Overview

The AudioStimRecSU 7 script allows you to generate acoustic stimuli via a loudspeaker or vibratory stimuli via a mini-shaker. Since the output of such devices is not flat over their entire frequency range, calibration curves are needed in order to derive the stimulus intensity for each frequency, given an input signal of constant amplitude. The $\texttt{SoundCalSU}_x$ script allows you to measure response intensities over a wide frequency range and generate calibration tables for acoustic and/or vibrational stimulators. The AudioStimRecSU script uses these calibration tables to generate stimuli of known intensity. The $\texttt{SoundCalSU}_x$ script also has facilities for testing the output of CED3505 programmable attenuators.

System Requirements:

Hardware Power 1401 interface, CED 3505 programmable attenuator(s), and equipment for measuring the intensity of sounds and vibrations.

Software Spike 2 version 7.07 or higher**.**

The following files must be saved to a folder named AudStimRec 7 inside the Spike2 root directory.

Scripts SoundCalSU 7.s2s and AudioStimRecSU 7

Sampling configuration SU.s2c.

Sequence SU7.pls

Calibration tables Up to four tables, one each for speaker and mini-shaker on two channels, are generated by the calibration script.

Play waves folder Time views containing waveforms to use as auditory stimuli are stored inside this folder

Performing the calibration

You can calibrate one or two channels of auditory stimulation. The attenuators must be controlled via their USB inputs. Depending on your operating system, you may need drivers to implement the computer end of the Virtual COM ports. You can download an appropriate driver from the FT232BM VP drivers on the FTDI website: www.ftdichip.com.

The input signals to the attenuators are generated by DAC0 or DAC1 with a digitisation rate of 40kHz, as supplied. You can edit a marked section of the SoundCAlSU script to increase the rate to 100 kHz. This will improve the sound fidelity for higher frequency tones at the expense of a reduction in the maximum allowed duration of stimuli and an increase in the time needed to load stimuli into 1401 memory.

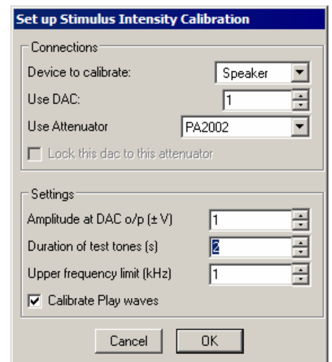
Connect the output of the attenuator to the power amplifier and speaker or mini-shaker. You can monitor the input to the attenuator by connecting the appropriate DAC output to ADC port 1 using a T-piece. The sampling configuration supplied records 5 waveforms each at 25kHz on ports 0 to 4. Thus, there are 4 other channels available for monitoring the output of your sound measuring amplifier or to monitor parameters such as particle velocity. You can easily modify the sampling configuration, for example, to add more channels or increase the sampling rate if required.

Set up the stimulation and recording equipment in the normal way but with the sound/vibration transducer at the position of the experimental animal. Then run the SoundCalSU script. The first time that you run the script, you will be prompted you to add a hotkey (labelled *AudCal*) to the Script bar. This button will give you 'one-click access to the script in future. The script then performs a series of preliminary checks, for example to establish communication with the attenuator(s), check the output voltage range of the 1401 and so on, reporting any errors found. Note that the script should generate tones with a maximum output range of ±5V irrespective of whether the 1401 output range is set to ±5V or ±10V. If all is well, then the script toolbar with seven buttons will appear. You can control the script by clicking on buttons on the toolbar and dialog items with the mouse. Alternatively, it should be possible to control the script using keyboard shortcuts, that is the underlined characters on toolbar button labels, the *tab* key to navigate dialog items and the keyboard arrow keys to increment values or select items from a list.

Sound Cal Toolbar

Set up

Click on Set up to begin. Select the device to calibrate (*Speaker* or *Mini-shaker*) from the drop-down list in the Set up dialog. The other items in the dialog will then enable. Select the DAC that generates the auditory signal and identify the attenuator to which it is connected by its serial number. If you have a single channel system, then there will be only one attenuator in the list and it is entirely up to you whether you drive it via DAC0 or DAC1.



However, if you are calibrating a two channel system then clearly it is important to let the script know which attenuator is connected to which DAC. When you have done this, you can *lock* this pairing by checking a checkbox. If the *Lock* box is checked then changing the DAC item in the dialog will change the attenuator item to match and *vice versa*.

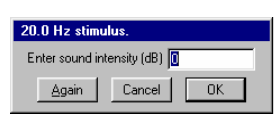
Items in the *Settings* group allow you to specify the amplitude and duration of tone bursts used in the calibration procedure. Choose an amplitude as close to 5V as is compatible with the input of your power amplifier for best possible signal quality. Select a stimulus duration that is long enough for the measuring amplifier to give a steady intensity reading. The upper limit on tone or playwave duration is the amount of an-board memory in the 1401. If Spike2 reports that "*system resources are low*" or "*Not enough memory in your 1401 ...*" then you will have to re-try with a reduced tone duration and/or shorter stored playwaves.

The `SoundCalSU` script generates sinusoidal stimuli of the selected amplitude and duration at fixed frequency intervals. If *Speaker* is selected then the range of frequencies tested is: 20 to100Hz @ 10Hz intervals and 100Hz to 10kHz @ 100Hz intervals. The *Mini-Shaker* calibration also includes a low frequency range extending from: 1 to 20Hz @ 1Hz intervals. You can limit the scope of the calibration in the dialog by setting the highest tone frequency to include. The final tone stimulus will be followed by a burst of low-pass filtered noise. The filter characteristic is flat in the range 0 to 999Hz falling to –3dB at 1.65kHz and > -50dB at 2kHz..

You can also include a series of arbitrary waveforms in the calibration by storing the data files containing the waveforms of interest to the *Playwaves* folder and checking the *Calibrate Playwaves* item in the dialog. See *Appendix 1* for a guide to creating playwaves to use as acoustic stimuli.

Calibrate

Click here to begin the calibration procedure. A new data file will open, sampling starts and the first test tone will be presented and monitored on channel 2. Enter the sound intensity reading of your measuring amplifier in the dialog.



Click Again to re-test the same tone or OK to advance to the next test frequency. The measured intensity will be added to a TextMark at the time of stimulus onset when you click on OK. Clicking on OK also starts the next test tone. Repeat this sequence of actions until all the tones and the noise signal have been tested. If Play waves were selected in the dialog, then these will be presented in the order that the play-wave files were created.

When the sequence of stimuli finishes, the script will save a table of intensity correction factors to disk inside the `AudStimRec 7` folder. You will also have the option to save the data file. If you click on *Cancel* during the calibration, you will have the option to abort the calibration or to update a pre-existing calibration of those frequencies that were tested so far leaving the others unchanged. The filenames of calibration tables are: `Spkcaldata_`*`n`*`.txt` for the speaker calibration and `Mscaldata_`*`n`*`.txt` for the mini-shaker calibration, where. *n* stands for *0* or *1*, the number of the DAC that provided the input signal

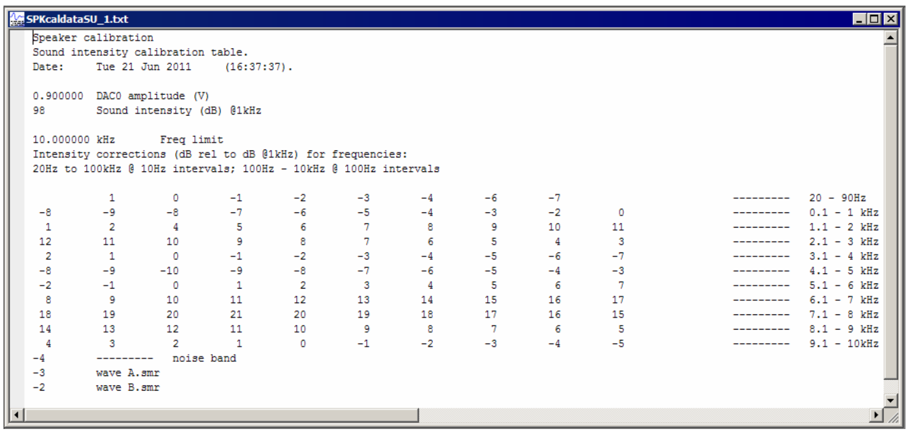
View/Hide/Print Calibration

Click on View Calibration to view the current calibration table on screen. Click the button again (Hide Calibration) to close the table and re-enable the other buttons on the script toolbar. The Print Calibration button is only active while the calibration table is visible on-screen.

The table shows:

- the device that was calibrated
- the date and time of calibration
- signal amplitude at the DAC output and the DAC number
- sound intensity of a 1kHz tone burst of this amplitude
- The highest tone frequency included in the calibration
- The body of the table shows intensity correction factors in dB relative to the value at 1kHz for all tone frequencies tested
- a list of calibration values for filtered noise and all waveforms in the `PlayWaves` folder identified by file name.

Example calibration table



SPKcaldataSU_1.txt

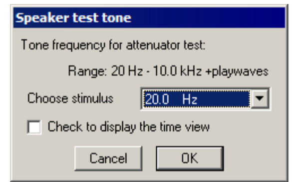
```
Speaker calibration
Sound intensity calibration table.
Date:     Tue 21 Jun 2011     (16:37:37).

0.900000  DAC0 amplitude (V)
98        Sound intensity (dB) @1kHz

10.000000 kHz       Freq limit
Intensity corrections (dB rel to dB @1kHz) for frequencies:
20Hz to 100kHz @ 10Hz intervals; 100Hz - 10kHz @ 100Hz intervals

          1        0       -1       -2       -3       -4       -6       -7                       ---------   20 - 90Hz
 -8      -9       -8       -7       -6       -5       -4       -3       -2        0              ---------   0.1 - 1 kHz
  1       2        4        5        6        7        8        9       10       11              ---------   1.1 - 2 kHz
 12      11       10        9        8        7        6        5        4        3              ---------   2.1 - 3 kHz
  2       1        0       -1       -2       -3       -4       -5       -6       -7              ---------   3.1 - 4 kHz
 -8      -9      -10       -9       -8       -7       -6       -5       -4       -3              ---------   4.1 - 5 kHz
 -2      -1        0        1        2        3        4        5        6        7              ---------   5.1 - 6 kHz
  8       9       10       11       12       13       14       15       16       17              ---------   6.1 - 7 kHz
 18      19       20       21       20       19       18       17       16       15              ---------   7.1 - 8 kHz
 14      13       12       11       10        9        8        7        6        5              ---------   8.1 - 9 kHz
  4       3        2        1        0       -1       -2       -3       -4       -5              ---------   9.1 - 10kHz
-4        ---------   noise band
-3        wave A.smr
-2        wave B.smr
```

Note that tone frequencies higher than the chosen upper limit will be included in the calibration table in order to maintain a consistent format. The intensity correction factors at uncalibrated frequencies will be listed as minus the intensity at 1kHz. You will not be allowed to apply stimuli at intensities that have not been calibrated.

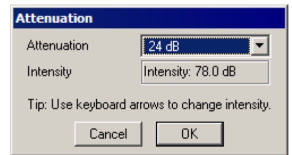
Test attenuator

The test facility generates a test tone via the currently selected device and based on the current calibration. Clicking on the Test Attenuator button displays a toolbar with 4 buttons.

Test Attenuator toolbar



Press Stimulus to select a tone frequency or playwave from the drop-down list in the dialog. Check the box if you wish to view the stimulus monitor on channel 2 (ADC1) and click on OK or press *Enter* to proceed.

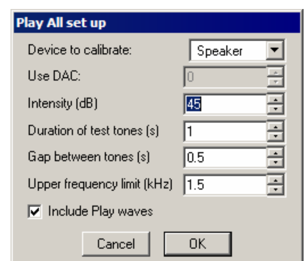


Next, click the Attenuation button and select an attenuation level. The attenuation range and step size available is determined by the attenuator hardware. The dialog shows the resulting intensity by applying the correction factor appropriate for that particular combination of stimulus and output device. Click on OK or press *Enter* to proceed.

Click on the Play button or press *Enter* to play the stimulus. The parameters of the stimulus will be shown on the toolbar and added to a TextMark in the data file at the onset of each stimulus. You can play the stimulus repeatedly by pressing *Enter*. As well as recording the stimulus monitor on channel 2, you can also record outputs from your measuring amplifier on channels 3 - 5. Bear in mind that the default sampling rate of the data file is 25kHz per channel which is not fast enough to display an accurate image of tones at the highest frequencies.

When you have finished testing, click on the Back button to return to the main script toolbar. You will have the option to save the data file if you wish.

Play All



Click here to play all of the internally generated tone stimuli plus filtered noise and playwaves sequentially at a selected intensity (-or as near as we can get given the attenuator step size). The *Set up* dialog allows you to choose the output device, target intensity, the duration of tones the gap between them and the range of stimuli to present. You are not allowed to change the DAC /attenuator used. These are determined in the initial *Setup* dialog.

Stimulus presentation begins when you click on OK. A dialog will display the frequency and intensity of the current tone stimulus or the name of the current playwave. You can interrupt the sequence by holding down the *Enter* key until the presentation stops. You will be prompted to save or discard the resulting data file.

Quit

Click here to close the calibration script. If sampling was in progress, then it will stop and the file will be closed and the display will be restored to its state when you first ran the script.

Appendix 1

Playwave stimuli

The audio calibration and stimulation scripts can use waveforms stored in Spike2 time views as auditory stimuli. The rules for creating these files are:

- The waveform should be recorded. In channel 1 of the time view.
- The sampling rate of the recording should match the time resolution specified in the scripts, -usually 40 kHz. Alternatively, the underlying time resolution of the file should permit the file to be interpolated to this rate (exactly) via a channel process.
- The entire playwave file, from zero to *MaxTime()* will be used as the auditory stimulus.
- Time views containing auditory stimuli must be stored in the *Playwave* folder.

Thus, the simplest way to create new stimulus waveforms is to create a sampling configuration with a single waveform channel sampled at 40 kHz, i.e., with a microseconds per time unit set to *25* on the *Resolution* tab of the sampling configuration. After recording all the stimulus waveforms using this configuration, you can bracket individual stimuli in this file with cursors and save each to its own playwave file using the *Export As…* facility on the Spike2 *File* menu. In order to achieve the highest fidelity during playback the recorded waveforms should fill a large proportion of the input voltage range, typically ±5V.

Custom tone stimuli

You can generate your own tone based stimuli using a Virtual channel. Simply create a new Virtual channel in a time view that was sampled at 40kHz. Set the sample interval manually to 1/40000

You can then use the *WEnv()* and *WSin()* options in the *Expression* box to generate sinusoidal stimuli within a trapezoidal envelope (to avoid distortion and clicks at start and end). Bracket the waveform with cursors and use the Export as… feature to save the waveform to its own data file. Ensure that the *Time shift…* and *Export RealWave channels as Adc channels* checkboxes are checked before exporting.

Example Virtual channel expression

*WEnv(1,2,1,0.1)*WSin(100,0.1)*5.0*

This expression creates an envelope with rise and fall time of 1s, a plateau time of 2s with the start of the envelope aligned to 0.1s. The envelope will contain a sine wave of 100Hz that crosses zero at time 0.1s. The amplitude (zero – peak) of the waveform will be 5V.

Amplitude of PlayWaves

Note that the peak –peak amplitude of the playwaves in the source time view will not correspond to that of the replay. The scripts automatically normalise the waveform prior to playing it so that the peak to peak amplitude matches that of the internally generated tone stimuli used in the calibration. This is determined, in turn by the *Set up* dialog of the calibration process.