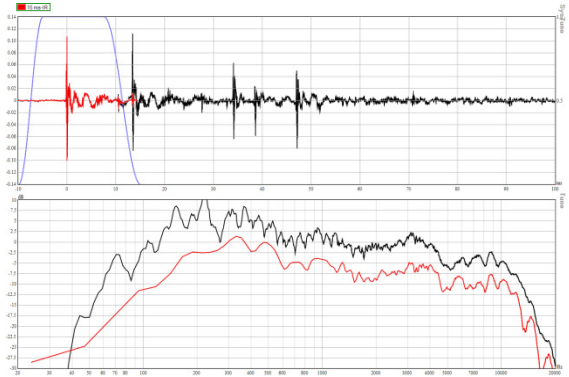
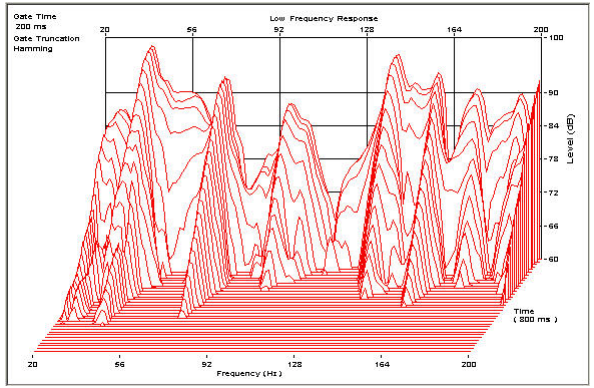
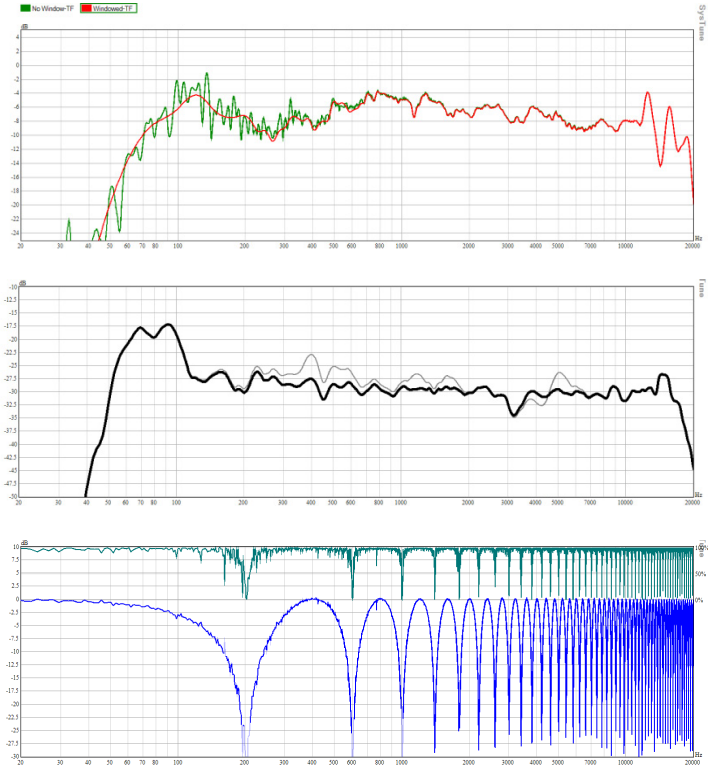
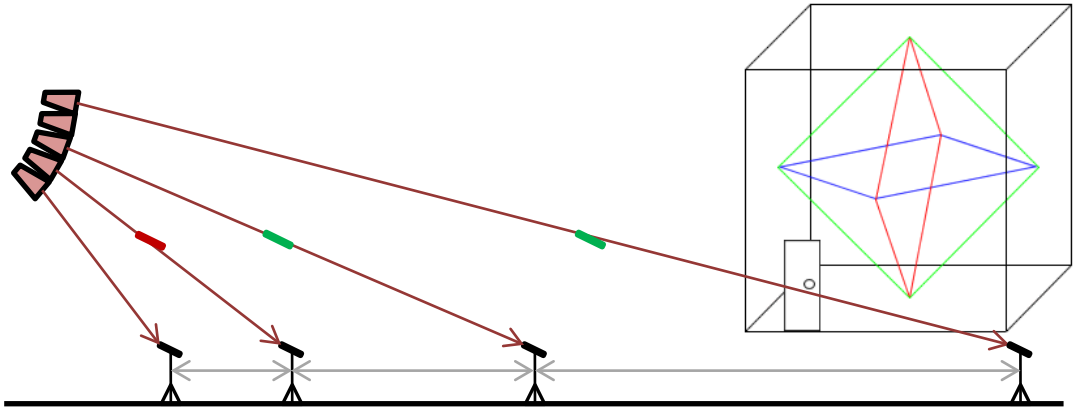
Properly Equalized Sound Systems









Introductions

John Murray

- B.S. in Broadcasting Production & Engineering, Ohio University 1975
- 13 yrs. in A/V system integration, designing, selling, & commissioning installed audio systems
- 13 yrs. in professional audio manufacturing, in product development, marketing & training for Electro-Voice, TOA Electronics, & Peavey MediaMatrix
- 7 yrs. as owner, ProSonic Solutions, professional audio sales representative firm
- 7 yrs. owner, Optimum System Solutions, professional audio system design & commissioning consultant firm
- Technical Editor: Live Sound International Magazine for 11 yrs.
- Technical Instructor of the Year: NSCA class on Line Array Speaker Systems, taught many classes for NSCA, InfoComm, & AES since '91
- Author: Dozens of technical articles for several audio trade magazines over 30+ yrs.
- Member: Equalization Standards Committees for all three trade organizations involved: AES, InfoComm, & SMPTE

Introductions

Charlie Hughes

- BS Physics – Georgia Institute of Technology 1988
- Senior Engineer, Loudspeaker Systems Design – Peavey Electronics (14 years)
- Chief Engineer, Professional Division – Altec Lansing Technologies (3 years)
- Founded Excelsior Audio in 2005 – Loudspeaker design, consulting, and measurements
- AFMG – Key Projects Manager, advanced technical support, and training since 2010
- Member, Audio Engineering Society since 1994
- Member, Executive Council for NC chapter of the Acoustical Society of America
- Member, Consumer Technology Association & International Electrotechnical Commission (IEC)
- Recipient of 2013 CTA Technology Achievement Award for work on standards development
- Standards committees for AES, CTA, IEC, and InfoComm
- Holds 5 patents relating to loudspeaker design and directivity control

PowerPoint Slides

www.excelsior-audio.com/EQ2016

The Purpose of this Class

The question isn't whether or not a speaker system should have a flat response. The flat response it should have is the anechoic, direct-sound response , which I call a *Neutral Transfer Function*. It produces the most natural timbre of the source from the speaker system.

The question should be "Is my measurement system, including the microphone quantities and positioning, set up to reveal the speaker system's isolated direct sound so that it can be equalized for a neutral transfer function."

The Purpose of this Class

I don't always want my measurement system to show me what I hear. Not everything I hear from a sound system in a room can be corrected by EQ. Most often I want my measurement system to show me what I can actually fix with equalization. This is the purpose of this class.

Syllabus

1. Types of Sound System Equalization (Murray)
   a. Source equalization
   b. Speaker equalization
   c. Purpose of speaker equalization – neutral transfer function of direct sound

2. What Can Be Equalized? (Hughes)
   a. Input signal / direct output from loudspeaker
   b. Very close reflections/adjacent loudspeakers (1/3 wavelength, < 120 degrees)
   c. Air attenuation
   d. Preventing excitation of ringing room modes (Murray, demo)

3. Measurement Systems (Murray)
   a. RTA
   b. TDS, TEF
   c. Dual-channel FFT (Hughes)

Syllabus

4. Measurement System Fundamentals (Hughes)
   a. Frequency resolution
   b. Signal acquisition window
   c. Matched length excitation & FFT block size (demo)
   d. Impulse response window
   e. Coherence & data validity

5. How to Make Good Measurements (Hughes)
   a. Thing to always keep in mind.
   b. Large, early reflection-free space (first reflections arrive long after the direct sound)
   c. Quite space & signal to noise ratio

6. Ways to Measure and EQ a Loudspeaker Array (Murray)
   a. In the shop
   b. On-site after installation
   c. Array shading

Syllabus

7. Measurement Compromises (Hughes)
   a. Eliminating/minimizing reflections & IR windowing
   b. Mic placement & typical cases
   c. Trade-off between isolation and resolution
   d. Linear vs. non-linear loudspeaker operation

8. The Lows Need to be Treated Differently (Murray)
   a. Long wavelengths and room dimensions
   b. The Schroeder Frequency
   c. In-the-shop using Keele's extreme-near-field technique
   d. Spatial averaging and steady-state measurements
   e. Quick demos

9. Making EQ Decisions
   a. Do I have valid data & is the response equalizable (minimum-phase)?
   b. Does the measurement seem to match what I'm hearing?
   c. Should the measurement match what I hear or show what I can fix with EQ?
   d. Should I try to equalize it?
   e. Demonstration

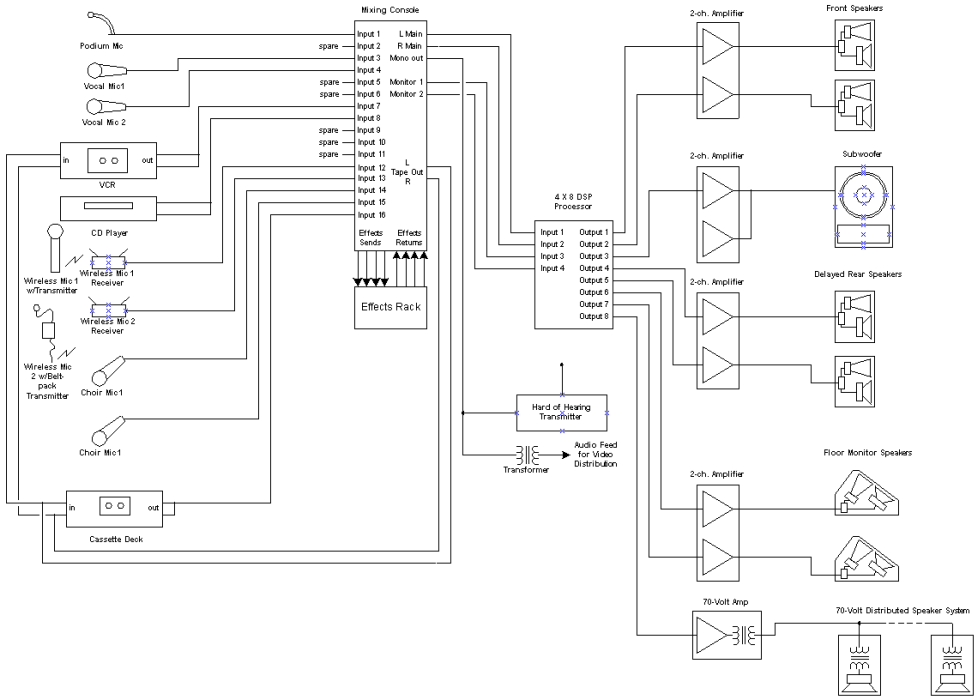
Section 1

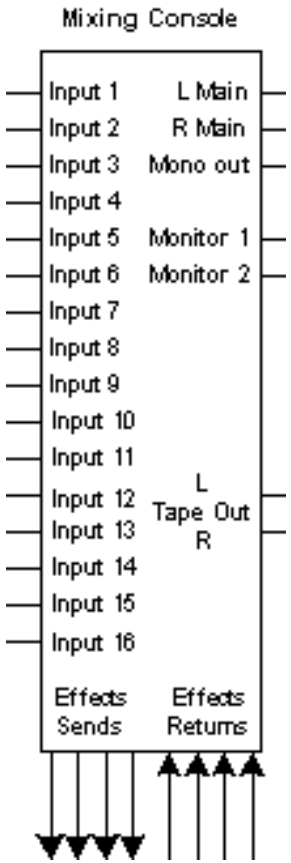
Types of Sound System Equalization

John Murray

Types of Sound System Equalization

A Typical Church Sound Reinforcement System

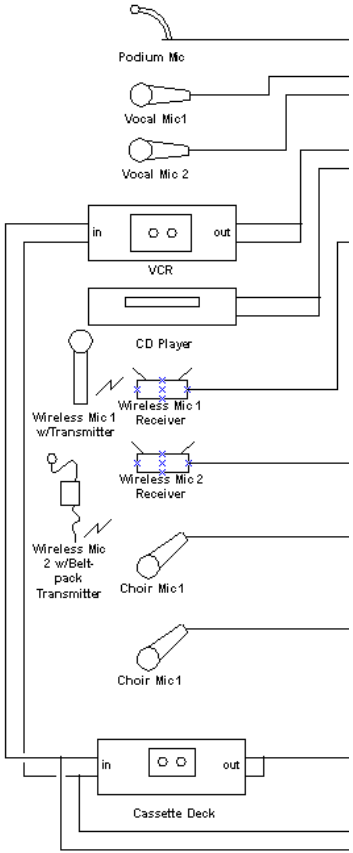




Source Equalization

- Source equalization is done on the input strip of the mixing console,
- It's purpose is to correct or modify the frequency response of the source and to restrict the response of a microphone where there is no source signal (HPF)
- It does not necessarily aim for a natural or accurate response, but is often used creatively to enhance the sound quality of the source

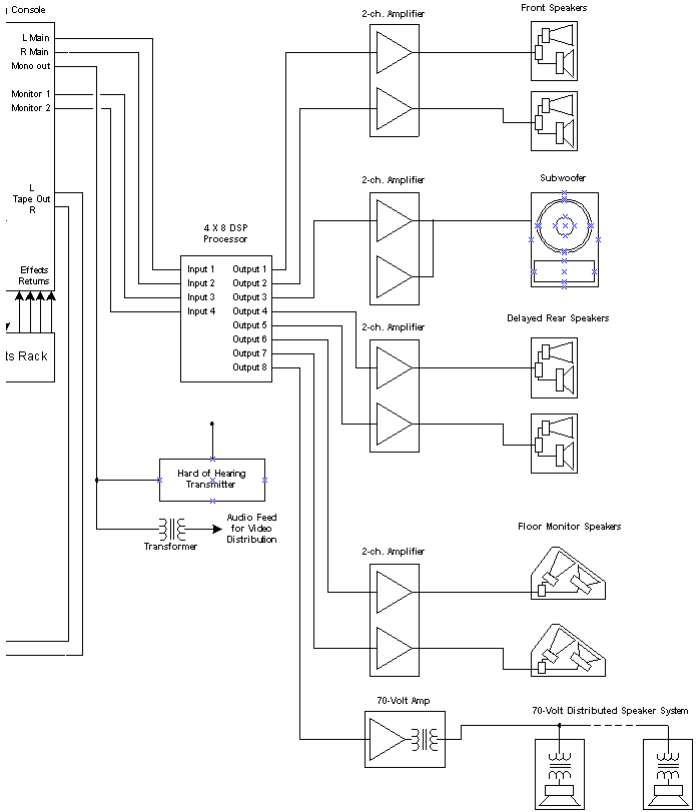
Source Equalization



- Note that if source equalization is needed to get a line-level program source like a CD player to sound right, this indicates that the sound system's speaker equalization is not set correctly
- Prerecorded line-level sources should need no EQ to sound right using a properly tuned speaker system
- We will not be covering source equalization in this class

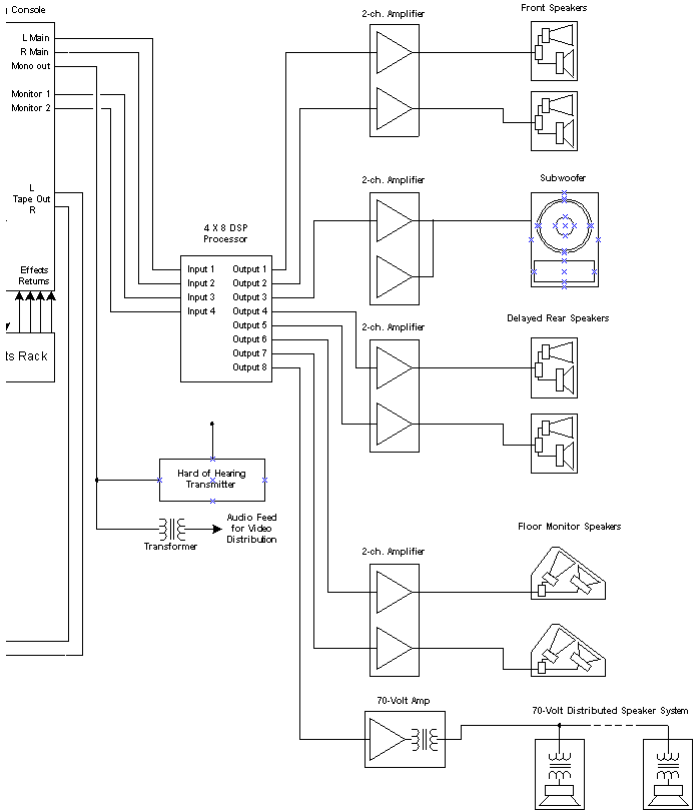
Speaker Equalization

- Speaker equalization is strictly technical, not a creative tuning
- That special kick-drum thump or extra cymbal sizzle should be done at the appropriate input EQ strip, not on the speaker system equalizer
- That way creatively tuned mics and program sources w/o EQ both sound right



Speaker Equalization

- The purpose of speaker equalization is to make the direct-sound (anechoic) frequency response of each speaker type in the system a flat response
- I call this a *neutral transfer function* and it ensures that the creative music mix is easy to obtain and a commercial application's mix sounds natural



Section 2a to 2d

What Can Be Equalized?

Charlie Hughes

What Is Equalization?

Dr. Eugene Patronis, Jr.

"Equalization as applied to sound reinforcement systems refers to the process of electrically modifying the input signal being fed to the system so as to correct for some anomaly or anomalies exhibited by the system's natural behavior."

"Equalization is both the most often used, as well as the most often abused signal processing function."

What Can be Equalized?

We can EQ the input signal to the loudspeaker.

By applying EQ to the input signal we can effectively change, hopefully improve, the transfer function (frequency response) of the loudspeaker.

We can't EQ the room!

EQ adjustments for room effects may help at some (most) listener locations, but it's only a band-aid.

EQ may optimize the overall subjective impression of the sound quality at the expense of the accuracy of the direct sound from the loudspeaker.

What Can be Equalized?

EQ Classifications

Technical Equalization – pass band restriction, driver protection & smoothing

Examples: crossover filters, high pass filters, parametric EQ (flat direct-field response)

Corrective Equalization – removing unwanted behavior from loudspeakers

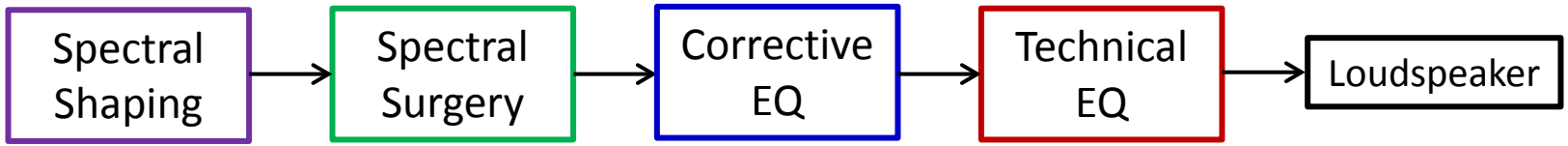
Examples: loudspeaker response smoothing, array EQ, boundary loading effects

Spectral Surgery – the absence of energy is often better than its presence

Examples: feedback notch filters, room mode notching

Spectral Shaping – obtaining a preferred spectral balance

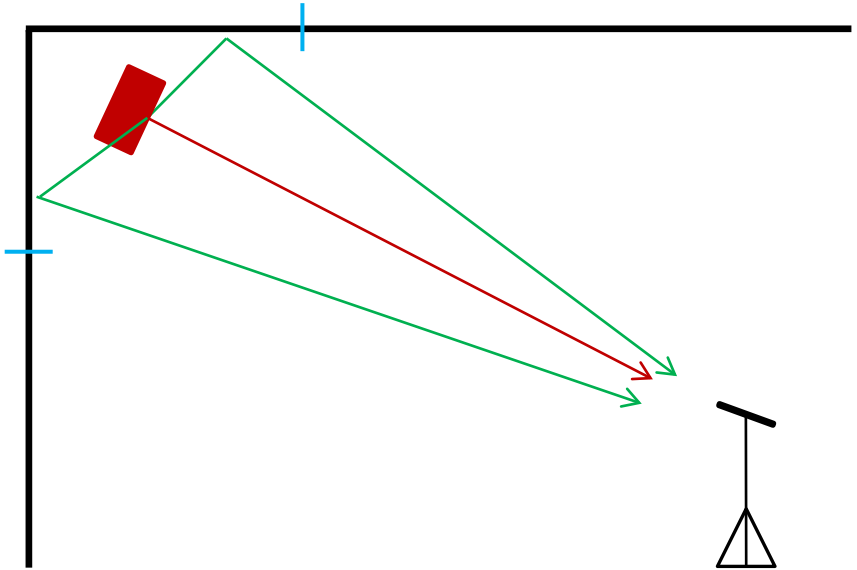
Examples: "smiley face" curve, bass haystack, gradual LF increase

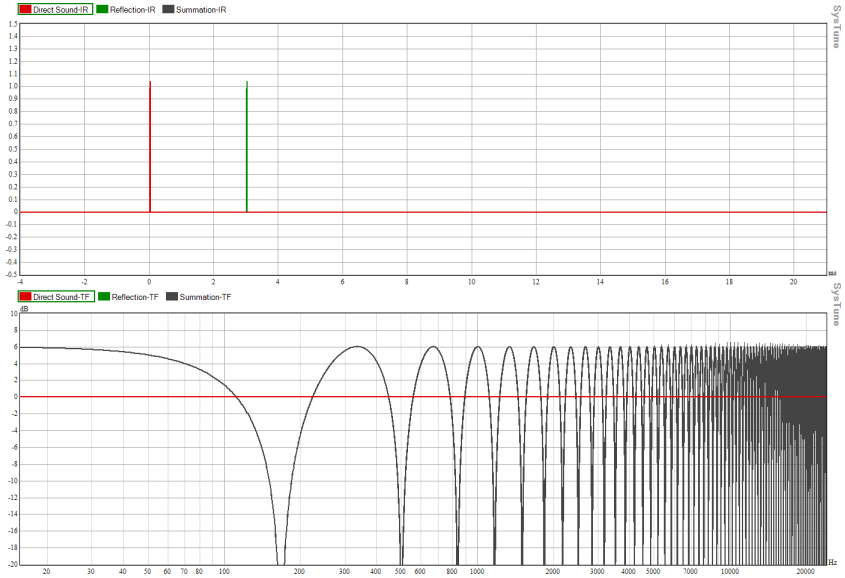


What Can be Equalized?

Boundary Loading (Corrective EQ)

The effects of reflections very close to loudspeakers can be equalized.
"Very close" is frequency (wavelength) dependent.





What Can be Equalized?

High Frequency Air Attenuation

Air molecules cause a decrease in SPL at higher frequencies (low pass filter). Sound travelling longer distances is subject to greater attenuation.

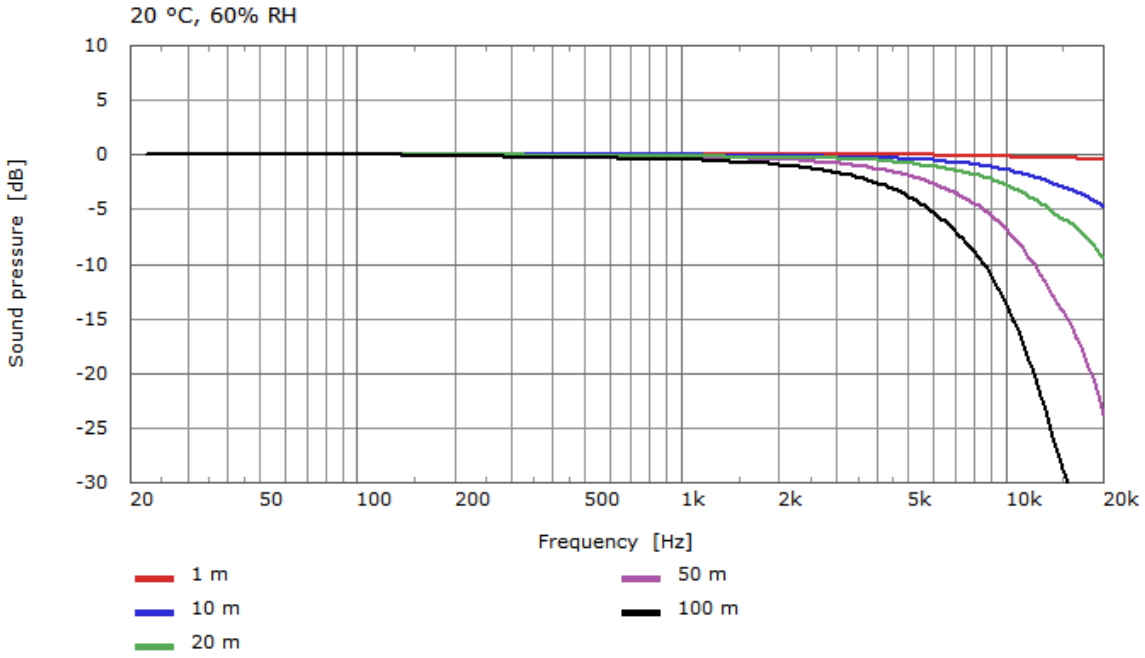
The amount of attenuation is frequency dependent.

The attenuation also changes with temperature and the moisture content of the air (relative humidity).

Tune for the average temperature and humidity in the venue during typical use conditions.

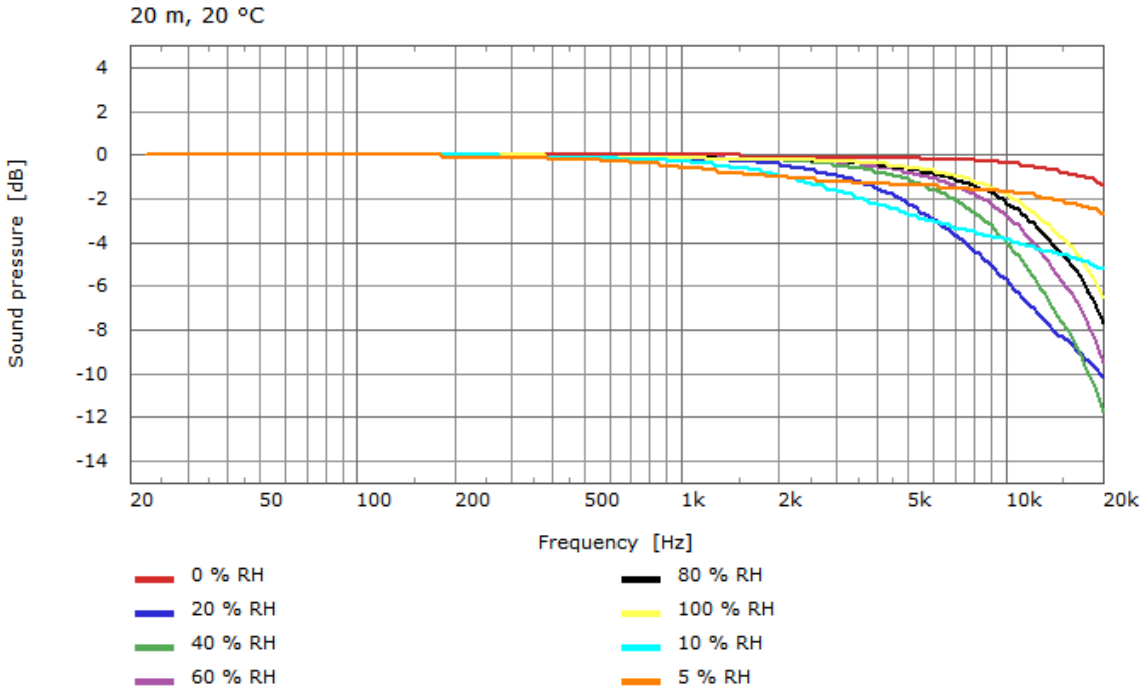
What Can be Equalized?

High Frequency Air Attenuation



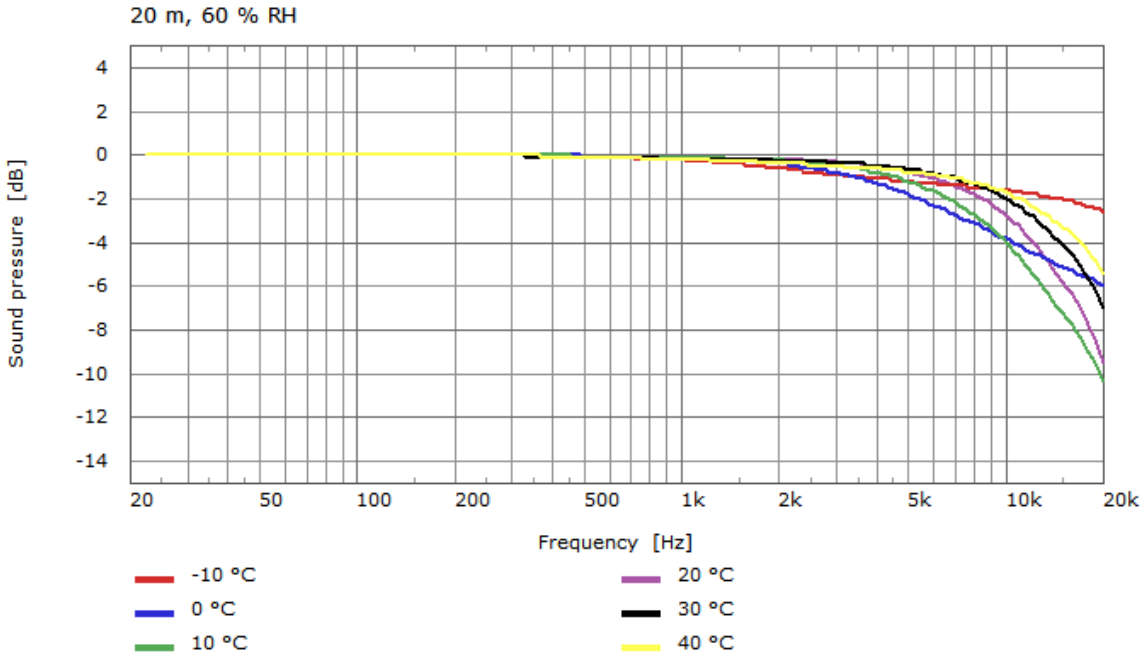
What Can be Equalized?

High Frequency Air Attenuation



What Can be Equalized?

High Frequency Air Attenuation



Section 2e

Preventing Excitation of Ringing Room Modes

John Murray

Ringing Room Modes

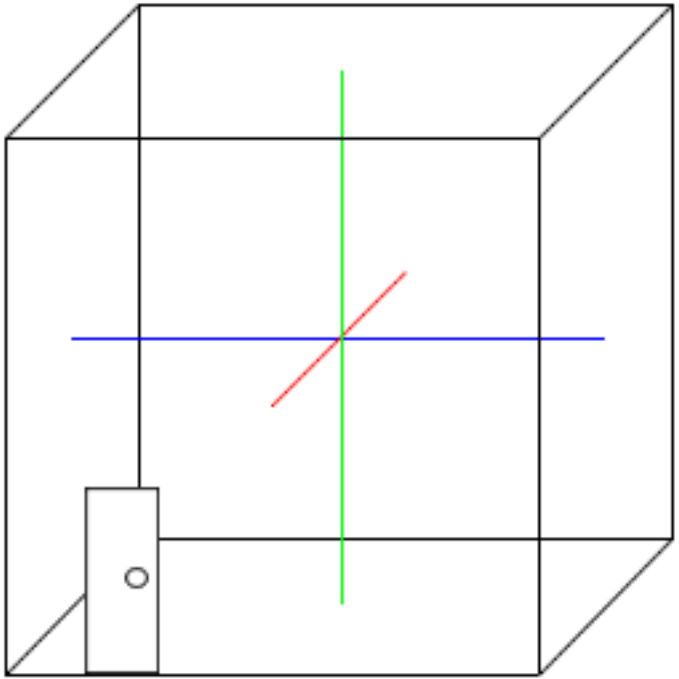
- What are modes, or standing waves?
- Let's have the Physics Girl, Dianna Cowern, who has a BS in Physics from MIT and was a Research Fellow at the Harvard-Smithsonian Center for Astrophysics, explain the underlying fundamentals to us…..

video placement: downloaded from

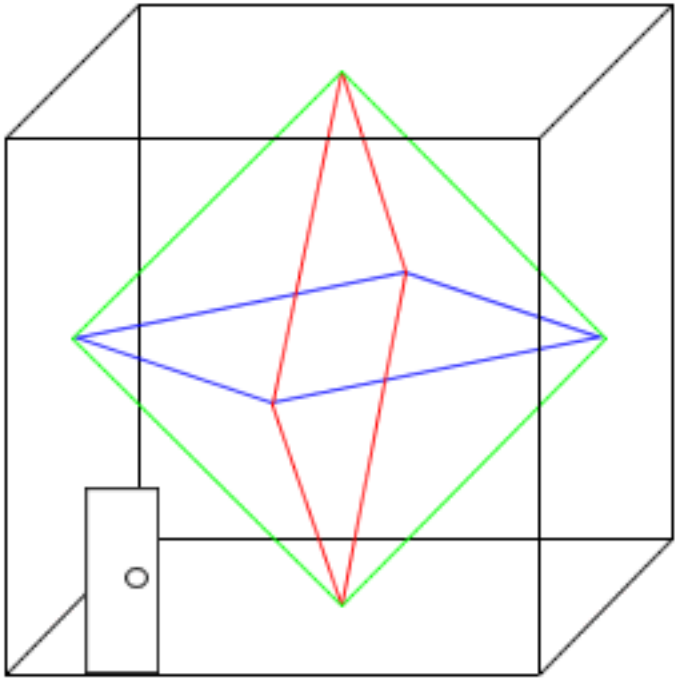
www.youtube.com/watch?v=wYoxOJDrZzw

Axial Room Modes

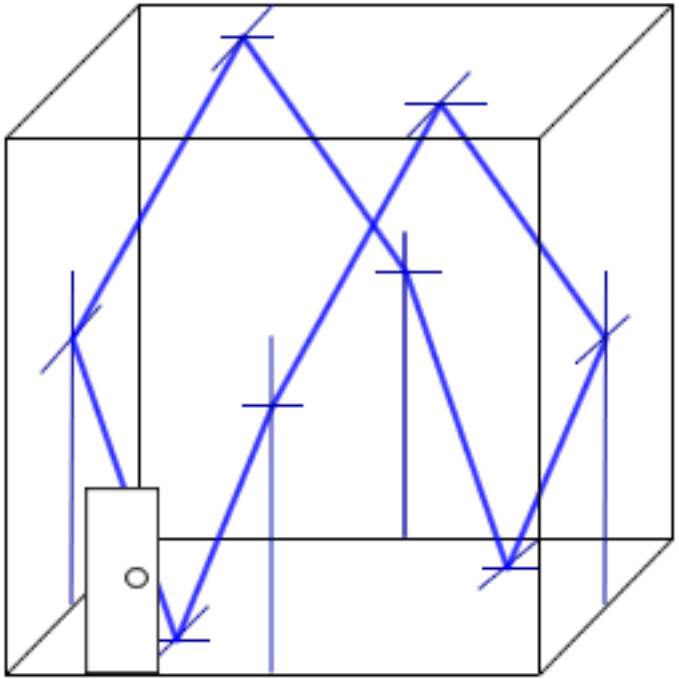
Note: the first mode occurs when the largest room dimension = ½ λ



Tangential Room Modes



Oblique Room Modes



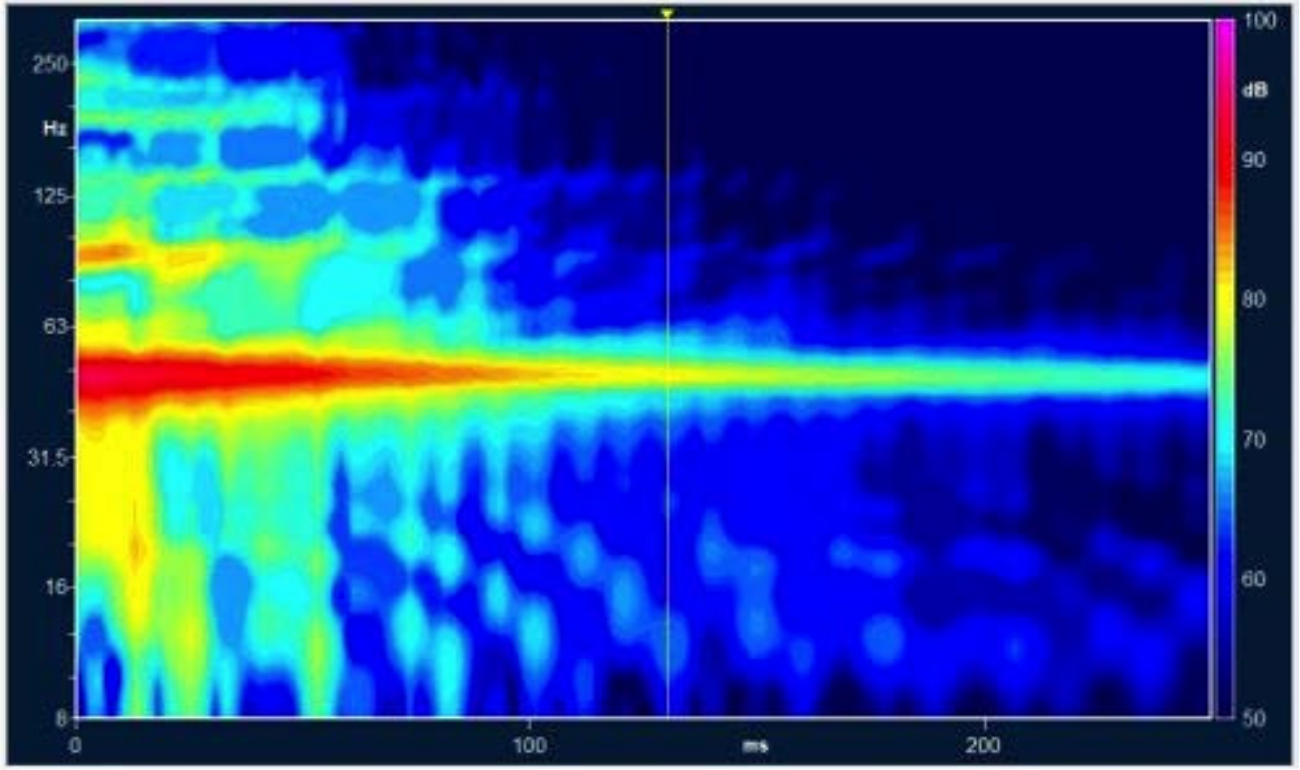
Ringing Room Modes

- The first to address the phenomena of ringing room modes in relation to sound systems was the originator of equalization, “Doc” C. P. Boner
- Doc Boner not only used broadband filters to correct speaker response, he would also add narrowband filters at those specific frequencies to minimize excitation of any of the room’s audibly ringing resonances
- Pipe organ tuners have been minimized these “Bull Notes”, otherwise know as standing waves or Eigentones, literally for centuries

Ringing Room Modes

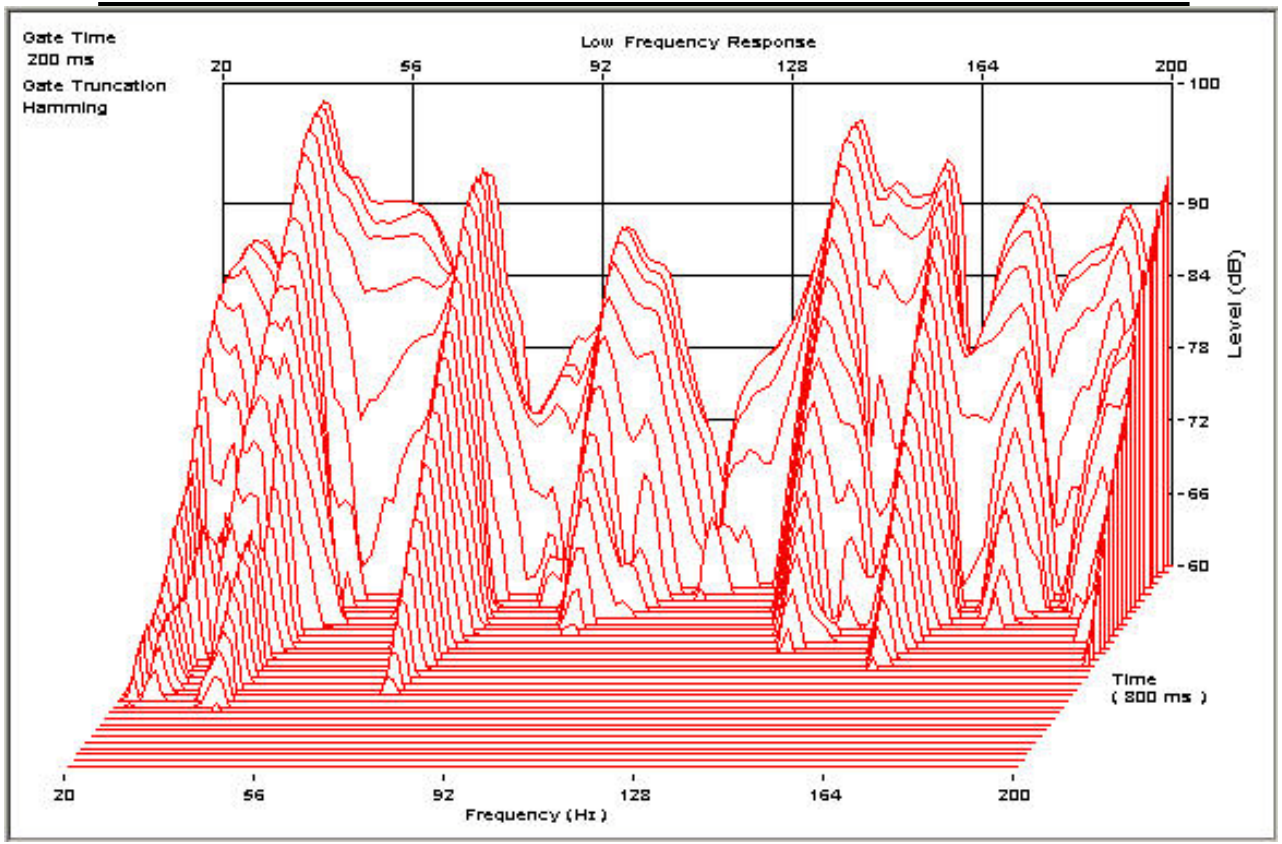
- Today's conventional equalization process is speaker correction, which does nothing to lessen the excitation of the room's resonances, unless a brute-force HPF is employed to kill all of the bass response
- In the case of rooms that have prominent resonances, it is far better to use narrowband 1/6 to 1/10 octave notch filters to prevent their excitation, and thus remove their masking effect to the clarity of the system's program
- We now have the tools to easily reveal and remove these problematic frequencies from the response of a sound system's speakers

A Room Mode as a Frequency vs. Time Display

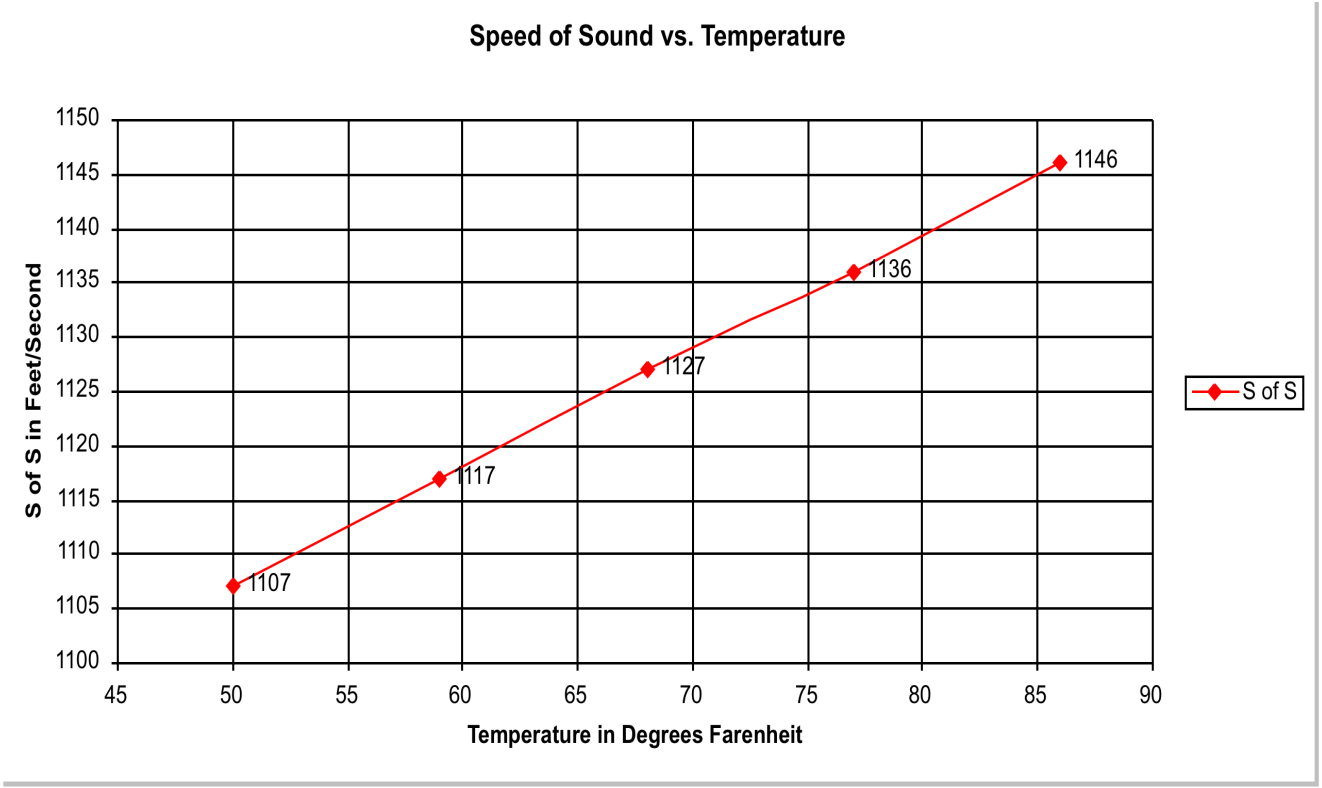


courtesy of www.acousticfrontiers.com

Room Modes as a 3-D "Waterfall"



Speed of Sound vs. Temperature



Bandwidth of Room-Mode Notch Filters

- Indoors, the temperature generally does not vary more than 10° F
- From the chart, a 10° F change results in approximately a 1% change in frequency for room modes
- Therefore, the filter bandwidth should be no wider than about 1/6 octave (Q of 8.5) and no narrower than about 1/10 octave (Q of 14)
- As for the amount of attenuation, it will vary with the particular resonance level
- I generally attenuate by ear until it no longer stands out audibly

Section 3a and 3b

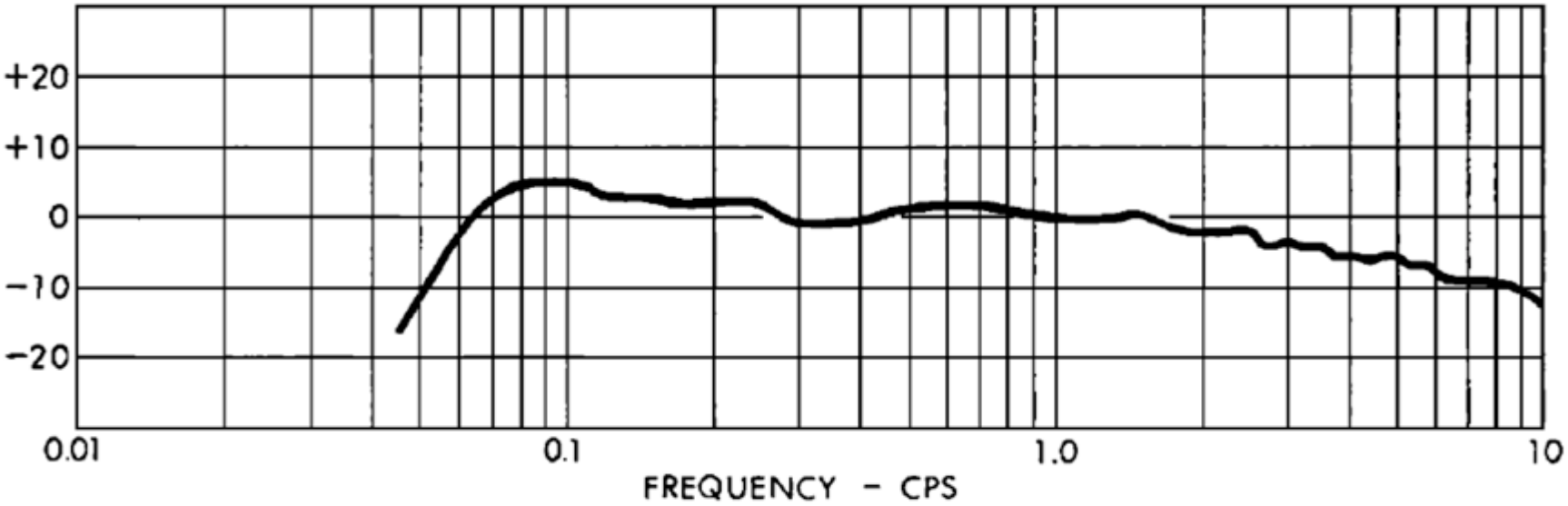
Measurement Systems:

RTA and TDS

John Murray

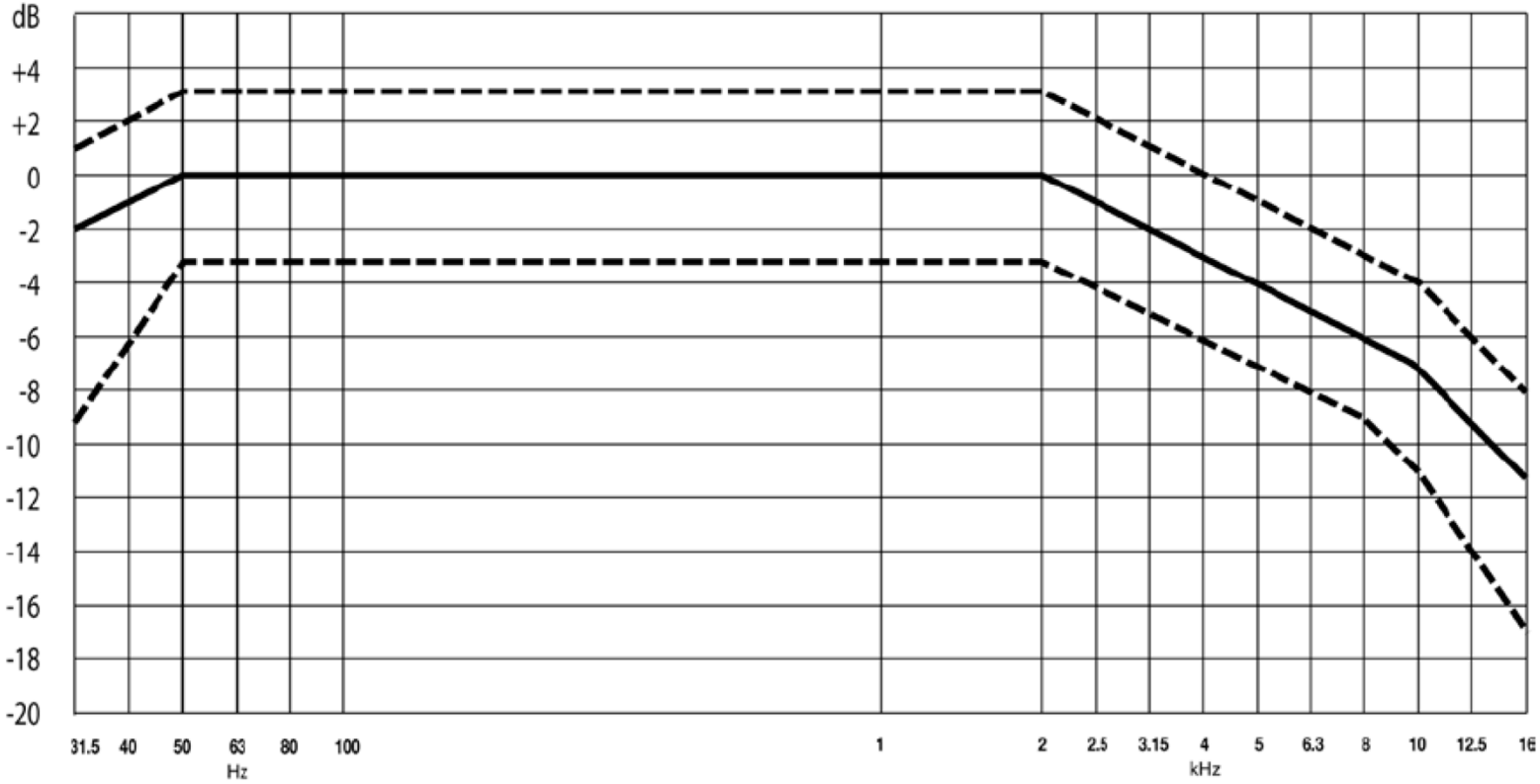
Tuning a System Using an RTA

"Doc" Boner's Time-Blind "House Curve"



***from Minimizing Feedback in Sound Systems and Room-Ring Modes with Passive Networks*, Journal of the Acoustical Society of America, vol. 37, no.1, 1965.**

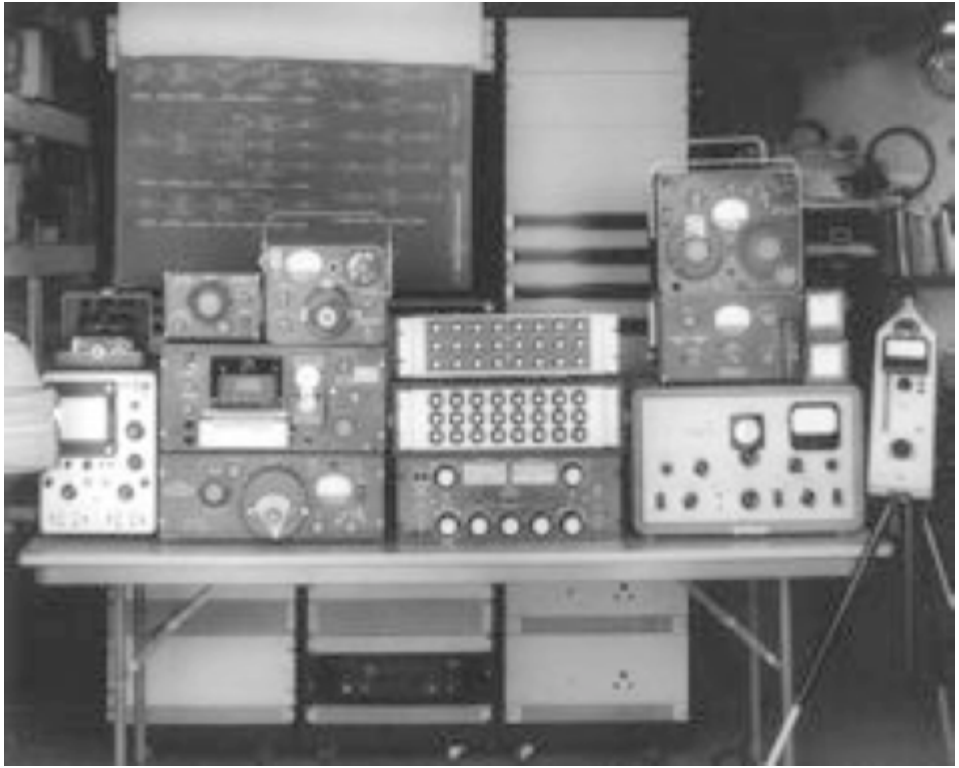
SMPTE X-Curve: 2010



"Doc" Boner's Time-Blind "House Curve"

- Boner's house curve was intended for large rooms that contained a statistical reverberant field
- It represents a speaker reproducing a natural sound quality in a relatively large room, with the measurement microphone beyond critical distance, in the reverberant field
- As discussed earlier, in addition to using broadband filters to create this curve, he also used narrowband filters to notch out room mode excitation by the speaker system

Pre-RTA Equalization Rig



Don Davis' Altec Lansing "Acousta Voicing" rig, not exactly carry-on luggage

The RTA Ended Narrowband Filtering

The Altec/HP 8050A, the first RTA

The Real-Time Analyzer

- In the near-field of a speaker, an RTA curve should be flat if the sound system is tuned properly
- Thus from a near-field to a far-field mic position, an RTA shows a gradually changing response, until the mic is beyond the critical distance, and the reverberant field dominates the direct sound level
- An RTA house curve in the reverberant field of a large, reflective room is the power response of the speaker system, as affected by the room's surface- and air-absorption

The Real-Time Analyzer

- Since an RTA's response varies with distance in the near-field, and is the power response of the particular speaker system in the reverberant far-field of a particular room, there is no one fixed RTA frequency response target that can be used for any speaker in any room
- With experience, this gradually dawned on the sound industry and tuning systems by instrument fell out of favor
- The use of an RTA, still good at finding feedback frequencies, took a back seat to tuning by ear for system EQ

3D Magnitude vs. Frequency vs. Time Measurement

3-D Audio Measurement

- The ultimate goal of audio measurement is to capture sound in terms of 3-Dimensional log-Magnitude (dB) vs. log-Frequency vs. Time
- It was introduced to the AES in the form of Time Delay Spectrometry (TDS) in a paper by Richard C. Heyser of Jet Propulsion Laboratory (JPL) in October of 1967
- This was 5 years before the introduction of the first real-time analyzer, the Altec 8050A
- In fact, this was really 12 years too early for the audio community to fully absorb yet

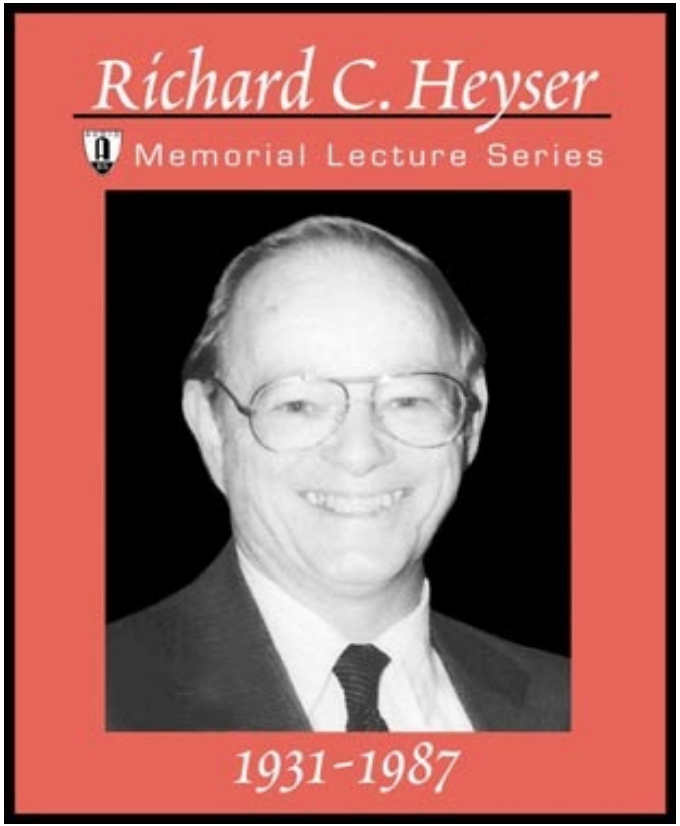
Time-Delay Spectrometry (TDS)

- It wasn't until Don Davis, formerly of Altec's Acousta Voicing training, created a SynAudCon workshop on TDS in 1979 that the audio community began to become aware of it
- This was the seminal event that began the careers of many people in the audio industry, including this presenter
- Never before had any of us seen sound as an Energy Time Curve (ETC), or seen a comb filter in both linear- and log-frequency scale, or a frequency response as a 3-D waterfall display with a measured time component

Time-Delay Spectrometry (TDS)

- In short, Dick Heyser revolutionized audio measurement by enabling anyone to make semi-anechoic measurements in the field for the first time
- TDS introduced us to FFTs, IFTs, ETCs, 3-D waterfalls, and the concept of a transfer function
- It is the reason I, and many others, have a career in audio today

Heyser's Time-Delay Spectrometry



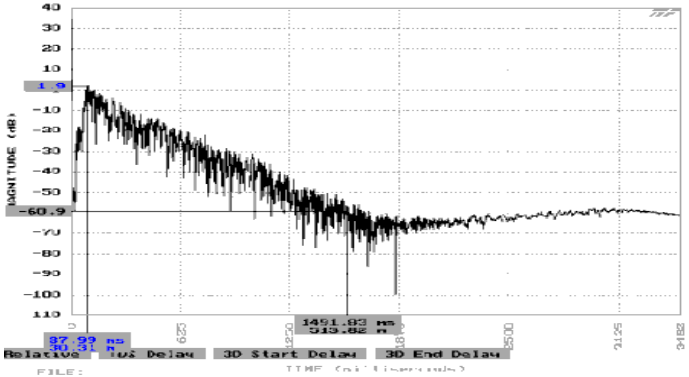
- It is the reason the AES has the Heyser Lecture Series every year at their convention

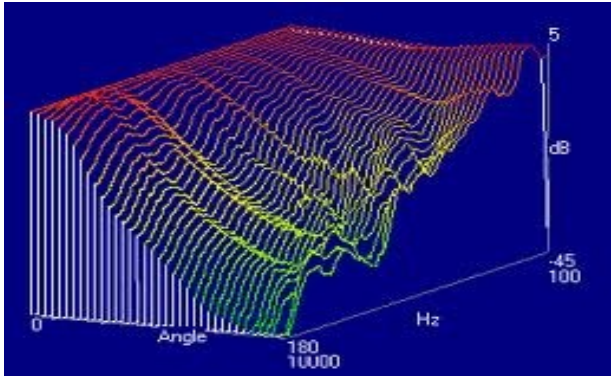
Techron TEF-10 & TEF-20

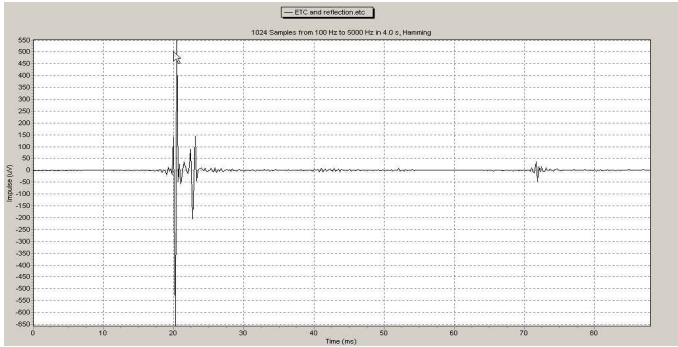
courtesy of audioheritage.org

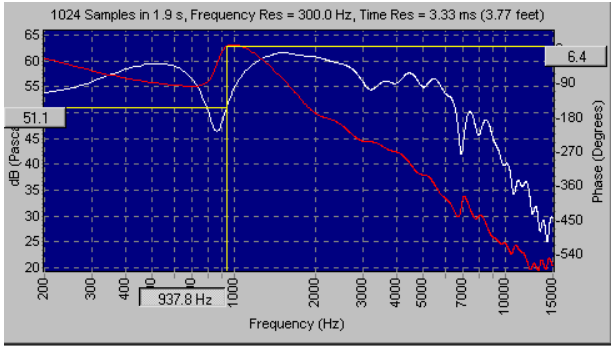
courtesy of sekaimon.com

TEF Displays: ETC, 3-D, Impulse, Magnitude/Phase

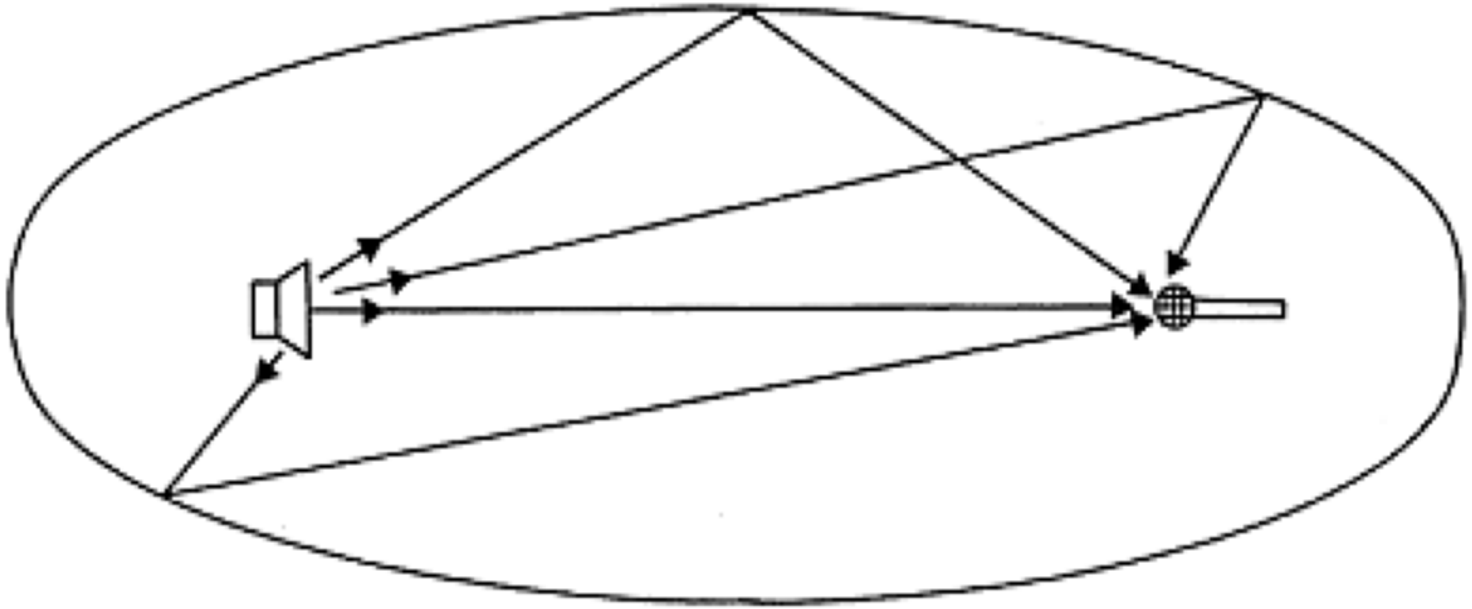








Heyser's ellipse: direct-sound isolation



courtesy of MC-squared

Focusing on the Direct Sound

- Human hearing is adept at discerning the "naturalness" of the direct sound of a source in a variety of acoustical environments.
- Floyd Toole relates - "The same sound source - a voice or musical instrument, for example - will take on some of the character of different rooms, and yet they can still be recognized as being the 'same' sound sources. Within some range of 'normal' rooms, we seem to have a built-in ability to 'listen through' a room to attend to even minute details of the sound source."

– Floyd Toole, *Sound Reproduction: Loudspeakers and Rooms*

Focusing on the Direct Sound

- If our hearing wasn't adept at discerning the direct sound of a source, we would be extinct. We would have failed to recognize a lion's growl in a canyon, as we would have in the bush.
- Likewise, a baby quickly learns to recognize their mother's voice, whether masked by the comb filtering of a room's surfaces, or from down the hall with a TV blaring.

Focusing on the Direct Sound

- This should make it rather obvious that, when the direct (anechoic) frequency response of a speaker system is equalized to be flat, like a near-field studio monitor, it will produce the most natural sound, regardless of room acoustics
- *Equalizing the direct sound of a speaker away from a flat direct-sound frequency response will only result in less natural sound quality, regardless of the speaker or room characteristics*

Section 3c

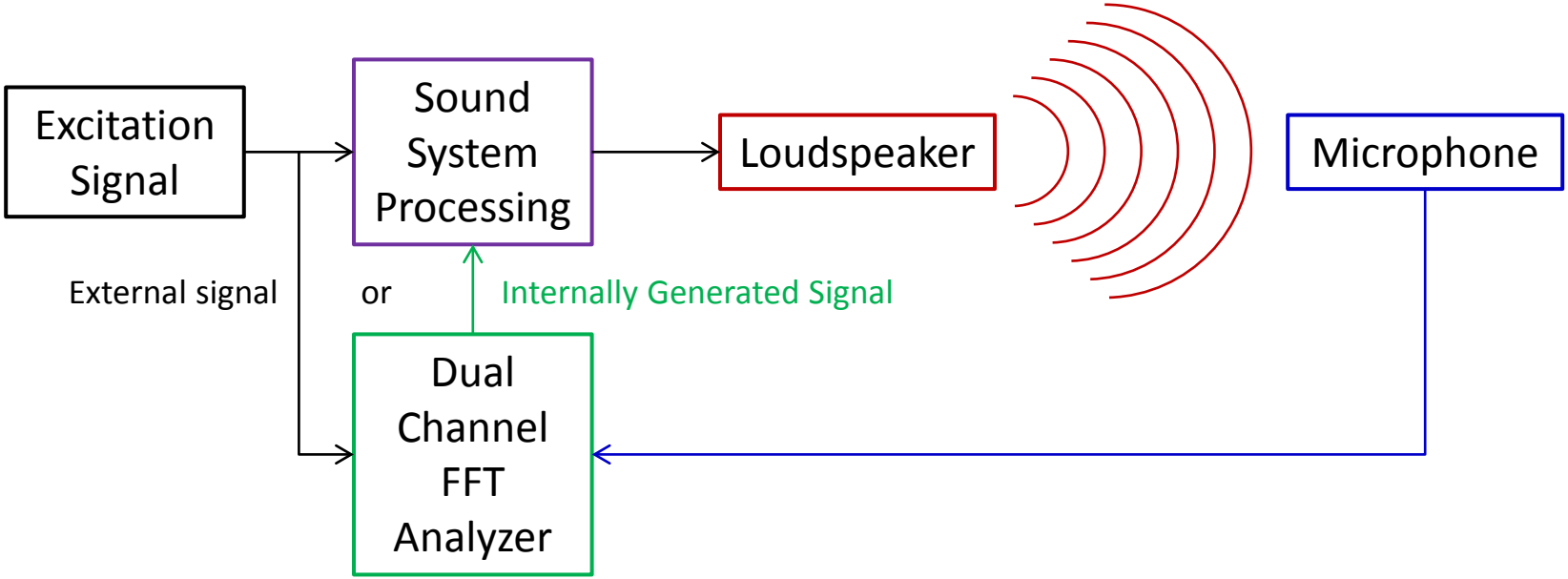
Measurement Systems:

Dual-Channel FFT

Charlie Hughes

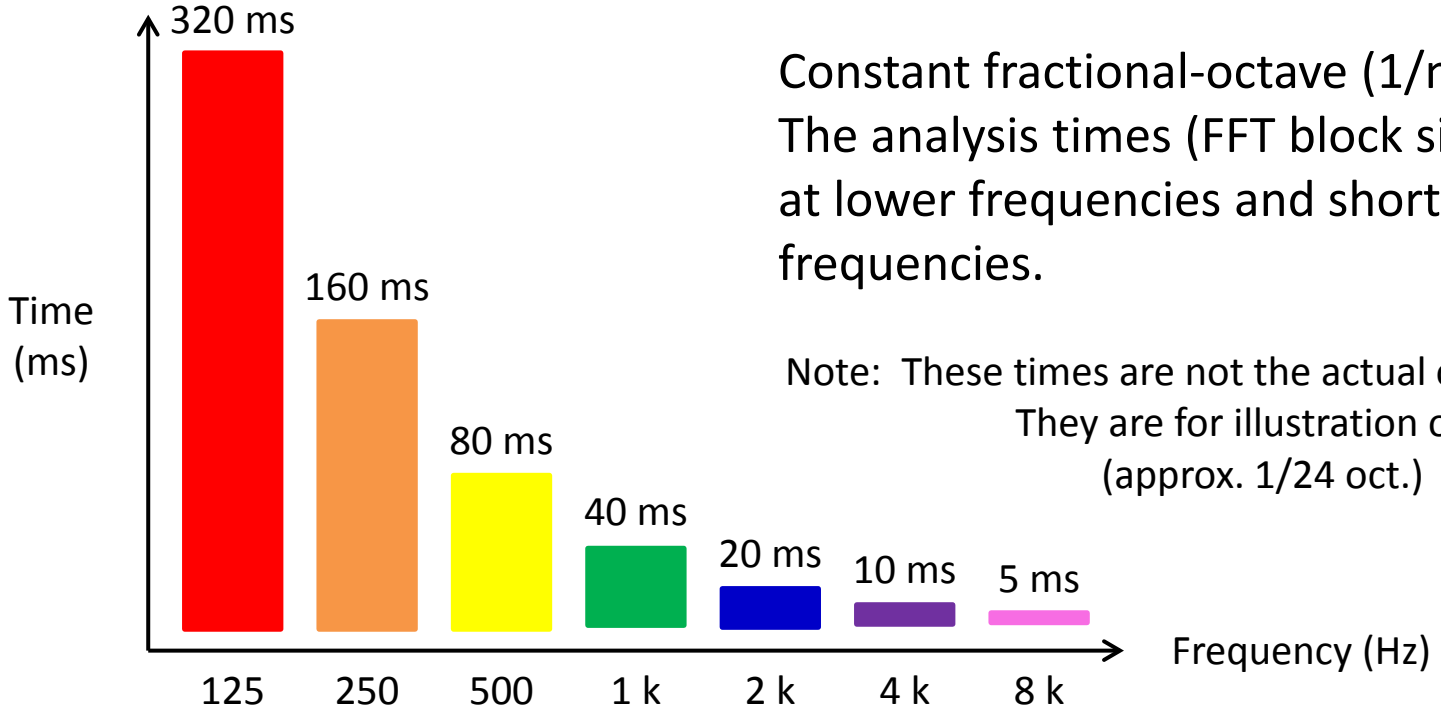
Measurement Systems

Dual-Channel FFT Measurement Systems



Measurement Systems

Dual-Channel FFT – SIM (Meyer Sound Labs)

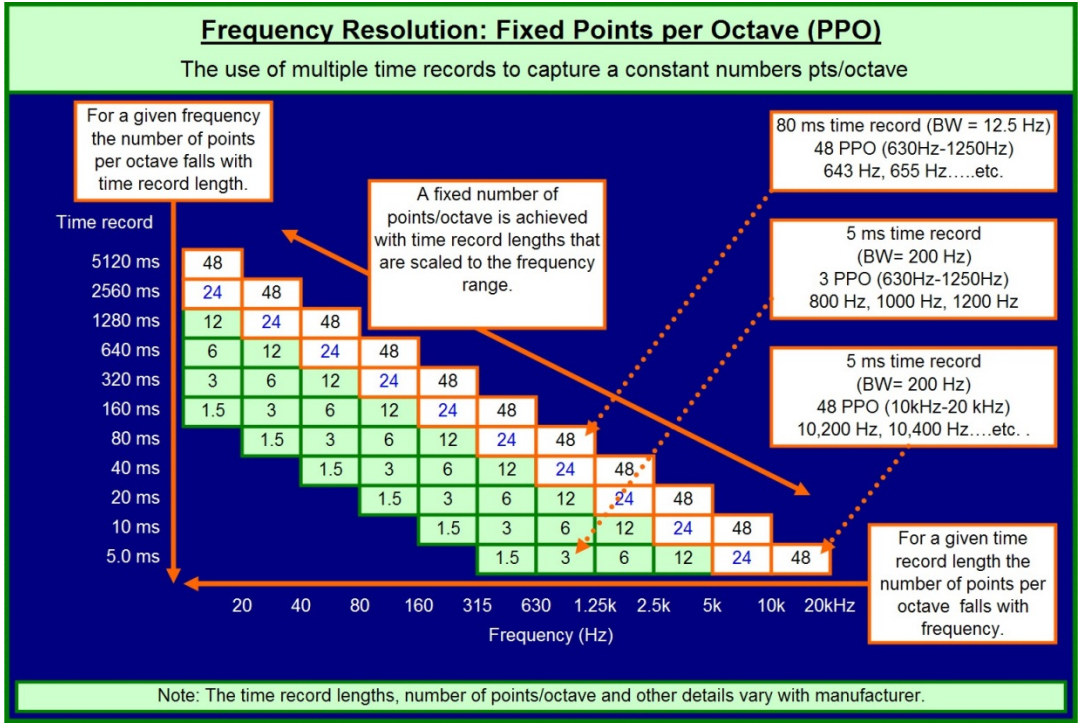


Constant fractional-octave (1/n) bandwidth: The analysis times (FFT block size) are longer at lower frequencies and shorter at higher frequencies.

Note: These times are not the actual ones used in SIM. They are for illustration only. (approx. 1/24 oct.)

Measurement Systems

Dual-Channel FFT – Fixed Points per Octave (FPPO)



The analysis times (FFT block size) are longer at lower frequencies and shorter at higher frequencies.

Graph courtesy of Bob McCarthy

Measurement Systems

Dual-Channel FFT – Smaart (Rational Acoustics)

- Can use an external stimulus (source independent)
- Like SIM, performs constant fractional-octave bandwidth analysis (FPPO), using “decimated time windows” (DTM)
- Can perform fixed-length FFT block size analysis (same analysis time for all frequencies)
- Some IR windowing capabilities
- Relies mainly on FFT block size to suppress the effects of reflections in the measurement data

Measurement Systems

Dual-Channel FFT – SysTune (AFMG Technologies)

- Can use an external stimulus (source independent)
- Only performs fixed-length FFT block size analysis (same analysis time for all frequencies)
- Traditional IR windowing to suppress the effects of reflections in the measurement data
- Time-Frequency-Constant (TFC) IR windowing

Measurement Systems

Dual-Channel FFT – Others

- EASERA
- ARTA
- Fuzz Measure
- Room EQ Wizard
- HOLM Impulse
- …

Section 4

Measurement System Fundamentals

Charlie Hughes

Measurement System Fundamentals

FFT Data Block

To perform an FFT we need a block of data for the analysis. This FFT block has a certain size associated with it. It is typically in binary increments (2^n) to make computer-based calculations faster.

$$FFT\ Block\ Size\ =\ Number\ of\ Samples$$

(Typical values 2,048; 4,096; 8,192; 16,384; 32,768; etc.)

Measurement System Fundamentals

Sample Rate

The sample rate can also be expressed as a period.
This is the reciprocal of sample rate.

$$Sample\ Period = \frac{1}{Sample\ Frequency}$$

(Typical values 22.7 us, 20.8 us, 11.3 us, 10.4 us, etc.)

Measurement System Fundamentals

Frequency Resolution

The FFT block size and the sample rate determine the frequency resolution and the LF limit of the measurement.

$$FFT\ Size\ =\ Sample\ Period\ *\ Number\ of\ Samples$$

(FFT Size = 20.8 us * 8,192 samples = 0.17 s)

$$\Delta frequency\ =\ 1/\Delta time$$

Freq. Res. = 1 / 0.17 s = 5.9 Hz

Measurement System Fundamentals

FFT Bins

In the frequency domain the FFT “fills” each discrete bin with the energy contained in the time domain signal.

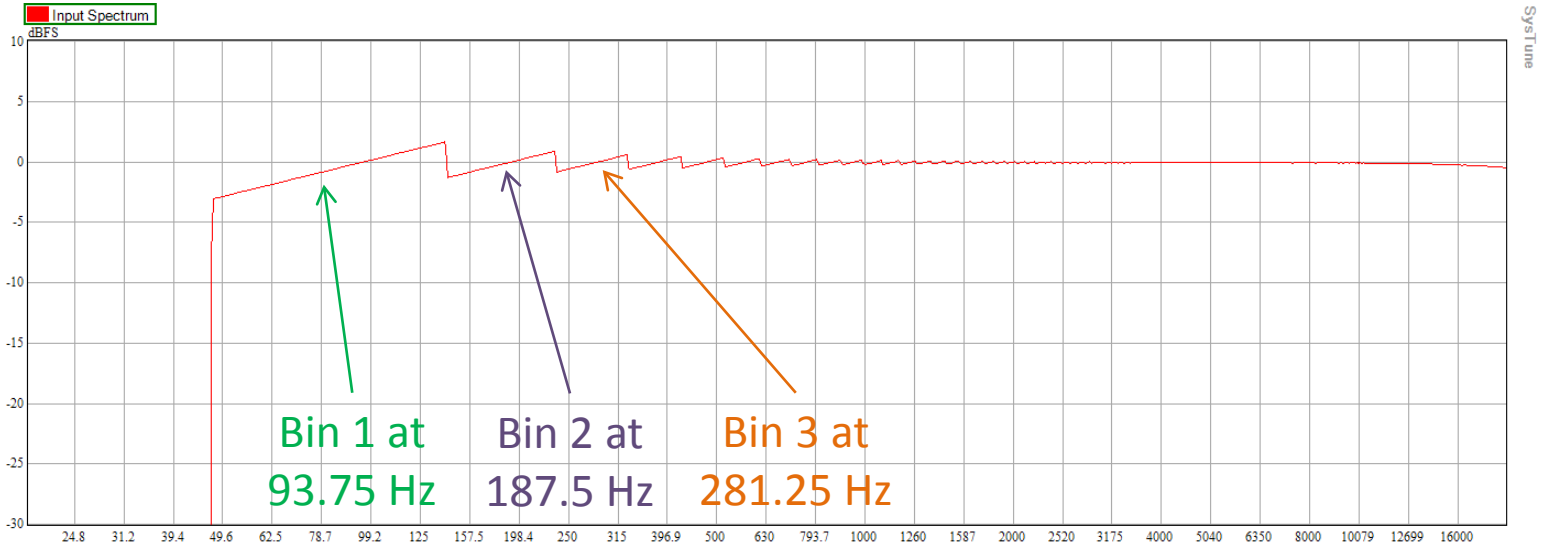
Each FFT bin is the same width as the frequency resolution. They are spaced linearly.

The energy can only go into a bin. It can’t go to frequencies between the bins.

Measurement System Fundamentals

FFT Bins – 93.75 Hz wide spaced every 93.75 Hz

48 kHz Sample Rate	512 Sample FFT Size	93.75 Hz Resolution



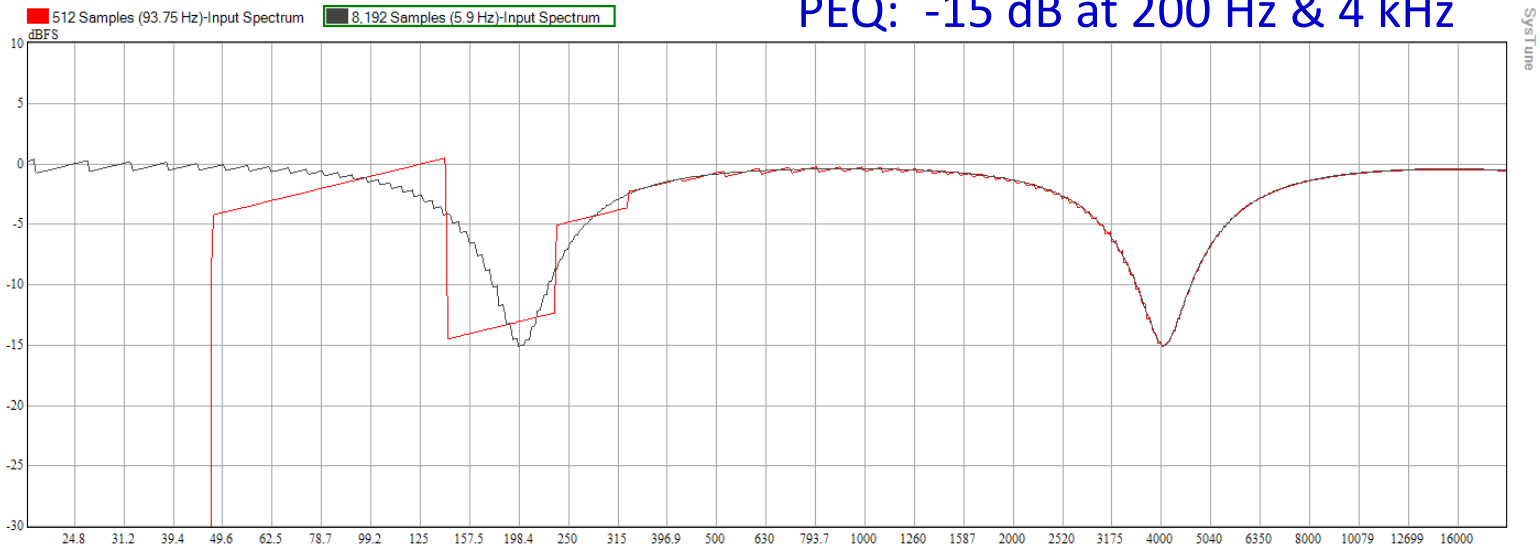
Measurement System Fundamentals

Single-Channel FFT

One signal is analyzed and its spectrum (frequency domain) is displayed.

The excitation signal is part of what is seen in the measurement results.

PEQ: -15 dB at 200 Hz & 4 kHz



Measurement System Fundamentals

Dual-Channel FFT

Two signals are analyzed:

Measurement channel and Reference channel

$$FFT\left[Deconvolution\left(\frac{Measurement}{Reference}\right)\right] \xrightarrow{\text{yields}} Transfer\ Function$$

(or Frequency Response)

Deconvolution in the time domain is the same as division in the frequency domain.

Because of the division (reference channel), the spectral content of the excitation signal is not seen in the measurement results.

Measurement System Fundamentals

Transfer Function (TF)

The transfer function is complex, meaning it contains both magnitude and phase.

The TF (frequency domain) and the Impulse Response, or IR, (time domain) are different views of exactly the same thing.

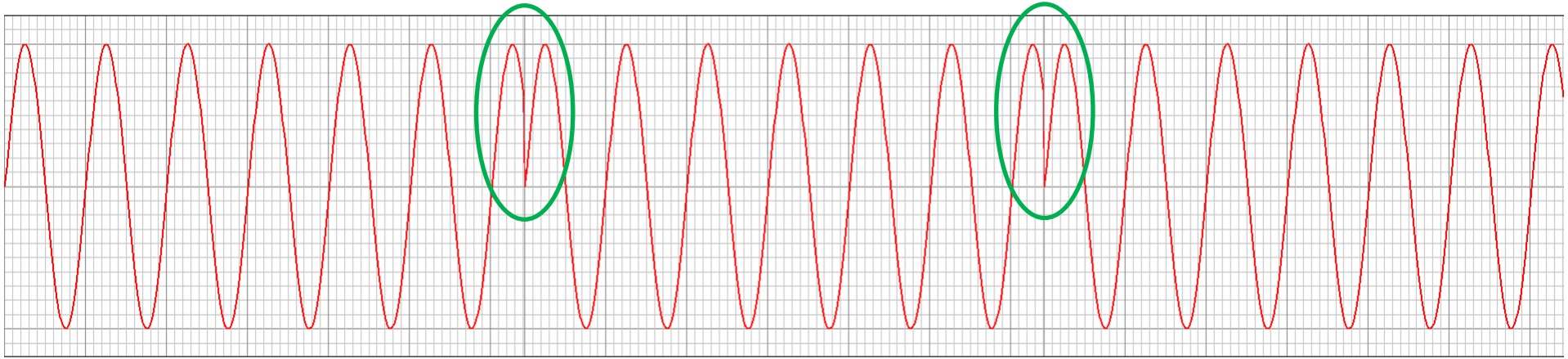
The frequency domain allows us to see ***what*** is happening.
The time domain allows us to see ***when*** it is happening.

Uncertainty Principle: $\Delta frequency = 1/\Delta time$

Measurement System Fundamentals

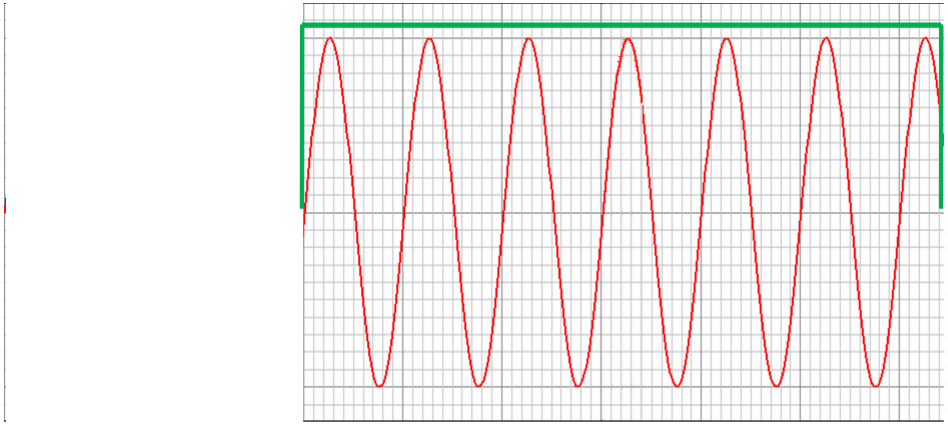
The Fourier Transform requires that the signal be infinite. For the FFT we assume the signal is infinite by “looping” it.

But there are consequences.



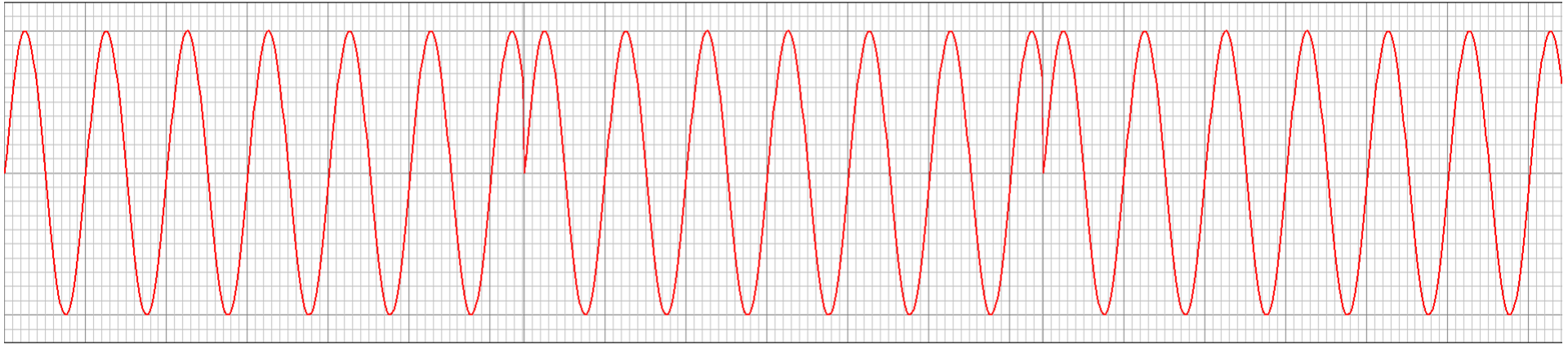
Measurement System Fundamentals

Just the act of taking an FFT block is applying a rectangular window to the data. This is a ***Signal Acquisition*** window.
(not an ***Impulse Response*** window)



Measurement System Fundamentals

This signal acquisition window can lead to our discontinuity problem.

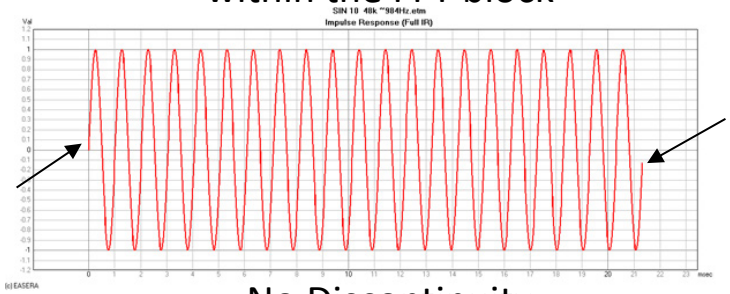


Measurement System Fundamentals

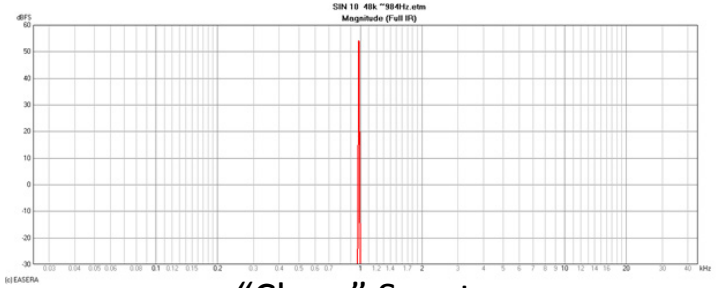
Example: 48kHz, 1024 Samples

997 Hz Sine

An integer number of periods ***does*** fit within the FFT block



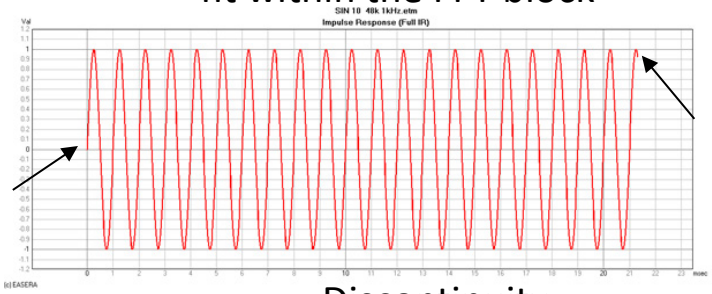
No Discontinuity



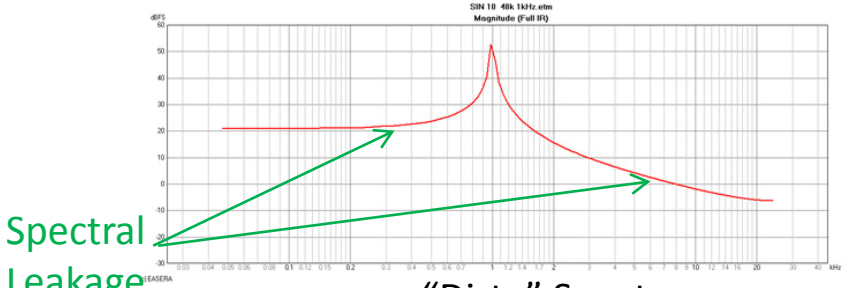
"Clean" Spectrum

1,000 Hz Sine

An integer number of periods ***does not*** fit within the FFT block



Discontinuity

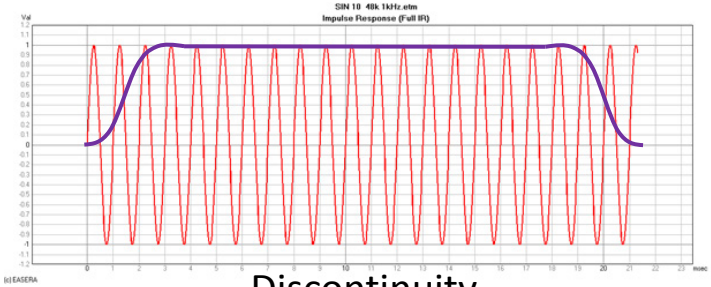


"Dirty" Spectrum

Measurement System Fundamentals

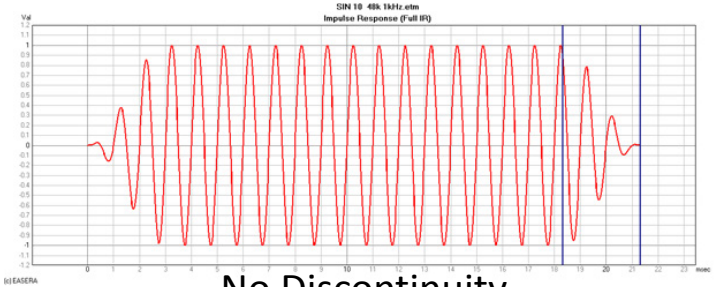
Example: 48kHz, 1024 Samples

1,000 Hz Sine



Discontinuity

Windowed 1,000 Hz Sine

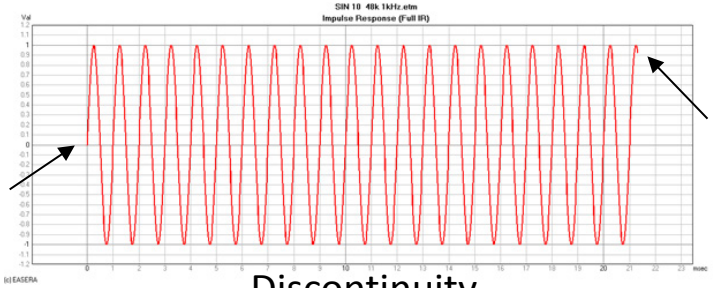


No Discontinuity

Measurement System Fundamentals

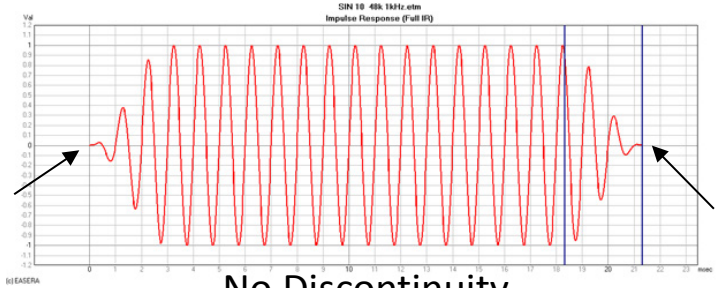
Example: 48kHz, 1024 Samples

1,000 Hz Sine

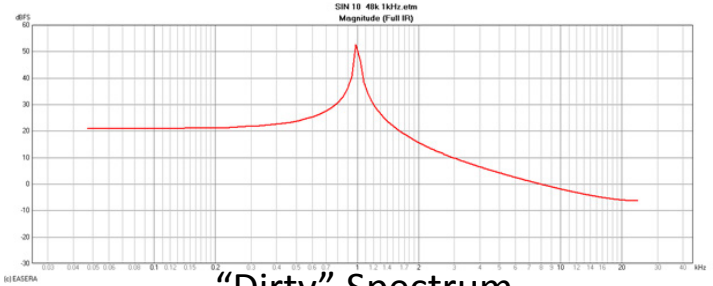


Discontinuity

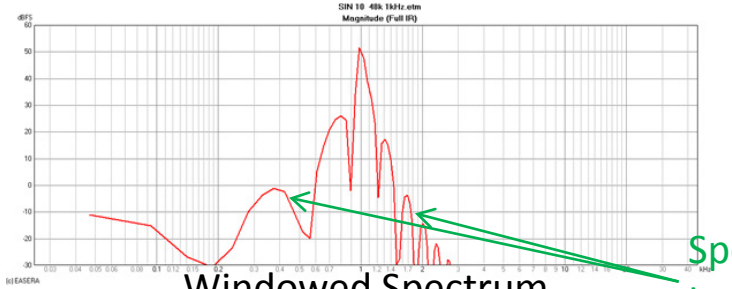
Windowed 1,000 Hz Sine



No Discontinuity



"Dirty" Spectrum



Windowed Spectrum

Spectral Leakage

Measurement System Fundamentals

Advantages of signal acquisition windowing

- Artifacts (spectral leakage) due to FFT block discontinuities can be reduced by using non-rectangular windows.

Disadvantages of signal acquisition windowing

- Windowing can't completely eliminate spectral leakage (side lobes).
- Information is reduced or lost due to the window.

One Solution – Matched Length Excitation & Analysis

- Using the same FFT size for generating the excitation signal and the analysis block can greatly reduce or eliminate these problems.

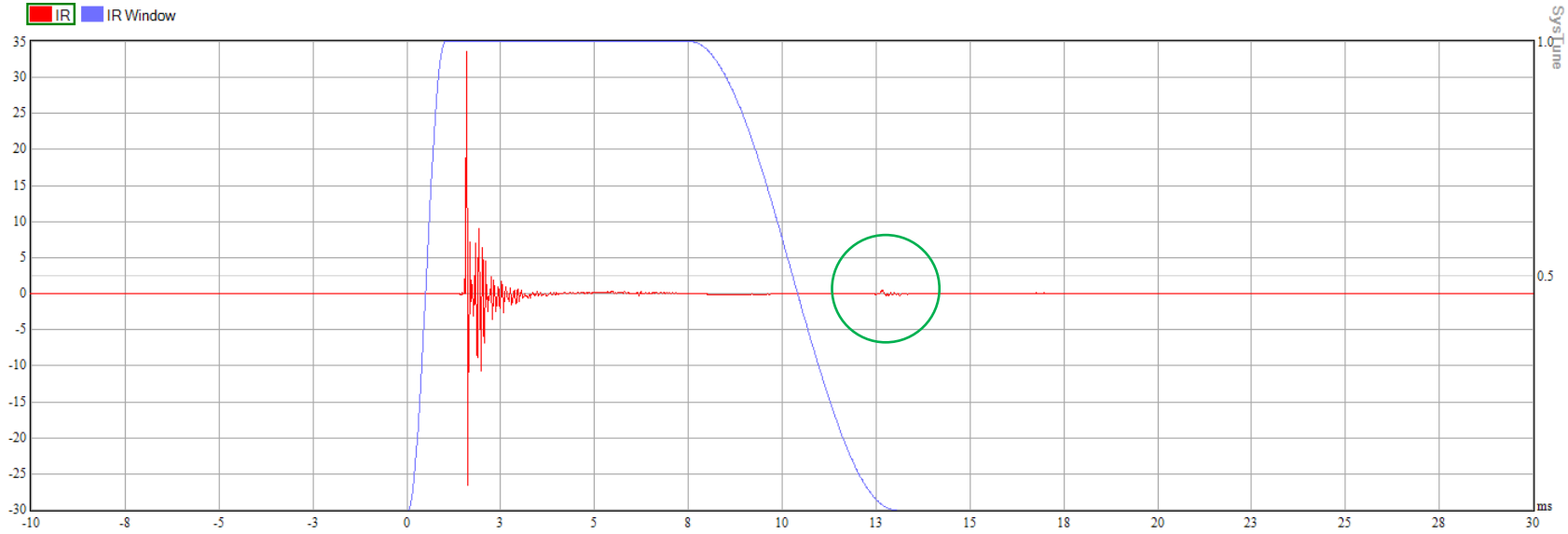
Measurement System Fundamentals

Using the same FFT size for generating the excitation signal and the analysis block can greatly reduce or eliminate these problems.

Demo with SysTune

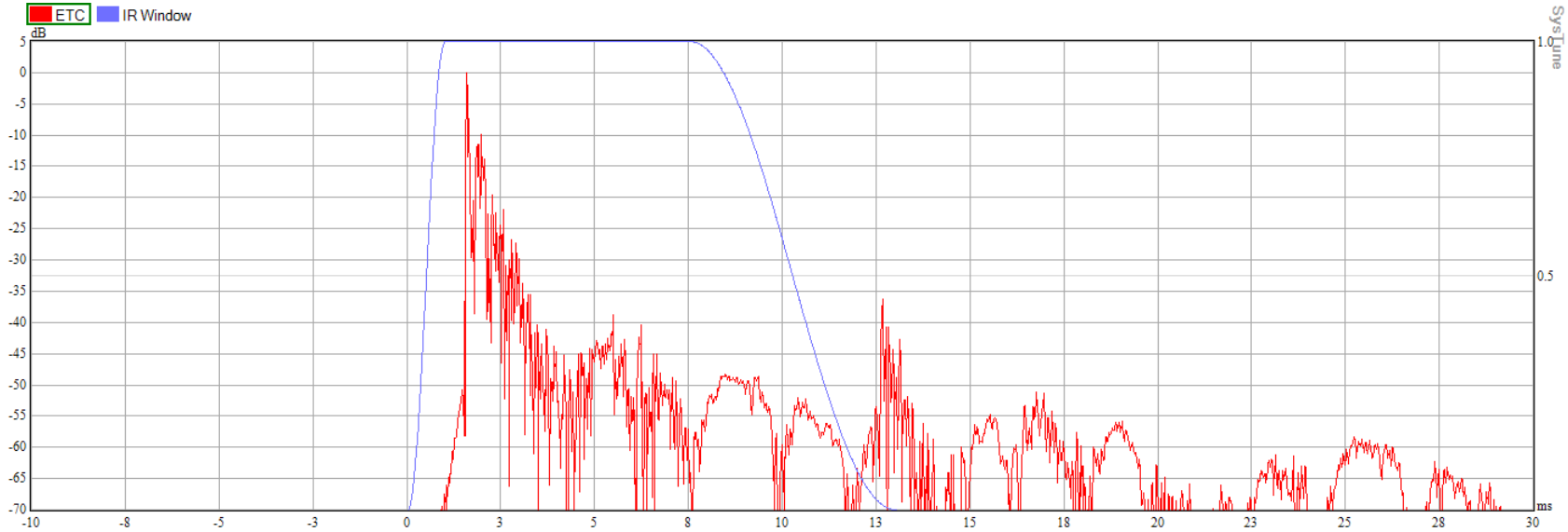
Impulse Response Windowing

An ***IR*** window is different than a ***Signal Acquisition*** window



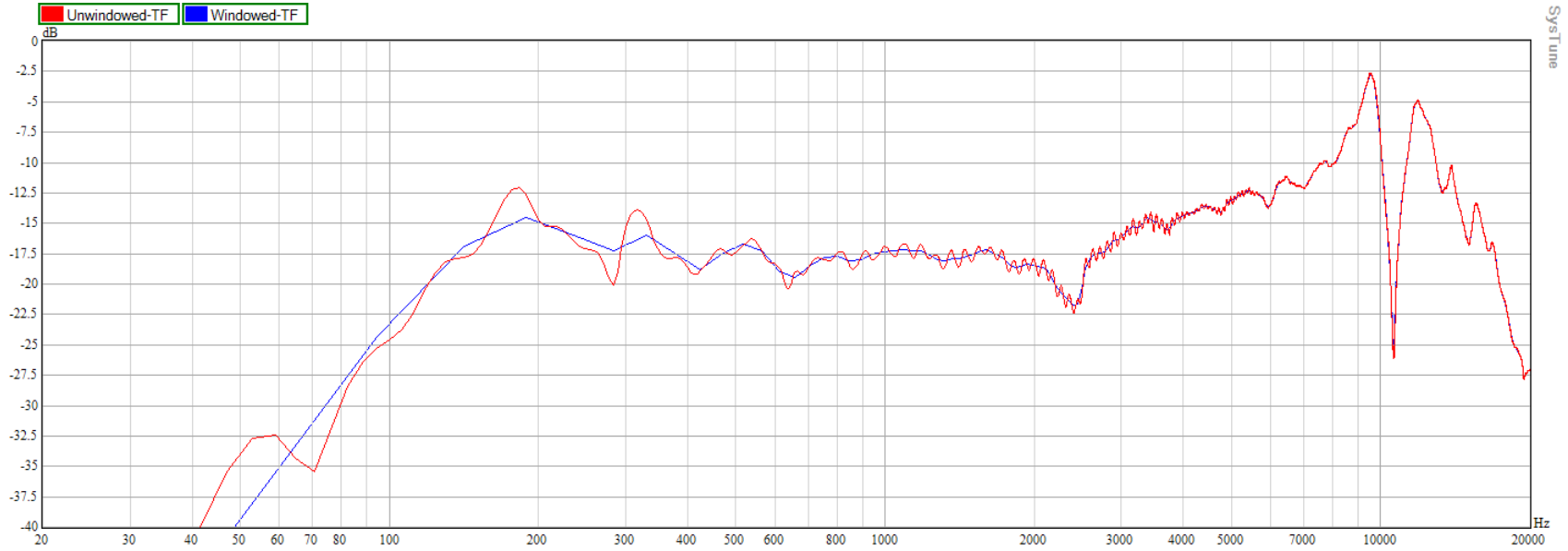
Impulse Response Windowing

An ***IR*** window is different than a ***Signal Acquisition*** window



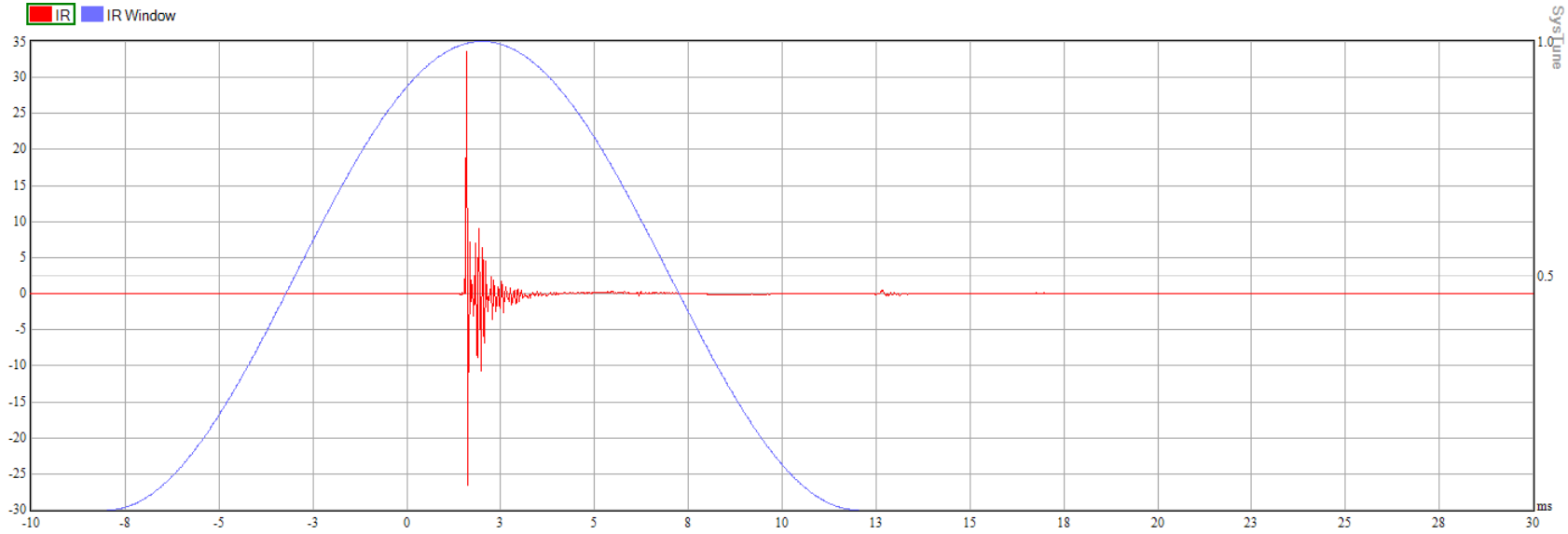
Impulse Response Windowing

Windowed and unwindowed transfer function



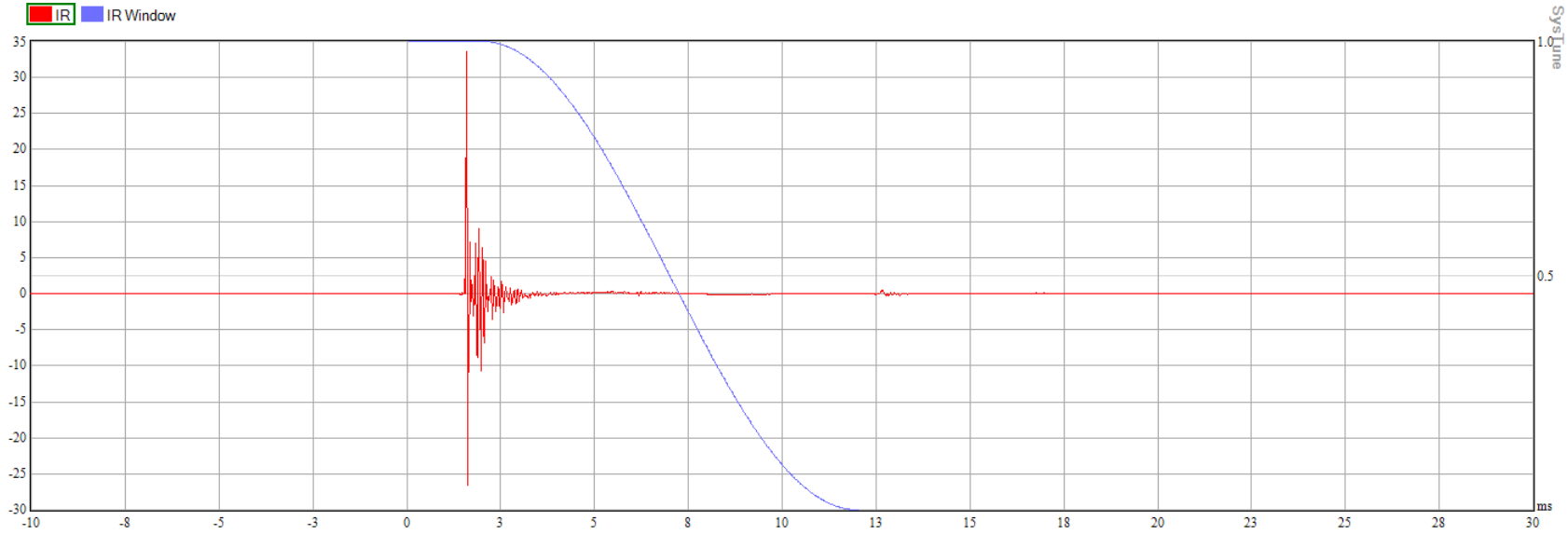
Impulse Response Windowing

Symmetrical Windows compared to Half-Windows



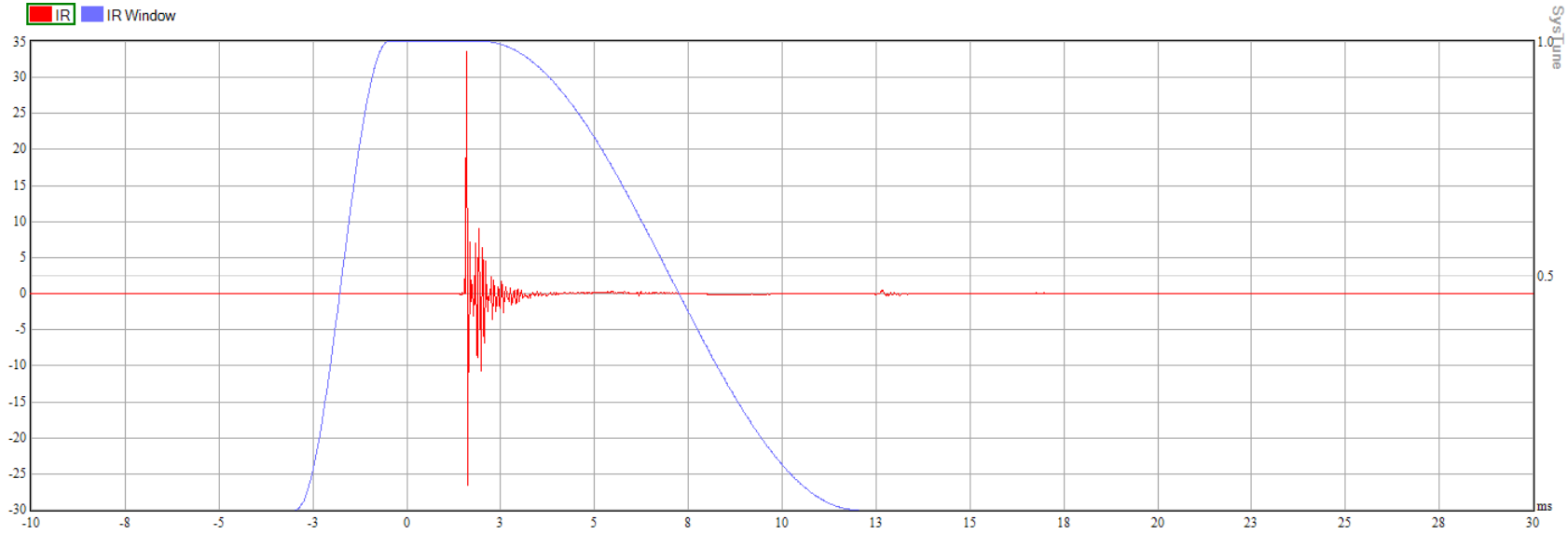
Impulse Response Windowing

Symmetrical Windows compared to Half-Windows



Impulse Response Windowing

Asymmetrical Left and Right Half-Windows

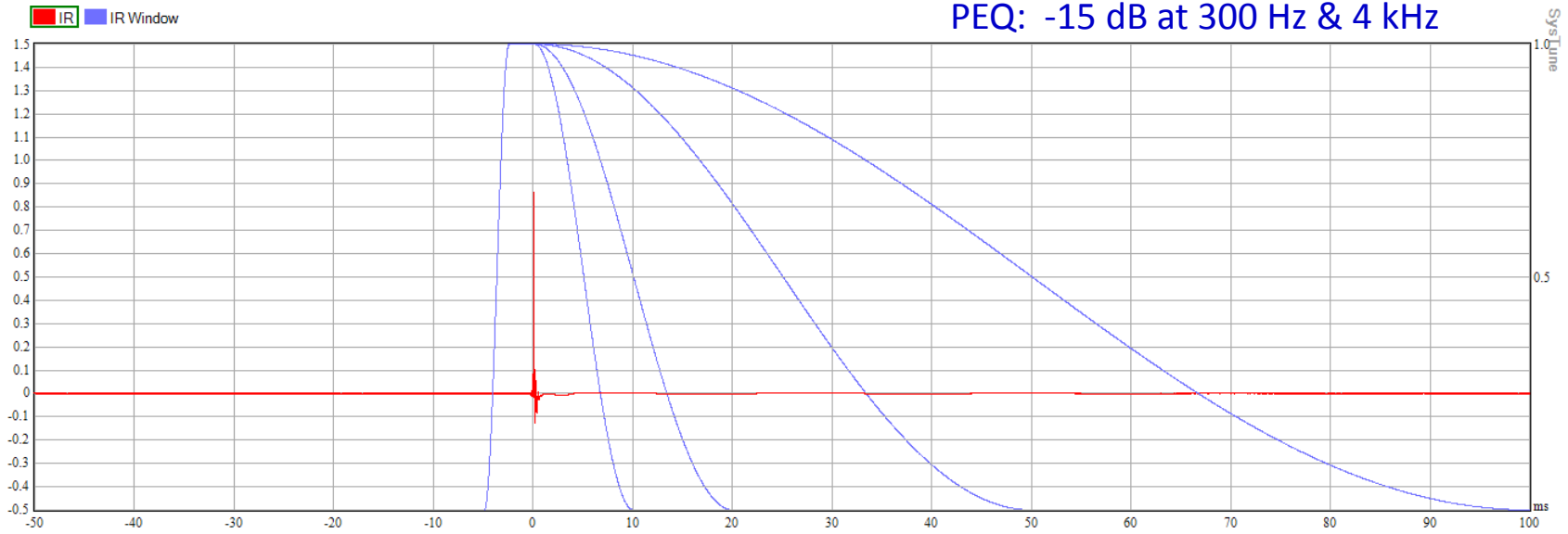


Impulse Response Windowing

Window length (time) affects frequency resolution

HP: 50 Hz, 24 dB/oct.

PEQ: -15 dB at 300 Hz & 4 kHz



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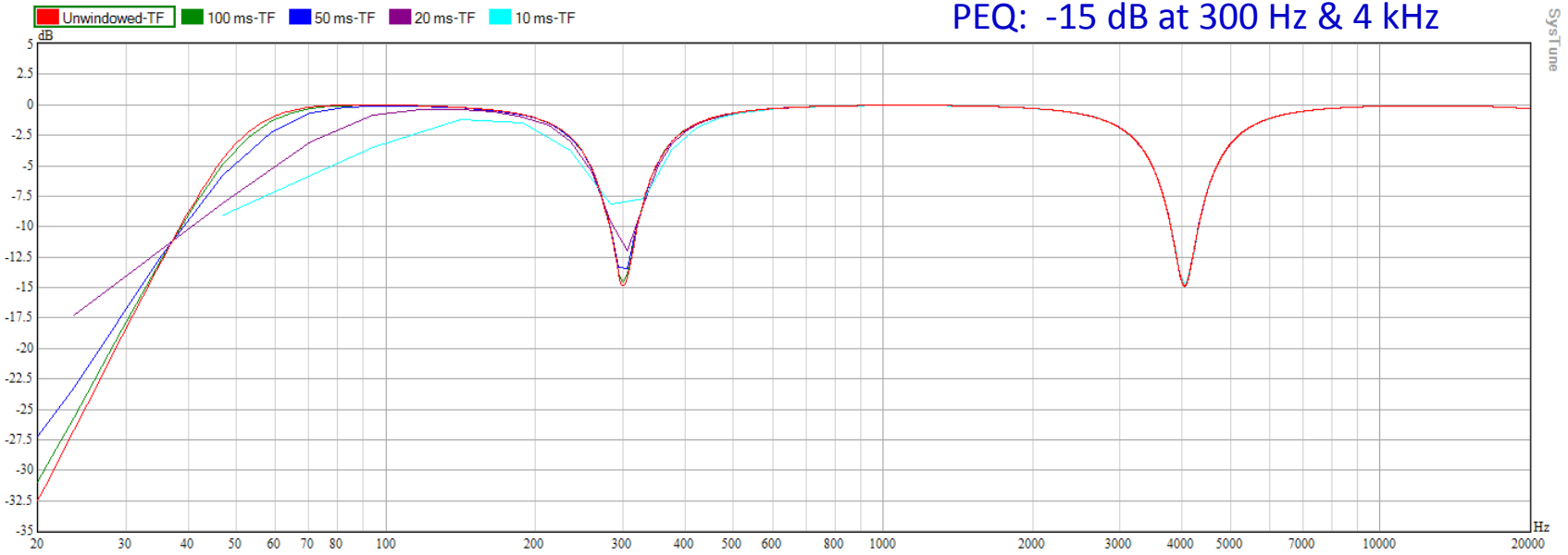

Impulse Response Windowing

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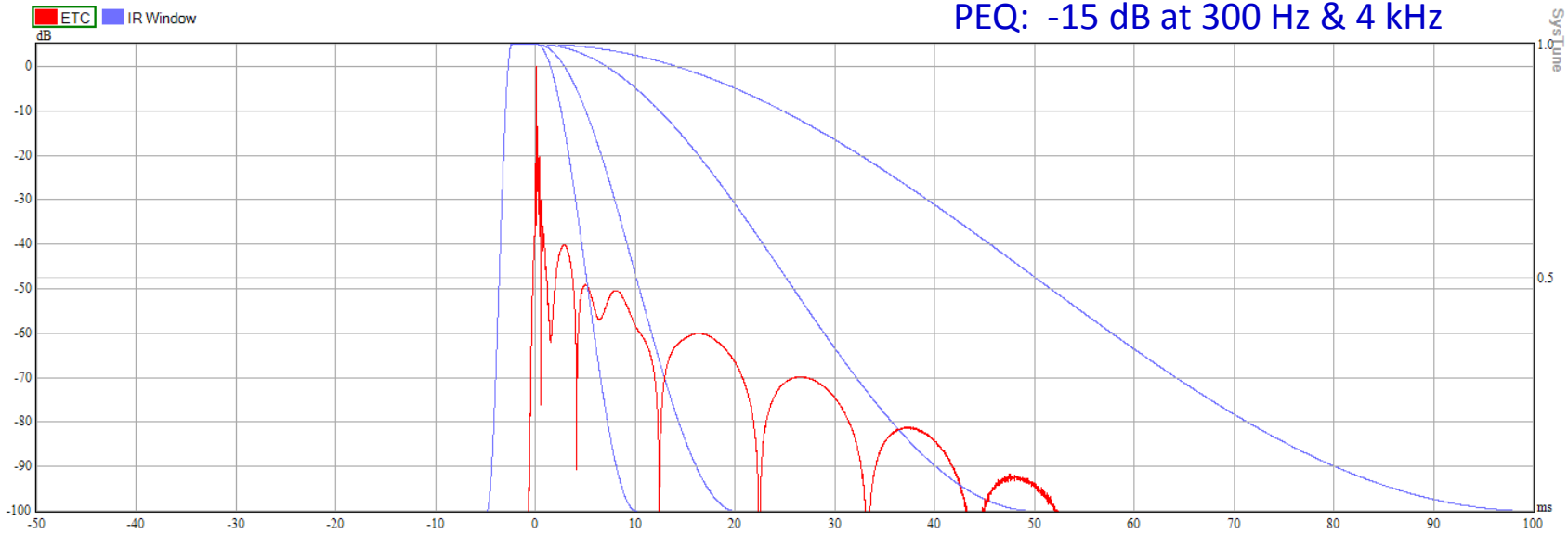


Coherence

Popular Meaning

High coherence tells us the measurement data is good and can be used for tuning the loudspeaker system.
Low coherence means the measurement data is questionable and shouldn't be used for tuning / EQ decisions.

Mathematical Meaning

High coherence means that at a given frequency there is a constant phase relationship between two sources. In a dual-channel FFT these sources are the reference channel and the measurement channel.

Coherence

- Not all analyzers are the same. There can be differences in the way analyzers internally process signals.
- The choice of excitation signal length and FFT size can also affect the measurement results.
- Both of these can influence the coherence values of a measurement.

Two different analyzers can measure the exact same loudspeaker, with the exact same microphone, in the exact same location, and get dramatically different coherence values.

Both analyzers are correct!

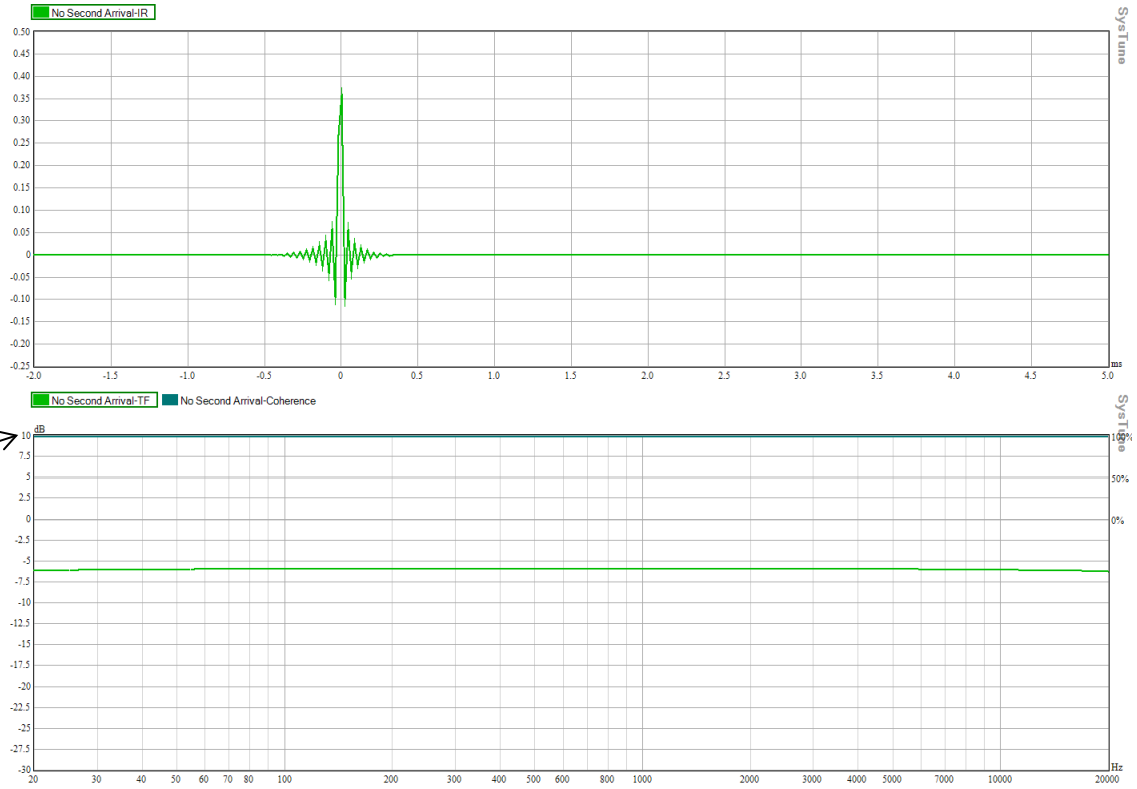
Coherence is not strictly a function of the device being measured.

Coherence

Example:

Matched length excitation signal and FFT analysis block.

No signal acquisition window (rectangular).



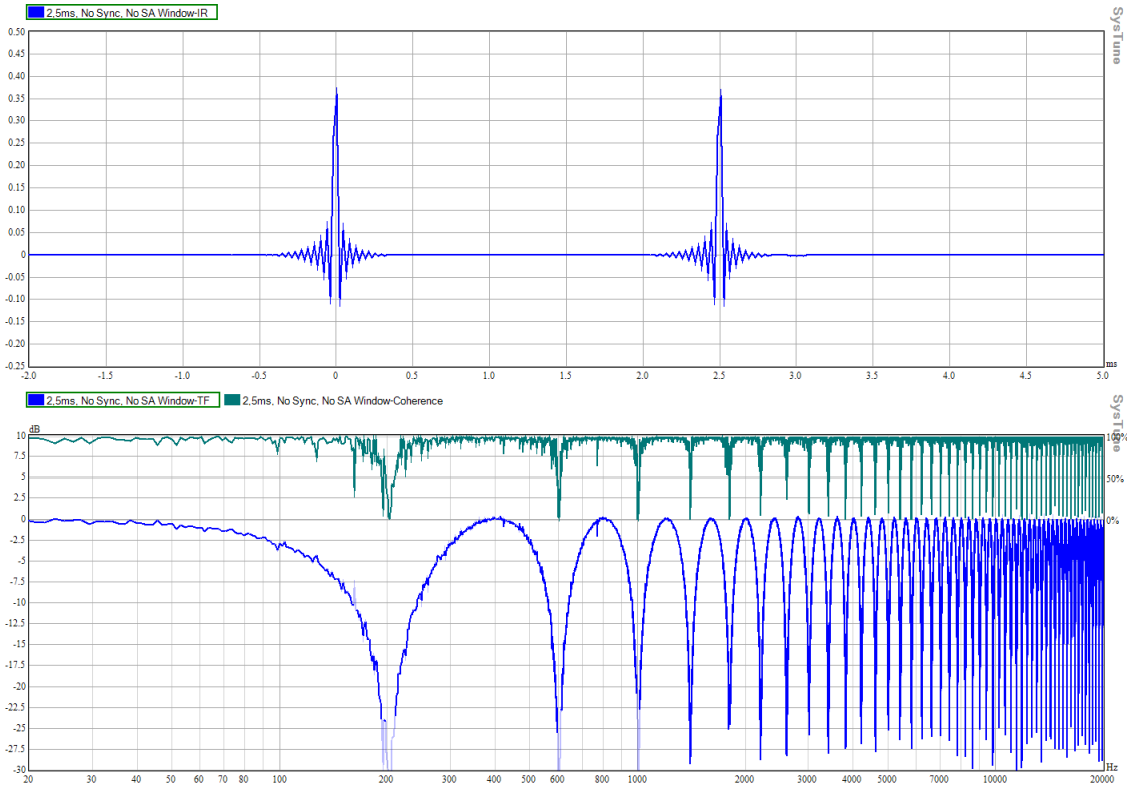
Note coherence value of 1.0 at all frequencies.

Coherence

Example:

Excitation signal length is not matched with the FFT analysis block.
No signal acquisition window (rectangular).

The excitation signal contains frequencies the FFT size can't place into a bin. This results in spectral leakage and a reduction in coherence at some frequencies.

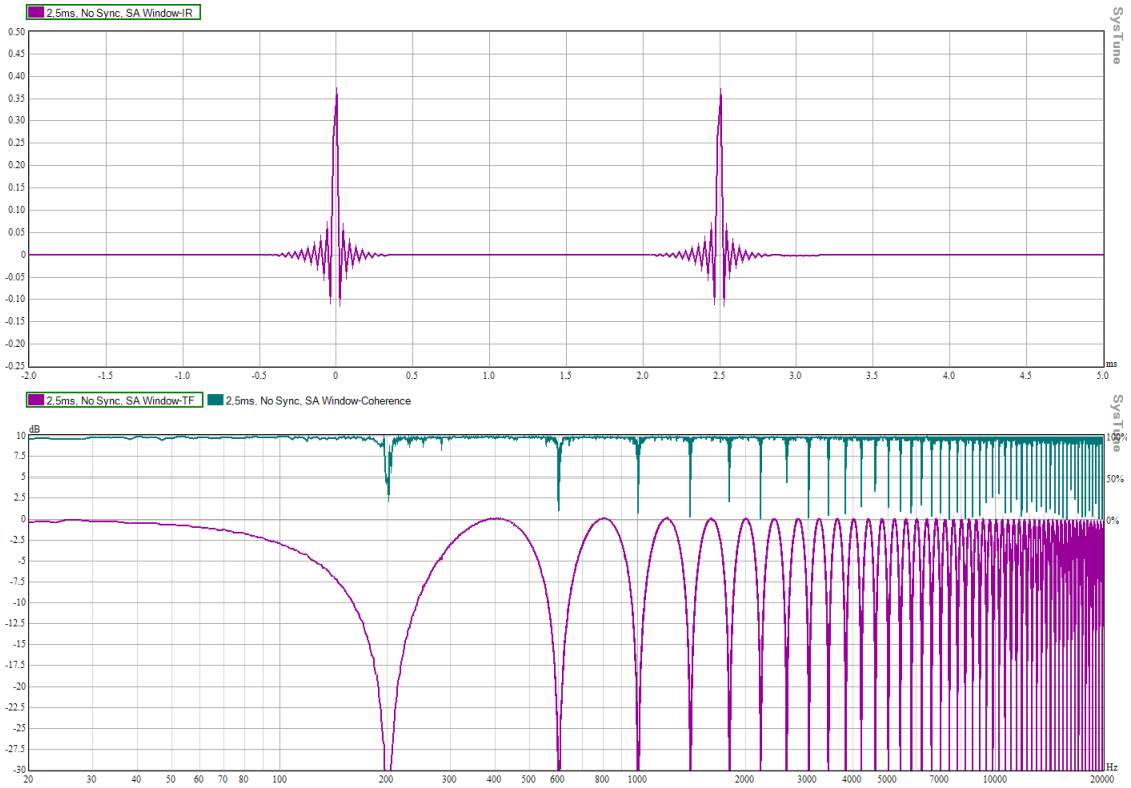


Coherence

Example:

Excitation signal length is not matched with the FFT analysis block.
With signal acquisition window (tapered at ends).

There is some improvement (reduction) in the spectral leakage. This results in an increase in the coherence values at some frequencies.

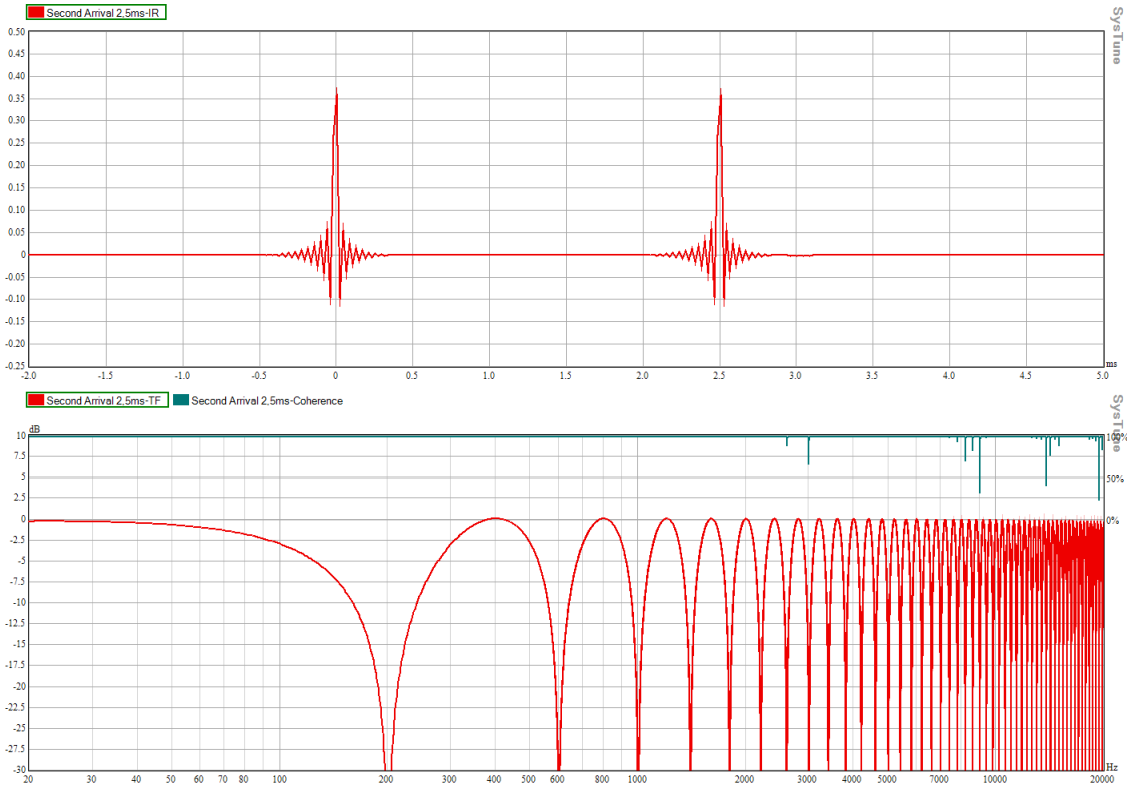


Coherence

Example:

Matched length excitation signal and FFT analysis block.
No signal acquisition window.

There is no the spectral leakage. This results in perfect coherence values of 1.0 at almost all frequencies.



Section 5

How to Make Good Measurements

Charlie Hughes

How To Make Good Measurements

Things to always keep in mind

- Why are we measuring?
- What are we measuring?
  - What do we think we're measuring?
  - What do we want to measure?
- Is it audible?
- Does it matter?

How To Make Good Measurements

Why are we measuring?

- What do we hope to accomplish?
- How will the data be used?
- Are we troubleshooting to find the cause of a problem?
- Are we documenting performance?

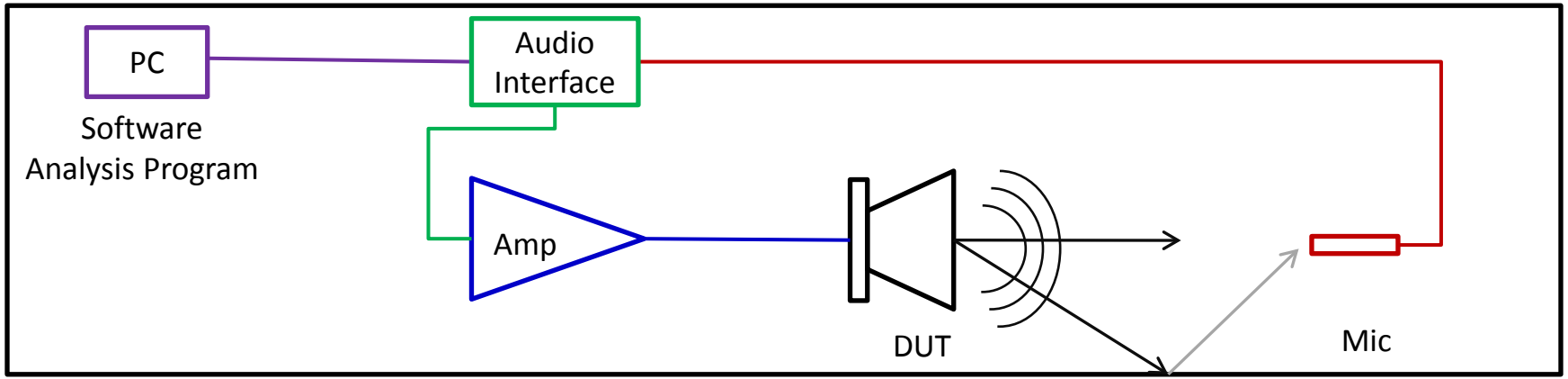
The answers will help to determine:

- Type of analyzer
- Test signal
- Resolution requirements (1/n octave)
- Calibrated or not

How To Make Good Measurements

What are we measuring?

- Mentally separate the Analyzer from the Device Under Test (DUT)
- Examine the signal flow
- Everything not the Analyzer is included in the measurement even though it may not be part of the intended DUT



How To Make Good Measurements

Do we need to modify the measurement set-up to address things that are not actually part of our intended DUT?

Compensate for, or exclude, unwanted effects

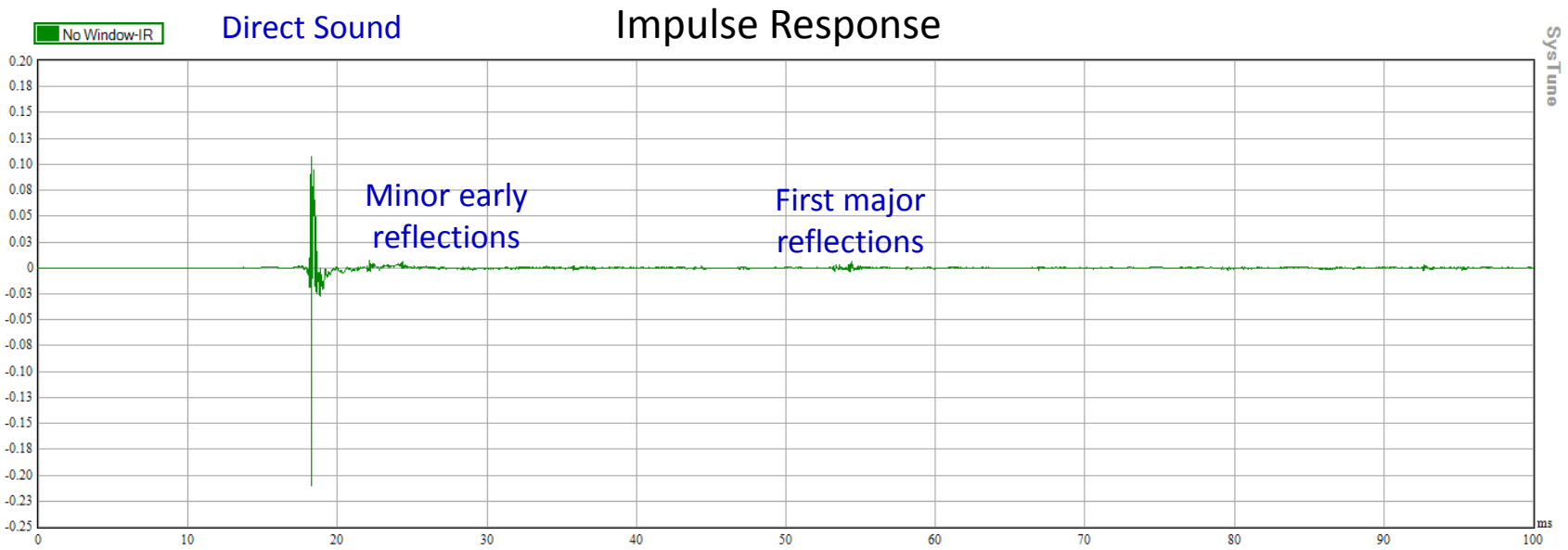
- Non-flat microphone response
  - Microphone correction data (typically on-axis response only)
- Reflections
  - Can reflections be windowed out or can the DUT be moved?
- Noise
  - Can we increase the SPL of excitation signal or can we modify the measurement parameters for increased S/N?

Is it audible? ***Does it matter?***

How To Make Good Measurements

Large space

The first reflections occur long after the arrival of the direct sound



How To Make Good Measurements

Large space

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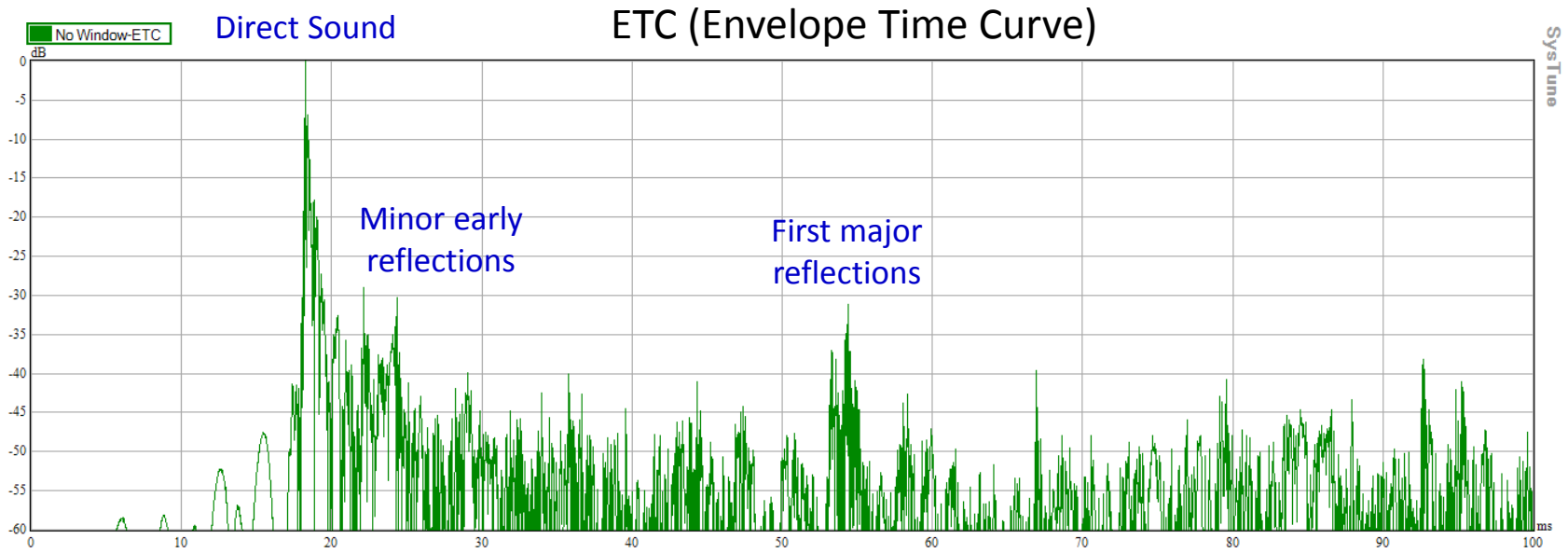
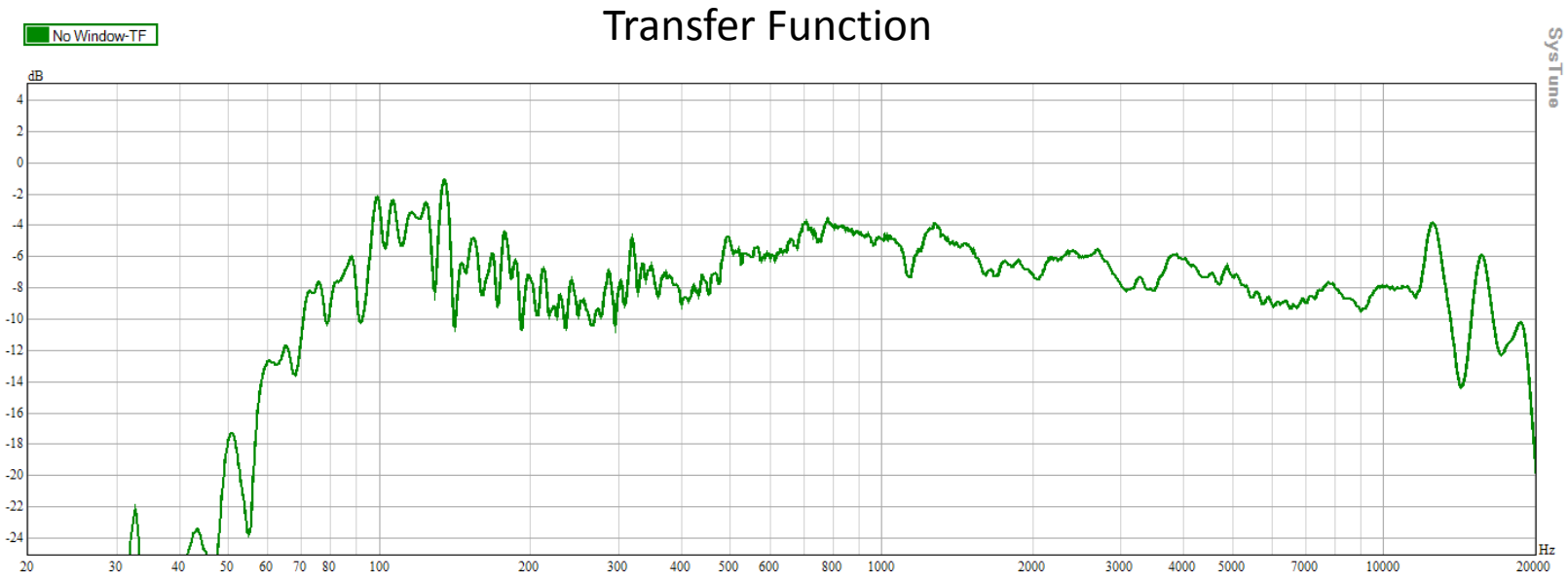


How To Make Good Measurements

Large space

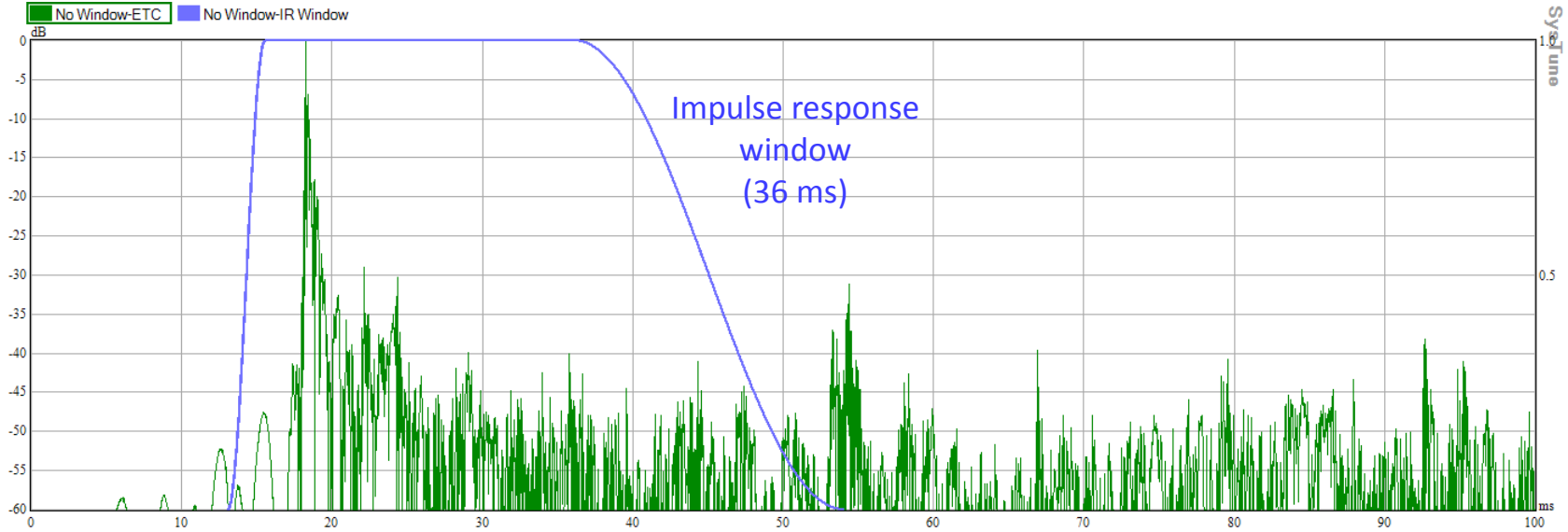
The transfer function (frequency response) of the measurement with the reflections included.



How To Make Good Measurements

Large space

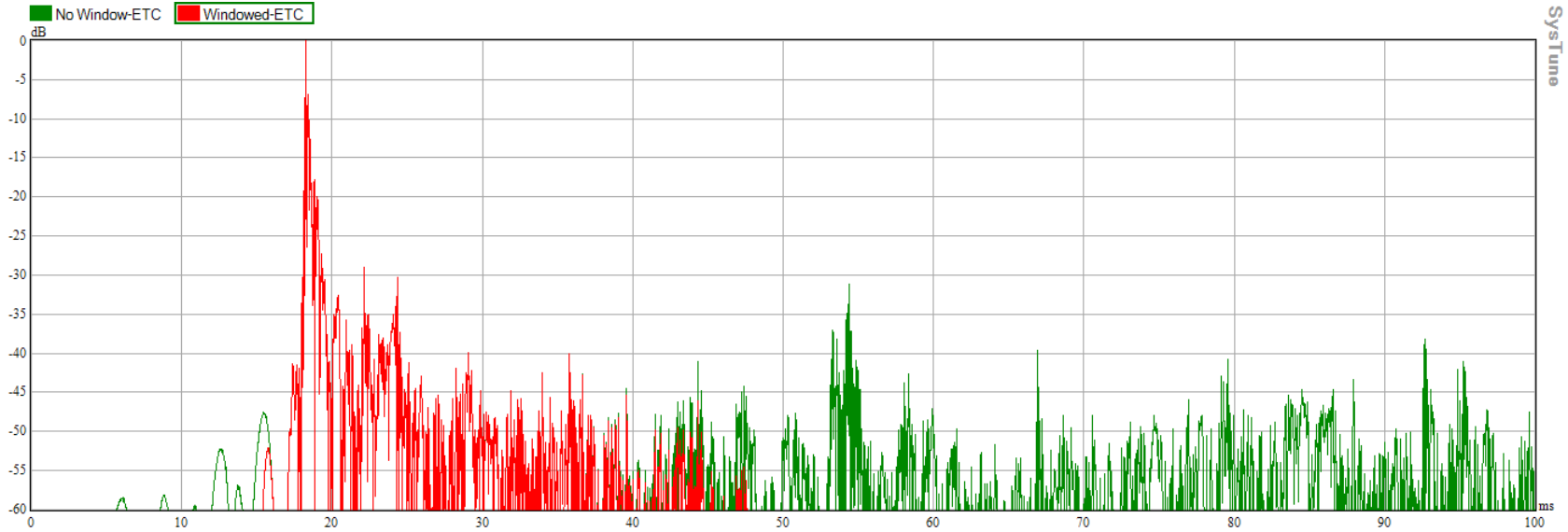
The IR window overlaid on the ETC



How To Make Good Measurements

Large space

ETC data with and without IR window applied



How To Make Good Measurements

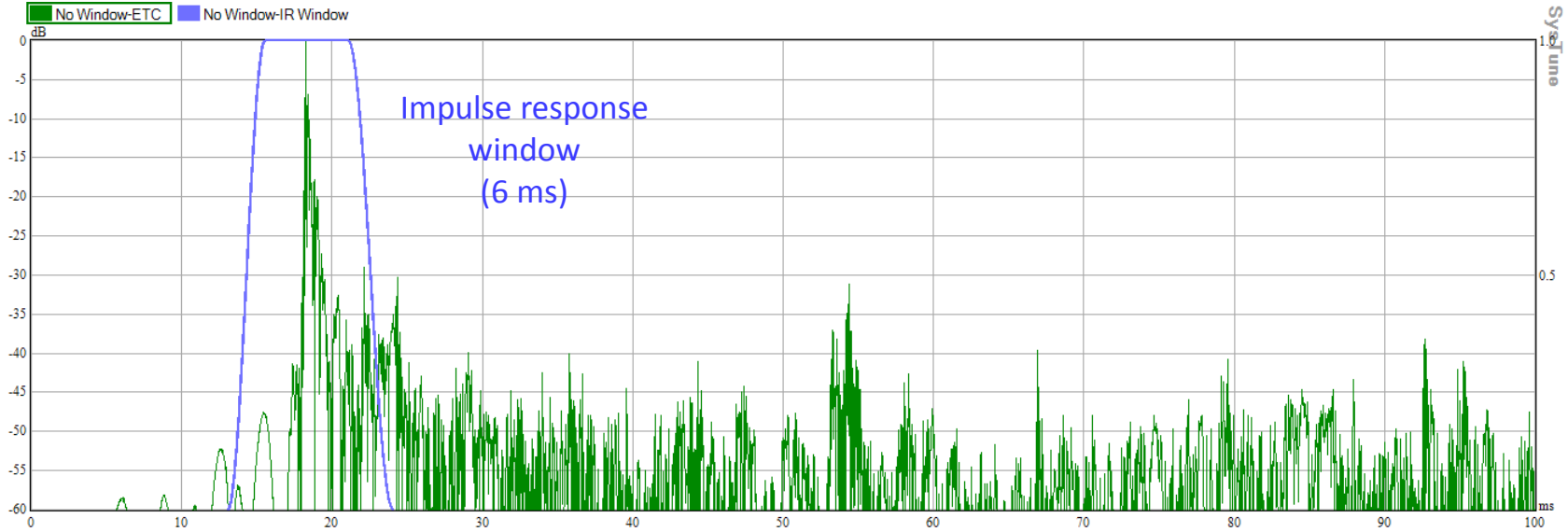
Large space

Transfer function with and without IR window applied

How To Make Good Measurements

Windowing the earlier reflections

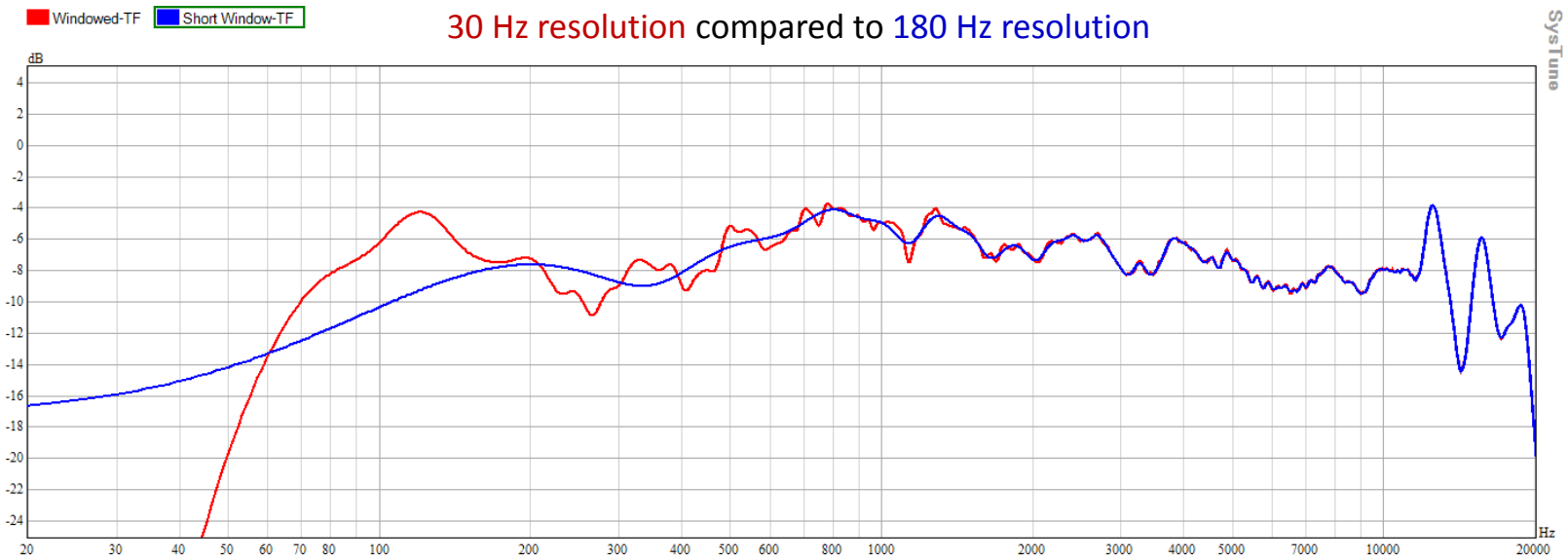
The IR window overlaid on the ETC



How To Make Good Measurements

Windowing the earlier reflections

Isolation from earlier reflections results in loss of frequency resolution



How To Make Good Measurements

Relatively quite space

We typically want a signal to noise ratio (S/N) of at least 20 dB. This helps insure we have good data. Having a S/N of 40 dB or more would be better, if possible.

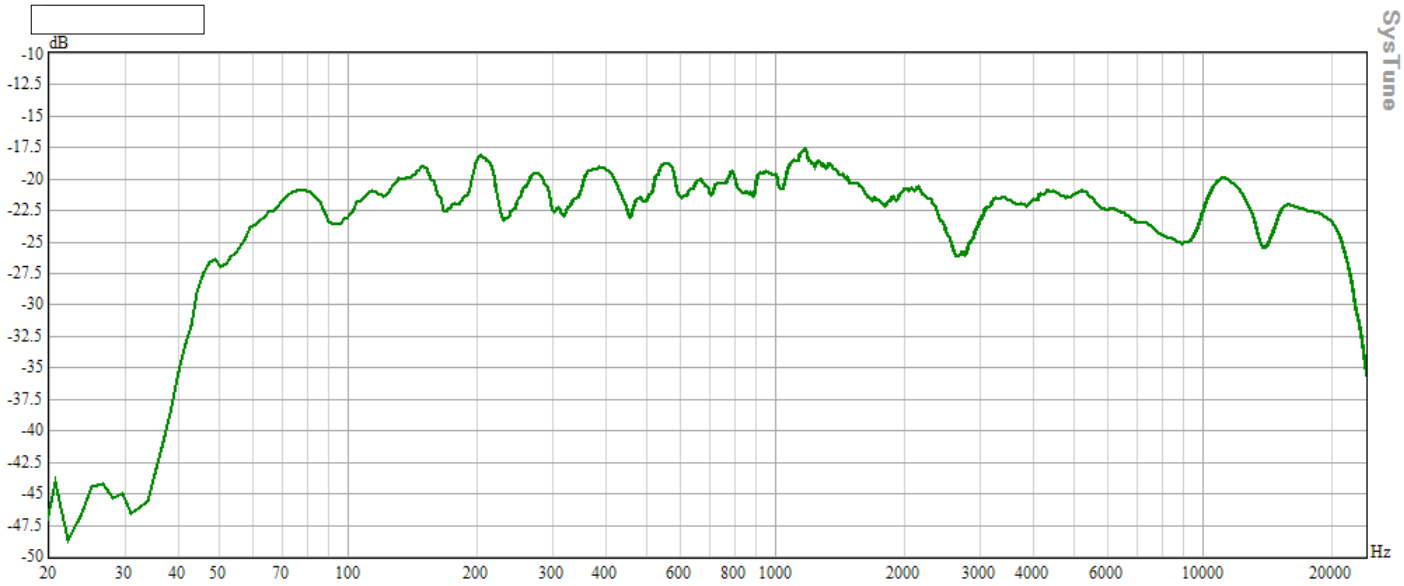
Noise is frequency dependent. Measure the noise floor in the room before doing a lot of other measurements to make sure there is at least 20 dB S/N at all frequencies of interest.

Don't drive the system beyond its linear operating range during measurements!

How To Make Good Measurements

Example: Is this a good response for a loudspeaker system?

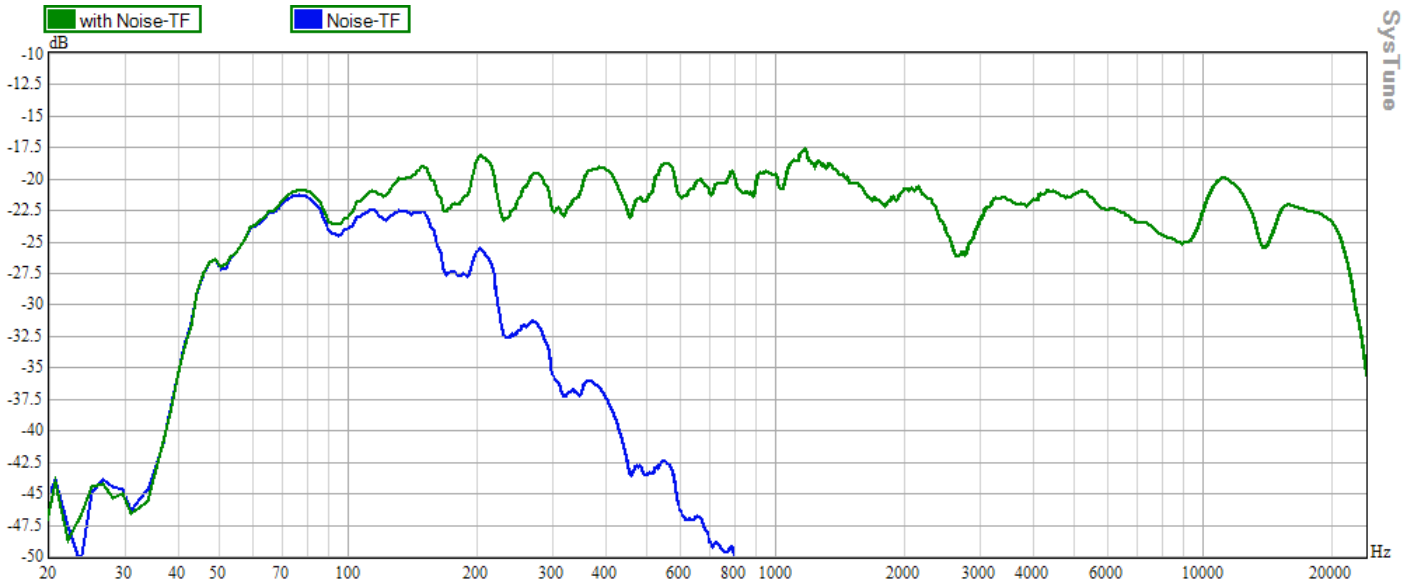
Two-way loudspeaker with an 8 inch woofer & an HF horn



How To Make Good Measurements

Example: Is this a good response for a loudspeaker system?

Do we know the spectrum & SPL of the noise in our room?

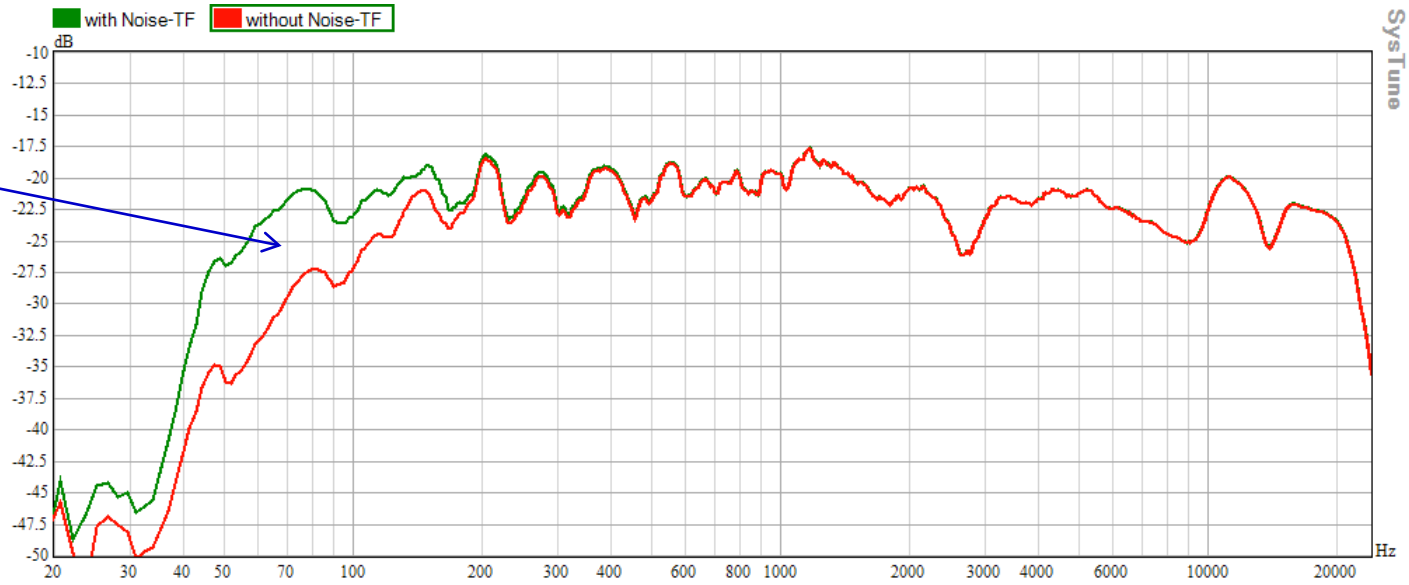


How To Make Good Measurements

Example: Is this a good response for a loudspeaker system?

Noise removed from the measurement via averaging or longer sweep

We may need a subwoofer



How To Make Good Measurements

Excitation Signal Types

Digitally generated pink noise can have a crest factor of about 12 dB.
Random pink noise can have a crest factor exceeding 20 dB.
Music can have a typical crest factor of 12 dB to 30 dB depending on type.

For a target SPL (think rms level) each of these will have a peak level at least 12 dB higher than the rms level!

A swept-sine signal has a 3 dB crest factor. This will cause less excursion of the loudspeaker drivers for the same SPL as a higher crest factor excitation signal.

Section 6

Ways to Measure and EQ a Loudspeaker Array

John Murray

In the Shop Measurement & EQ

- The following pictures are courtesy of Deward Timothy of Poll Sound
- He is probably the best in the industry at physical equalization

High-Bay Movable Shop Crane

Diffractional Effects

Lower Frequency Mutual Coupling

Proper Splay Angling

Configuring a 6-Box Array

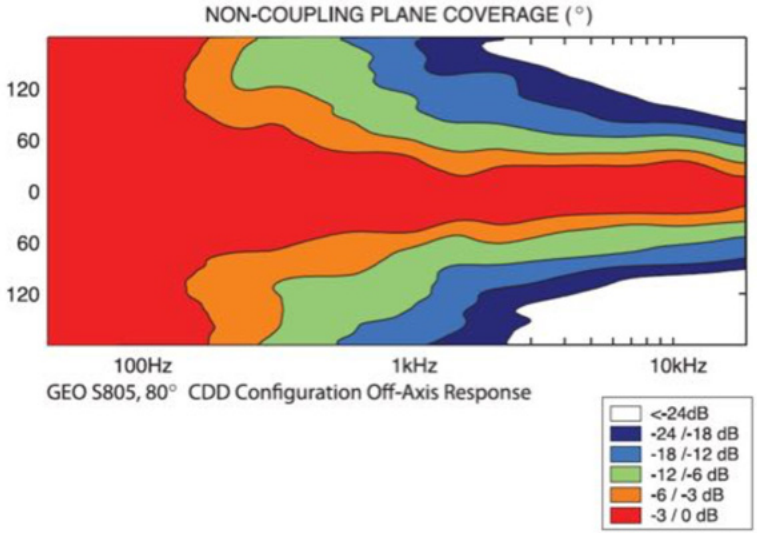
Point Of Geometric Origin (POGO) Laser

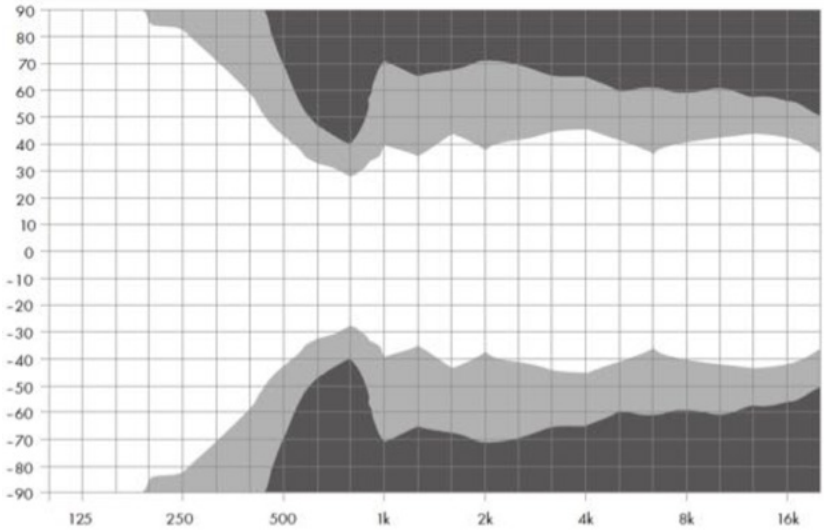
Measurement & EQ On-Site After Installation

On-Site Measurement & Equalization Issues

- Other than doing what can be done to isolate the direct sound, which cannot be done at longer wavelengths in rooms, the other major issue when performing on-site measurement and equalization is whether to use a single microphone position or multiple power-averaged positions
- Generally, a single source does not require averaging multiple microphone positions to resolve its response, however when the on-axis and off-axis response is not the same, an averaged response can be useful for equalization

Averaging On- and Off-Axis Single Box Response



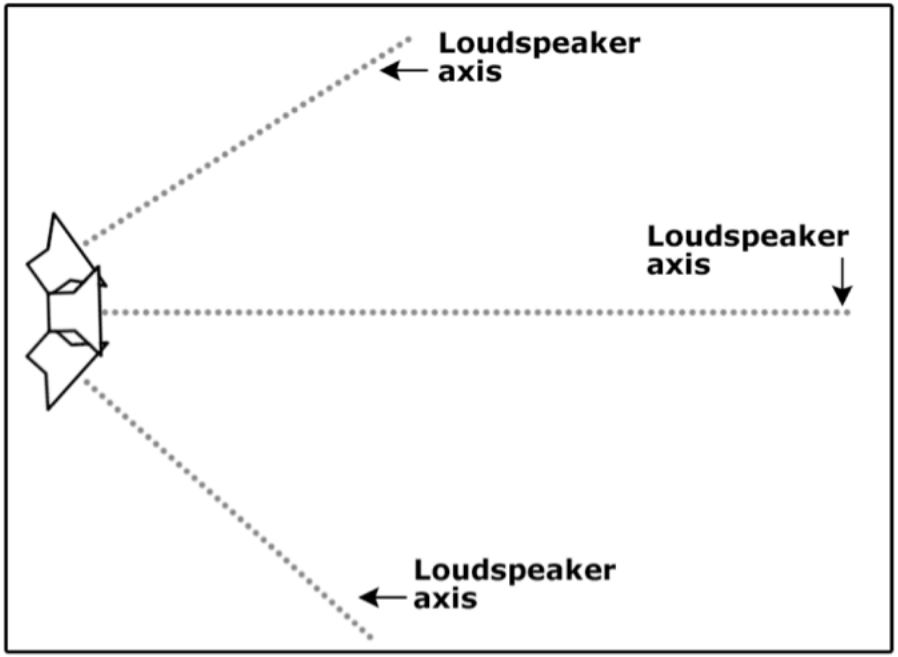


- As can be seen in these typical examples, the response at the horizontal -6 dB down-points (+/- 40° here) are only 3-6 dB lower than the on-axis response @ 10 kHz

Averaging On- and Off-Axis Single Box Response

- If on-site time allows, averaging on-axis and off-axis measurements of a single speaker system, or single line-array can be done, though it would only be 2-3 dB of HF shelving boost for well-behaved speakers
- However, if the off-axis position is at the -6dB down-point, another speaker should be providing overlapping coverage from that point on, and would add level to off-set the off-axis HF droop, no longer requiring EQ averaging
- So, if time permits, averaging single-box on- and off-axis horizontal response can be employed, but it is not crucial

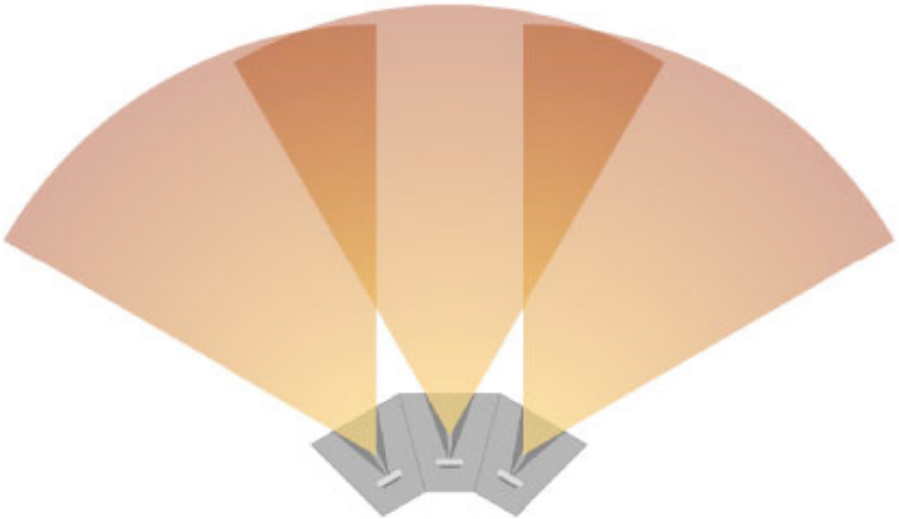
Averaging Multi-Box Arrays



from the InfoComm Spectral Balance Standard draft

- If a non-line array is driven by a single DSP channel, then power averaging their on-axis responses would be the most efficient way to equalize the array
- If each box is separately driven and bi- or tri-amped, then array shading can be considered

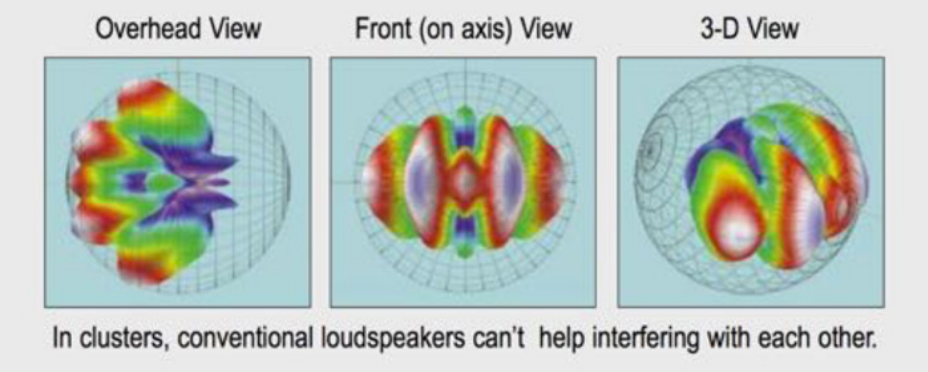
Non-Line-Array Multi-Box Shading

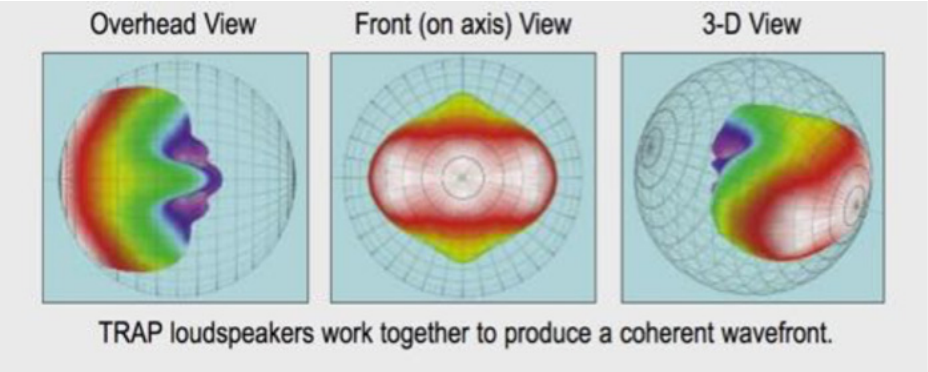


courtesy of www.gtaust.com

- Moderately sized pro speakers are physically incapable of uniform dispersion across their entire bandwidth
- Therefore, lobing overlap at some frequencies is unavoidable and this is where array shading enters the picture

Non-Line-Array Multi-Box Shading

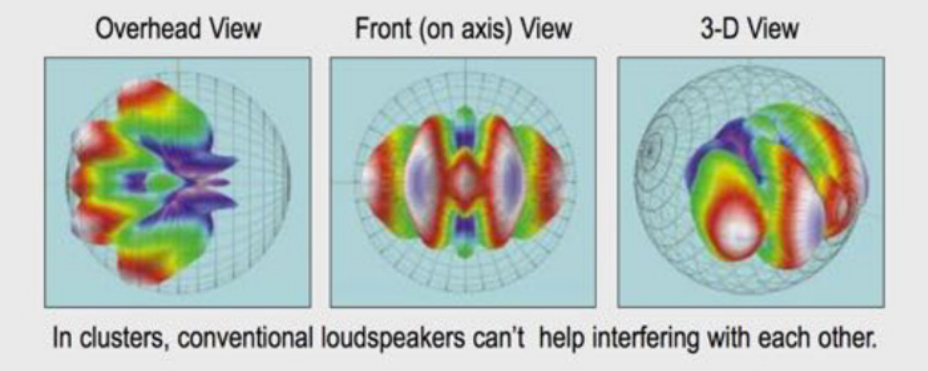


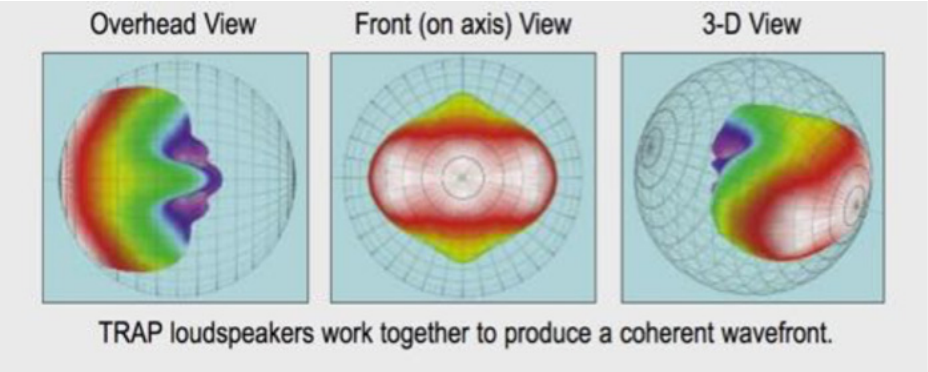


courtesy of Renkus-Heinz

- Array shading can incorporate dissimilar crossover points, HP/LP/AP filters, and delay
- Be cautious about array shading on-site as unintended lobe steering can aim the sound in unexpected directions

Non-Line-Array Multi-Box Shading





courtesy of Renkus-Heinz

- It's best to contact the manufacturer for array shading parameters
- If you are going to roll your own array shading, be sure to take the time to set up the parameters in your shop first to confirm not only the frequency response, but to spot check polars as well

Section 7

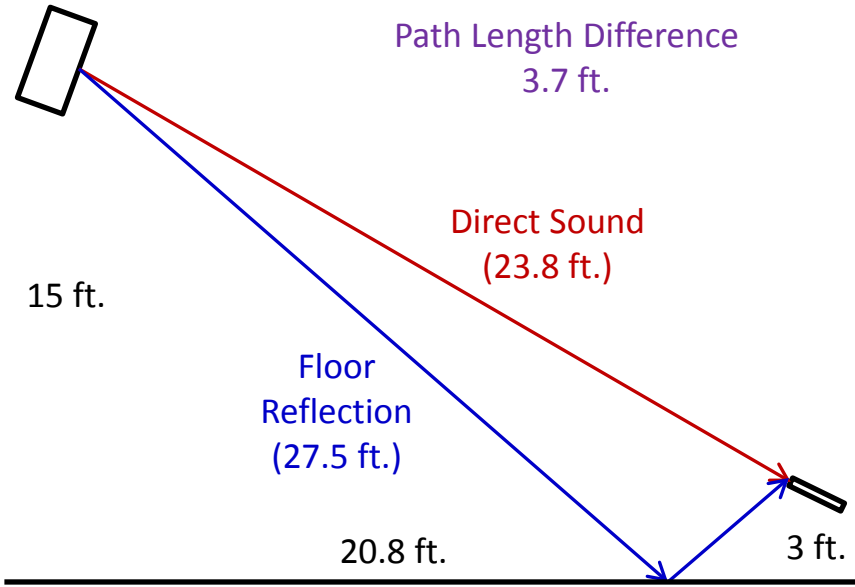
Measurement Compromises

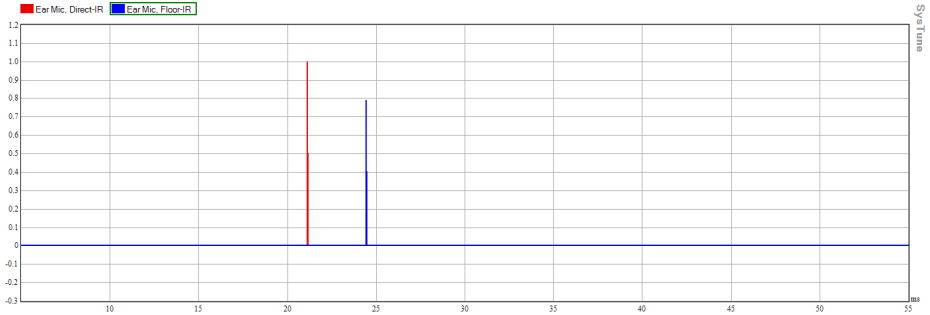
Charlie Hughes

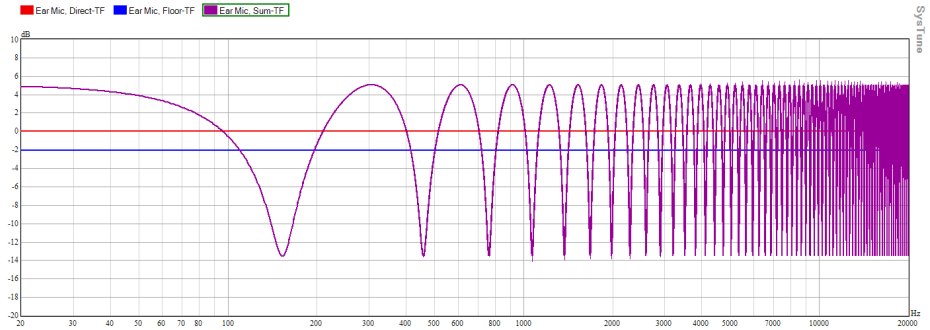
Measurement Compromises

Ground Bounce

A mic at ear height may have a nasty reflection from the floor.



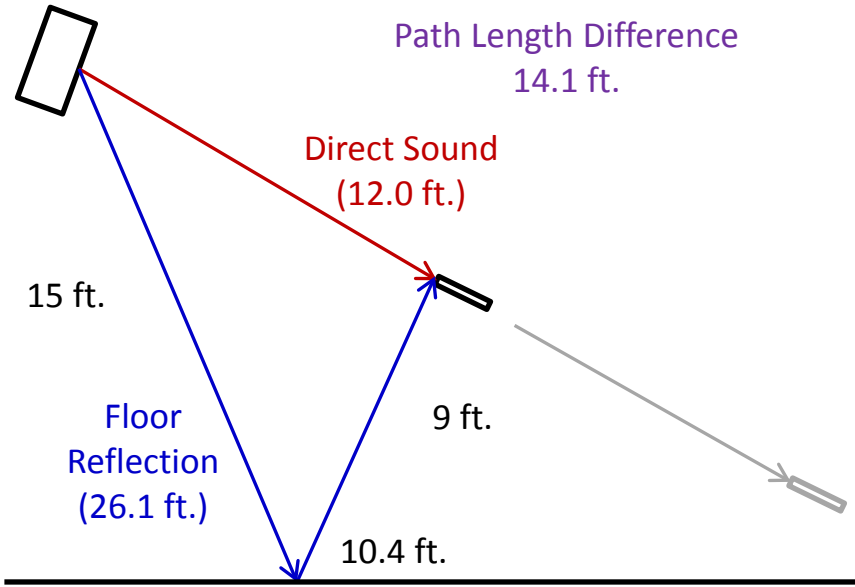


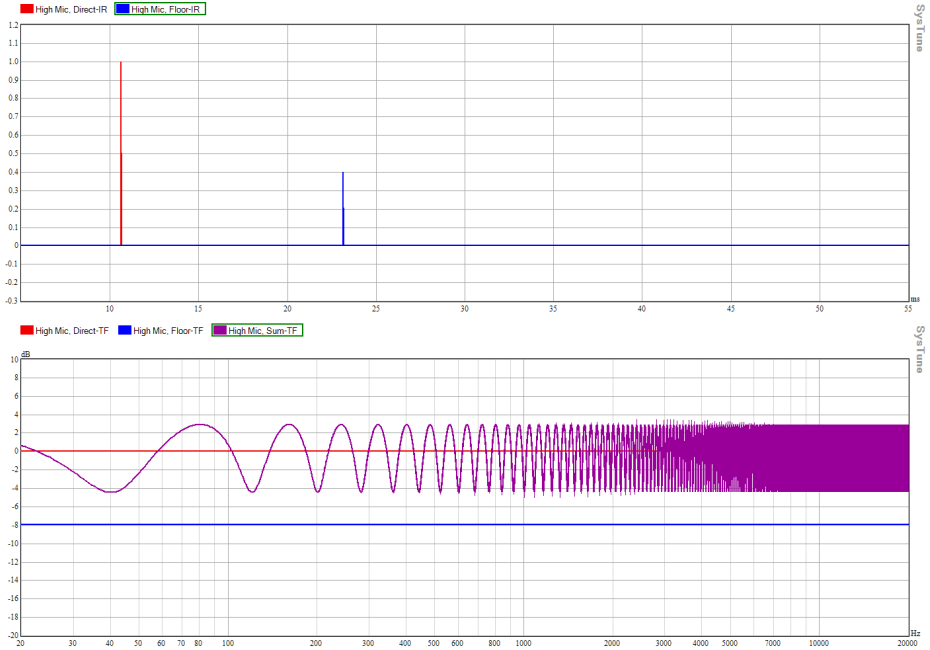


Measurement Compromises

Ground Bounce

A “high mic” technique increases the distance the floor reflection must travel.

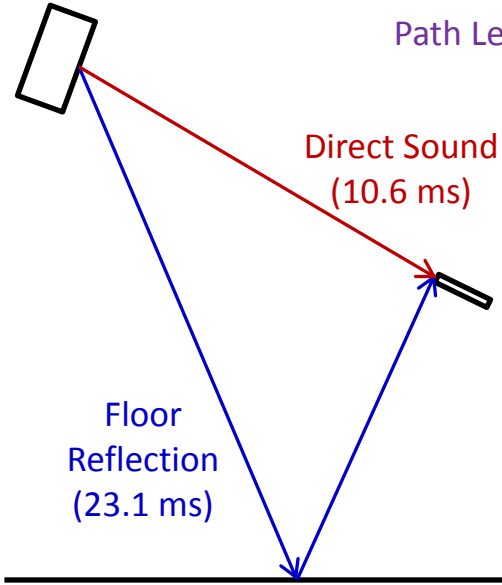




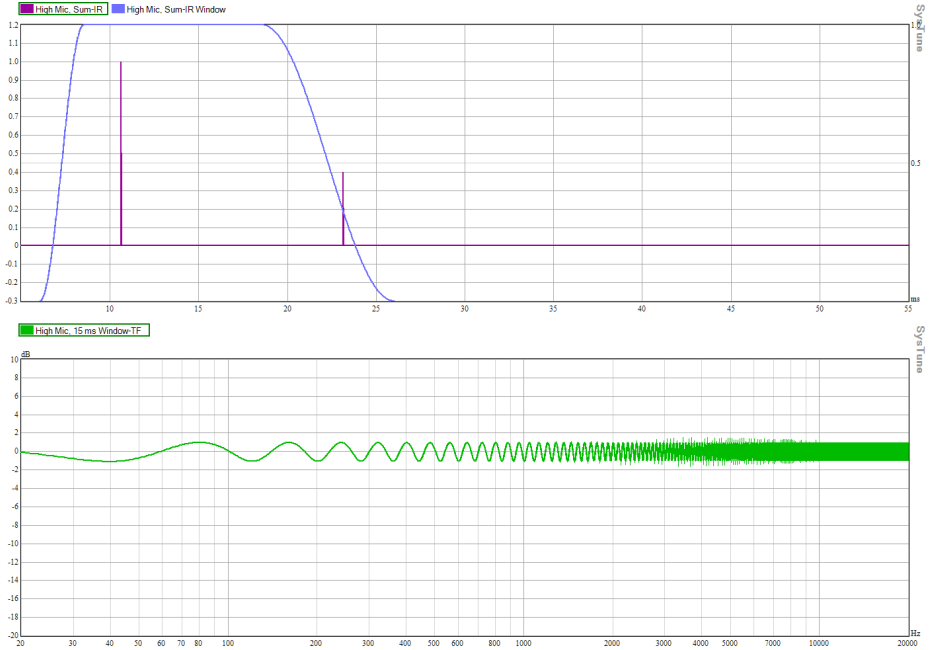
Measurement Compromises

Ground Bounce

The reflection can be windowed & the measurement can still have reasonable frequency resolution.



IR window of 15 ms yields approx. 100 Hz resolution



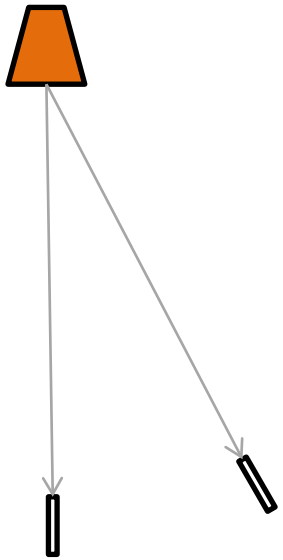
Measurement Compromises

Mic Placement – Single Loudspeaker

- Measure on-axis of the loudspeaker.
- Measure off-axis as well.
  The loudspeaker's response will not be exactly the same in all directions.
- Average the measurements using a "power average".

EQ based on the average response, but don't boost too much.

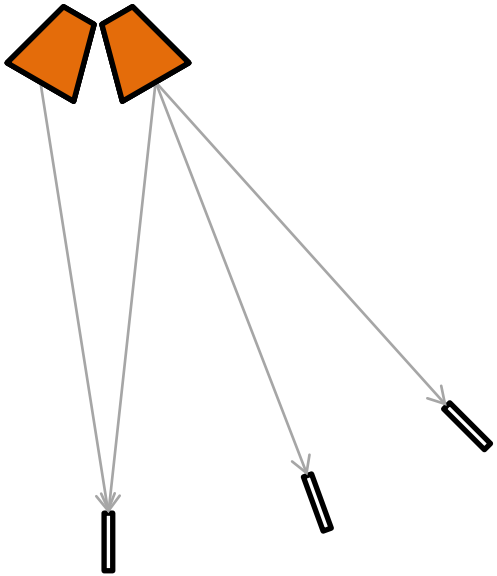
Too much energy on-axis will sound worse than less energy off-axis.



Measurement Compromises

Mic Placement – Clusters/Arrays

- Measure on and off-axis of the loudspeaker
- Measure in the overlap region of adjacent loudspeakers.
- Measure each loudspeaker separately & with both (all) loudspeakers on.

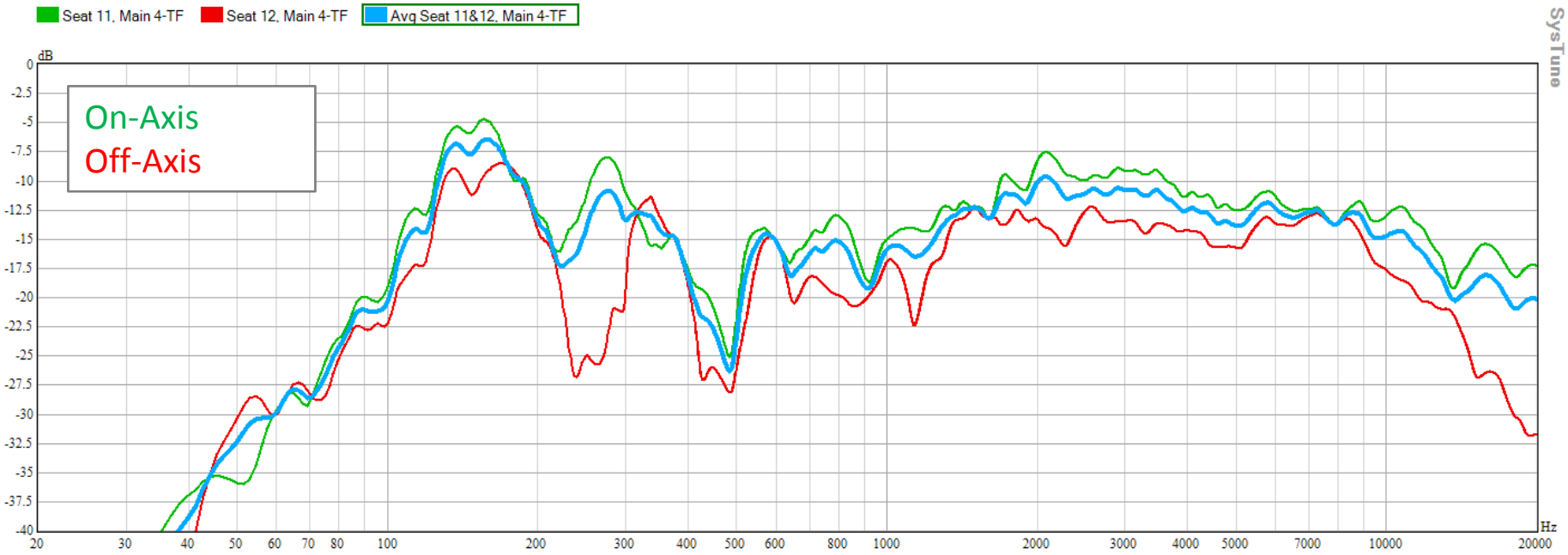


Make EQ decisions based on the combined response of all loudspeakers, but with knowledge of what each one is doing individually.

It's not possible to boost a cancellation!

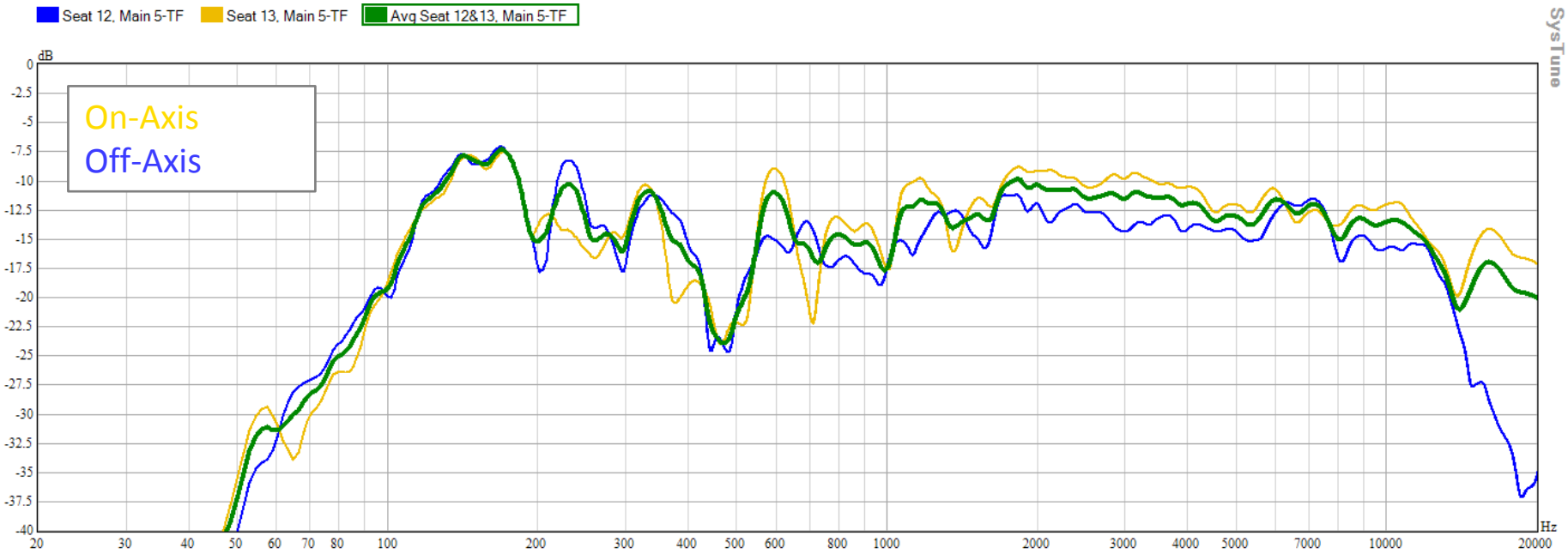
Measurement Compromises

Mic Placement – Clusters/Arrays: Example, Exploded Cluster



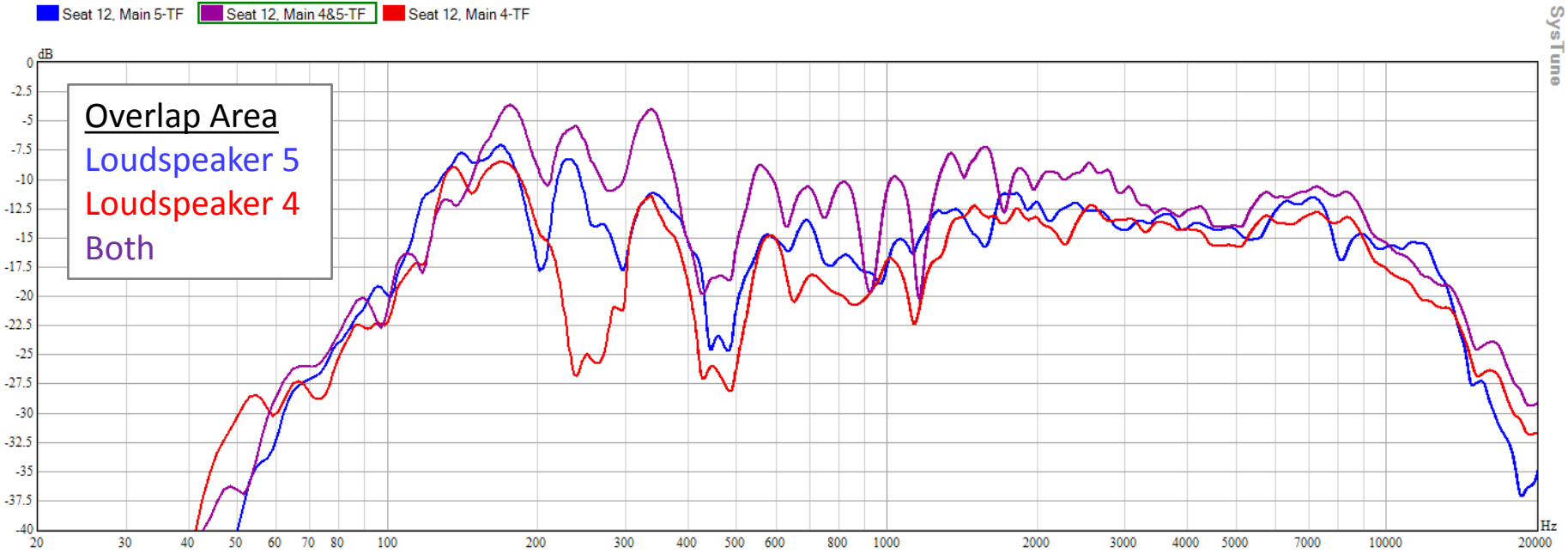
Measurement Compromises

Mic Placement – Clusters/Arrays: Example, Exploded Cluster



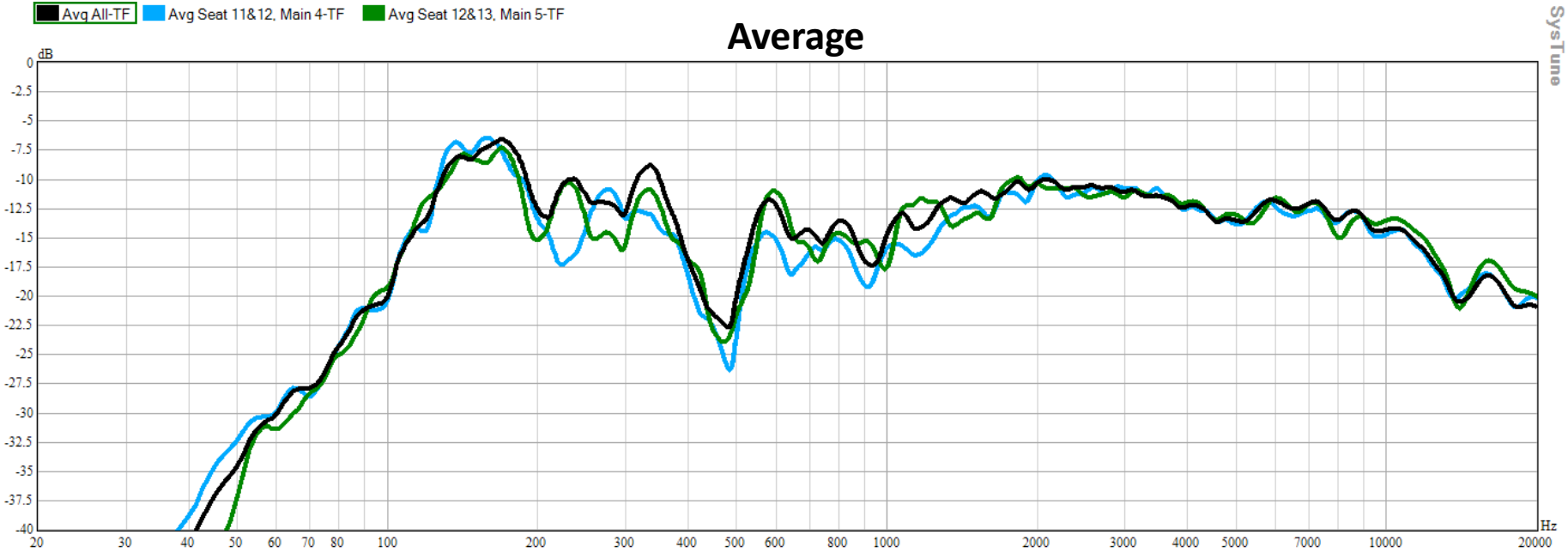
Measurement Compromises

Mic Placement – Clusters/Arrays: Example, Exploded Cluster



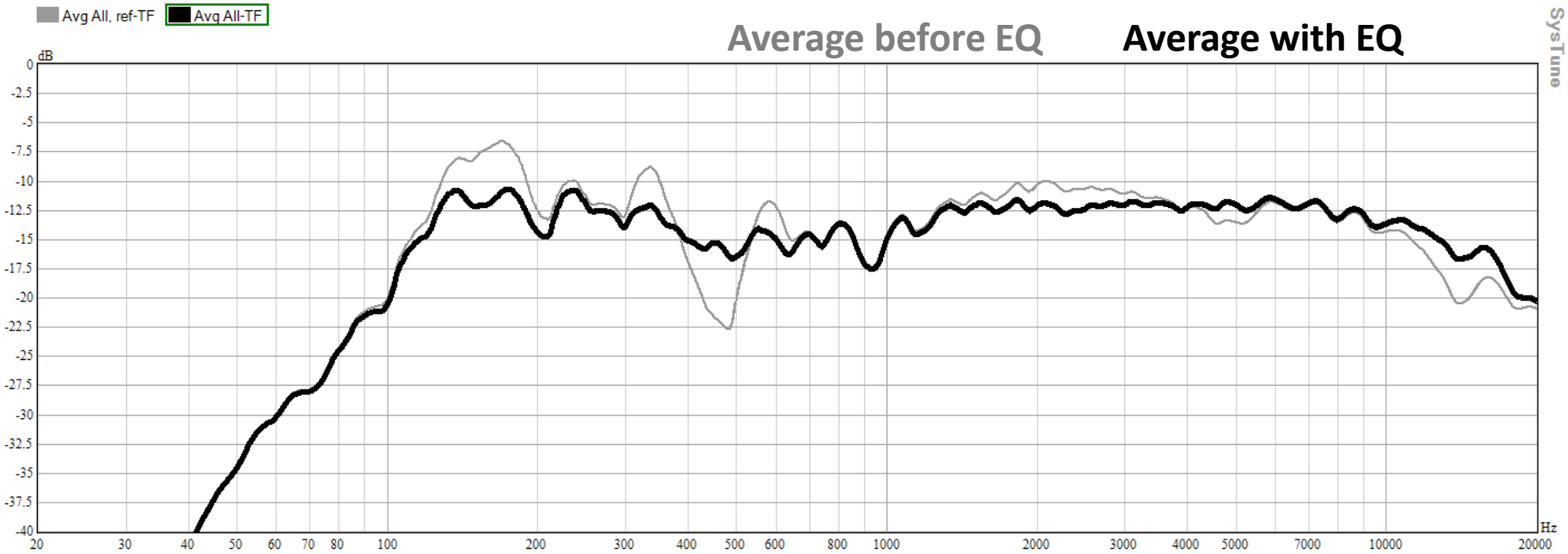
Measurement Compromises

Mic Placement – Clusters/Arrays: Example, Exploded Cluster



Measurement Compromises

Mic Placement – Clusters/Arrays: Example, Exploded Cluster



Measurement Compromises

Mic Placement – Ceiling Loudspeakers *("high mic" may not be possible)*

A. Measure on-axis of a loudspeaker.

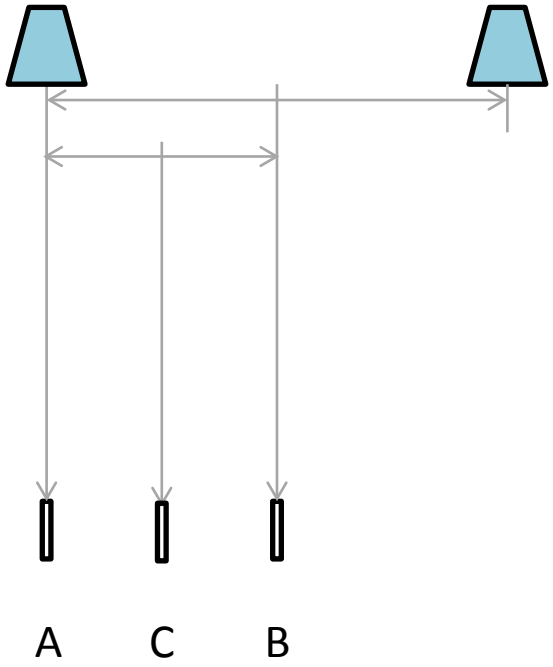
B. Measure midway between two adjacent loudspeakers.

C. Measure halfway between points A and B.

- Average all 3 measurements.

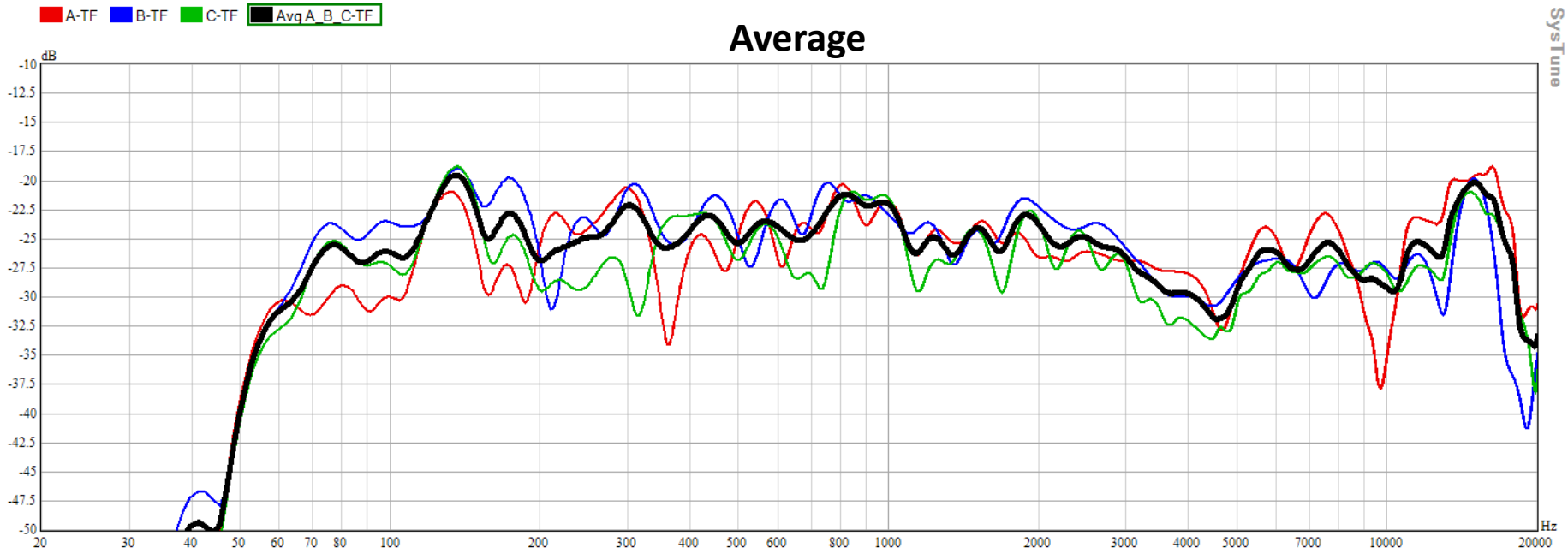
EQ based on the average response, but don't boost too much.

Too much energy on-axis will sound worse than less energy off-axis.



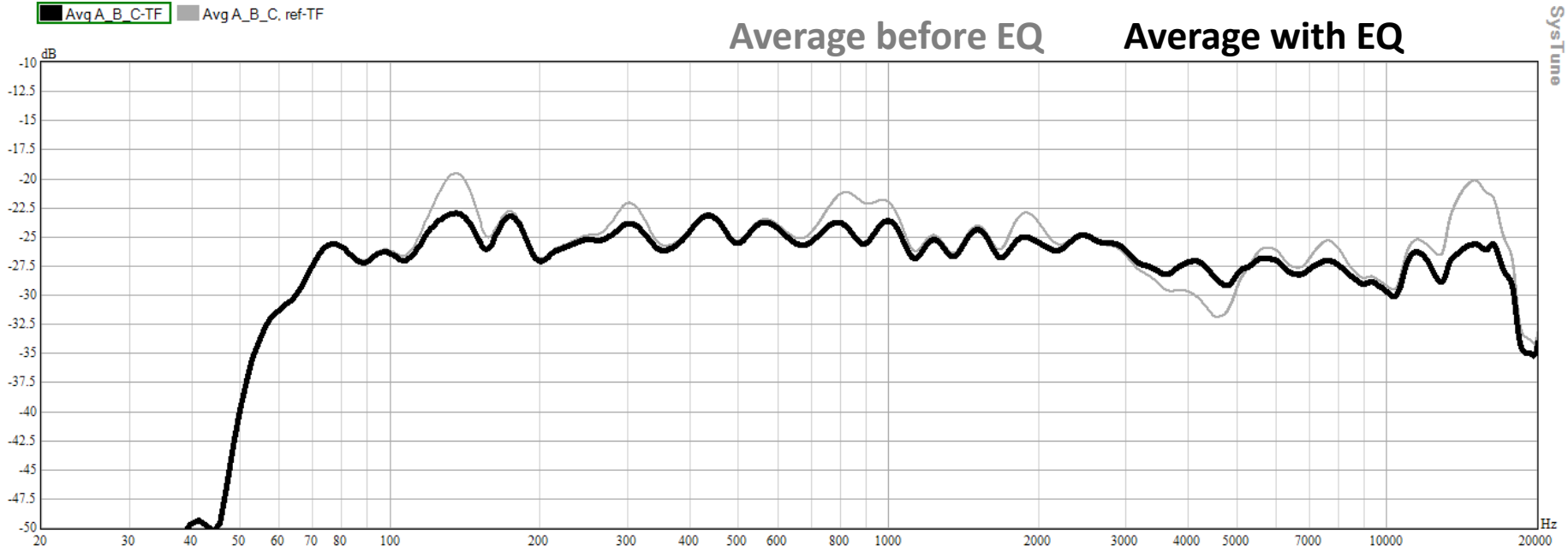
Measurement Compromises

Mic Placement – Ceiling Loudspeakers: Example



Measurement Compromises

Mic Placement – Ceiling Loudspeakers: Example



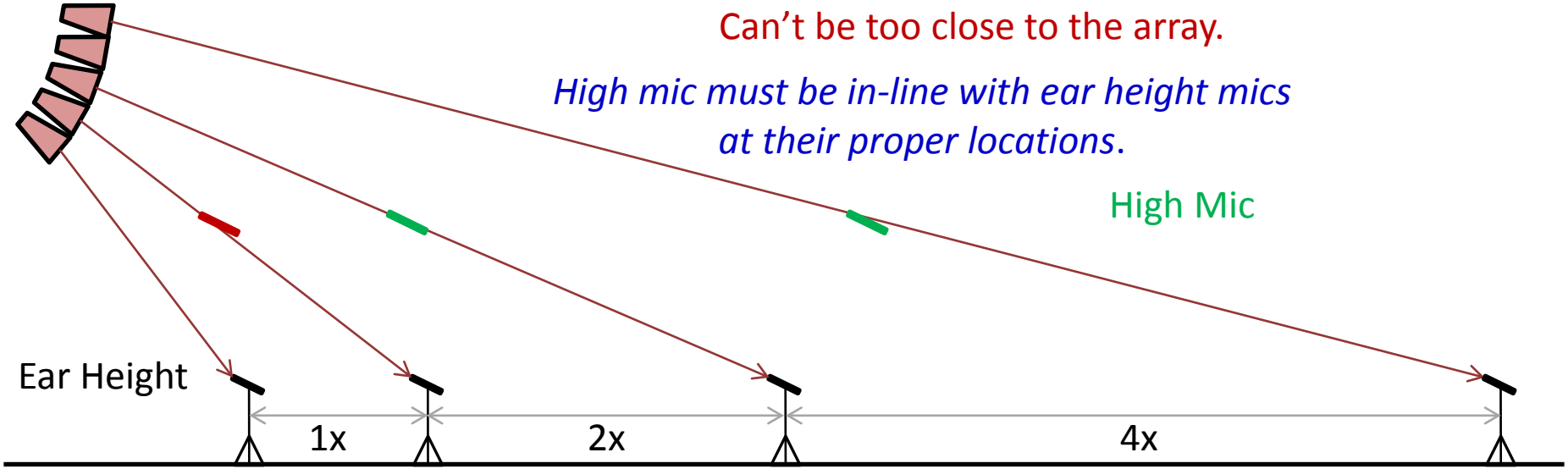
Typical Measurement Scenarios

Mic Placement – Line Arrays

- Measure on-axis of the array at log-spaced distances.
- Average the measurements.

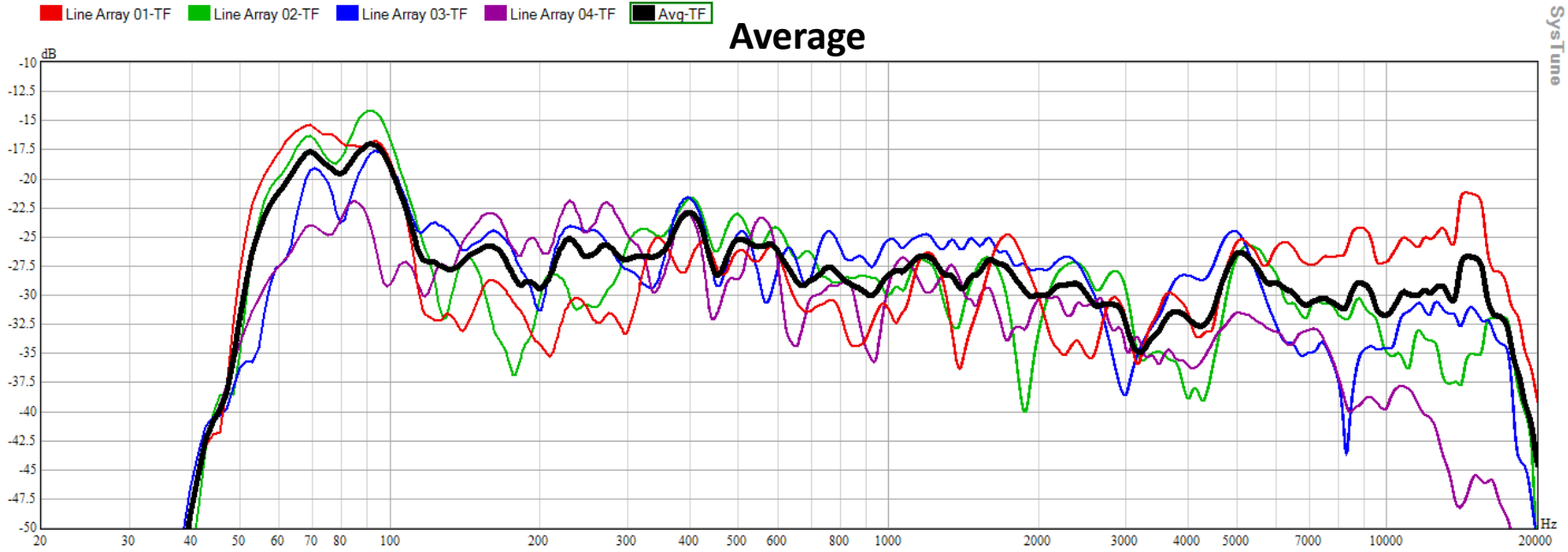
High mic may not be possible for all locations.
Can't be too close to the array.

High mic must be in-line with ear height mics at their proper locations.



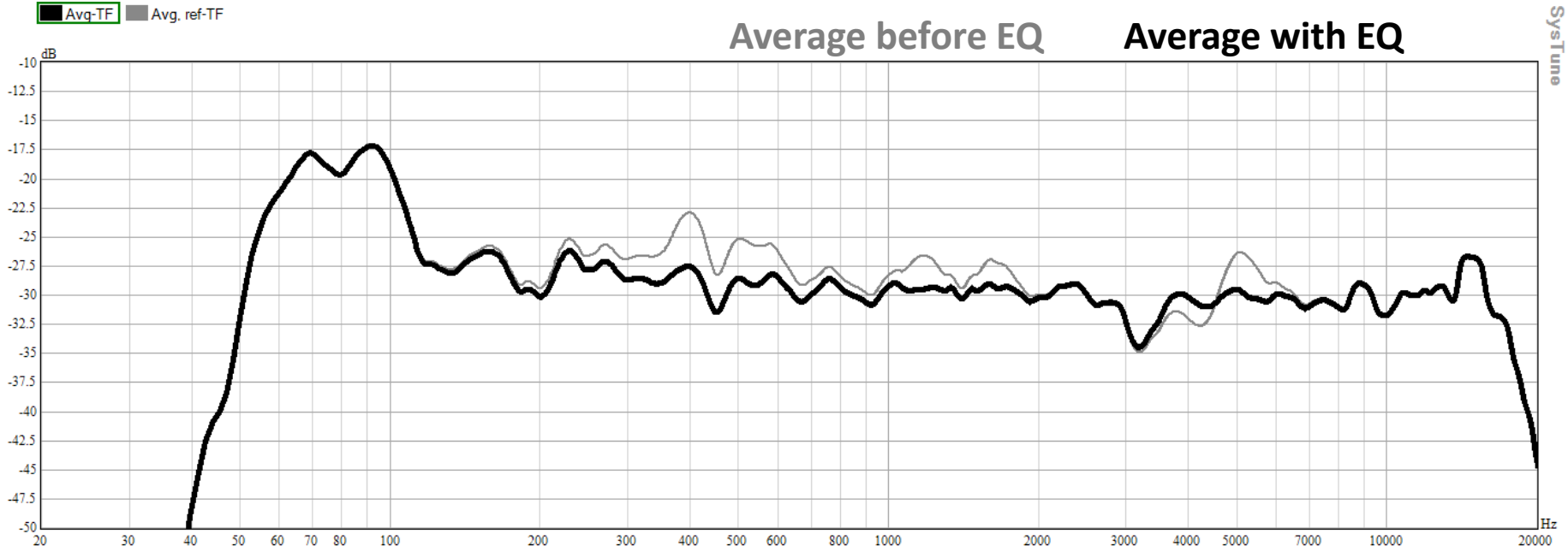
Typical Measurement Scenarios

Mic Placement – Line Arrays: Example



Typical Measurement Scenarios

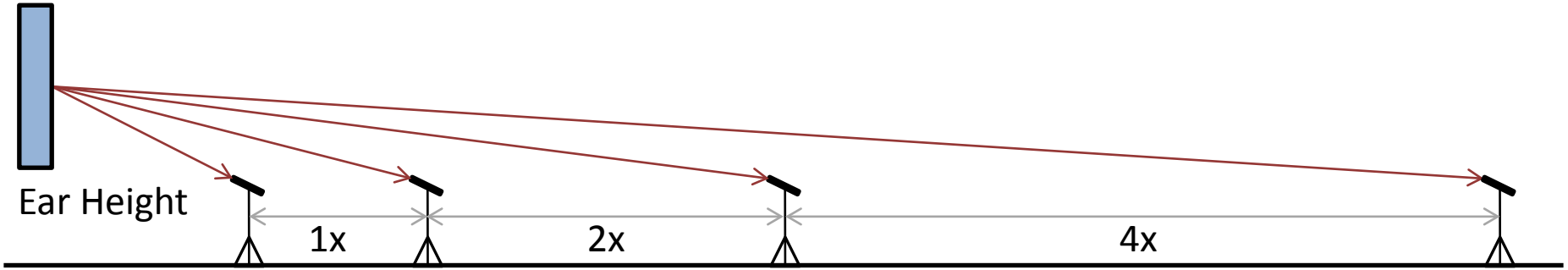
Mic Placement – Line Arrays: Example



Typical Measurement Scenarios

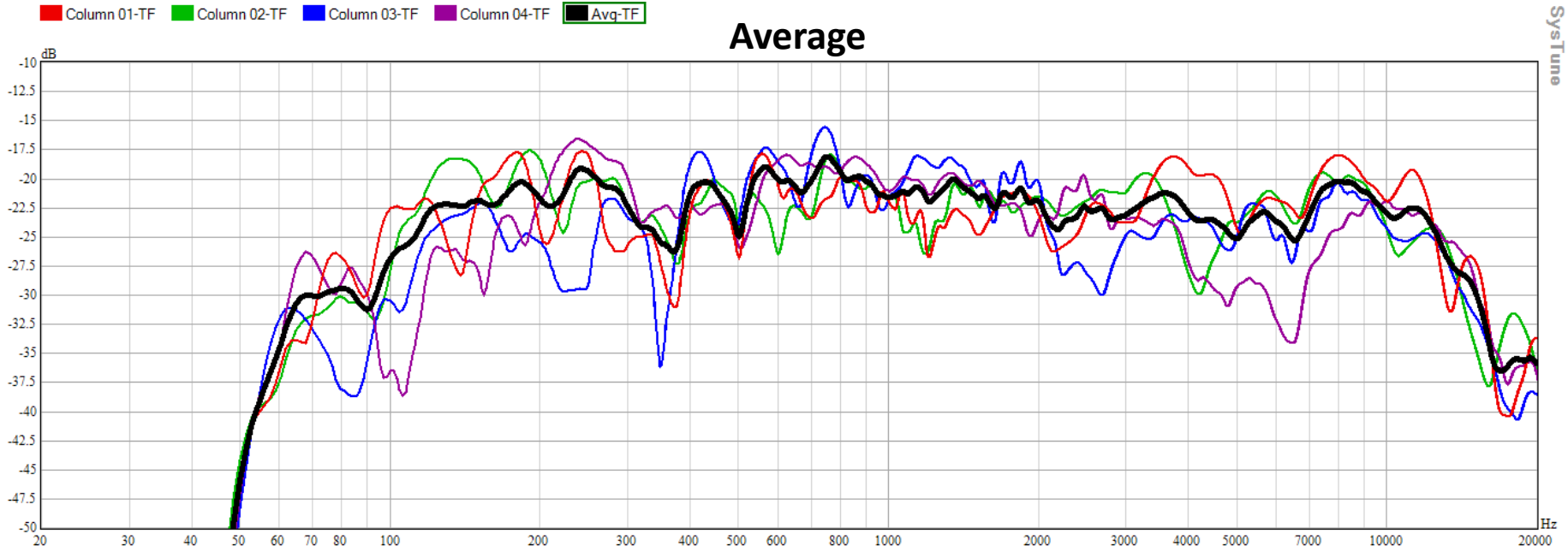
Digitally Steered Line Arrays & Columns

- Measure on-axis of the array at log-spaced distances.
- Average the measurements
- High mic method may not be possible (must be in the coverage pattern).



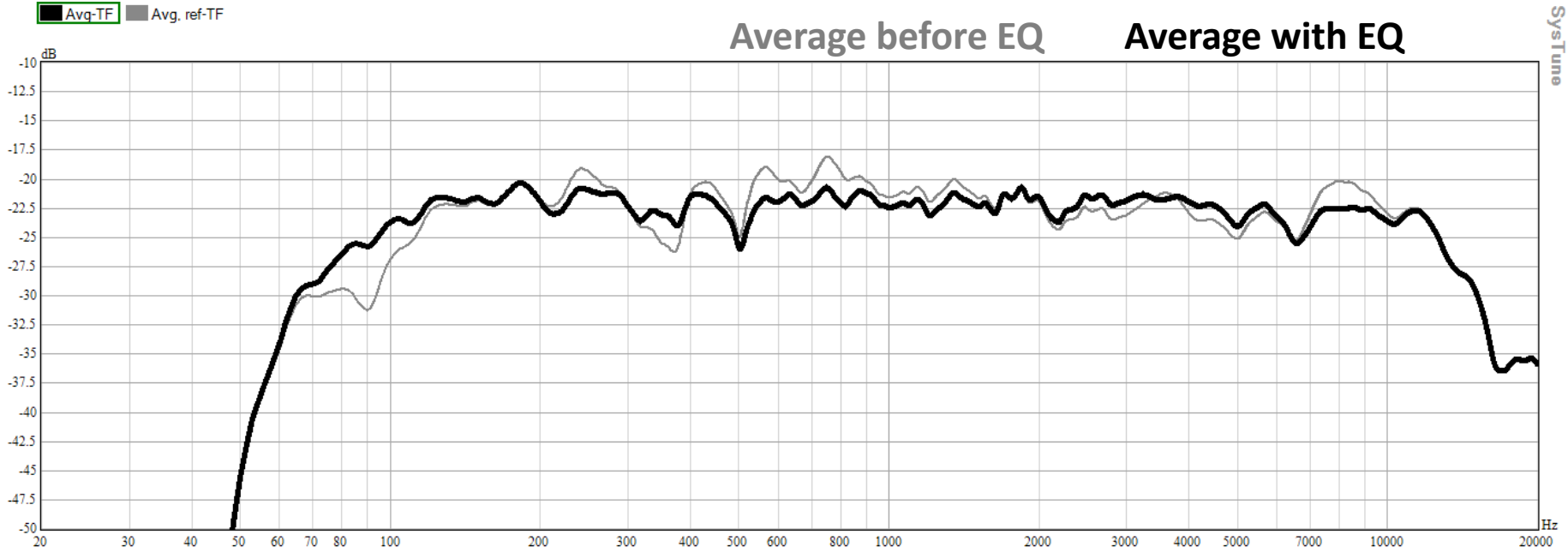
Typical Measurement Scenarios

Digitally Steered Line Arrays & Columns: Example



Typical Measurement Scenarios

Digitally Steered Line Arrays & Columns: Example

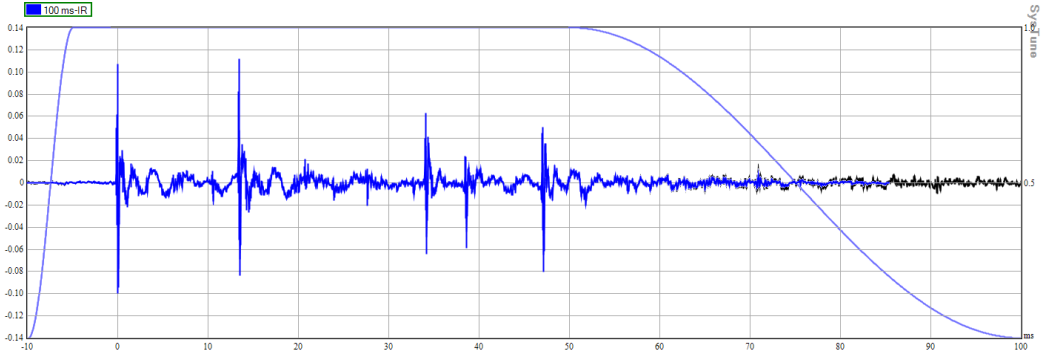


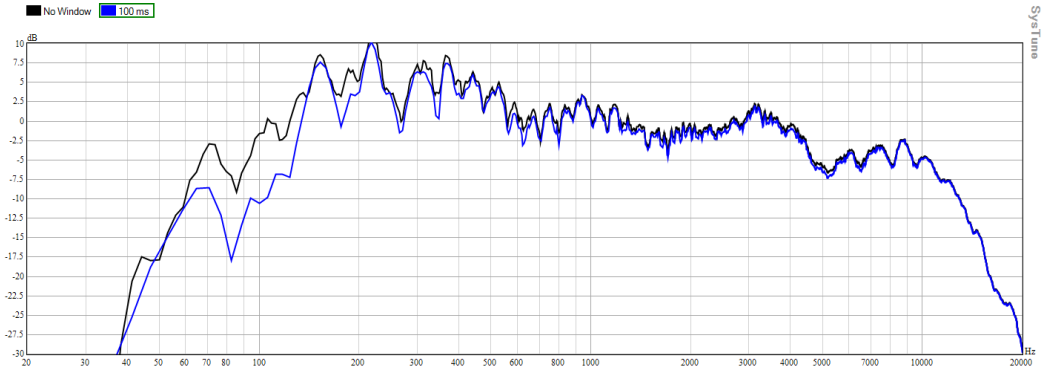
Measurement Compromises

Trade-off between isolation & resolution

To get better frequency resolution more time is required. This means using a longer IR window.

A longer IR window could allow more reflections into the measurement.



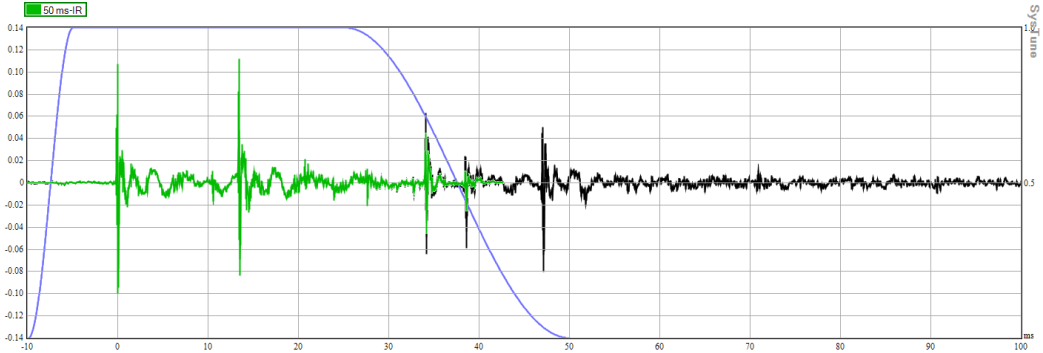


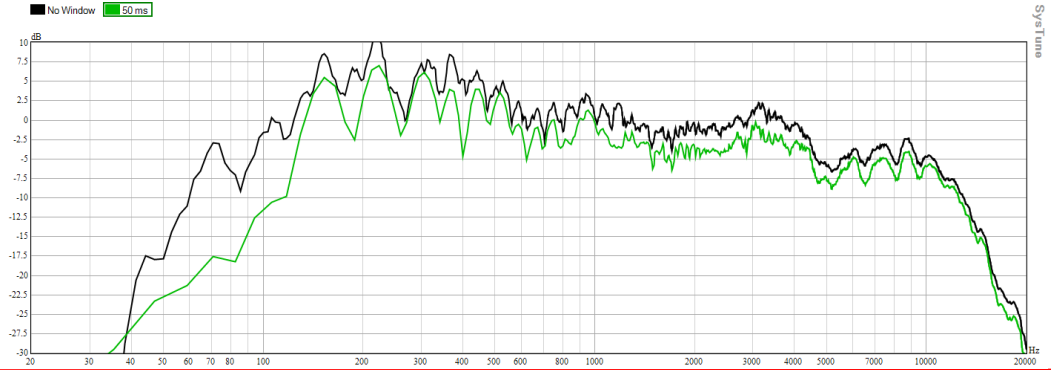
Measurement Compromises

Trade-off between isolation & resolution

How much frequency resolution do we need when doing on-site measurements?

Is 40 Hz needed?



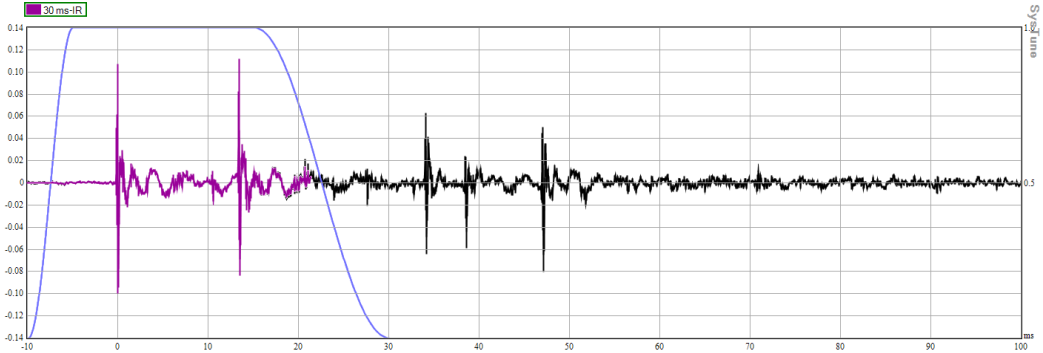


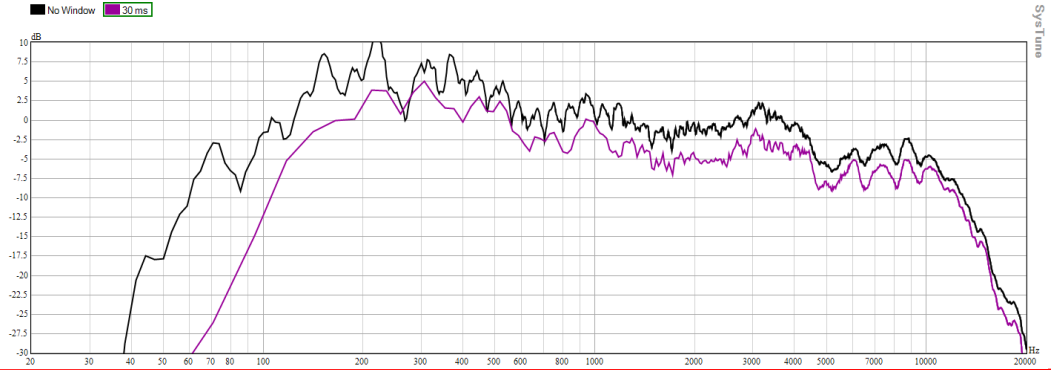
Measurement Compromises

Trade-off between isolation & resolution

How much frequency resolution do we need when doing on-site measurements?

Is 70 Hz sufficient?



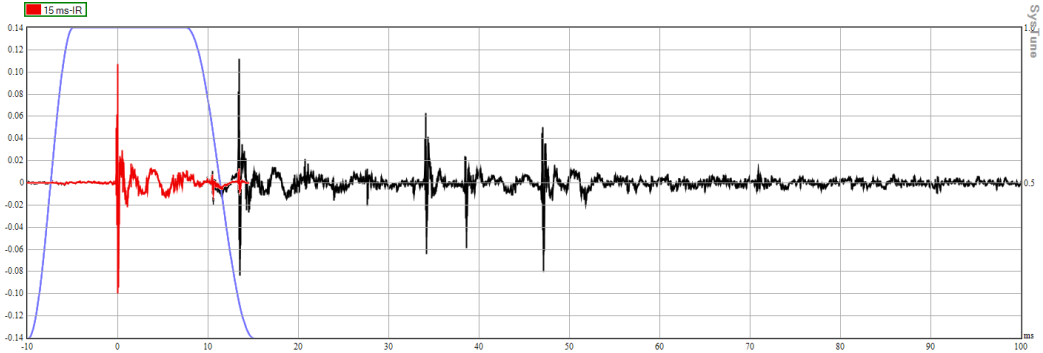


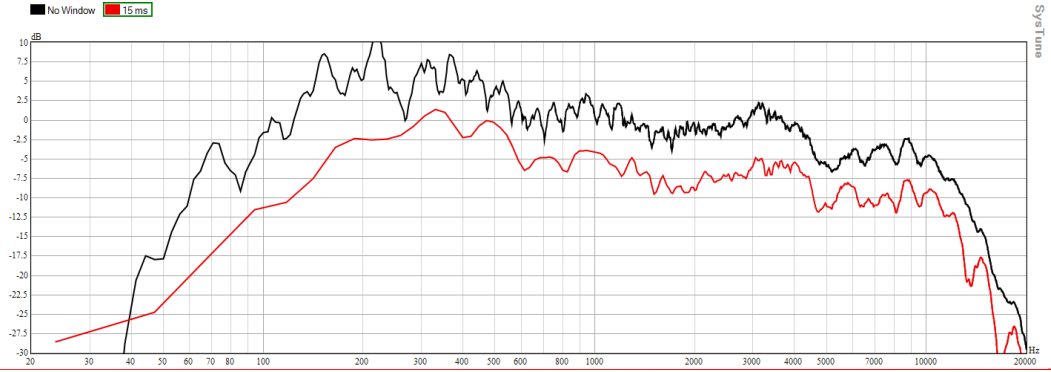
Measurement Compromises

Trade-off between isolation & resolution

How much frequency resolution do we need when doing on-site measurements?

Will 140 Hz be OK?



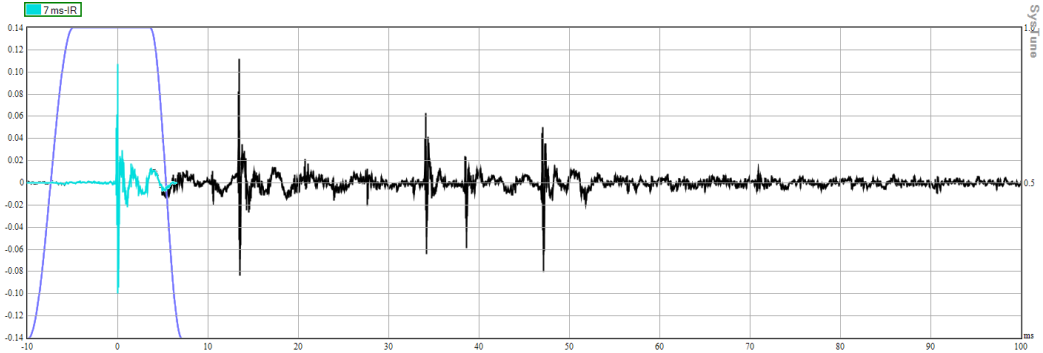


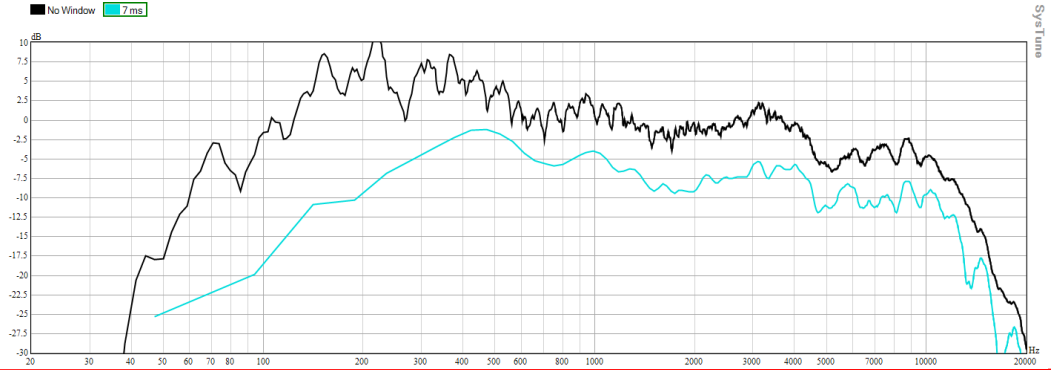
Measurement Compromises

Trade-off between isolation & resolution

How much frequency resolution do we need when doing on-site measurements?

What about 290 Hz?



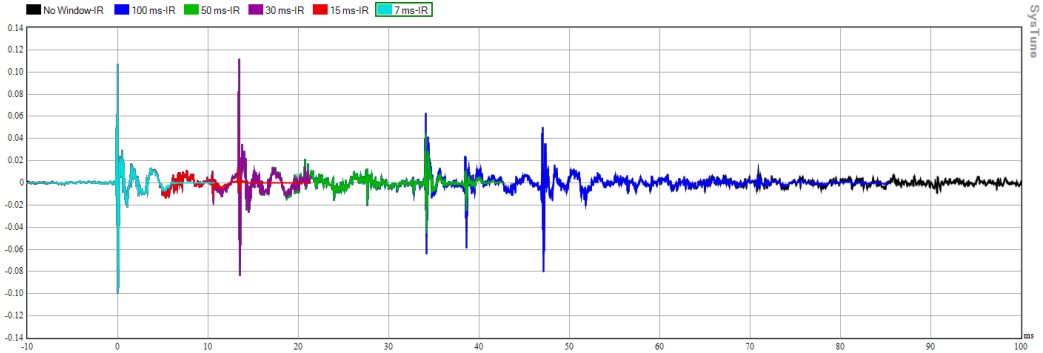


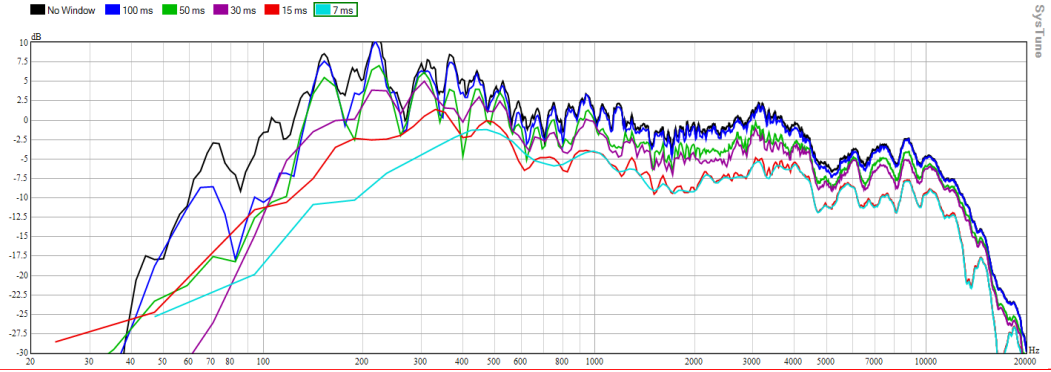
Measurement Compromises

Trade-off between isolation & resolution

Are we going to try to apply narrow-band (high Q) filters?

Given the frequency resolution limitations of IR windowing to get at the direct sound from the loudspeaker, will we be able to accurately apply narrow-band filters?

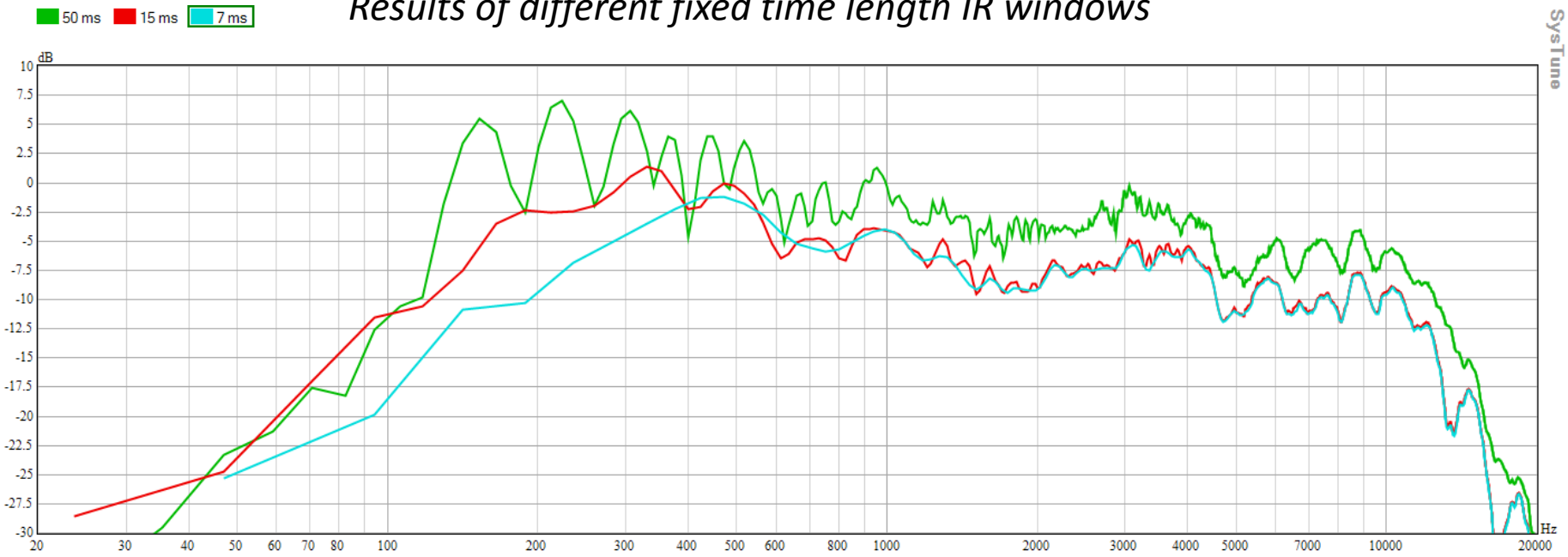




Measurement Compromises

Trade-off between isolation & resolution

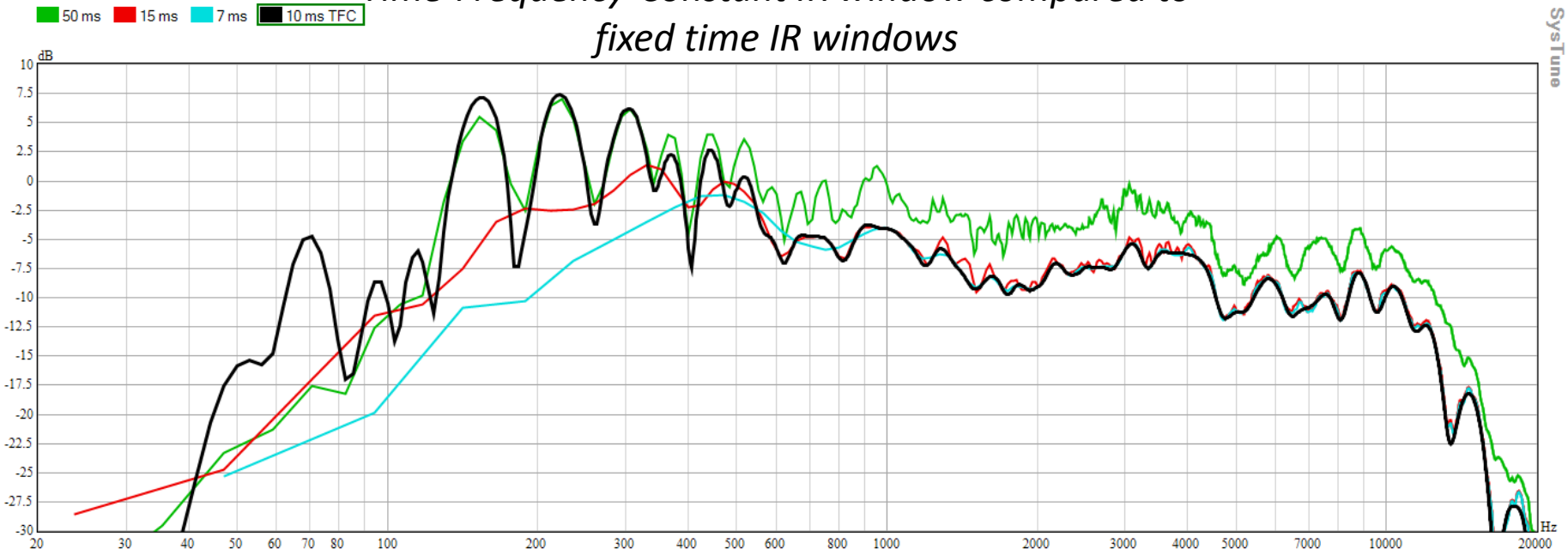
Results of different fixed time length IR windows



Measurement Compromises

Trade-off between isolation & resolution

Time-Frequency-Constant IR window compared to fixed time IR windows



Measurement Compromises

Linear vs. non-linear operation

- Initial tuning should always be done in the linear operating range of the loudspeaker system.

What if the system will typically be run at, or beyond, its maximum recommended limits? (i.e. not enough rig for the gig!)

Requote your maintenance contract!

- Listen at typical levels (hearing protection may be required).
- Power compression is frequency dependent. Typically lower frequency pass bands will have more reduction that high frequency pass bands.
- Distortion will generally add higher frequency content to the output.
- These can alter the perceived spectral balance of the sound system. EQ as needed by ear to try to get acceptable sound quality at the target SPL.

Section 8

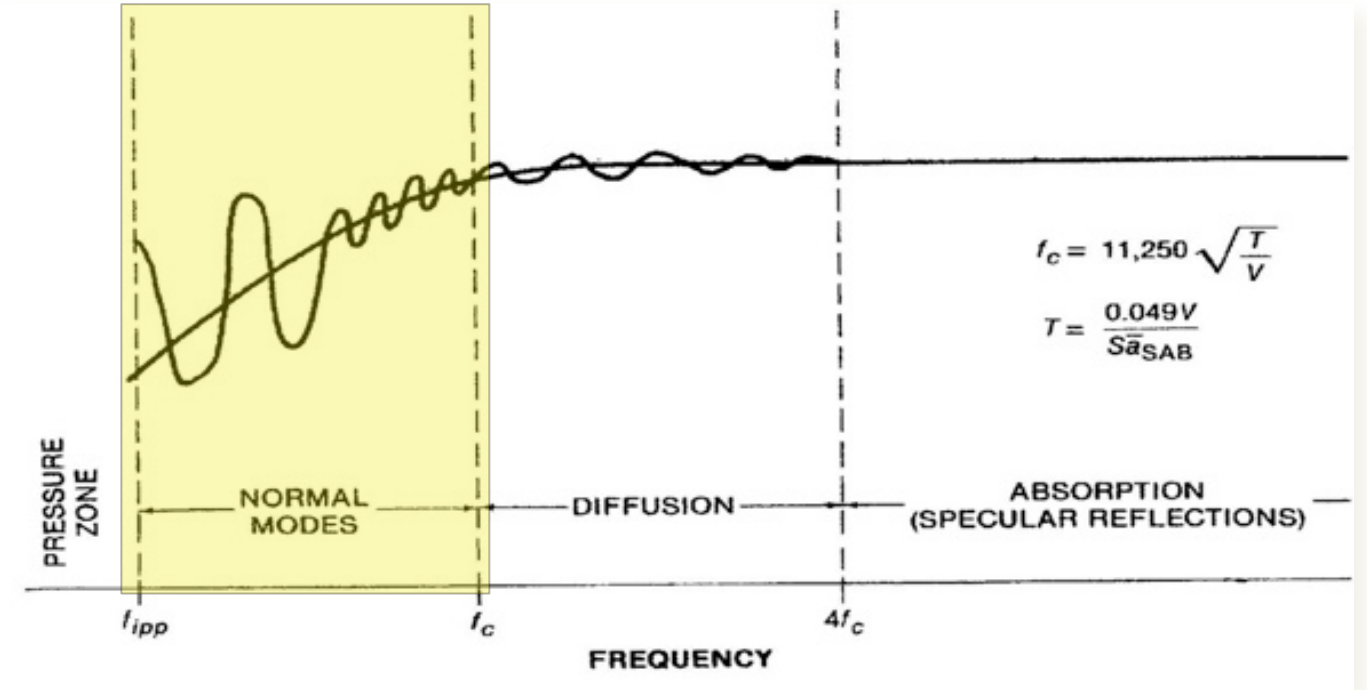
The Lows Need to be Treated Differently

John Murray

The Direct Sound's Long Wavelengths

- The long wavelengths of the direct sound cannot be isolated from the room's reflections
- A 20 Hz wavelength is 56 ft./17 m long, so isolating the direct sound from room reflections at very-low frequencies is an impossibility in all but the most cavernous spaces
- Likewise, room modes will also mask the long wavelengths of the direct sound in our measurements
- Therefore, for equalization, the low frequencies will have to be handled differently than the mid and high frequencies because we cannot isolate them from the room

Room Resonances/Normal Modes



Leo Beranek's Controllers of Steady-State Room Acoustic Response, courtesy of BB&N/*Sound from System Engineering*, Second Edition, D. & C. Davis

The Schroeder Frequency

- These standing waves are the room's resonances that are very noticeable in the lower frequencies
- Fc, the critical frequency, is commonly known as the Schroeder Frequency as he created the formula Beranek applies
- Note that in the graph, Fc was calculated from the T60 of a reverberant room
- In a small or large dead room without a statistically reverberant field, we cannot use this formula to calculate the onset of standing-wave acoustics

The Schroeder Frequency

- Instead a simple formula, developed by Don Davis and Chips Davis during their development of Live-End Dead-End (L.E.D.E.) Technology, can be used for Fc:

$$SF_{DD} = 3 \times (\text{speed of sound}) / (\text{room's smallest dimension})$$

- It is based on having at least 3 wavelengths across the room's smallest dimension to begin generating a homogeneous sound field in the space

The Schroeder Frequency

- Any single frequency response, or transfer function, measurement made below the Schroeder Frequency using a measurement mic anywhere in the room cannot be trusted for system equalization
- For any frequency below SF_{DD}, the frequency response is hopelessly corrupted by the comb-filtering of multiple room modes

Standing Wave Demonstration

The Schroeder Frequency

- Since the level difference can be as great as 30 dB, averaging less than a great number of mics at different positions would probably not yield a statistically accurate average response
- So, how do we equalize a system reliably below the Schroeder Frequency?

Don Keele's Extreme Near-Field Method

- In 1973 Don Keele presented an AES paper that demonstrated and proved a technique used earlier by Ray Newman of Electro-Voice to obtain the far-free-field lower pass-band response of any cone speaker
- When the measurement mic is positioned very close to the dust cap of the cone, the response measured will be very close to the far-free-field response of the device up to a limiting frequency

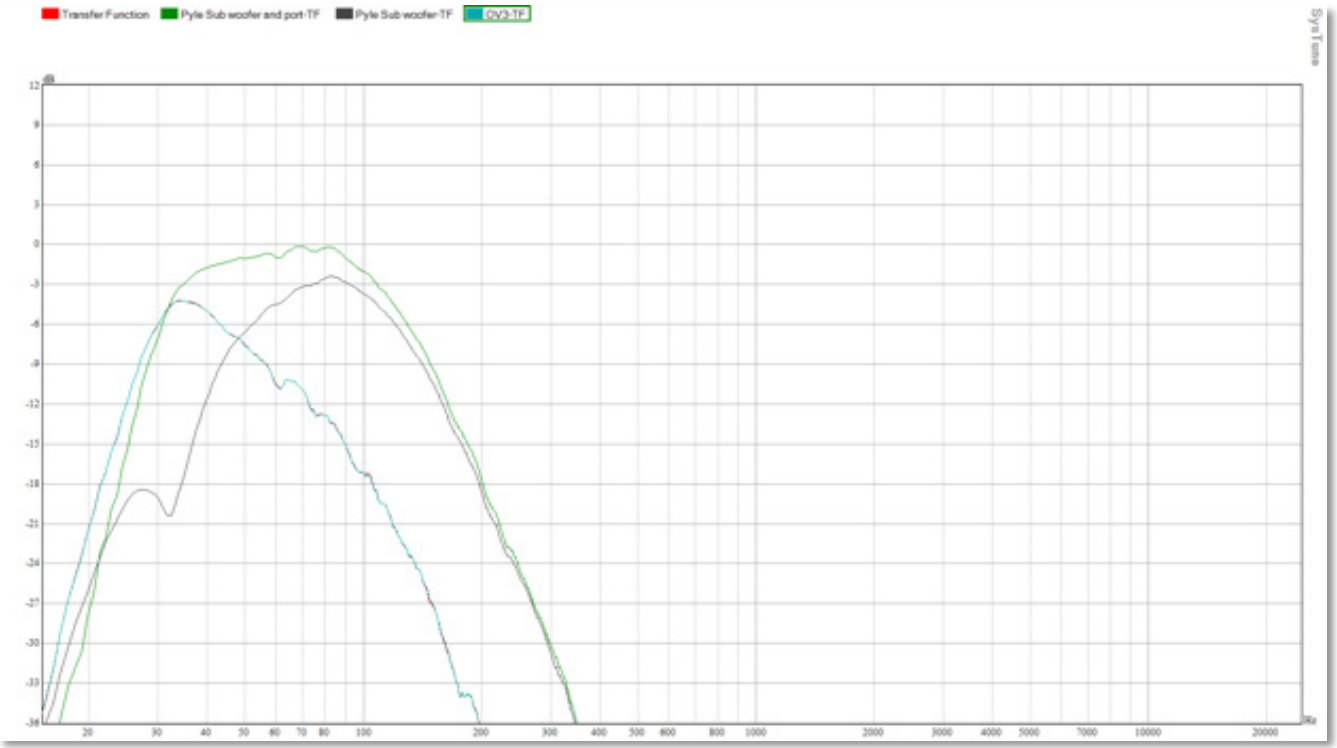
Don Keele's Extreme Near-Field Method

- If the speaker is in a ported enclosure, the response of the cone and the port opening(s) are summed, with the level of the port measurement power-weighted in dB proportionally by the port-to-cone area ratio

Don Keele's Extreme Near-Field Method

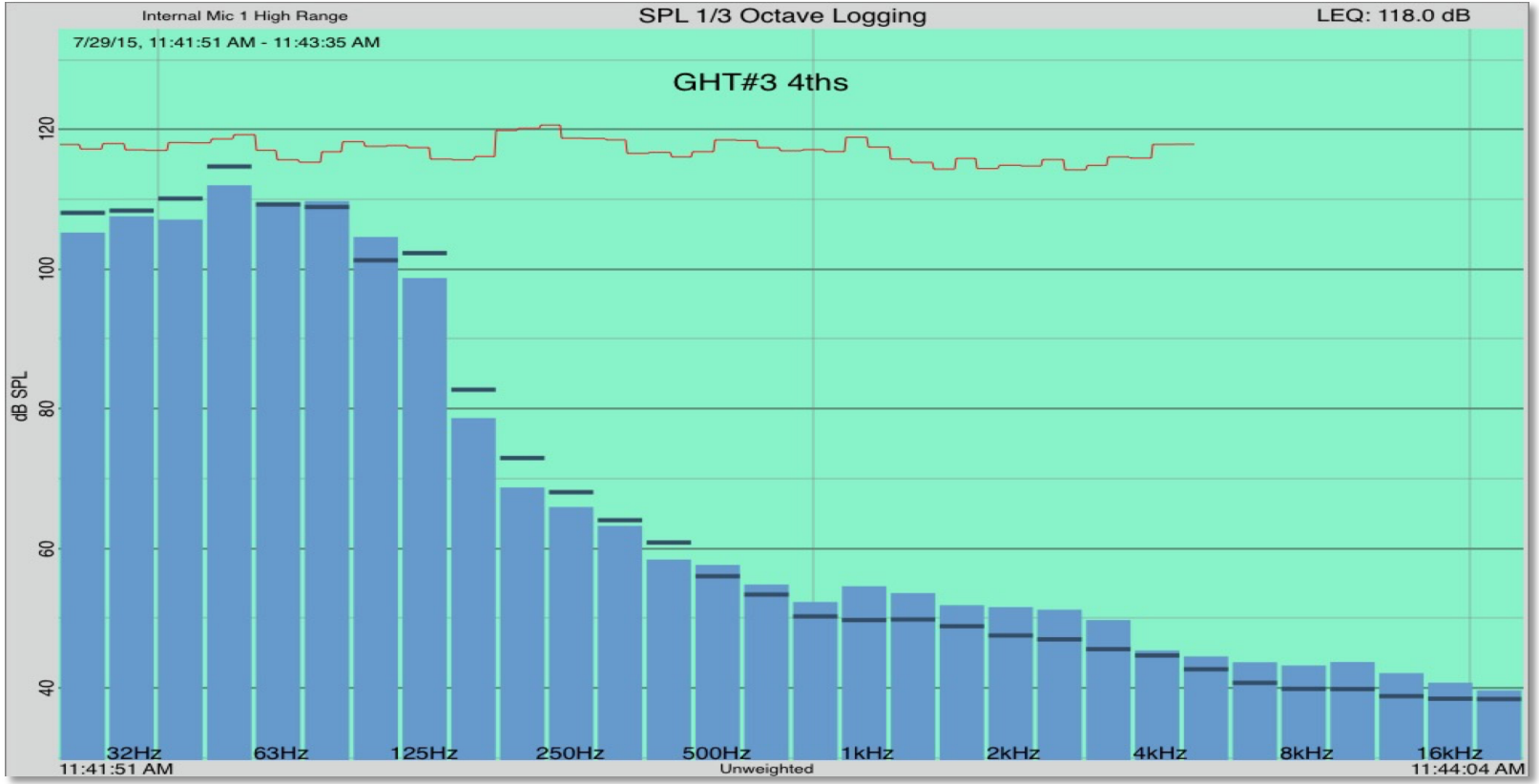
Extreme near-field low-frequency measurement technique

Don Keele's Extreme Near-Field Method

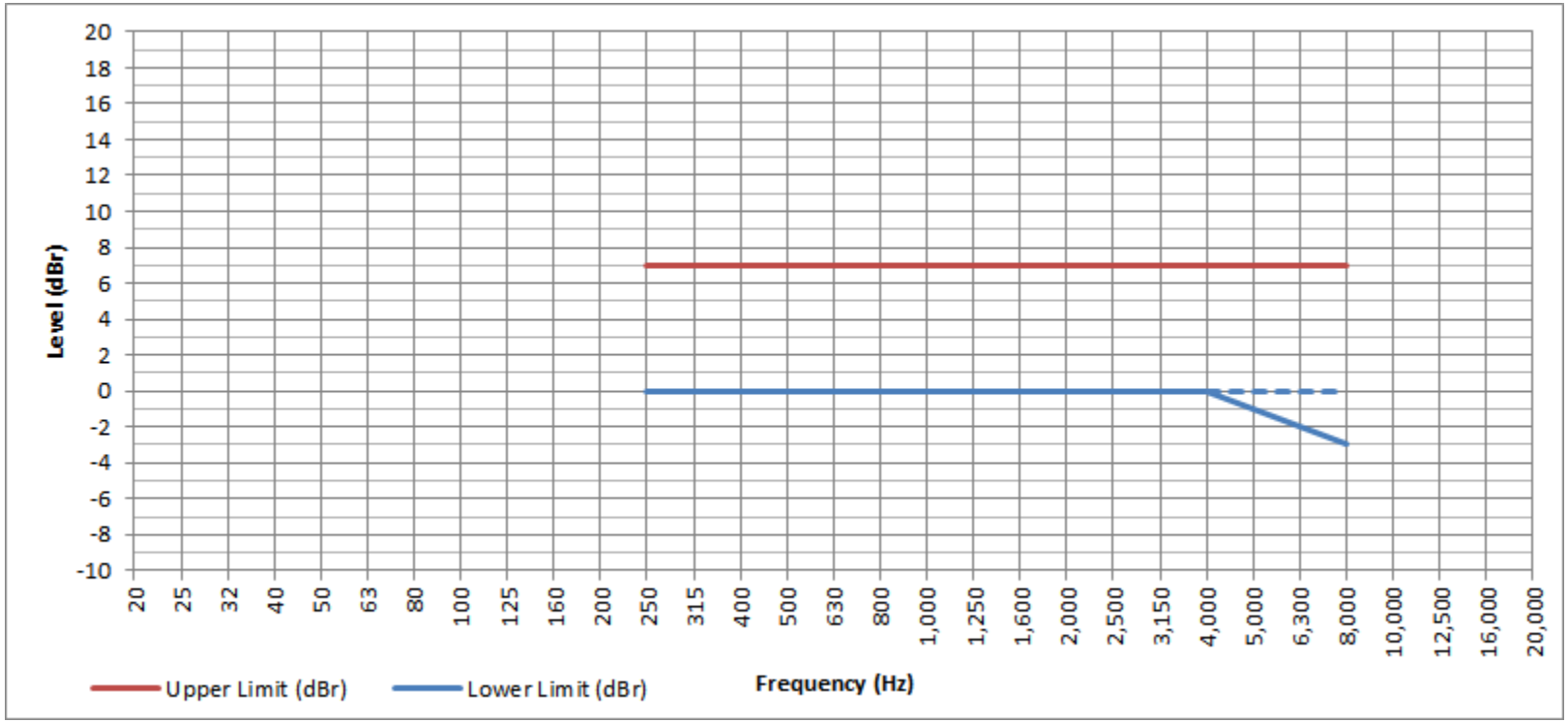


Left: port @ -5dB Right: cone Top: summed LF response

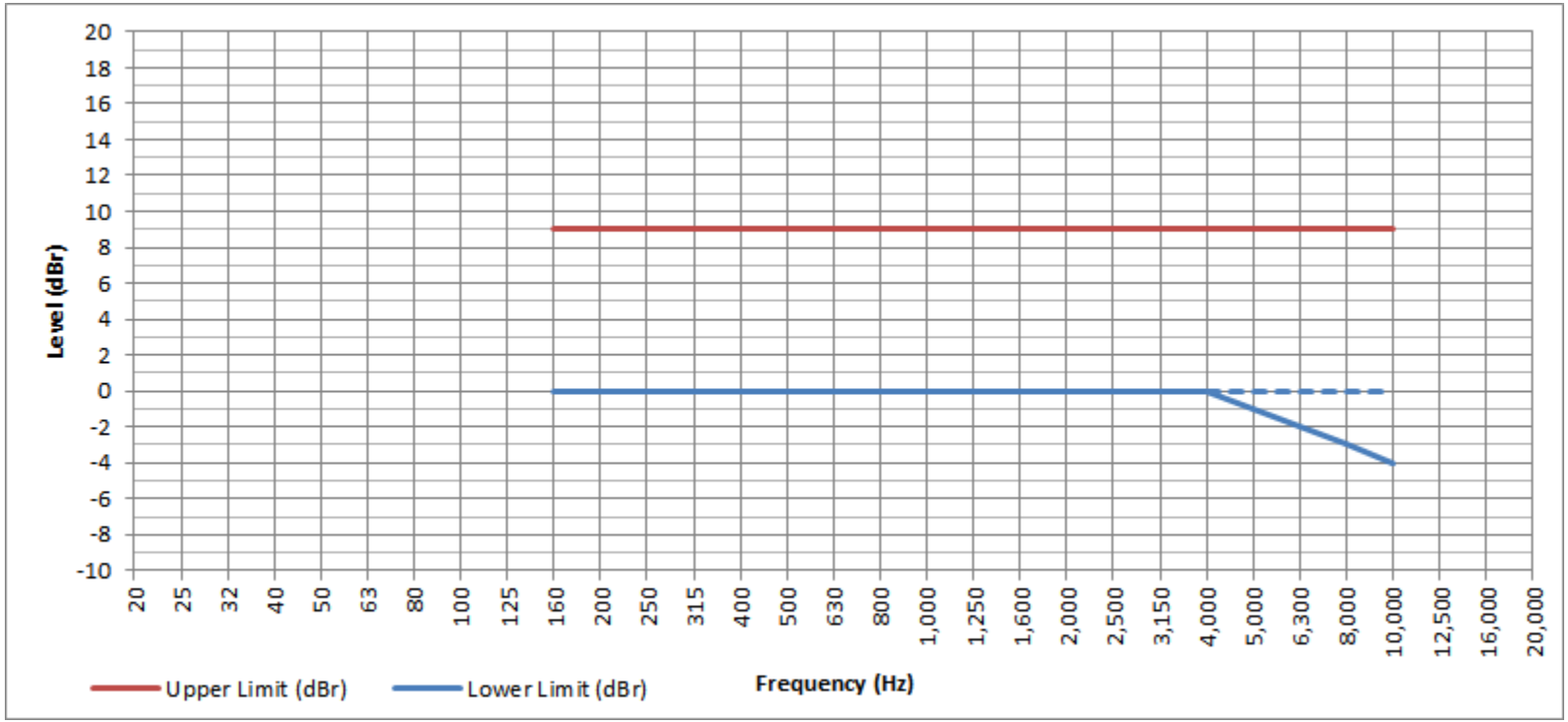
1/3-Octave LEQ Room Average



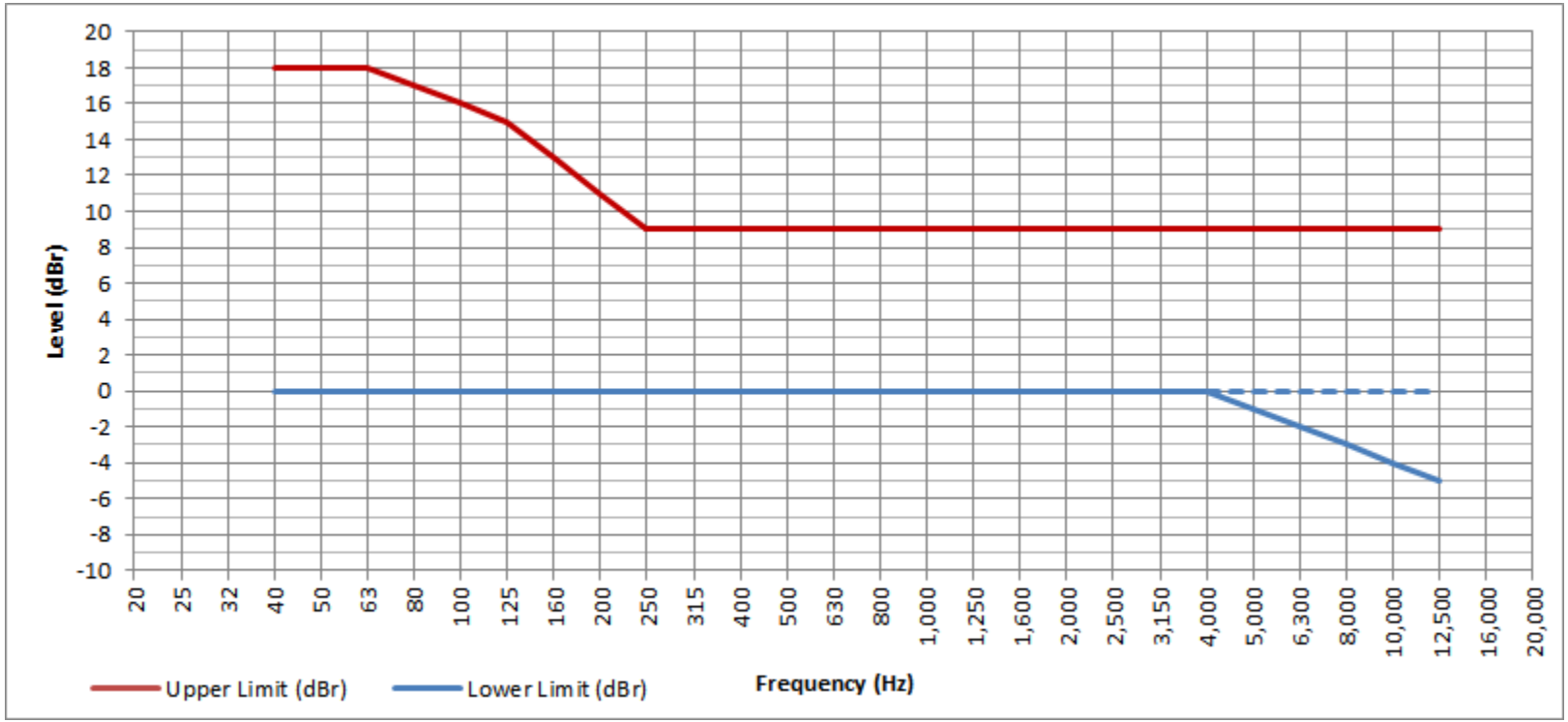
InfoComm 8M-201x Voice-Communications Tolerance Limits



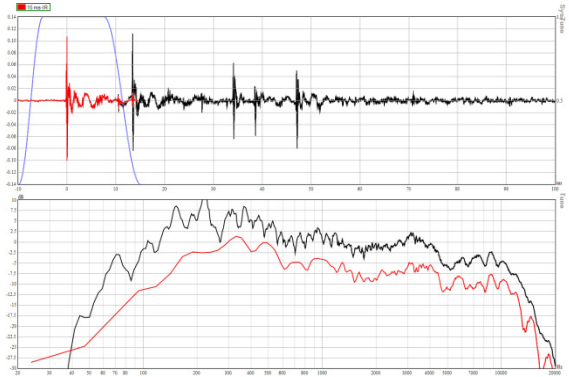
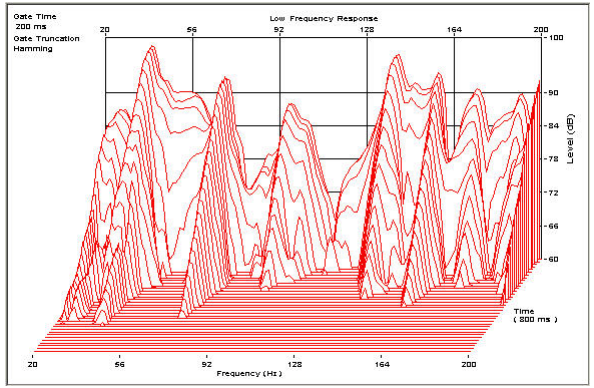
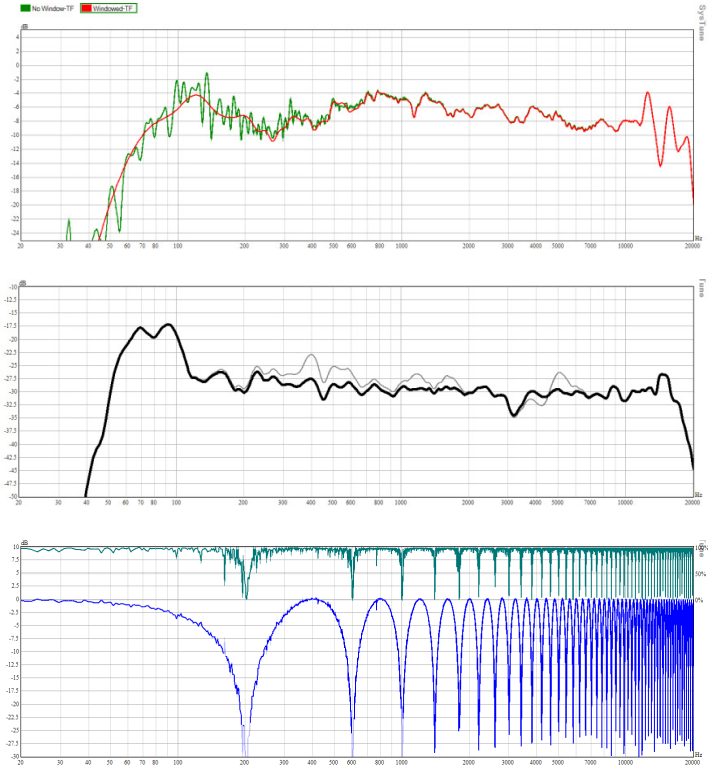
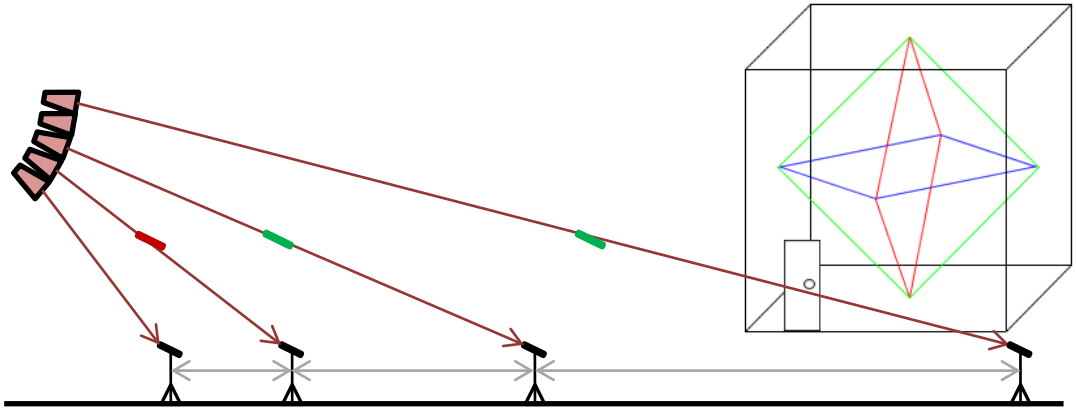
InfoComm 8M-201x Limited-Bandwidth Tolerance Limits



InfoComm 8M-201x Full-Bandwidth Tolerance Limits



Section 9

Making EQ Decisions

Properly Equalized Sound Systems