Stop Counting Samples

Thomas Lund

TC Electronic A/S
Denmark

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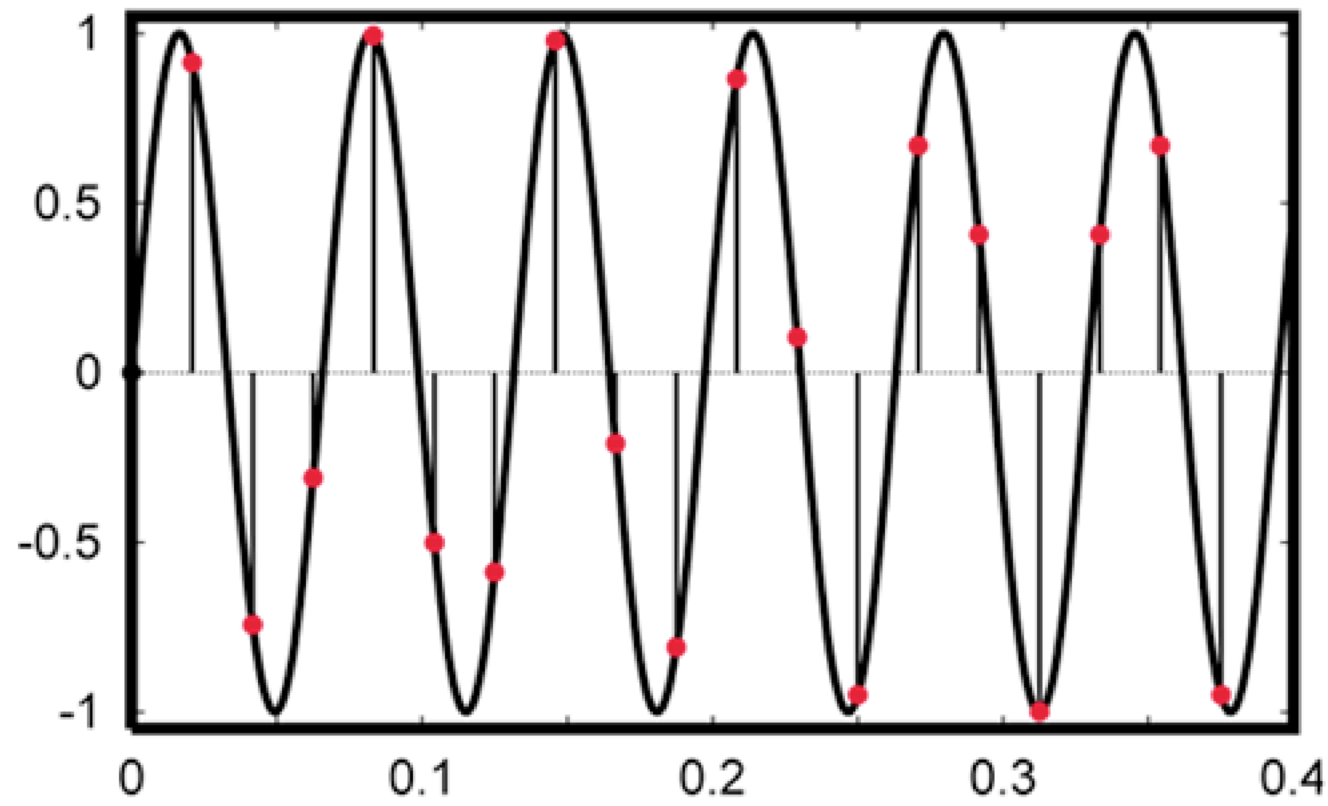
Agenda

Definitions	Intrinsic Level, 0 dBFS+ PPM, Sample and Signal Peak Meter
The Case	Summary of Papers and Articles Linear Audio Listening examples
Status	Production and Delivery Today Codec Audio Listening examples
Looking Ahead	Loudness Control, ITU-R BS.1770 Production Advice

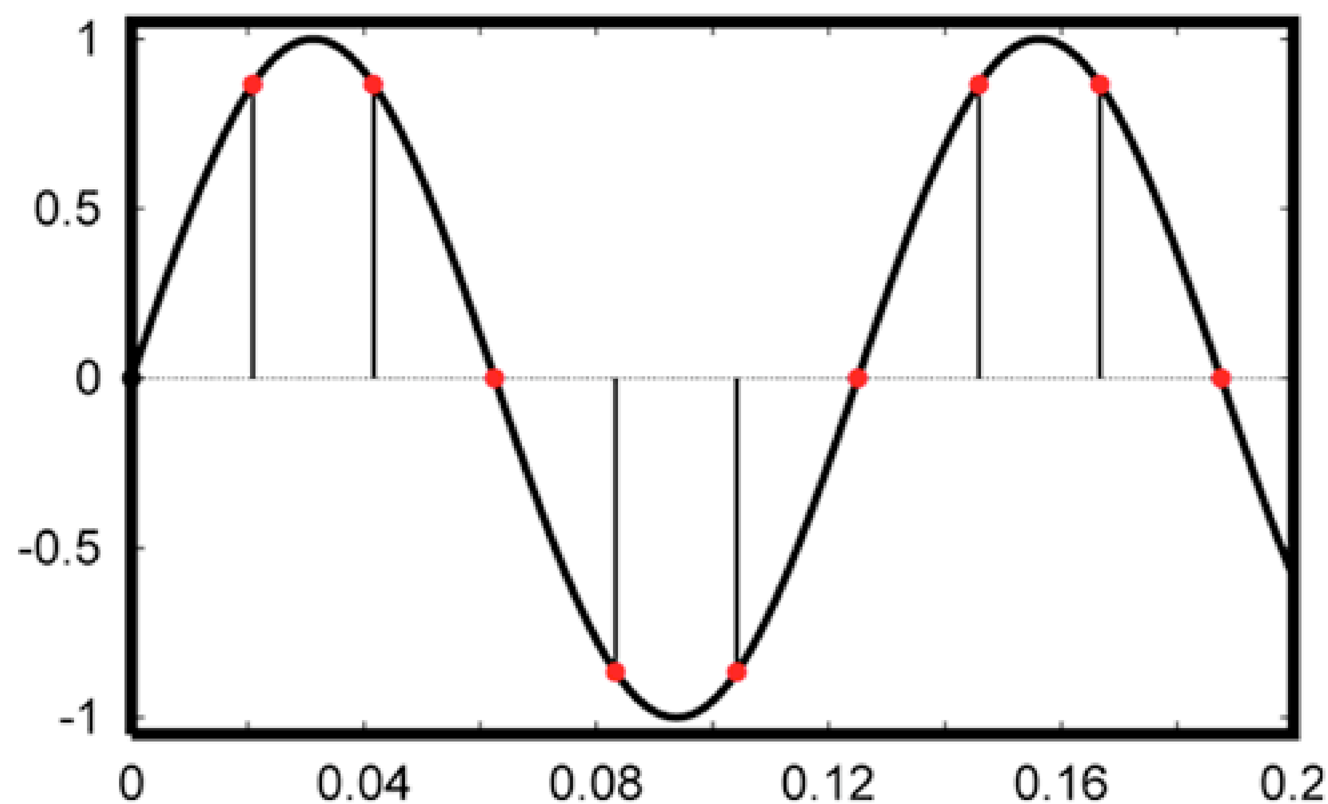
Intrinsic Level

Even in a linear audio system, analog and digital level is not the same.

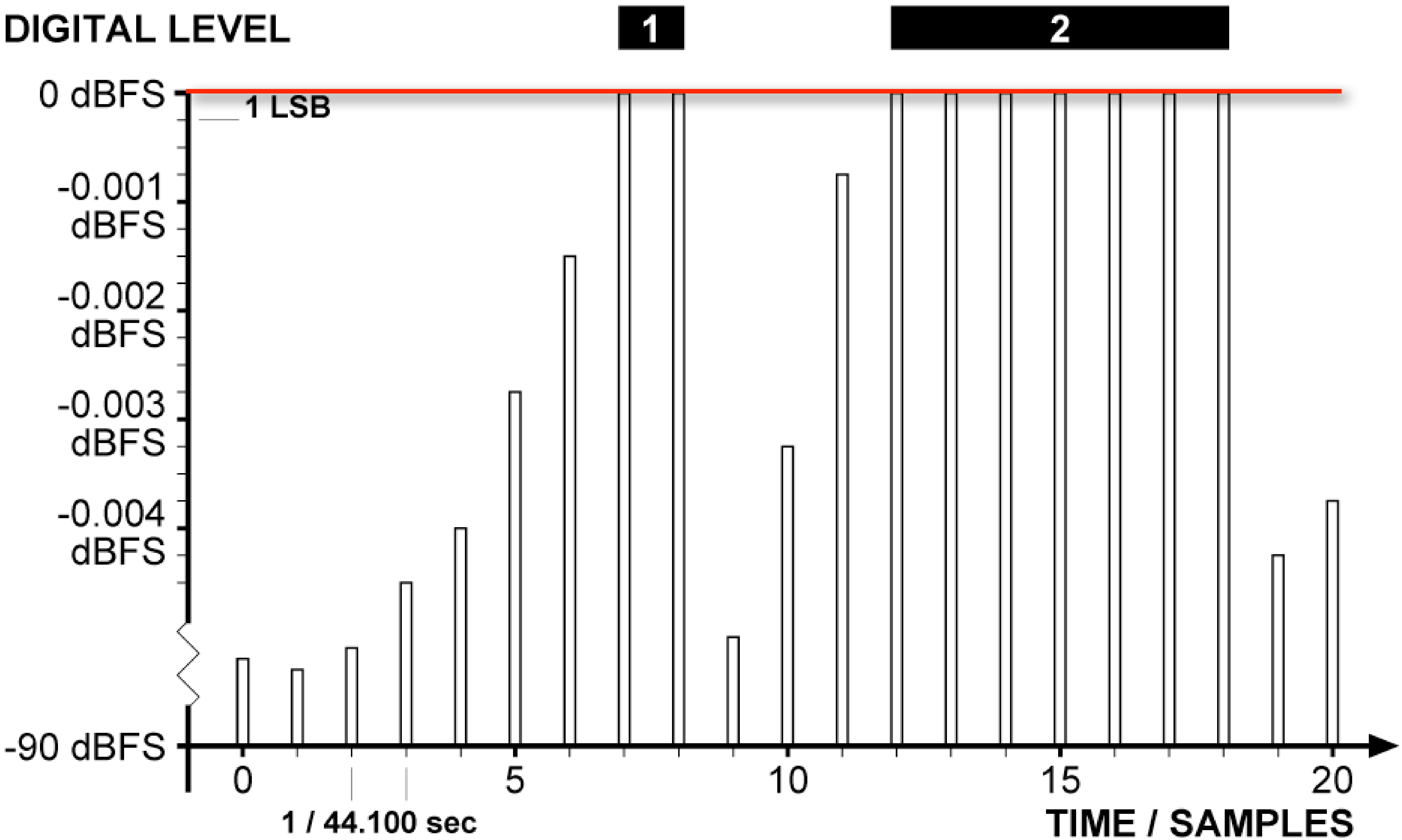
1



2



Sample Counting



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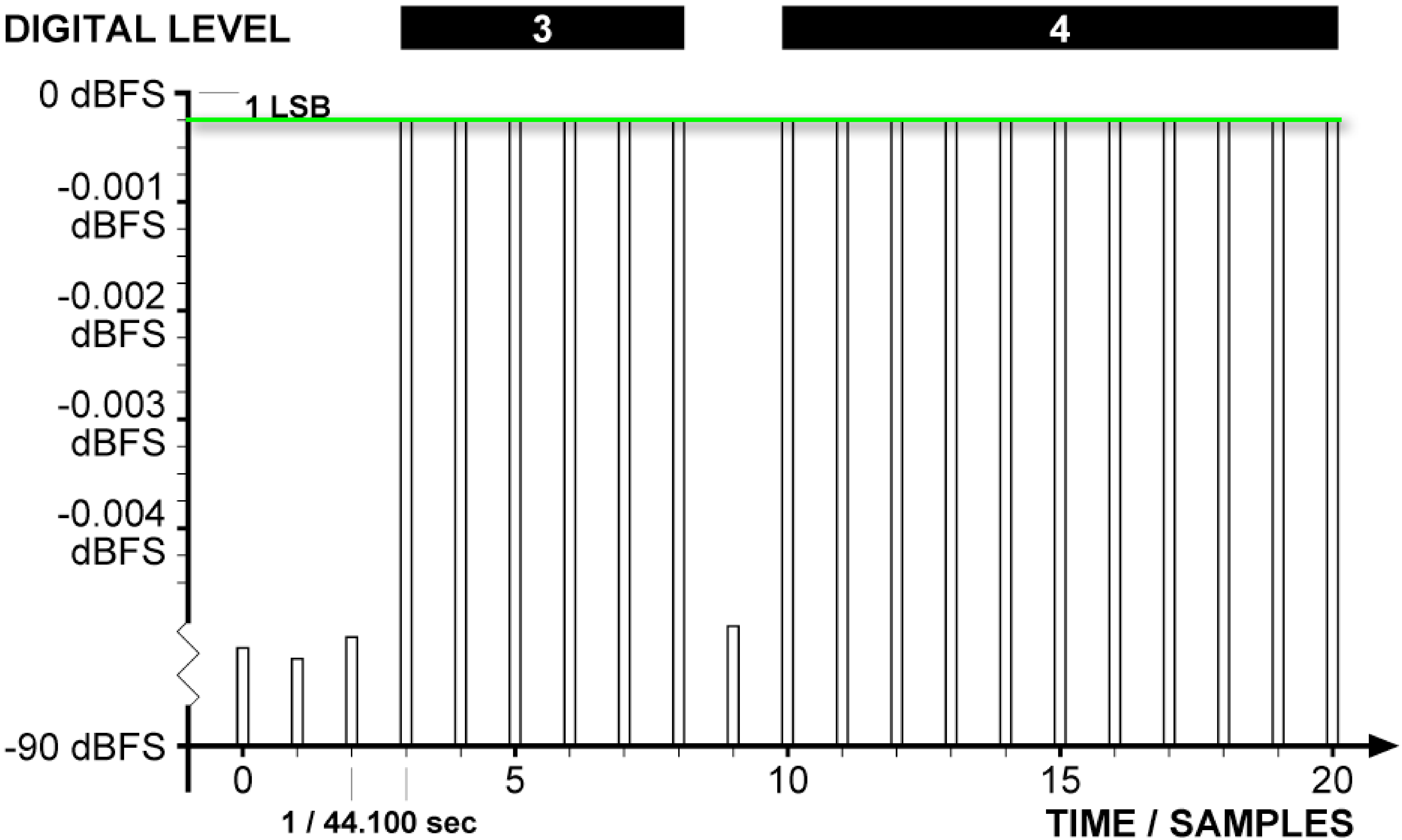
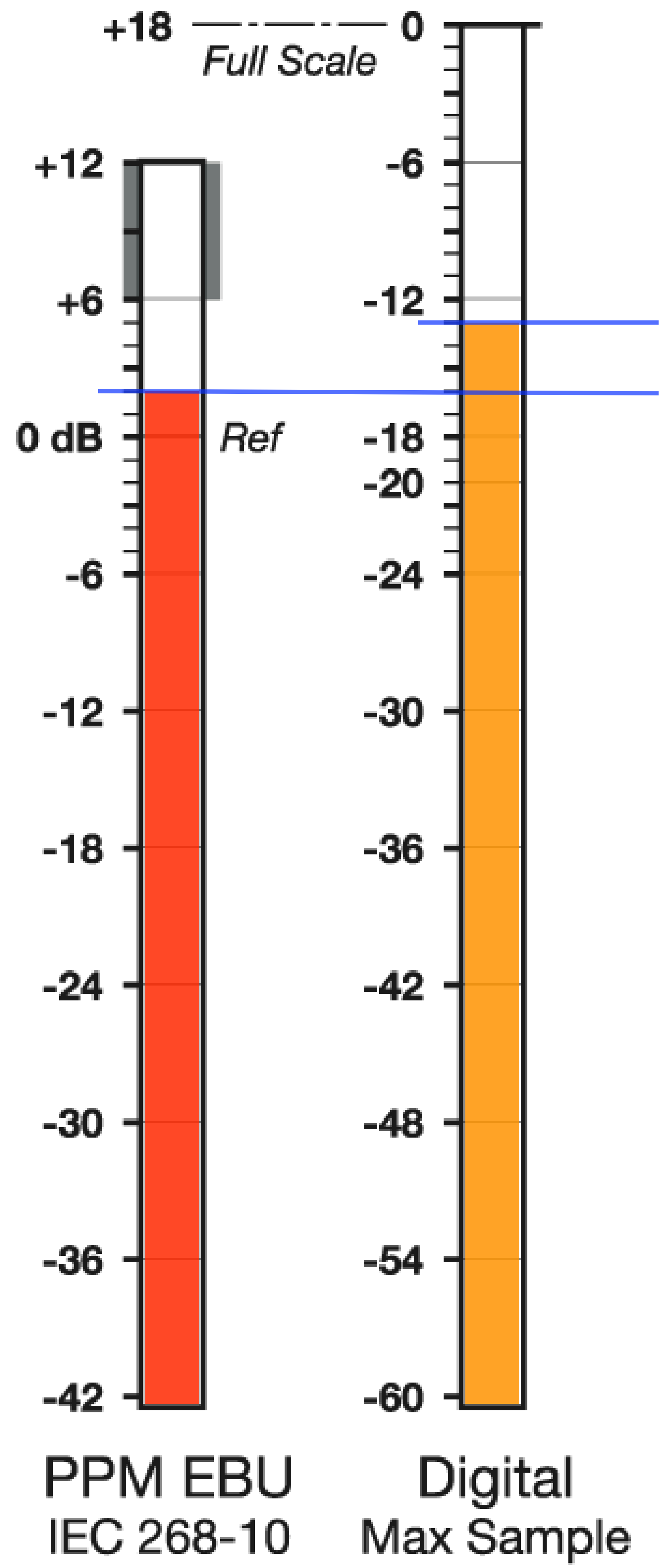


Peak Meters

With program material, PPM and Digital meters do *not* show the same.



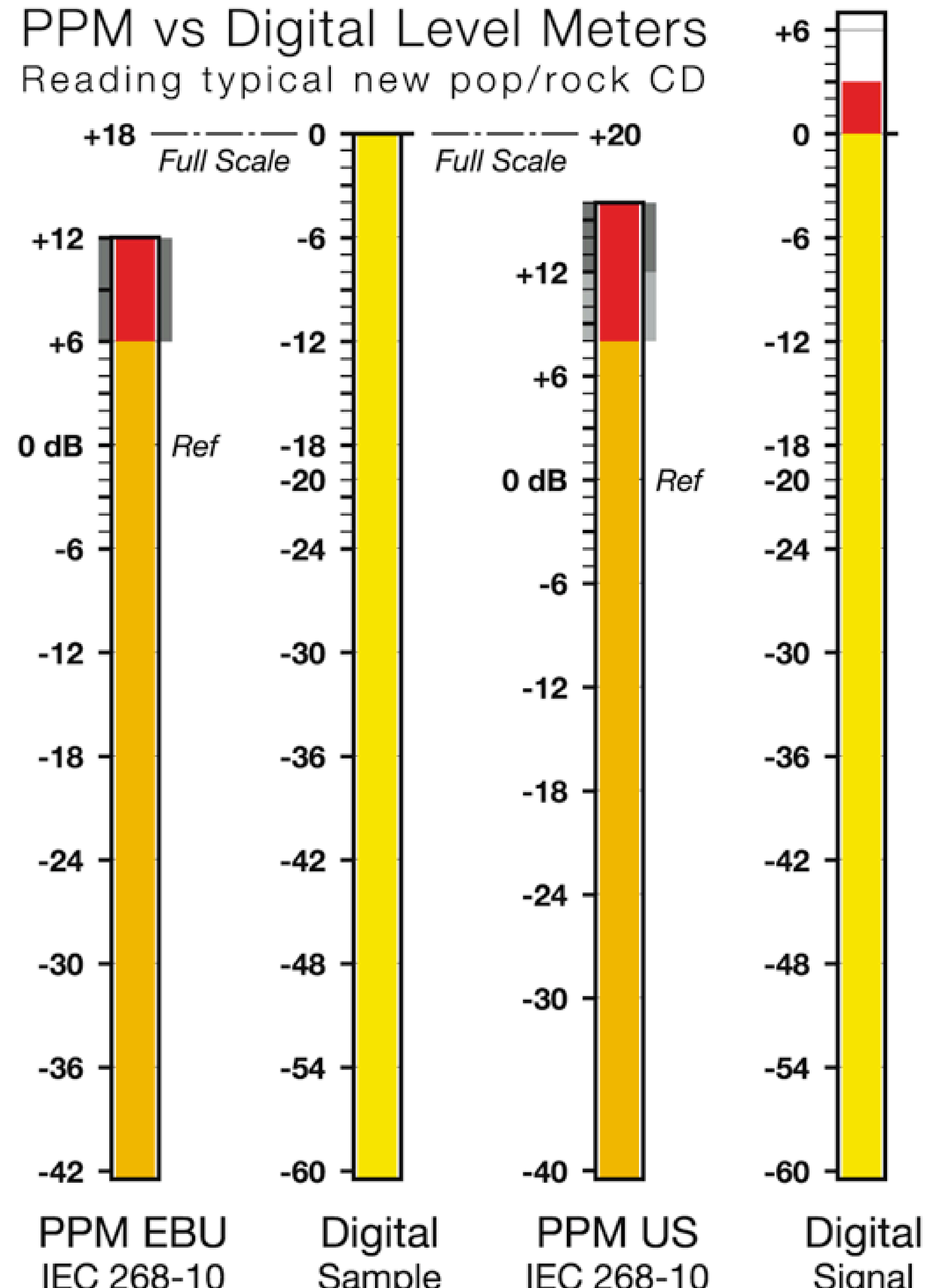
The difference may be 3-4 dB or more with many types of program material.

Peak Meters

Comparison of

- PPM
- Digital Sample
- Digital Signal

Peak Level meters



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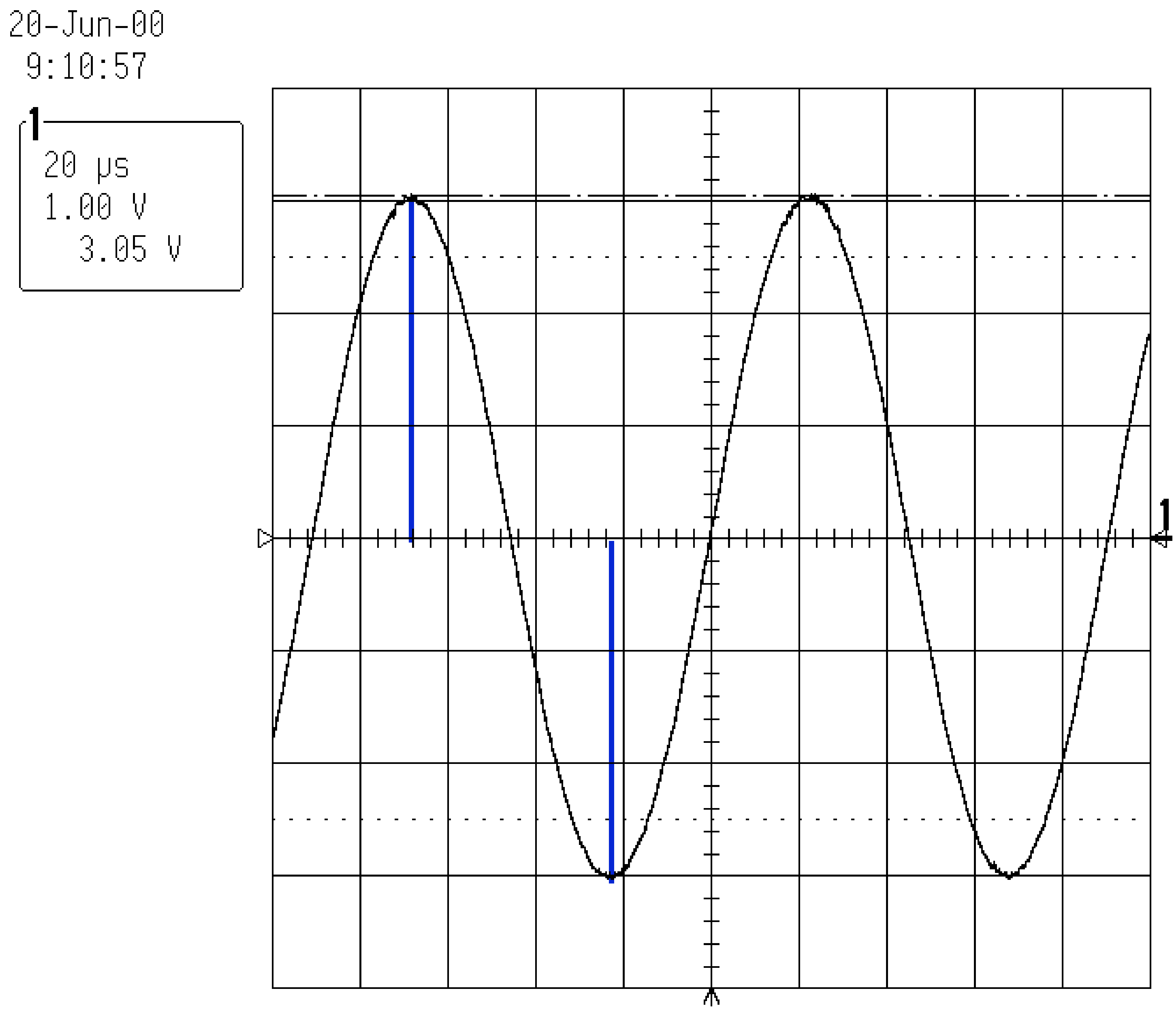
Audio Level

Distortion & Listening Fatigue in Digital Audio

Examples

Sine @ fs/4
Full Scale

Start Phase
90 degrees



Le Croy 9350A
NAD C 520
Sample rate: 44.1 kHz
Signal: 11.025 kHz

Audio Level

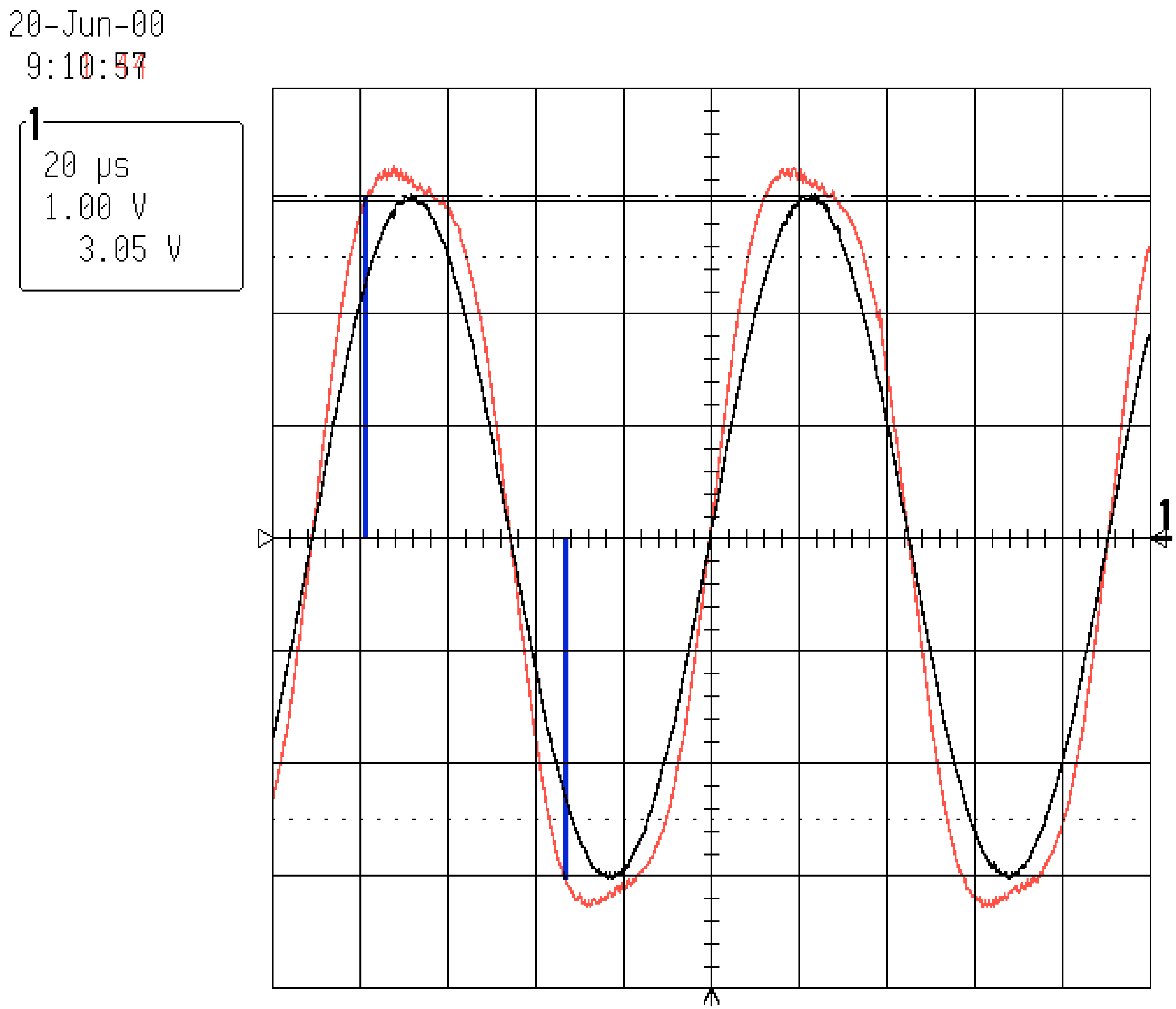
Distortion & Listening Fatigue in Digital Audio

Examples

Sine @ fs/4
Full Scale

Start Phase
90 degrees

Start Phase
45 degrees



Le Croy 9350A
NAD C 520
Sample rate: 44.1 kHz
Signal: 11.025 kHz

Background & Summary

Distortion to The People

Model	Ref Trk 0 dBFS
Denon DCD725	-61.3 dB
Marantz CD4000	-58.8 dB
NAD 514	-74.3 dB
NAD 520	-67.9 dB
Sony C11	-78.1 dB
Sony D50	-82.9 dB
Yamaha CDX390	-70.9 dB

CD Player Hot Level Reproduction

THD+n
20 Hz - 80 kHz

Background & Summary

Distortion to The People

Model	Ref Trk 0 dBFS	fs/8 +0.69 dBFS	fs/6 +1.25 dBFS	fs/4 +3.0 dBFS
Denon DCD725	-61.3 dB	-34.8 dB	-27.0 dB	-18.1 dB
Marantz CD4000	-58.8 dB	-36.6 dB	-30.7 dB	-20.7 dB
NAD 514	-74.3 dB	-30.6 dB	-24.9 dB	-17.2 dB
NAD 520	-67.9 dB	-30.4 dB	-25.8 dB	-19.3 dB
Sony C11	-78.1 dB	-30.2 dB	-24.6 dB	-16.8 dB
Sony D50	-82.9 dB	-65.0 dB	-59.3 dB	-29.0 dB
Yamaha CDX390	-70.9 dB	-33.9 dB	-26.4 dB	-18.3 dB

CD Player Hot Level Reproduction

THD+n
20 Hz - 80 kHz

Background & Summary

Distortion to The People

Model	Ref Trk 0 dBFS	fs/8 +0.69 dBFS	fs/6 +1.25 dBFS	fs/4 +3.0 dBFS
SRC 1 44.1>48k	-79.8 dB	-30.2 dB	-25.0 dB	-20.0 dB
SRC 2 44.1>48k	-78.2 dB	-31.4 dB	-25.8 dB	-21.5 dB
Broadcast Proc.	-71.3 dB	-31.6 dB	-25.7 dB	-18.5 dB

Pro Equipment Rate Conversion

THD+n
20 Hz - 80 kHz

Audio Level

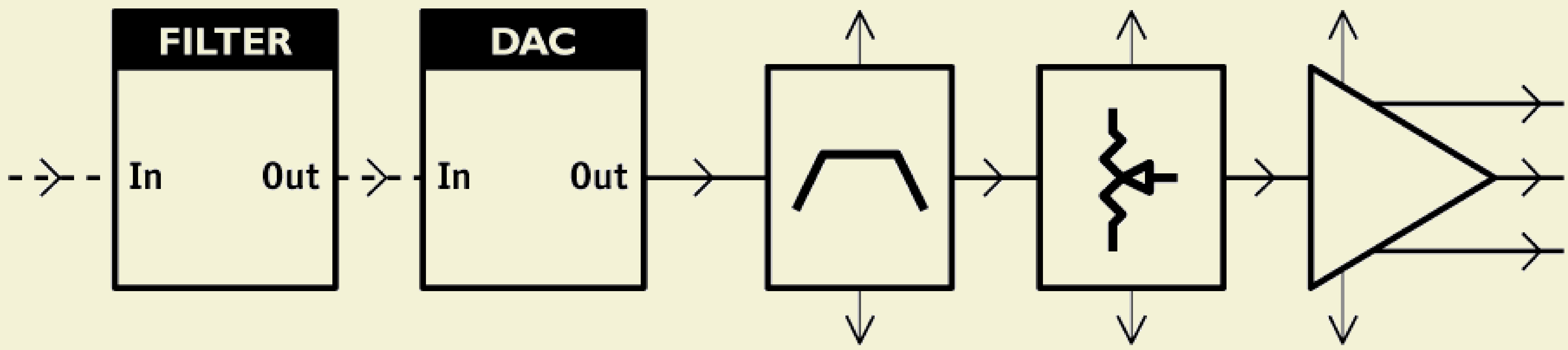
Critical Areas

Headroom is needed several places in the signal-path:

DA Converters
Filters, analog and digital
Sample rate converters
Data reduction codecs (e.g. MP3)

Digital Filter

Conversion



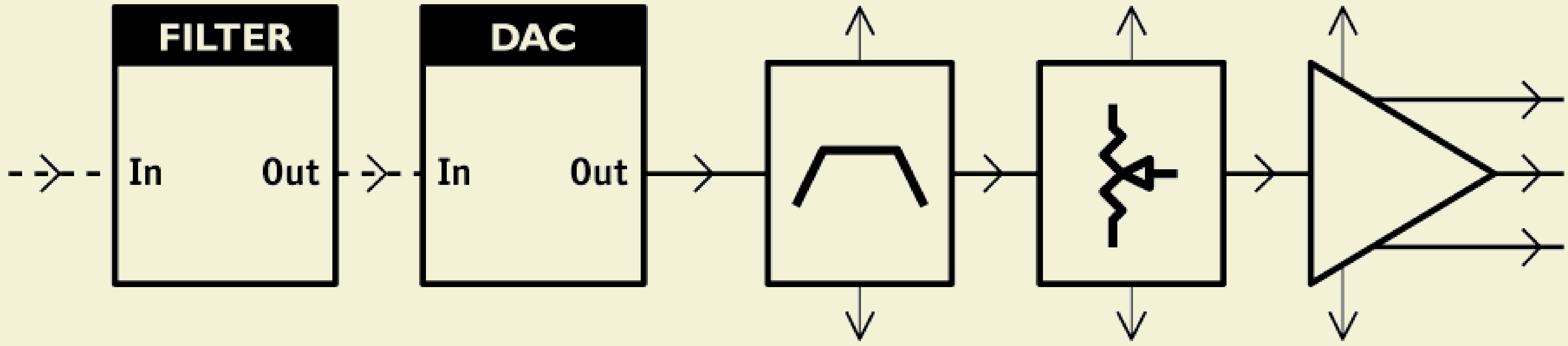
DA Conversion

Elements of the DA Conversion signalpath
sensitive to 0 dBFS+ level

Headroom Required

Distortion & Listening Fatigue in Digital Audio

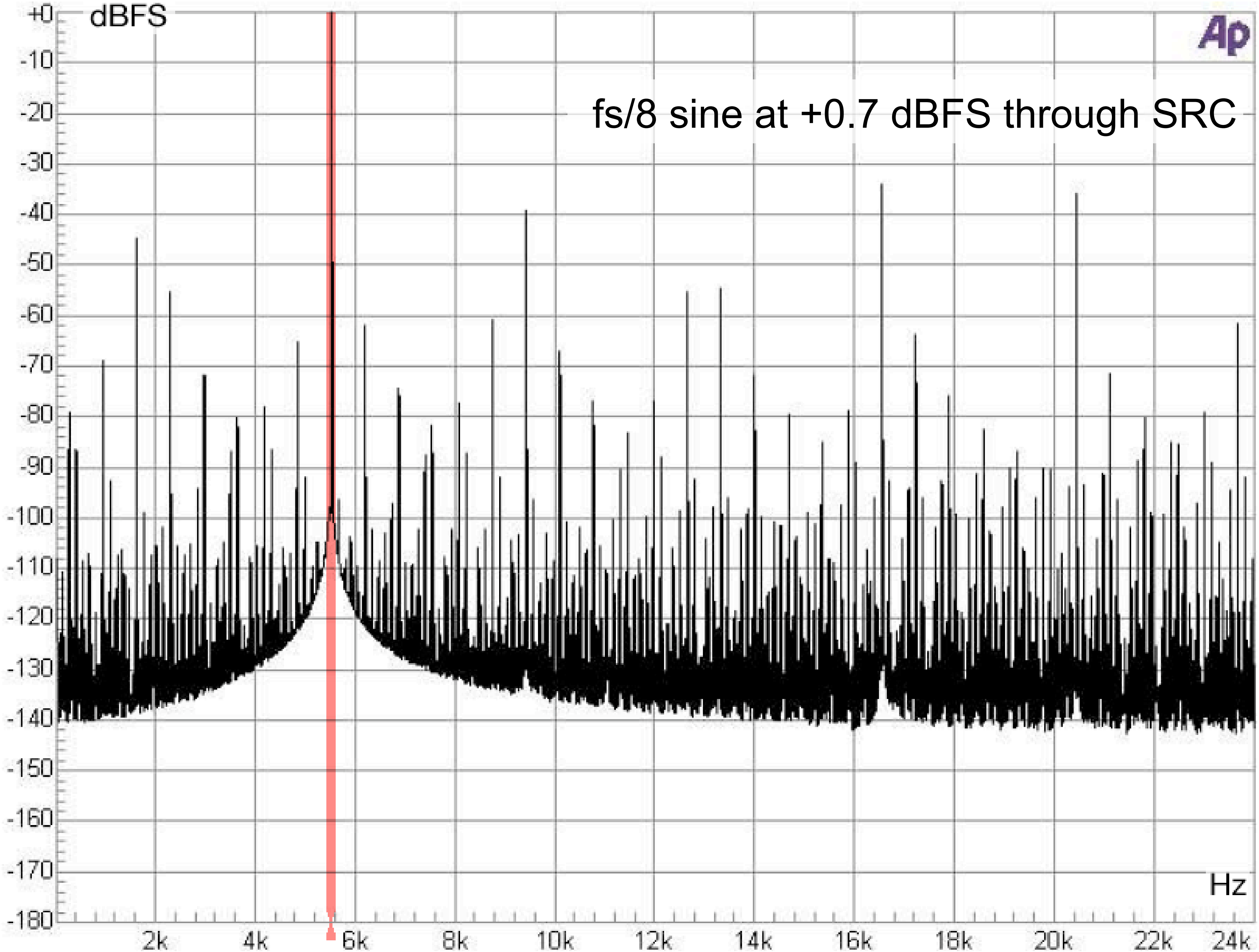
Digital Filter | Conversion | Analog Filters | Gain Scale | Output Buffer



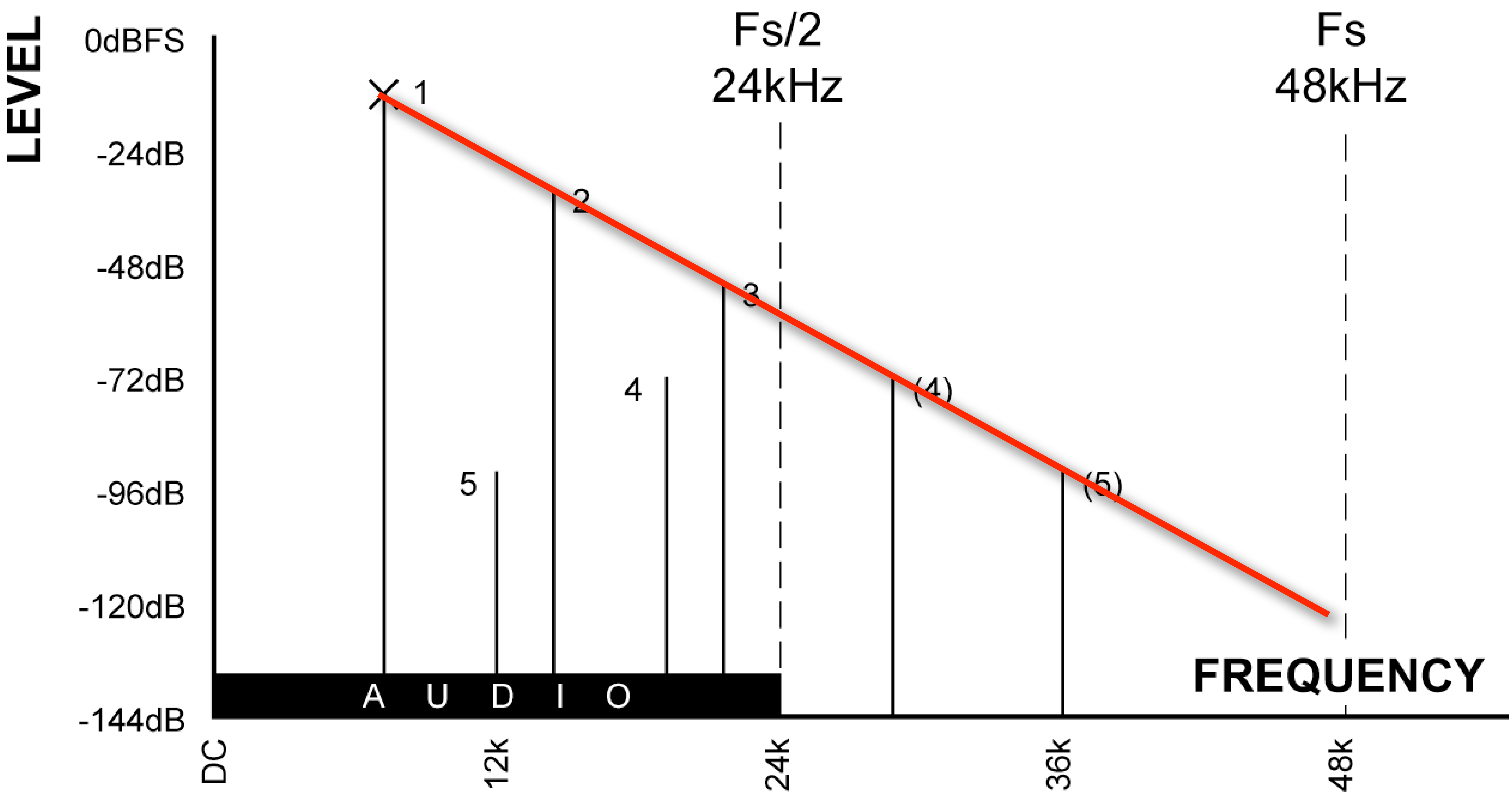
DA Conversion

Elements of the DA Conversion signalpath sensitive to 0 dBFS+ level

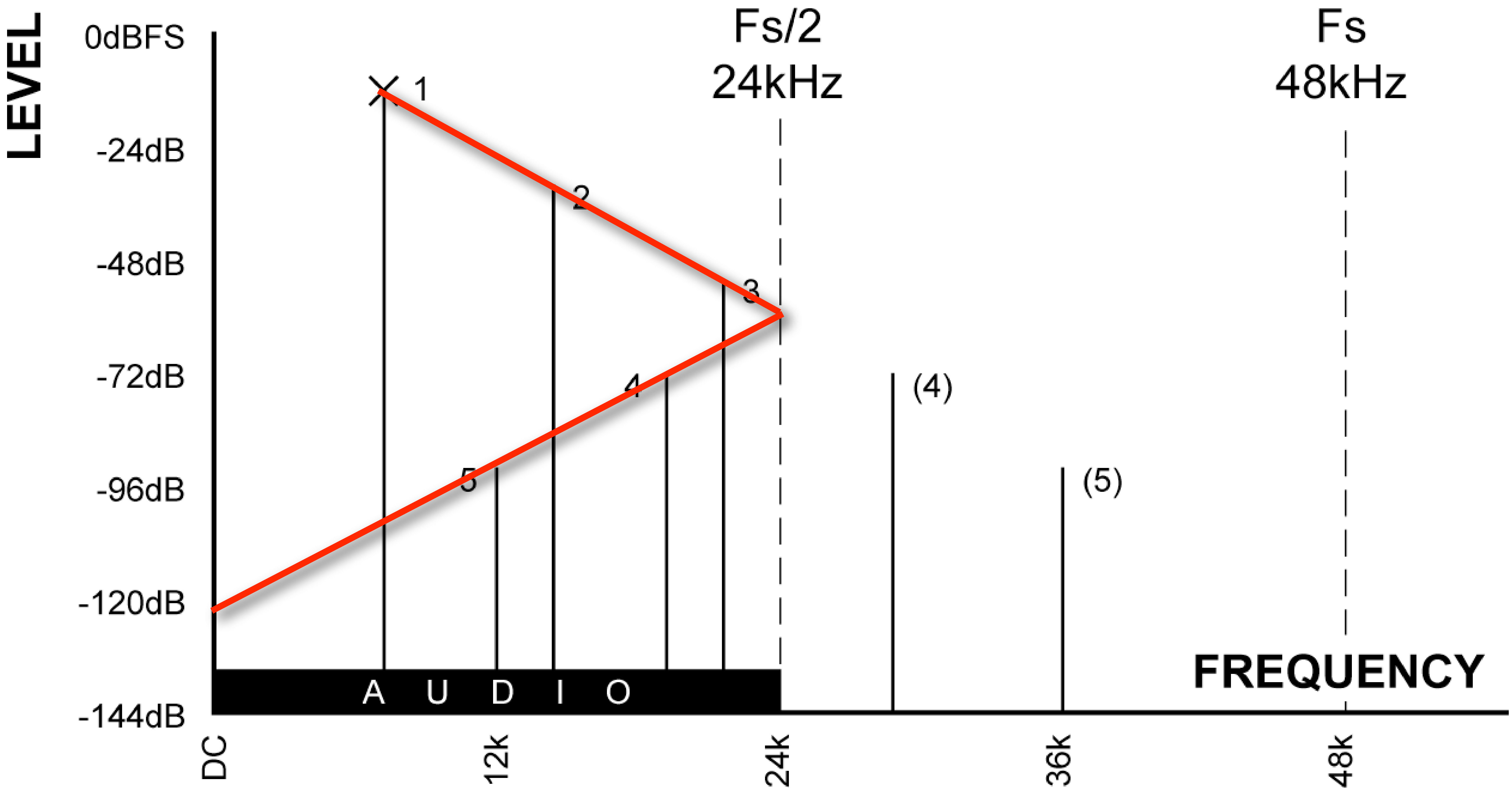
Sample Rate Conversion



Alias Distortion



Alias Distortion



Background

Overload History

It could take CD players 200-700 ms to get out of distortion latch-up.

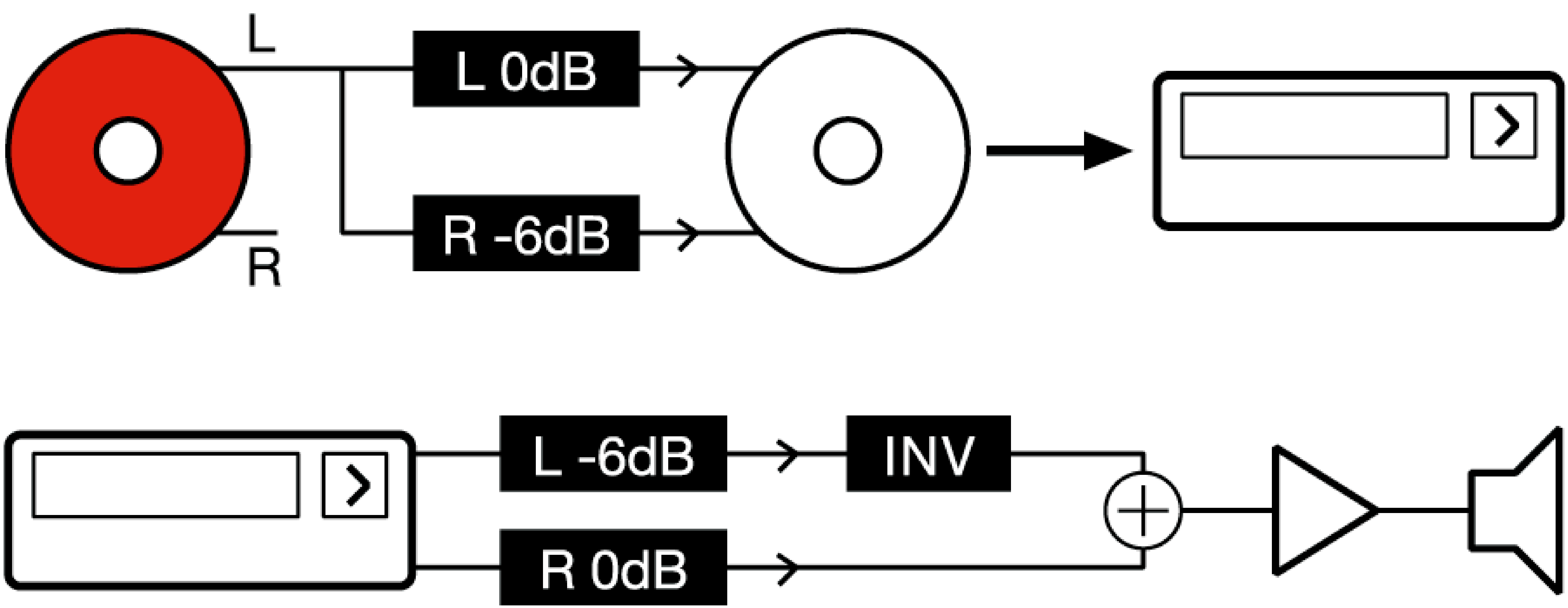
0 dBFS+ level is hit more and more frequently on new pop music releases.

AES 23 paper...

0 dBFS+

Hyper-optimized audio creates distortion and listener fatigue on CD, Film and Broadcast Commercials.

TC papers about 0 dBFS+ level and its consequences for DA, SRC's and Data Reduction Codecs available through AES.



Audio Level

0 dBFS+

0 dBFS+ level normally isn't generated unless digital processing is used.

Freshly AD converted audio generates very few such peaks, if any at all.

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CD Production

Analog
vs.
Digital

Before 1992
Analog multitrack (emphasis)
Analog interfacing, mix and processing
Mastering to 1/2” or DAT

CD Production

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What you hear is what you get
Sample counting ok

CD Production

Analog
vs.
Digital

Before 1992
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Analog interfacing, mix and processing
Mastering to 1/2” or DAT
Sample counting ok

Now
Digital recording
Digital interfacing, mix and processing
Mastering to AIFF or WAV files

CD Production

Analog
vs.
Digital

Before 1992
Analog multitrack (emphasis)
Analog interfacing, mix and processing
Mastering to 1/2” or DAT
Sample counting ok

Now
Digital recording
Digital interfacing, mix and processing
Mastering to AIFF or WAV files

You don’t see and hear what you get
Sample counting not ok

CD Production

“Desktop Audio”

Clipping and bad limiting can create a lot of alias distortion

CD Production

"Desktop Audio"

Clipping and bad limiting can create a lot of alias distortion

Clipping may happen
- on the mix buss of a DAW
- in plug-in processing
- imported from sample libraries

and may not be seen or heard

CD Production

"Desktop Audio"

Clipping and bad limiting can create a lot of alias distortion

Clipping may happen
- on the mix buss of a DAW
- in plug-in processing
- imported from sample libraries

and may not be seen or heard

DDD is not a sign of quality

Broadcast

Ingest

Digital inputs or file transfers can fool sample based level measures to underestimate the true peak level of, for example, commercials

Data Reduction

AES23 Distortion.pdf (13 Pages)

Algorithm	*Mode*	*Datarate [kbit/s]*	*Avg. per ch. [kbit/s]*	*Max. peak re. 0.5*
MPEG-1 L II	stereo	384	192	+1.3 dB
MPEG-1 L II	stereo	224	112	+1.3 dB
MPEG-1 L III	stereo HQ	320	160	+1.7 dB
MPEG-1 L III	stereo HQ	160	180	+2.3 dB
MPEG-1 L III	int-st HQ	128	64	+5.3 dB
MPEG-1 L III	int-st fast	128	64	+3.0 dB
MPEG-1 L III	int-st HQ	96	48	+4.7 dB
MPEG-2 L III	22.05 kHz, i-st HQ	80	40	+1.7 dB
DTS	6 ch.	1234	206	+0.6 dB
Ogg Vorbis	stereo	var., Q=10	157-193	+0.3 dB
Ogg Vorbis	stereo	var., Q=5	49-64	+1.8 dB

Table 4: *Maximum peak values observed in 12 hot CD excerpts (length 14-33 s) perceptually coded with various algorithms, data rates and modes.*

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MPEG-1 L III	int-st HQ	128	64	+5.3 dB
MPEG-1 L III	int-st fast	128	64	+3.0 dB
MPEG-1 L III	int-st HQ	96	48	+4.7 dB
MPEG-2 L III	22.05 kHz, i-st HQ	80	40	+1.7 dB
DTS	6 ch.	1234	206	+0.6 dB
Ogg Vorbis	stereo	var., Q=10	157-193	+0.3 dB
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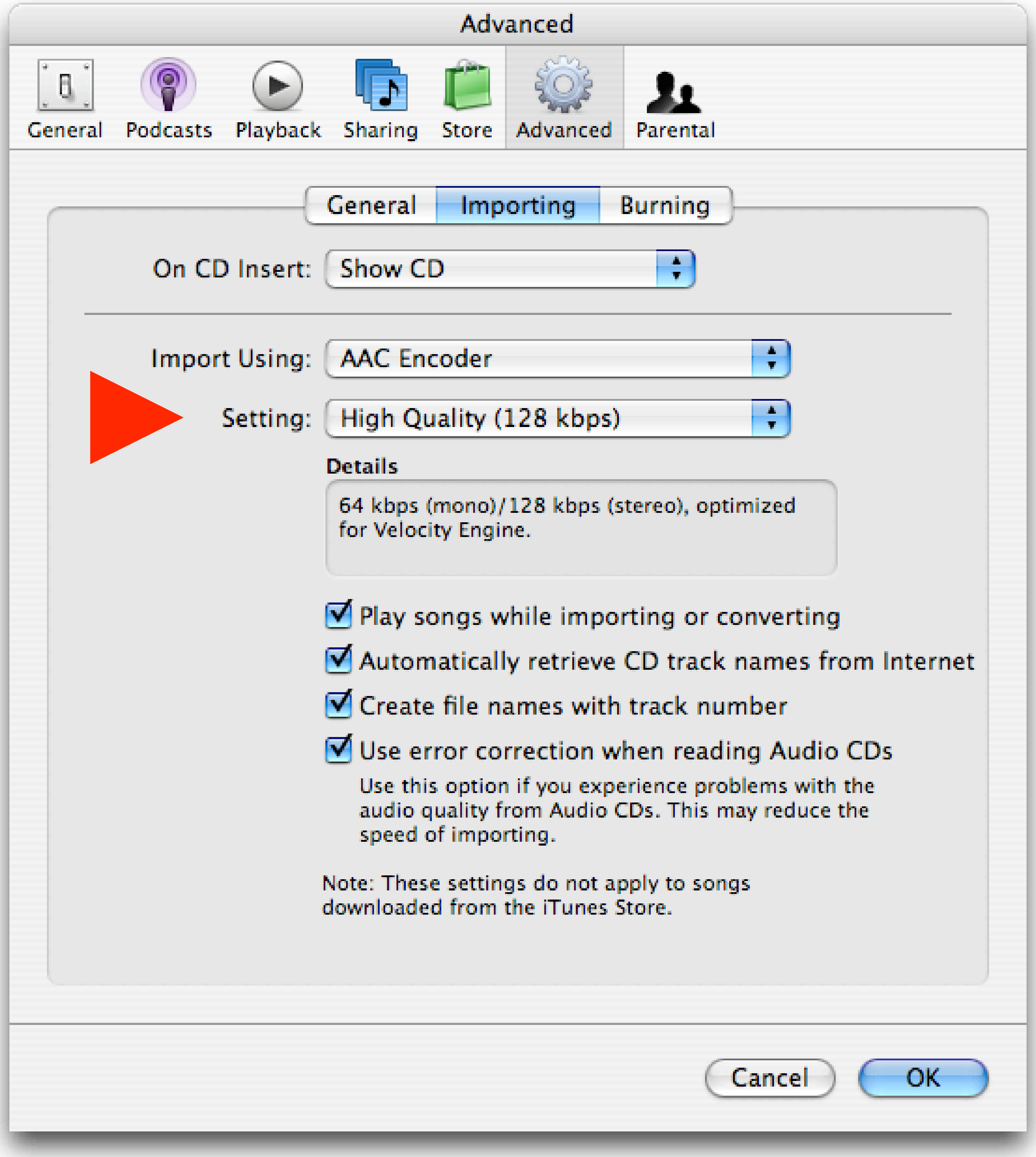
Table 4: *Maximum peak values observed in 12 hot CD excerpts (length 14-33 s) perceptually coded with various algorithms, data rates and modes.*

iTunes import

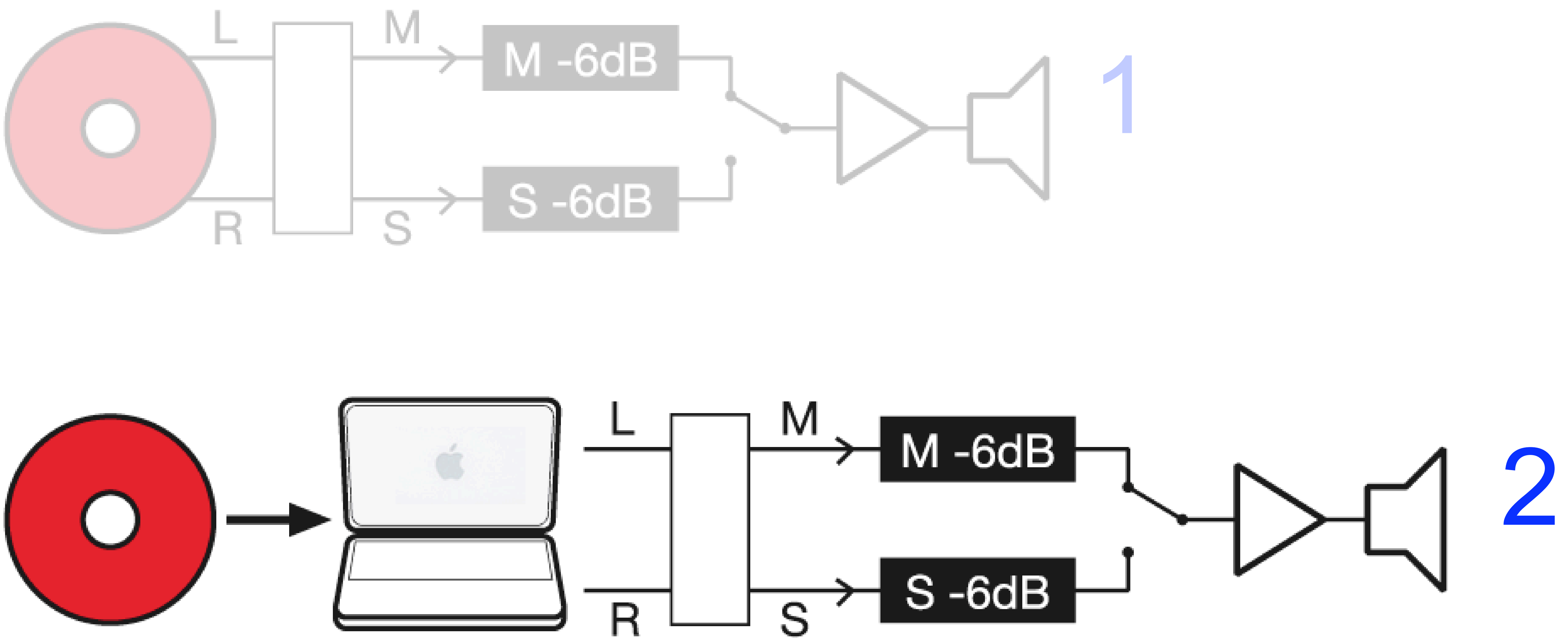
Encoder default is AAC at 128 kbps, joint stereo.

"High Quality"?



iTunes Codec Listening

iTunes Codec Listening



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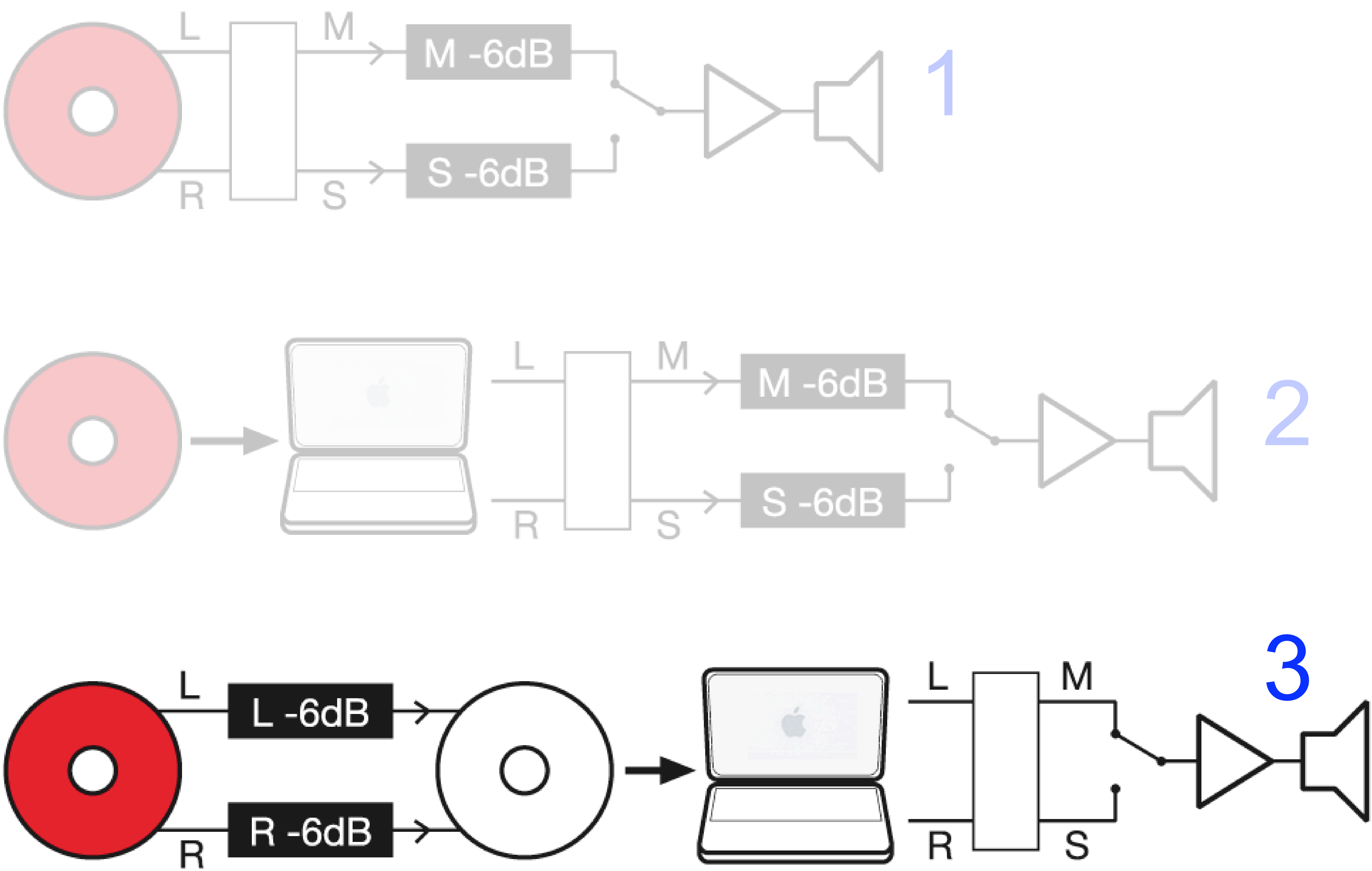
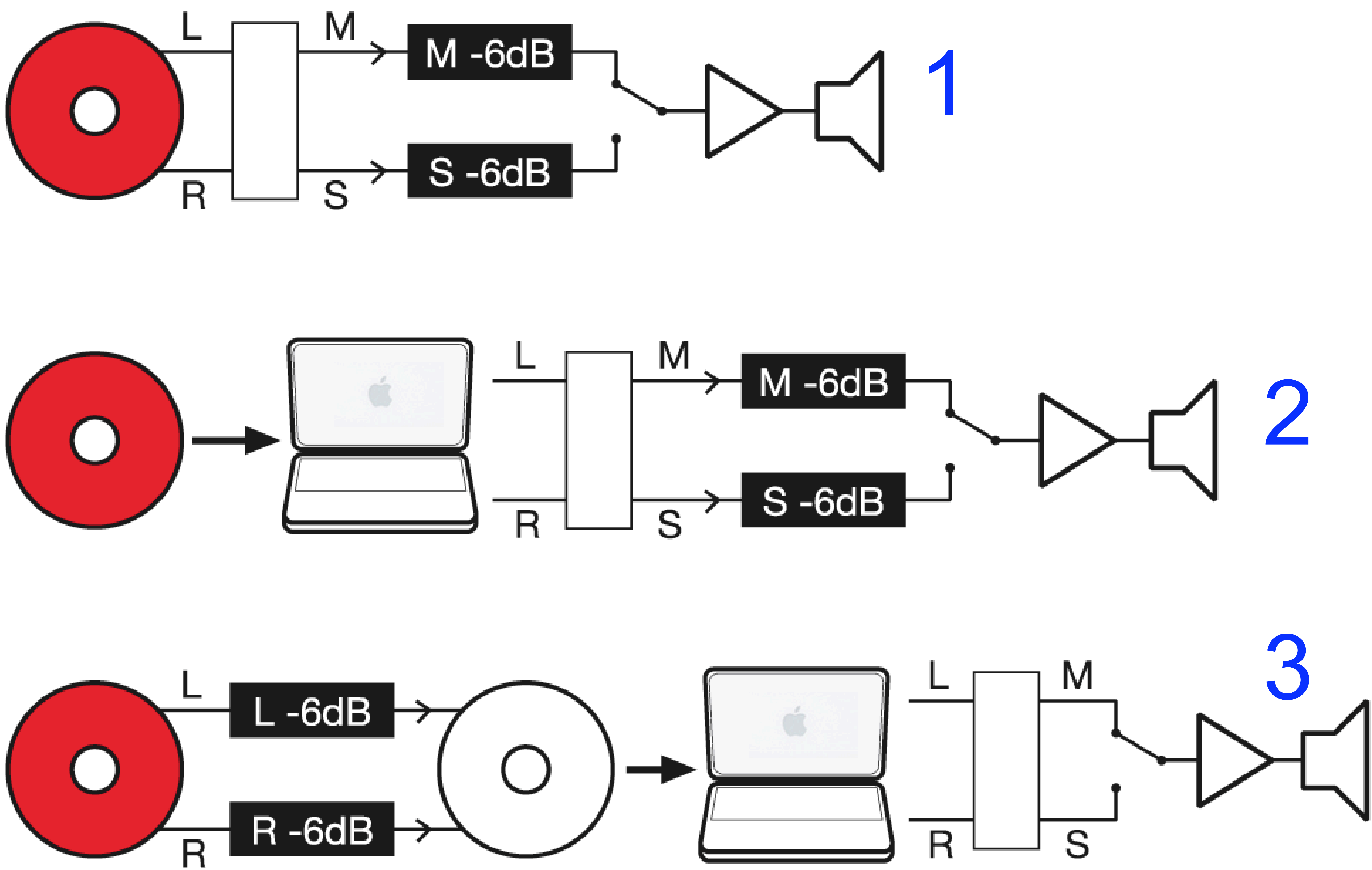


iTunes Codec Listening

Stop Counting Samples



Data Reduction

Is it really necessary?

Bandwidth is going up. Why risk future compatibility and audio quality?

Data reducing or SR converting pop music without attenuation leads to distortion in radio station archives

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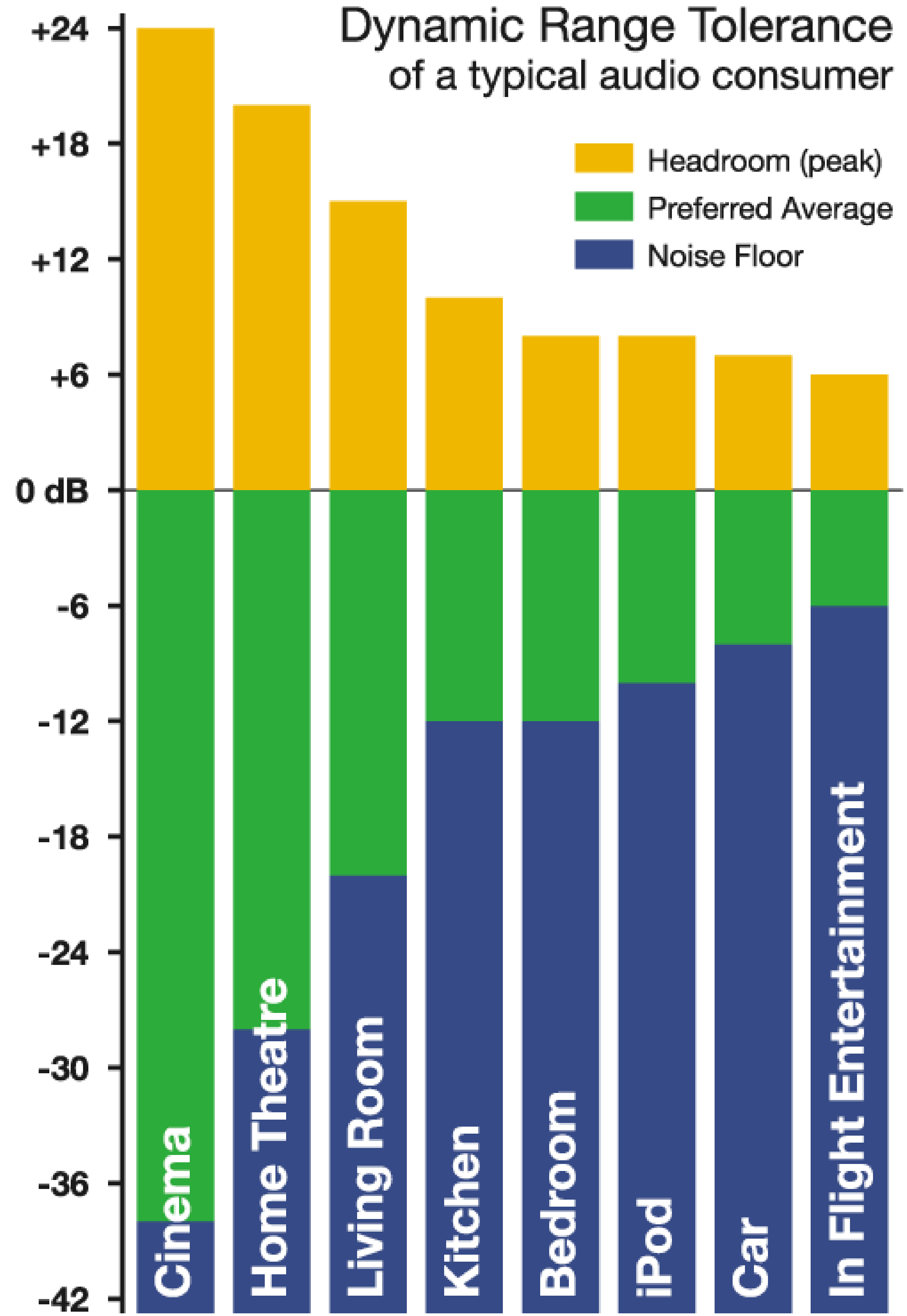
Peak Level

Safety Limit Guidelines

PPM based restriction at -9 dBFS
Sample based restriction at -3 dBFS
Signal based restriction close to FS

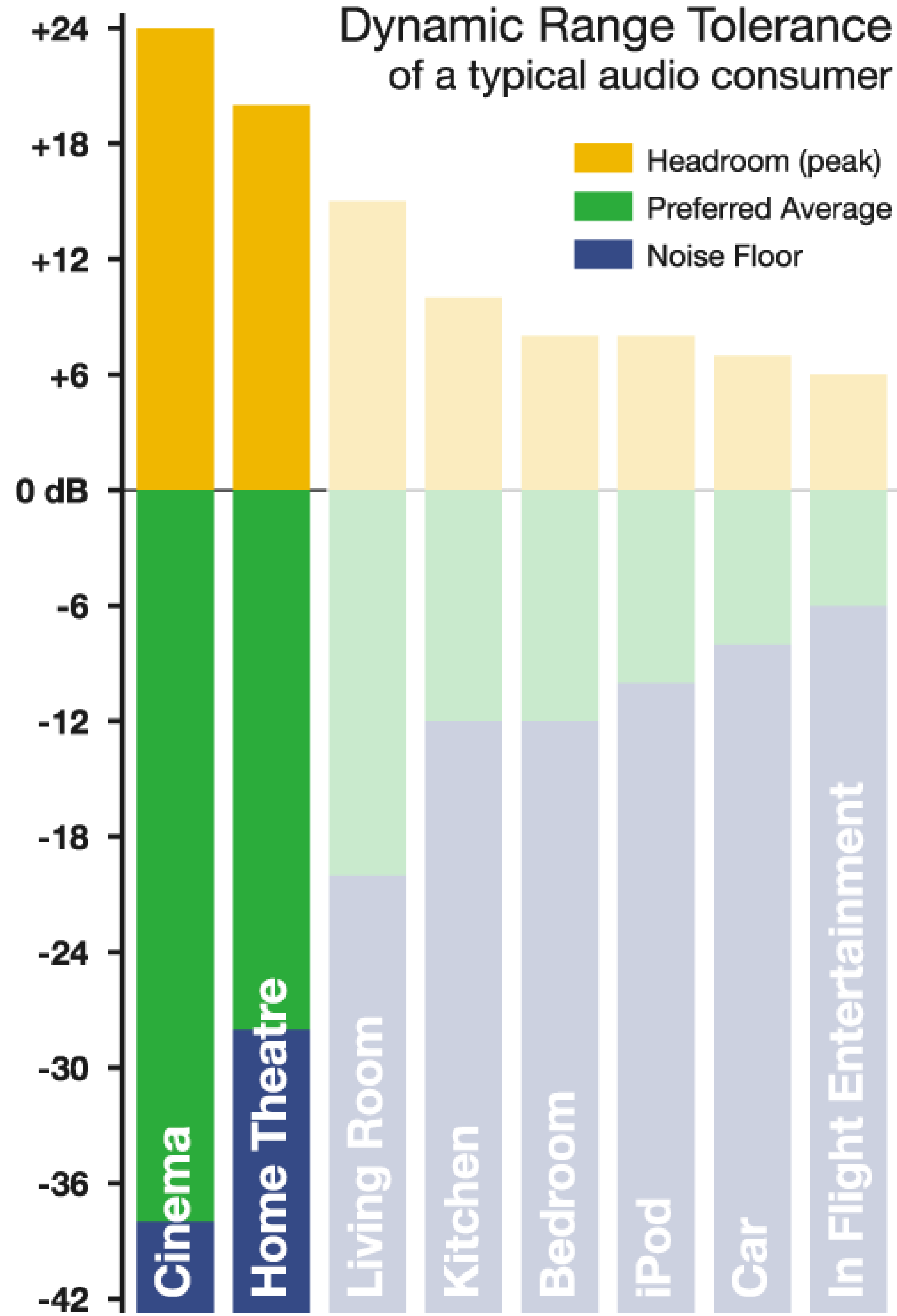
D R T

The ideal dynamic range of program depends on the listener's situation.



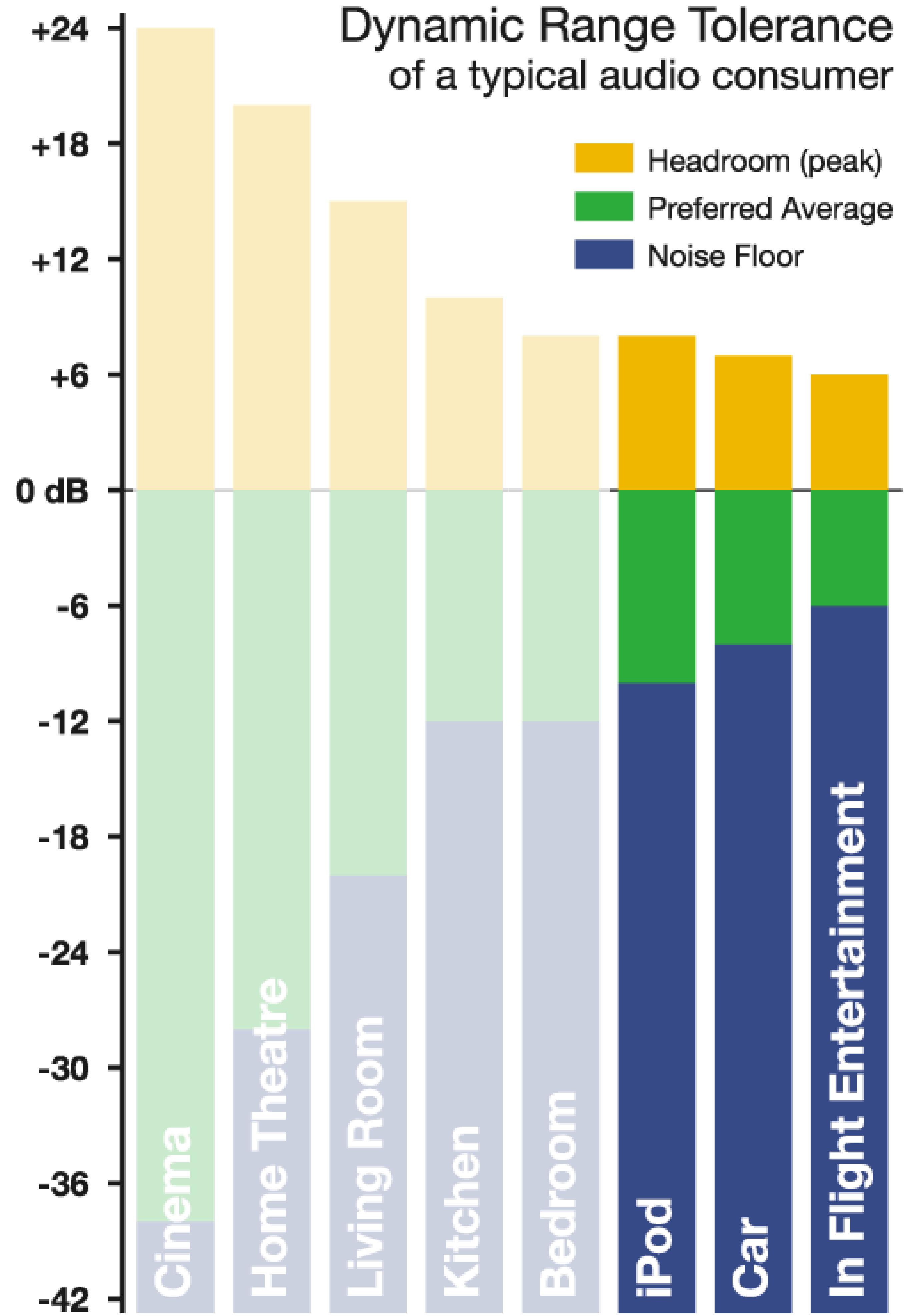
D R T

A wide dynamic range is desirable in Cinema, on DVD, and for digital Classical Music broadcast.



D R T

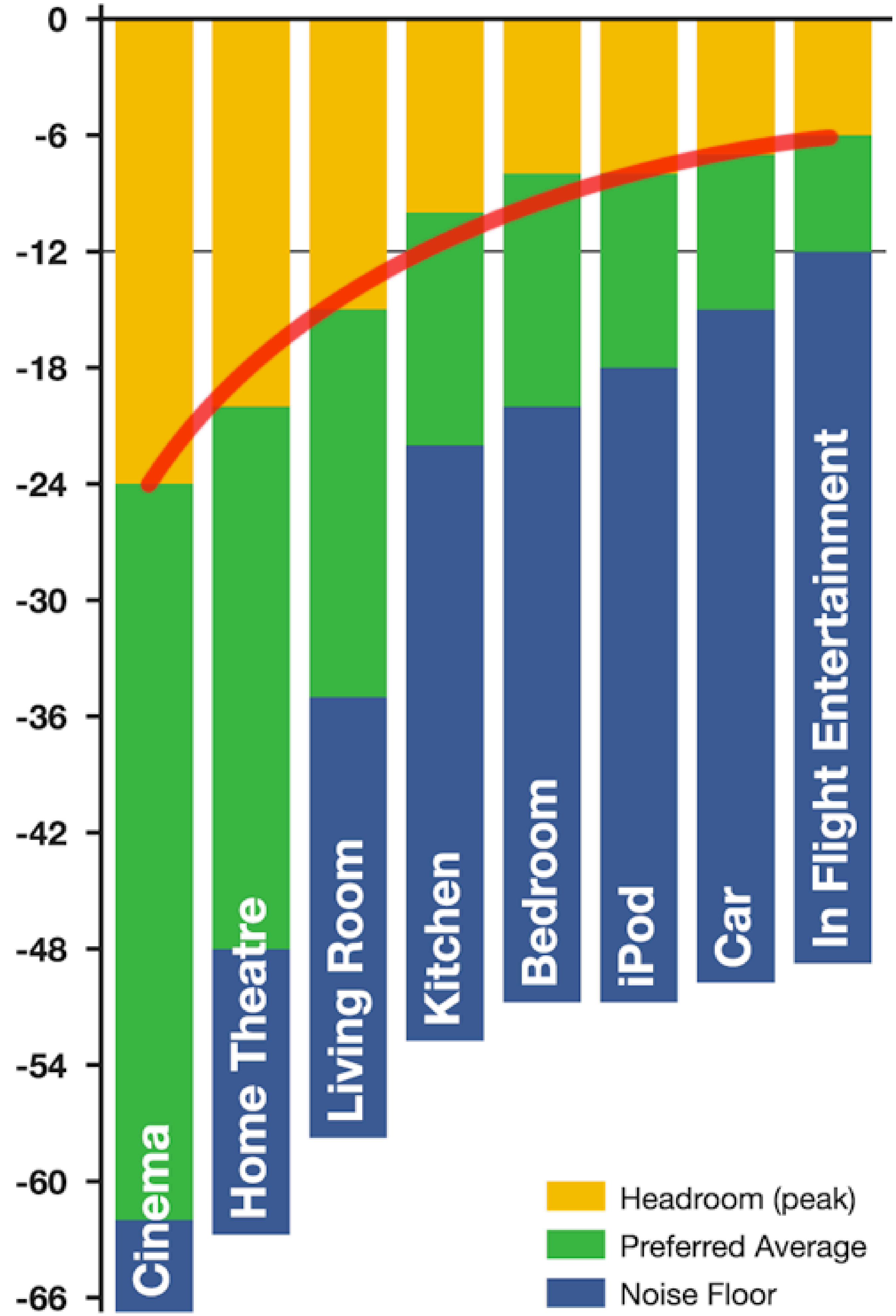
A narrow dynamic range is desirable where background noise is high.



Peak Level Normalization

Low dynamic range material end up loud when level control is based only on a peak level measure.

That’s what happened to CD.



Loudness

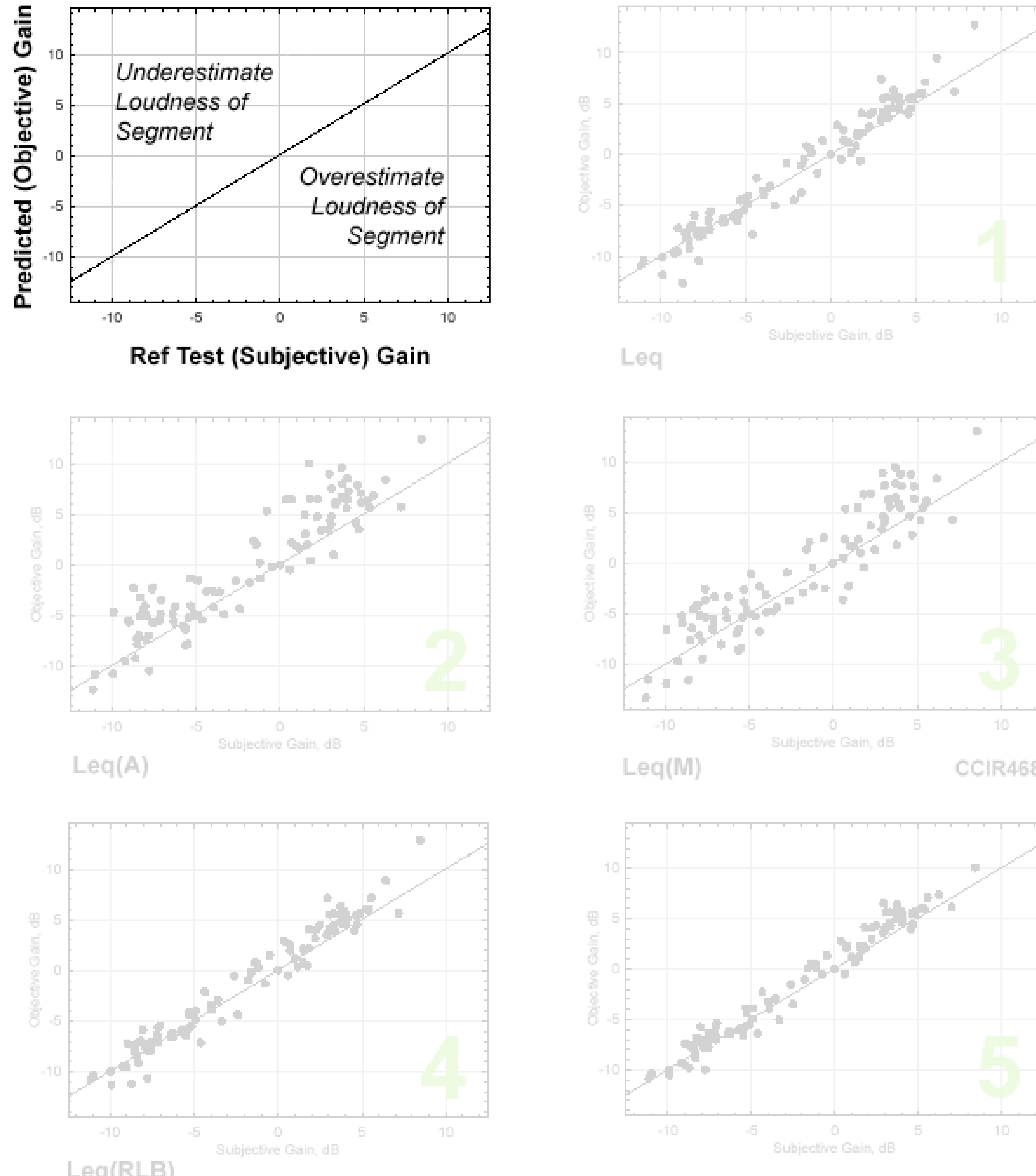
Unlike Level, Loudness is subjective. Listeners weigh a number of factors differently:

- Sound Pressure Level
- Frequency Contents
- Duration

Variability between listeners: BLV (Age, Culture, Gender etc.)
Variability within the same listener: WLV (Mood, Focus etc.)

A Loudness measure must be based on statistics.

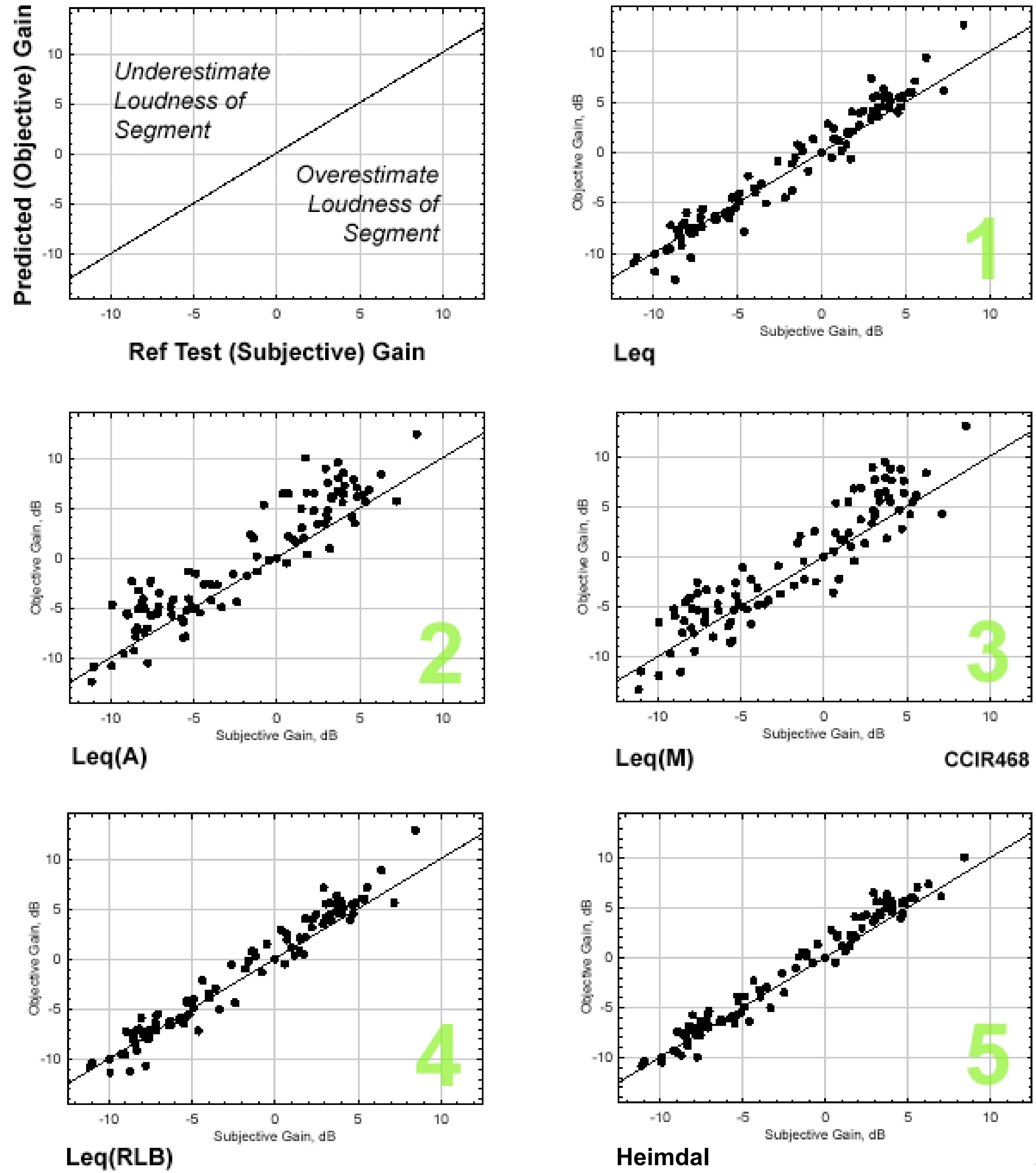
Comparison Scatter Plots



2

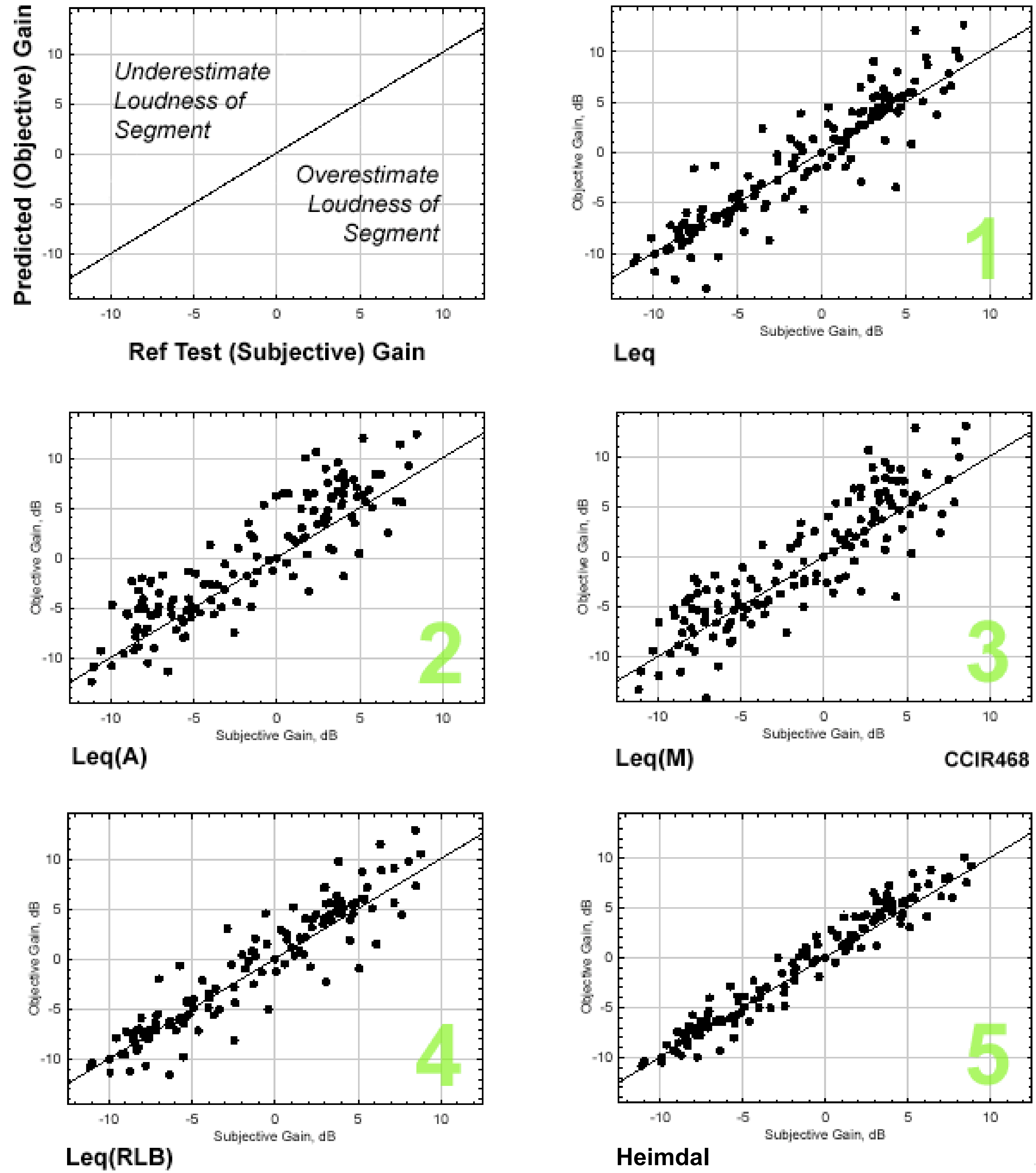
Comparison Scatter Plots

Uniform material, mainly compressed speech, as ITU test



Comparison Scatter Plots

As before...
plus more music
plus less processed
speech segments



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Comparison Scatter Plots

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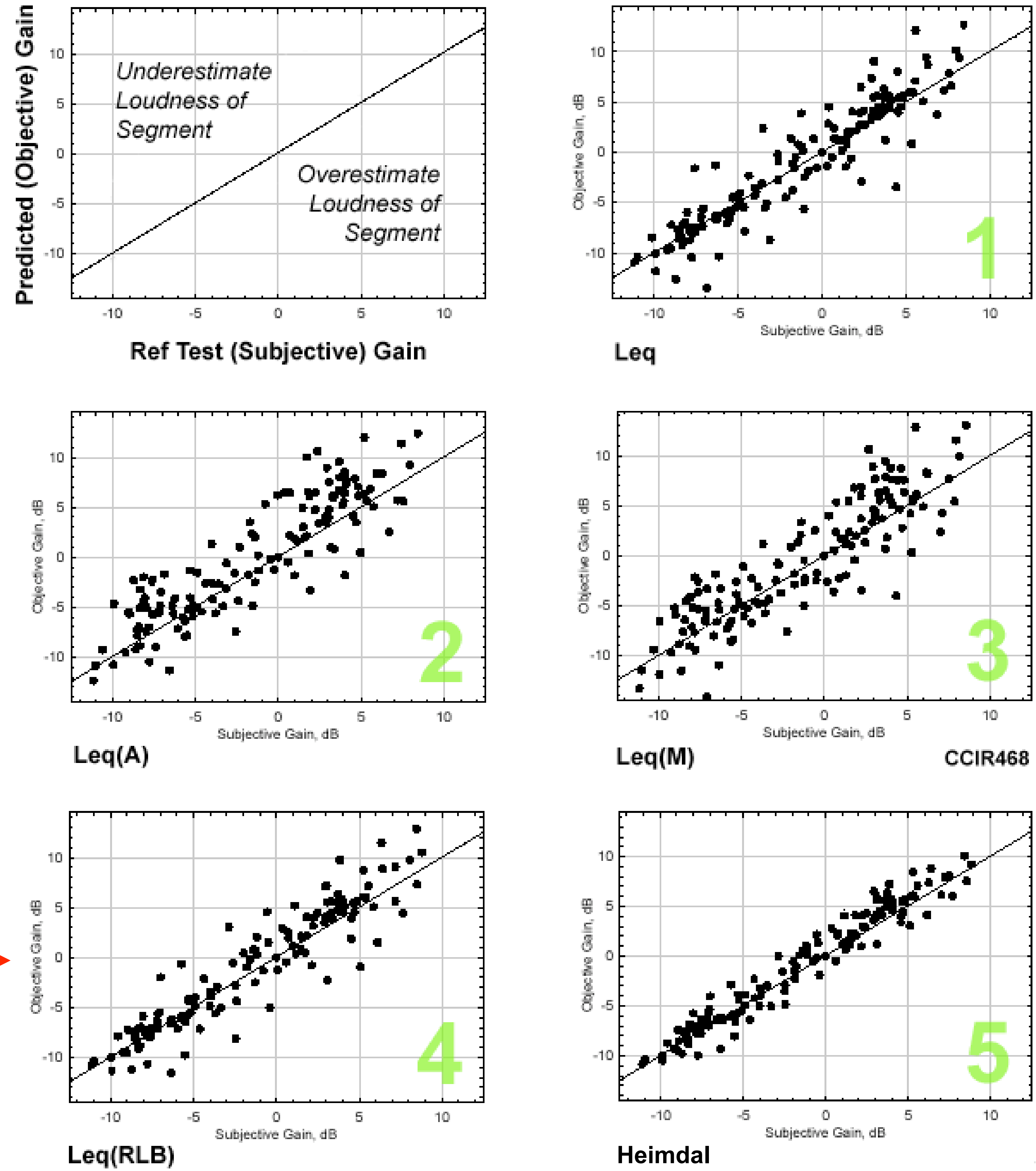

2

BS.1770

Loudness

A *baseline* measure based on an update of Leq(RLB), Leq(R2LB), has already been drafted.

Short-term aspects and realtime use to be further investigated.

BS.1770

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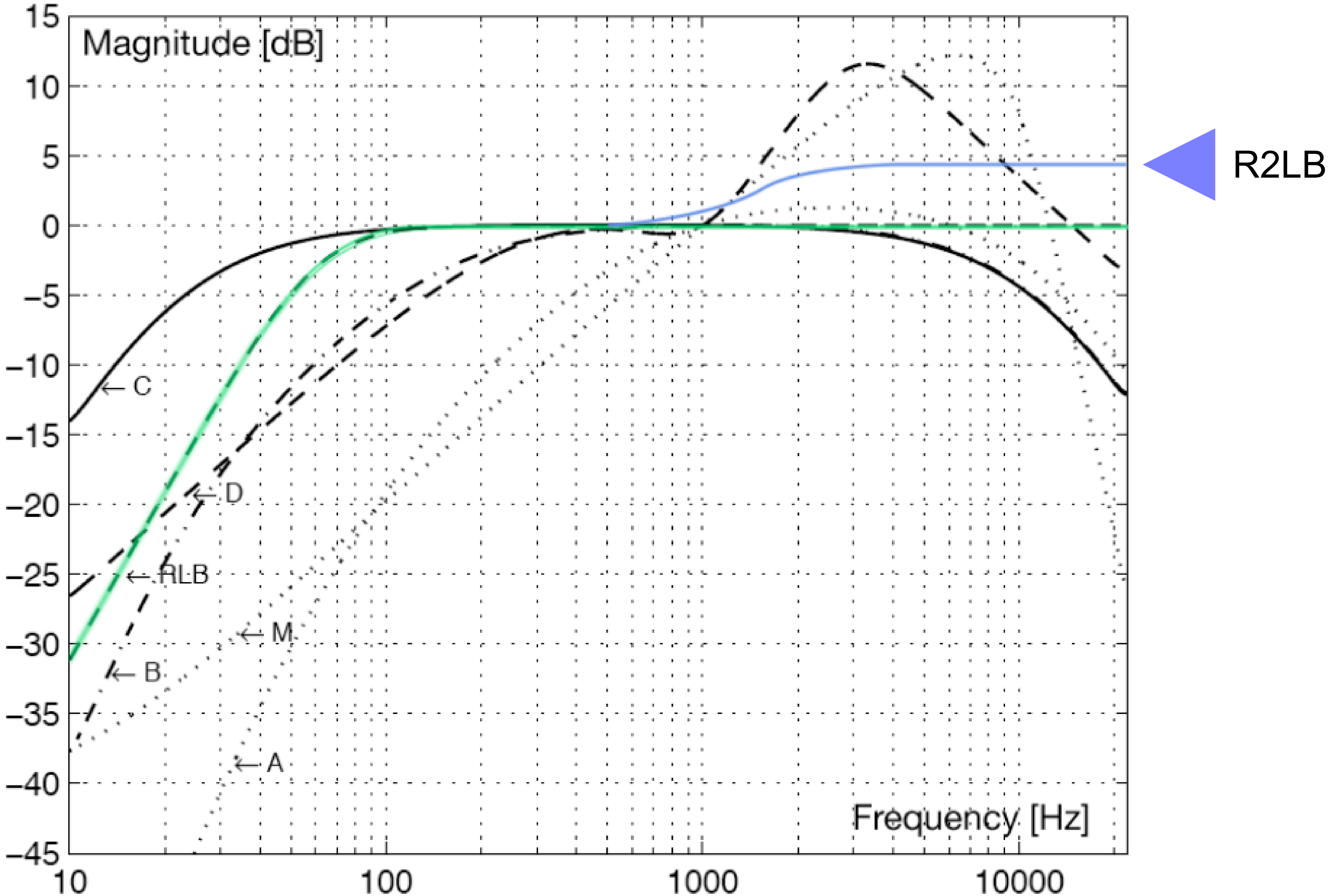
Peak Level

SC-02-01: Digital “true peak” level measure based on over-sampled detection.

Leq Weighting Curves



True Peak Meter

Quote from SC-02-01 report

maximum under-read (in dB) = $20.\log(\cos(\pi.f_{norm}/n))$

This equation was used to construct the following Table, which probably covers the range of interest:

Over-sampling ratio	**Under-read (dB) maximum f_{norm} = 0.45**	**Under-read (dB) maximum f_{norm} = 0.5**
4	0.554	0.688
8	0.136	0.169
10	0.087	0.108
12	0.060	0.075
14	0.044	0.055
16	0.034	0.042
32	0.008	0.010

Loudness

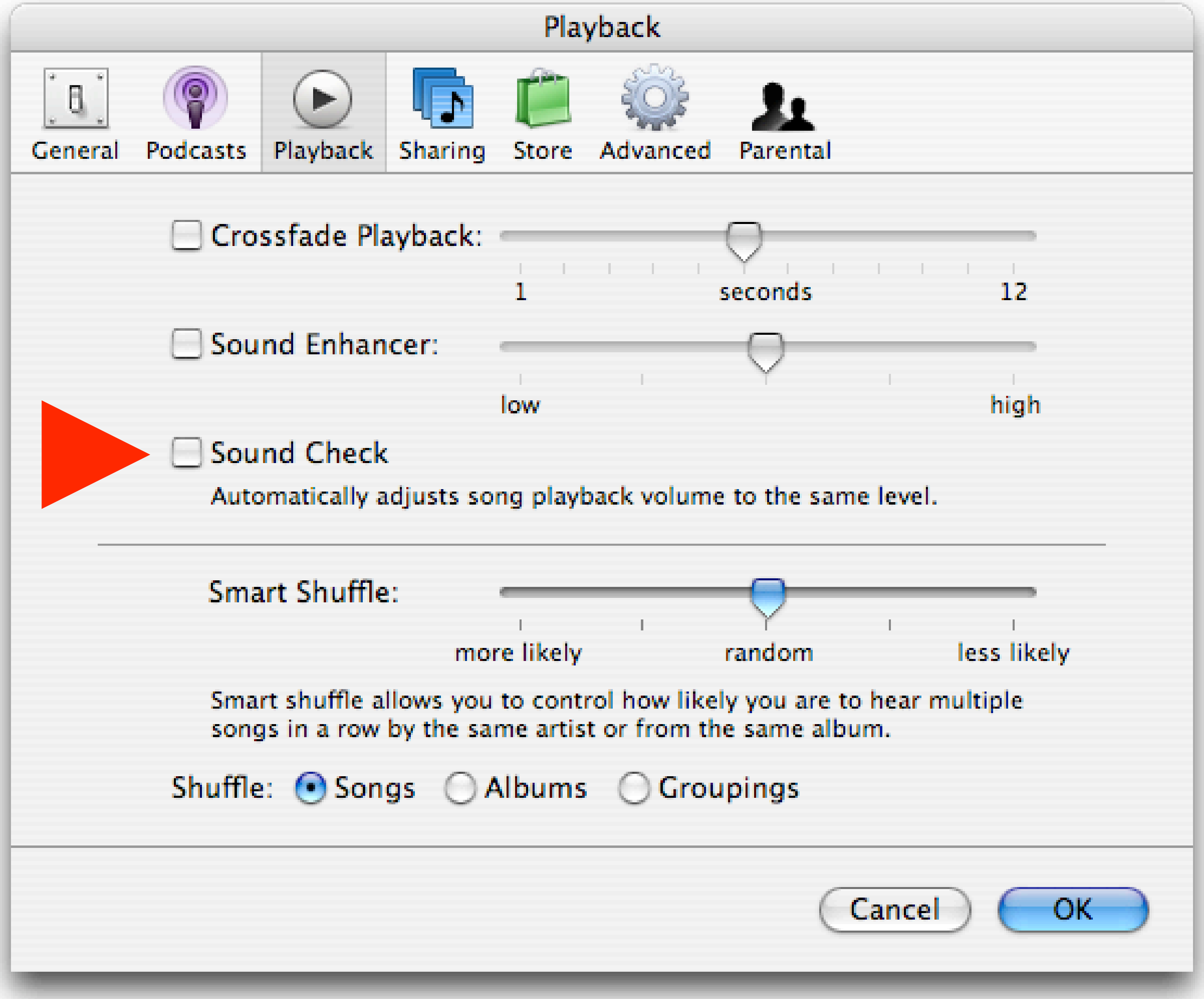
Soon, Loudness control at the consumer will take away the level advantage from hyper compressed and clipped audio...

...but its distortion will remain forever.

iTunes is just the first step.

iTunes Playback

Even a crude loudness check is better than none

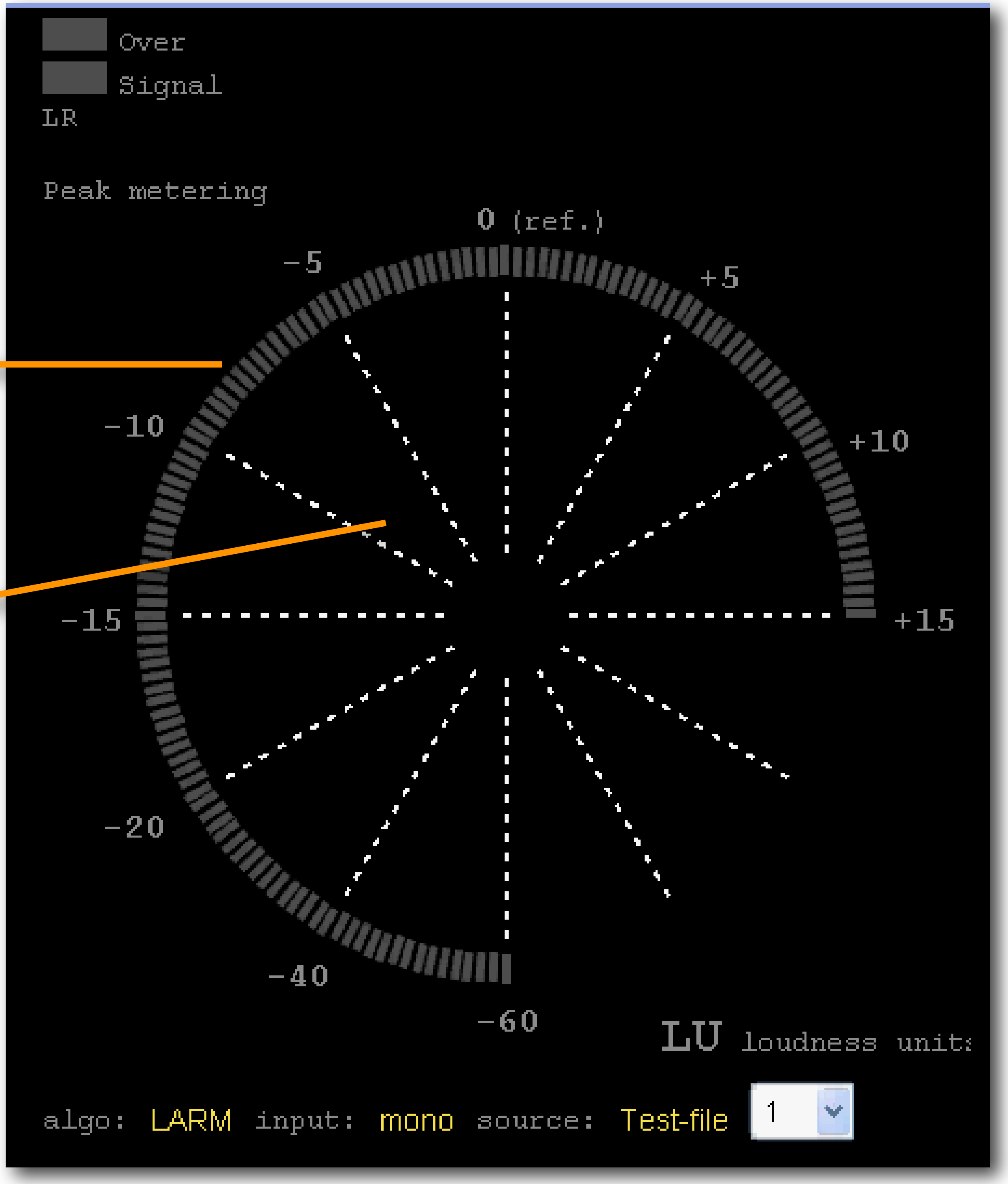


Loudness Meter Study

Short-term loudness in the outer ring.

Loudness history in the radar view.

Currently set for one revolution per minute.



Radar View

Loudness history in the radar view.

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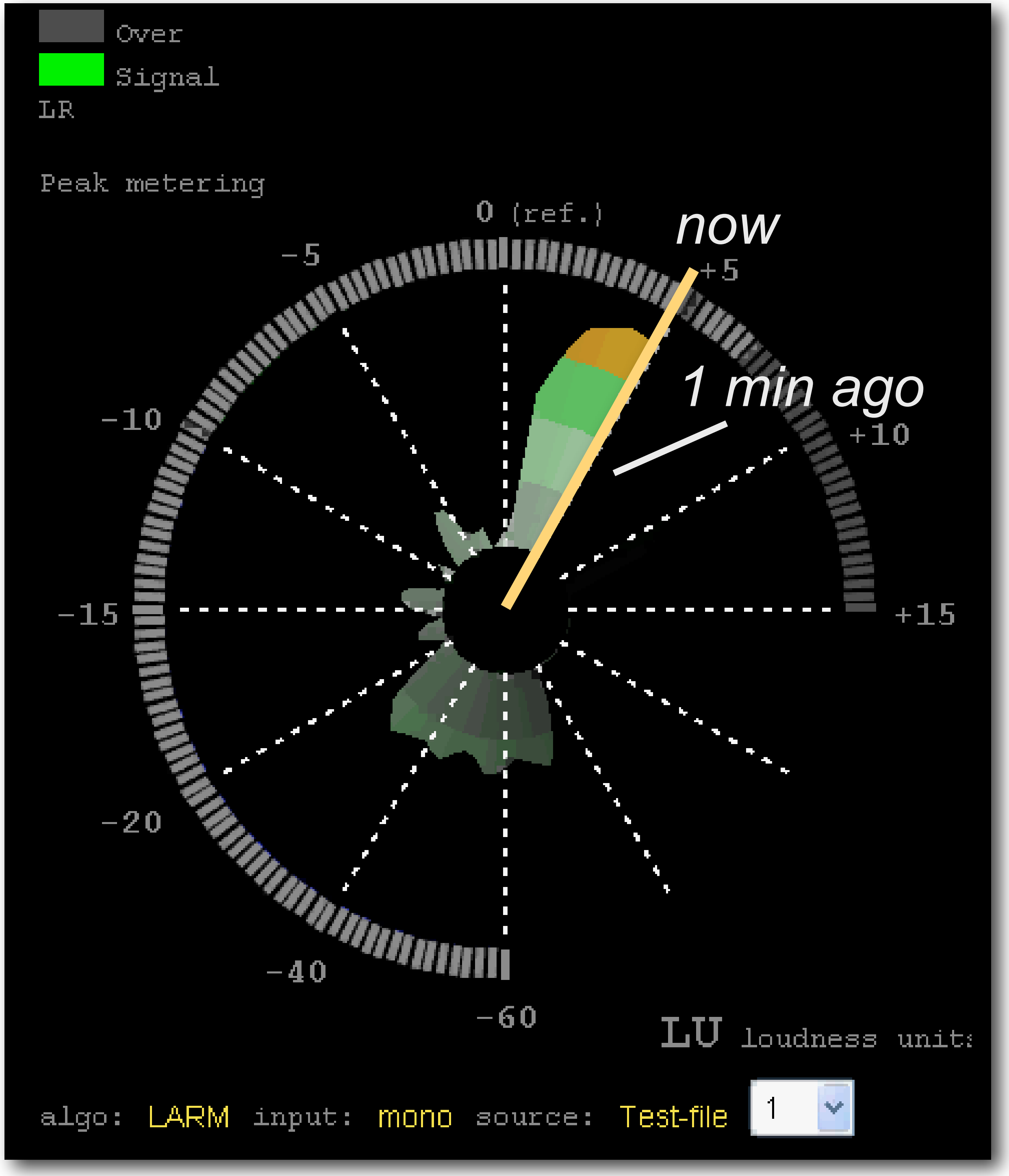

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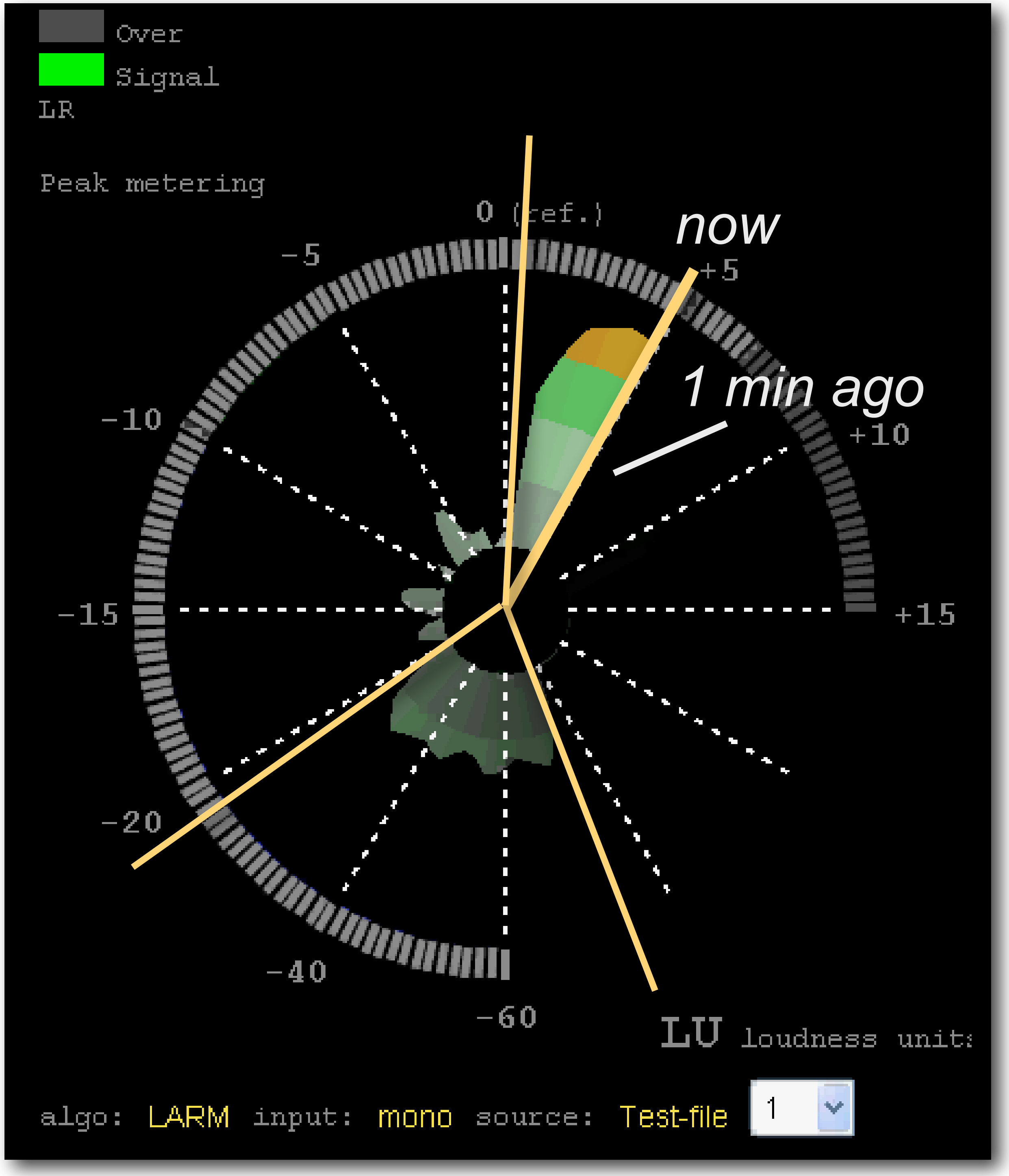

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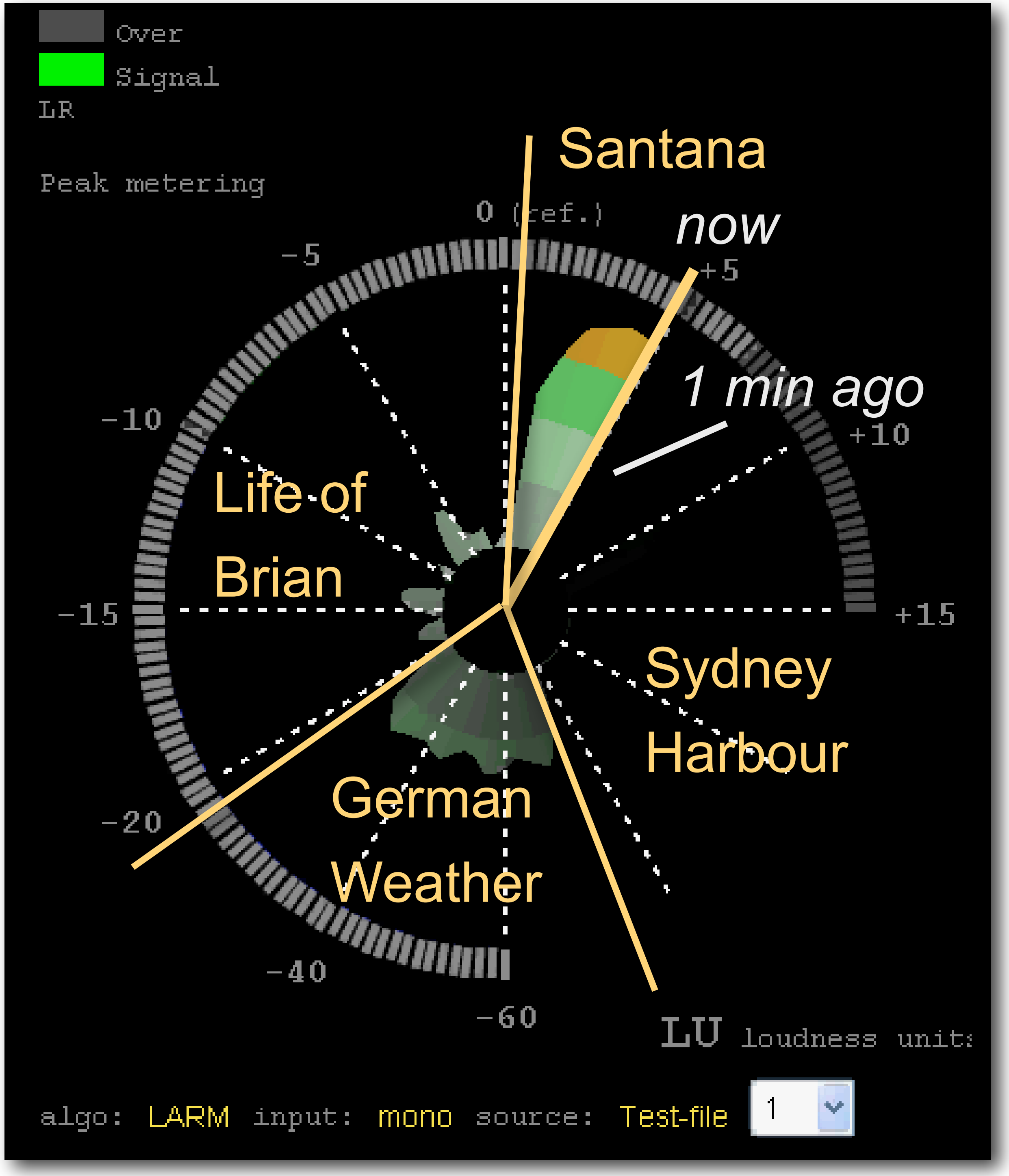

Hot will get caught

Loudness

The basic Loudness measure will keep improving

Hot will get caught

Loudness

The basic Loudness measure will keep improving

Realtime Correction

Broadcasters and consumers will be able to over-attenuate excessively loud material, if they wish

Hot will get caught

Loudness

The basic Loudness measure will keep improving

Realtime Correction

Broadcasters and consumers will be able to over-attenuate excessively loud material, if they wish

The Signal/Sample ratio can be used as a quick identifier of hot material, and level the field between analog vs. digital interfacing and file transfers

Conclusion

Mix and Normalize to -3 dBFS

Conclusion

1. Mix and Normalize to -3 dBFS

Respect digital rules: Don’t clip

Conclusion

1. Mix and Normalize to -3 dBFS
2. Respect digital rules: Don’t clip

Use low level dynamics processing

Conclusion

1. Mix and Normalize to -3 dBFS
2. Respect digital rules: Don’t clip
3. Use low level dynamics processing

Use upsampled limiting

(or process in the analog domain)

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1. Mix and Normalize to -3 dBFS
2. Respect digital rules: Don’t clip
3. Use low level dynamics processing
4. Use upsampled limiting
   (or process in the analog domain)

Use upsampled metering

Thanks SC-02-01!

Conclusion

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2. Respect digital rules: Don’t clip
3. Use low level dynamics processing
4. Use upsampled limiting
   (or process in the analog domain)
5. Use upsampled metering

Use loudness calibrated speakers

Conclusion

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6. Use loudness calibrated speakers

Data reduced delivery: Lower level

Conclusion

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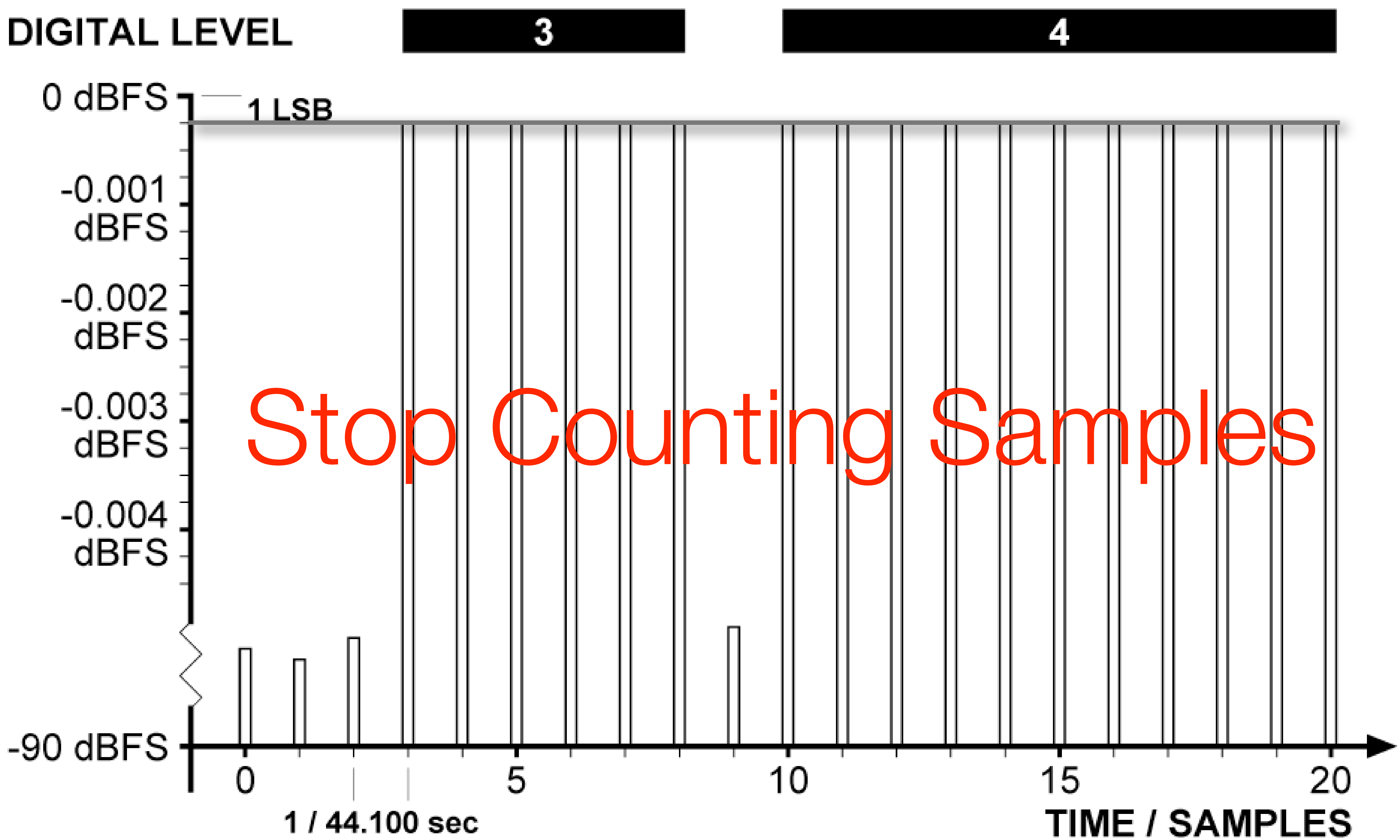
3. Use low level dynamics processing
4. Use upsampled limiting
   (or process in the analog domain)
5. Use upsampled metering
6. Use loudness calibrated speakers
7. Data reduced delivery: Lower level

A Loudness advantage will Vanish,
The Distortion will Remain

Protect The Music



Conclusion

References

ISO 226, ISO 532, MPEG4, MPEG7

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Psychoacoustics - Facts and Models

Dunn, 2000 (Audio Precision paper)
Digital Filter Overshoot and Headroom

Moore et al. December 2003 (JAES no 12)
Why are Commercials so Loud?

Nielsen & Lund, 1999 - 2003 (AES 107, 109, 111, 23 reg.)
0 dBFS+ Level in Mastering and Audio Production

Skovenborg & Nielsen, 2004 (AES 117)
Evaluation of Different Loudness Models with Music and Speech

Thomas Lund
TC Electronic A/S
Risskov, Denmark

End