Virtual Instruments for Audio Testing

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(Invited)

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With today's personal computer technology and software development tools, the capabilities of dedicated instrumentation can be recreated in a virtual environment. The test system accuracy is limited only by the quality of the data acquisition board or sound card. All of the signal generation, filtering, analysis and data handling can be performed in software such that the PC becomes the test platform. The user-interface can even be made to emulate familiar analog instrumentation using standard Windows controls.

1 INTRODUCTION

Recent improvements in graphical programming languages, data acquisition boards and sound cards now make it possible to use a PC as the measurement platform for audio testing. Selection of the appropriate hardware to be used as the "front-end" to the test software is also important and will be discussed. An actual test system developed using a graphical programming language, with a PC and a sound card is shown.

DSP-based signal generation, filtering, and analysis software has many advantages over traditional analog instrumentation. As an example of this, a test algorithm will be demonstrated that can simultaneously measure all harmonics, yet only a single sine sweep is necessary.

It will be shown that properly implemented "virtual instruments" can be a cost-effective, flexible, portable, and powerful solution for collecting accurate audio test data.

2 VIRTUAL INSTRUMENTS

Many companies still use old swept sine measurement systems consisting of separate standalone hardware linked together by BNC cables (often referred to as a "rack-and-stack" system). In a traditional hardware-based test system (see Figure 1), a sine generator performs a continuous sweep through the frequency range of interest. As the sine generator sweeps, a band pass filter "tracks" the frequency of the sine generator to suppress noise and harmonics that occur at other frequencies. The voltmeter or "measuring amplifier" then detects the RMS level, and outputs a proportional DC voltage to a synchronized chart/level recorder or an oscilloscope. The level on the chart recorder corresponds to the sound pressure level measured at each frequency.

The corresponding frequency resolution is limited by the sweep rate, filter bandwidth and the level recorder writing and paper speed. If the sweep rate is too fast, frequency and amplitude "smearing" will occur. This is particularly noticeable if the transducer's response exhibits any sharp peaks or dips, which is usually the case.

A "virtual" test system was developed that operates on the same principles as a traditional swept sine measurement system consisting of a sine generator, voltmeter, tracking filter, and level recorder. The main difference is that all these functions are implemented in software as virtual instruments, "VIs" (see Figure 2).

This virtual system uses a similar but improved approach, sweeping in discrete steps. Each frequency step can contain numerous cycles that are synchronously averaged to minimize the effect of background noise (see Figure 3). Increased averaging has the effect of narrowing the filter bandwidth for better noise rejection.[i] Transducer settling (transient ringing) is also minimized by discarding the first few cycles at each new frequency step and providing a phase continuous transition between frequencies. This provides higher frequency, amplitude, and phase accuracy, as well as excellent noise suppression.

3 HARDWARE

When performing digital signal analysis, the test system accuracy is ultimately limited by the quality of the measured analog signal that is converted to the digital domain for analysis. Therefore it is important to choose quality transducers, signal conditioning, and A/D & D/A converters for the front-end of the test system.

3.1 Transducers & Signal Conditioning

In acoustics, the finest available measurement microphone ensures that the signal is not corrupted. It is important to make it easy for the user to interface the test system to these transducers by providing a compatible microphone power supply with adjustable gain to maximize the signal to noise ratio of the measured signal. By keeping the signal conditioning external to the PC, the measurement system dynamic range is maximized.

3.2 DAQs and Sound Cards

With the virtually unlimited memory capability of modern PCs, it is no longer necessary to combine the data gathering and analysis into the same step as is done with traditional hardware-based test systems (e.g., a tracking filter and level recorder). The entire sine sweep is played and the response of the device under test is synchronously recorded. The frequency analysis and limit comparisons can then be performed as post-processing steps, taking advantage of existing DSP software analysis tools. The data can also be analyzed and reanalyzed as desired. The analysis can be in as much detail as is appropriate for the application.

This processing requires a good D/A converter for transforming the digital representation of the sine sweep created in the Stimulus.vi into an analog signal and a good A/D converter for transforming the measured analog signal back into the digital domain for analysis in the Analysis.vi. In the audio range of frequencies, good A/D & D/A converters are easy to find, in fact, most sound cards provide at least 16-bit resolution. Some sound cards can even measure frequency response from DC – 20 kHz ± 0.25dB and distortion down to 0.003%.[ii]

Original development was done on a National Instrument's AT-DSP2200 data acquisition board. This data acquisition board has an on-board DSP processor. The computational speed of the PC's CPU, however, was more than adequate for this application and consequently it was not necessary to utilize the on-board DSP. In order to reduce the system price, a sound card driver was developed using standard Windows multimedia calls.

Another key requirement was that the system could be calibrated and traced to the National Institute of Standards and Technology (N.I.S.T.). Therefore, a comprehensive set of calibration routines was created including a microphone calibration that uses N.I.S.T. traceable acoustic calibrators (see Figure 4).

4 ANALYSIS

A common technique for distortion testing is to attenuate the fundamental with a notch filter (see Figure 5). This measurement technique, however, cannot distinguish the amplitude of individual harmonics (e.g., 2nd, 3rd, and 4th harmonics) or distinguish distortion from noise. Moreover, these measurements are typically performed in the presence of high background noise. In this situation, it is impossible to discern distortion from noise.[iii]

Alternatively, a tracking filter can be used to measure an individual harmonic. This technique, however, is slow and must be repeated if additional harmonics are desired. Instead of using a tracking filter that can only measure one harmonic at a time, a special FFT-based algorithm was developed to measure all the harmonics in a single sine sweep (see Figure 6). This is equivalent to a parallel bank of individual tracking filters that measure all selected harmonics simultaneously. This parallel analysis technique saves considerable measurement time over the traditional serial analysis method. The ability to separate out individual harmonics is ideal for detecting and analyzing different kinds of transducer faults, such as rubbing voice coils, loose particles in the gap, and buzzing or rattling.

It was decided to minimize the number of necessary setup parameters in order to make this routine easy to use. By default, the appropriate measurement algorithm and setup is chosen depending upon the number of harmonics selected (see Figure 7).

5 POST-PROCESSING

Once an accurate measurement is performed, one possible post-processing step is to test the DUT response against limits according to user's specifications. This was achieved by providing very flexible, user-definable, tolerance limits (see Figure 8). The curves are defined by either specifying the frequency "knee points," using actual transducer measurement data, or selecting from a list of standardized curves. Depending on the test requirements, the tolerance curves can be Absolute, Floating or Aligned. In Absolute mode, the measurement curve is tested against "fixed" tolerances. In Floating mode, the transducer curve "floats" for a best fit between the tolerance limits. For Aligned mode, the measurement curve is "anchored" to a single reference point.

After the measurement is completed, the limit curves' resolution is matched to the measurement's resolution (using a straight-line interpolation if necessary) and the result and tolerance margins are displayed. The tolerance margin is the degree to which the device passes or fails, a useful statistical indicator of production quality (see Figure 9).

6 DATA HANDLING & MANAGEMENT

No two measurement applications are exactly alike. Each user requires a certain amount of customization. In order to aid in optimizing tests to specific customers needs, a test sequencer was designed that would allow for a variety of tests. Different steps and sequences can be stored and modified. With the Sequence Editor, custom test procedures are easy to setup and automate. The big advantage of this approach is being able to setup a variety of tests and add new modules and features in the future.

Very often it is desirable to decide a test action based on information from a previous test or step. This requires information about whether a previous test passed or failed and the curve data from the previous step(s). Each step can be configured for display options and conditional branching. In addition, the overall sequence can be configured for display options and conditional execution for trouble shooting (e.g., single stepping) as well as memory management. The sequence editor treats each step as an individual test with its own data and results. This allows the user to configure each step depending on the data and results from its own step and previous steps. The flow diagram in Figure 10 shows how each step can be configured individually and the overall sequence configured globally.

7 HUMAN INTERFACE

The user-interface can be made to emulate familiar analog instrumentation using standard Windows controls. This provides a consistent and user-friendly interface that does not change from instrument to instrument, making it much easier for the average instrumentation user to quickly start performing measurements.

The Sine Generator.vi, for example, was designed to look, feel, and sound like a real analog sine generator. When changing frequency or level, the sine generator.vi actually sweeps to the new frequency or level for a smooth transition without any dropouts. In addition, the user can control the sweep rate to their personal taste to create a certain amount of hysteresis, similar to the feel of a real knob on an analog sine generator.

8 CONCLUSION

Graphical programming languages provide powerful tools for designing, testing, and creating virtual instruments. Because VI's are software-based, they never become obsolete. As the power and performance of computers improve, so does the test system. Virtual Instruments make programming and using the hardware much easier. A minimum of external test equipment is required.

Properly implemented "virtual instruments" can be a cost-effective, flexible, portable, and powerful solution for collecting accurate audio test data.

9 REFERENCES

[i] C.J. Struck, "An Adaptive Scan Algorithm for Fast Response Measurements", presented at the AES 91st Convention – New York, (1991 October 4 – 8)

[ii] Digital Audio Labs CardD+ sound card specifications

[iii] S.F. Temme, "Why and How to Measure Distortion in Electroacoustic Transducers", presented at the AES 11th International Conference on Audio Test and Measurement, Portland, Oregon (1992 May 29 – 31)

Figure 1 Traditional Hardware-based Test System

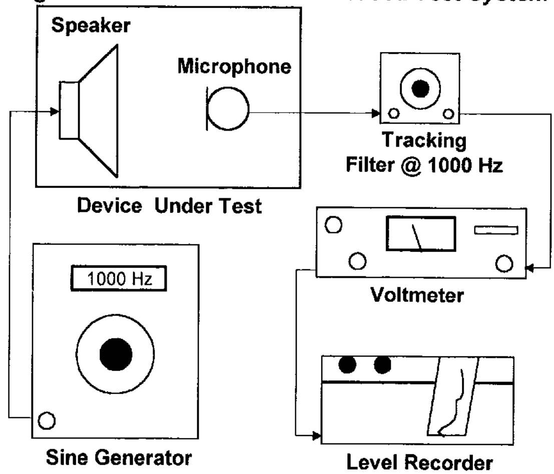
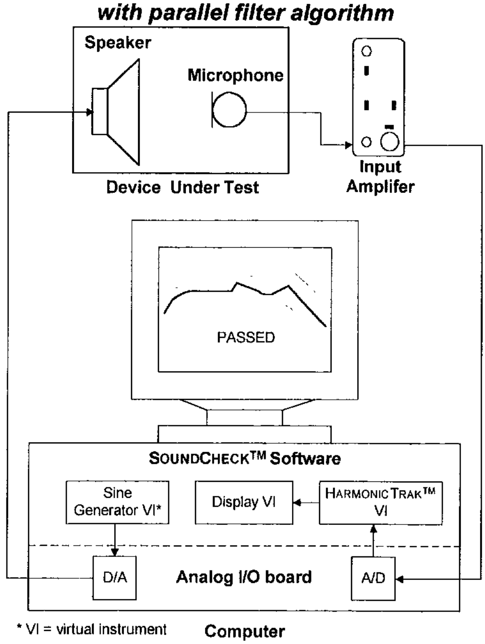


Figure 2 Software-based Test System with parallel filter algorithm



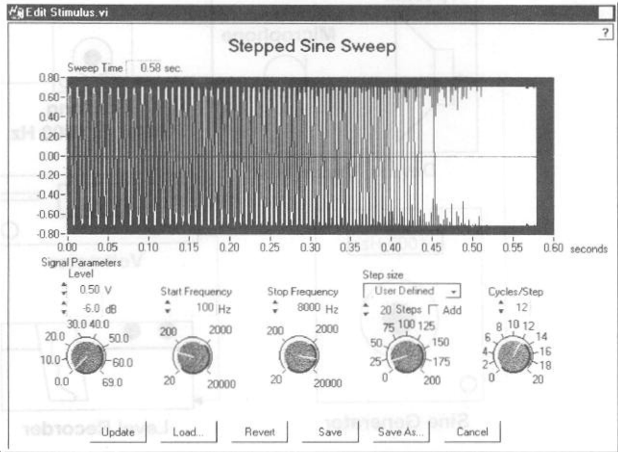


Figure 3 The Edit Stimulus.vi replaces the traditional sine generator found in hardware-based test systems

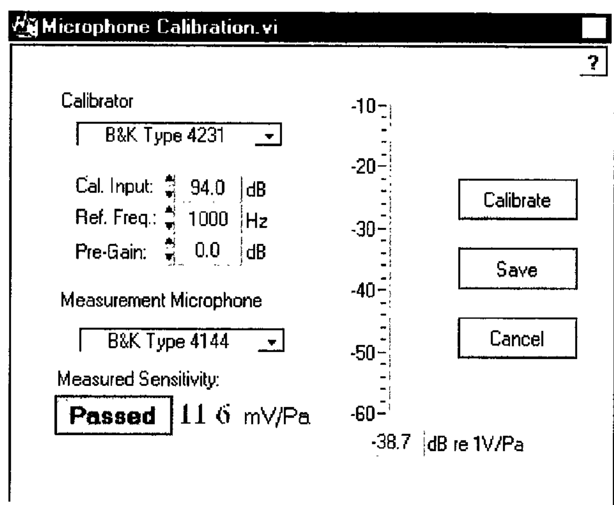


Figure 4 Acoustic Microphone Calibration for checking system conformance

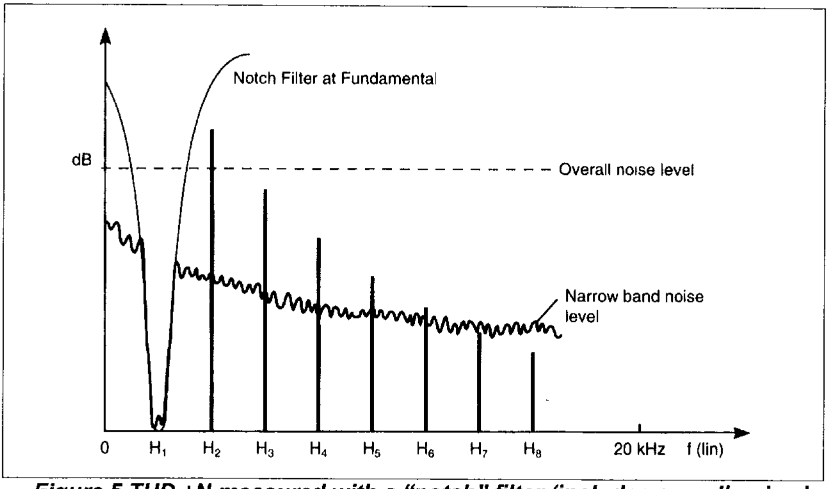


Figure 5 THD +N measured with a "notch" filter (includes overall noise level)

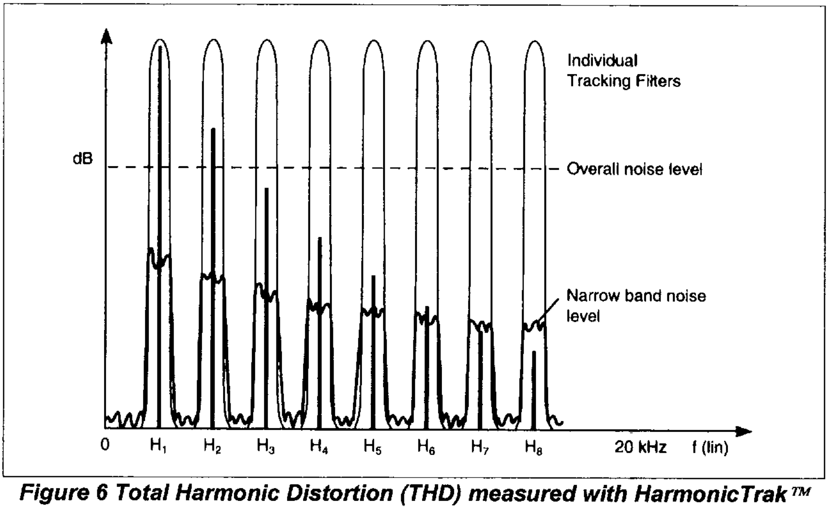


Figure 6 Total Harmonic Distortion (THD) measured with HarmonicTrak™ (includes selected distortion components)

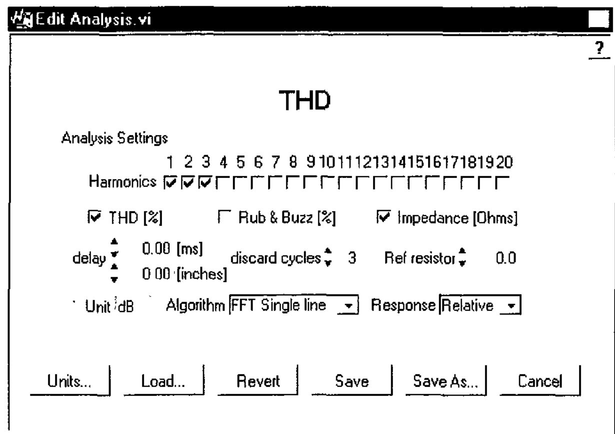


Figure 7 Analysis Setup automatically selects correct measurement algorithm

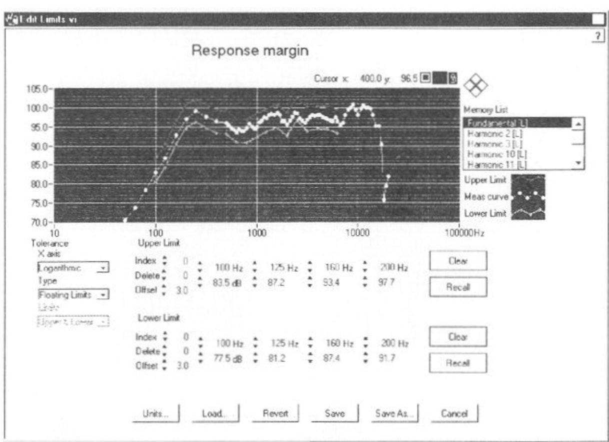


Figure 8 Limits Setup for specifying tolerances

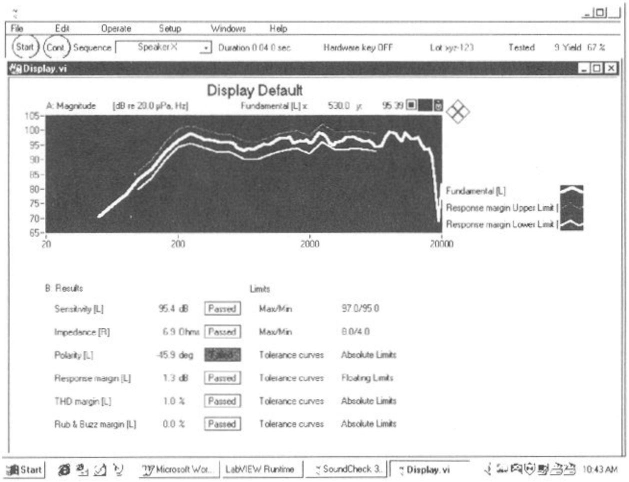


Figure 9 Final Data and Results Display with Pass/Fail Informationon a Variety of Different Tests

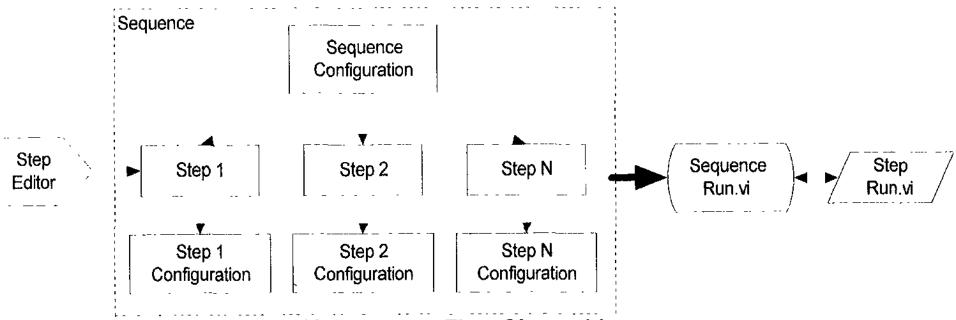


Figure 10 Sequence Flow Chart with Conditional Branching for Individual Steps