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Registered to ISO 9001:2015

APPLICATION NOTE

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02.14.2019

Application Note 105c   ABR Calibration and Test for MP160/150 & Acq*Knowledge* 4.1 or above

This Application Note outlines the precise test equipment calibration required to ensure proper signal-to-noise levels (S/N) and signal time delays for Auditory Brainstem Response (ABR) measurements using an MP160 or MP150 data acquisition system and Acq*Knowledge* 4.1 or above software.

Overview

The calibration procedure can occur in three steps:

a. **Baseline noise measurement:** With the input terminals to the ERS100C amplifier grounded, the acquired input noise voltage signal amplitude should be a flat-baseline trace with an amplitude range of 150 mV p-p or less, assuming 2000 signal pass averages. Use two of JUMP100C cables to short both Vin+ and Vin- to GND. Figure 1 shows a schematic for the noise measurement prior to an ABR test.

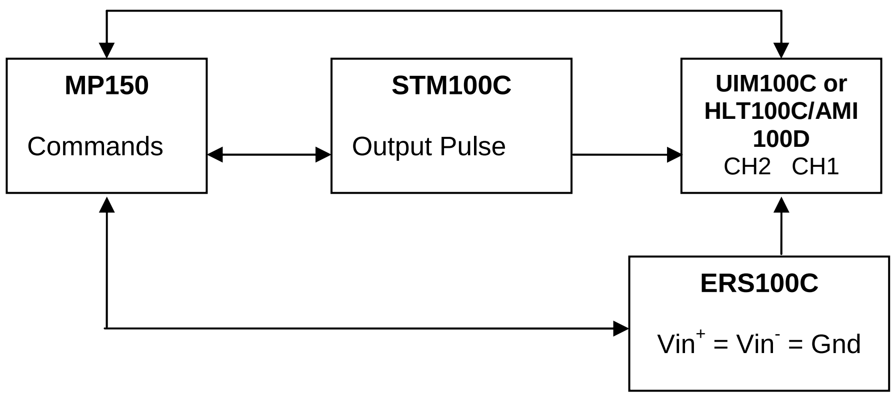

Figure 1 **Schematic of Noise Measurement**

b. **Time delay measurement**: Precise time delay calibration between the trigger of the auditory and the ABR signal is required to establish valid measurements for the Auditory Brainstem Response. This response occurs within 10 msec after audio (click or pip) stimulation.

c. **Artifact rejection**: Spurious signals, primarily from moment artifact, should be filtered during ABR measurements.

Equipment

BIOPAC Part #	Function/Purpose
MP160/MP150	Data acquisition system that performs hardware-based signal averaging.
UIM100C	Signal interface module to the MP150.
HLT100C	Signal interface module to the MP160.
AMI100D	Signal interface module to the MP160 (supports 100D Series Smart Amplifiers)
STM100C	Stimulator module for amplifying stimulus signals.
ERS100C	Evoked Response Amplifier module. This module amplifies the physiological signal sourced from strategically placed surface electrodes.
OUT101	An electrically-generated sound delivery unit (tube phone) calibrated to deliver precise sound pressure levels. A foam-encased tube is placed within the ear canal for proper signal delivery.
Optional Calibration Microphone	To calibrate the OUT101 Tubephone, use an Etymotic ER-7C Probe Microphone—this microphone provides a calibrated output voltage which is a function of applied Sound Pressure Level (SPL). The sensitivity is 50 mV/Pascal (-46 dB re: 1 V/uBar): 0 dB SPL = 0 dBuV. Place the Probe Microphone insert tube in the auditory canal prior to the insertion of the OUT101 foam tip. The OUT101 Tubephone sound delivery tube and the Probe Microphone sound input tube will then be exposed to the same auditory chamber. The SPL will be recorded, via the Probe Microphone, simultaneously with applied auditory stimulus from the OUT101 Tubephone.

TSD108	Physiological Sounds Microphone can record a variety of acoustical signals.
EL503 x 3	Electrodes are 3.5 cm diameter adhesive backing, with 1 cm contact area diameter.
LEAD110	Unshielded signal lead to be connected between the ground scalp electrode and the ERS100C module GND.
LEAD110S x 2	Shielded signal leads to be connected between the scalp electrodes and the ERS100C module (Vin+, Vin- inputs).
GEL1	Conductive gel to be applied between the skin and electrode, if required.
ELPAD	Abrasive pad to gently exfoliate the areas where the scalp electrodes are to be attached.
JUMP100C x 2	Jumpers are used to short amplifier inputs for a noise test.
CBL100	Cable used to connect between STM100C - Output (50 Ω) and UIM100C - CH2.
CBL122*	Cable used to connect to CBL100 between STM100C - Output (50 Ω) and HLT100C/AMI100D - CH2.

*__IMPORTANT:__ CBL122 is unisolated and must not be used with external equipment when a human subject is connected to the MP System unless the external equipment has its own built-in isolation.

Setup

Hardware Setup

Figure 2 shows the acceptable hardware arrangement of the signal acquisition modules for the noise measurement, for verifying system performance prior to performing an ABR.

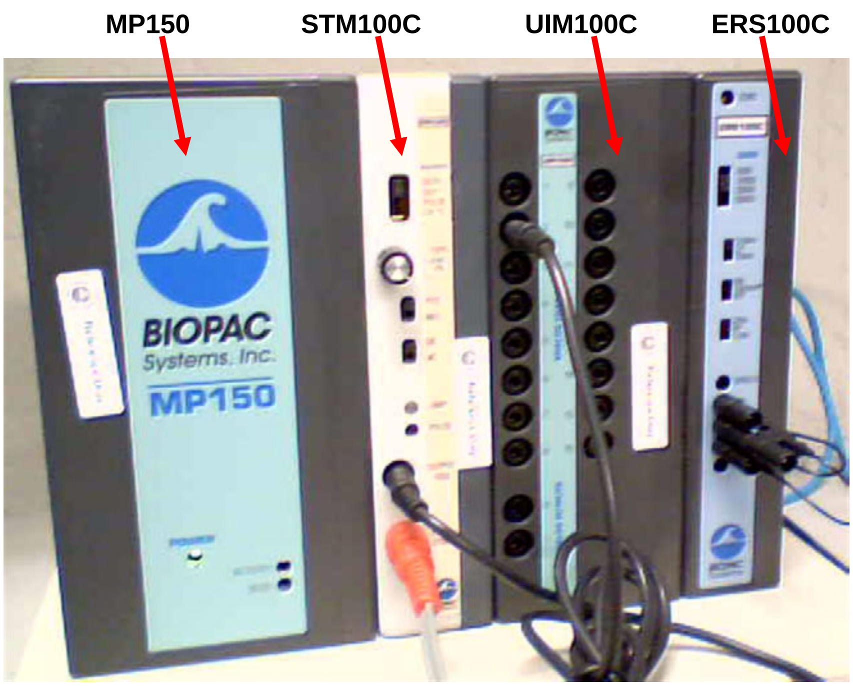

Figure 2 **Test Setup for Noise Measurements**

MP150 ⇔ STM100C ⇔ UIM100C ⇔ ERS100C

The module placements are shown in the photograph marked with red arrows.

The module switch/dial settings and connections are listed in Table 1.

Module	Switch/Dial	Value/Setting
MP150/MP160	None	None
STM100C	Source	OUT0 - will receive commands from MP160/150 Stimulator window
	Level	Variable depending on test requirements
	Pos/Neg	Pos
	DC/AC	DC
	Output (50 Ω)	Connect to CH2 of UIM or HLT/AMI module using cable (stimulus ref.)

	Ext STIM	Connect the Ext STIM output to the OUT101 tubephone using the connector built into the OUT101 (Not required for noise measurement). • If using the optional calibration microphone: place the Etymotic ER-7C Probe Microphone insert tube in the auditory canal prior to the insertion of the OUT101 foam tip.
UIM100C	Analog CH2	MP150: Receives signal from STM100C Output (50 Ω) (stimulus reference)
HLT100C or AMI100D	Analog CH2	MP160: Receives signal from STM100C Output (50 Ω) (stimulus reference)
ERS100C	Gain	50000
	LP	3 kHz
	100 Hz HP	On
	HP	20 Hz (but 100 Hz HP is dominant)
	Vin+, Vin-, Gnd	All connected using two JUMP100C connectors
	Channel Setting	(slider switch atop the module) CH1 to the UIM100C

***Table 1* Module settings**

Software Setup

Software setup in Acq*Knowledge* begins with a new graph file and invoking the “View by Channels” window for channel activation as shown in Figure 3 (MP160/150 > Set Up Data Acquisition > Channels). Channel A1 will accept signals from the ERS100C module. Channel A2 will accept signals from the STM100C module.

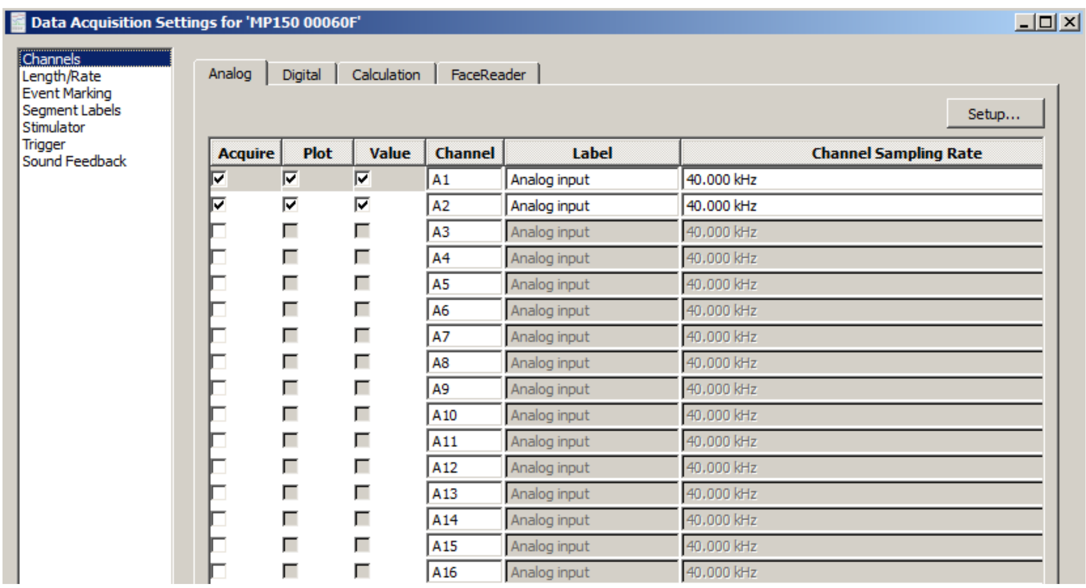

***Figure 3* Channel setup window invoked via MP160/150 menu > Set Up Data Acquisition > Channels…**

The next step is to configure the acquisition parameters (Figure 4). Click "Setup" in the “Length/Rate” window to generate the Averaging options (see Figure 5 on following page).

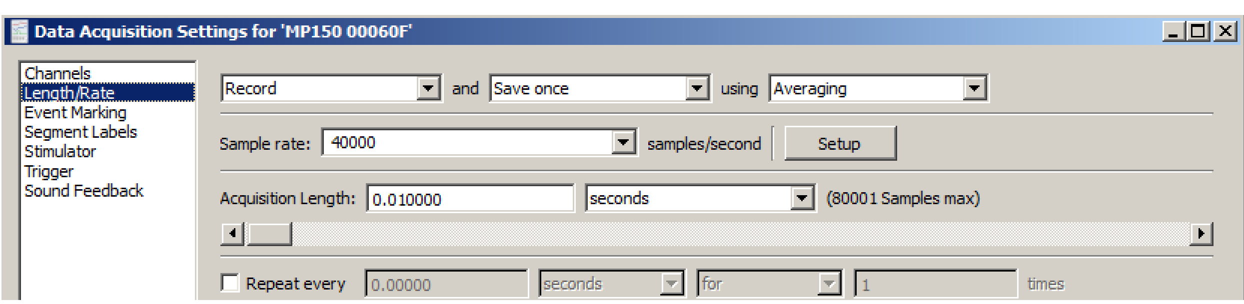

***Figure 4* Acquisition parameters**

Select the channel(s) the averaging pass will be applied to in the "Enable" column.

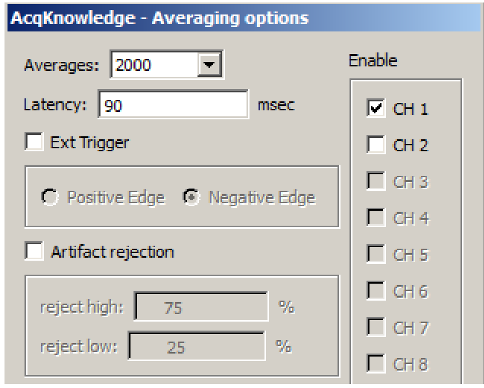

Figure 5 **parameter setup for signal Averaging**

The MP160/150 stimulator is configured to provide the output signal to the STIM100C. Figure 6 shows the Stimulator setup window, which is used to create the stimulus pulse that is routed to the OUT101 tubephone. Figure 7 shows the manual attenuation control window for the STM100C. Here, the user can set the attenuation of the output signal to appropriate levels independently of the manual attenuation (vial rotary knob) settings on the STM100C.

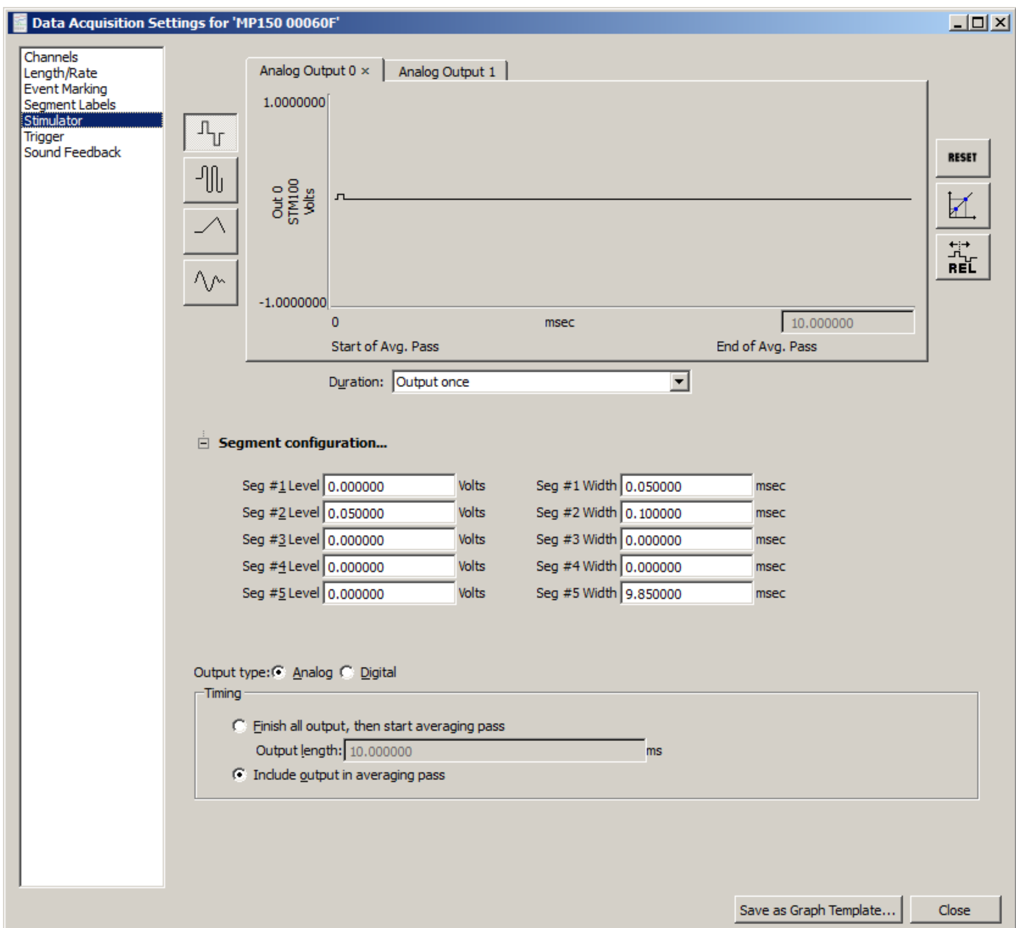

Figure 6 **Stimulator setup window**

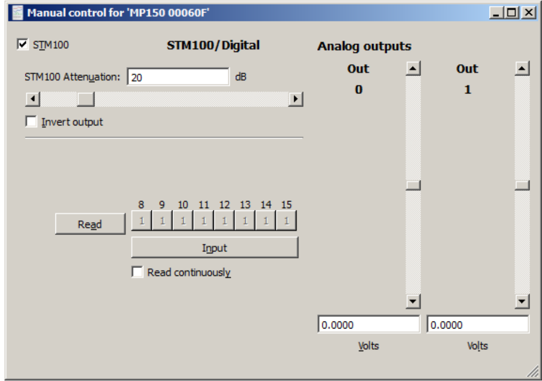

Figure 7 **Manual attenuation control for the STM100C module, dB value shown is the sound level reduction from the case of maximum SPL (0 dB)**

After hardware and software configurations are complete, click the "Start" button to observe signal averaging being applied to graph Channel A1 (ERS100C). The trace on Channel A2 is the stimulus signal (STM100C,) which is not a factor in this particular examination but will be for later tests. Figure 8 shows the end of the noise averaging run.

- **Note that a voltage amplitude measurement shows that the noise is roughly 92 nV p-p. This value is less then the acceptable maximum of 150 nV p-p.**

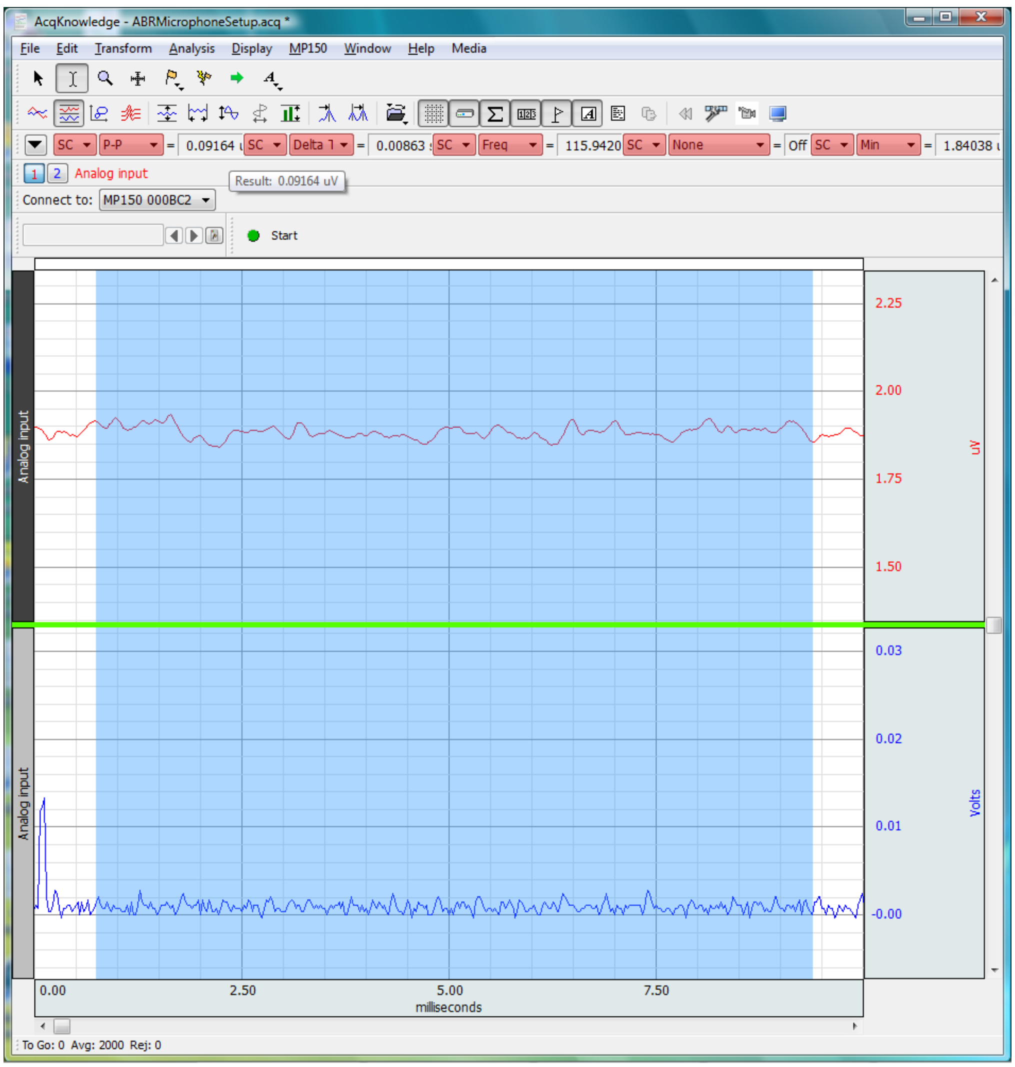

Figure 8 **Waveform traces after averaging. CH1 is the noise signal when inputs Vin+ and Vin- of the ERS100C module are grounded. CH2 is the stimulus signal.**

Timing Measurements

Using OUT101 to invoke an acoustic stimulus requires calibration of the delay between the onset of the electrical stimulus from STM100C to the onset of the acoustic response (generation of the acoustic click and the reception of the click by the TSD108 transducer). This determines how much of the acquired data from a subject must be ignored due to this delay. Figure 9 shows the equipment configuration for the measurement.

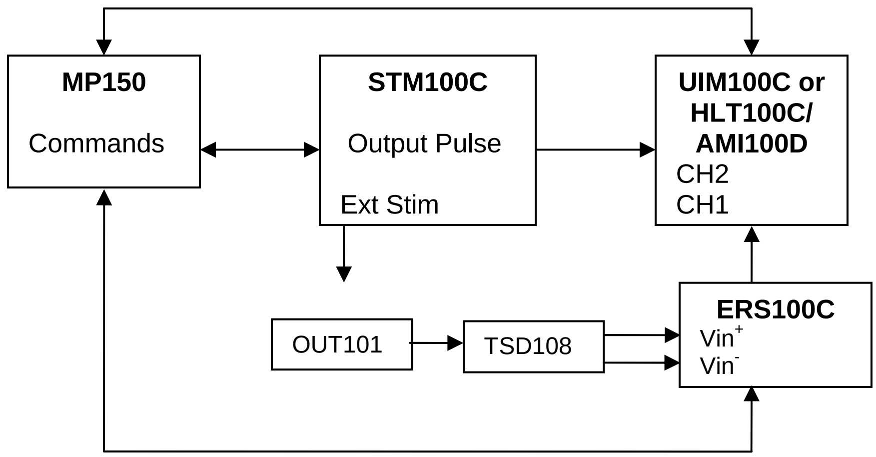

Figure 9 **Setup for timing measurements**

Figure 10 shows a photograph of the actual time delay measurement configuration. Note that the OUT101 tubephone is taped perpendicularly to the front face of the physiologic microphone for effective acoustic energy transference.

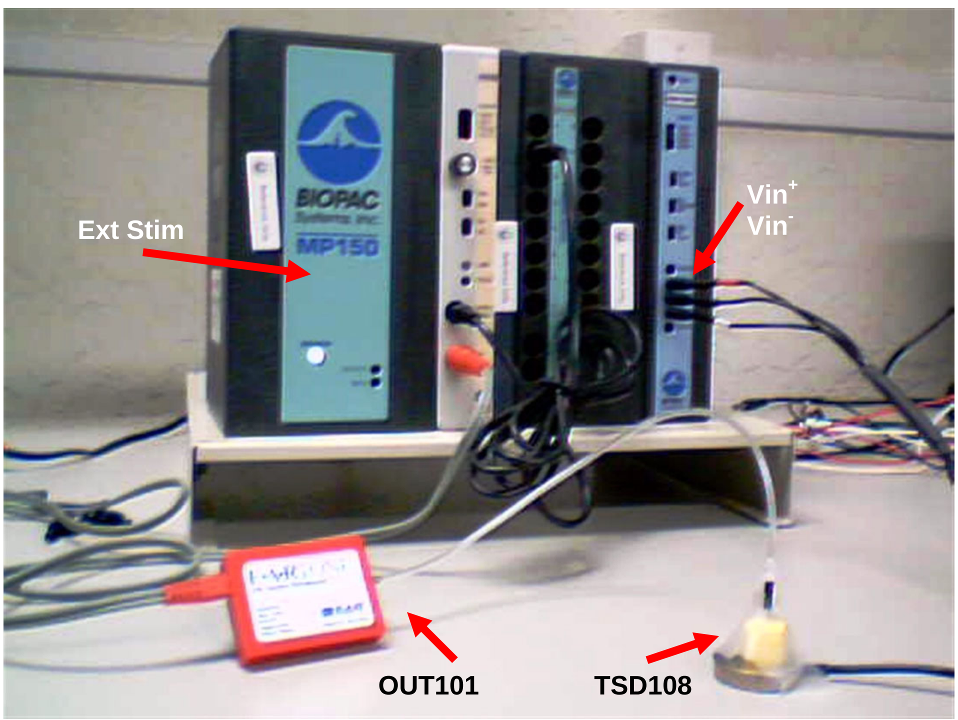

Figure 10 **Photo of the arrangement between OUT101 (acoustic stimulus) and TSD108 (acoustic receiver)**

Keeping the software configurations the same as in section 3.0, the following waveform traces should be seen during the averaging passes. Note that the "ringing" signature of the trace in CH1 is the microphone response upon receiving an acoustic impulse. Taking a measurement from the electrical stimulus onset in CH2 to the beginning of the acoustic impulse response, the user should find that the delay should be in the order of 1 msec. This 1ms delay is due to the tubing length associated with the OUT101 tubephone. This delay is by design in order to time isolate the tubephone electrical activation impulse energy from the delivered sound energy. Accordingly, any tubephone electrical activation signal will occur prior to the point of auditory stimulus (by 1ms) and will not distort any ABR measurement during the averaging process. Figure 11 on the following page shows the graph output.

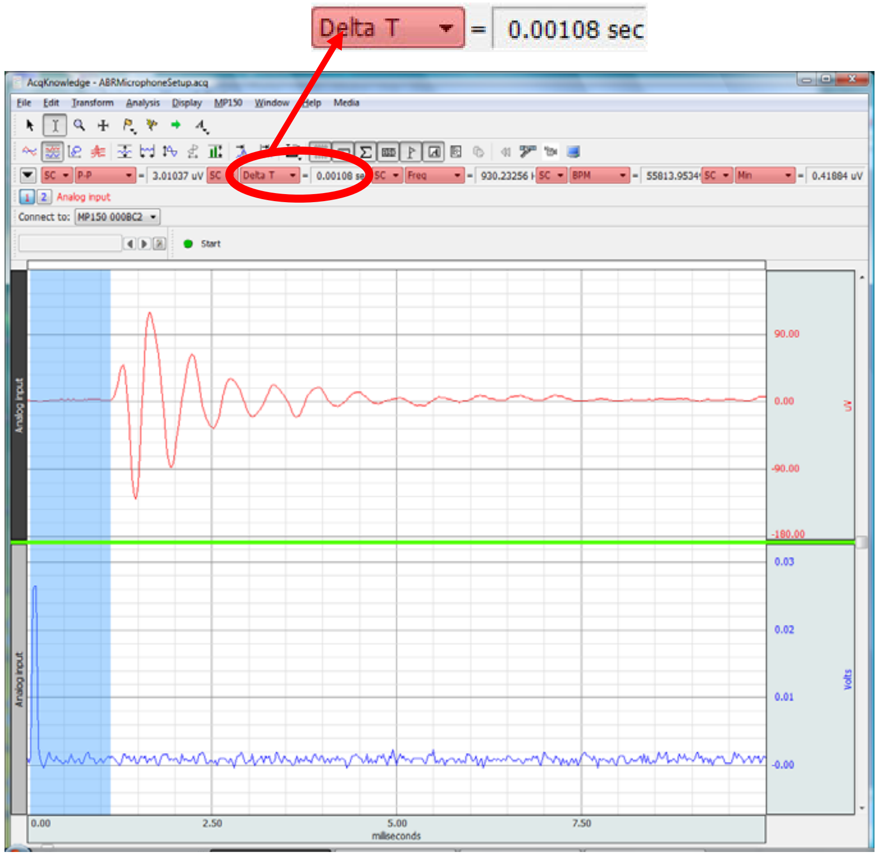

Figure 11 **Time delay between the onset of the electrical stimulus to the onset of the acoustic response. This measurement verifies that the delay from electrical stimulus to delivered auditory "click" is approximately 1 ms.**

Artifact rejection

During an acquisition, extreme levels of artifact may be present for one reason or another. Check "Artifact rejection" in the Averaging setup window to determine which signal levels constitute artifact. This will enable the MP160/150 System to reject these trials.

When artifact rejection is enabled, the MP160/150 System will (1) ignore any trials containing signals that exceed the artifact rejection thresholds, (2) keep track of how many trials have been rejected, and (3) add that number of trials to the total number of trials to be acquired. Trials that were rejected due to the presence of artifact can then be "re-tried." To set these parameters, establish a high threshold and a low threshold; both thresholds refer to the scale limits (nominally ±10 Volts).

If the high and low artifact rejection thresholds are set to 80% and 30% (respectively), the MP System will reject any trial where the signal exceeds +8 Volts or -3 Volts. When the channel scaling feature is used to change the range of Map (Scale) values to something other than ±10 Volts, the artifact rejection formula for symmetrical limits is:

$$y = ((2 \cdot PV)/100) \cdot x - PV \quad \textit{where } y = \text{voltage threshold}$$

where: *PV = Peak Value*
x = percent threshold (whole number)

If non-symmetrical limits are used, the following equation is used:

$$y = ((V1-V2)/100) \cdot x + V2$$

where: y = voltage threshold
V1 = Higher Peak Value
V2 = Lower Peak Value
x = percent threshold (whole number)

TSD108 Response Curve and Audible Levels

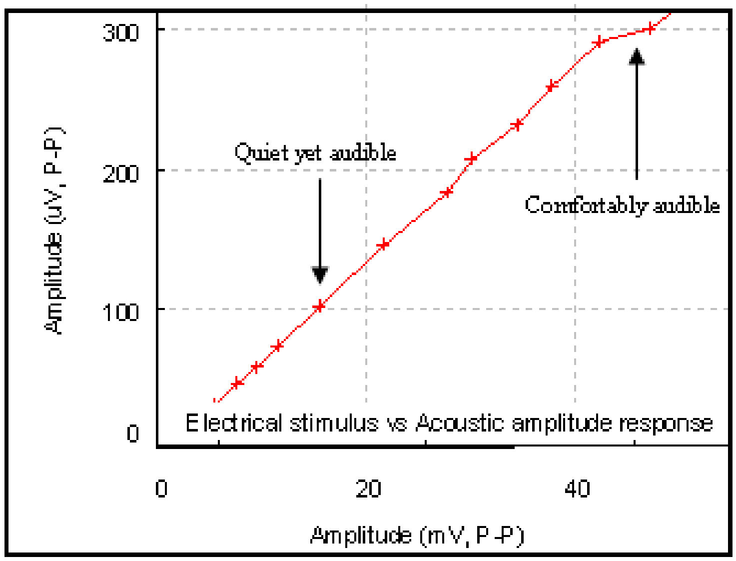

Figure 12 **Transducer profile of TSD108. Dial on STM100C set so marker is pointing downward (south).**

The OUT101 Tubephone will provide acoustic stimulus to a subject with the option of varying the levels of output energy. Figure 10 shows a rough profile of electrical trigger level versus acoustic amplitude output for the OUT101. The STM100C provides electrical triggering to the OUT101 device through the EXT Stim output jack. The microphone used in the time measurement task (Sec. 4.0) is also used to create this profile. The output level dial on the STM100C is set so that the rotary dial marker is pointing downward (south). The electrical signal to the OUT101 acoustic source was adjusted in the Manual Control Window, using the dB attenuation slider with a starting value of 40 dB.

Figure 12 shows Electrical stimulus level input to OUT101 Tubephone versus TSD108 recorded signal level response.

Channel A2 indicates the input electrical stimulus and Channel A1 indicates the acoustic response. Peak to peak measurements were used in acquiring the data points. Annotated on the graph is a person's nominal perception of the "loudness" and comfort level of the acoustic signal.

At zero dB attenuation on the Manual Control attenuation slider, and with the STM100C Level dial set so the marker is pointing "west" (50% amplitude), the perceived sound intensity is loud, yet tolerable. If the the STM100C dial marker is pointed "north" (75% amplitude), the sound intensity can become uncomfortable after a few minutes.

ABR Test

Figure 13 **Sensor placements. On Subject for ABR measurements**

After calibration of noise and time delay, the user can proceed to the ABR test phase.

The test setup would be the same as that in Sec 4.0, but with body electrodes (EL503, LEAD110, LEAD110S) replacing the physiologic microphone (TSD108). OUT101 acoustic source is placed within the subject's right ear and connected to Ext Stim of the STM100C module, ground electrode (LEAD110) is placed on the forehead and connected to the GND of the ERS100C module, the ipsilateral earlobe electrode is placed on the right earlobe (LEAD110S, white lead) and connected to Vin- of the ERS100C module, the contralateral earlobe electrode is placed on the top of the head and held with an ace bandage and connected to Vin+ of the ERS100C module. Figure 11 shows the placement of the electrodes on a subject for ABR measurements.

Enable the “Artifact rejection” checkbox in the Averaging setup and set the values to 60% and 40% to filter outliers. (See Figure 5 for example.)

Before plugging the body electrodes into the ERS100C module, a test for skin-electrode impedance should be performed. Contact preparation by using electrode gels and ace bandages help reduce the skin-electrode impedance levels to values under 5 kohms, however 5 to 10 kohms is acceptable. Figure 14 shows BIOPAC’s EL-CHECK meter that can be used to measure skin-electrode impedance values.

Using the EL-CHECK meter, plug the connected body electrode leads into the top receptacles and read the indicated impedance value. If the impedance values read high (>10 kohm) reseat the electrode to that part of the body, striving for improved contact, namely a reading at 5 kohm - 10 kohm or lower.

During recording, the waveforms generated by the body electrodes will change as averaging progresses. Near the end, the waveforms will stabilize. Figure 15 shows the result of one trial.

Figure 14 Electrode meter

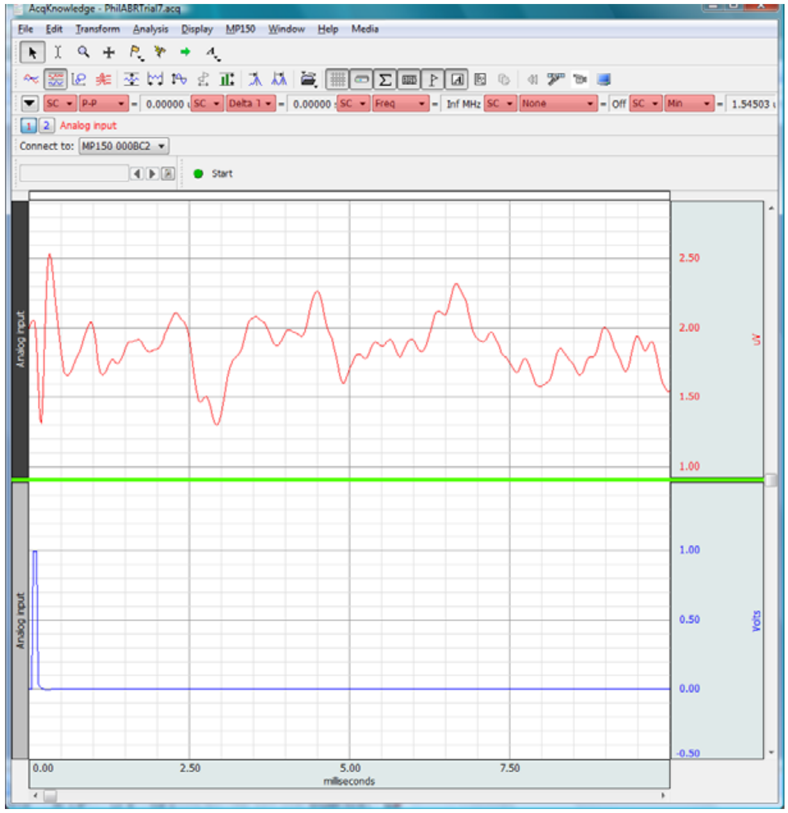

Figure 15 An ABR trial measurement with 2000 averaging passes with 40,000 samples per second acquisition rate and 10 msec acquisition time

A smoothing transform can be applied to ABR signals to further reduce high frequency components. The Jewett Waves III, IV, V can be readily seen in Figure 17, where two consecutive ABR averaging trials are shown overlapped.

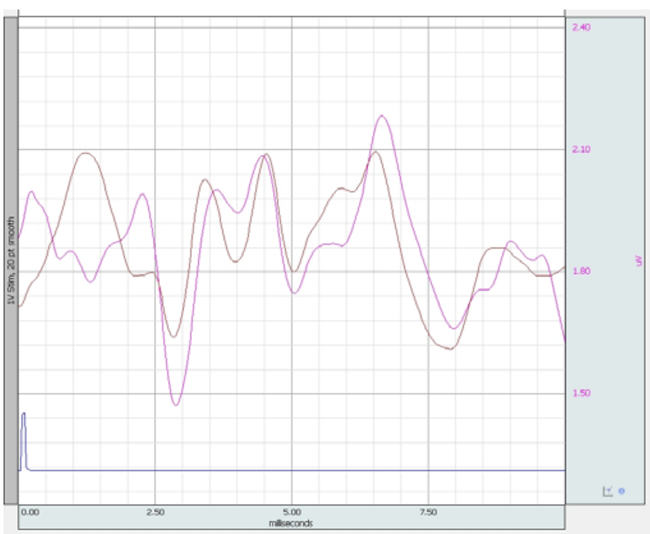

Figure 17 ABR trial combination
1 V Stim, 20 pt Smoothing (overlapped mode)

Frequently Asked Questions

How are waveforms overlapped in the display?

Use the overlap function in the Acq*Knowledge* software. This is a toolbar button in the software window; see the software manual under the Help menu for details.

Is baseline noise measurement something that should be done for each subject before the test?

We recommend to perform this test, before a series of trials, to verify setup. This is an input channel verification test. If the recorded noise is under a specific amount when doing this test, confidence in the validity of the measurement results can be assumed, given the equipment used. This test verifies amplifier performance and data acquisition system performance.

Should the timing measurement be performed for each MP system? If so, is a TSD108 also required?

Timing measurement is not as important as the baseline noise previous test, simply because there is less that can go wrong. However, verification of timing is paramount, in order to perform accurate ABR. This test is excellent in the sense that it provide unambiguous results regarding stimulus presentation, stimulus amplifier (STM100C) acoustic delay, stimulus transducer (OUT101) performance. The baseline noise test verifies input channel performance and the timing test verifies the output channel performance. Ideally, a simple mechanical jig would be developed which would position the OUT101 Tubephone output a fixed and known distance away from the TSD108. Benchmark response tests could be performed and then kept on file to look for changes (if any) as the system is used over a period of time.

For the artifact rejection, the manual says, “the MP160/150 System will ignore any trials that contain signals exceeding the artifact rejection thresholds, keep track of how many trials have been rejected, and add that number of trials to the total number of trials to be acquired.” Regarding artifact rejection, after the test is completed, do I have to go back and do a couple minutes to make up for the missed trials? Or does the system automatically do that? If the latter, how exactly does this work?

Artifact rejection, when turned on, simply throws out data passes that exceed a specified threshold. Waves that exceed the threshold, and get rejected, are typically ones where the subject moves during the test. Subject movement nearly always causes artifact, especially if electrode impedances are not low enough or electrodes are not well attached to skin surface. If the number of rejected trials exceeds 10% of the total number of trials, it's probably best to simply repeat the full recording. To obtain a proper artifact rejection setting, perform an averaging test using one average and perform this test about 10 times. Take a look at each trace and examine the highest and lowest value in each trace. If the reject thresholds are set about 25% above and below the highest and lowest values recorded over 10 (single averaging pass trials) then that should be about right. Once artifact rejection is set, for a certain class of ABR measurement, in terms of amplifier gain, amplifier filter settings, type of subject, electrode attachment points, electrode lead type, gel used and local environment, then it's unlikely that software rejection thresholds would need to be changed once established.