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Acoustics

- The scientific study of sound, especially of its generation, transmission, and reception
- · Common units of measurement in Acoustics

Mass (M)	м	kilogram (kg)	gram (g)	1 kg = 10 ³ g
Time (t)	t	second (s)	5	
Area (A)	A	m²	cm ²	1 m ² = 10 ⁴ cm ²
Displacement (d)	d	meter (m)	centimeter (cm)	1 m = 10 ² cm
Velocity (v)	v = d/t	m/s	cm/s	1 m/s = 10 ² cm/s
Acceleration (a)	a = v/t	m/s²	cm/s ²	1 m/s ² = 10 ² cm/s ²
Force (F)	F = Ma = Mv/t	kg · m/s² newton (N)	g - cm/s² dyne	1 N = 10 ⁵ dyne
Pressure (p)	p = F/A	N/m2 pascal (Pa)	dyne/cm² microbar (ubar)	2 × 10 ⁻⁵ N/m ² or
		,	0	20 μFa (reference volue)
				2 × 10 ⁻⁴ dyne/cm ²
				orµbar
				(reference value)
Work (W)	W = Fd	N · m Ioule	dyne - cm	1 J = 10 ⁷ erg
Power (P)	P = W/t	joule/s	erg	1 w = 1 l/s
, onci (,)	= Fd/t = Fv	watt (w)	watt (w)	= 10 ⁷ erg/s
intensity (I)	I = P/A	w/m²	w/cm²	10 ⁻¹² w/m ²
				(reference value)
				10-16 w/cm ²
				(reference value)





- Sound: air molecule pressure that sets the eardrum in motion and leads to the perception of sound.
- Noise: sounds that are unpleasant and unwanted.





Acoustics

Requirements:





Waveforms and frequencies

- Three examples of the relationship between the waveform of a signal in the time domain compared to its spectrum in the frequency domain.
- Most natural sound signals are complex in shape. The signal is composed of a number of discrete frequencies at individual levels present simultaneously.
- The number of discrete frequencies displayed is a function of the accuracy of the frequency analysis.





Intensity: Range of Sound pressure

The human audible sound pressure variations range from about 20 µPa to 100 Pa.

- 20 μPa (0dB) is the quietest sound that can be heard by an average person, therefore is called the <u>threshold of hearing</u>.
- A sound pressure of approximately 100 Pa (120dB) is so loud that it causes pain, therefore is called the <u>threshold of pain</u>.
- The ratio between these two extremes is more than a million to 1.







- The direct application of linear scales, in Pa, to the measurement of sound pressure would therefore lead to the use of enormous and unwieldy numbers.
- Additionally, the ear responds not linearly but logarithmically to stimulus.
- Therefore, it has been found more practical to express acoustic parameters as a logarithmic ratio of the measured value to a reference value
 - a logarithmic ratio called a decibel or just dB.

Systems Neuroscience & Martin Reurotechnology Unit	Decibels
Decibel Sound Pre	essure Level (SPL): $dB(SPL) = 20Log\left(\frac{P_0}{P_{ref}}\right)$
Linear scale: Unwiedly numbers	Log. scale: Reference sound pressure: 20 [µPa] Managable scale
Sound Pressure, p [Pa] 100 10 10 1 0.1 0.01 0.001 0.000 1	Sound Pressure Level, 140 [dB] 120 100 80 60 40 20 0 0



An increase of 3 dB in pressure (corresponding to 1.4 times) is just perceptible.

A change of 10 dB or 3.16 times is perceived as twice as loud. There is no linear relationship between the loudness level in dB and the perception by man.

		140	-	Threshold of pain
		130		
Change in Cound	Change in	120		Threshold of discomfor
Level (dB)	Perceived Loudness	110		
		100		Pneumatic breaker
3	Just perceptible	90		
5	Noticeable difference	80	·	Busy traffic
		70		
10	Twice (or 1/2) as loud	60		Conversation
15	Large change	50		
20	Four times (or 1/4) as loud	40		Living room
20		30		
		20		Quiet countryside
		10		
		0	· · · · · ·	Threshold of hearing



Frequency: Auditory Range

 Young human beings can detect sounds ranging from 20 to 20000 Hz





The wavelength depend on:

- 1. the speed of sound
- 2. the frequency or period of interest





Diffraction of Sound

- Objects placed in a sound field may cause diffraction.
- But the size of the obstruction should be compared to the wavelength of the sound field to estimate the amount of diffraction.
- If the obstruction is smaller than the wavelength, the obstruction is negligible.
- If the obstruction is larger than the wavelength, the effect is noticeable as a shadowing effect







- Diffusion occurs when sound passes through holes in e.g. a wall.
- If the holes are small compared to the wavelength of the sound, the sound passing will re-radiate in an omnidirectional pattern similar to the original sound source.
- When the hole has larger dimensions than the wavelength of the sound, the sound will pass through with negligible disturbance.





Reflection of Sound

- When sound hits obstructions large in size compared to it's wavelength, reflections take place.
- If the obstruction has very little absorption, all the reflected sound will have equal energy compared to the incoming sound.







- Near field: Locations close to the sound source between the source and the far field. The near field is typically characterized by large sound pressure level variations with small changes in measurement position from the source.
- Far field: That part of the sound field in which sound pressure decreases inversely with distance from the source. This corresponds to a reduction of approximately 6 dB in level for each doubling distance.
- Free field: An environment in which a sound wave may propagate in all directions without obstructions or reflections. Anechoic rooms can produce such an environment under controlled conditions.
- Diffuse field: An environment in which the sound pressure level is the same at all locations and the flow of sound energy is equally probable in all directions.





Sound Fields

- Reverberation room: a room with hard reflecting surfaces, all the energy will be reflected and a so-called diffuse field with sound energy uniformly distributed throughout the room is set up.
- Anechoic room : In a room with highly absorbent surfaces all the energy will be absorbed by the surfaces and the noise energy in the room will spread away from the source as if the source was in a free field.





Psychoacoustics



Definitions

 Psychoacoustics: deals with perception of sound. How we perceive physical attributes.







- Pitch
- Loudness
- Timbre



- Pitch: Term used to describe the subjective impressions of the "highness" or "lowness" of a sound.
- ASA: "that attribute of auditory sensation in terms of which sounds may be ordered on a musical scale"
- Pitch relates to frequency.
- Unit: mels.

Pitch vs Frequency

• Frequency: number of cycles per unit of time.

$$f = \frac{\# cycles}{time}$$

Units: Hz







- Subjective aspect of pitch can be measured using a unit called mel.
- 1000 mels is the pitch of 1kHZ tone at 40dB SL.
- Frequencies can be adjusted:
 - Twice as high: 2000 mels
 - Half as high: 500 mels
- 'bass' = low pitch
- 'treble' = high pitch





The ear is more sensitive to F_0 differences in the low frequencies than the higher frequencies. This means that:

300 vs. 350 ≠ 3000 vs. 3050

That is, the difference in *perceived pitch* (not F_0) between 300 and 350 Hz is **NOT** the same as the difference in pitch between 3000 and 3050 Hz, even though the *physical* differences in F_0 are the same.





- The pitch of complex tones depends to a large extent upon the perception of the harmonics in the sound as opposed to the place of maximal displacement along the cochlea.
- It would seem that it is the harmonic structure that determines our perception of pitch, rather than simply the frequency of the lowest harmonic that is physically present in the signal.
- It is as though our brains calculate the difference in Hertz from one harmonic to the next to decide what the real "pitch" of the tone is.





Pitch of complex tones: Missing Fundamental Frequency

- The pitch the ear and brain "hear" is in each case not based on the lowest frequency component.
- It can be "hear" rather the tone as having the pitch of the original fundamental frequency, *even when it is not physically present in the signal*!





Loudness vs Intensity

- Loudness: the subjective impression of the power of a sound.
- Term to relate intensity of a sound.
- Intensity: amount of sound energy per unit of area. Intensity= Power Area



"The noise in the Room is 60dB loud"



Loudness vs Intensity

- Sounds of different frequencies are not equally audible.
- Loudness depends on frequency: Equal Loudness Contour curves
- The unit for loudness level is phon. Sone refers to the comparison of the loudness of a 1000Hz tone at different intensities.
 40 phons curve: All the





"that atribute of auditory sensation in terms of which a listener can judge that two sounds similarly presented and having the same loudness and pitch are dissimilar"

 Different voices and instruments are recognised as having a different quality when making the same note. This individual timbre results because different instruments produce different mixtures of overtones that accompany the fundamental.





• Timbre differences are partly related to differences in spectrum envelope- differences in the relative amplitudes of the individual harmonics.





Source Localization



Source Localization

- Source localization: Ability to determine the specific location of a sound source.
- Binaural hearing: general term used to describe the nature and effects of listening with 2 ears, instead of one.
- Binaural fusion:separated signals received by the 2 ears are perceived as a single fused auditory image.





HRTF

Head-Related Transfer Function (HRTF) or Anatomical Transfer Function (ATF).

- HRTF: reflects accumulated effects of factors that influence the sound traveling from the sound source to the eardrum, including acoustical shadows, reflections, and difraction due to the head and body, as well as the ear canal resonance.
- HRTF is an individual function for every person and every sound source location.

Azimuth effect: HRTF are different for the R ear when the loudspeaker is at an azimuth of 45° to the right compared with when is 45° to the left (45°vs 315°), morphology of curve also changes.





Directional Filter: Directional earprint of the Pinna.

- Useful to determine elevation and front-back direction of a sound.
- The Pinna effectively adds a directional earprint to the sound spectrum.
- The Pinna-induced amplification pattern is direction dependent as the underlying acoustic depend on the angle of incidence of the sound waves to the Pinna. (specially for frequencies above 3-4 KHz)
- Spectral cues are of major importance for the perception of sound elevation and frontback angle.

Pinna Transfer Functions



Figure 4 Directional spectral *eaprints* of the author's right pinna. Top: Amplitude of the pinna transfer in the frequency range 1-20 kHz for sound elevations [-40, -35, ..., +55] deg in the vertical median plane (curve thickness increases with elevation). Note that the pinna can cause a substantial amount of amplification and attenuation, and that these effects depend on sound direction for frequencies above 4 kHz. Bottom: The same data shown in the elevation-frequency plane. Amplitude now increases with intensity. Note that peaks and valleys vary systematically with elevation at the high frequencies.



Head Shadows

 Head shadows: occur for frequencies that can be obstructed by the head, i.e., wavelenghts that are shorter with the size of the head, specially f over 1.5kHz.







ITD

- Difference between the times sounds reach the 2 ears.
- Robust and powerful for horizontal localization.
- When distance to each ear is the same, there are no differences in time.
- When the source is to the side of the observer, the times will differ.
- ITD: It constitutes a localization cue for low frequencies (up to about 1.5 KHz). The auditory system apparently chooses to ignore this cue for higher frequencies.





Interaural Difference (Binaural) Cues:

• Difference in sound pressure level reaching the two ears.

ILD

- Second important cue for horizontal localization.
- Reduction in sound level occurs for high frequency sounds for the far ear.
- The head casts an acoustic shadow.
- ILD: Intensity differences serve as a localization cue for high frequencies (above approximately 1.5-2 KHz).



Interaural Difference (Binaural) Cues:

- Duplex Theory: the auditory system derives horizontal sound directions from time differences for low frequencies, and from intensity differences for the high frequencies. They complement each other.
- Interaural cues are not sufficient for elevation or frontal back sound source orientations.





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Decibels Classification



Decibel Sound Pressure Level (SPL): dB (SPL)= 20 log (Po/Pref)

- Reference sound pressure: amount of pressure against the eardrum, caused by air molecules when sound is present, that vibrates the eardrum and can just be detected by a normal human ear.
- Reference sound pressure: . Dyn/cm2, 20 micropascals [μPa] RMS, 2*10^-5





dB (HL)

Decibel Hearing Level (HL)

- Standarized (ISO 226:2003)
- Referenced to normal hearing standards or "audiometric zero"
- Curves obtained from large scale studies of hearing.

Equal-loudness-contour





dB (HL)

Equivalence between Decibel Hearing Level (HL) and dB (SPL)

Frequency [Hz]	dB SPL	dB HL
250	12	0
500	> 5	0
1000	2	0
2000	- 2	0
4000	- 5	0
8000	13	0





Decibel Hearing Level (nHL)

- Referenced to normal hearing level for a specific AER stimulus.
- Behavioral threshold levels are determined for a relatively small group of people (with normal hearing).
- Idea: find out the smallest intensity level on the dial or the equipment screen that can be just barely detected by each subject.





Comparison between diferent types of dB for Clicks



Figure 1 – Temporal characteristics of an electric reference pulse

TABLE 7-1. COMPARISON OF CLICK-STIMULUS INTENSITY LEVELS IN dB SPL CORRESPONDING TO 0 dB nHL AMONG SELECTED STUDIES

Study	Year	Rate (/sec)	peak SPL (dB)
Burkard & Hecox	1983	27	40.0
Campbell et al.	1981	10	40.0
Hood & Berlin	1986	27.7	36.0
Ozdamar & Stein	1981	20	nit, general Quitane
Selters & Brackmann	1977	20	38.0
Stapells, Picton, & Smith	1982	10	36.4



Decibel Sensation Level (SL)

- Referenced to sensation level.
- in other words, the patient's behavioral thershold is determined and then referred to as 0 dB SL.
- Subsequent intensity levels are described with reference to this level.
- A normal-hearing subject may actually have 0 dB HL=0 dB SL.
- Lack of standarization or comparability among facilities or laboratories.



dB Classification

Measured by <u>physical</u> measures of sound:

Their references are physical measures of sound, made with a calibrated measuring devices (dB SPL, dB peSPL).

<u>Physiologic</u> measures of sound: (biological reference level)

Their reference is either a group of normal hearers' average thershold for the stimulus (dB HL, dB nHL) or an individual patient's threshold for the stimulus (dB SL).





- Calibration is the adjustment of equipment, or the application of a defined correction factor when using the equipment.
- Relevance?
 - Amplifier
 - Stimulus: rate, duration, polarity, intensity.
 - Accuracy



....Reproducibility

Sometimes, like for Audiometers, periodic calibration are necessary to ensure precision and predictability in stimulus presentation and response acquisition.



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Calibration procedure

BRITSCH BRITISH STANDARD	BS EN 60645-3-2007
C C C C C C C C C C C C C C C C C C C	
Electroacoustics — Audiometric equipment —	 Standards: IEC 60645-series –equipment for audiometry (stimulus def.)
Part 3: Test signals of short duration	 ISO 389-series - reference 0 for calibration of equipment for audiometry
	 ISO 8253-series - audiometric methods
U	 International standards concerning electroacoustic measurements: IEC 60318- coriac
The European Blandard EN 60645-3:2007 has the status of a British Standard	International standards connecting bearing side 60118
NEW 12.140, 17.140.30	series
NO COPYING WITHOUT BSI PERMISSION EXCEPT AN PERMITTED BY COPYRIGHT LAW	British Standards



IEC 60645-3 (2007)

EN 60645-3:2007

-4-

INTRODUCTION

Developments in the field of hearing measurements for diagnostic, hearing conservation and rehabilitation purposes have resulted in the availability of a wide range of audiometers. In addition it is possible to consider the audiometer in terms of a set of functional units that can be specified independently. By specifying these functional units it is then possible to specify the performance of other audiometric equipment that uses these units. IEC 60645, *Electroacoustics – Audiometric equipment*, consists of a number of parts. Part 3 covers the requirements for reference and other test signals of short duration.

Examples of test methods, where such signals are commonly used, are the recording of brainstem evoked potentials and evoked otoacoustic emissions. Reference signals are described in order to provide a basis for calibration and as a recommendation for use when there is no specific reason to have an alternative signal. The method of measurement of acoustic and vibratory signals is described.



• Everything with a duration of less than 200 ms



Calibration procedure



Conditions:

- Make measurements inside of a sound-treated room.
- Use authorized equipment to make measurements
- Keep the enviromental conditions according to norms.



Calibration- Artificial Ear

Artificial ear:

Has an acoustical impedance similar to that the human ear. USES:

- Frequency response and sensitivity measurements on insert earphones and headphones
- · Calibration of audiometers







- A device designed for measurement of the intensity of sound waves in air.
- It consists of a microphone, an amplifier, a frequency-weighting circuit, and an output meter calibrated in decibels with a reference of 20µPa.





- Top figure is the equal loudness contour for 40 dB at 1 kHz.
- The popular A-weighting at the bottom is shown in comparison to the inverted 40 dB equal loudness contour.





Frequency Weighting Curves

- The A-weighting, B-weighting and C-weighting curves follow approximately the 40, 70 and 100 dB equal loudness curves respectively.
- D-weighting follows a special curve which gives extra emphasis to the frequencies in the range 1 kHz to 10 kHz. This is normally used for aircraft noise measurements.





Calibration procedure

Equipment

- Oscilloscope
- Signal Generator
- Sound Level Meter w. Microphone & calibrator
- Sound proof chamber
- Headphones
- sound stimulus
- Humidifier
- Humidity & temperature sensor











Set-up: peSPL



Fig. 2.3. Typical measuring arrangement for the checking of headphone characteristics



PeSPL measurement:

- 1. Measure environmental conditions: temperature and humidity.
- 2. The peak-to-peak voltage of the stimulus is measured in the oscilloscope.
- 3. A sinusoidal wave with a freq. of 1KHz and the peak-to-peak voltage of the stimulus is created in the signal generator.
- 4. Earphone is enclosed within the artificial ear.
- 5. SLM is calibrated
- 6. Microphone of SLM is placed inside the artificial ear.
- 7. The artifical ear is placed in position as required in the norms. Inside the chamber with headphones separated from each other by 15 cm.
- 8. The sinusoidal wave is sent to the earphones.
- 9. The acoustic stimulus is received by the SLM
- 10. Close the chamber.

11. Obtain the measurement from the SLM corresponding to the peSPL for that specific stimulus.





Calibration Procedure

SPL measurement:

- 1. Using the same set-up, send now the stimulus file and obtain the SPL value displayed in the SLM.
- 2. Compare the 2 measurements.





Auditory Processing and Perception BMT 925, Course 2013 Hands on Sessions

Farah I. Corona-Strauss



Hands on Sessions

- 1. Audiogram
- 2. Rinne and Weber Test, OAEs
- 3. ABR-Obtain latency shift graphics
- 4. ALR Obtain ALRs related to Tinnitus
- 5. Artificial Ear Calibration procedure See response for different frequencies and different stimulation levels.
- 6. The computational ear



The Computational Ear



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	Lyons Passive Long Wave Cochlear Model
	LyouPatriveExc(21) DesignLyouCascade (15) soccascade (46) sgc (6) SecondOrderFiller (37) EpsiloneFromTandFS (17) sonfilters (47)
	Seidain (44)
	Patterson-Holdswortn EKB Futer Bank MakeEEBFührs (24) EEBFührsBank (18) MeddsiHairCell (27)
	Seneff Auditory Model
	SenstEar (39) SenstEarSemp (42)
	Alternate Analysis Techniques mfcc (29) spectrogram (49) prockpc (33) symbol (50) rasin (35)
	Correlogram Processing
	CorrelogramPitch (12)
	CorrelogramFrame (10) Demonstrations
	MakeVowel (26) FMPoints (19)
	WhiteVowel (52)
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 Auditory T Auditory T Record a a sampling Using mat Compute Compute 	Foolbox-Matlab sound file with your name using g frequency of 16000Hz. tlab plot the waveform the spectrogram and plot it the correlogram/movie and plot it





- The autocorrelogram, or simply correlogram, is a visual display of sound periodicity and an important representation of auditory temporal activity that combines both spectral and temporal information. It is normally defined as a three-dimensional volumetric function, mapping a frequency channel of an auditory periphery model, temporal autocorrelation delay (or lag), and time to the amount of periodic energy in that channel at that delay and time.
- · The periodicity of sound is well represented in the correlogram.
- If the original sound contains a signal that is approximately periodic, such as voiced speech, then each frequency channel excited by that signal will have a high similarity to itself delayed by the period of repetition.





	Auditory Toolbox
 Calculate (MakeERBfilters filterbank and apply it to yo 	-ERBFilterBank) a ERB our sound file.
ERBFilterBank	0 · · · · · · · · · · · · · · · · · · ·
Purpose Fiber sa sudio signal with a bank of Gammatone fibers SynopSiS output = ERBFiberBank(x, fooff.)	-20-
MakeERBFilters	

• The gammatone filter is widely used in models of the auditory system and is physiologically motivated to mimic the structure of peripheral auditory processing stage. The gammatone function is defined in time domain by its impulse response:

Purpose Design the filters needed to implement an ERB cochlear model.

 $g(t)=at^{n-1}cos(2\pi ft+\phi)e^{-2\pi bt}$

10³

10⁴

 where is the order of the filter which largely determines the slope of the filter's skirts; is the bandwidth of the filter and largely determines the duration of the impulse response; is the amplitude; is the filter centre frequency; is the phase.



A simple cochlear model can be formed by filtering an utterance with these filters. To convert this data into an image we pass each row of the cochleagram through a half-wave-rectifier, a low-pass filter, and then decimate by a factor of 100. A cochleagram of the 'A huge tapestry hung in her hallway' utterance from the TIMIT database



 Make a vowel make a white vowel and perform the same previous processing as for your sound file, i.e., spectrogram, correlogram, filters....

Sumple votes symbols y = NsharVowel(lea, pitch, sampleRate, fl, fl, fl, fl) YhiteVowel Purpose Demonstrate the effect of whitesing one portion of a speech signal Synopsis [output_sCoeff] = WhiteVowel(data_st.L.pos)	
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WhiteVowel Purpose Demonstrate the effect of whitening one portion of a speech signa Synopsis [output_sfCoeff] = WhiteVowel(data_sr,Lpos)	Synopsis v = MakeVowel(len, witch, sampleRate, fl, f2, f3)
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Open Matlab

Systems Neuroscience & Marine Participation Systems Neurotechnology Unit

- Record a sound file with your name using a sampling frequency of 32000Hz.
- Type HAP on the command window
- Select different sounds, including your created sound file and run the simulation using different parameters, such as frequency range, number of channels, intensity.



HAP: model of the BM





Using the Auditory Toolbox compute for the previously recorded sentence the following:

- Spectogram
- Correlogram
- Compute ERB filter bank and apply it to the sentence
- Remove the high frequency channels and play the sentence
- Make a "white" effect of the original sentence.





MATERIALS:

MAICOAudiometer

METHODOLOGY:

- Write information of the subject: Last name, Name, Birth date, and gender.
- Place the headphones making sure the placement is correct (L on the left ear and R on the right ear). The subject should not be able to see the operator of the audiometer nor the screen of the laptop.
- The program starts with the right ear and a frequency of 1 KHz.



Experiment: Audiogram

- To increase the intensity of every frequency, move slowly the cursor with the arrows of the keyboard.
- When a response of the subject is show on the display, press enter and the intensity value will be stored. Then, a new frequency will be available to be applied. When the high frequencies are applied, the program will continue with the low frequencies.
- When the audiogram of the right ear is ready, press the blue L in order to start testing the left ear.
- Determine if a hearing loss in one or both ears exists (Hearing loss: if one or more frequencies are larger than 15 dB HL).



MATERIALS: – Tuning Fork METHODOLOGY

- The Weber test can detect unilateral conductive hearing loss and unilateral sensorineural hearing loss.
- Take a tuning fork and perform the Weber test. The procedure is as follows: the vibrating tuning fork is held against the forehead at the midline and conduction deafness is indicated if the sound is heard more loudly in the affected ear and nerve deafness is indicated if it is heard more loudly in the normal ear.
- What were the results?



Experiment: Rinne Test

MATERIALS

– Tuning Fork METHODOLOGY

- The Rinne test compares perception of sounds, as transmitted by air or by bone conduction through the mastoid. Thus, one can quickly suspect conductive hearing loss.
- Make the Rinne test by placing a vibrating tuning fork initially on the mastoid until the sound is no longer heard, then place immediately the fork just outside the ear. Normally, the sound is audible at the ear.
- Write the comments of the subjects, is the subject a normal hearing subject?



MATERIALS:

OAE device

METHODOLOGY:

- The OAE device has a probe that fits in the outer ear canal of the subject.
- Before placing the probe, the correct size of the olive should be determined (the olive is the small rubber piece that serves as seal in order to avoid leakage of the auditory stimulus).
- Find the correct size of the olive according to the ear canal of the subject, when it is selected, place it in the probe and next, introduce it inside the ear canal. Clean the used olives with alcohol.
- Turn on the OAE device and press the start button. The subject should remain quite and still during the measurement time.
- The device will give a pass or fail result. When this result is collected, remove the olive from the probe, and clean it with alcohol.
- Turn on the device.
- What were the results?



Experiment: ABR-Obtain latency shift graphics

OBJECTIVE: Obtain the latency shift graphic of ABRs for different stimulation types (clicks and chirps) & intensity levels.

METHODS: Obtain 4 files of ABRs with different stimulation levels and then create the latency shift plot.



ABR Setup







METHODOLOGY

- Turn on devices
- Review setup.
- STIM UNIT: max. volume, repeat mode selected, open stimuli under: "D:\AP&P_handson\ABRs\"
- ACQ. UNIT:
 - Open & run the user interface under the pahtway: D:\ AP&P_handson\ABR\ABRguid4.fig
 - Open the icon of the amplifier: check serial number, channels selected (1,2,9), sampling frequency (19200), filters (none)







Experiment: ABR-Obtain latency shift graphics

- Set the filename by double click on the name box:
 - Subject Name+stimulus+number:
 - example: Carlos_ck_01 for "click" stimulation using "clickalt20hz4tb.wav"
 - and Carlos_ch_01 for "chirp" stimulation using "deBoer_Chirp.wav"
 - File 1: name_ck_01 ; Intensity=60
 - File 2: name_ck_02 ; Intensity=50
 - File 3: name_ck_03 ; Intensity=40
 - File 4: name_ck_04 ; Intensity=30
 - File 5: name_ch_01 ; Intensity=60
 - File 6: name_ch_02 ; Intensity=50
 - File 7: name_ch_03; Intensity=40
 - File 8: name_ch_04 ; Intensity=30

STIM-UNIT:

· Open corresponding stimulation file

ELECTRODE PLACEMENT

• Identify the positions on the scalp: Ch1: Right mastoid, Ground: front forehead, Reference: Cz.



- Clean carefully the placement area using an abrasive gel. This will ensure low impedances for the recordings.
- Fill the electrode with electrode gel and prevent air inclusions. And plug the electrode at the desired position



Systems Neuroscience & Marine Participation Systems

Experiment: ABR-Obtain latency shift graphics

 Check the impedances of the electrodes against a reference electrode using Impedance check icon in the SaveToFileABRs.mdl file. (The impedances should be lower than 10 kOhm. Impedances around 5 kOhms are optimal).



 If the impedance of an electrode is > 20 kOhms repeat steps 2 to 4. Connect now the electrodes to the biosignal amplifier e.g. <u>g.BSamp</u>.



ACQUISITION

- 1. Place the headphones on the subject and verify the impedances again (this step is only to make sure that no changes had occurred after placing the headphones). Make sure the headphones are placed correctly (the stimulus sounds on the right ear).
- 2. Select a stimuli file and play it (computer 2: Stim-unit).
- 3. Select intensity level in the user interface (computer 1: *Acq-unit*), and select "attenuate".
- 4. Make sure that the subject hears the stimuli on the right ear.
- 5. Ask the subject to remain quiet and without moving, sleep if possible.
- 6. Turn off the lights and start the measurement by pressing the Start button in the user interface (computer 1: *Acq-Unit*).
- 7. When the acquisition is done wait until the two colormaps are shown before starting the new acquisition. To start a new measurement change the name of the file and the intensity level if necessary, both in the user interface.
- 8. Select the stimulation file and play it.
- 9. Start acquisition again and repeat steps 7 and 8 until all the files are acquired.



10. In order to process all the files use the left side of the user interface:



- Select the "Trace option" and write the name without number. On the field "No. of files" write 4. Press process and wait until the data is displayed.
- What were the differences in amplitudes and latencies in general comparing clicks and chirps?





- Turn off first the amplifier
- Turn off the rest of the devices: trigger box,mp3 player, cd player, computers.
- Remove eletrodes, washing them with cold water.



HANDS ON SESSIONS Auditory Late Responses



- OBJECTIVE: Obtain the Auditory Late Responses (ALRs) using a high entropy paradigm and verify the effects of attention in the ALRs.
- PARADIGM: using the paradigm specified in the next figure obtain ALRs for 20 minutes: A sound file with three different tones will be played during the entire time of the experiment. In the first ten minutes, the subject has to press a button when he/she hears a specific tone (target tone). The last ten minutes, the subject should ignore the tones and do not pay attention. With the data collected, the changes of the ALRs during the different phases of the experiment will be analyzed. Later, the synchronization stability of the ALRs will be calculated, and the results should be also analyzed.



A stimulus file is sent to the right ear while the left ear is receiving music. The stimulus file has three different tones, each one with duration of 300 ms, and with an interstimulus interval of I-3 seconds. The sequence of the tones is randomly arranged, and has a total duration of 20 minutes. During the first 10 minutes the subject has to identify one of the three tones and count the number of times that he recognizes it. The tones are very similar, so the experiment requires that the subjects are really focused on the target tone. The next 10 minutes, the subject has to relax, not pay attention, ignore the tones and stop the counting. The data is recorded during the 20 minutes of the experiment. The processing is made offline (filtering, averaging and calculation of the ALR and the synchronization stability for the two stages, attention and no attention).



• Tone A, B, C, attention and no attention:

target tone





ALRs experiment

INFORMATION TO THE SUBJECT:

- The subject will receive training, in order to identify a target tone.
- The subject will hear 3 tones, one is the target and he/she should press a button every time when the target tone appears, but only during the first part of the experiment (first 10 minutes).
- The next ten minutes (11-20), the subject will receive a **sign** which indicates that he/she should ignore the tones henceforward and should not press the button anymore.
- During the entire experiment the subject should remain with the eyes closed.
- Make sure the subject do NOT sleep at any moment of the experiment.





METHODOLOGY:

- Turn on the computers, attenuator, trigger box, and amplifier.
- Verify the connections according to the setup (have special consideration while checking the headphones: left connector to the mp3player, and right to the attenuator; and also check if the button is connected to the triggerbox).

ACQ-UNIT

- Open PAH_2ch executable program (desktop):
 - Select serial port (COM1); Select attenuation of 30dB for the stimulus (Right).
- Open MATLAB and go to the directory D:\AP&P_handson\ALR_Attention\three_tones and open the file "ALR4channels.mdl".
 - Select the icon of the amplifier and check: serial
 - number, channels selected (1,2,5,9), sampling frequency (512), filters(none)
 - In the same program check the acquisition time (1210 seg).
 - Set the filename (icon named 'To File 1').

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ATT left	ATT right	CATTING	ATTRIN
(• 0d8	(€ 0.dB	@ 0.48	C 0.49
C 10 d8	C 10:00	C 10.49	C 10.40
C 20 d8	C 20.48	C 20.48	(C 200 dD)
C 30 d8	C 30 dB	C 20 00	(* 20 ab
C 40 dB	C 40 dB	C JU dB	C 30 68
C 50 dB	C 60 dB	C 40 dB	C 40 dB
C 60 08	C 60 dB	C 50 dB	C 50 dB
C 90-40	C 70 dB	C 60 dB	C 60 dB
C 90.49	C 00 40	C 70 dB	○ 70 dB
C 100.48	C 100.40	C 80 dB	C B0 dB
C 110 dB	C 110 dB	C 90 dB	C 90 dB
C 120 dB	C 120-68	C 100 dB	C 100 dB
Cimute	C mate	C 110 dB	C 110 dB
		C 120 dB	C 120 dB
COCOMPO.		C muto	C nute
CON2	A Éd		
COME			
COM64		COM6 v	X Ext



Acquisition







Stim-Unit

- Open the folder "AP&P\handson\ALRs_attention\three_tones" placed at the desktop.
- · Verify if the volume of this computer is at the maximum level .
- Make sure to hear the stimuli in the headphones and make sure that the option "repeat" is NOT selected in the sound player program.
- Turn on mp3player (music for right ear): play the song "01 Listen to Dreamy Sounds", select the option "repeat 1", and set the volume at 07.



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ELECTRODE PLACEMENT

 Identify the positions on the scalp: Ch1: Right mastoid, Ground: front forehead, Reference: Cz.



- Clean carefully the placement area using an abrasive gel. This will ensure low impedances for the recordings.
- Fill the electrode with electrode gel and prevent air inclusions. And plug the electrode at the desired position

 Check the impedances of the electrodes against a reference electrode using Impedance check icon in the SaveToFileABRs.mdl file. (The impedances should be lower than 10 kOhm. Impedances around 5 kOhms are optimal).



 If the impedance of an electrode is > 20 kOhms repeat steps 2 to 4. Connect now the electrodes to the biosignal amplifier e.g. <u>g.BSamp</u>.



Systems Neuroscience & Neurotechnology Unit

ALRs experiment

Starting an Acquisition

- Place the headphones on the patient and verify the impedances again (this step is to make sure that the electrodes remained connected after placing the headphones).
- Make sure the subject hears the music of the mp3player (music).
- Play the training file 1-3 times. (target.wav). At this point the subject should be able to identify the target tone.
- Tell the subject the experiment will start and he/she should press the response button every time he/she hears the target tone until he receives your signal which means (the first ten minutes are over and now starts the second part of the experiment=no attention) that he should stop solving the paradigm and should from now on ignore the stimulations.
- Make sure you have ready the stimulation file ""threetonesstim1" (de repeat option on windows media player should be off)
- Play "ALR4chanels.mdl" and immediately after play the stimulation file "threetonesstim.wav".
- After 10 minutes let the patient knows that he should stop pressing the button and paying attention to the tones.
- Wait until the acquisition is finished.



Processing

- · Go to the command window of MATLAB:
- Type: ALRapp('filename')

Results:

- Explain the results in the plot of the synchronization stability (target tone- tone B) in terms of attended /no attended conditions.
- Verify amplitudes for ALRs and synchronization stability of the ALRs at N100 for the target tone.
- Compare with the results of the other group.



Fig. 2. The difference of the synchronization stability for a subject (a = 40 as example) for 3 different tones.



- Turn off first the amplifier
- Turn off the rest of the devices: trigger box,mp3 player, cd player, computers.
- Remove eletrodes, washing with cold water and storing with NaCl solution.



Questions?