Music and Hearing Aids

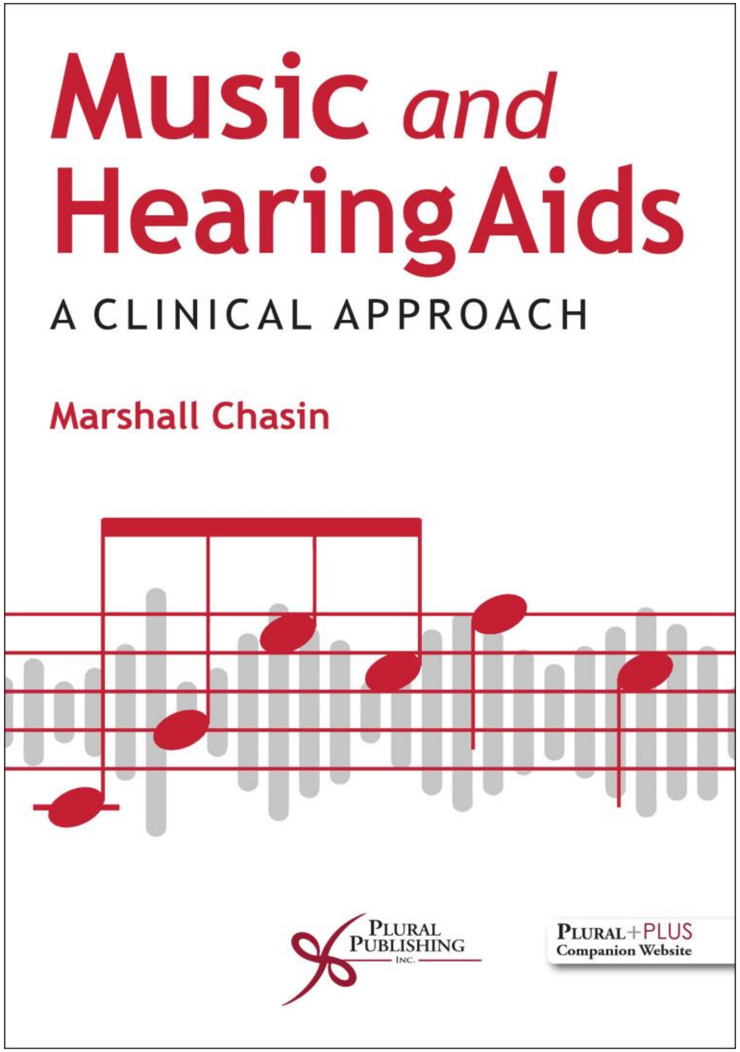
Marshall Chasin, AuD

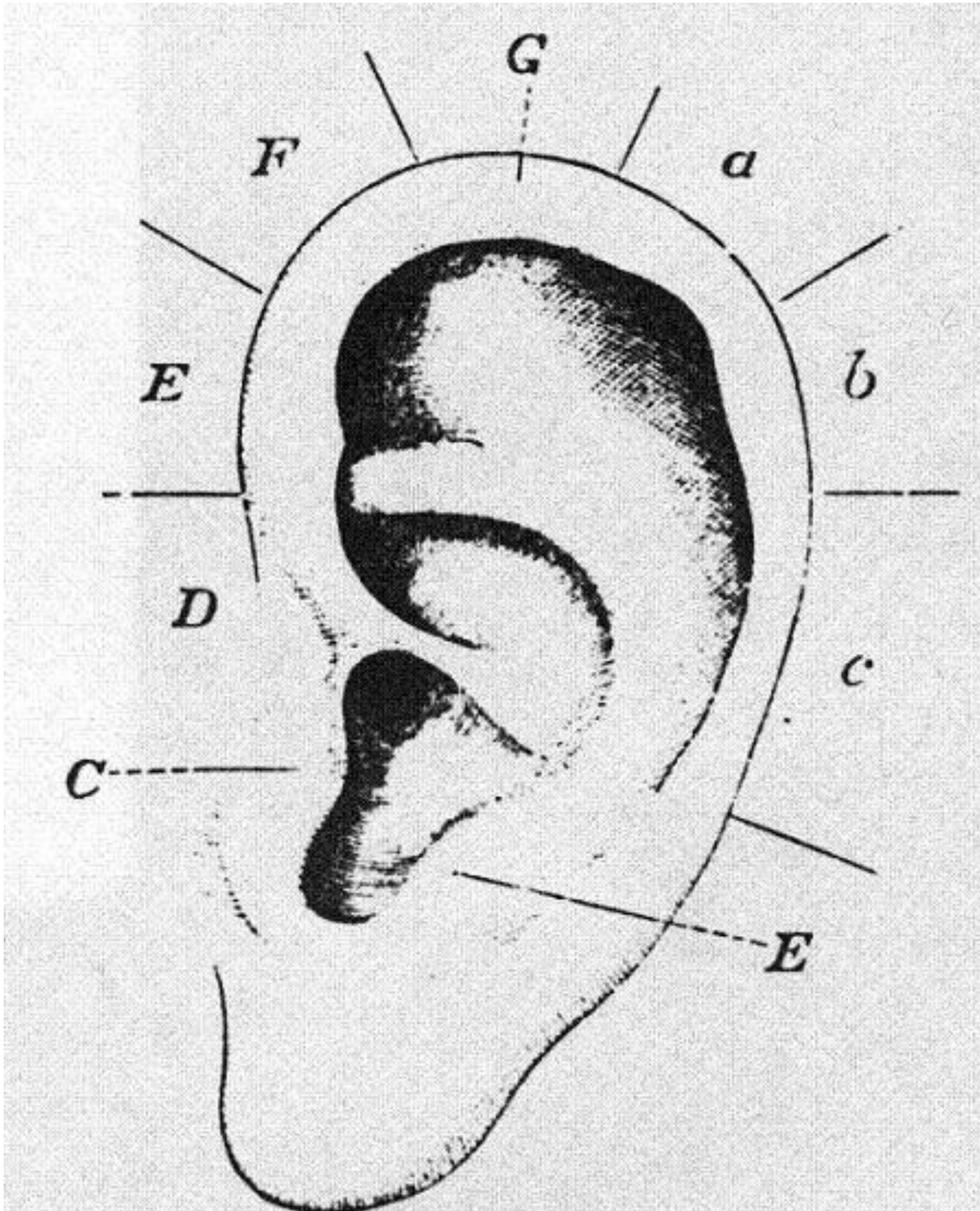
Musicians' Clinics of Canada

www.MusiciansClinics.com

CAA, Kelowna, BC

Conflict of Interest declaration…




G
F
a
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Four Differences ...

- Speech vs. Music Spectra
- Differing sound levels
- Crest factors
- Speech is narrow band

Four Differences …

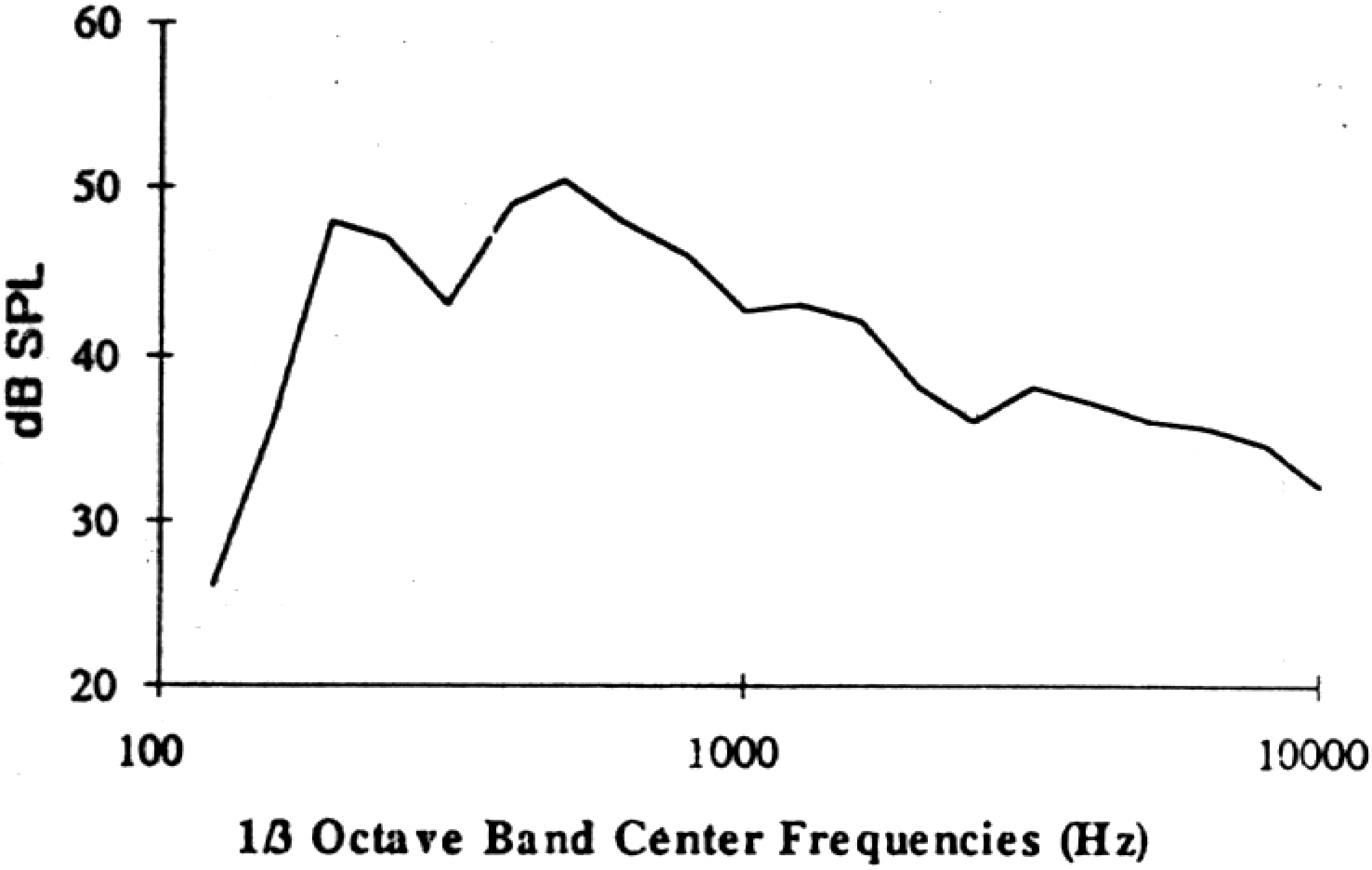
(1) Speech vs. Music Spectra:

Speech has a relatively uniform spectrum

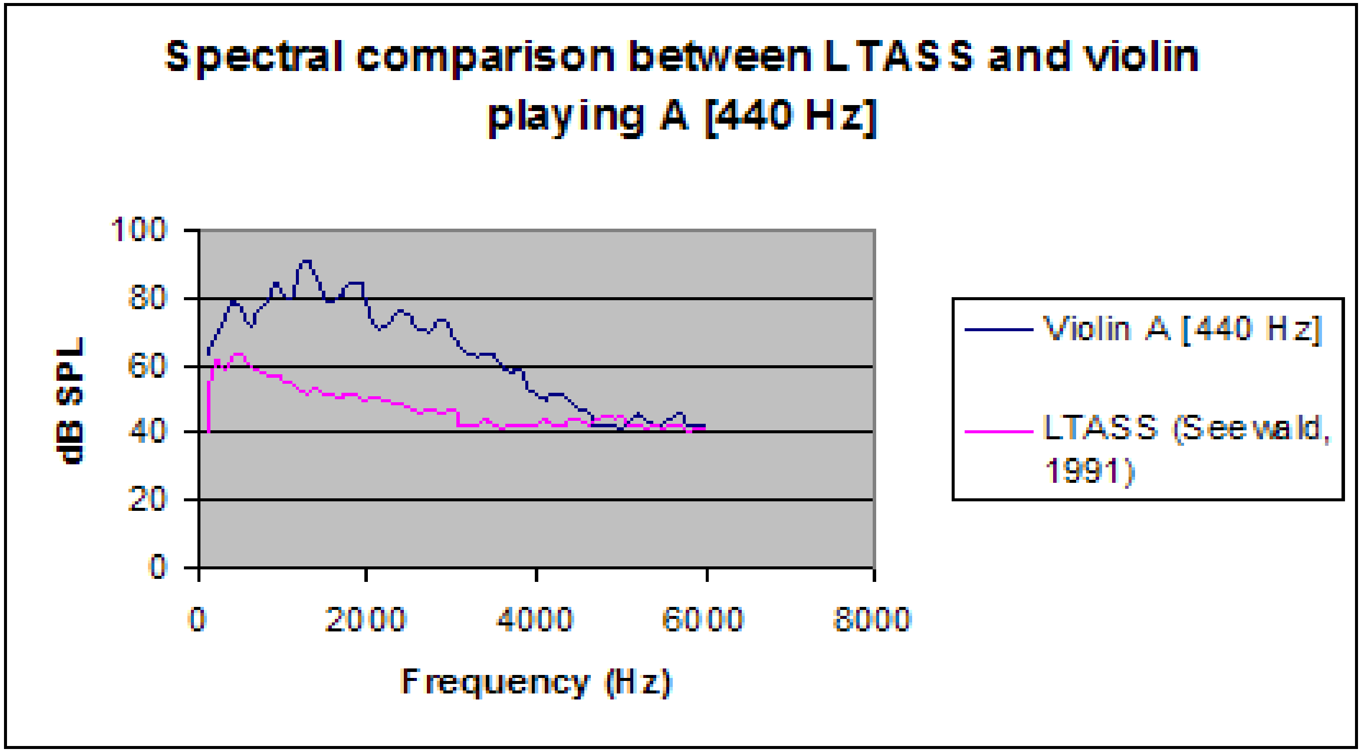
- Human vocal tract source
- Long-term speech spectrum "target"

Music has many sources

- Highly variable
- No "music target"



60
50
40
30
20
dB SPL
100
1000
10000
1/3 Octave Band Center Frequencies (Hz)


Spectral comparison between LTASS and violin playing A [440 Hz]
100
80
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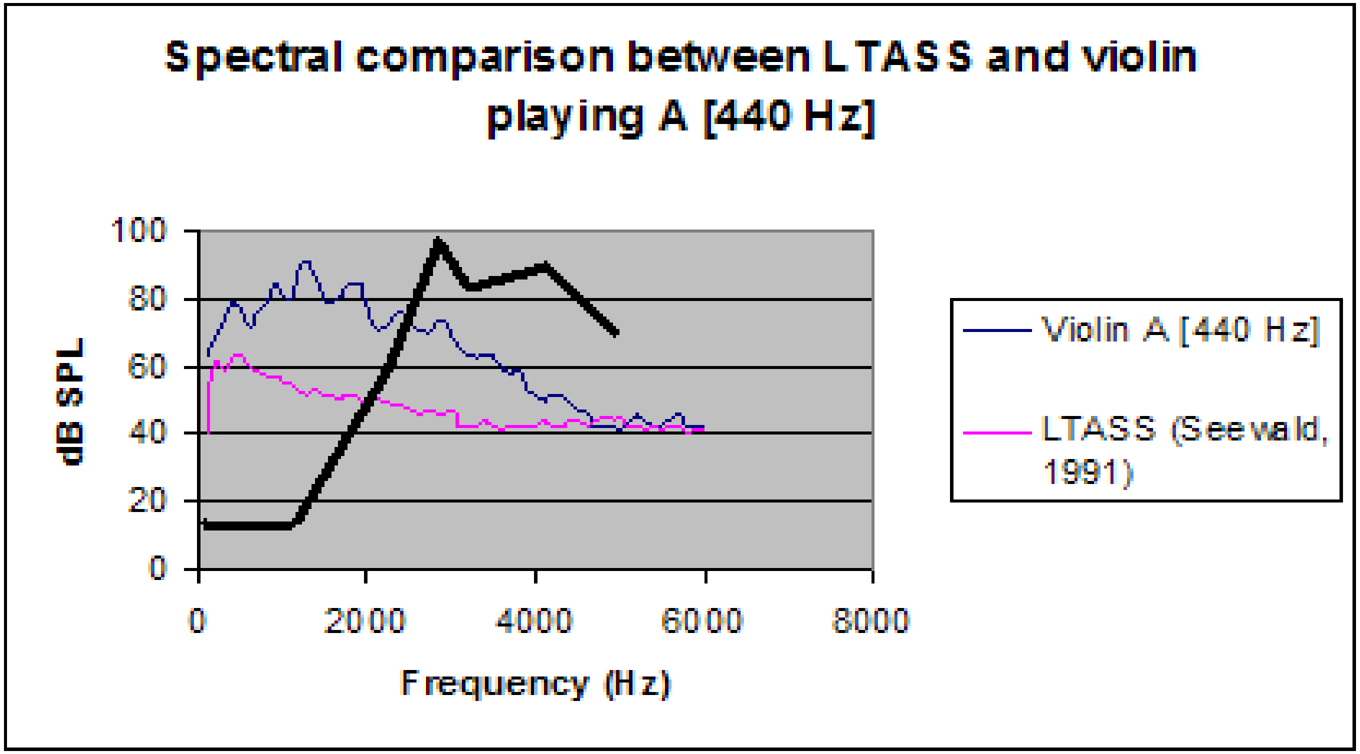
20
0
dB SPL
0
2000
4000
6000
8000
Frequency (Hz)
Violin A [440 Hz]
LTASS (Seewald, 1991)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100
80
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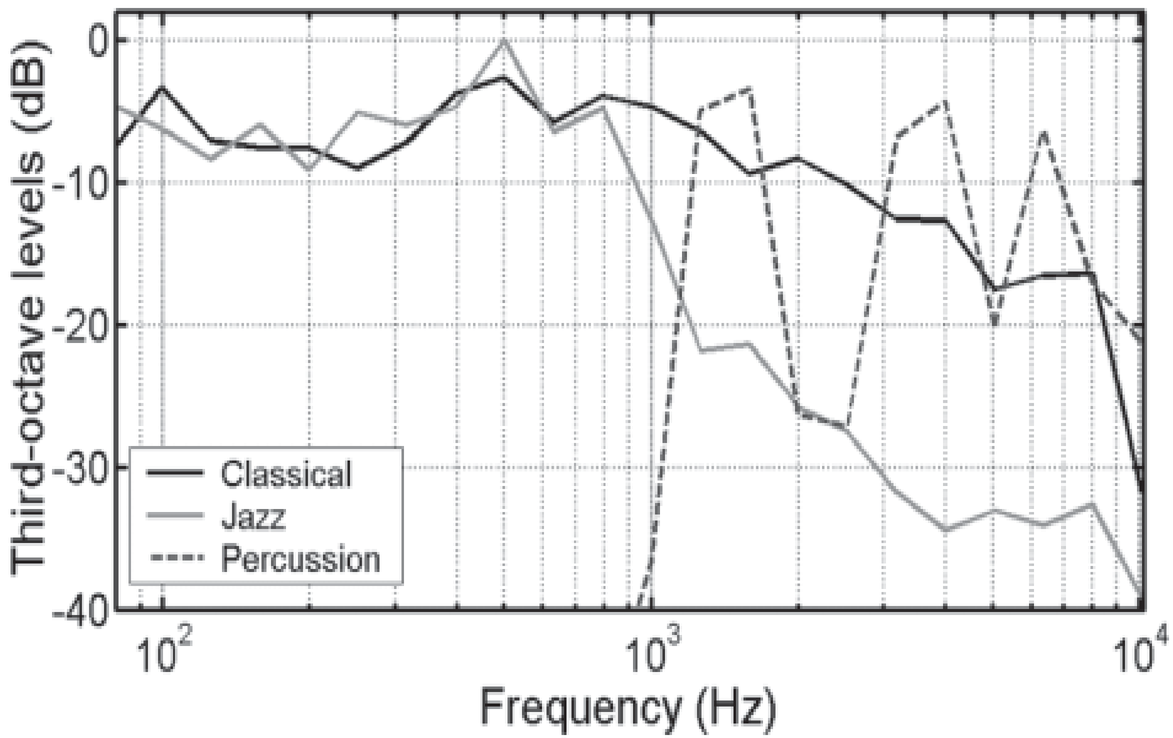
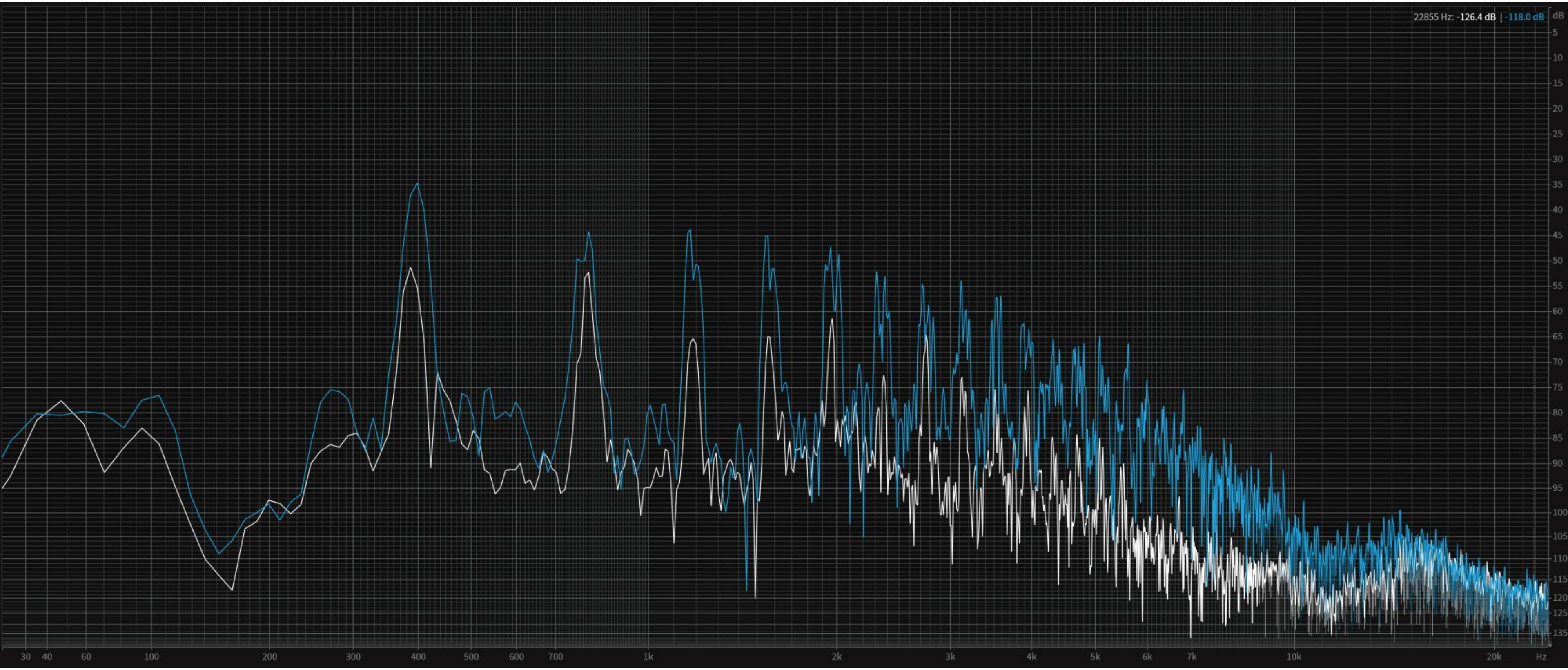
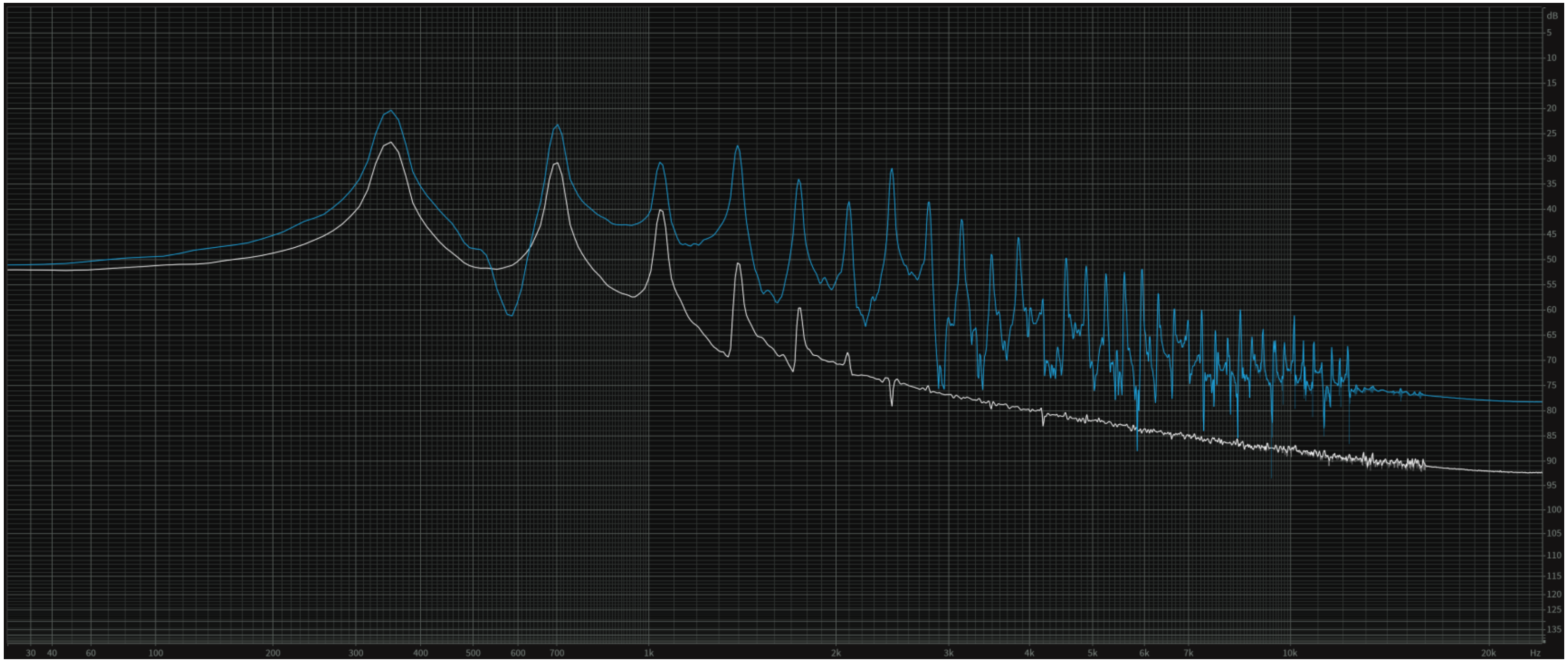


Moore, B. C. J. (2012). Effects of bandwidth, compression speed, and gain at high frequencies on preferences for amplified music. Trends in Amplification, 16(3), 159–172.

Violin (soft and loud)



French horn (soft and loud)



Spectral shapes differences (soft to loud)

	Low Frequency Region	*High Frequency Region*
Vocals	11-15 dB	0-10 dB
Stringed Instruments	11-15 dB	11-15 dB
Brass Instruments	< 10 dB	> 30 dB
Reeded Woodwinds	< 10 dB	> 30 dB

Four Differences …

(2) Differing sound levels:

Speech is 65 dB SPL ± 12 dB

- (53 dB SPL to 77 dB SPL)
- Shouted speech can be 82 dB SPL

Music can reach 105 dBA; peaks of 120 dB SPL

Table 2-1. Average sound levels of a number of musical instruments (from over 300 musicians) measured from 3 meters on the horizontal plane. *Also given is the sound level for the violin measured near the left ear of the player. (Chasin, 2006).

Musical Instrument	**dB(A) ranges measured from 3 meters**
Cello	80-104
Clarinet	68-82
Flute	92-105
Trombone	90-106
Violin	80-90
Violin (near left ear)*	85-105
Trumpet	88-108

Four Differences …

(3) Crest factor: (instantaneous peak – long term RMS)

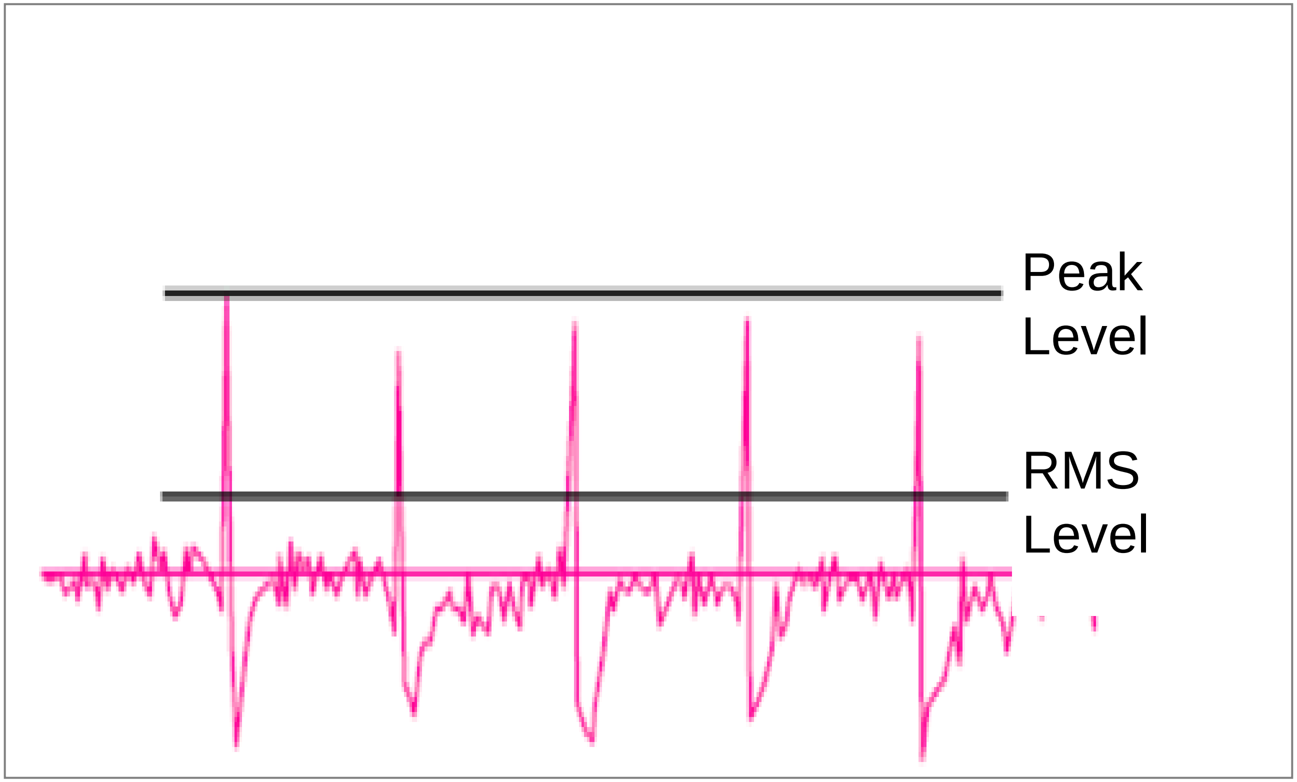
(Used in hearing aid testing… OSPL90 – 77 dB = Ref. Test Gain)

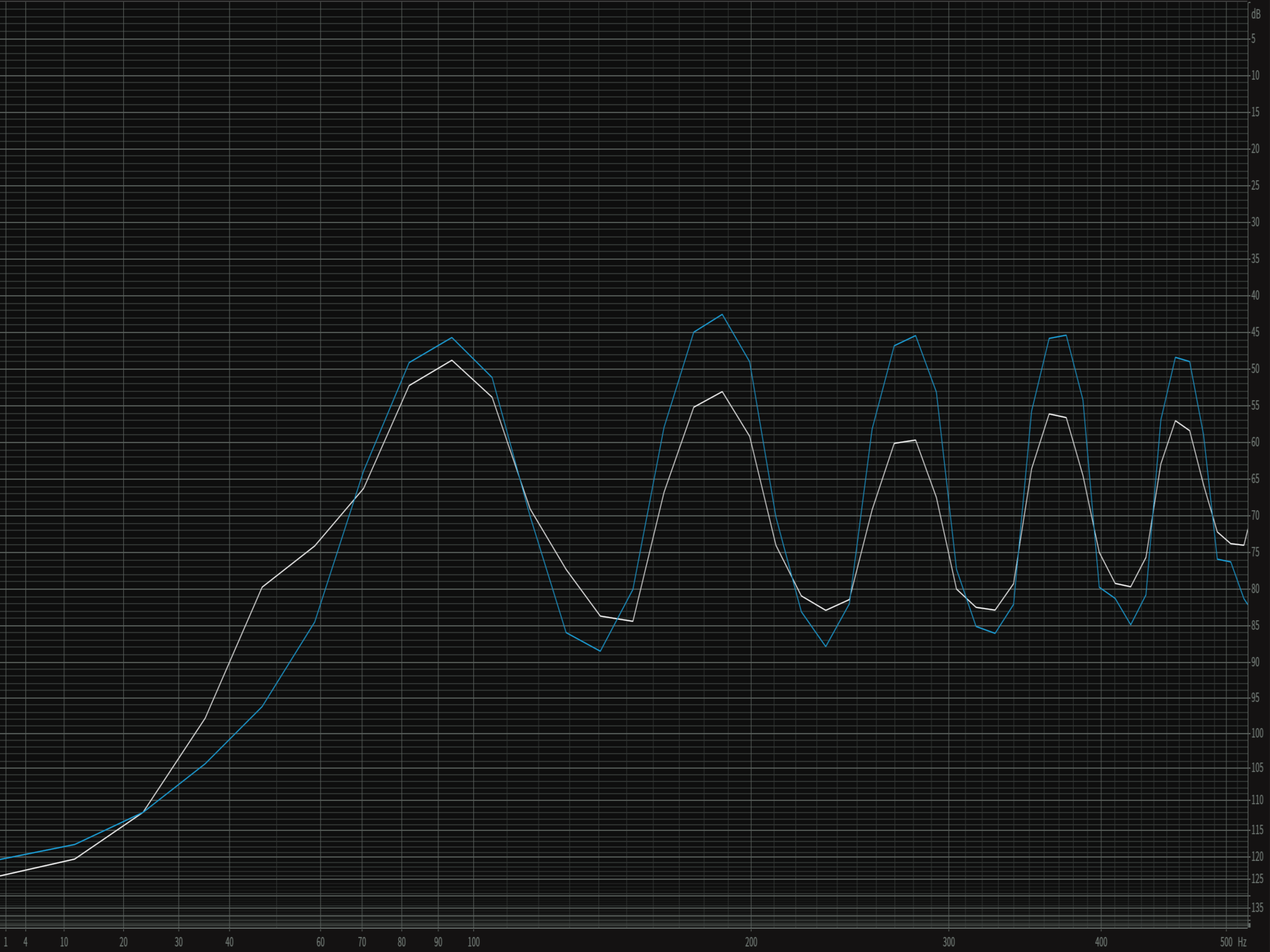
Speech has a crest factor of 12 dB- 16 dB

Music has a crest factor of 18 dB-22 dB

- Less damping.

Crest factor




dB
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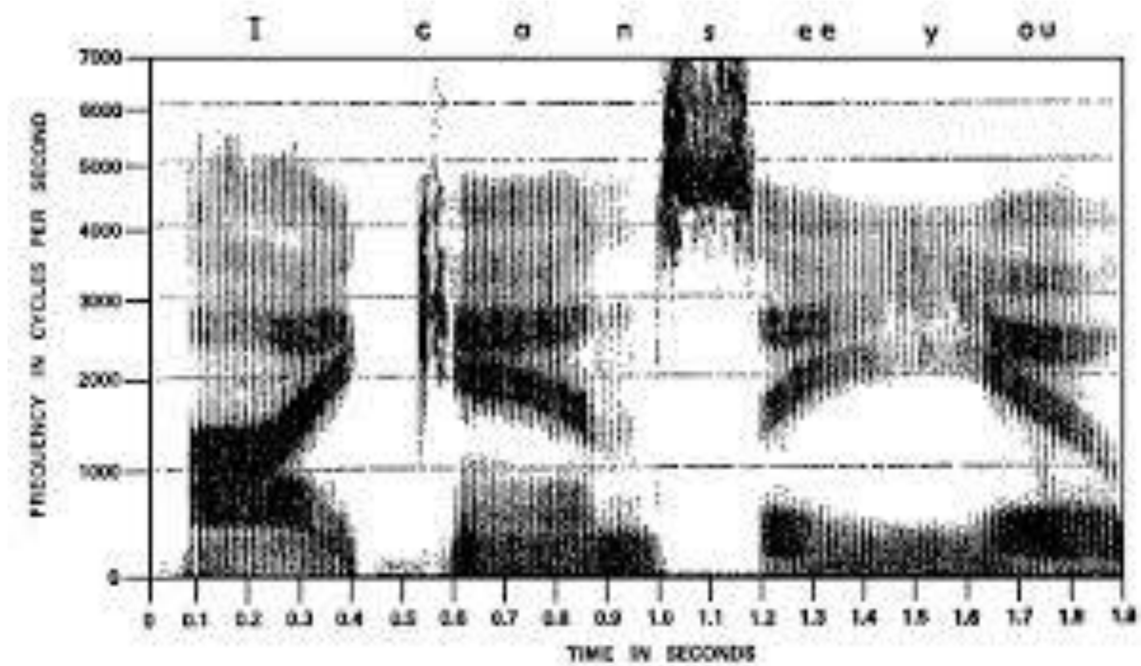
95
100
105
110
115
120
125
135
1
4
10
20
30
40
60
70
80
90
100
200
300
400
500
Hz

Four Differences …

(4) Speech is narrow band:

At any one point in time, speech is EITHER low frequency sonorants (e.g., vowels and nasals) OR high frequency obstruents (e.g., fricatives, siblants,…) but NEVER both

… Whereas music is always broadband…



Speech

65 dB SPL RMS

12-16 dB crest factor

-6 dB / octave

Well defined SII and target

Low frequency boost for louder speech

Speech is narrow band at any one point in time

Music

>100 dB SPL RMS

18-22 dB crest factor

Variable slopes

No “MII” and no target

Variable boosts for louder music

Music is always broadband

What about hard of hearing musicians …

… or non-musicians who like to listen to music?

Peak input limiting level

Peak input limiting level of many hearing aids (up until very recently) limits sound above 90 dB SPL.

(…Only 1 analog aid of the 1980s had 115 dB)

…. shouted [a] is about

82 dB SPL

Peak input limiting level

This occurs just after the microphone, and is related to the A/D converter.

- Overloading the "front end".

If distortion occurs this early in the circuitry, then nothing later (e.g. software adjustments) can improve things.

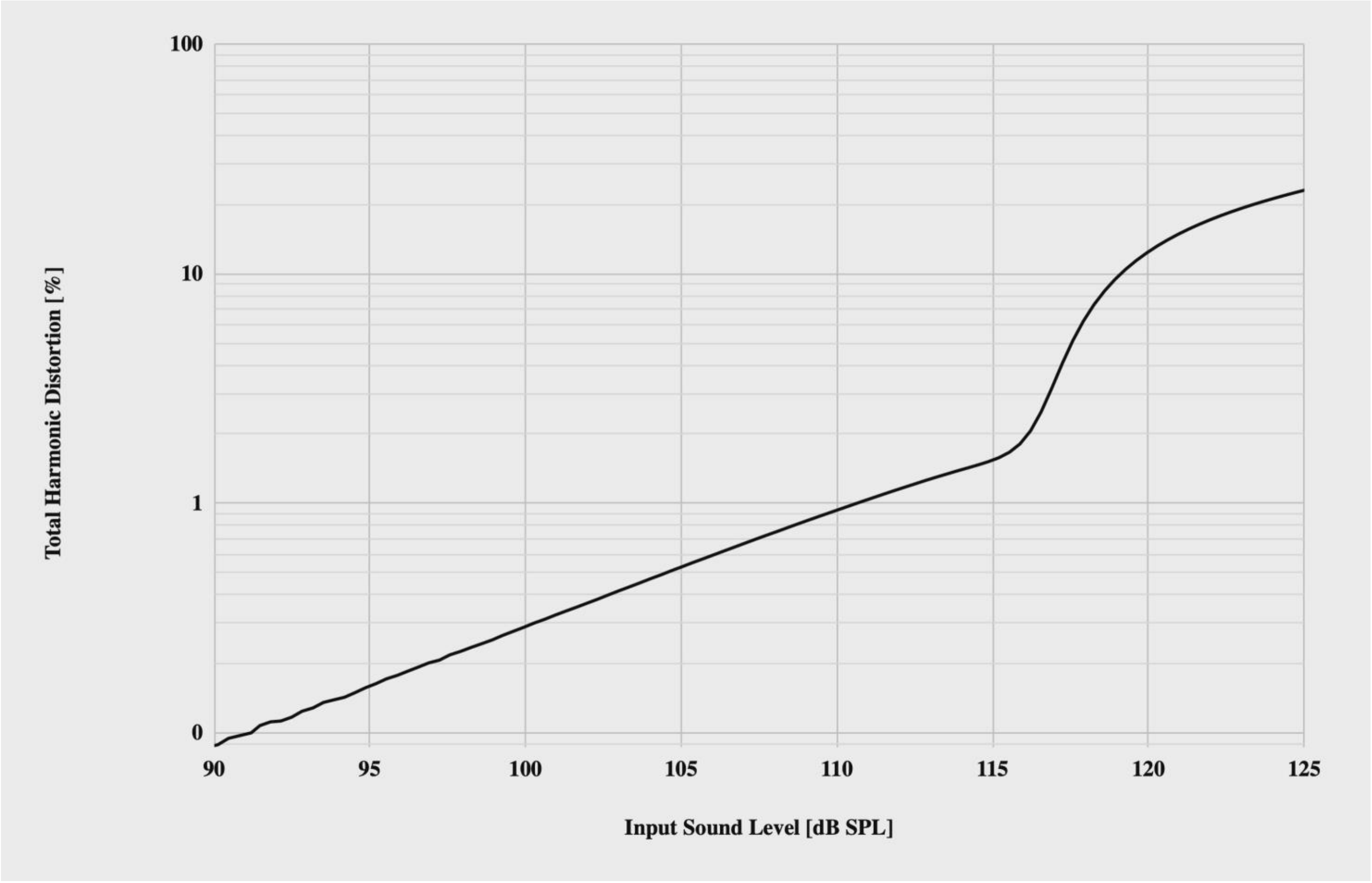
Chasin, M., & Russo, F. (2004). Hearing aids and music. Trends in Amplification, 8(2), 35–48.

Oeding, K., & Valente, M. (2015). The effect of a high upper input limiting level on word recognition in noise, sound quality preferences, and subjective ratings of real-world performance. Journal of the American Academy of Audiology, 26(6), 547–562.

Plyler, P., Easterday, M., & Behrens, T. (2019). The effect of extended input dynamic range on laboratory and field-trial evaluations in adult hearing aid users. Journal of the American Academy of Audiology, 30(7), 634–648.

Before we get to the nature of the music program(s), we need to take care of the "front end"…

Its not the microphone…



Max Headroom

The overpass analogy

80 dB
dynamic
range

High level music and low "ceiling"

An Experiment:

A hearing aid was constructed where the peak input limiting level can be successively reduced from 115 dB SPL, to 105 dB SPL, to 95 dB SPL … and back to 115 dB SPL.

Acknowledgments: Mead Killion, Russ Tomas, Norm Matzen, Mark Schmidt, Steve Aiken.

Speech

Music

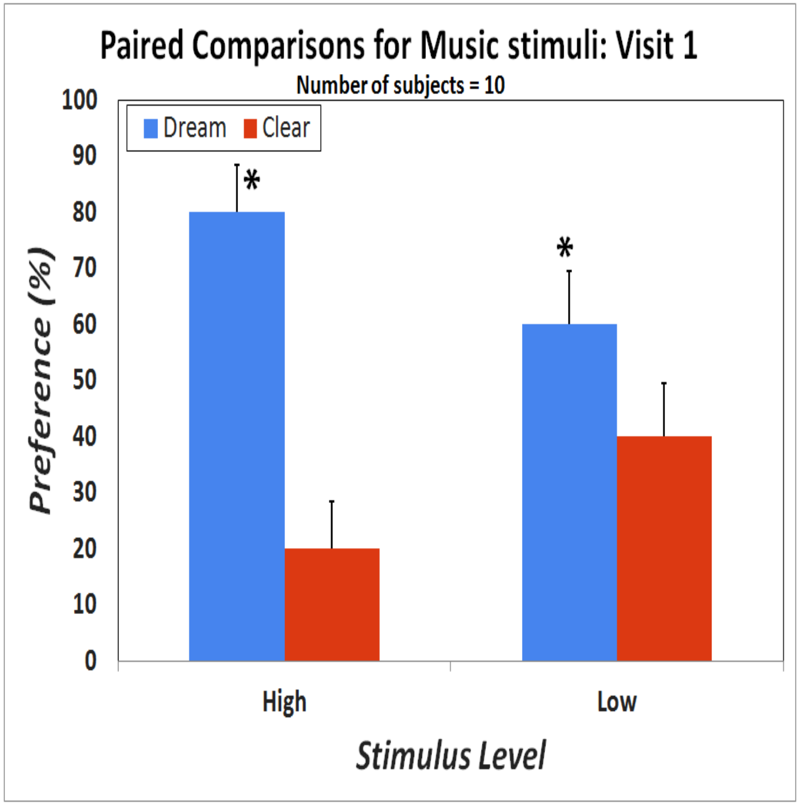
Therefore ….

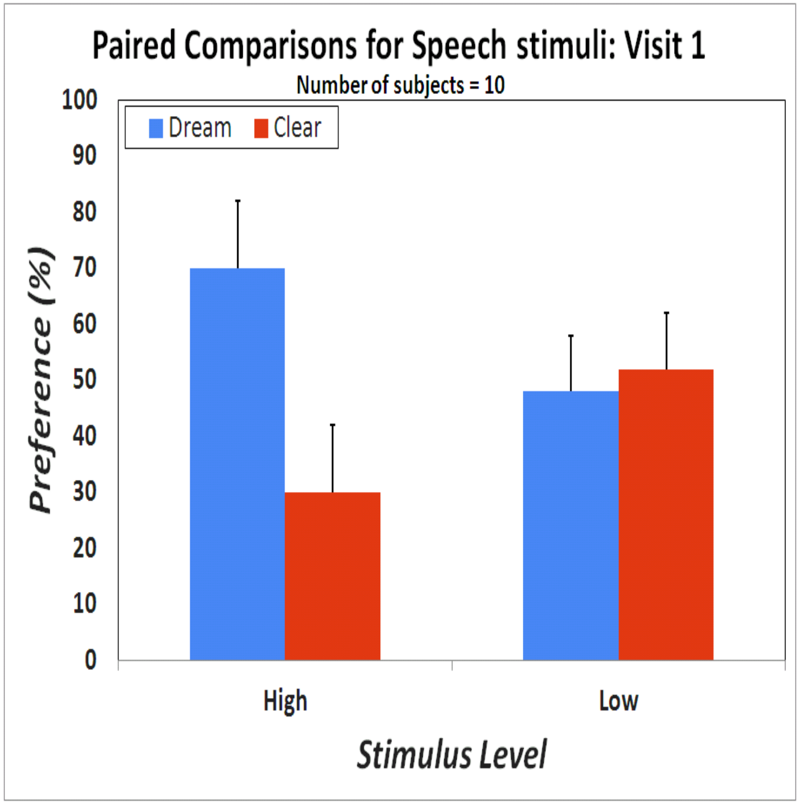
Peak Input Limiting Level should be at least 115 dB SPL …. Or maybe even up to the limit of modern hearing aid microphones ?

… And it's not just for music. What about the level of a hard of hearing person's own voice at their hearing aid?

Chasin M. The "best hearing aid" for listening to music: Clinical tricks, major technologies, and software tips. Hearing Review. 2014;21(8):26-28.

Benefit even for high level speech…





Chasin, M. :A hearing aid solution for music listening", Hearing Review, 2014: 21 (1), 28-30.

What the crest factor can tell us about speech…

A hard of hearing person’s own voice can overdrive their own hearing aid!

84 dB input + 16 dB crest factor = 100 dB ish

Four clinical strategies…

* Lower volume on stereo or other input and increase gain on aid.

* You can use an ALD (WITH A VOLUME) as input.

* Use (creative) microphone attenuators such as Scotch tape. (3-4 layers of tape will provide 10-12 dB of flat attenuation up to 4000 Hz).

* Take off the hearing aids

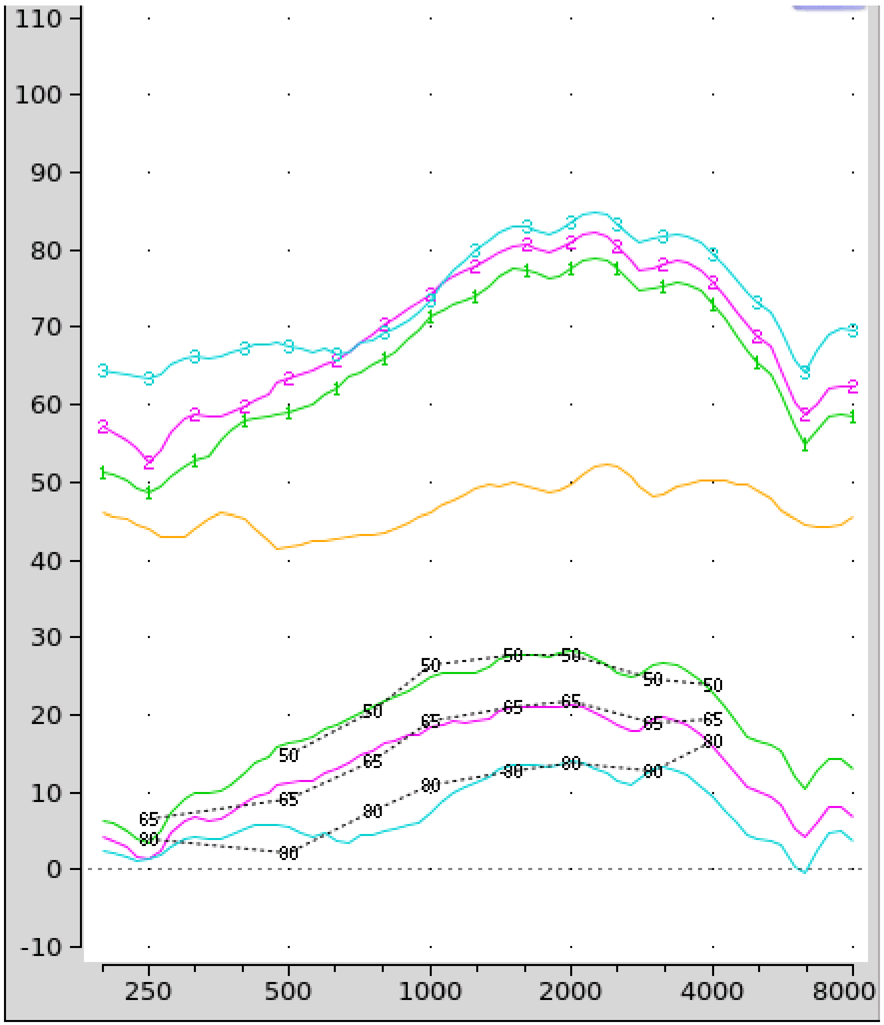
Use microphone attenuators such as Scotch tape.

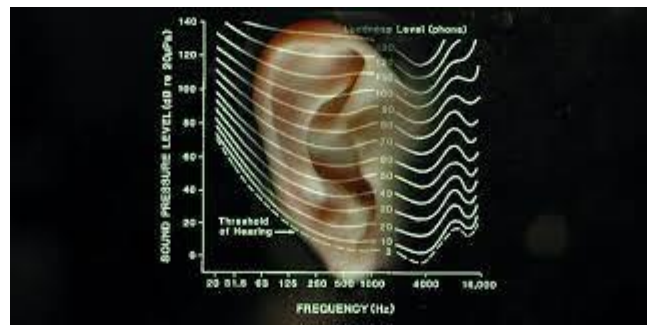
3-4 layers of Scotch tape will attenuate the input by 10-12 decibels…

Take off the hearing aids

dB HL at 1000 Hz	*65 dB input*	*80 dB input*	*95 dB input*
15	**0**	**0**	**0**
25	2	1	0
35	8	4	0
45	14	7	0
55	20	10	1
65	28	15	2
75	36	20	3
85	44	24	4

Higher inputs require lower gains…





Low gain/high output test

Set an aid for 5-10 dB of gain, but maximum OSPL90 with a 105 dB SPL input and if distortion (>20%) then cannot handle loud (live) music.

Input + gain << Output

Chasin, M. "Can your hearing aid handle loud music? A quick test will tell you", The Hearing Journal Dec. 2006, 59 (12), 22-24.

Four technical innovations…

1. -6 dB/octave low cut microphone

2. Shifting the dynamic range upwards

3. Front end compression prior to the A/D converter

4. Post 16 bit architecture

1. -6 dB/octave low cut microphone

Non-occluding BTE provide gain above 1000 Hz and do not occlude the ear canal.

Useful for those with a high frequency loss

BUT still has a front end limiting problem…

1. -6 dB/octave low cut microphone

SO…. We can use a desensitized microphone.

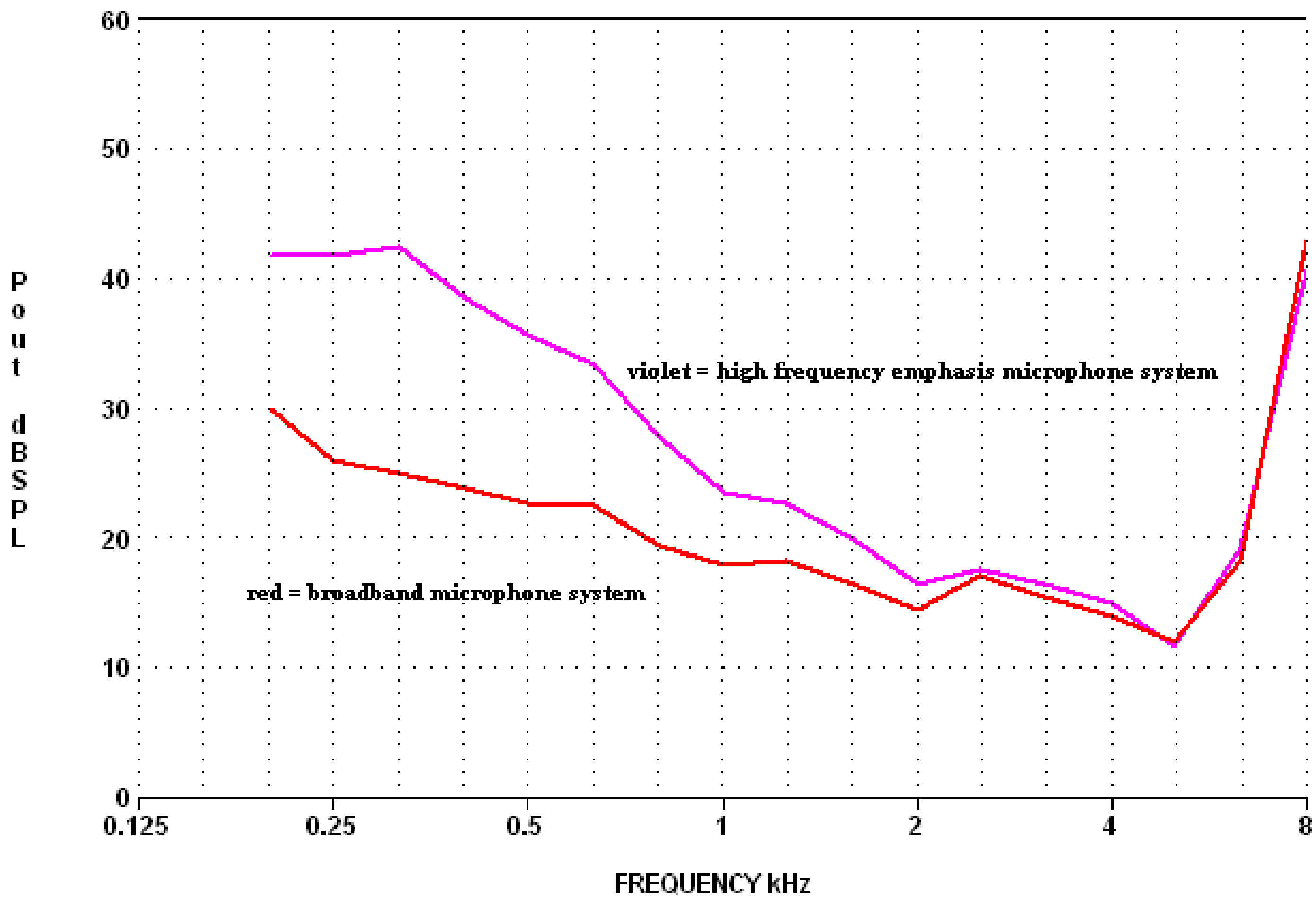
Use a high frequency emphasis (-6 dB low frequency roll-off) microphone.

Same frequency response but less front end distortion.

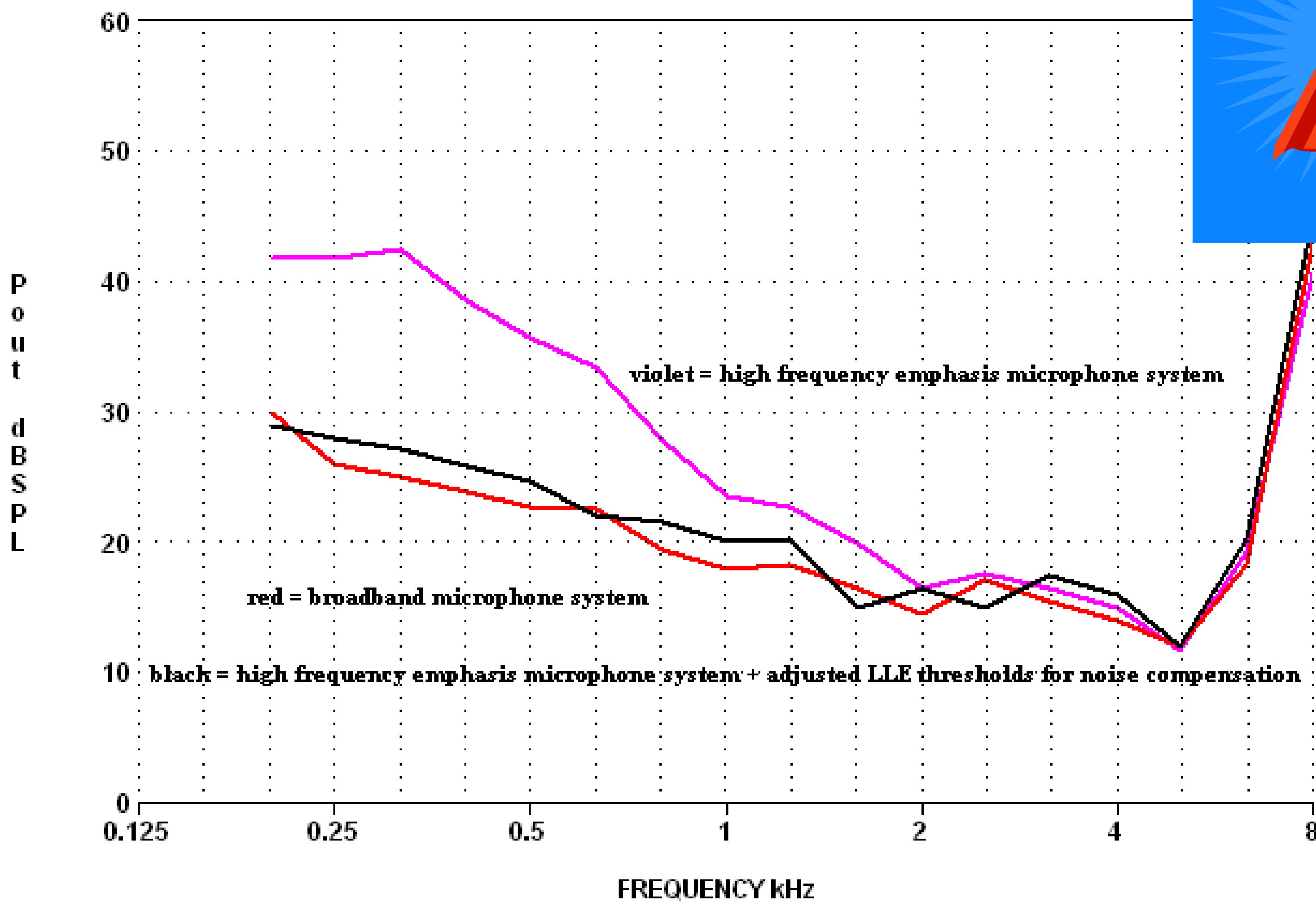
Chasin, M., & Schmidt, M. (2009). The use of a high frequency emphasis microphone for musicians, Hearing Review, 16(2), 32–37.

Schmidt, M. (2012). Musicians and hearing-aid design—Is your hearing instrument being overworked? Trends in Amplification, 16(3), 140–145.

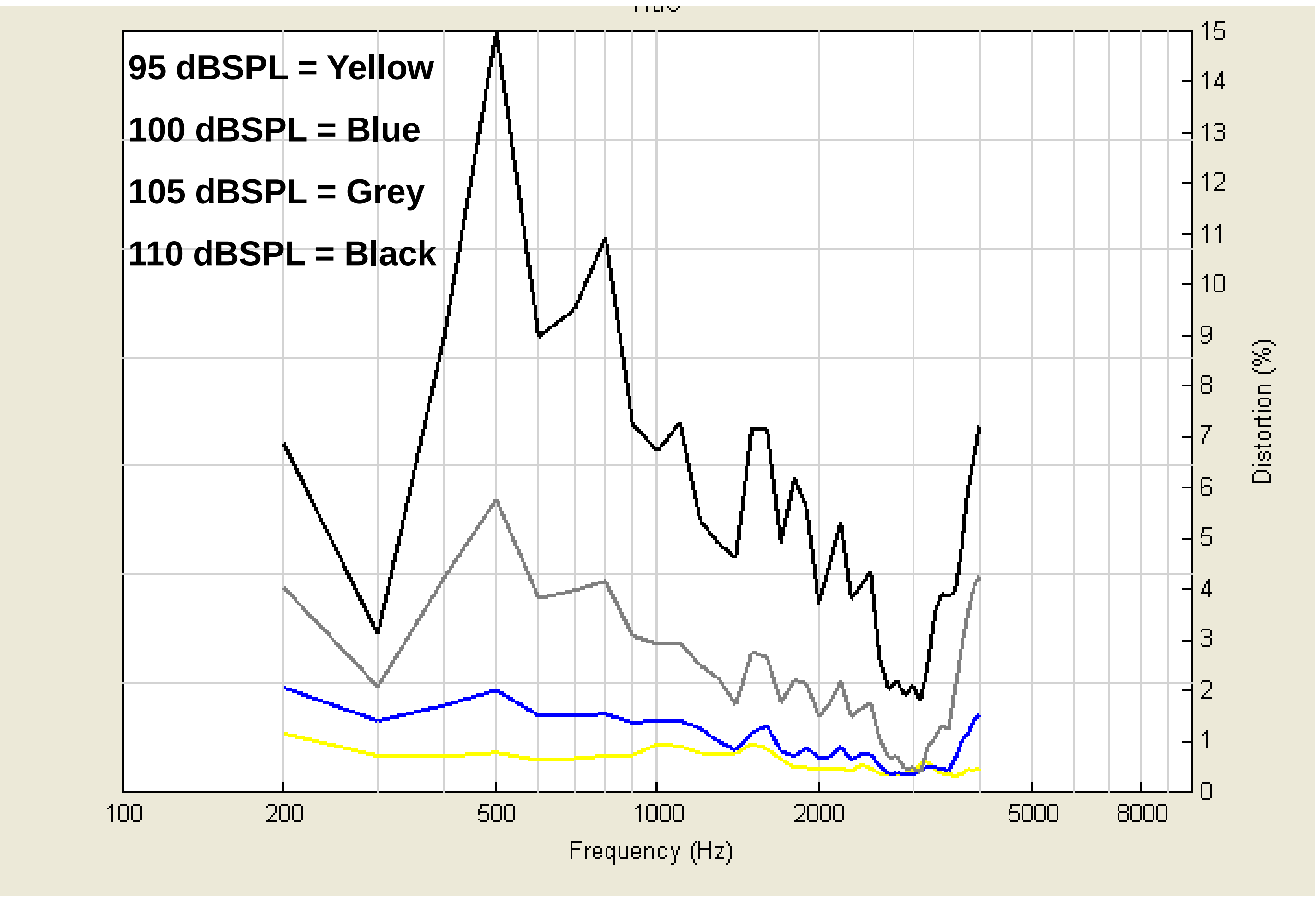
Microphone noise…
you will need expansion…



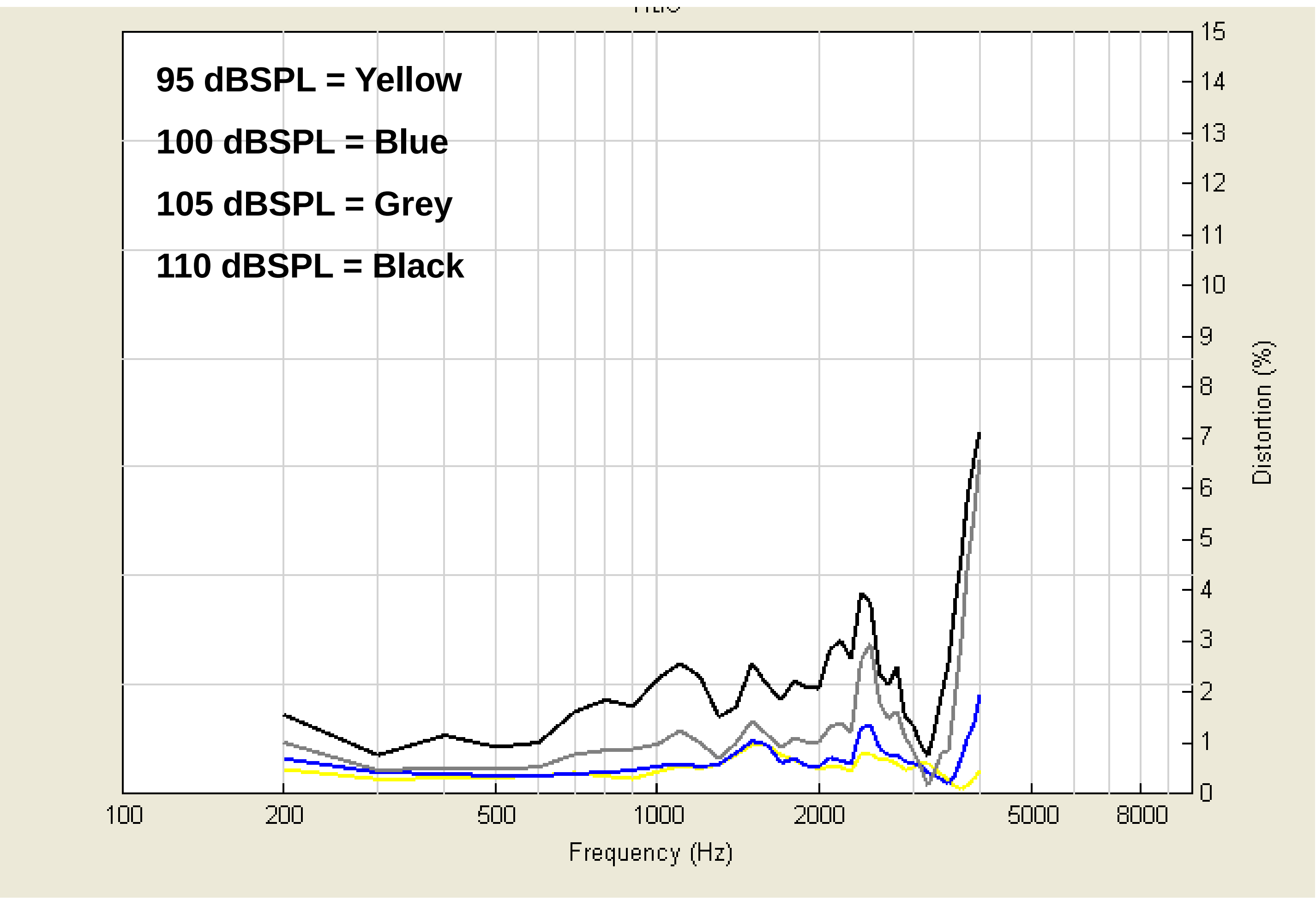
Expansion comes to the rescue



THD Results with Broad Band Mic for 95, 100, 105 & 110 dB SPL inputs



THD Results with High Frequency Mic for 95, 100, 105 & 110 dB SPL inputs



2a. Shifting the dynamic range upwards

Based on the definition of “dynamic range”

“90 dB dynamic range”

0 dB SPL to 90 dB SPL	15 dB SPL to 105 dB SPL
24% distortion	< 3% distortion

2b. Shifting the dynamic range upwards

Another approach …

Transformer effect by doubling the voltage

Increases the high end of the dynamic range by up to 6 dB (can be accomplished off-IC chip)

2b. Shifting the dynamic range upwards

And other approaches …

Auto-ranging of the A/D converter that detects the input and ensures that the input is within the optimal operating range. (e.g. HRX)

“Stacked A/D converters”, while still commercially available, aren’t widely used.

3. Front end compression prior to the A/D converter

Some hearing aid manufacturers are now using an analog compressor prior to the A/D converter …

… and then digitally re-establish gain after…

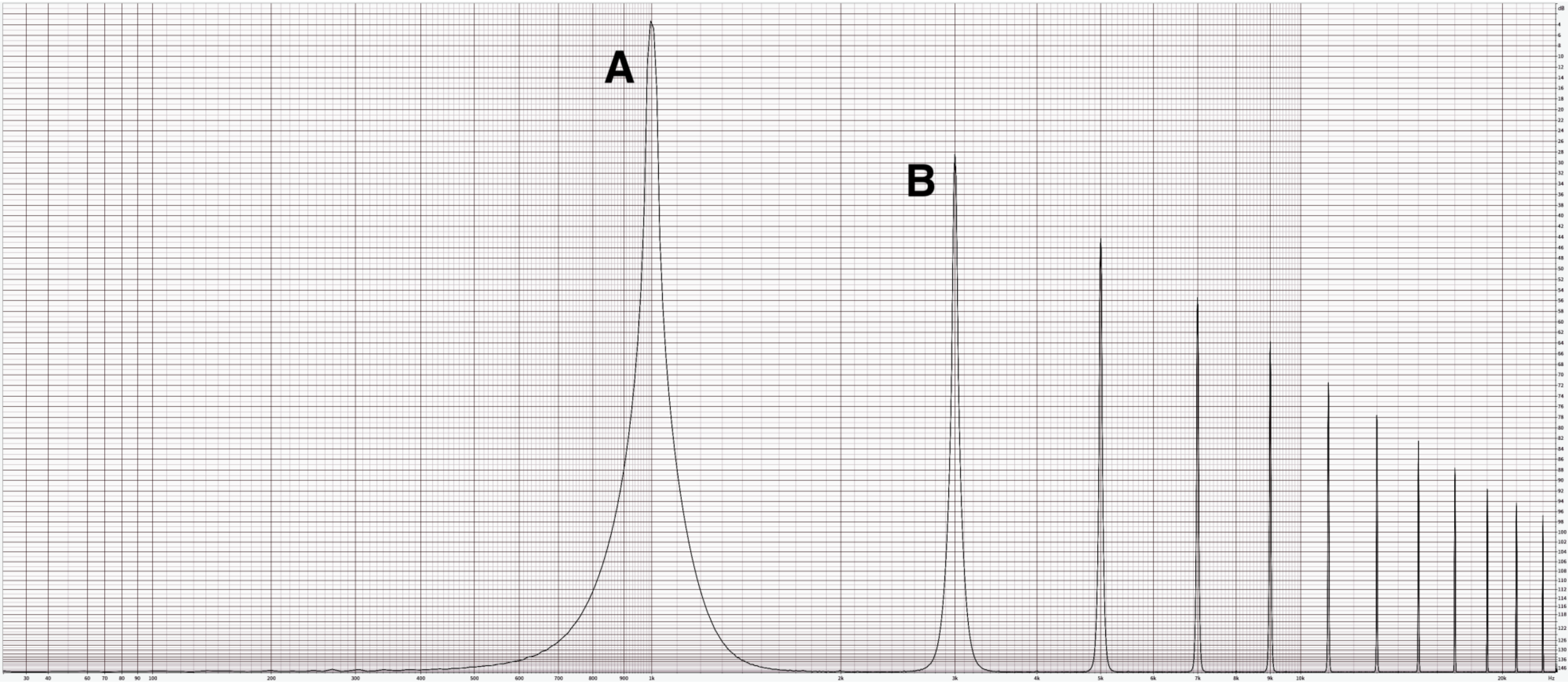
4. Post 16 bit architecture

20 and 24 bit architecture A/D converters that have > 96 dB dynamic range.

- For each bit (n) add 6 dB to dynamic range
  $(20n)\log 2 = 20n \times 0.3 = 6n$


HIT OF THE PARTY
OH, Marshall YOU'RE WONDERFUL!
Did he say 20nlog2!!

One way that manufacturers determine this…



4. Post 16 bit architecture

As reported by manufacturers, some 18 or 19 bit systems, all have >115 dB SPL

Unitron Vivante

Phonak Lumity

Oticon Intent

Widex Moment

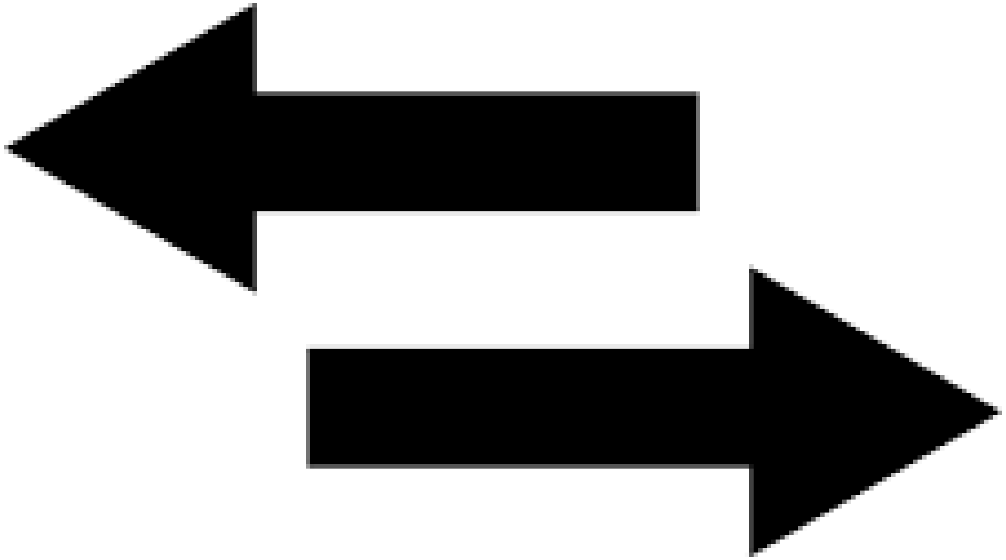
Bernafon Encata

Resound LiNX Quattro/Key

Starkey Evolve AI

....

Frequency response… the two ends…



Frequency response… the low end

Low frequency end (down to 55 Hz):

- affected by earmold/eartip venting

-Hirsch and Bowman (1953)

  - distortion occurred for low frequency inputs >80 dB SPL

- High level, low frequency sounds rarely would be seen by speech but commonly seen even by unamplified music.

Moore, B. C. J., & Tan, C-T. (2003). Perceived naturalness of spectrally distorted speech and music. Journal of the Acoustical Society of America, 114(1), 408–419

Hirsch, I. J., & Bowman, W. D. (1953). Masking of speech by bands of noise. Journal of the Acoustical Association of America, 25, 1175–1180.

Frequency response… the high end

- Hallowell Davis et. al (1950):
  - American servicemen (and Davis himself) volunteered to have TTS created unilaterally.
  - They were given two unmarked knobs; one controlled frequency and the other, sound level. Asked to match the sound in the good ear with that of the TTS affected ear.
  - Low frequencies = good correspondence
  - High frequencies = heard as louder, but not higher pitched

    …. Subjects heard the sound as flat relative to good ear.

    Diagnostic clue = if a client hears sounds as flat…

    (opposite study has yet to be done…)

Davis, H., Morgan, C. T., Hawkins, J. E., Galambos, R., & Smith, F. W. (1950). Temporary deafness following exposure to loud tones and noise. Acta Otolaryngologica, 88(Suppl.), 1–57.


I think that French horn
is a little flat
imgflip.com

Frequency response… the high end

In the early 1990s Brian Moore and his colleagues noted that sometimes if the hearing was very severe, then cochlear dead regions may exist.

TEN test which takes around 10 minutes for 4 frequencies

1. Reduce the amplification in this region
2. Frequency shift to a lower frequency (healthier) region

Moore, B. C. J. (2004). Dead regions in the cochlea: Conceptual foundations, diagnosis, and clinical applications. Ear and Hearing, 25(2), 98–116. 3.

Moore, B. C. J., Glasberg, B. R., & Stone, M. A. (2004). New version of the TEN Test with calibrations in dB HL. Ear and Hearing, 25, 478–487.

A clinical trick to assess dead regions…

Chasin M. "Testing for cochlear dead regions using a piano". Hearing Review. 2019;26(9):12.

Summary of limitations of the high frequencies… (based on speech, but…)

Mild hearing loss	>55 dB hearing loss	Steeply sloping audiogram
Broad bandwidth	Narrow bandwidth	Narrow bandwidth

Aazh, H., & Moore, B. C. (2007). Dead regions in the cochlea at 4kHz in elderly adults: Relation to absolute threshold, steepness of audiogram, and pure-tone average. Journal of the American Academy of Audiology, 18(2), 97–106.

Ricketts, T. A., Dittberner, A. B., & Johnson, E. E. (2008). High frequency amplification and sound quality in listeners with normal through moderate hearing loss. Journal of Speech-Language-Hearing Research, 51, 160–172.

Sometimes we can just change the input…

A client came in with a high frequency hearing loss as a result of chemotherapy as a child. Great mezzo soprano but was having difficulty passing an ear training class (intervals, chords,…)

Was able to convince the administration to perform the test one octave lower…

…. She passed!

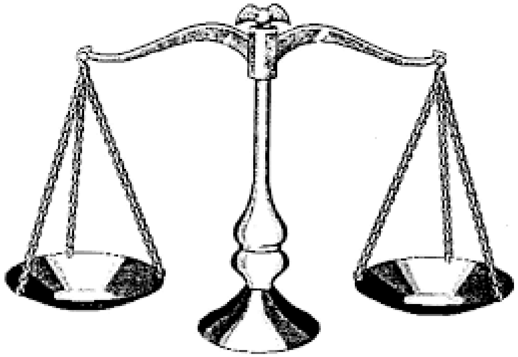
… or, we can just change the music!

Music can be composed with a bass lead rather than a treble lead…

But Skinner and Miller (1983)…

For each increase in high frequency gain, there must be an associated low frequency gain increase as well in order to maintain or improve speech discrimination.

Based on speech but an hypothesis is that this will also apply for music.



One cannot specify a high frequency figure without also specifying an associated low frequency limit as well.

Skinner, M. W., & Miller, J. D. (1983) Amplification bandwidth and intelligibility of speech in quiet and noise for listeners with sensorineural hearing loss. Audiology, 22(3), 253–279.

No frequency lowering for music

No Frequency Lowering

1. **FREQUENCY TRANSPOSITION**: linear decrease up to 2 octaves (e.g. Widex Audibility Extender)

2. **FREQUENCY TRANSLATION**: spectral envelope warping. A "copy and paste" approach where high frequencies are pasted right onto a lower frequency region (e.g., Starkey Spectral IQ)

3. **SOUND RECOVERY**: non-linear compressing by as much as 2:1, but typically not more than 1.5:1 (e.g., Phonak and Unitron)

No Frequency Lowering

Music: line spectrum (except percussion).

Speech: line spectrum for low frequencies
continuous spectrum for high frequencies

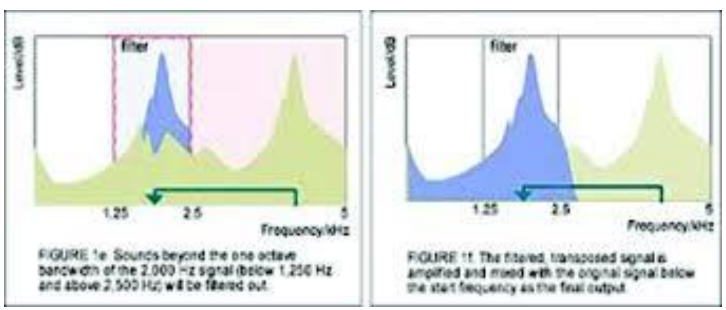
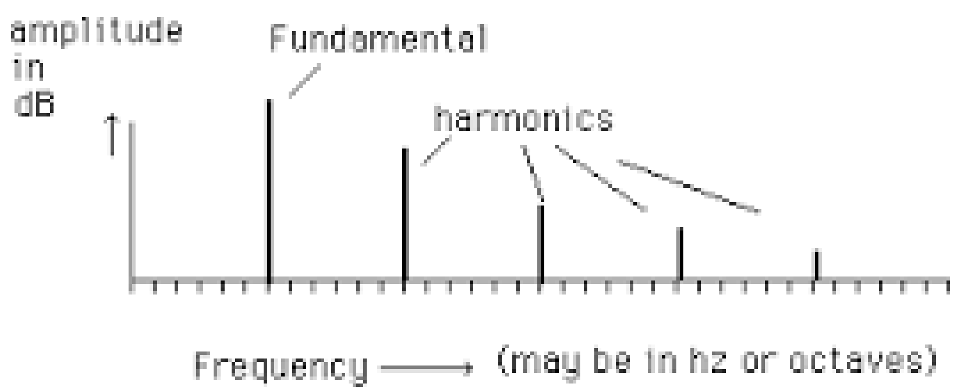


FIGURE 1e: Sounds beyond the one octave bandwidth of the 2,000 Hz signal (below 1,250 Hz and above 2,500 Hz) will be filtered out.

FIGURE 1f: The filtered, transposed signal is amplified and mixed with the original signal below the start frequency as the final output.



Frequency lowering is for speech; not for music

Speech:

1. Low frequency line spectra Sonorants
2. High frequency continuous spectra

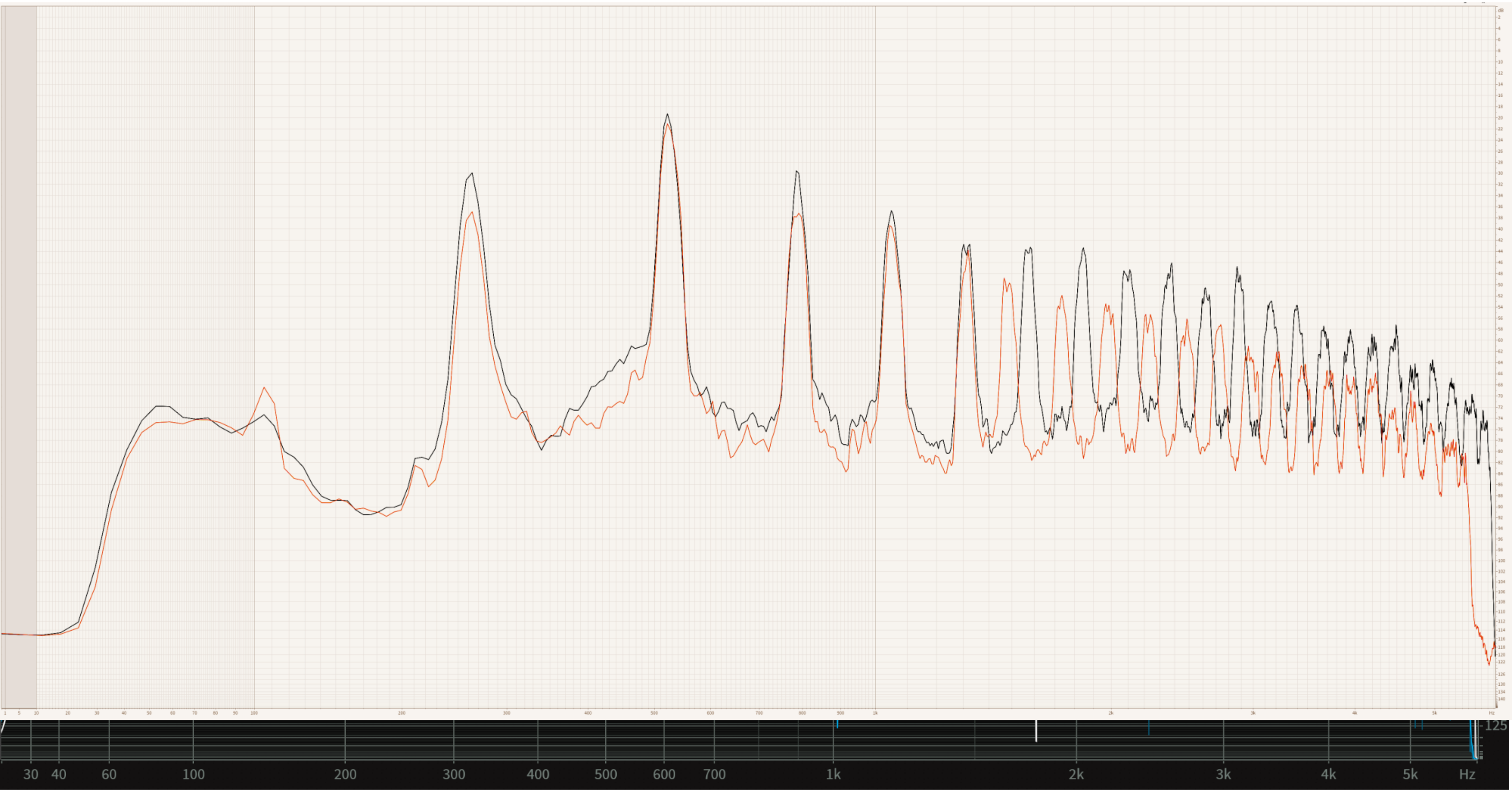
   Obstruents (e.g., ‘s, sh, th, f,…’)

Music:

All frequencies have line spectra

Chasin M. Back to Basics: “Frequency Compression Is for Speech, Not Music”. Hearing Review. 2016;23(6):12.

½ of one semi-tone decrease above 1500 Hz



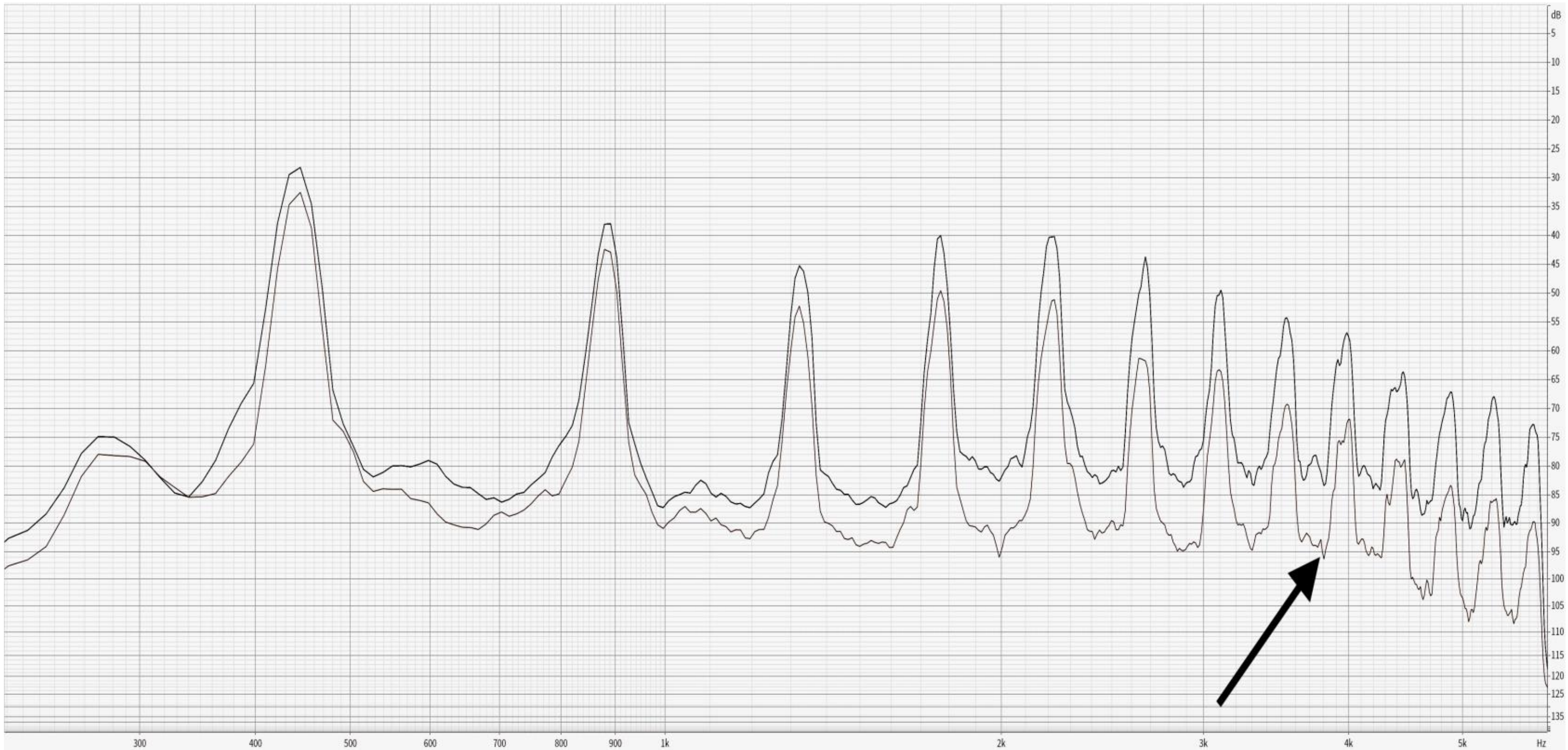
No Frequency Lowering

½ of one semi-tone decrease above 1500 Hz with one note (1.059/2)

Same thing with full music score

But OK for speech

-6 dB/oct roll-off above 1500 Hz



-6 dB/oct roll-off above 1500 Hz

Do not change the frequencies of the harmonics, but instead gradually reduce the amplitudes slightly

Similar compression…
… A cochlear issue

Compression is implemented in order to re-establish the loudness growth function of the damaged cochlea and not the characteristics of the input signal.

Davies et al. (2009)… "Chasin and Russo (2004) suggested that WDRC … may be better for music…. That hypothesis was supported by the present data." (p. 696).

Hearing aids and compression (ABA)

Slight compression for music and for speech (BUT the volume has been turned up for the B part):

MUSIC

SPEECH

Hearing aids and compression (ABA)

TOO MUCH compression for music and for speech (AND the volume has been turned up for the B part):

MUSIC

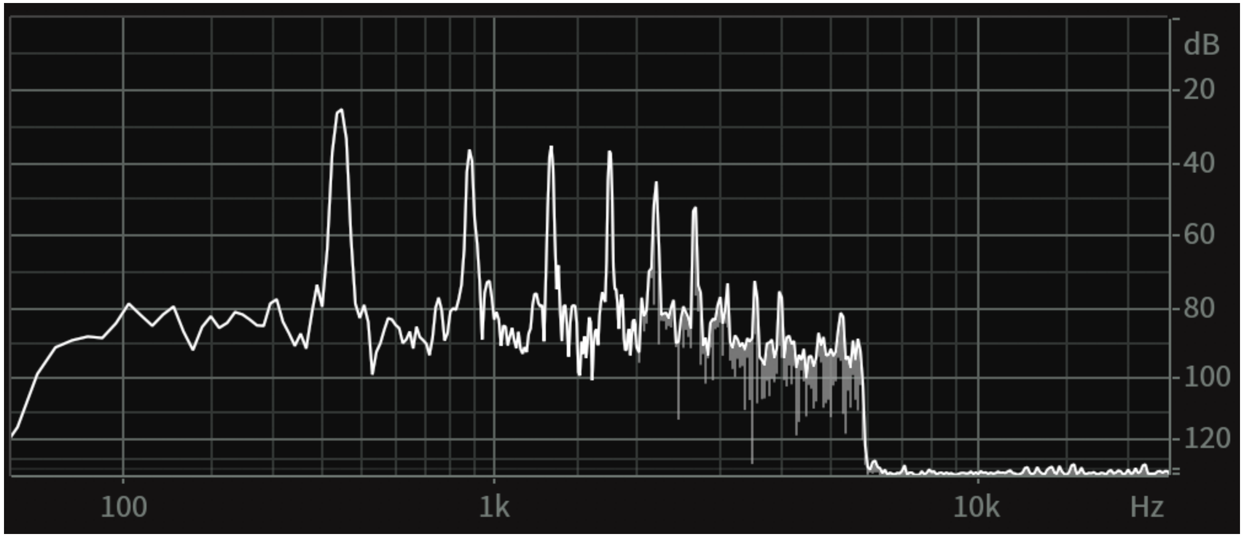
SPEECH

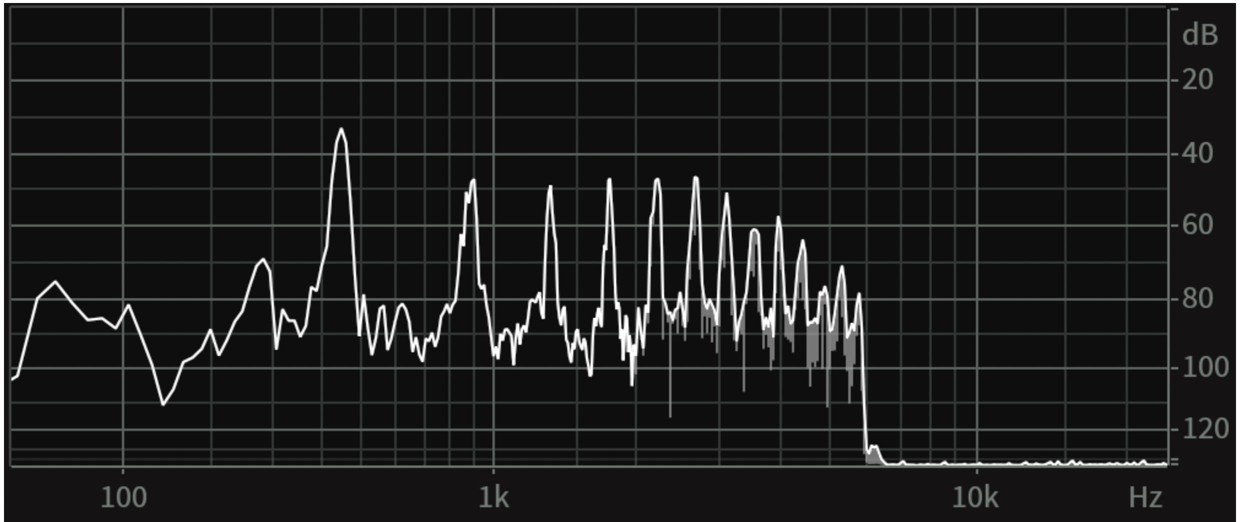
The (potential) problem with multi-band compression

Many musical instruments vary by only the amplitude of the harmonics (and not the frequency of the harmonics).

Examples: violin, piano, flute, guitar, saxophone,… only are all ½ wavelength resonator instruments and all have integer multiples of their fundamental (or tonic)- they only differ in the amplitude of the harmonics.

The (potential) problem with multi-band compression (A: flute/B: oboe)





Settings for a speech and music program

Music is the SAME as Speech

1. Compression
2. Frequency response

Music is DIFFERENT than speech

1. No advanced features
2. No frequency lowering

Less is more when it comes to music!

Whoever said "less is more" clearly wasn't a coffee drinker.

DEATH WISH COFFEE CO.

Music Program #1- Live Music

1. The "front end" has been taken care of
2. The frequency bandwidth has been optimized for that client
3. Frequency lowering has been disabled
4. Similar compression characteristics to a speech-in-quiet program
5. Advanced features such as feedback management has been "managed"

What about streamed music?

Thank you to:

Crogan, N., Arehart, K, and Kates, J. (2012). Quality and loudness judgments for music subjected to compression limiting. JASA, 132(2), 1177-1188.

Music has been compression limited once already during the creation of the MP3/MP4 process. This is a relatively benign form of compression but does reduce the volume. (Some radio stations tend to use this as well).

Should not “doubly” compress this streamed music, so the music program should be close to linear.

Music Program #2- Streamed Music

1. The “front end” has been taken care of
2. The frequency bandwidth has been optimized for that client
3. Frequency lowering has been disabled
4. Linear, or only slight WDRC compression for this program
5. Advanced features such as feedback management has been “managed”

Digital Delay

Up to 6 or so years ago, for just A/D and D/A and no DSP, the digital delay was 5-6 msec.

Today, for just the A/D and D/A and no DSP, the digital delay is now 50 microseconds

- a 100 fold improvement

BUT what about digital accessories and apps? ...

Digital Delay

Modern hearing aid accessories based on a Bluetooth platform have typical digital delays of 28-33 msec… and proprietary "links" can add 10 msec…

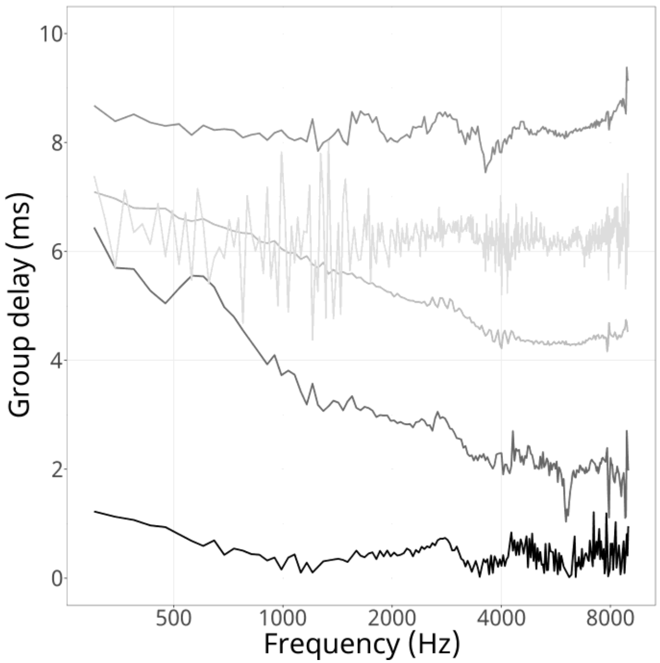
Many Smartphone "noise stripping" apps have digital delays on the order of 50-70 msec (e.g., "HeardThat" https://heardthat.ai)

With the advent of Bluetooth LEAudio, these numbers will improve somewhat

Digital Delay

Two types of processing in hearing aid industry:

1. FFT based where delay is the same for all frequencies (e.g., HeardThat app)
2. Band of filters where delay is greater for lower frequency sounds than higher frequency sounds



Digital Delay

Recall another difference between speech and music

- At any one time, speech is narrow band

  (either low frequency sonorants or high frequency obstruents, but never both)

- At any one time, music is broadband

  (Harmonics for both low and higher frequencies)

If the digital delay is different for different frequency ranges, this can be more problematic for music

Digital Delay… are we overly concerned?


EXIT

Marshall.Chasin@rogers.com

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All audio files can be found at:
MusiciansClinics.com/Demos

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