

Why Higher Sampling Rates Are Better... And Worse

Jon Boley

DSP Engineer

Shure Incorporated

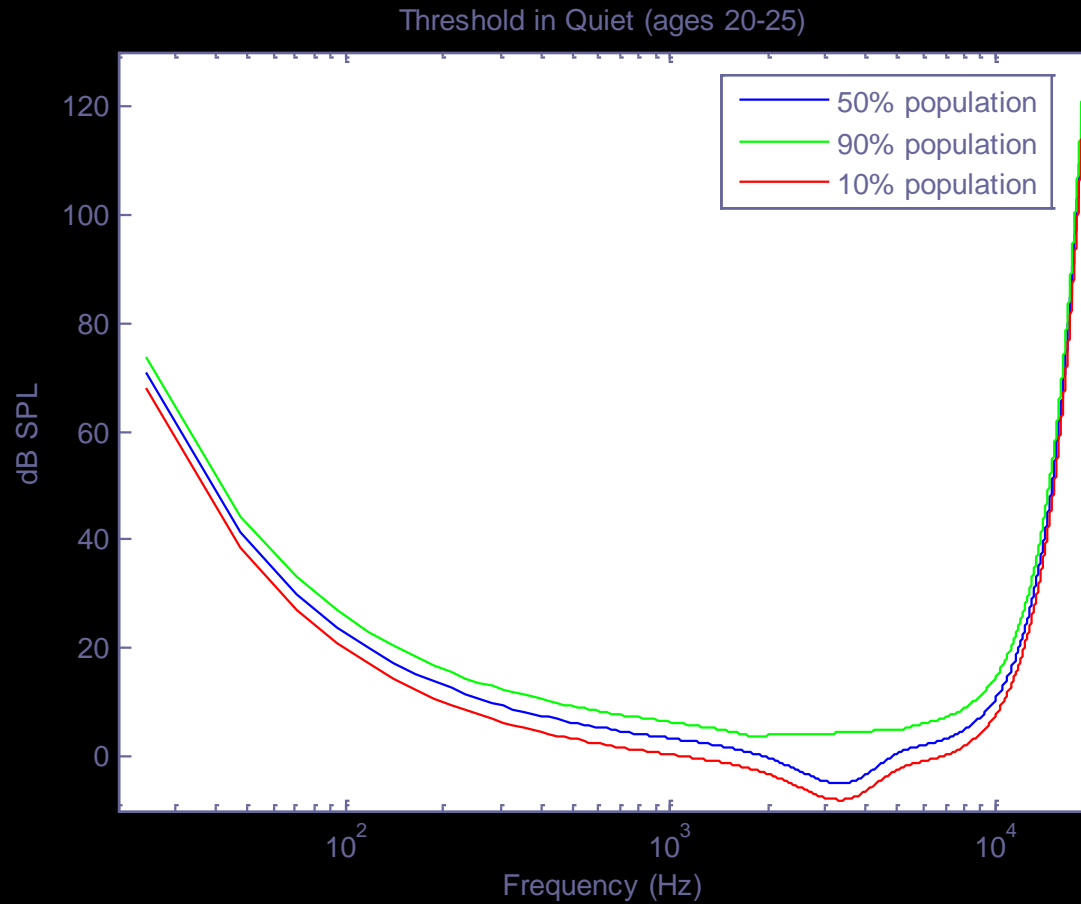
GMUE '05

Oct 26 2006

Today's Topics

- Limits of hearing
- Filters, Phase, and Group Delay
- Mixing & Monitoring
- Stereo Imaging
- Jitter & other effects of too much caffeine

Threshold of Hearing



Adapted from Zwicker & Fastl

So we can't hear high freqs...

How about “sensing” them?

- Oohashi, et al. found that brain electrical activity and blood flow changed significantly when frequencies above 20kHz were present.
- Interesting note- it took about 20sec for the brain to respond to high frequencies.
- To my knowledge, this research has not yet been duplicated.

Filters, Phase, and Group Delay

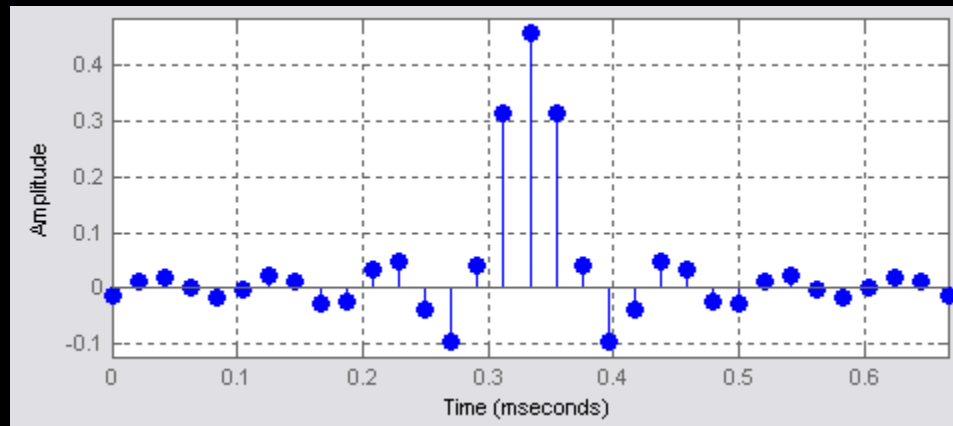
- Sidenote on temporal “smearing” / Pre-Echo
 - Frequency -vs- Time
- Phase / Group delay
 - Linear Phase / Constant Group Delay
 - Nonlinear / Frequency Dependent Group Delay

“A New Perspective on Decimation and Interpolation Filters”
by Steve Green (Cirrus Logic)

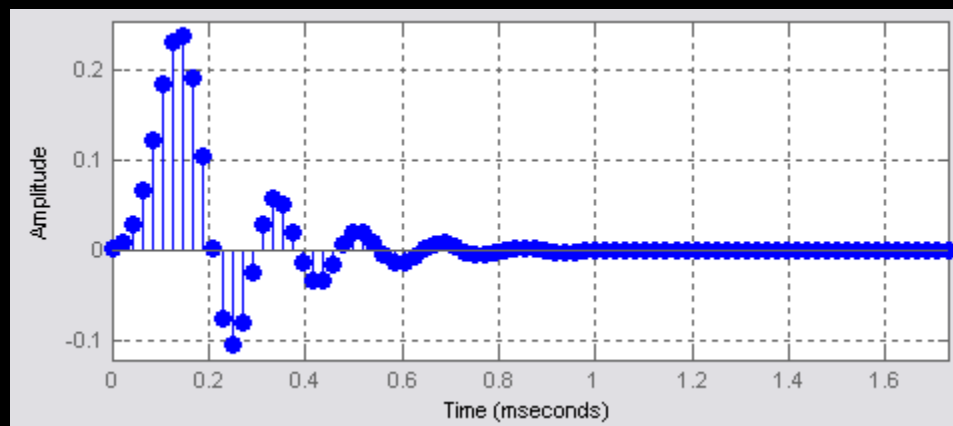
<http://www.cirrus.com/en/pubs/whitePaper/DS668WP1.pdf>

Pre-Echo

- Typical [linear phase] FIR filter response:

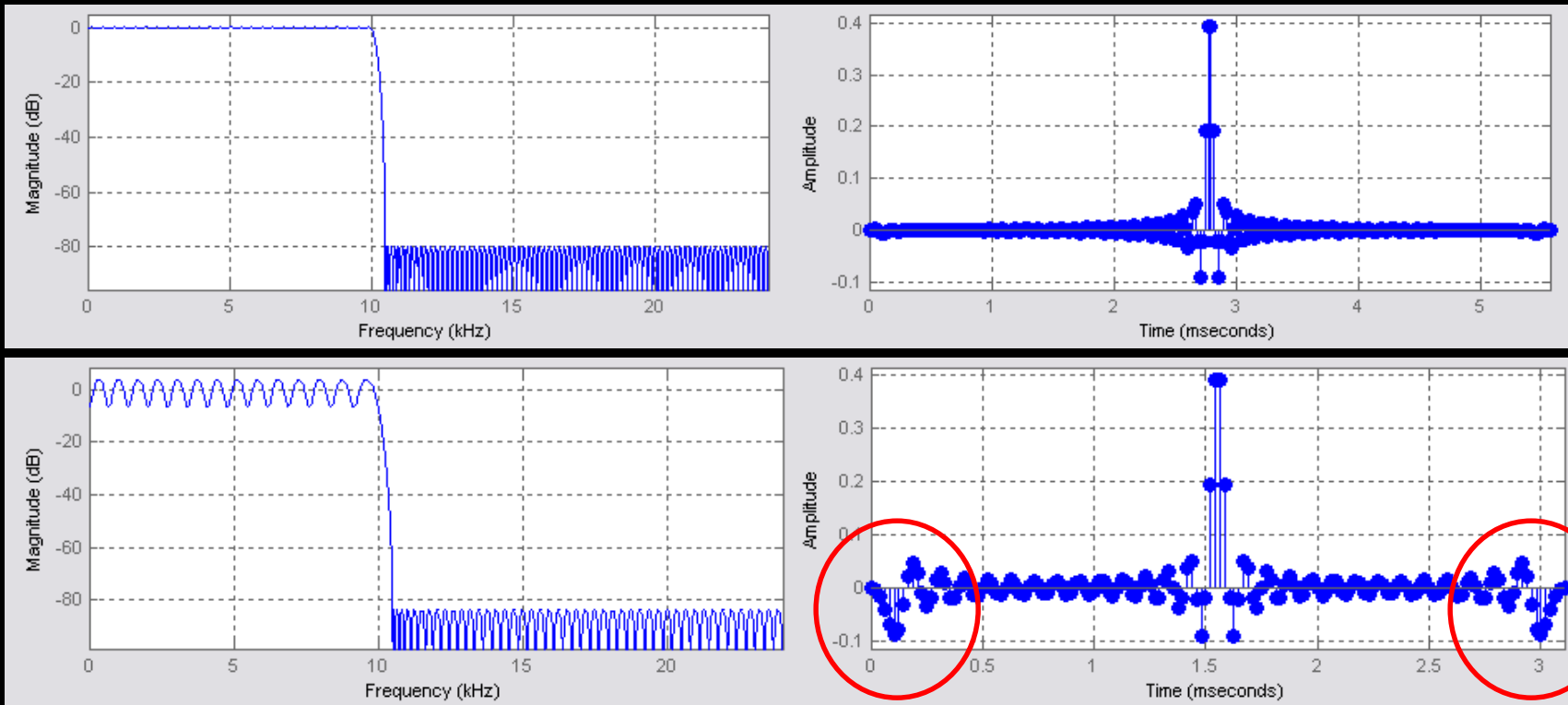


- Typical IIR filter response:

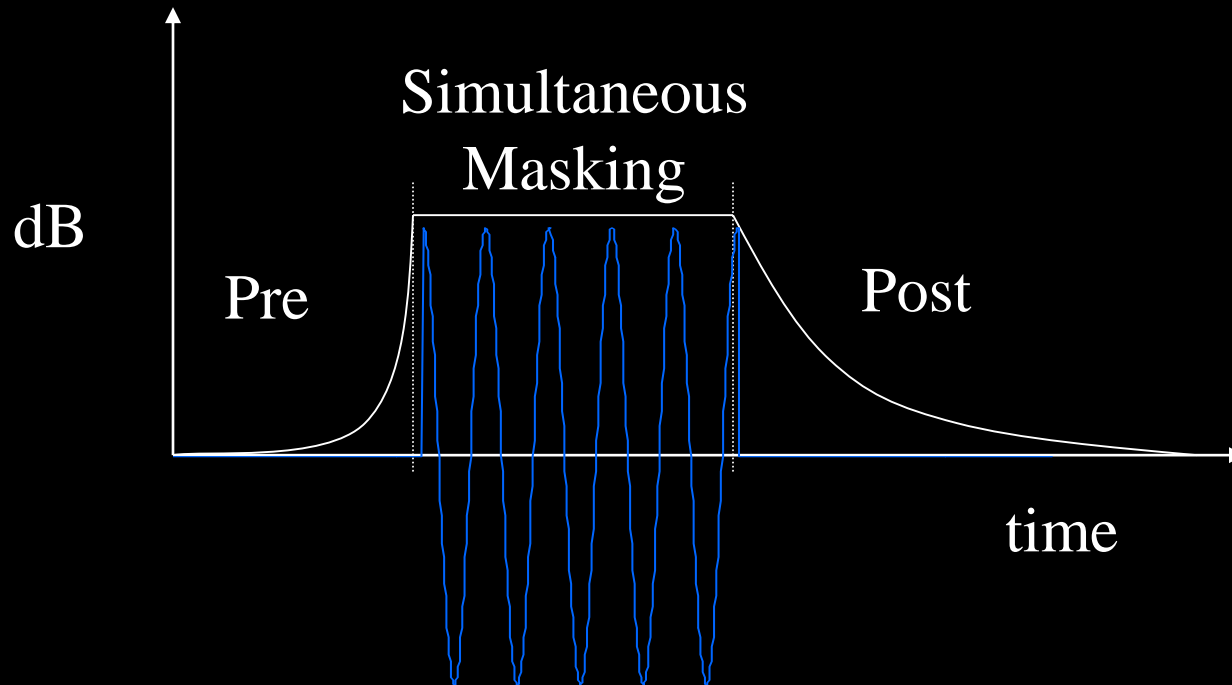
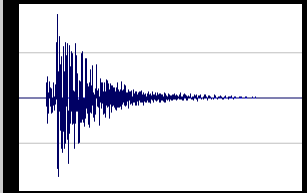
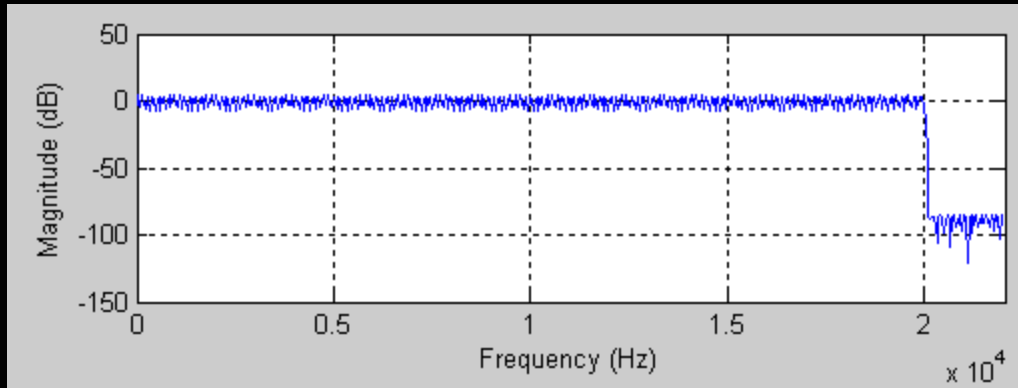
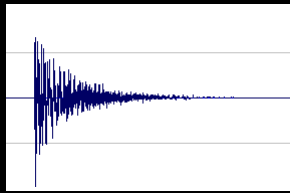


Pre-Echo

- Pass-band ripple can result in pre-echo
 - Sine wave in frequency domain = Impulse in time domain

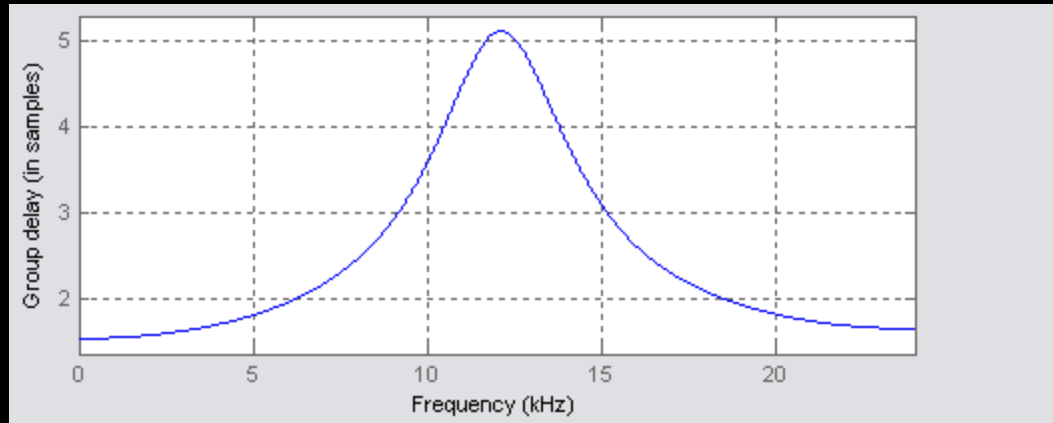


Pre-Echo



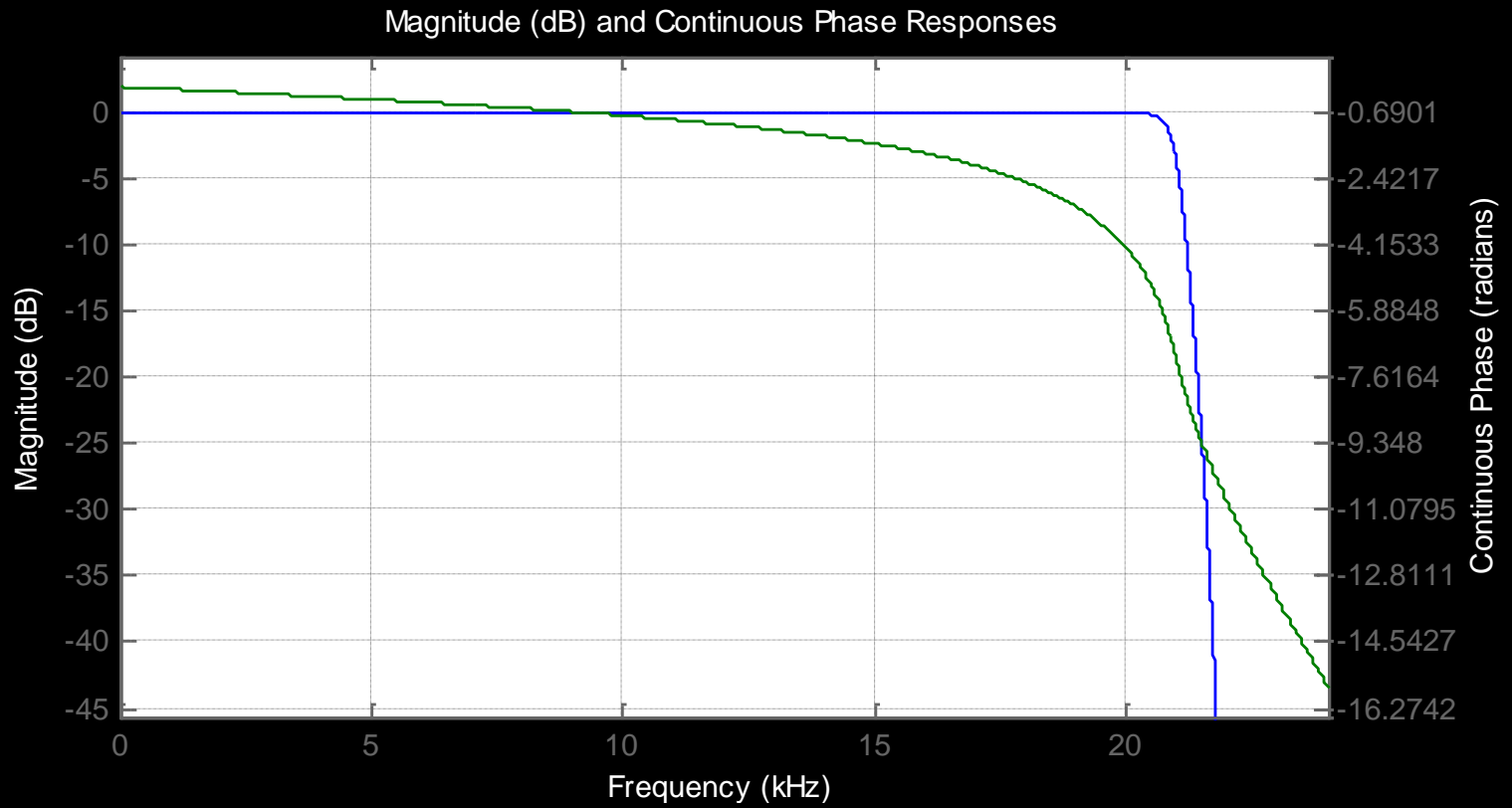
Phase Linearity

- Linear vs. non-linear phase (and resulting group delay)



- Anti-aliasing filter delay (linear or near-linear phase)
 - At 48kHz, 12 samples = $250\mu\text{s}$
 - At 192kHz, 12 samples = $62.5\mu\text{s}$
- Non-linear phase is not just in digital filters... also in analog filters and mechanical systems! This is where the argument for speakers with extended frequency response comes into play.

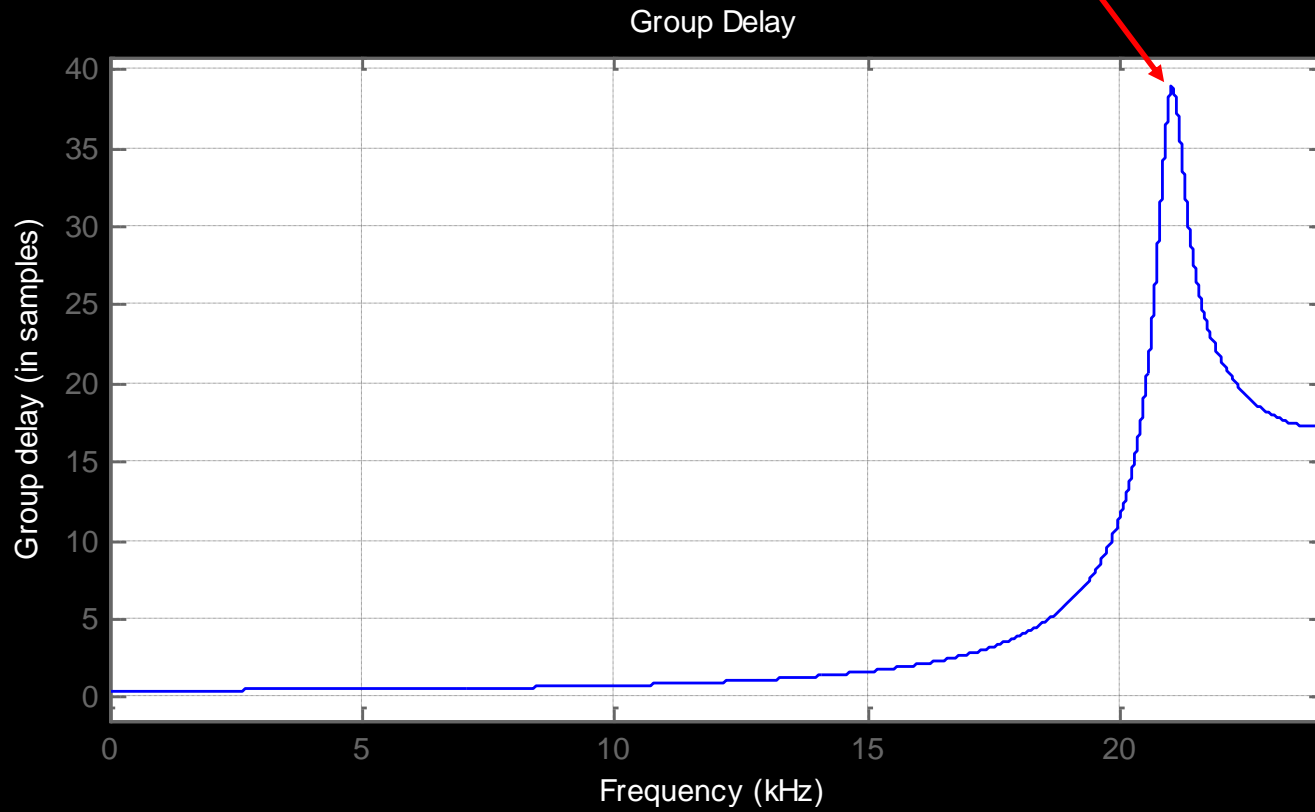
Filters



$F_s=48\text{kHz}$

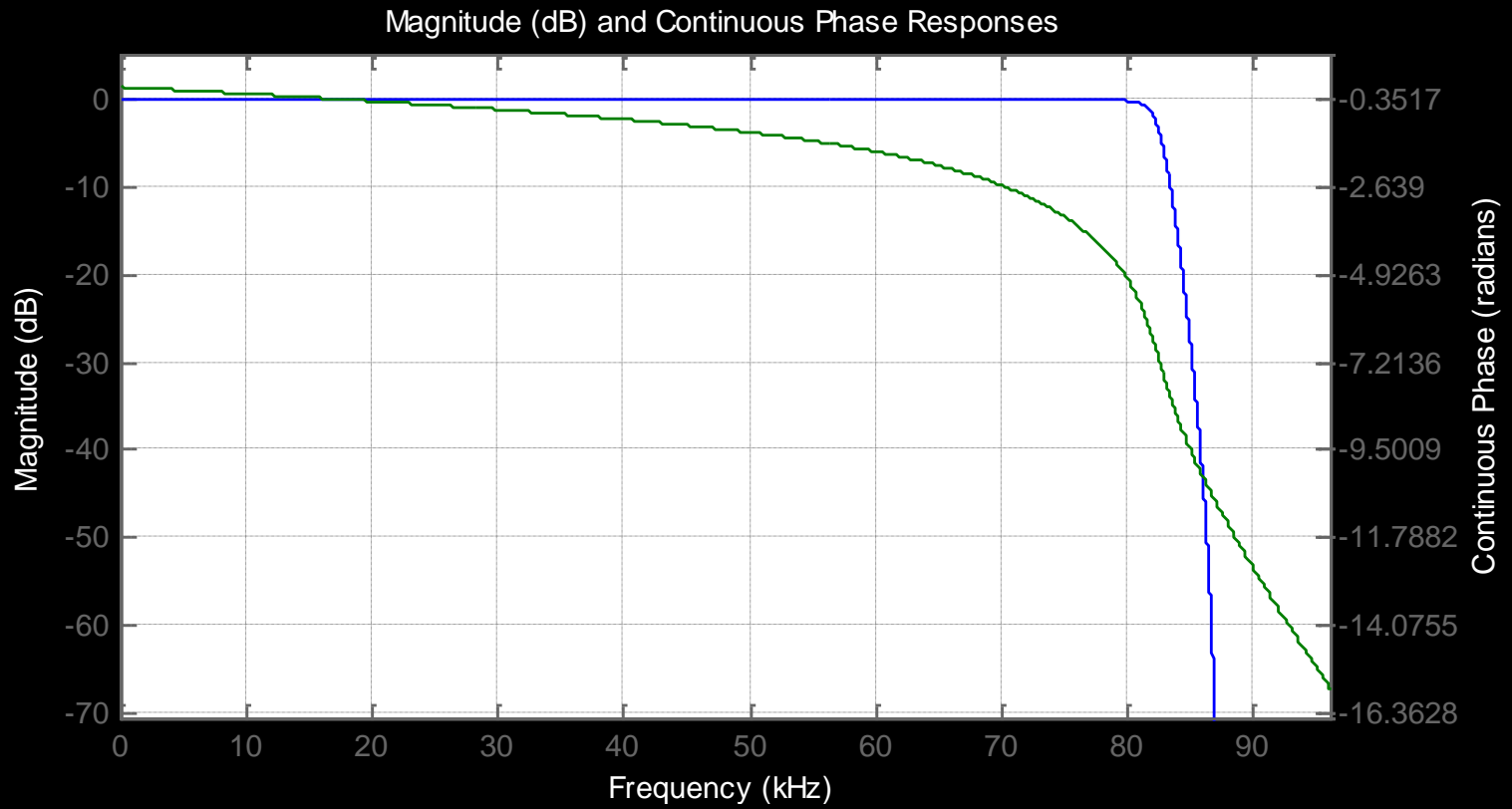
Filters

< 1.0ms of ringing @ F_c



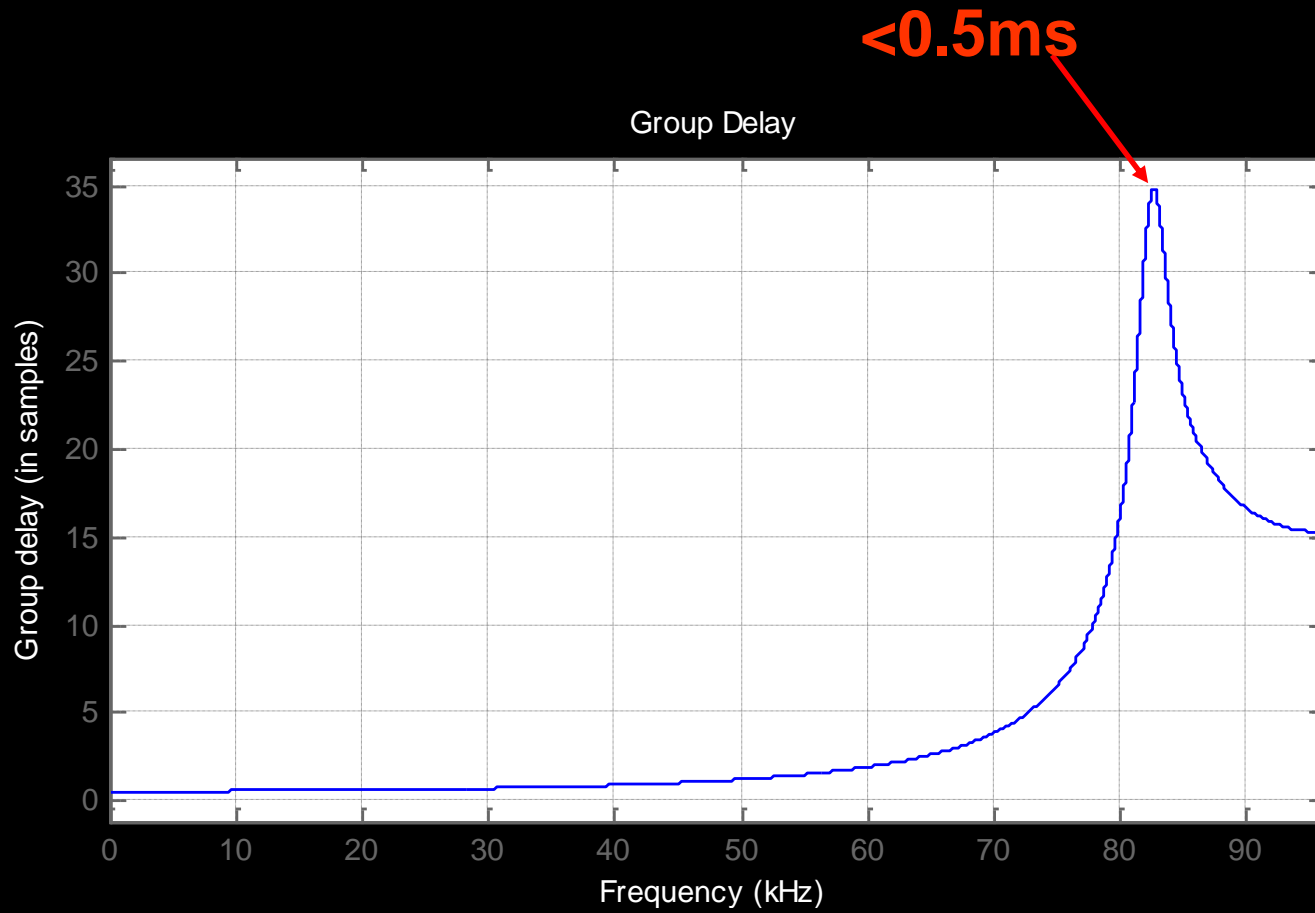
$F_s=48\text{kHz}$

Filters



$F_s = 192\text{kHz}$

Filters

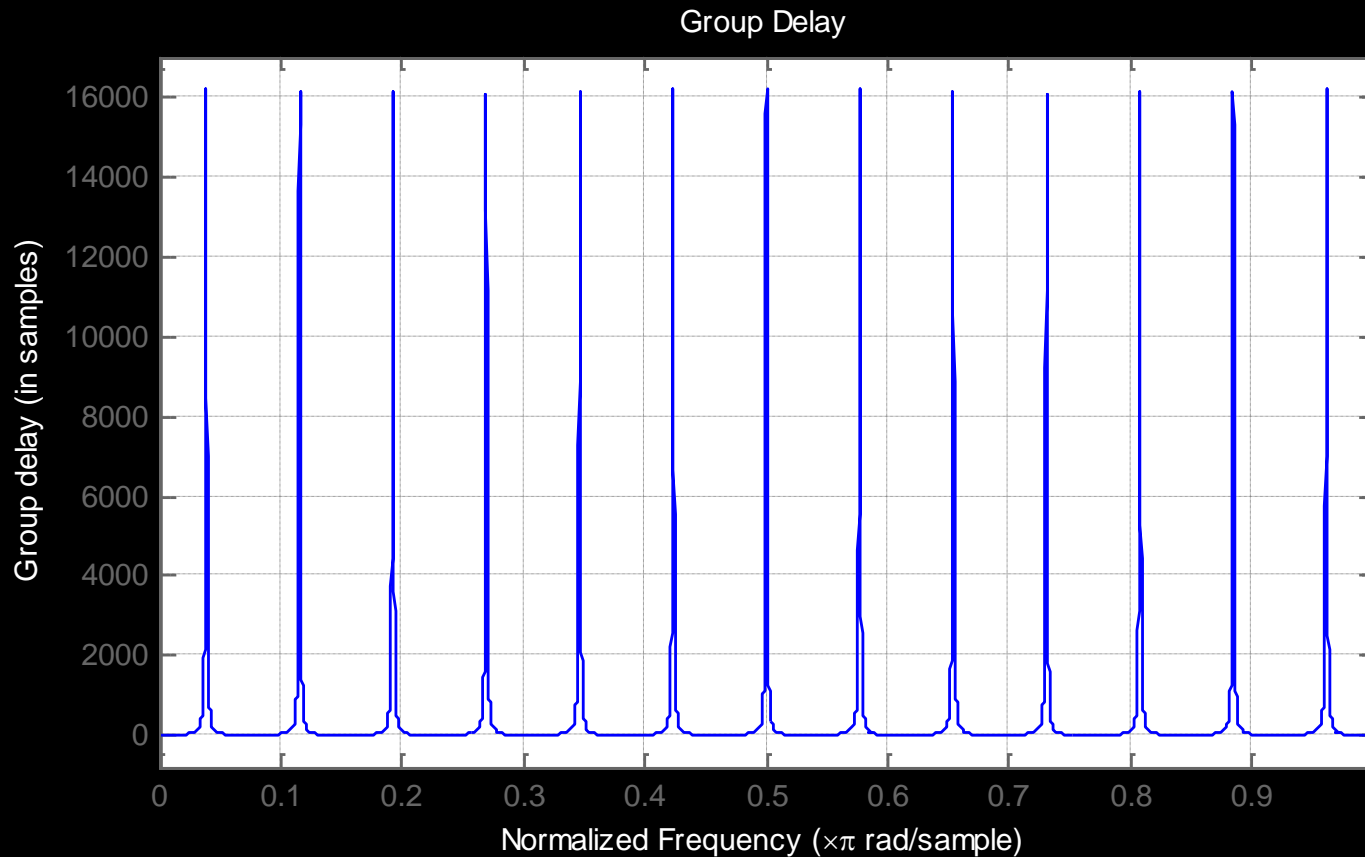


<0.5ms

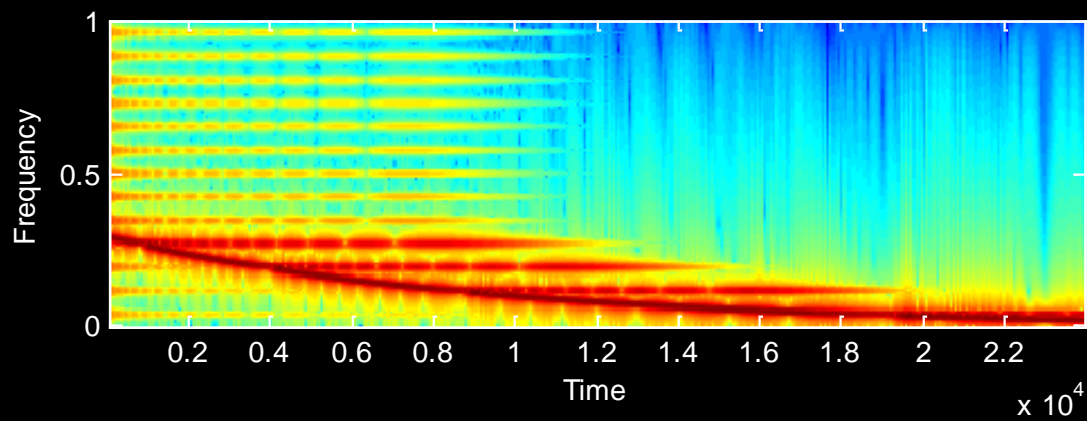
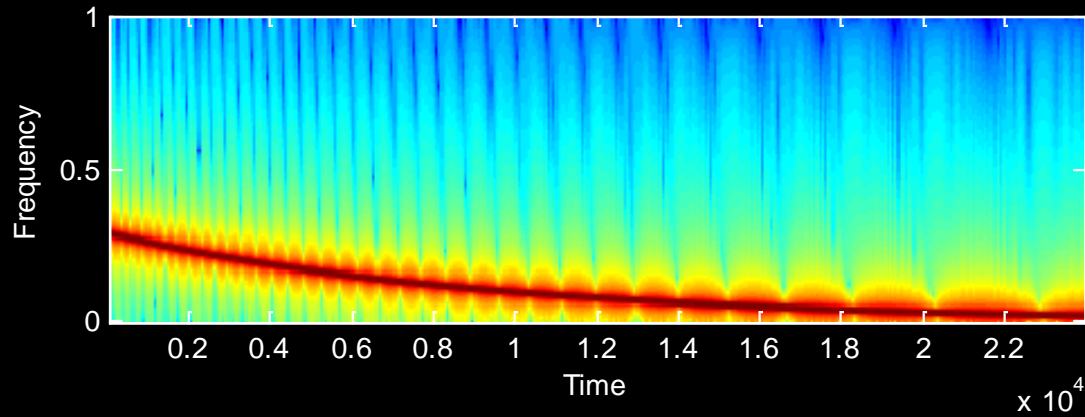
$F_s = 192\text{kHz}$

Severe Group Delay

To make it really audible, let's increase the group delay at several frequencies...



Severe Group Delay



Mixing & Monitoring

- Here's where phase becomes critical...
 - You can't hear small phase nonlinearities of a mono signal in an anechoic environment... but in any real condition, you can, especially when you start

mixing!

pink



allpass



2*pink






pink+allpass

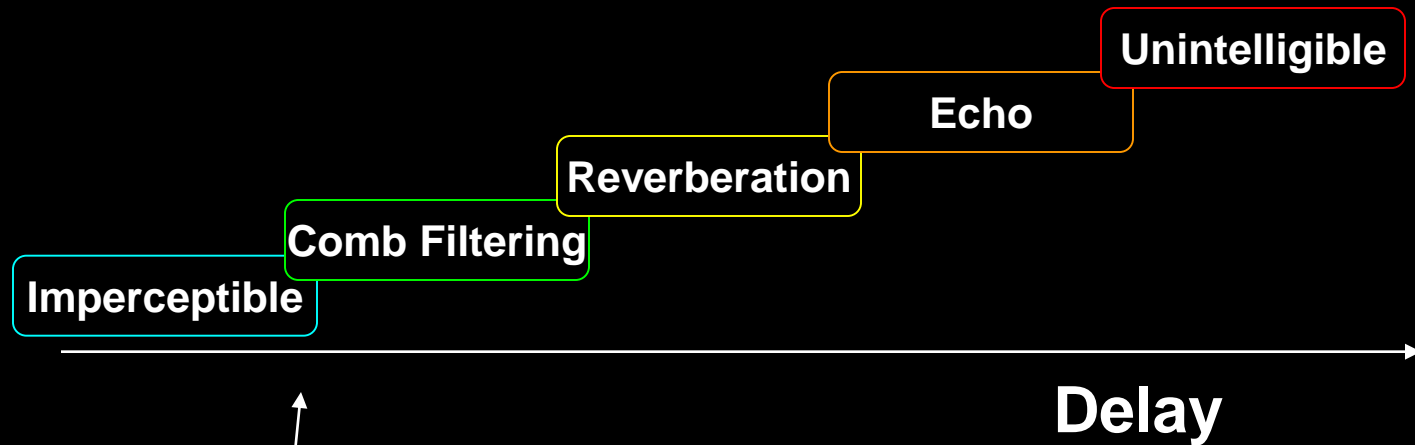


- Delay
 - How much delay is too much?

Delay

- Doesn't matter if you don't know it's been delayed (i.e. broadcast audio doesn't arrive instantly)
- But if you can hear the original and the delayed version mixed... comb filtering! 
 - But we hear varying degrees of this all the time
- What if we increase the delay?  

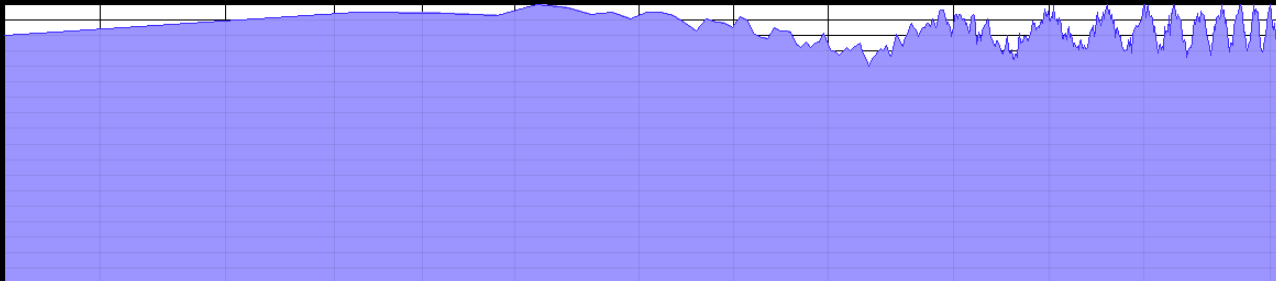
Monitoring with Delay



6 samples @ 48kHz = first comb filter null @ 4kHz
(latency due to ADC can be audible!)

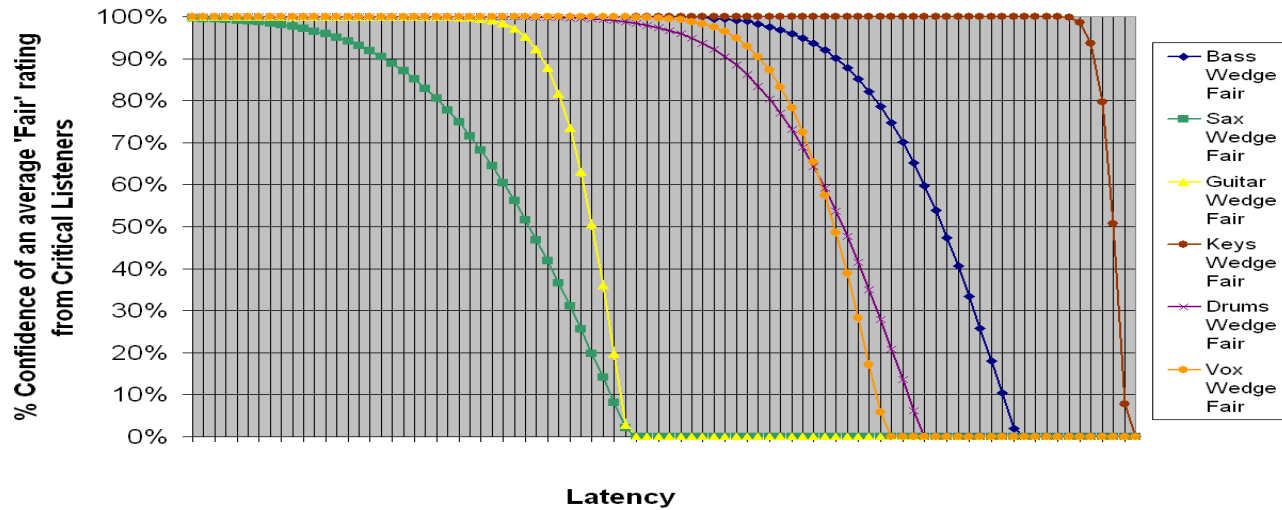
Monitoring with Delay

- If we have a null at 4kHz, the depth of this null is determined by the relative levels of the source and the processed sound
 - Source/Monitor Ratio = 0dB ... null = $-\infty$ dB
 - Source/Monitor Ratio = \pm big ... null = small
- The solution – to reduce comb filtering, either keep your monitor mix turned down, or use in-ear monitors with good isolation
- Note: This and following slide use constant group delay

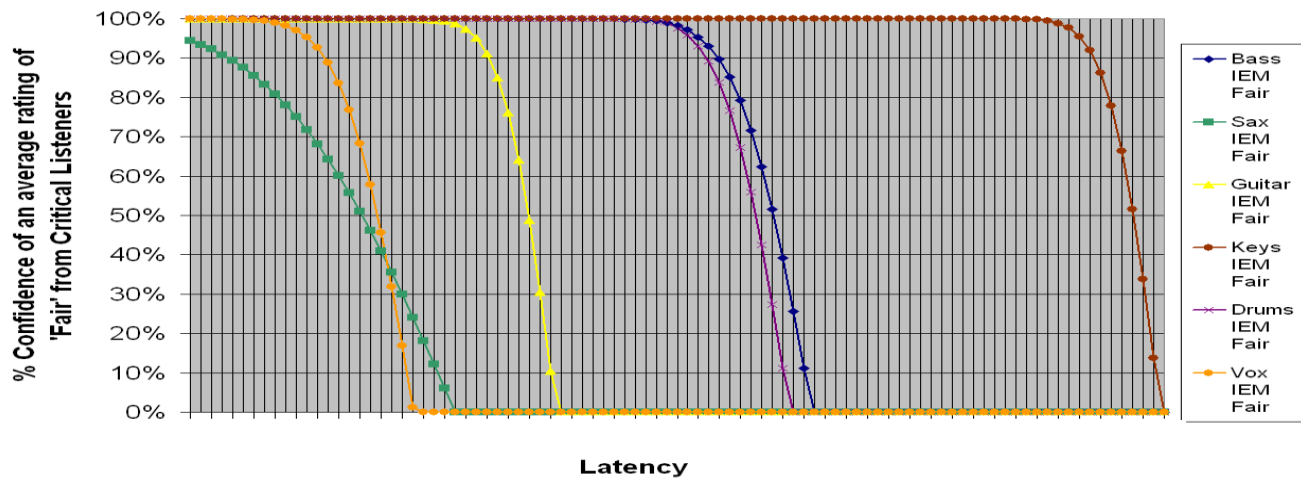


Monitoring with Delay

Instrument Comparison of Wedge with 'Fair' Rating

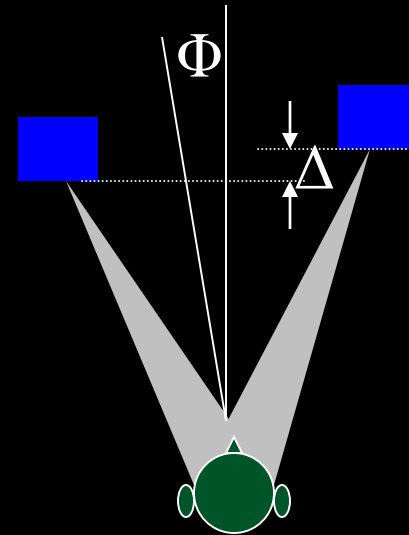


Instrument Comparison of IEM for 'Fair' Rating



Stereo Imaging

- This question seems to have first appeared around 1980 when some of Sony's designs used alternating samples for stereo converters.
- 48kHz
 - 20.8 μ s sampling period
 - $\Delta = 6.9$ mm per sample ; $\Phi = 1.9^\circ$
- 96kHz
 - 10.4 μ s sampling period
 - $\Delta = 3.5$ mm per sample ; $\Phi = 0.95^\circ$
- 192kHz
 - 5.2 μ s sampling period
 - $\Delta = 1.7$ mm per sample ; $\Phi = 0.47^\circ$



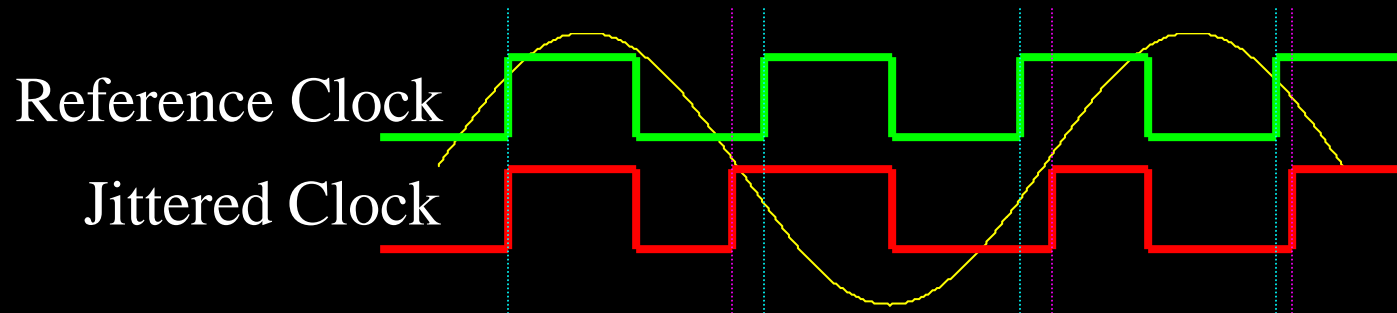
Stereo Imaging

- Most research shows that the JND for IPD is approximately $10\mu\text{s}$ (for clicks and/or very low frequencies)... although some research suggests as little as $2\mu\text{s}$ (Blauert)
- The timing of the ADC and DAC are the most critical factors here
 - Use the same converters for everything where possible (i.e. same anti-alias/anti-imaging filters)
 - Make sure they are based on the same clock signal
- In any case, these image shifts will be no larger than those caused by turning your head just slightly ($< 2^\circ$)

Why Higher Sampling Rates Are Worse

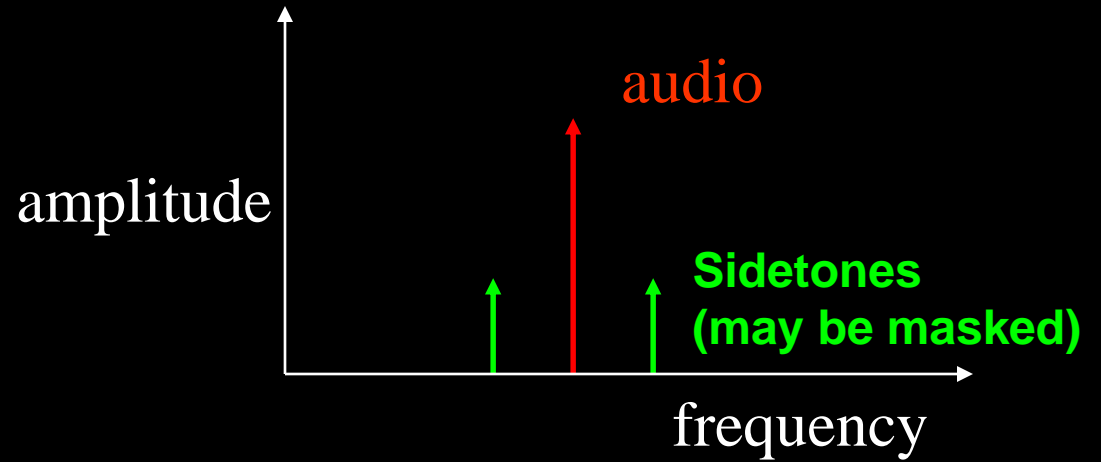
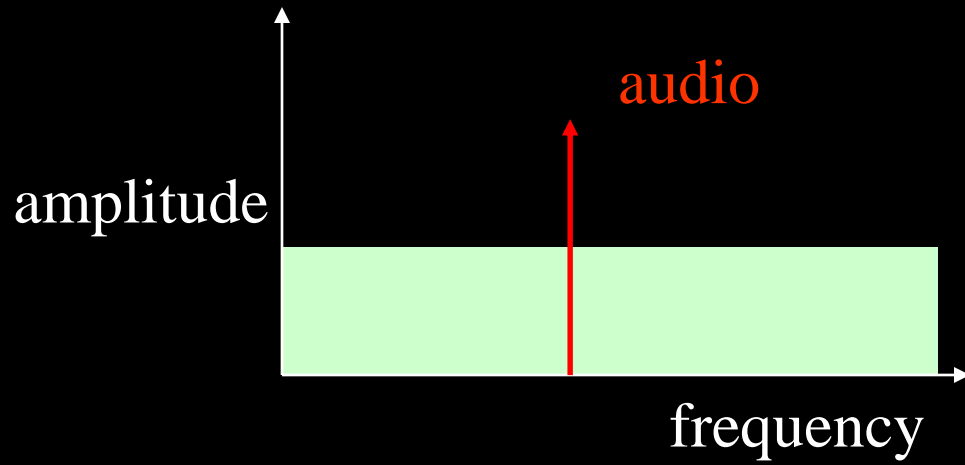
- Computational complexity & memory constraints
 - 192kHz requires 4x more processing than 48kHz
 - 30-band GEQ: 172 MIPS -vs- 43 MIPS
 - Buffers often must also be extended to 4x the length at 48kHz
 - 10ms (24bit audio): 46kb -vs- 11.5kb
- Jitter
 - Magically appearing sounds!

Jitter

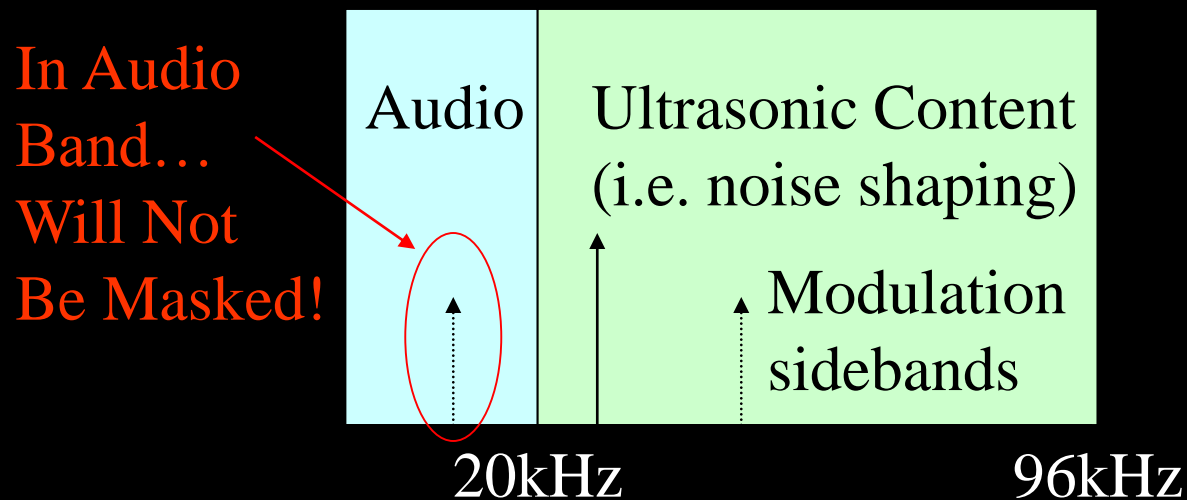


- Common Causes
 - Temperature variation of crystal
 - Inaccurate PLL for frequency/phase conversion
- Effects
 - Degraded audio quality (THD+N)... especially if ADC clock is bad!

Jitter Spectra



Jitter @ $F_s=192\text{kHz}$



Julian Dunn, Audio Precision Application Note #5: Measurement Techniques for Digital Audio, Published by Audio Precision, Inc., 2004.

Summary

- Audible differences between 44.1, 48, 96, 192kHz sampling rates are all due to issues other than Nyquist itself.
- For better sound, use better anti-alias/anti-imaging filters or increase sample rate... Both methods produce similar results.
- Be careful with phase nonlinearities in your audio band.
- Sample rate *may* affect stereo imaging, but probably not much.
- Sample rate drastically affects computational requirements.
- Keep your clocks clean, especially when using high sample rates.

References

- Dunn, Julian. *Anti-alias and anti-image filtering: The benefits of 96kHz sampling rate formats for those who cannot hear above 20kHz*. 104th AES Convention, May 1998.
- Oohashi, et al. *Inaudible high-frequency sounds affect brain activity: hypersonic effect*. Journal of Neurophysiology 83: 3548–3558, 2000.
- Julian Dunn, *Audio Precision Application Note #5: Measurement Techniques for Digital Audio*, Published by Audio Precision, Inc., 2004.
- Blauert, Jens, *Spatial Hearing, Revised Edition*. MIT Press, Cambridge, MA. 1997.