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Quantum Analogs

Appendices

Advisor and Student Manual

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Appendix 1

TeachSpin's Quantum Analogs Controller Box
Technical Description

The following chart provides a description of the role of each component on the Controller Box.

Controller Label	Function
Microphone Input	provides a source of +5 V dc (for biasing of capacitor microphones), and accepts the ac signal placed atop that bias by the microphone
AC Amplifier	provides a fixed gain, of order 100, from about 20 Hz to 20 kHz; ac-coupled at input
Attenuator	10-turn scale, provides *attenuation* of amplified ac signal, by a factor given by (dial setting)/10. Example -- a dial setting of 9.5 turns implies an *attenuation* of 9.5/10 = .95 or 95%. Only 5 % of the signal is transmitted! NOTE: A higher reading means less signal.
AC Monitor	provides a direct view of the amplified ac signal at the attenuator's output
Envelope Detector	a rectifier system, giving the amplitude of the sine-wave signal present at the AC Monitor output, on a cycle-by-cycle basis
Detector Output	a dc-coupled positive voltage, the output of the envelope detector
Sine Wave Input	provides the entry point for ac signals from signal generator or computer sound-card
Speaker Output	directly coupled to Sine Wave Input below it on the panel; provides the point of attachment for 3.5-mm speaker plug
Frequency-to-Voltage Converter	when toggled to On, this module derives a signal from Sine Wave Input, and converts its frequency to a voltage, at conversion ratio 1 Volt per kHz
DC Offset	10-turn dial, allowing the addition of a 0 to -10 Volt offset to the output of the Frequency-to-Voltage converter
DC Output	the (possibly dc-offset) output voltage of the F-to-V converter module

Quantum Analogs uses sound waves in cylinders and spheres to model the quantum states in semiconductors, hydrogen atoms, and hydrogen molecules. In these experiments, our dependent variable is usually the amplitude of the sound detected at the microphone. The Controller accepts the microphone signal at its Microphone Input and amplifies the signal's ac component. The Attenuator is then used to decrease or increase the magnitude of the amplifier output to keep the signal size in an appropriate range. For example, in experiments using the computer, the signal size must be in the range for which the sound card has a linear response. If the AC Monitor signal exceeds the peak-to-peak voltage limits for your particular computer, the sound card input will begin to saturate. This will give distorted response curves. **Appendix 2 – Recognizing and Correcting Saturation** provides a detailed explanation.

Appendix 2

Recognizing and Correcting Saturation

When you are using a computer sound card both as the source of the speaker waveforms and as the detector of the resulting waveforms from the microphone, you must check to make sure that the signals are not saturating. Because the specifications for computer soundcards vary widely, we cannot be sure that keeping signals below one or two volts peak-to-peak will be sufficient. You will have to experiment with your own system to see what works.

In the Quantum Analogs computer-based experiments, we use the amplitude of the microphone signal on the computer screen as an arbitrary measure of the intensity of the sound the microphone is receiving. For these *relative* measurements to be accurate, the system must be operating in a region where the relationship of signal to response is linear. This means that any change the input signal from the microphone must result in proportional changes in the heights of the peaks.

The following instructions assume that you have an experiment set up, and have the Quantum Analogs program “SpectrumSLC.exe” running. One way to determine if you are operating in the linear range is to perform repeated scans over some feature, such as a single resonance, and to vary the attenuator setting on the Controller.

First, use the Quantum Analogs program to adjust the speaker output and microphone input strengths on the computer. To set the speaker intensity, move the slider marked Amplitude Output Signal to the middle of its range. (The slider is in the lower left corner.) To set the microphone input, go to the menu across the top of the screen. Choose Configure > Input Channel/Volume > Microphone. Set that slider to the middle of its range also.

Now take a series of spectra while adjusting the ATTENUATOR knob on the Controller. The microphone signal coming from the apparatus first passes through a built-in amplifier. It then goes through the attenuator before reaching the *AC-MONITOR* connector. The ten-turn knob on the attenuator *diminishes* the incoming signal by a factor ranging from zero to one, so a setting of 9.8 turns stands for an *attenuation* of 0.98 (or 98 %) relative to the maximum possible. Only 2 % of the signal is being transmitted.

After taking an initial wide-range spectrum, choose a section that includes the highest peak and a smaller one next to it. Readjust the scan to cover just this portion. Using the option that allows you to keep successive spectra visible, take Spectrum 1, 2, 3, etc. with the attenuator knob set at 9.8, 9.7, 9.6, and 9.5 turns (out of ten). The nesting heights of the peaks will tell you whether or not the system is behaving in a linear fashion. Continue to go to lower numbers on the 10-turn dial setting until you have visual evidence of saturation. (The peak heights will no longer be increasing evenly.)

Once you have reached saturation, increase the attenuation until you are back into the linear range. Now you can operate with confidence that the signals you see really are proportional to the amplitude of the sound wave you are studying.

Appendix 3

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Connecting to a Computer – Instructions & Troubleshooting

Choice of Sound card

You should use a sound card of reasonable quality. It may cost about $50 to $100. A very cheap soundcard (including some on-board cards) may have one of the following problems:

(1.) The sample rate may not be high enough. (44100 Samples/second is needed)
(2.) There might be high cross-talk between output and input signal.
(3.) The card might not be able to deal with input and output simultaneously
(4.) The Line-In mode may be lacking.

Installation of Soundcard

Install the drivers of your soundcard as directed by the vendor. Some soundcard drivers come with a set of filters and sound-effects which may be switched on by default at installation. Be sure to switch *off* all these filters and effects.

Connecting the soundcard output signal to the Quantum Analogs Controller

Build a simple experimental system, such as a one-dimensional layout of the speaker, three cylinders, and the microphone laid out in the V-channel. Attach a BNC splitter or “tee” to *SINE WAVE INPUT* on the Controller. Using the adapter cable, connect the output of the sound card to one arm of the splitter. With a BNC cable, convey the soundcard output signal from the splitter to Channel 1 of your oscilloscope. Plug the lead from the speaker end of your experimental tube to *SPEAKER OUTPUT* on the Controller. The sound card signal is now going to both the speaker and Channel 1. (Some soundcards have more than one output connector. In this case, you should check the manual of the soundcard to determine which connector is used for headphones, and use this connector.)

Choose Line-In mode of the soundcard

The microphone signal of the Quantum Analogs experiment has been amplified in the Controller. A signal with up to 10V, rms, is provided at the *AC-MONITOR* of the Controller. Depending on the strength of the resonances in your experimental setup, you will have a signal of a few volts on the *AC-MONITOR* connector. Soundcards have two different modes of input that can be used, the Line-In mode and the Microphone mode. ***Whenever possible use the Line-In mode.*** The mode of the soundcard can be set in the program SpectrumSLC.exe by going in the menu to Configure > Input Channel/Volume. In the window that opens, there is a box showing all available input-modes of your individual soundcard. Choose Line-In. If there is no Line-In mode, choose Microphone and read the next section. In the Line-In mode, the soundcard samples a signal of about 1V rms maximum. The output level of the Controller at *AC-MONITOR* may need to be attenuated using the *ATTENUATOR* knob on the Controller to avoid saturation of the analog-to-digital converter in the sound card. Read the section on saturation on page A3-4. Be sure that the voltage at AC-Monitor does not exceed 1V rms maximum. In the worst case, the soundcard could be damaged by excessive voltages.

Using Microphone-mode instead of Line-In-mode

Whenever possible, the Line-In mode should be used. If your sound card does not provide such a mode, it may be possible to use the microphone mode. However, it is likely that you will have problems with saturation and/or distortion if the signal is not adjusted to the optimum range. In the Microphone-mode, the soundcard sends out a DC-voltage as the bias voltage for the microphone. This DC-voltage can make a significant offset of the input signal. In addition, the input is much more sensitive (by a factor of about 100) compared to the Line-In mode. The Input signal will already be saturated at a level of about 10 to 100 mV rms. To avoid saturation, you will need to attenuate the signal much more than in the Line-In mode. The attenuator may need to be high as 9.9 turns to prevent saturation. Read section on saturation below.

Connecting the soundcard-input to the Quantum Analogs Controller

The microphone signal will be sent two different places. Connect the microphone on your experimental tube to *MICROPHONE INPUT* on the Controller. Put a BNC splitter on the Controller connector labeled *AC-MONITOR*. From the splitter, use the adapter cable to send the amplified microphone signal to the Line-In input on the computer soundcard, and a BNC cable to send the same signal to Channel 2 of the oscilloscope. Channel 2 will show the actual signal coming from the microphone. Some Soundcards have different connectors for Line-In and for Microphone mode. If this is the case be sure to use the Line-In connector. Other sound cards use the same connector for both, Line-In and Microphone mode. The electrical properties of the connector are switched by switching the mode in the software. Some of the soundcards use the same connector even for digital input. You should choose the correct mode in the software before connecting the cable. You should also take care not to plug a cable with voltage on it into the soundcard. This might destroy the sound card. Please avoid any static electrification, and do not touch the central wire of the connector when plugging in or out. Be sure that the voltage at sound-card input does not exceed 1V rms maximum. In the worst case, the soundcard may be damaged by excessive input voltages.

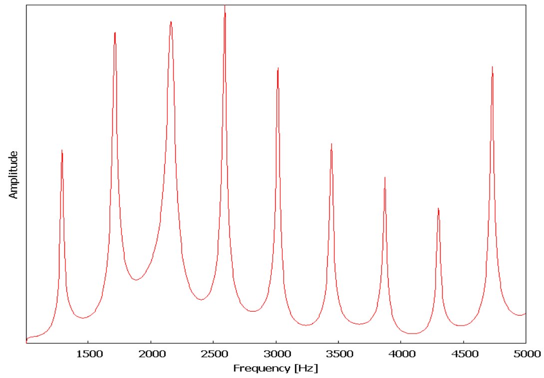
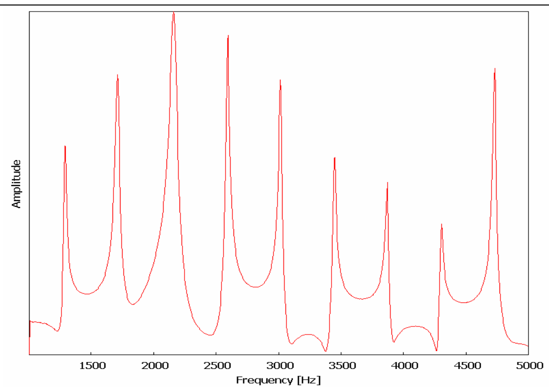
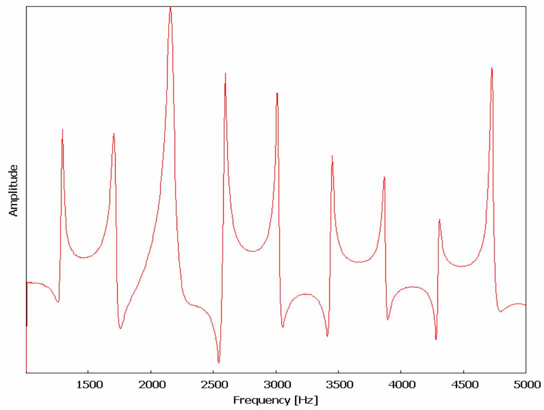
Setting input and output levels in the computer.

There are four different places where input and output levels can be controlled. You should be aware of all of them, to be sure that you are using your system with the optimal settings.

1. The Output level of the speaker can be set within Windows as it is done when you hear music from the computer. You may open System-Control > Sounds and Audio > Audio > Volume and adjust the output volume. On some computers, there is an output-volume slider provided for easier adjustment.
2. In the program SpectrumSLC.exe, there is a slider in the lower left corner labeled Amplitude output signal. It determines the amplitude of the sine wave sent to the output by the program. The slider can be used to change output level easily in measurements. Be aware that (1) and (2) are independent ways to adjust the output level. They are both effective.
3. The input level in either the Line-In mode or the Microphone-mode can be adjusted by the slider provided in the program SpectrumSLC.exe that is available in the menu of the program when going to Configure > Input Channel/Volume.
4. The *ATTENUATOR* knob on the Quantum Analogs Controller also determines the level of the microphone signal before it goes into the sound card.

Cross-talk of the channels

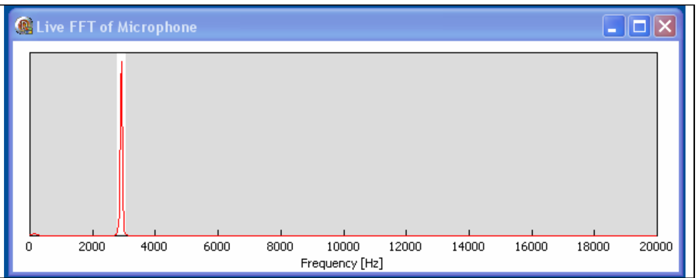
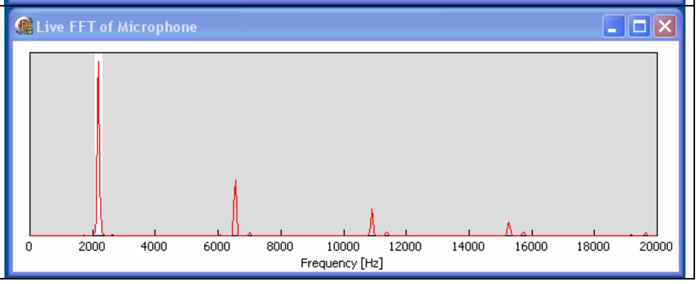
Depending on the quality of your sound card and the actual settings, there might be a problem with cross-talk between the output and input channels. If the speaker output signal is getting to the input channel internally on the sound card, this creates a flat background in the measurement which interferes with the detected signal. This can be a serious problem, since line-shapes of the resonances are modified significantly. See the figure below for examples. If you detect cross-talk in your experiment, it may help to reduce the amplitude of the speaker output signal. In very cheap sound cards, a feedback loop may build up due to the cross-talk. In this case, a fixed frequency is observed in the resonator, which does not sweep when sweeping a frequency scan. When this happens, the only solution is to purchase a better soundcard. Read the next section on detecting problems with the signal.

<table>
<tr><td>Spectrum without cross talk</td><td>
</td></tr>
<tr><td>Spectrum with cross talk</td><td>
</td></tr>
<tr><td>Spectrum with severe cross talk</td><td>
</td></tr>
</table>

Detecting problems with the signal

A reliable way to detect problems with the signal is to have a look at the live image of the Fourier-transformation of the input signal. The program SpectrumSLC.exe provides this information in an extra window that can be opened by going to the Menu > Windows > Live FFT of the Microphone signal. When sweeping a spectrum, there should be a single peak sweeping from low frequency to high frequency. If there are additional peaks, you have a problem with the signal. There can be peaks at higher harmonics (double or triple frequency) which indicate distortion or saturation of the signal. If there are peaks at *fixed* frequency during a measurement which do not sweep, you probably have a feedback loop (read about cross talk) or you might have another external signal coupling into your experiment (external sound or external electrical AC signal). Additional peaks can also be created by software filters or sound-effects of the sound card. In this case, switch off all software filters and sound-effects in the sound card software.

If, during this sort of sweep measurement, you have a single peak on a low background, the signal is perfect. See figure below.

<table>
<tr><td>Fourier-transformation of a perfect signal</td><td>
</td></tr>
<tr><td>Signal with higher harmonics due to distortion or saturation</td><td>
</td></tr>
</table>

Saturation

Problems with saturation of the analog-to-digital converter in the sound card are particularly likely when using the Microphone mode instead of the recommended Line-In mode. To prevent saturation, the input-signal needs to be reduced to an optimum range. In the newest version of the SpectrumSLC.exe software (versions starting with 7.1), there is a blue bar in the lower left corner of the main window which indicates saturation. If the blue bar blinks while you are passing through the top of a peak, the signal is saturating the analog-to-digital conversion of the sound card. In this case, you need to reduce the level of the signal by increasing the reading on the ATTENUATOR knob. Saturation is observed at amplitudes above 100 units on the computer display.

There is another way to check the linearity of the signal transmitting path. In the Quantum Analogs computer-based experiments, we use the amplitude of the microphone signal on the computer screen as an arbitrary measure of the sound-amplitude the microphone is receiving. If these relative measurements are to be accurate, the system must be operating in a region where the relationship of signal to response is linear. This means that increases in the input signal from the microphone must result in corresponding increases in the heights of the peaks.

One way to determine if you are operating in the linear range is to perform repeated scans over some feature, such as a single resonance, and to vary the attenuator setting on the Controller. The following instructions assume that you have an experiment set up and have the Quantum Analogs program "SpectrumSLC.exe" running.

Take a series of spectra while adjusting the ATTENUATOR knob on the Controller. The microphone signal coming from the apparatus passes first through a built-in amplifier, and then goes through the attenuator before reaching the *AC-MONITOR* connector. The ten-turn knob on the attenuator diminishes the incoming signal by a factor ranging from zero to one, so a setting of 9.8 turns stands for an attenuation of 0.98 relative to the maximum possible, or a transmission of 2% of the signal.

After taking an initial wide range spectrum, choose a section that includes the highest peak and a smaller one next to it. Readjust the scan to cover just this portion. Using the option that allows you to keep successive spectra visible, take Spectrum 1, 2, 3, etc. with the attenuator knob set at 9.8, 9.7, 9.6 . . . turns (out of ten). The nesting heights of the peaks will tell you whether or not the system is behaving in a linear fashion. Continue to go lower on the 10-turn dial setting until you have visual evidence of saturation.

Once you have reached saturation, move back into the linear range. Now you can operate with confidence that the signals you see really are proportional to the amplitude of the sound wave you are studying.