# 0dBFS+ Levels in Digital Mastering

Søren H. Nielsen and Thomas Lund
TC Electronic A/S
DK-8240 Risskov, Denmark

Presented at
the 109th Convention
2000 September 22-25
Los Angeles, California, USA

AES

*This preprint has been reproduced from the author's advance manuscript, without editing, corrections or consideration by the Review Board. The AES takes no responsibility for the contents.*

*Additional preprints may be obtained by sending request and remittance to the Audio Engineering Society, 60 East 42nd St., New York, New York 10165-2520, USA.*



**AN AUDIO ENGINEERING SOCIETY PREPRINT**

# 0dBFS+ Levels in Digital Mastering

SØREN H. NIELSEN AND THOMAS LUND

TC Electronic A/S
Sindalsvej 34, DK-8240 Risskov, DENMARK
soerenn@tcelectronic.com, thomasl@tcelectronic.com

A sine tone at 0dBFS is often believed to be the maximum level obtainable from a digital medium. Therefore it is typically the maximum level digital filters and analog circuitry in consumer equipment is aimed at reproducing.

As we have showed in previous papers, inter-sample peaks may be considerably higher than 0dBFS.

This paper examines the sonic consequences when 0dBFS+ signals are reproduced in typical consumer equipment. The performance of a variety of domestic CD players exposed to such signals are presented and evaluated.

## 0. INTRODUCTION

Several golden ears in the pro audio industry tend to believe that the best sound in pop / rock music generally was produced between 1982 and 1995.

Despite higher resolution in converters and DSP, lower jitter and probably a better overall understanding of digital media, we seem to be on a declining rather than inclining sound quality slope these years; even though people buying records and film may not be aware of it.

Obviously there could be many reasons for this we cannot directly influence: Trends, basic recording and microphone placement skills, more semi-pro equipment being used, shorter production times and therefore less attention to detail etc.

But if the public do not care, why should we?

Because pride in our industry, craftsmanship and conservation of talent tell us to be concerned. And because more bits, more resolution and more channels can only be justified by the end quality and listener involvement going up.

Being a supplier of equipment for professional music and film production, TC Electronic therefore has a continued interest in discussing goals and rules for the production and mastering process with a pronounced focus on quality.

In this paper we have investigated millennium sound quality from a level point of view. Even though distortion in a linear digital audio system is generally lower at high levels, there may be situations where domestic equipment is not capable of reproducing hot signals created in a mix or mastering process.

As we have shown in our paper at the 107th convention, Level Control in Digital Mastering [1], such level peaks are readily derived from mastering tapes conforming to normal rules of permitting a number of consecutive samples at full scale, or even without a single sample hitting 0 dBFS.

Areas in which 0 dBFS+ levels could be of concern are discussed, including professional equipment for domain and sample rate conversion, data compression encoders and decoders plus, most noticeably, end user reproduction equipment.

We will disclose our findings of how various consumer CD players react to hot levels and discuss if this should have an influence on how digital audio is measured and mastered.

We will also investigate whether signals fulfilling the sample rate criteria versus artificial or DSP generated signals should be looked upon differently, or if common measurement guidelines and principles could be attained.

Finally we will discuss if the findings give us reason to continue to work on a DRA (Dynamic Range Approval) draft as suggested in the previous paper.

## 0.1 Definition of 0dBFS+ Level

In the digital domain the peak level may deviate from the peak level in the analog domain.

There are two main reasons for this [1]:

1. Basic sampling theory. Sampling occurs at regular intervals, and at frequencies near integer fractions of fs, such as fs/4 and fs/2, the phase of the signal compared to the sampling times may generate a digital peak value somewhat below the analog peak value. If the signal is not exactly at one of the critical frequencies mentioned above, the peak value in the digital domain will get very close to the analog peak value. If the analog signal prior to sampling was properly bandwidth-limited, the output after digital to analog conversion will be substantially equal to the analog input signal.

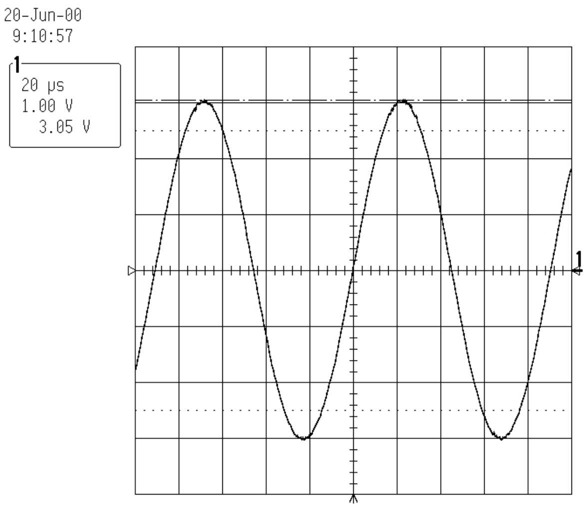



*Figure 1*

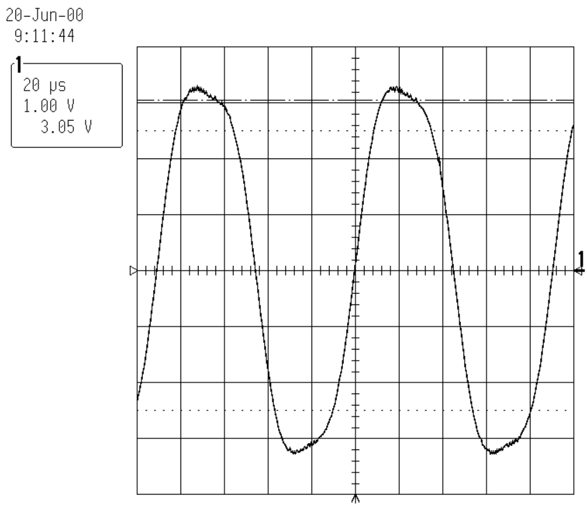


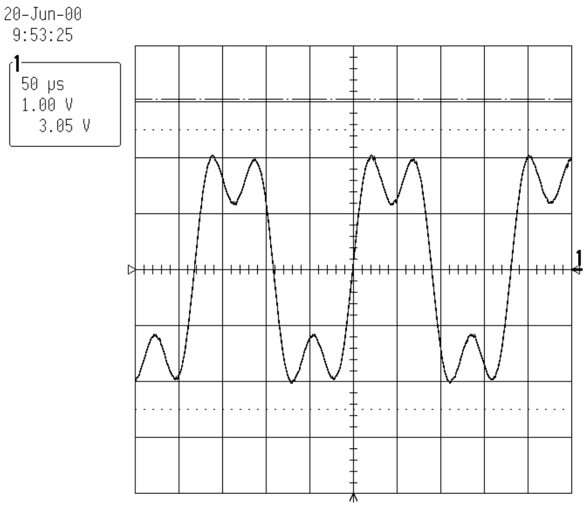

*Figure 2*

*11025 Hz sine waves at full scale (0 dBFS) sampled @ 44.1 kHz.*
*Example shows consumer CD player, NAD C 520, measured on LeCroy 9350A digital oscilloscope.*

*Figure 1: Starting phase of 90°. Analog and digital peak values are identical.*
*Figure 2: Starting phase of 45°. Analog peak value should be +3 dBFS. Notice the clipping.*

2. Gibb's phenomenon. Occurs when limiting the bandwidth of a wide-band signal (or truncating an impulse response). This is particularly important when the signal is clipped in the digital domain, but it

applies generally. What happens is that a square wave (or hard clipped signal) can be viewed upon as a sum of individual sine waves of frequencies 1, 3, 5,... times the fundamental frequency. The flat top of the square wave depends on the presence of all harmonics at the right levels and phases. If some of the harmonics are removed by lowpass filtering, the peak value of the signal rises. When converting from digital to analog a low pass filter is always applied, so the analog level may be higher than expected.



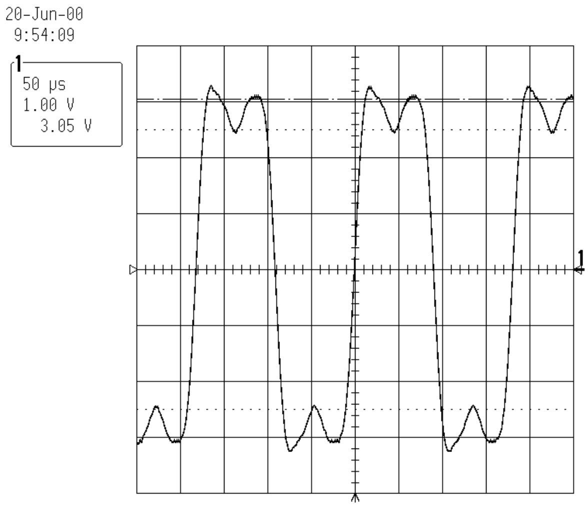




*Figure 3* *Figure 4*

*5512.5 Hz square waves sampled @ 44.1 kHz.*
*Example shows consumer CD player, NAD C 520, measured on LeCroy 9350A digital oscilloscope.*

*Figure 3: -6 dBFS. Notice the dip at the flat top due to lowpass filtering at the output. Only the third harmonic fits within 22.05 kHz.*

*Figure 4: Digital full scale. Notice the clipped peaks. They are supposed to be twice as high as in Figure C but clearly there is no level capabilities above a full scale sine wave.*

In this paper we will refer to signals that may typically be reconstructed at a higher peak level than a sine wave asynchronous to the sample rate for “0dBFS+” levels.

# 2. CRITICAL ELEMENTS OF THE SIGNAL PATH

We have not systematically investigated how performance is affected in all the types of processing depending on intersample level computation, but many elements of the audio signal chain may be affected by 0dBFS+ levels. E.g. filters, asynchronous and synchronous sample rate converters, data-compression and data-expansion circuitry etc.

## 2.1 D/A Conversion

Our scope was to investigate consumer reproduction equipment so most of our attention has been given to the digital to analog conversion process.

In a digital to analog reproduction chain there are several filters - and other limiting factors.

The most basic D/A converter is rarely used for audio in modern equipment, but may actually have an advantage when it comes to behavior at high levels. The signal is sampled out of the D/A chip at the basic sample rate, so no filtering takes place in the digital domain. An active analog output reconstruction filter is often run at +/-15 V supply which in most cases gives plenty of analog headroom. The fact that analog reconstruction filters typically have non-linear phase may or may not be a disadvantage.

Many early D/A stages use a low oversampling factor like 2, 4 or 8 combined with a digital reconstruction filter before the samples are output at this higher rate through a conventional D/A chip. In the analog domain a simple filter attenuates mirror signals around the oversampled Nyquist frequency.

A modern D/A converter stage typically consists of one chip using a very high oversampling rate and a built-in digital filter for reconstruction. Also with this type of converter a simple analog filter is removing unwanted images at the output.

To summarize, these are the sources of distortion and clipping in the D/A conversion process and the subsequent analog signal path:

1. Digital filter before the D/A converter
2. The D/A converter chip, especially the output stage
3. Analog gain stage after the converter, including AC coupling
4. Gain adjustment circuit
5. Analog output stage, possibly limited by supply voltage or current driving capability

When low pass filtering takes place in the digital domain special care must be taken to avoid overload due to critical input signals. Also the AC coupling in the analog domain may generate high peaks - up to 6 dB when fed by a square wave.

# 3. TEST AND PROCEDURE

Several different domestic and professional CD players have been submitted to a variety of test signals constructed to reveal difficulties regarding level handling, analog circuitry lock-up etc. Test signals include tones at different levels, program like material and artificial signals.

To check the headroom capacity of different consumer CD players, CDs were made with a collection of test signals that could reveal problems in digital filter or analog signal-path design.

Measuring equipment consisted of LeCroy 9350A, AP System 2 Cascade and Prism DScope.

## 3.1 Test Signals

Signals entering a digital signal processing system from the analog world should be limited in bandwidth and be free of aliasing. Metering of analog inputs stages is often based on the digital samples [3], [4], [5] although some DAT recorders use analog based meters. When digitally based meters are used the analog input gain can be set to avoid overload in the digital domain. Some obscure bandlimited signals may have a higher amplitude in the analog than in the digital domain. Most signals, however, will be well-behaved.

In the digital domain some processing of the signal will be done, and some of it may be non-linear. In mastering much of the processing can be non-linear: Expanding, compressing, limiting and clipping. But also linear processing like filtering takes place: Equalisation and DC removal.

It is therefore relevant to investigate the response to simple bandwidth-limited signals as well as complex ones which do not stick to Nyquist.

Three categories of test signals were used:

1. Sine waves
2. Square waves
3. A pseudorandom sequence

Sine waves of four frequencies were used: 997, 5512.5, 7350 and 11025 Hz. The first frequency has no simple relationship to the samplerate of 44100 Hz whereas the three others are fs/8, fs/6 and fs/4 respectively.

Sine waves with a simple relationship to the sample rate can be sampled in a way that the analog peak value is substantially larger than the digital peak value.

The square wave signals can be divided into two categories with specific purposes. Two low frequency square waves, 20 and 50 Hz, can be used to test the behavior of the AC coupling at the output. Although only rarely full scale square waves will be generated by a signal processing algorithm something similar will be the result if bass guitar or bass drum is clipped in order to maximize loudness. Even these low frequency signals with only low level harmonics above the Nyquist frequency will show some output filter ringing at the edges.

The remaining square wave signals of 551.25, 5512.5, 7350 and 11025 Hz are all of frequencies with simple relationships to the sample rate. This is primarily done in order to avoid problems like asymmetry and jitter of the digitally generated signals. These square waves have significant harmonic components above the Nyquist frequency, so some overshoot due to Gibb's phenomenon must be expected in the output filtering process. Like the low frequency square waves they are unlikely to occur in clean form in real material but clipping does generate signals with flat tops.

As an extreme signal a pseudorandom sequence has been chosen. It is a sequence repeating every 32767 samples, consisting of only +1 and -1 (or appropriately scaled values). The frequency spectrum is white.

The peak level in the analog domain is about 6 dB higher than in the digital domain so this signal will push filters and converters to their limits.

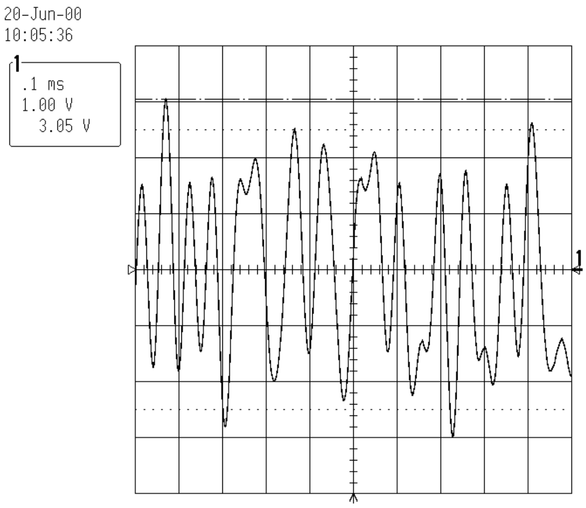



*Figure 5*

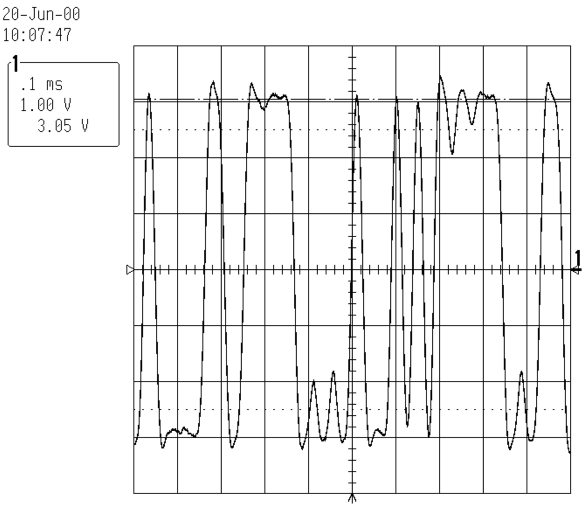



*Figure 6*

*Pseudorandom signals sampled @ 44.1 kHz.*
*Example shows consumer CD player, NAD C 520, measured on LeCroy 9350A digital oscilloscope.*

*Figure 5: -6 dBFS amplitude in the digital domain. Notice that peaks may be around 6 dB higher than the raw signal amplitude.*

*Figure 6: 0 dBFS amplitude in the digital domain. Notice that no peaks reach +6 dBFS as they theoretically should. They are clipped.*

All signals were 30 seconds in length with a half-cosine envelope at both start and end - suitable for making simple time domain inspection as well as spectral analysis.

## 3.1 Phase and Level

The synchronously sampled sine waves were generated with two start phases: One with the theoretical maximum value present as a sample value in the digital domain and one with the highest possible analog peak level within the limitation of +/-1 in the digital domain.

5512.5 Hz: 90 and 67.5°. At 67.5° the analog peak level is up to +0.69 dBFS.

7350 Hz: 90 and 60°. At 60° the analog peak level is up to +1.25 dBFS.

11025 Hz: 90 and 45°. At 45° the analog peak level is up to +3.0 dBFS.

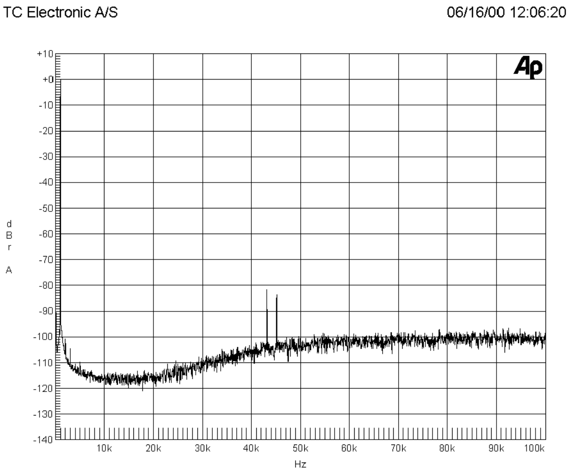
The CDs made for testing contained the signals at several levels including these which have the maximum analog peak level. As the present investigations are concerned about changed system behavior at high levels and not about general system performance a relative measurement may be sufficient. If advanced test equipment is available a spectral analysis will show detailed information. But a simple oscilloscope can also tell a lot.

The main test signal is recorded on one of the two stereo channels on a CD. The other channel contains the same signal but attenuated by 6 or 12 dB so that overload should not occur in that channel. By using a mixer or an oscilloscope with different gain in the two channels combined with a subtraction (phase reverse) feature the error signal can be heard or seen directly.

# 4. RESULTS

Sample rate synchronous sine waves were not chosen because of their program-like constitution but because they reveal level handling limitations using conventional distortion analyzers and enable easy comparison between units.

As a reference for the distortion measurements, asynchronous signals were used as shown below.



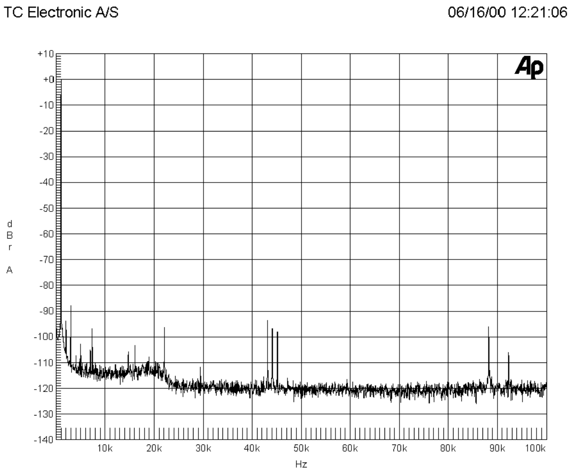




*Figure 7* *Figure 8*

*997 Hz @ 0dBFS sampled @ 44.1 kHz. 0 dB = Reference level for Sine Distortion Tests.*
*20 Hz to 100 kHz FFT on AP Cascade.*

*Figure 7: Yamaha CDX390 consumer CD player*
*Figure 8: Sony D50 consumer CD player (portable)*

## 4.1 Sine Test Results

A fuller picture can be obtained looking at FFT's as shown in Fig. 9-20, but a condensed comparison is shown in the table below.

Bandwidth of the THD+n measurements is 20 Hz - 80 kHz.

| | 997 Hz sine<br>Peak = 0.0 dBFS | 5512.5 Hz sine<br>Peak = +0.69 dBFS | 7350 Hz sine<br>Peak = +1.25 dBFS | 11025 Hz sine<br>Peak = +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 |

*Table 1. THD+n comparison*

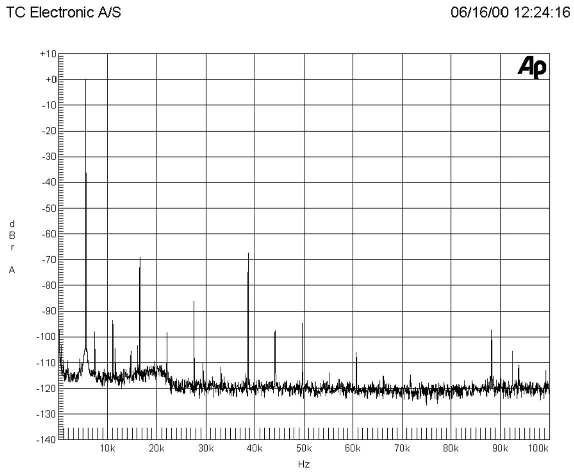



*Figure 9*

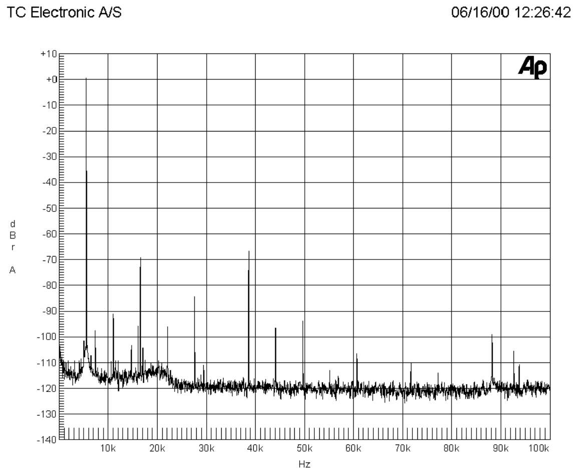



*Figure 10*

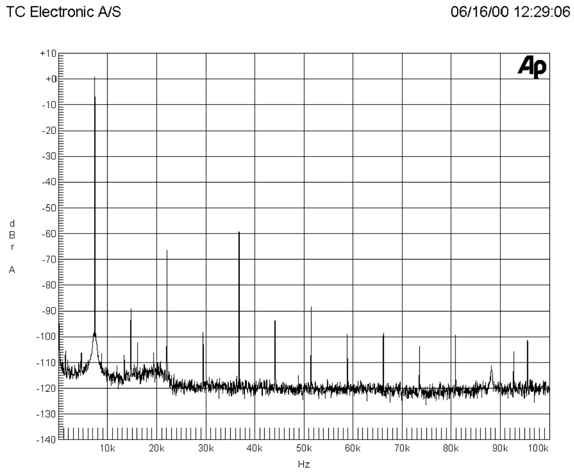



*Figure 11*

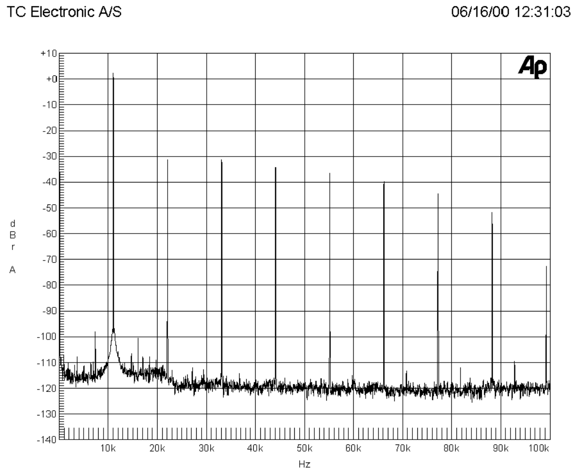



*Figure 12*

*Sony D50 consumer CD player (portable). Analog output performance.*
*Sampled @ 44.1 kHz. 0 dB = Reference level for asynchronous sine wave.*

*Figure 9: 5512.5 Hz Sine. Theoretical analog level at 0 dBFS*
*Figure 10: 5512.5 Hz Sine. Theoretical analog level at +0.69 dBFS*
*Figure 11: 7350 Hz Sine. Theoretical analog level at +1.25 dBFS*
*Figure 12: 11025 Hz Sine. Theoretical analog level at +3.0 dBFS*

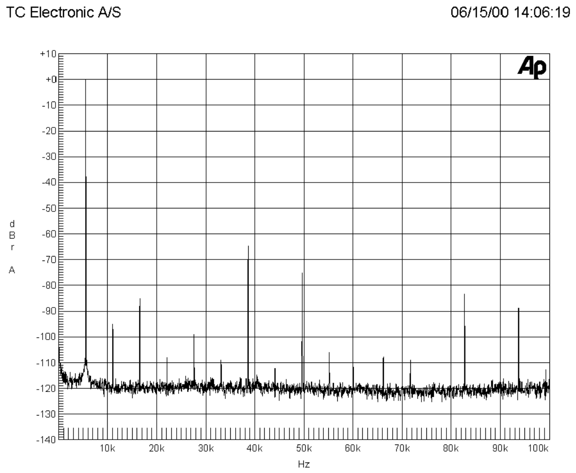



*Figure 13*

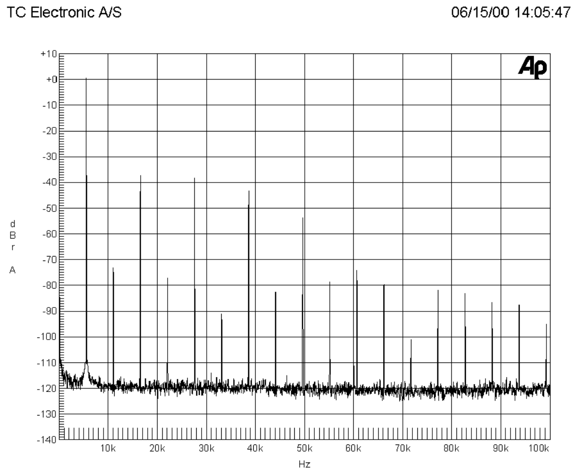



*Figure 14*

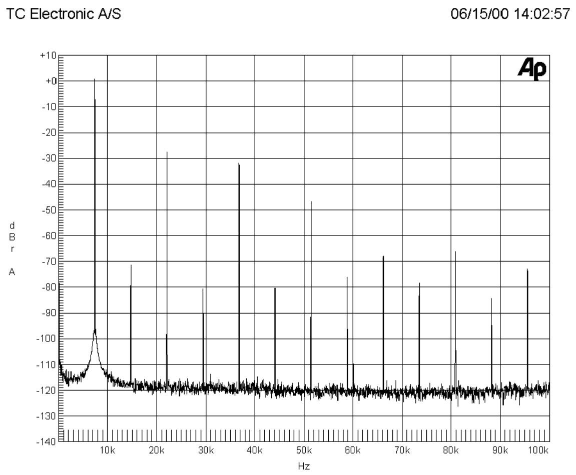



*Figure 15*

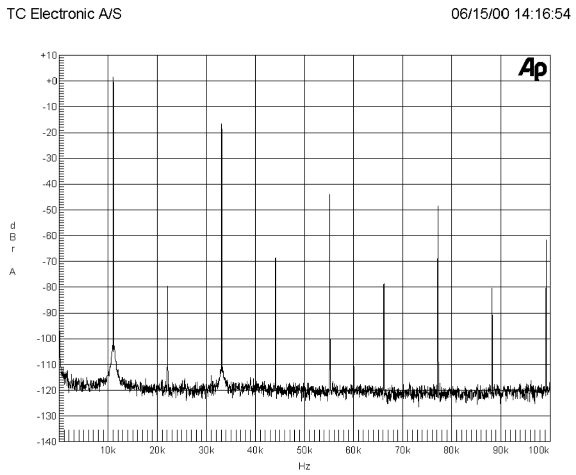



*Figure 16*

*Denon DCD 725 consumer CD player. Analog output performance.*
*Sampled @ 44.1 kHz. 0 dB = Reference level for asynchronous sine wave.*

*Figure 13: 5512.5 Hz Sine. Theoretical analog level at 0 dBFS*
*Figure 14: 5512.5 Hz Sine. Theoretical analog level at +0.69 dBFS*
*Figure 15: 7350 Hz Sine. Theoretical analog level at +1.25 dBFS*
*Figure 16: 11025 Hz Sine. Theoretical analog level at +3.0 dBFS*

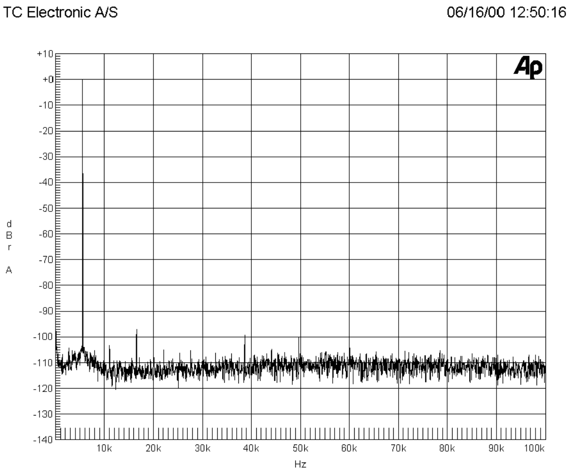



*Figure 17*

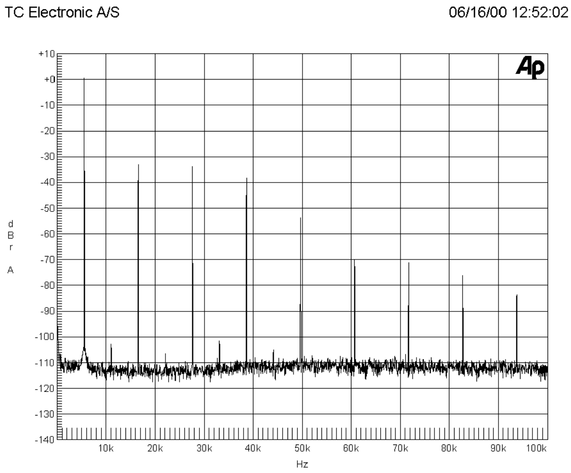



*Figure 18*

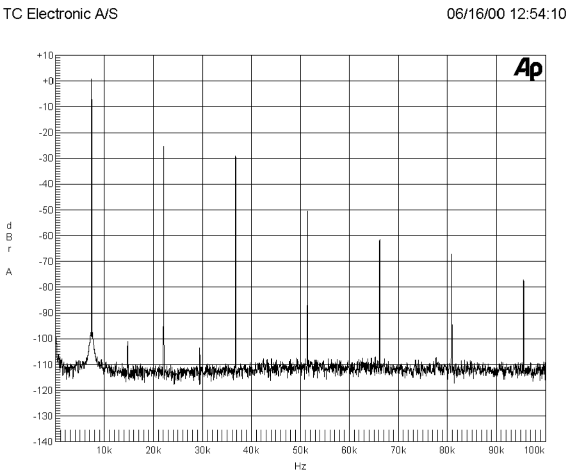



*Figure 19*

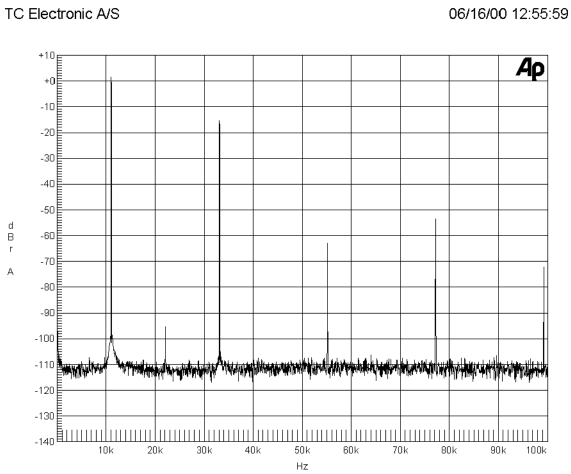



*Figure 20*

*Sony C11 professional CD player. Analog output performance.*
*Sampled @ 44.1 kHz. 0 dB = Reference level for asynchronous sine wave.*

*Figure 17: 5512.5 Hz Sine. Theoretical analog level at 0 dBFS*
*Figure 18: 5512.5 Hz Sine. Theoretical analog level at +0.69 dBFS*
*Figure 19: 7350 Hz Sine. Theoretical analog level at +1.25 dBFS*
*Figure 20: 11025 Hz Sine. Theoretical analog level at +3.0 dBFS*

# 5. LEVEL CONTROL IN MASTERING

Level measurement in CD production has historically been a matter of counting consecutive samples at digital full scale, 0 dBFS. Master tapes may be rejected if they contain too many consecutive samples at full scale.

Rules based only on the number of consecutive samples clipped are useless if we want to prevent 0dBFS+ levels from occurring because it is easy to subtract a few LSBs from the signal every time full scale is hit. LSB-cheating clearly does nothing to reduce neither level nor distortion.

## 5.1 Monitoring

Monitoring in most mastering studios is done using expensive stand-alone converters or mastering devices where distortion associated with 0dBFS+ levels may be less pronounced or not exist at all.

Under such circumstances the engineer will not stand a chance to find out if the consequences could be listening fatigue or even unmasked distortion at the end user.

## 5.2 Mastering against -3dBFS

Discussing deteriorating sound quality with mastering engineers, we discovered that some of them had started using a conservative level approach several years ago. Because of experiences from the real world, their ears had told them to generally keep peak levels below -3dBFS. Bad experiences with on-air signals or domestic reproduction equipment may be the reason why.

Before we get more precise level monitoring tools or new level guidelines for what is allowed before a master tape is rejected, this conservative approach certainly seems appropriate in order to avoid the kind of consumer equipment distortion we have described in this paper.

# 6. CONCLUSION

All of the domestic CD players investigated have shown difficulty dealing with 0dBFS+ levels that can easily occur on modern CDs. New models are actually worse than older types relying less on oversampling and more on analog filters.

We have not investigated how seriously audio quality is subjectively affected, nor have we made any listening fatigue tests concerning 0dBFS+ levels. However, modern CDs contain these kind of signals and modern CD players are not designed to reproduce them without distortion.

There appears to be plenty of reasons for concern about the quality of audio when hot mastering levels are to be reproduced at the end listener.

To make things worse, the mastering engineer is neither able to hear nor see when the level danger-zone is reached.

Regardless of whether Dynamic Range Approval guidelines are adopted by the industry or not, visual inspection tools aimed at 0dBFS+ detection should find their way into the quality-conscious mastering studios.

It is our belief that findings like this stress the need for a continued development of a DRA. In general it would appear that more focus should be given to how the upper end of the digital recording and reproduction dynamic range is utilized.

# REFERENCES

[1] Søren H. Nielsen & Thomas Lund (1999): Level Control in Digital Mastering, Presented at the 107th AES Convention, Preprint 5019.

[2] E. Zwicker & H. Fastl (1990): Psychoacoustics - Facts and Models, Springer-Verlag, Berlin.

[3] International Electrotechnical Commision (1995): IEC 268-18, Peak programme level meters - Digital audio peak level meter, First edition.

[4] DK Audio MSD600C oversampling meter (1999), http://www.dk-audio.dk.

[5] M. Ankerman et al. (1992): Aussteuerungsmesser mit Anzeige der Kurzzeit-Abtastwerte-Verteilung, ITG-Fachbericht 118, VDE-Verlag GmbH, p. 163-169.

[6] Internal report and Test CD, “Full Scale Reproduction Test no 1”, TC Electronic A/S.
Sine, Square and MLS signals at various levels.