflection added in, or if there is destructive interference causing that comb filtering.

Today there are several measurement platforms that employ some type of windowing to essentially take out the effect of reflections. My favourite is the TEF 20 originally developed by the Techron division of Crown, now owned by Goldline. It's my favourite because I know how to use it the best and I haven't learned the newest systems, such as the Smaart system we are using here. All my measurements you see on the slides were done with the TEF 20.

TEF means time, energy and frequency. It uses a swept sine wave signal with an accompanying swept filter to gather its information. This makes it very immune to other noise. We can zero right in on the direct sound or a reflection to look at the frequency characteristics. We can use the TEF for EQ setting, but we would have to take a new measurement with each EQ adjustment. This is time consuming. The Smaart system allows us to do this in real time. More on this shortly.

DSPs; Let's turn now to the current technology in equalizers. The 'old' days machine was the analogue graphic EQ with 'X' number of filters. Powered mixers included basic 6 to 8 band units. Economy stand alone models had 10 to 20 bands. We only used 1/3 octave models with 27 to 30 bands. Then digital processing came along and we suffered through the early generations of audio DSPs. Now manufacturers are into the 6 to 8th generation devices, and they really do work very well. On a per channel basis, they are less expensive than the old analogue units ever were. And they do so much more!

Today's DSPs offer parametric filters. On the old analogue graphic units we could only adjust the volume of the filter. We could not change it's frequency center or the width (Q) of the filter. There were analogue parametric models and they were good, but were expensive, as most models only had four or five filters. We have one at the shop if you want to get a good deal on one!

With DSP we can have many parametric filters on which we can adjust all three parameters; filter width, amplitude and frequency center. We can also apply delay to the signal, allowing us to align speakers that are not physically aligned. We can add compression and limiting for system protection. DSPs work fantastically as the main EQ/control portion of today's sound system. This is the only thing we use in our systems. Let's have Don do a demonstration of EQ setting for our speaker here.
Have DSP software up on powerpoint screen; have Smaart on other screen.

Mixer Input Strip 'EQ';

Once we have the system EQ all tweaked up as best we can, the system owners or users take over and have to operate it day in and day out. Not everyone using a microphone will sound great. Or you may feel that band needs a little sweetening. This is where we turn to the tone controls on the mixer input channels. Remember, this is NOT equalizing; it is TONAL ENHANCEMENT. But if you really want to, you can call it EQ when I'm not around.

Here's some graphs to illustrate what happens when we adjust the bass and treble controls on out JBL EON unit here. I measured with these controls at the 9:00 oclock, 10, 2, and 3 oclock positions; moderate cutting and boosting.

Now lets look at our mixer here and listen to some music etc while Don demonstrates what happens.

Have camera on strip on screen. Explain each control. Play music while Don adjusts controls. Dave talks while Don adjusts to demonstrate getting better intelligibility. Give guidance for settings. Smaart is on second screen showing the changes.

Conclusion; Well that's Equalization 101. Feel free to ask any questions you want. If none of us can answer them right now, we will find the answer for you later. Some of you may feel you want to do some fundraising in order to purchase some better quality equipment. Carl & Ron can help you with that, and this last slide may give you an idea to publicize your cause.

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