True Peak Measurement

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Summary

As a supplement to the ITU recommendation for measurement of loudness and true-peak level [1], [2], a set of test signals for testing the conformance of the true peak measurements to the recommendations has been proposed [3].

This paper describes an extension of that set of test signals with some which exercise the very high levels normally causing overload when signals are converted. Furthermore, arguments for leaving out the optional DC-filter and pre-emphasis are given.

Introduction

Widespread use of the sample peak measure, in combination with a desire to produce loud music CDs and commercials, has allowed distortion to systematically develop in professional audio. For more than a decade, audible overload has been generated in digital to analog converters, sample rate converters and in lossy data reduction systems. Before the true-peak measure, such hostile signals could not be recognized on meters used in audio production [4].

In a series of AES papers starting 1999, consequences of basing audio normalization and peak level detection on the dysfunctional sample peak measure has been investigated and reported [4], [5], [6], [7].

With the advent of the "true-peak" measure, defined by ITU with the aid of AES SC-02-01, a more appropriate technique is now available; and true-peak metering is also part of the ITU-R BS.1770-2 standard [1]. This document describes new, relevant test signals for checking meter compliance with the true-peak standard.

Overall remarks to standards and recommendations

There is no need to reinvent the wheel, so instead of defining new methods and performance requirements, existing recommendations and practices should be used whenever appropriate. One beauty of standards is that there are so many of them...

In the spirit of the IEC international standards and reports on audio level meters [8] analogue PPM) and [9] (digital PPM) it is proposed in the present context that the individual characteristics of a true peak level measurement method and meter are described by:

- Characteristic to be specified
- Method of measurement
- Performance requirement

In general, the IEC documents on level meters do not prescribe a specific method of implementation. Instead they use a set of requirements which must fulfilled in order for a meter to bear the mark of the relevant standard. The standards specify that the accuracy etc. must be specified quantitatively in the product documentation. This ensures that meters of different make and model can be compared on equal terms.

For each characteristic / parameter, the actual performance should be indicated in the documentation for the meter (and in a few cases directly on the meter itself). In order a meter to qualify as a type IEC xxx meter, the performance of each characteristic should be at least as good as required by the corresponding clauses in the standard. So it is not sufficient just to state that a meter fulfills IEC xxx - the actual meter performance should also be stated in the documentation.

Compared to the accuracy possible in the digital age, many of the performance requirements in the IEC meter documents may seem relaxed. On the other hand, an accuracy of the amplitude-frequency response of e.g. +/-0.5 dB in the range from 20 Hz to 20 kHz is well in line with the operational needs for accuracy. And even in digital implementations there are necessary trade-offs to make between accuracy and cost.

Elements to be Included in a True Peak Meter

The true-peak meter of [1] is not based on a specific algorithm, but on a block diagram in combination with a set of minimum requirements.

This implementation-agnostic approach helps the technique spread because the measurement may be tailored to various platforms without imposing unnecessary costs. However, in previous versions of BS. 1770, the true-peak measurement includes two potentially confusing options: DC block (high pass filter) and pre-emphasis.

Where the up-sampled measure is necessary to predict overload in digital-to-analogue converters and in sample rate converters, extra conservatism in the fundamental measurement would need to be justified carefully. Every dB of headroom is important for audio quality, so false positive warnings are as bad as false negative. A high-pass filter may cause the peak level of clipped audio to rise, but there is no reason to expect such a filter in a linear PCM signal-path downstream of the meter. Furthermore, DC blocking prevents the operator from recognizing a potential DC problem in the signal. Similarly, there is no reason to expect worst case conditions or errors when considering ripple and frequency response of down-stream devices. Therefore, meter pre-emphasis makes the measurement less accurate, not more.

Our aim should be to satisfy linear audio and an ideal sample rate converter. A compromised signal-path may be dealt with by setting the true-peak threshold lower, not by adding belts and braces to the fundamental measure.

In order to keep true-peak measurements accurate and repeatable, and to avoid unnecessary or even counterproductive complications, we consequently suggest to discard of the DC block and of the pre-emphasis option in BS.1770. It is therefore also not necessary to specify true-peak meter test signals to check for such properties.

Test Signals

In order to unambiguously define DUT meter readings, test signals should comply with Nyquist criteria as discussed in [3], [4], [6], [7]. Sample rate synchronous ("SRS") sine waves were described and used for the testing of true peak performance in [5] and [6]; and they are also the main static signal test probe suggested in [3].

One motivation for measuring true peak instead of sample peak is to enable use of calibration signals with frequencies close or equal to an integer fraction of the sample rate together with a peak meter. Due to the nature of sampling, the true peak value may differ significantly from the sample peak value. This is a well-known phenomenon, and is also recognised in [9].

However, the main motivation for measuring true peak instead of sample peak is the risk of clipping due to interpolation filters, either in the digital (e.g. sample rate converters) or analogue domain (digital-to-analogue converters with their reconstruction filters). The risk of clipping is not just theoretical, as shown e.g. in [4]. Furthermore, it has been demonstrated [7] that also certain perceptual audio coding systems have difficulties handling legal full-scale signals.

For these reasons we propose that the collection of test signals proposed by [3] is extended by some which exercise the potential overload region. Such signals are essential to test the very purpose of a true peak meter.

For an illustration of SRS sine waves similar to Fig 1 in [3], see Fig 2 in [7]. The signals are described in the following subsection.

Test Signals at Critical Frequencies

An important property to test in a true peak meter is how accurately the peak value is measured, particularly in two cases:
1) The sample peak is coinciding with the true peak.
2) The difference between true peak and sample peak is maximum.

When considering continuous sine waves, a few frequencies are of particular interest:
1) fs/4, at two different phase shifts - a max. deviation between true and sample peak of 3.01dB.
2) fs/6, at two different phase shifts - a max. deviation between true and sample peak of 1.25 dB.
3) fs/8, at two different phase shifts - a max. deviation between true and sample peak of 0.69 dB.
4) 1 kHz - an often used calibration frequency.

The odd integer ratios are not as suitable for testing overload behaviour as the even ratios, due to the fact that maximum difference between sample peak and true peak is smaller for the odd ratios. The odd-ratio frequencies may be useful for testing symmetry behaviour, however.

There are only a few sample rates in use in pro audio, so the number of critical frequencies with respect to true peak values is limited, as listed in the table below.

frequency \ fs	**32 KHz**	**44.1 kHz**	**48 kHz**	**88.1 kHz**	**96 kHz**
fs/4	8 kHz	11.025 kHz	12 kHz	22.050 kHz	24 kHz
fs/6	5.333.. kHz	7.35 kHz	8 kHz	14.7 kHz	16 kHz
fs/8	4 kHz	5512.5 Hz	6 kHz	11.025 kHz	12 kHz
1 kHz	1 kHz	1 kHz	1 kHz	1 kHz	1 kHz

Table 1. Sample rates used in pro audio and integer ratio test tone frequencies.

The levels to exercise the meter at full scale and beyond are just two per frequency: One where the sample peak is at full scale and one where the true peak is at full scale.

Due to the relative many samples used to represent a 1 kHz signal at typical pro-audio sample rates, the maximum difference between true and sample peak is small: At fs = 32 kHz the max. difference is 0.04 dB, at fs = 48 kHz the max. difference is 0.02 dB. Therefore we consider it safe to leave out the 1 kHz signal in the true peak test suite, except as a calibration signal, where is should be sampled with a cosine phase of zero degrees (starting at sample maximum).

The table below shows our proposal for test frequencies and three sets of cosine starting phases and corresponding true peak levels.

Frequency	starting phase of cosine [degrees]	true (and sample) peak level [dBFS]	starting phase of cosine [degrees]	true peak level [dBFS]	starting phase of cosine [degrees]	true peak level [dBFS] (max. sample peak level 0.00 dBFS)
fs/4	0	0.00	45	0.00	45	+3.01
fs/6	0	0.00	60	0.00	60	+1.25
fs/8	0	0.00	67.5	0.00	67.5	+0.69

Table 2. Three sets of integer ratio test tones.

A test signal suite could be composed of a sequence of these frequencies in triplet sequences, e.g. with a duration of 5 seconds per signal (sufficiently long for easy reading) with a pause of a 2 seconds in order for the peak meter to return to a low value. Between each frequency triplet the pause could be a little longer, e.g. 5 seconds.

There may be a practical issue if the operator does not know the sample rate. This is hopefully a hypothetically one, and in broadcast one would assume sampling at 48 kHz, but an 8 kHz signal at different starting phases may actually be used to determine the sample rate if desired.

SRS sine waves may also be used to assess the *amount* of headroom available in a meter, in a processor or of an entire signal-path. Despite testing for more than a decade [6], [7], we haven't found consumer or professional linear audio equipment that clipped for level lower than 0 dBTP. However, depending on the design of, for instance, a DA converter, it runs out of headroom at an unforeseeable level above this point. A wider selection of sample rate synchronous sine waves could therefore include more level steps from -0.1 to +3 dBTP, for instance at 0.1 dB intervals.

Conclusion

It is well indicated to specify a suite of test signals for the verification and accuracy of true-peak meter conformance.

In this document, paramount true-peak meter test signals to complement the probes suggested in [3] have been described. Also, it is justified how static as well as dynamic test signals have to satisfy the Nyquist criterion.

It is further proposed to dispose of the DC block (high pass filter) and of the pre-emphasis option in ITU-R BS.1770-2. Consequently, neither function needs testing.

The main purpose of the true-peak meter is to reveal likely overload of a downstream, linear audio signal-path including sample rate and DA converters. The DUT meter therefore must be tested for true-peak calculations being precise, and for its internal headroom being sufficient; the latter lacking in [3].

References

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