Software Audio Test Suite User's Guide

Release: GitHub

Version: 1.0

Table of Contents

Introduction

The purpose of this document is to provide information concerning the use of the Dolby® software audio test suite (SATS), *GitHub Release* version 1.0. The SATS consists of a collection of software audio test tools for testing digital audio products or devices. The audio test tools are designed to be run from a PC command line (Command Prompt) or shell (such as Bash), and operate on *.wav* files. Alternatively, the audio test tools can be incorporated into scripts for automation purposes. A test framework with a collection of test signals and reference results are provided for checking limits so that simple pass/fail decisions can be generated. The common test framework is designed to test every tool provided with the package generated using either Linux or Windows platform. SATS also consists of a KISS_FFT library package for FFT calculations. SATS supports INTEL® MKL FFT (for Linux and Windows MSVS platforms) but this package must be downloaded separately and organized as per the details mentioned in README document.

We recommend that the SATS be used only to test implementations or software-based encoder or decoder products. Products that provide hardware digital audio inputs and outputs must capture the output to a compatible *.wav* file before testing with the SATS.

Refer README document for system requirements and installation.

Usage

2.1 command line

All the tools have the same command line options:

```
Usage: tool [OPTION]... -i WAVFILE
-c <chan>,        selects channel in multichannel file
                  -c 0 for first channel
                  -c a for all channels
-s,               turn off stripping lead silence
-t,               send text output to standard output (default)
-to <name>,       create a text file
-xmin <lim>       selects minimum x-axis limit
-xmax <lim>       selects maximum x-axis limit
-powermin <lim>   selects dB level to which very low power values will
                  be clipped.
-thr_db <thr>     selects db level to which power values will be set
                  as a power threshold for silence stripping
-thr_s <thr>      selects bit depth level to which sample values will
                  be set as a sample threshold for silence stripping
```

Detailed description about these and more options which are specific to a tool are addressed in general in Section 2.2 and under individual tools in Chapter 3.

The following are examples of the SATS usage at the command line:

Spectrum_avg.exe -c 2 -thr_db -100 -i inputfile.wav

- Use the spectrum averaging tool
- Analyzes the third channel in the multichannel input file (inputfile.wav)
- Set the power threshold as -100db to strip the signal head part with the power which is smaller than -100db

Thd_vs_level.exe -to result -c 0 -i inputfile.wav

- Use the total harmonic distortion plus noise (THD+N) vs level tool
- Analyzes the first channel in the multichannel input file (inputfile.wav)
- Save the result in .txt format to result.txt

Test Signals (Input File)

The test signal files to be used with these tools are provided with the SATS package. The tools that require frequency or amplitude sweeps are derived from files provided in SATS package for this purpose. Failing to use these files will likely cause unexpected results. All the tools strip any leading silence present on the input signal. The leading silence is assumed, due to encoding and decoding delay. `-s` option can be used to turn off silence stripping.

Test Tool

The audio test tool is used as the executable application file of the command line test (see Chapter 3).

2.2 Options

The available general SATS command options are detailed in this section. **Some tool specific options are described in Chapter 3 under the respective tools.** Use `-h` option to see all the options available for a specific tool.

Input Channel (`-c`) Option

The `-c` option indicates which channel the tools should use as input. The default channel is the first, selected using `-c 0`. The last channel is selected by using the number of channels minus one. The `-c a` option can be used when calculating the output of the tool for all channels simultaneously.

Textual Output (`-t`) Option

The `-t` option (default) requests textual output. The output is provided as two columns of numerical data separated by a comma and tab character. The left column corresponds to the horizontal or x-axis, whereas the right column corresponds to the vertical or y-axis. The results are directed to the standard output upon completion of the measurement. The output of the tool can be saved to a file by using the `-to` option with the name of a file.

Turn off silence stripping (`-s`) Option

All the tools strip any leading silence present on the input signal. The leading silence is assumed, due to encoding and decoding delay. `-s` option can be used to turn off silence stripping.

Threshold of Silence Stripping (`-thr_db`, `-thr_s`) Option

The `-thr_db` and `-thr_s` options realize a user adjustable choice for silence stripping. If `-thr_db` is chosen, a number given by the user will processed as a power threshold. The beginning of a signal will be stripped, if it's power is below the given threshold. If `–thr_s` is chosen, an integer number ranging from 0 to 31 will be given by the user, which represents the number of least significant bits (LSB). A corresponding sample threshold will be calculated and every sample below the given LSB value will be removed from the beginning of an input file. It is noted that `-thr_db` and `-thr_s` cannot be chosen at the same time. If none of them is chosen, a default setting of `-thr_s` 0 is used.

Minimum Limit for Power Values (`-powermin`) Option

The `-powermin` option will select dB level to which very low power values will be clipped during measurement.

Test Tools

A collection of specific-function test tools is available for use with the SATS. The audio test tool is used as the executable application file of the command-line test. The build files for these executables are provided with SATS package. Refer README document for build instructions and locations.

3.1 Frequency Response (freq_resp)

The frequency response tool (freq_resp.exe) requires that the input file contains a frequency sweep. The sweep must be stepped, that is, the sweep must contain only one frequency for a period of time. This time is called a dwell. All dwells ≤ 200 Hz must be at least two seconds in length, whereas all dwells > 200 Hz must be at least one second in length. Dwells must contain frequencies between 20 Hz and 20 kHz. The peak amplitude during each dwell is measured. Dwells should be spaced a minimum of ten percent apart. For example, for a low-to-high sweep after a 20 Hz dwell, the next dwell must be at least 22 Hz. SATS package provides these frequency sweep test files ready for use.

3.2 Dynamic Range (dyn_rng)

The dynamic range tool (dyn_rng.exe) requires a 200 Hz input tone. Normally, this tool has an amplitude of –60 dBFS. The 200 Hz tone is removed using a notch filter. The residual noise is filtered using a CCIR-RMS 2K perceptual weighting filter and a 20 kHz lowpass filter. The RMS level of this filtered signal is calculated against time every 100 ms. The RMS metering is calibrated such that a full-scale sine wave provides a value of 0 dBFS.

3.3 THD+N vs Frequency (thd_vs_freq)

The THD+N vs frequency tool (thd_vs_freq.exe) uses the same stepped frequency sweeps as the frequency response tool. The same dwell length and frequency constraints apply. This test removes the fundamental frequency of the tone and measures the RMS level of the residual noise after applying a 20 kHz lowpass filter. The residual noise level is calculated against the frequency of the incoming tone. The RMS metering is calibrated such that a full-scale sine wave provides a value of 0 dBFS. Dwells should be spaced a minimum of ten percent apart. For example, for a low-to-high sweep after a 20 Hz dwell, the next dwell must be at least 22 Hz.

3.4 THD+N vs Amplitude (thd_vs_level)

The THD+N vs amplitude tool (thd_vs_level.exe) requires a 4 kHz amplitude sweep as input. This sweep does not need to be stepped. The fundamental of the 4 kHz tone is removed, and the RMS level of the residual noise is measured after the application of a 20 kHz lowpass filter. The residual noise level is calculated against the absolute peak amplitude of the 4 kHz tone. The RMS metering is calibrated such that a full-scale sine wave provides a value of 0 dBFS.

3.5 Power vs Time (pwr_vs_time)

The name of the power vs time tool (pwr_vs_time.exe) is somewhat confusing, because the tool actually plots amplitude vs time. Measurements are averaged in an adjustable interval (100ms default) for the duration of the file. These measurements are calculated against time. This tool is useful for testing dynamic range control and dialogue normalization level compensation. The metering is calibrated such that a full-scale sine wave provides a value of 0 dBFS. Following additional command line options for usage with this tool are available:

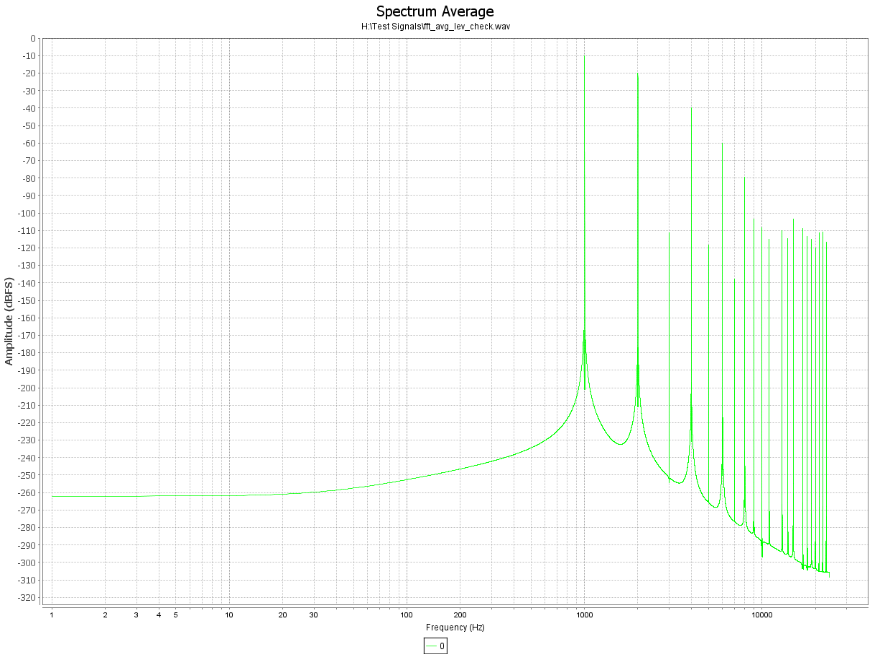
Block Size (`-blksz_t, -blksz_s`) Option

The `-blksz_t` and `-blksz_s` options are used to pre-define the size of block in the input sample for which the results will be calculated. The block size can be specified as time (in ms) using `-blksz_t` or as number of samples using `-blksz_s`. When the block size value is not specified, a default value setting of `-blksz_t` 100 is used. Note: blksz_t and blksz_s cannot be set simultaneously.

3.6 Spectrum Averaging (spectrum_avg)

The spectrum averaging tool (spectrum_avg.exe) is normally used with a frequency domain stationary signal. Periodograms are computed twice per second and averaged over the entire file to produce a very detailed spectral representation. This routine uses Fast Fourier Tranforms (FFTs) and a fourth-term Blackman-Harris analysis window. Points are plotted at 1 Hz intervals up to half the sampling frequency.

An example plot of the result file obtained using this tool is shown below.



3.7 Noise Modulation (noise_mod)

The noise_mod.exe tool requires a 41 Hz amplitude sweep as input. This sweep does not need to be stepped. The noise level at 4 kHz is measured using a narrow bandpass filter. The noise level is calculated against the absolute peak amplitude of the 41 Hz tone. The RMS metering is calibrated such that a full-scale sine wave provides a value of 0 dBFS.

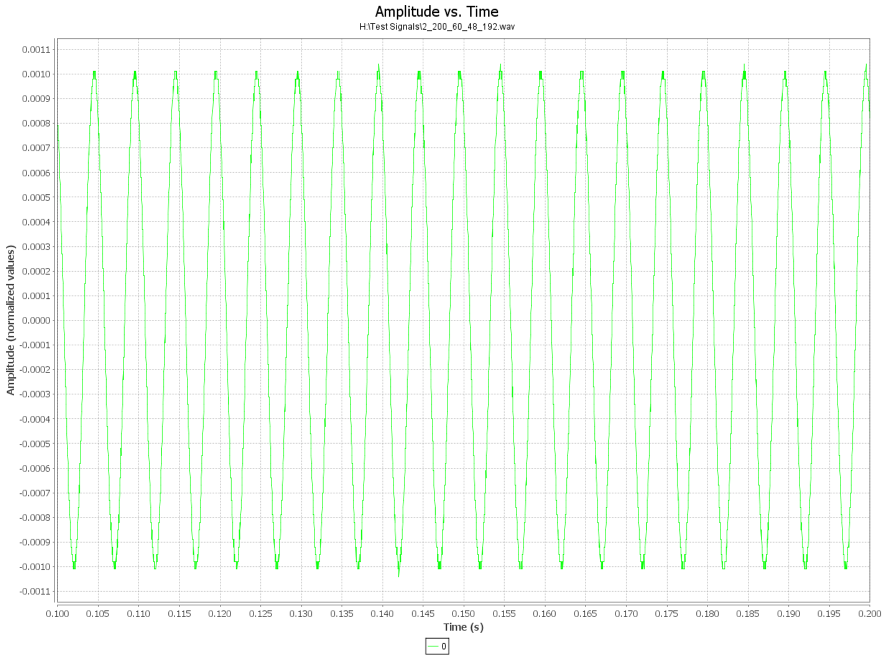
3.8 Amplitude vs Time (amp_vs_time)

The amplitude vs time tool (amp_vs_time.exe) is similar to the pwr_vs_time.exe tool in section 3.5, except that no averaging is performed. The individual samples are displayed using a linear scale up to a range of three seconds. The amp_vs_time.exe tool allows the user to inspect the waveform contained within the .wav file without the use of a digital audio editing tool. Each point is at least an individual sample. Following additional command line options for usage with this tool are available:

Output Limits (`-xmin and -xmax`) Options

The `-xmin and -xmax` options specify the limits for the output.

An example plot of the result file obtained using this tool is shown below.



3.9 Spectrum of NFFT Points (spectrum_NFFT.exe)

The spectrum of NFFT points (spectrum_NFFT.exe) tool is similar to the spectrum_avg.exe tool, except that it provides additional command options for the user to specify the number of points analyzed during the periodogram measurement, the number of averages to calculate and the choice of window. The following lists the spectrum_NFFT.exe command options:

```
Usage: spectrum_NFFT.exe [OPTION]... FILE

-t,                 Send text output to standard output (default)
-to <name>          Create a text file
-c <chan>,          Selects channel in multichannel file
                    0 = first channel
-window <type>      Selects the type of window: Bartlet, Bartlet-
                    Hann, Blackman-Harris (default), Rectangular,
                    Triangular or Hann
-n <NFFT>           Selects the FFT size of spectrum: 512, 1,024,
                    2,048, 4,096, 8,192 10,240,16,384, 32,000,
                    44,100 or 48,000
-z <NAVGs>          Selects the number of spectrum averages to
                    compute (if not provided, as many averages
                    based on file length will be computed)
```

If the number of FFT points option (`-n`) is not specified, the sample rate of the input signal determines the size of the FFT measured for the periodogram. The number of averages option (`-z`) specifies how many periodogram averages to compute. If the `-z` option is not specified, the tool will compute as many averages within its limits based on the input file length. The type of window for periodogram can be chosen using the `-window` option. If the `-window` option is not specified, the tool will use Blackman-Harris as the default analysis window.

Note: If the `-n` option is specified, the frequency resolution of the measured points will no longer be 1 Hz, but instead depends on the number of FFT points specified and the sample rate of the input signal.

3.10 Multiple Tone Frequency Response (mult_freq_resp.exe)

The multiple tone frequency response (mult_freq_resp.exe) tool measures the spectral density at specified frequencies. By using a multitone input signal (a signal containing multiple tones that occur simultaneously), the tool measures the power of each tone in the input signal and linearly interpolates a frequency response.

The SATS package provides multitone test signals ready for use. Other multitone signals, however, may also be used. For best results, use as many equal-level tones in the signal as possible. Ensure that the total power of all the tones does not exceed full scale, or the distortion can lead to inaccurate results. To avoid unnecessary cancellation of the tones, do not use harmonically related tones or random phase settings. The multitone signal must be at least one second long.

The following lists the mult_freq_resp.exe command options:

```
Usage: mult_freq_resp.exe [OPTION]... FILE

-t,           Send text output to standard output (default)
-to <name>    Create a text file
-c <chan>,    Selects channel in multichannel file
              0 = first channel
-f <name>     Selects file containing a list of multitone frequencies
              (must be provided)
```

A file containing the frequency of each tone in the input signal must be provided using the `-f` option. The file must specify each frequency on a new line.