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INVITED REVIEW

PSYCHOACOUSTICS: Software package for psychoacoustics

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Abstract: Current research in the field of psychoacoustics is mostly conducted using a computer to generate and present the stimuli and to collect the responses of the subject. However, writing the computer software to do this is time-consuming and requires technical expertise that is not possessed by many would-be researchers. We have developed a software package that makes it possible to set up and conduct a wide variety of experiments in psychoacoustics without the need for time-consuming programming or technical expertise. The only requirements are a personal computer (PC) with a good-quality sound card and a set of headphones. Parameters defining the stimuli and procedure are entered via boxes on the screen and drop-down menus. Possible experiments include measurement of the absolute threshold, simultaneous and forward masking (including notched-noise masking), comodulation masking release, intensity and frequency discrimination, amplitude-modulation detection and discrimination, gap detection, discrimination of interaural time and level differences, measurement of sensitivity to temporal fine structure, and measurement of the binaural masking level difference. The software is intended to be useful both for researchers and for students who want to try psychoacoustic experiments for themselves, which can be very valuable in helping them gain a deeper understanding of auditory perception.

Keywords: Psychoacoustics, Software

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1. INTRODUCTION

Current research in the field of psychoacoustics is mostly conducted using a computer to generate and present the stimuli and to collect the responses of the subject. The package of software described here, named "*PSYCHOACOUSTICS*" allows basic and advanced psychoacoustic research to be conducted without programming by the user. Some other such software packages have a similar purpose. See for example, http://samcarcagno.altervista.org/pychoacoustics/pychoacoustics.html and https://exporl.med.kuleuven.be/web/index.php/Public:Software/APEX. However, we believe that our package is more comprehensive and user-friendly than these other packages.

Possible experiments include measurement of the absolute threshold, simultaneous and forward masking (including notched-noise masking), comodulation masking release, intensity and frequency discrimination, amplitude-modulation and frequency modulation detection and discrimination, gap detection, discrimination of interaural time and level differences, measurement of sensitivity to temporal fine structure, and measurement of the binaural masking level difference. The package also includes several tools for analysing the results of the experiments.

2. INSTALLATION

Installation of the software is typical of that for other software packages. Installation is initiated by clicking on a file called "PSYCHO_Install.msi." The installation procedure creates all necessary directories and installs all experiment-related files in the directory c:/users/public, so that they are available to all users. Finally, the installation procedure adds an icon to the desktop. The program can be run by clicking on this icon. The software operates on a PC (operating systems from Windows XP to 10) and requires a sound card and headphones. A 16-bit sound card can be used, but a 24-bit external sound card is recommended. Ideally, the sound card should be in a metal or shielded case that reduces the influence of external electromagnetic fields. High-quality stereo headphones, e.g. Sennheiser HD580 to 650, are recommended.

When the program is first started, a message is put on the screen instructing the user to disable Windows sounds effects and telling them how to do so (The option "No

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sounds" should be chosen). This is necessary to prevent such sounds from being played during a measurement. Once this has been done the program closes and then restarts. At this stage, calibration is required.

3. CALIBRATION

To perform the calibration procedure it is necessary to measure the voltage generated by the sound card when a calibration signal is played. This calibration signal is a 1,000-Hz sinewave generated at the highest level allowed by the sound card (the full-scale level). The voltage can be measured using a voltmeter or some other device such as an oscilloscope. Ideally, the measurement should be made while the headphones are connected to the sound card, since the headphones may "load" the sound card and change the voltage. The user has to enter the measured voltage in the software and also to enter the "sensitivity" of the headphones, which is the sound level produced by a 1-V RMS input signal, usually with a frequency of 1 kHz. Once this information has been entered, the software indicates the maximum and minimum sound levels that can be generated.

If the indicated range is unsuitable, the calibration procedure should be repeated with the gain of the sound card adjusted using the appropriate sliders in the "Sound Mixer." If it is desired to measure absolute thresholds accurately using a 16-bit sound card, the maximum level should 75–80 dB SPL. With a 24-bit soundcard, a maximum level of 85–90 dB SPL allows sound levels down to −20 to −15 dB SPL, which is adequate for most applications. Once the *Stop* button is pressed, the voltage and the Sound Mixer settings are saved and access to the main program is allowed.

When starting any experiment, the software recalls the saved settings, so that calibration is required only once. However, if the soundcard and/or headphones are changed, recalibration is required.

4. EXPERIMENTAL METHODS

For all experiments, it is required to specify the name or initials of the subject, the tested ear (L or R), and parameter values of the stimuli for that experiment.

The experiments can be run using two- or three-alternative forced-choice methods (*2AFC* or *3AFC*). Each trial contains two or three observation intervals, but only one of them contains a signal. The task is to indicate the interval with the signal. The variable of interest (e.g. the signal level) is varied from trial to trial using an adaptive procedure. Two procedures are available, called *2d1u* and *3d1u*. For the *2d1u* procedure, the variable parameter is decreased following two successive correct responses, and increased following one incorrect response. This tracks the variable value corresponding to 70.7% correct [1]. For the *3d1u* procedure, the variable parameter is decreased following three successive correct responses, and increased following 1 incorrect response. This tracks the variable value corresponding to 79.4% correct [1]. The number of turnpoints, n, can be 8, 10 or 12. The threshold is calculated as the mean of the value of the variable at the last $n-4$ turnpoints. For each experiment, the method (*2AFC* or *3AFC*) and the decision rule (*2d1u* or *3d1u*) need to be specified.

For experiments on modulation detection, identification, and discrimination, it is also possible to determine psychometric functions (percent correct as a function of signal magnitude). To do this, maximum and minimum signal parameter values must be specified. The software calculates three intermediate parameter values, giving five in total. Signals with these parameter values are presented in a series of 30 trials (five for practice and five for each parameter value) or 55 trials (five for practice and ten for each parameter value). The outcome is the number of correct responses for each parameter value.

During all experiments, each observation interval is indicated by lighting up a box on the screen, and correct-answer feedback is (optionally) provided.

The *Start* button for each experiment launches the experiment and saves the input data in a text file (**.in*) in the directory for a given experiment and subject. The result of each measurement is saved in a text file (**.out*), and the information about the file location is displayed. The name of the file is created based on the subject name, the current date and the measurement number. For each experiment, a sample input file (read-only) is included to facilitate its use.

5. TYPES OF EXPERIMENT

The software is organised into a series of modules, each of which enables a specific type of experiment to be performed. The main modules are described below.

5.1. Absolute Threshold

This module allows determination of the absolute threshold. It is necessary to specify the signal frequency, the duration of each observation interval, and the gap between intervals.

5.2. Masking

This module allows determination of the detection threshold of a sinusoidal signal in simultaneous, backward, and forward masking, using a noise masker whose passband has to be specified. The module can be used to replicate and extend several classic experiments [2,3]. The duration of the masker and signal, the rise and decay times of the signal and masker, and the time between the onset of the masker and the onset of the signal (*Asynchrony*) must be specified. Backward masking is achieved by setting

Asynchrony < 0, while forward masking is achieved by setting *Asynchrony* > *Masker duration*. The phenomenon of "overshoot" [4] can be measured by choosing *Asynchrony* so that the (brief) signal falls at different temporal positions within the masker duration. The masker level can be specified as the total level (dB SPL) or as the spectrum level (level per 1-Hz wide band) within the masker passband. Either the signal level or the masker level can be varied adaptively to estimate the threshold.

5.3. Notched-noise Method

This module implements the notched-noise method for determining the shape (the frequency response) of the auditory filter [5,6]. The experiment involves measuring the detection threshold of a sinusoidal signal presented in a spectral notch between two bands of noise. The difference between the signal frequency and the upper cut-off frequency of the lower band, divided by the signal frequency, is denoted *Deltaf*/f_{lower}. The difference between the lower cut-off frequency of the upper band and the signal frequency, divided by the signal frequency, is denoted *Deltaf*/f_{upper}. At least three pairs of values of *Deltaf*/f_{lower} and *Deltaf*/f_{upper} should be specified. The spectra and levels of the two noise bands are independent. However, the two bands have the same duration and rise/decay time, and their level must be specified in the same way (overall level or spectrum level). The duration of the signal and its onset asynchrony must be chosen such that the signal ends no later than the end of the masker (simultaneous masking).

5.4. ROEX_BG

This module allows the determination of the auditory filter shape based on data from the notched-noise experiment [6,7]. The information appearing in *Pocket help* describes the parameters to be entered. The experimental data (*Enter data*) are entered in a separate window. The signal threshold should be entered for all pairs of values of *Deltaf*/f_{lower} and *Deltaf*/f_{upper}. To derive auditory filter shapes from the data, it is assumed that the auditory filter shape can be approximated by a simple mathematical expression with a small number of free parameters. Patterson *et al.* [7] suggested that each side of the passband of the filter could be modeled as an exponential with a rounded top, called rounded-exponential or "roex." It is convenient to measure frequency in terms of the absolute value of the deviation from the centre frequency of the filter, f_c, and to normalize this frequency variable by dividing by the centre frequency. The new frequency variable, g, is:

$$g = |f - f_c|/f_c. \tag{1}$$

The roex(p) filter shape is then given by:

$$W(g) = (1 + pg)\exp(-pg), \tag{2}$$

where p is a parameter that determines both the bandwidth and the slope of the skirts of the auditory filter. The higher the value of p, the more sharply tuned is the filter. When the filter is assumed to be asymmetric, then p is allowed to have different values on the two sides of the filter: p_l for the lower side and p_u for the upper branch. The filter model also includes a parameter r (specified in dB) that limits the dynamic range of the filter.

The software uses an iterative procedure to determine the values of p_l, p_u, and r that predict the obtained thresholds as accurately as possible. The software plots the frequency response of the resulting auditory filter on linear and logarithmic scales. It also displays the final values of the p_l, p_u, and r and the equivalent rectangular bandwidth (*ERB*) of the auditory filter.

5.5. Intensity Discrimination

This module is used to determine the difference limens (DLs) in intensity (also called just noticeable differences, JNDs) for sinusoidal signals (*SS*) or noise bands (*NB*). In this module (and in some others), *Frequency roving* and *Level roving* can be used. This means that the frequency (or centre frequency) and signal level are selected randomly from within a user-specified range. The roving of level occurs across trials but not within trials. The roving encourages the subject to compare the stimuli presented within each trial rather than to compare each stimulus with a representation in long-term memory. The duration of the stimuli and their rise/decay time are selected by the user, as is the baseline level. This allows measurement of intensity difference limens as a function of duration and level.

5.6. Frequency Discrimination

This module is used to determine frequency DLs for sinusoidal signals and bands of noise. Otherwise it is similar to the module described in Sect. 5.5. Frequency DLs can be measured as a function of frequency, duration, and level. *Frequency roving* and *Level roving* are also available with this module.

5.7. Gap Detection

This module allows determination of the shortest detectable silent interval in a sinewave or a band of noise. The centre frequency and the passband and level of the noise are defined by the user. Abrupt switching off (or on) of a signal causes a significant broadening of its spectrum, called "spectral splatter." To prevent the splatter being used for detection of the gap, ramps of a user-specified duration can be applied at the start and end of the gap. If it is desired to use very short ramps, for which splatter may still be audible, there is an option to present an additional noise to mask the splatter. The parameters defining this noise are specified independently of those for the "target" band of noise or sinewave. The additional noise background is presented as one continuous burst during all observation intervals in a trial.

5.8. Modulation Detection

This module allows measurement of amplitude modulation (AM) and frequency modulation (FM) detection thresholds using a sinusoidal or noise band carrier. The level, centre frequency, duration, bandwidth (for a noise carrier), and modulation rate are specified by the user. The threshold for detection of AM as a function of AM rate is called the temporal modulation transfer function (TMTF), and it has often been used to characterise temporal processing in the auditory system [8,9]. The detection of FM as a function of FM rate has been used to explore the mechanisms underlying FM detection. FM detection is thought to depend on three mechanisms: for very low rates ($\leq$5 Hz) and for carrier frequencies below 5 kHz, a mechanism based on the use of temporal fine structure information (phase locking) may be dominant [10–12]; for rates from 5 Hz up to about 10% of the carrier frequency, a mechanism based on the conversion of FM to AM in the auditory system may be dominant [11,13,14]; for still higher rates, the spectral sidebands of the FM signal may be resolved in the auditory system, and threshold is determined by the detection of the sidebands [15,16].

5.9. Discrimination of Interaural Differences

Interaural differences of time and intensity (ITDs and IIDs) are used to judge the direction in azimuth of sound sources. For sinewave signals, ITDs play a dominant role at low frequencies, while IIDs play a dominant role at high frequencies [17]. This module allows measurement of just detectable changes in ITD or IID from reference ITD and IID values of 0, for sinewaves and for bands of noise. The user specifies the frequency (or centre frequency and bandwidth for noise signals), baseline level at the left and right ears, signal duration, the interval between signals, and the initial value of the ITD or ILD.

5.10. Binaural Masking Level Difference (BMLD)

The threshold for detecting a sinusoidal signal presented in a diotic background noise (same noise at the two ears) can be much lower when the interaural phase of the signal is 180° than when the interaural phase is 0° (diotic signal) [18,19]. The difference between thresholds for the two cases is called a binaural masking level difference (BMLD). For a low-frequency signal, the BMLD can be as large as 15 dB. More generally, a BMLD can occur whenever the interaural phase or level of the signal differ from the interaural phase or level of the masker [20,21].

This module allows measurement of the BMLD for six different cases (*Measurement type*). For the signal, the following cases are possible: in phase at the two ears (So), in antiphase at the two ears (Sπ), or monaural (Sm). For the noise, the following cases are possible: in phase at the two ears (No), in antiphase at the two ears (Nπ), or uncorrelated at the two ears (Nu). The BMLD can be measured by comparing thresholds for six pairs of conditions: NoSo vs. NoSπ; NoSo vs. NπSo; NoSo vs. NoSm; NoSo vs. NπSm; NoSo vs. NuSo; and NoSo vs. NuSπ.

The software allows the user to specify the characteristics of the signal and masker, such as frequency, bandwidth, level and duration. Generally, the BMLD is largest for low-frequency signals [20], but it can be large for high-frequency signals when the masker bandwidth is very small [22].

5.11. Comodulation Masking Release (CMR)

The threshold for detecting a sinusoidal tone presented at the centre frequency of a random band of noise with a fixed spectrum level increases with increasing masker bandwidth up to a certain bandwidth (called the "critical bandwidth"), and then remains roughly constant with further increases in masker level [2]. This is illustrated by the points labelled R in Fig. 1. However, if the noise is amplitude modulated using an independent lowpass noise as the modulator, a quite different pattern of results is observed: the signal threshold at first increases with increasing noise bandwidth and then decreases markedly

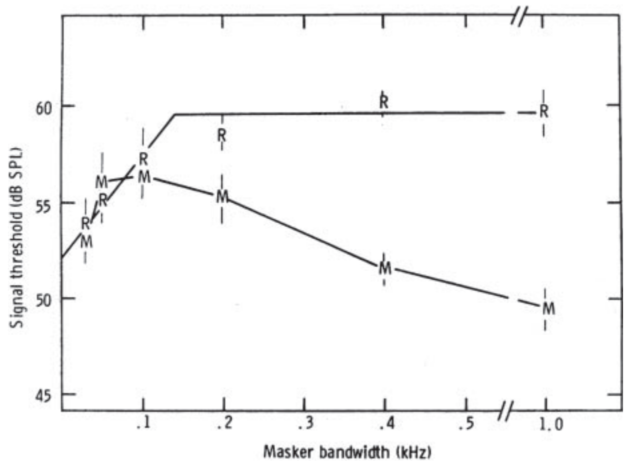

Fig. 1 The points labeled "R" are thresholds for detecting a 1-kHz signal centered in a band of random noise, plotted as a function of the bandwidth of the noise. The points labeled "M" are the thresholds obtained when the noise was amplitude modulated at an irregular, low rate. From [23] by permission of the author.

[23]. This is illustrated by the points labelled M in Fig. 1. The difference in threshold for the random noise and the modulated noise is called comodulation masking release (CMR).

This module allows measurement of CMR. The signal is a tone while the masker is a band of noise. The user can specify the signal parameters (level, frequency, duration), the masker parameters (centre frequency, level, bandwidth, duration) and the parameters of the modulator, which can be a sinewave, a narrow band of noise, or a lowpass noise.

When a noise is used as the modulator, the inherent amplitude fluctuations in the noise limit the modulation depth that can be used. To allow large modulation depths to be used, the software includes an option for creating a noise modulator with reduced amplitude fluctuations; this is called low-noise noise, LNN [24].

5.12. Modulation Detection (Discrimination) Interference (MDI)

The detection or discrimination of amplitude modulation (AM) or frequency modulation (FM) applied to a carrier C1 can be significantly more difficult if an additional AM or FM sound is present, even when the carrier frequency of the additional sound, C2, is remote from that of C1. This effect is called modulation detection interference (MDI) [25,26]. MDI can occur for AM sounds and FM sounds, and across modulation types: AM on C2 can interfere with the detection and discrimination of FM applied to C1, and FM on C2 can interfere with the detection and discrimination of AM applied to C1 [27]. The MDI effect shows tuning in the modulation domain: MDI is greatest when the modulation on C2 has a similar rate to the modulation on C1 [28,29].

This module allows determination of AM and FM detection thresholds in the absence or presence of up to four additional modulated sounds (*flankers*). The signal and flankers can be sinusoidal tones or noise bands, although FM can only be applied to the tones. The module requires selection of the type of signal to be used (tones, *T*, or noise bands, *N*), and their parameters (frequency or centre frequency, modulation rate, modulation depth, level, duration). The parameters defining the characteristics of the *flankers* must also be specified.

In the absence of flankers, this module allows measurement of AM detection thresholds when interfering FM is present (in the same interval of a forced-choice trial) and of FM detection thresholds in the presence of interfering AM (in the same interval of a forced-choice trial).

5.13. Fast Method for Determining Psychophysical Tuning Curves

Psychophysical tuning curves (PTCs) are often used to characterize the frequency selectivity of the auditory system [30,31]. To measure a PTC, the sinusoidal signal is fixed in frequency and presented at a fixed (usually low) sensation level (about 10–15 dB SL). A narrowband noise is usually used as the masker. For each of several masker centre frequencies, the level of the masker required just to mask the signal is determined. This module incorporates a fast method of determining PTCs, called *SWPTC*, using a band of noise whose centre frequency is slowly and continuously swept from below to above the signal frequency or vice versa [32,33]. The masker level required for threshold is measured using a method similar to Békésy audiometry. The masker level is increased whenever the subject indicates that the signal is audible (by pressing the space bar), and is decreased whenever the space bar is released. The signal is pulsed on and off regularly to help the subject to "know what to listen for." A similar method was used by Zwicker [34].

The basic version of this module (*SWPTC*) requires specification of the signal parameters (frequency, level, duration, interval between pulses) and the noise parameters (lowest and highest centre frequencies, bandwidth, direction and speed of noise level changes in dB/s, and initial noise level). The software estimates the sharpness of the tip of the PTC based on the measure Q10 dB (tip frequency divided by the bandwidth measured 10 dB above the tip), using several methods. The software also estimates the frequency at the tip of the PTC. If the tip frequency is shifted by more than about 10% from the signal frequency, this may indicate the presence of a "dead region" in the cochlea at the signal frequency [35–39]. A dead region is a region with very few or no functioning inner hair cells, synapses, or neurons [40].

An option is provided, called *SWPTC_ERB*, in which the masker bandwidth is equal to the ERB of the auditory filter at the centre frequency of the noise [6]. This can be effective in reducing the influence of beats while minimising the broadening of the tip of the PTC produced by the finite bandwidth of the masker [41,42].

Another option is called *SWPTC_LPN*. This allows a fixed lowpass noise to be presented in addition to the signal and sweeping masker. This noise is intended to mask combination tones that result from the interaction of the signal and sweeping masker in the auditory system [42–44]. A related option is called *SWPTC_TEN*. This allows a Threshold Equalizing Noise (TEN) [36] to be presented in addition to the signal and sweeping masker. For all these options it is possible to present a variety of sounds to the opposite ear to that receiving the signal and sweeping masker, to investigate effects of contralateral stimulation. This may be useful, for example, to investigate the effects of the efferent system [45,46].

5.14. Sensitivity to Temporal Fine Structure (TFS)

Within a normal cochlea, broadband sounds like speech and music are decomposed into narrowband signals, each of which can be considered as a relatively slowly varying envelope (ENV) imposed on a rapidly oscillating carrier (the temporal fine structure, TFS) [47]. This module (*TFS section*) provides a group of programs that assess sensitivity to changes in TFS [48,49].

A monaural test of sensitivity to TFS is the TFS1 test [48]. The test involves discrimination of a harmonic complex tone (H), with a fundamental frequency F0, from a tone in which all harmonics are shifted upwards by the same amount in Hertz, resulting in an inharmonic tone (I). The phases of the components are selected randomly for every stimulus. Both tones have an envelope repetition rate equal to F0, but the tones differ in their TFS. To prevent discrimination based on spectral cues, all tones are passed through a fixed bandpass filter, centred on the high unresolved harmonics. A background noise is used to mask combination tones. Figure 2 gives examples of waveforms at the output of a simulated auditory filter in response to H and I tones (without the background noise, for clarity).

It is assumed that the pitch of such sounds corresponds to the reciprocal of most prominent time interval between peaks in the TFS close to adjacent envelope maxima [50]. For the two H tones (top), the most prominent time interval is 10 ms (1/F0). When the harmonics are shifted by 50 Hz (bottom left), the most prominent time interval is 9.5 ms, while when the shift is 25 Hz (bottom right) the most prominent interval is 9.75 ms. In the TFS1 test, a two-interval 2AFC method is used. In one interval of a trial (selected randomly), there are four successive bursts of tone H. In the other interval, tones H and I alternate, giving the pattern HIHI. The task is to choose the interval in which the sound changes across the four tone bursts. The frequency shift is adaptively varied to find the smallest detectable shift. This task has been used to assess the effects of hearing loss and age on sensitivity to TFS [51,52].

Binaural sensitivity to TFS can be assessed using the *TFS_LF* option [53]. This measures the ability to detect a shift in the interaural phase of a sinusoid with fixed frequency. The structure of this task is similar to that for the TFS1 test. In one interval the interaural phase difference (IPD) of all four tone bursts is 0. In the other interval, the IPD alternates between 0 and φ. The task is to pick the interval in which the sound appears to change in some way, for example to move within the head. The value of φ is adaptively varied to determine the threshold. This task has been used in many studies to assess the effects of age and hearing loss on binaural sensitivity to TFS. For a meta-analysis, see Füllgrabe and Moore [54].

One problem with the TFS-LF test is that some subjects are completely unable to perform the test using the selected test frequency (often 500 Hz). A variation on the test that can be performed by most subjects is the TFS-AF test [55,56]. This is similar to the TFS-LF test except that φ is fixed (usually at 180°) and the frequency is adaptively varied to determine the threshold. The frequency starts at a low value, such as 200 Hz. Most subjects can perform the task well when the frequency is low, but performance worsens when the frequency increases, and above some frequency the task becomes impossible [57,58]. Hence, the highest frequency at which the task can be performed gives a measure of binaural sensitivity to TFS. As for the TFS-LF test, performance worsens with increasing hearing loss and with increasing age [56,59].

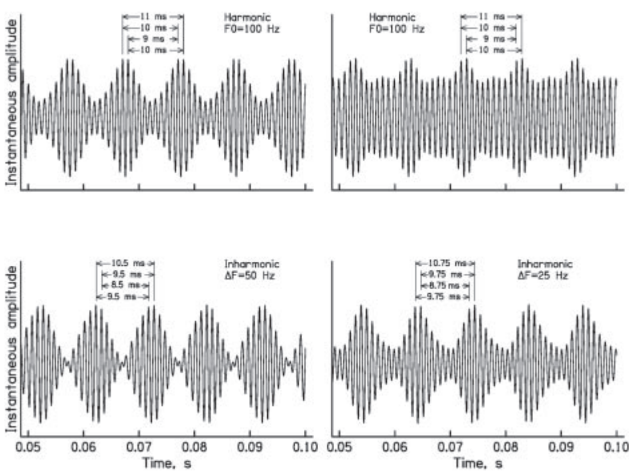

Fig. 2 Waveforms at the output of a simulated auditory filter centred at 1,000 Hz evoked by H tones (top) and I tones (bottom) with shifts of 50 Hz (left) and 25 Hz (right).

6. CONCLUDING REMARKS

The experiments implemented in the *PSYCHOACOUSTICS* package have been used successfully in many laboratories. We hope that the package will be widely used by researchers as a convenient and time-efficient method for running experiments. We also hope that the package will be used by teachers and students to demonstrate and try out experiments. The package is not in final form: the implementation of additional modules is ongoing. We encourage suggestions from fellow researchers as to modules that should be added. We also encourage suggestions about modifications and extensions to existing modules.

The software is currently being debugged. When this is finished, the software will be available from https://www.psychol.cam.ac.uk/hearing and also from http://ia.amu.edu.pl/psychoacoustics/.

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