Noise and Vibration Data Acquisition using a 16 bit PC Sound Card

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ABSTRACT: Noise and vibration data acquisition cards for the Personal Computer have been available for about 10 years, commencing with 8 bit, then 12 bit and now 16 bit plug in cards. The data acquisition capabilities available range from two up to 64 or more channels, with sampling rate capabilities from a thousand samples per second to well over one million samples per second. The range of plug in cards available on the market is enormous. For a particular application, it is no easy task to determine the cheapest card available to meet the actual requirements. For data acquisition and analysis of rotating machinery noise and vibration, a frequency bandwidth of 10 to 20 kHz per channel is normally required. Unfortunately, the cost of special purpose 12 or 16 bit cards with this bandwidth tends to be at least two thousand dollars. This paper presents a detailed investigation, calibration and analysis of a mass market 16 bit sound card which is now available for under two hundred dollars and which has adequate performance for the analysis of rotating machinery noise and vibration. The data acquisition capabilities under the windows environment are discussed along with channel calibration, amplitude and phase response and the ease of transferring the data to post-processing environments such as MATLAB.

1. INTRODUCTION

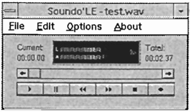
The measurement and analysis of noise and vibration is recognised as being essential for routine condition monitoring for most types of rotating machinery. Special purpose data collectors tend to be used as the prime means of collecting, storing and analysing the routine vibration measurements in a large number of industries [1]. For more specialist data acquisition and machinery monitoring problems, tape measurements may be made prior to analysis with dynamic signal analysers or Personal Computer data acquisition systems may be used to capture and post-process the data [2]. An unfortunate aspect of the current generation of PC based data acquisition systems is the cost which starts at around two thousand dollars and extends up to five thousand dollars for typical 12 or 16 bit multi-channel cards. For university or research use, the cost for single or multiple cards can be prohibitive.

An alternate data acquisition capability is now available through the use of plug in sound cards [3]. The mass-market sound cards are typically used to produce high quality sound from the PC, and are used routinely for computer games, educational software and general multi-media applications. The major reason for the current interest in sound cards is the cost, with most two channel 16 bit cards selling for under two hundred dollars. In order to determine the suitability of the sound cards for general analogue data acquisition, a number of aspects of a Sound Blaster 16 Basic card were thoroughly investigated including the windows data acquisition capabilities, transfer of the WAV files into MATLAB, phase and amplitude response, input voltage ranges, calibration and the cross channel noise. This paper details the results of the investigation, provides a summary of the advantages and disadvantages of using sound cards for general data acquisition and describes a range of suitable applications for their use in Mechanical Engineering.

2. GENERAL DATA ACQUISITION CAPABILITY

The Sound Blaster 16 bit card provides two main options for general data acquisition using either the microphone or line inputs. The microphone input provides a single channel of data whereas the line input can be configured as either one or two (stereo) channels [4]. The Sound Blaster card comes complete with a range of DOS and Windows software products for controlling the channel gain, sampling rate and resolution for general data acquisition. The Windows software which was found to provide the easiest means of transfering the resulting WAV files into MATLAB was the Soundo'LE application. This could be run as an external executable application from within MATLAB using the *!c:\sb16\winappl\soundole.exe* command to execute the application.

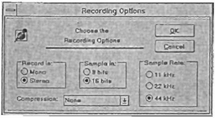
The graphical user interface of the Soundo'LE application is shown in Figure 1 where the basic record, play back, pause, rewind and fast forward buttons are available. The recording option available under the options menu shown in Figure 2 allow the number of channels, sample rate and the resolution to be selected as required. The sample rate is per channel with simultaneous sampling on both channels. Figure 3 shows the menu to choose the actual data channel for recording purposes and the channel gain selection. The channel gain is independant of the channel selected which allows for the acquisition of the same data channel at different gains if required.



Current 00:00.00

Total: 00:02.37

Figure 1. Sound Blaster 16, Soundo'LE Graphical user Interface.



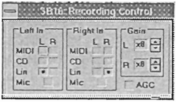
Record in: ○ Mono ● Stereo

Sample in: ○ 8 bits ● 16 bits

Sample Rate: ○ 11 kHz ○ 22 kHz ● 44 kHz

Compression: None

Figure 2. Control of the sample rate, resolution and number of channels.



Left In: Lin L ■

Right In: Lin R ■

Gain: L x8, R x8

□ AGC

Figure 3. Channel and gain control interface.

Having specified the data acquisition requirements, clicking on the record button on the lower right portion of the menu shown in Figure 1 digitises the data and transfers it to the Windows environment using DMA. The data acquisition capabilities of the Sound Blaster card under Windows are impressive. At the maximum sample rate of 44.1 kHz per channel, for two channels and 16 bit resolution, the data is digitised and transferred continuously to the Windows environment at 176400 Bytes/sec until the acquisition is stopped or the hard disk is full. After acquisition, the data files are available to be saved as industry standard WAV files and are compatible with the Windows Media Player and Sound Recorder, allowing some basic editing to be done if required.

It is also possible to transfer the WAV files into MATLAB by writing a MATLAB M-file to read the WAV file once the format of the WAV file is known. Appendix A shows a typical M-file which can be used to read any version of the WAV files supported by the Sound Blaster series cards. The M-file first reads the file header to determine the format of the data, whether 8 or 16 bit acquisition was used, the number of channels and the sampling rate, etc. After the WAV file is read into MATLAB, the data can then be treated as any other MATLAB vector allowing very sophisticated signal processing techniques to be used to analyse the data [5].

3. INPUT VOLTAGE RANGE

The Sound Blaster 16 bit card has menu selectable independant gain control on the two channels. When the card is used to digitise analogue voltage waveforms through the line input, the maximum input voltage range for the various gains has to be determined as it is initially unknown and not specified in the technical documentation. In order to determine the input voltage range, harmonic distortion analysis was used to specify the maximum input voltage level as a function of gain. A Hewlett-Packard 35665A Dynamic Signal Analyser was used to provide a 1 kHz analogue sine waveform as input to the line-in on the Sound Card for various amplitudes and gains. With the input signal sampled at 44.1 kHz, power spectrum analysis was used to measure the amplitude ratio between the 1 kHz component and the next highest harmonic component in dB. Figure 4 shows the harmonic distortion results as a function of gain and the input voltage amplitude.

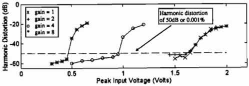


Figure 4. Harmonic distortion as a function of gain and input voltage.

With 0.001% harmonic distortion or 50dB being used as a basis for determining a maximum input voltage range, the voltage ranges can be read from Figure 1 as a function of gain. Table 1 provides a summary of the input voltage ranges corresponding to a harmonic distortion of 0.001%.

Gain	Input Voltage Range
1	± 1.65
2	± 1.6
4	± .95
8	± .47

Table 1. Input voltage ranges for harmonic distortion of 0.001% (50 dB) and 44.1 kHz sampling rate.

The input voltage range was not a linear function of gain. As shown in Table 1, the maximum input voltage range for a gain of 1 and 2 were very similar. The conversion of input voltage to quantum level at these gains did not cover the complete range of the A/D converter, (-32767 to + 32768 using 16 bit conversion) indicating that the conversion was less than 16 bit. This would have to be taken into account in determining the resolution and calibration of the card at these gains.

4. RESOLUTION

The resolution of the sound card was determined by grounding the input channel and digitising the resultant noise floor as a function of channel gain for both 8 and 16 bit conversion. The results are shown in Table 2 as standard deviation of the number of quantum levels in the noise floor.

Gain	8 bit conversion	16 bit conversion	Resolution Standard Deviation (mV)
1	0	2.2	0.27
2	0	3	0.18
4	0	5	0.145
8	0	9	0.13

Table 2. Typical noise level measured with grounded inputs and specified as standard deviation of the number of quantum levels.

With 8 bit conversion, the grounded input had a zero quantum level noise floor across all gains. With 16 bit conversion, the resolution became poorer as the gain was increased and had a typical maximum standard deviation of 9 quantum levels at a gain of 8. With a maximum input voltage range of 0.47 Volts, this corresponds to a standard deviation resolution of 0.13 mV as shown in Table 2. The resolutions obtained as a percentage of the input voltage range are also shown in Table 2. The resolution available with gains of 1 and 2 were found by taking into account the reduction in the number of quantum levels available for the input voltage ranges.

5. CALIBRATION

Knowing the input voltage range as a function of gain, a crude calibration from quantum level into voltage can be made. For a more accurate calibration, a statistical analysis of the input and digitised waveform was used. A 399.89mV peak sinusoidal waveform from the HP Analyser was captured at a sample rate of 32.768 kHz using the HP analyser and 20480 time points were transfered to MATLAB for analysis. The waveform was also digitised by the sound card and then transfered into MATLAB for analysis at the various gains. By computing the RMS of the two time signals, a calibration factor was then computed to convert the digitised sound card quantum level into Volts. Table 3 provides the resulting calibration values for the various gains and resolutions.

Gain	8 bit conversion	16 bit conversion
1	29.772×10^{-3}	11.663×10^{-5}
2	14.910×10^{-3}	5.814×10^{-5}
4	7.451×10^{-3}	2.913×10^{-5}
8	3.744×10^{-3}	1.464×10^{-5}

Table 3. Calibration factors as a function of gain and resolution.

The calibration factors will vary from card to card. The values shown above should be taken as indicative only.

6. DUAL CHANNEL FREQUENCY RESPONSE

The amplitude and phase response of the sound card was measured by using a random noise signal from the HP analyser. The signal was connected to both channels of the sound card and after digitising the signal and saving the data as a WAV file, the two channels of data were read into MATLAB and the amplitude and phase of the complex transfer function computed as a function of frequency. Using the 16 bit resolution and the 44.1 kHz sampling frequency, the amplitude and phase response of the sound card is shown in Figure 5. An external analogue anti-aliasing filter was used with a low pass cut-off frequency of 15 kHz, and so the results up to 15 kHz are shown.

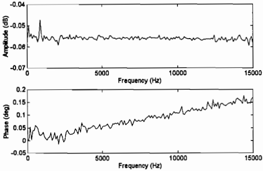


Figure 5. Amplitude and phase response across the two channels of the sound card.

The amplitude response between the two channels appears to be flat within 0.01 dB across the frequency range as shown except for the low frequency region. A closer inspection of the low frequency response of the sound card was undertaken to ascertain the low frequency behaviour. Table 4 shows the amplitude response of the card below 12 Hz. The amplitude response was found to be within 1 dB above 10 Hz which suggests AC coupling as would be expected for sound measurement.

Frequency (Hz)	Amplitude Response (dB)
1	-14
2	-8.3
4	-3.9
6	-2.2
8	-1.35
10	-0.9
12	-0.6

Table 4. Low frequency amplitude response of the sound card.

The phase response of the card shown in Figure 5 shows a linearly increasing phase within 0.2 degree up to 15 kHz which is representative of a small time delay between channels. At 15 kHz, the phase lead across the two channels was approximately 0.16 degrees which corresponds to a negligible time difference of 0.03 msec between channels. The time difference is most probably due to some small delay in triggering the A/D conversion on both channels. For the maximum sampling rate of 44.1 kHz, the data acquisition can be considered to be simultaneously sampled. Excluding the linearly increasing phase, the phase response is within 0.1 degree across the whole frequency range.

7. CROSS CHANNEL NOISE

The cross channel noise was investigated by digitising both channels at the maximum sampling rate using 16 bit resolution, with channel 1 having 0.4, 0.8, 1.4 and 1.4 peak Voltage sinewave inputs at the gains of 1, 2, 4 and 8 respectively for various frequencies while channel 2 was grounded. The resulting cross channel isolation from the input to the grounded channel is shown in Table 5 in dB. The worst case scenario occurs at the highest frequency with a gain of 8 where the inter-channel isolation is approximately 63 dB.

Frequency (kHz)	Gain of 1	Gain of 2	Gain of 4	Gain of 8
1	> 90	> 90	> 90	> 80
5	> 80	> 80	> 80	> 75
10	> 75	> 75	> 75	> 68
15	> 70	> 70	> 72	> 65
20	> 68	> 68	> 70	> 63

Table 5. Cross channel isolation as a function of frequency and gain (dB) sampled at 44.1 kHz.

8. DISCUSSION

From the investigation in using a PC sound card for data acquisition, a summary of the advantages and disadvantages which were found is shown in Table 6. The major advantage in using the sound card for data acquisition over the more expensive dedicated A/D plug in cards is the cost. For less than $200 the sound card provides 16 bit simultaneous 2 channel sampling capability under the MS-Windows environment, including DMA for continuous data capture. The software provided was very robust and did not crash even for long time capture records of tens of megabytes. The menu driven software was very easy to use and was seen as providing a usable and efficient data acquisition capability.

The inter-channel isolation of 63 dB at the maximum gain and frequency range is of concern and is probably the major deficiency of the card as compared with the more expensive dedicated plug in cards which can provide up to 90 dB of cross talk over the full frequency range [6]. The sound card had no DC A/D capability and the response dropped off below 10 Hz. For those applications which require a low frequency or DC measurement, then the sound card will not be suitable.

Advantages	Disadvantages
16 bit resolution	No calibration from bits to Engineering units
Adjustable gain, sampling rate and resolution	No anti-aliasing filtering
2 channel, simultaneous sampling	No DC capability
A/D capability under MS-Windows	Low level of technical information available
DMA for continuous data capture	Difficult to write MS-Windows A/D software
Easy to use MS-Windows software	63 dB cross channel isolation at 20 kHz
.wav files can be read using MATLAB	
Line and mic inputs	
Extremely cheap (< AUS$200)	
Readily available for educational and home use	
Available on desktop PC's, laptops and notebooks	

Table 6. Summary of the advantages and disadvantages of PC sound cards for vibration data acquisition.

A number of vibration and noise applications will be suitable areas for using the sound card such as sound intensity, gear vibration condition monitoring and general dual channel transfer function analysis. The sound card has been shown to provide high quality amplitude and phase response across a broad frequency range which is required for all of these signal analysis areas. Aside from the dual channel signal analysis areas, the card will also be useful as a general purpose single channel data acquisition system for those applications which require analogue signals with bandwidths below 20 kHz to be digitised. The card has the potential to be very useful in Mechanical Engineering undergraduate programs in laboratory situations as a general purpose data acquisition capability. This could include measurements such as pressure, strain, vibration, force, etc, for laboratories and final year projects where the expense of the normal data acquisition cards becomes prohibitive.

The sound cards present somewhat of a problem for the application specific data acquisition area where special purpose software is required to do acquisition and analysis all in one. While software libraries are available using 'C' and pascal for the sound cards, the task of writing the software under the windows environment is a difficult one which in the opinion of the author should be left to computer programmers. The ability to develop Matlab compatible 'C' data acquisition code under Windows would be very beneficial as a MATLAB graphical user interface could then be developed integrating the data acquisition and analysis.

9. CONCLUSION

The Sound Blaster 16 basic card has been shown to be a very capable general purpose 16 bit data acquisition card with 2 channels sampled simultaneous at a maximum rate of 44.1kHz. The inbuilt DMA provides a continuous sampling capability direct to the MS-Windows environment and will allow data to be collected until the hard disk becomes full. The easy to use menu driven software allows the selection of sample rate, resolution, number of channels and gain control. The WAV files can be read into MATLAB once the format of the WAV file is known and a sample version has been provided. The price of the 16 bit cards (< $200) means that they are readily available on the normal desktop computer at work, home, and for undergraduate student laboratory or project purposes. While the single channel amplitude and phase response of the card was not measured, the cross channel amplitude and phase response of the card contained less than 0.01 dB and 0.1 degree variation respectively across the bandwidth which was tested (<15 kHz).

The calibration of the sound card was relatively straight forward given a precision waveform generator. The calculated calibration factors then allow for the digitised waveform to be converted to Volts and then to engineering units once the sensitivity of the particular transducer is known.

The input Voltage range which can be used on the line input can pose a problem as it is initially unknown. By using a harmonic distortion test, the input voltage range was determined as a function of gain and the maximum input range was determined to be ± 1.65 Volts with a harmonic distortion of 0.001% or 50dB. The card does not provide any anti-aliasing protection so a low pass analogue filter must be used with the appropriate setting to filter the input waveform prior to digitisation.

The sound cards are seen to provide a very inexpensive alternate data acquisition capability. Further investigation into various technical aspects not shown in this paper would seem to be warranted.

10. ACKNOWLEDGMENT

The time and effort given by Mr. Wang-Juin Yee in the initial investigation of the capabilities of the sound card is greatfully acknowledged. Thanks also goes to Dr R.M. Howard for the helpful technical discussions.

11. REFERENCES

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Appendix A: MATLAB M-file for reading WAV files

```
function [chan1,chan2,dt,format]=lwave(wavefile)
% This function load any windows compatible
% .wav file
%
% the format of the file is returned in format
% the data is returned in chan1 and chan2
% the sample time period is returned in dt
%
% Modified from the MATLAB wavread.m file
% pop up window to select file
[filename,path] = uigetfile('*.wav');
if filename==0
    % cancel operation
else
    % rearrange path and filename of desired          % mat file
    wavefile = [path,filename];
    % load file into workspace
    eval('fid=fopen(wavefile,"rb");');
end
if fid == -1
    error('Can"t open .WAV file for input!');
end;
if fid ~= -1
    % read header
    header=fread(fid,4,'uchar');
    header=fread(fid,1,'ulong');
    header=fread(fid,4,'uchar');
    header=fread(fid,4,'uchar');
    header=fread(fid,1,'ulong');
    % read format from file header
    % PCM format
    format(1)=fread(fid,1,'ushort');
    % Number of channels
    format(2)=fread(fid,1,'ushort');
    % Sampling frequency
    fs=fread(fid,1,'ulong');
    % specify sampling rate
    dt = 1/fs;
    format(3)=fs;
    % average bytes per second
    format(4)=fread(fid,1,'ulong');
    % block alignment
    format(5)=fread(fid,1,'ushort');
    % bits per sample
    format(6)=fread(fid,1,'ushort');
    % read header
    header=fread(fid,4,'uchar');
    % Number of samples in file
    nsamples=fread(fid,1,'ulong');
    % read data in correct format
    if format(6)==8   % 8 bit data
      % read data in 8 bit format
      d=fread(fid,nsamples,'uchar');
    elseif format(6)==16 % 16 bit data
      % read data in 16 bit format
      d=fread(fid,nsamples,'short');
    end
    % reorder data if in two channel mode
    if format(2)==2
      chan1 = d(1:2:length(d));
      chan2 = d(2:2:length(d));
    elseif format(1)==1
      chan1 = d;
    end
    % close file
    fclose(fid);
end
```