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USE OF SYNCHRONOUS TIME AVERAGING TO EXTRACT SIGNALS FROM NOISE

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INTRODUCTION

In the manufacture of industrial noise control equipment such as silencers and enclosures, it is important to be able to measure performance parameters related to the silenced and unsilenced sound pressure levels. For high performance silencing equipment, the silenced levels are often well below the ambient noise levels and cannot be measured with conventional instrumentation.

In order to extract the silenced signal from the total signal plus noise, we have developed a computer based measurement system based on synchronous time averaging. The system consists of hardware and software to: (1) synthesize a broad band periodic signal to be output on the digital to analog (D/A) converter, (2) synchronously capture and average a microphone signal at the analog to digital (A/D) input, and (3) analyze the resulting signal to produce narrow band, octave band and all pass sound pressure levels.

The ability to control all output and input signal parameters independently allows us to optimize the output signal to the sound production system and to eliminate errors associated with aliasing and leakage. The application of this system to field measurements has shown that improvements in signal to noise ratio of 30 dB are easily achieved making it possible, for example, to produce a high quality measurement of a 30 dB signal buried in a 50 dB ambient.

MEASUREMENT PROCEDURE

An analog input/output (I/O) card installed in a portable computer is used to output a signal to loudspeakers and to capture the resulting acoustic signal via a microphone connected to the input.

The output signal is first synthesized using separate software. Amplitude and phase angle spectra are defined over the frequency range of interest along with the sampling rate (Sr) and number of points (Np) associated with the time series signal to be output on the D/A converter.

The amplitude and phase information is used by an inverse FFT to generate the required digital time series signal. The resolution of the frequency domain signal is Sr/Np (Hz) and the duration, or period, of the time series signal is the inverse Np/Sr (seconds).

The resulting digital time series signal is broad band periodic containing all frequencies of interest and can be amplitude optimized for the frequency response of the sound production system being used; see reference 1.

The digital time series signal is output continuously (with no gaps at signal end points) as an analog voltage signal on the D/A converter of the analog I/O board. The analog voltage signal is fed through an amplifier to the sound production system (e.g. loudspeakers, horn driver, etc.).

The resulting acoustic signal is converted into a corresponding voltage signal by a microphone. This voltage signal is then amplified and returned to an A/D input channel of the analog I/O board.

The capture rate of the microphone input signal is synchronized exactly with the periodicity of the output signal; i.e. identical values of Sr and Np are used. As each period of the signal is captured, it is averaged with all prior captured periods. For systems of this type, synchronous signals average to themselves while non-synchronous signals average to zero.

Since all frequencies in the output signal are periodic in the time window, there is no "leakage" of signal energy into sidebands and the amplitude of each frequency component is precisely measured. Note that we would specifically not use tapered windows, e.g. Hanning, for signals such as those defined here. As well, the signal can be synthesized at high enough sampling rates to prevent aliasing errors without using low pass filters.

Theoretically, we expect non-synchronous signals, e.g. ambient noise, to decrease in proportion to "10*Log A" where "A" is the number of ensemble averages. Therefore for non-synchronous signals, 10 averages would be expected to reduce the level by 10 dB, 100 averages by 20 dB, etc. This effect is frequency independent; see reference 2, page 53.

The system is calibrated by applying a known rms sound pressure at the microphone and computing the zero-mean rms at the analog input which allows the channel calibration factor to be calculated. For a 1% accuracy, equivalent to 0.1 dB, a signal duration of at least eight periods is required; see reference 3. As a check, the microphone output voltage is also recorded. This allows the channel calibration factor to be calculated from the channel components. These two calibrations typically agree to within 1.5% which is equivalent to 0.15 dB.

Once the averaged input signal and calibration data has been obtained separate software is used to produce a narrow band Fourier transform which is then integrated into linear and A-weighted octave bands and all pass levels. For these calculations the A-weighting filter values are linearly interpolated between their one-third octave band center frequency values; see reference 2, page 15.

EXAMPLE

Fig. 1 shows one period of a time series signal that will be used to demonstrate the results of the measurement procedure. The signal parameters are: flat amplitude spectrum and random phase angle spectrum between 200 Hz and 2000 Hz, Sr = 20,000 Hz, Np = 1024 points, resolution = Sr/Np = 19.5 Hz, period = Np/Sr = 0.0512 seconds.

If this signal is fed from the D/A output directly to the A/D input and Fourier transformed, the resulting amplitude spectrum is as shown in Fig. 2. This figure shows that the computer I/O system has a S/N of 70 dB.

We now construct an experiment in which this signal is used to drive a small loudspeaker. A small tape recorder playing a broad band random signal is used to generate an interfering noise signal. A microphone attached to a sound level meter senses the acoustic signals; the A-C output of the SLM is fed into an A/D input of the analog I/O board. The microphone, loudspeaker and tape recorder are all in close proximity, within 5 cm to 10 cm of each other. By adjusting the volume controls and the distances between the microphone and sound sources, the sound pressure levels of the two sources can be set independently. The levels chosen for this experiment are: signal = 87 dBC, noise = 97 dBC, so that the S/N = -10 dBC.

If the computer generated signal, Fig. 1, is turned off and the interfering noise is played by itself, captured and analyzed, the results are as shown in Fig. 3 to Fig. 6 for 1, 10, 100 and 1000 averages. The corresponding data acquisition times are 0.0512, 0.512, 5.12 and 51.2 seconds. It is seen that the spectrum level decreases by approximately 10 dB for each factor of ten increase in the number of averages at all frequencies.

If the computer generated signal is now added to the noise signal, the results are as shown in Fig. 7 to Fig. 10 for 1, 10, 100 and 1000 averages. For one average, Fig. 7, the signal is 10 dB below the noise and the result is similar to Fig. 3 as expected. As the number of averages increases the interfering noise decreases and the signal emerges from the masking noise. Between 1 and 1000 averages the S/N increases from -10 dBC to +20 dBC.

For comparison, if the interfering noise is turned off and the computer generated signal is played by itself, the results are as shown in Fig.11 to Fig. 14 for 1, 10, 100 and 1000 averages. These figures demonstrate that the computer generated signal spectrum, 200 Hz to 2000 Hz, is independent of the number of averages and that the signal plus noise for 1000 averages, Fig. 10, produces the correct signal spectrum.

SUMMARY

The synchronous time averaging method can provide extremely high quality measurements that are consistent and repeatable and behave in a predictable manner. This method allows a direct and accurate measurement of sound pressure levels even when these levels are 20 dB or more below ambient levels.

REFERENCES

1. "Measurement Techniques for Engine Induction System Components", M.P. Sacks and S. Hackney, Proceedings of the Canadian Acoustical Association, Calgary, Alberta, Oct. 5 - 9, 1987
2. L.L. Beranek and I.L. Ver, Noise and Vibration Control Engineering, John Wiley & Sons, Inc., New York, 1992
3. "Use of Sine-Wave Calibration in the Frequency Domain", S. Braun, J. Acóust.Soc. Am., 58, 1653-1655 (1974)

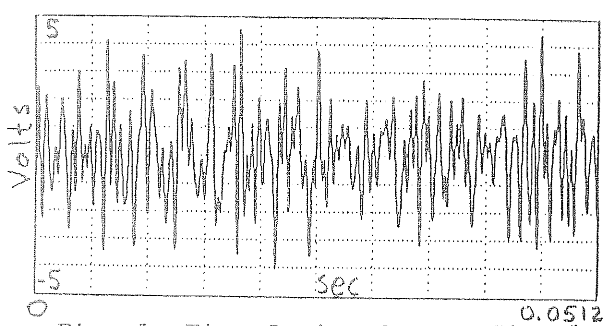

Fig. 1 Time Series Output Signal

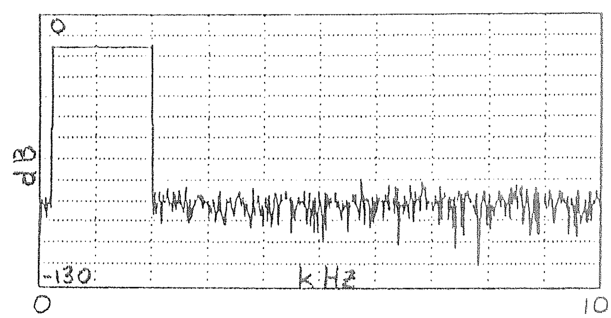

Fig. 2 Spectrum of Fig. 1

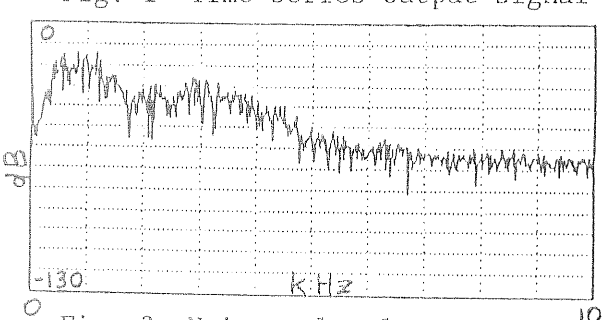

Fig. 3 Noise only, 1 avg

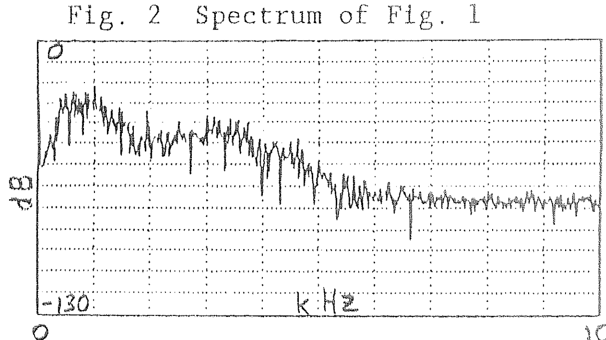

Fig. 4 Noise only, 10 avgs

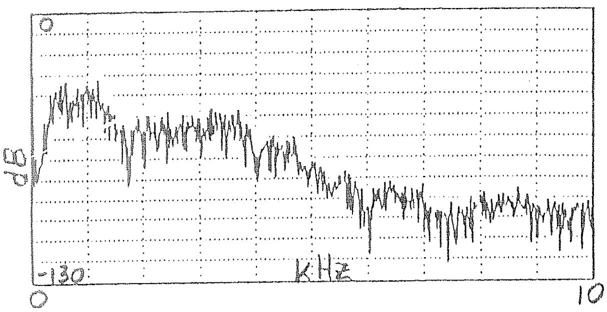

Fig. 5 Noise only, 100 avgs

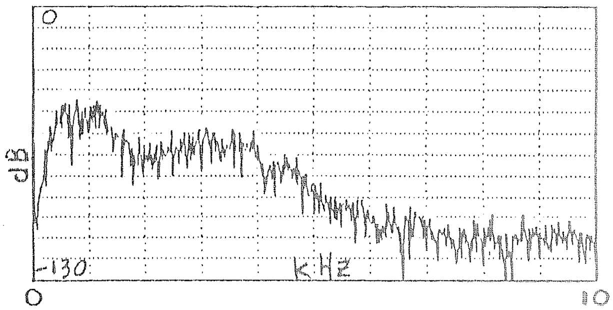

Fig. 6 Noise only, 1000 avgs

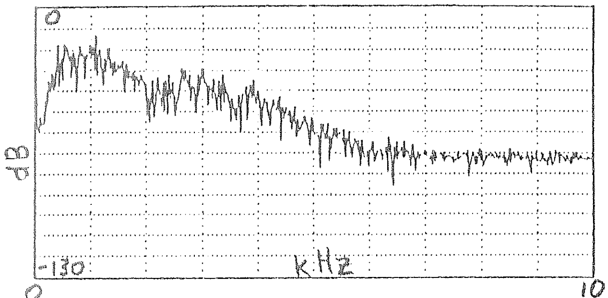

Fig. 7 Sig + Noise, 1 avg

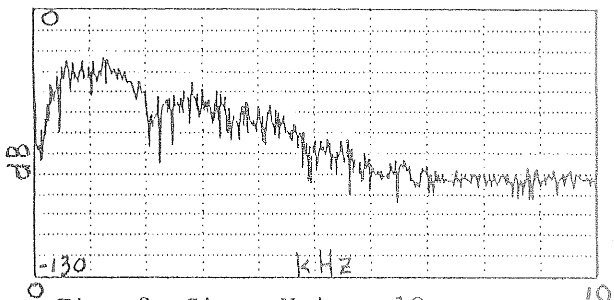

Fig. 8 Sig + Noise, 10 avgs

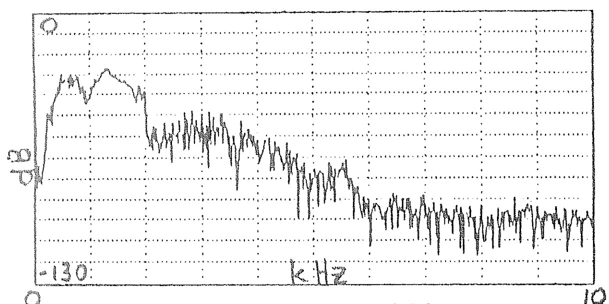

Fig. 9 Sig + Noise, 100 avgs

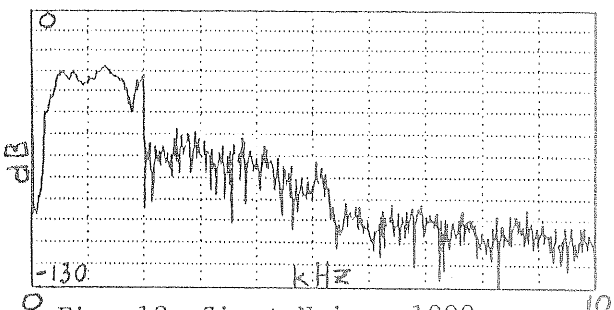

Fig. 10 Sig + Noise, 1000 avgs

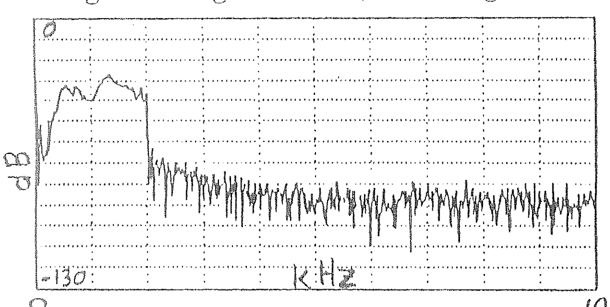

Fig. 11 Signal only, 1 avg

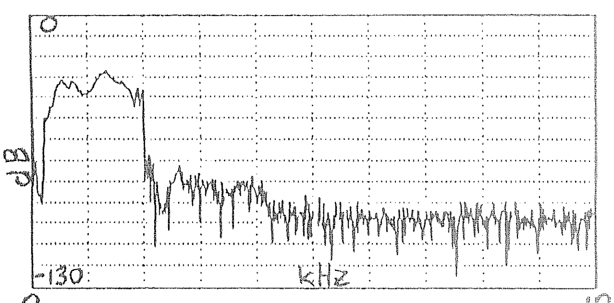

Fig. 12 Signal only, 10 avgs

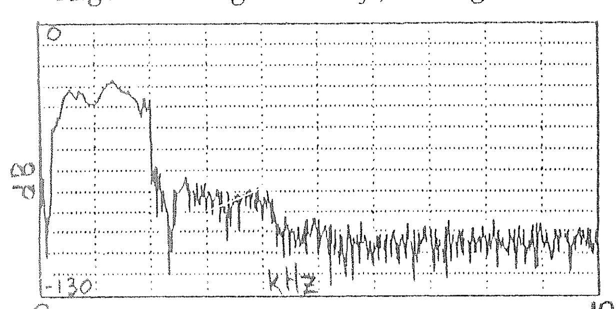

Fig. 13 Signal only, 100 avgs

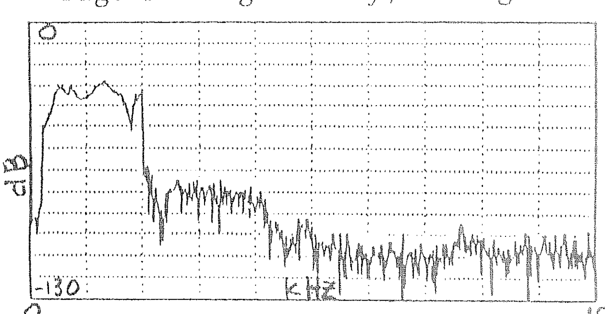

Fig. 14 Signal only, 1000 avgs