The Calibration and Setup of the NCVS Dosimeter

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This memo is intended to document the technical procedures used by the NCVS researchers when measuring vocal doses with the NCVS Dosimeter (a Pocket PC-based voice dosimeter). The equipment settings, the calibration recordings and the analyses needed to obtain the pieces of information required by the dosimeter software are described. Information is provided on the dosimeter setup and on downloading the dosimeter software to the NCVS Dosimeter. Finally, the analyses for obtaining the relationship allowing estimation of sound pressure levels from the skin acceleration levels (SAL) are described. Functional examples of the Matlab scripts and of the calibration recordings can be downloaded at http://www.ncvs.org/ncvs/library/tech.

Keywords: vocal dose measurement, skin vibration, accelerometer, sound pressure level (SPL), pocket PC, voice dosimeter, NCVS Dosimeter

1. Introduction

Measurement of the amount of talking has been challenging both theoretically and practically. Various vocal doses have been defined measuring such items as the voicing time, number of oscillatory cycles of the vocal folds, the distance traveled by the vocal folds, and the heat dissipated in the vocal fold tissue [7;10] in order to quantify the long-term vocal load. A pocket-PC-based voice dosimeter (hereafter referred to as the NCVS Dosimeter) was developed by the researchers of the National Center for Voice and Speech (NCVS) in order to measure the amount of talking and vocal rest for the course of the whole day [4;5;8]. The device uses an accelerometer attached to the skin of the neck to pick up the person’s voice, and all the necessary values (fundamental frequency, SPL, etc.) are derived from the skin vibration of the neck. This memo is intended to provide information on the procedures related to the use of the device.

In order to measure the vocal doses in a particular person using the NCVS Dosimeter, the device should be adjusted (using software) to fit the person’s voice and skin vibration characteristics. Particularly, a) the phonation threshold level in dB shall be set so that the signal below that level will be recognized as unvoiced and above as voiced, b) the fundamental frequency (F0) extraction range needs to be determined by finding out the lowest and highest F0s the subject can produce, and c) the “artifact frequency” should be specified to allow the
dosimeter to recognize skin vibration artifacts unrelated to vocalization (swallowing, running, walking and other physical activities, unconscious hitting of the accelerometer, etc.). Furthermore, in order to determine how loudly the subject speaks during the day, the skin acceleration levels (SALs) registered with an accelerometer placed at the neck should be related to the sound pressure levels (SPLs) registered with a sound level meter.

Before the voice dosimetry is started, the subject is asked to come for a “calibration session,” which involves sets of laboratory measurements aimed at obtaining the above-mentioned pieces of information.

2. The successive steps

The procedure of voice dosimetry using the NCVS Dosimeter consists of several steps. The relevant steps are listed below in chronological order (the steps described in this memo are marked in bold):

- **equipment preparation** (before the subject arrives for the calibration session)
- the subject arrives
- the necessary documentation is filled out with the subject (the consent form, IRB documents, etc.) and the subject is explained the purpose of the study and the calibration procedure
- the accelerometer is attached to the neck of the subject
- the subject is seated in the recording room (sound booth)
- **recording C1**: two-step calibration for absolute SPL measurement with a head-mounted microphone
- **recording C2**: comfortable, soft and loud reading of the Rainbow passage and of the Marvin Williams passage for determining the SPL/SAL relationship in speech
- **recording C3**: soft phonation tests and a few sentences of comfortable reading for determining the SALs of the softest phonations and the frequency range
- the recordings are finished, the subject leaves the recording booth
- while one researcher is explaining the functioning of the NCVS Dosimeter to the subject, the second researcher performs analysis of the recording C3
- the **NCVS Dosimeter** is setup based on the results of the analysis of the recording C3
- **the NCVS Dosimeter software is downloaded to the pocket PC**
- the NCVS Dosimeter use is practiced by the subject (i.e., cabling, switching on, starting the application, dosimeter tests, switching off the application, un-cabling)
- the accelerometer is detached from the neck of the subject
- the NCVS Dosimeter is handed to the subject together with all the necessary materials and equipment

- the SAL/SPL calibration relationship for estimating the SPLs from the accelerometer signal is determined by analyzing the recordings C1 and C2

- after a day of wearing the voice dosimeter, the data files are collected and downloaded from the NCVS Dosimeter to a PC

- the SAL/SPL calibration relationship is entered in the Matlab script (e.g., dosi2_7_read6.m) for NCVS Dosimeter data extraction so that the skin acceleration levels (SAL) recorded by the NCVS Dosimeter can be converted into sound pressure levels (SPL)

- the NCVS Dosimeter data files are extracted and processed to obtain the vocal doses

### 3. The software and files needed to complete all the steps given above

**Four audio recordings (WAV files):**

- SPL calibration recording for obtaining absolute SPL values of subject’s voice (e.g., MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV, available with this memo)

- Two synchronized recordings of subject’s speech (e.g., Rainbow passage, Marvin Williams Passage). These two recordings provide a means of determining the relationship between SPL and SAL so that the SPL can later be estimated from the skin vibration of the neck:
  - Microphone recording (here done on DAT recorder, digitally transferred to PC; e.g., MT07_12_12_2003_9_06am_dataset.wav, available with this memo)
  - Recording of subject’s skin vibration registered with an accelerometer (this shall be recorded on a Pocket PC, with the same equipment as used in the field voice dosimetry; e.g., MT07_ppc_12_12_2003_9_06am_dataset.wav, available with this memo)

- Recording of subject’s skin vibration during the softest phonations and during reading (recorded with a Pocket PC). This recording is used to adjust the phonation threshold levels so that the signal with values above the threshold will be detected as voiced and below as unvoiced. The reading signal is used to adjust the range of fundamental frequencies for F0 extraction and to set the artifact frequency for recognizing accelerometer artifacts not related to vocalization (e.g., MT07_ppc_12_12_2003_9_07am_softR.wav, available with this memo).
Pocket PC to Computer synchronization software:
- Microsoft ActiveSync 3.7.1 or equivalent (usually provided with the pocket PC, free download at [http://www.microsoft.com](http://www.microsoft.com))

NCVS Dosimeter Application Software:
- **Dosimeter2.7final.zip** (not provided with this memo, contact the NCVS team to discuss potential distribution)

MS eMbedded Visual Tools software, Version 3.0 – 2002 Ed.

Matlab Software
- for performing analysis of the recorded signals (scripts provided as described below but the Matlab software is not provided with this memo, see [http://www.mathworks.com](http://www.mathworks.com))

Matlab Scripts:

Set 1 (for all analyses)
- **Script1a_SALxSPLcalibrationR4.m**: the first script for obtaining the SAL-SPL relationship (provided with this memo)
- **Script1b_SALxSPLcalibrationR4.m**: the second script for obtaining the SAL-SPL relationship (provided with this memo)
- **Script2_SALxSPLcalibrationR4.m**: the third script for obtaining the SAL-SPL relationship (provided with this memo)
- **Script3_SALxSPLcalibrationR4.m**: the fourth script for obtaining the SAL-SPL relationship (provided with this memo)
- **Script4_SALxSPLcalibrationR4.m**: the fifth script for obtaining the SAL-SPL relationship (provided with this memo)
- **Script5_SALxSPLcalibrationR4.m**: the script for exporting the SAL-SPL calibration values (provided with this memo)
- **dosi2_7_read6.m**: the script for extracting the data collected by the NCVS Dosimeter (not provided with this memo)
- **row.m**: script that is needed by nearly all other scripts (provided with this memo). It could be placed in a folder with all the other scripts or in a general directory, the path of which is used by Matlab.
- *Jan_FFTpeak_picking23.m*: the script for analyzing the accelerometer signal giving the SAL, F0 and FEC (spectral centroid) values that are similar to those given by the NCVS Dosimeter, no user editing needed (provided with this memo)

Set 2 (optional, as described later in Section 7, option 2)

- *GetThresholds.m*: a GUI-based script that is used to streamline the acquisition of thresholds needed to customize the NCVS Dosimeter to an individual (for analyzing the accelerometer signal giving the SAL, F0 and FEC [spectral centroid] values that are similar to those given by the NCVS Dosimeter) (provided with this memo)

- *getFoLimits2.m*: the function script used by *GetThresholds.m* to pick frequency ranges, no user editing needed (provided with this memo)

- *gettrim.m*: the function script used by *GetThresholds.m* to pick a region for analysis, no user editing needed (provided with this memo)

- *Jan_FFTpeak_picking23e.m*: the script used by *GetThresholds.m* to analyze data using the potential values of thresholds, no user editing needed (provided with this memo)

**Sound (Speech) Processing Software:**

- for determining the instances of impulses in order to synchronize the pocket PC and DAT recordings. For this purpose Matlab can be used by using the plot and zoom features, or commercially available packages may be used; e.g., Multispeech by Kay Elemetrics (www.kayelemetrics.com), GoldWave (recommended) (www.goldwave.com), or ProTools (www.digidesign.com) (not provided).

- to perform narrow-band spectrographic analysis to find out the range of F0 for a given subject (optional for Section 7, Option 1). For this purpose, the software such as CSL or Multispeech by Kay Elemetrics (www.kayelemetrics.com) or VoceVista: Visual Feedback for Instruction in Singing (www.vocevista.com) can be used (neither are provided with this memo) Note that Section 7, Option 2 does not require these.

### 4. Equipment Setup

The equipment setup for the SAL/SPL calibration recording is displayed in Figure 4.1. The detailed list of the concrete equipment can be found in the paper [9].

#### 4.1. Pocket PC Settings:

The set-up for the NCVS Dosimeter (Compaq iPAQ, model 3765, modified with an external accelerometer input [3;5;8]) shall be done in the following way:

- **Time**
  1. From the "Start" menu (see Appendix A), select "Settings" (third from bottom).
  2. Select the "System" tab.
3. Tap on the Clock symbol.
4. Set Time Zone, Time and Date for "Home"
5. Tap "ok" at the top right of the screen to exit this window.

- Microphone AGC (disable)
  1. From the "Start" menu (see Appendix A), select "Settings" (third from bottom).
  2. Select the "System" tab.
  3. Tap on the Microphone AGC symbol.
  4. Select the radio button labeled "Disable".
  5. Tap "ok" at the top right of the screen to exit this window.

- Voice recording format
  1. From the START menu (see Appendix A), select "Settings" (third from bottom).
  2. Tap on the up arrow (▲) at the bottom right of the screen.
  3. Select "Options..." at the top of the pop-up menu.
  4. Select the "Options" tab (third tab, near bottom of screen)
  5. The highlighted window of the drop-down menu under the heading "Voice recording format:" should read "11.025 Hz, 16 Bit, Mono (22 KB/s)". If it doesn't, tap on the down arrow to the right of the drop-down menu window and select that option.
  6. Tap "ok" at the top right of the screen to exit this window.

![Figure 4.1: The equipment setup. (Ch 1 – left channel, Ch 2 – right channel).]
• **Sound Notifications (disable to prevent recording artifacts)**
  1. From the "Start" menu (see Appendix A), select "Settings" (third from bottom).
  2. Select the "Personal" tab.
  3. Tap on the "Sounds and Notifications" symbol.
  4. Make sure all boxes are unchecked.
  5. Tap "ok" at the top right of the screen to exit this window.

  Note that the setting from these last two steps are also needed when the device is in the field, however they are automatically set by the ‘Dosimeter’ software as described later.

### 4.2. Sound Level Meter Settings:

Before starting the recordings, the batteries in the sound level meter should be checked. The signal cable shall be attached to the Aux1 output of the Sound Level Meter. The following setup of the Sound Level Meter (Brüel&Kjaer 2238) has been used for the calibration recordings:

<table>
<thead>
<tr>
<th>Output</th>
<th>Aux 1 (AC)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Range</td>
<td>30 – 110 dB (General)</td>
</tr>
<tr>
<td>Freq. Weighting</td>
<td>C</td>
</tr>
<tr>
<td>Detector 1</td>
<td>Bandwidth: BB/F, Frequency Weighting: C</td>
</tr>
</tbody>
</table>

### 4.3. DAT Recorder Level Settings:

Adjust the DAT recording levels for the head-mounted microphone and the sound level meter signals. Optimally, the loud phonations are used for this purpose and the levels are adjusted so that the loudest phonations levels are close to the upper limit of the dynamic range of the DAT recorder. For the sound level meter channel, it is important to check that the calibration tone delivered by the calibrator is not clipped in the DAT recording.

### 4.4. Reading Material:

Two reading passages have been used in order to determine the relationship between the skin vibration of the neck and the radiated sound pressure levels. The first one is the beginning of the Rainbow passage, the authorship of which is attributed to Fairbanks [2]. This reading passage was designed to include all speech sounds of the English language. The second reading passage is the Marvin Williams passage, which is all-voiced (i.e., it does not contain any unvoiced consonants, only vowels and voiced consonants). The author of this passage is not known to the authors of this memo. Its text has been taken from the book of Sataloff [6]. The text of the two passages is given below.
The Rainbow Passage (the beginning portion only):

When the sunlight strikes raindrops in the air, they act like a prism and form a rainbow. The rainbow is a division of white light into many beautiful colors. These take the shape of a long round arch, with its path high above, and its two ends apparently beyond the horizon. There is, according to legend, a boiling pot of gold at one end. People look but no one ever finds it. When a man looks for something beyond his reach, his friends say he is looking for the pot of gold at the end of the rainbow.

Marvin Williams:

Marvin Williams is only nine. Marvin lives with his mother on Monroe Avenue in Vernon Valley. Marvin loves all movies, even eerie ones with evil villains in them. Whenever a new movie is in the area, Marvin is usually an early arrival. Nearly every evening Marvin is in row one, along the aisle.

5. Accelerometer Attachment

The procedure of attaching the accelerometer to the skin has been described in detail in the publication [4]. The reader is referred to that publication for more information. Correct attachment of the accelerometer to the skin of the neck is important for obtaining reliable results.

6. The Recording Procedure:

The purpose of the recording procedure is twofold: a) to find out how the skin acceleration levels of the subject relate to the absolute sound pressure levels recorded by the sound level meter; and b) to determine the softest phonation levels and the frequency range of the subject.

A head-mounted microphone (omnidirectional B3 Lavalier microphone by Countryman Associates) is used to register the speech simultaneously with the sound level meter. The head-mounted microphone signal is less influenced by the ambient noise than the sound level meter and is used to obtain the SPLs of the running speech. In order to measure the absolute SPLs of speech (in dB re 20 µPa at 30 cm distance), the head-mounted microphone signal is related to the sound level meter (SLM) signal through a two-step calibration procedure. This procedure is basically similar to the one described in [7] (although the sound level meter distance used for this experiment is 30 cm in contrast to the 50 cm used in the previous article). The head-mounted microphone shall be placed at about 5-10 cm distance from the mouth, slightly to the side so that it is out of the air-stream. The exact distance of the head-mounted microphone from the mouth is not critical (as the head-mounted microphone is omni-directional and its signal is related to the sound level meter signal at 30 cm distance) but once set the distance shall be kept constant throughout the whole recording.
The steps of the recording are as follows:

1) Place the subject in the sound booth.
2) Check the settings of the pocket PC and of the sound level meter (as described in section 3).
3) Place the head-mounted microphone (HMM) on the subject’s head (mouth-to-mic distance about 5-10 cm).
4) Place the Sound Level Meter (SLM) to the distance of 30 cm from the mouth.
5) Prepare the texts of the Rainbow Passage and of the Marvin Williams passage for the subject to read.
6) Connect the HMM to the 1st channel and the SLM to the 2nd channel of the DAT recorder and check the appropriate levels.
7) Explain the course of the two-step calibration procedure to the subject.

**Recording C1: the two-step SPL calibration procedure**

8) Start the DAT recording.
9) Announce the date, time and the subject code on the recording.
10) Put the sound level calibrator on the sound level meter and play the calibration tone for several seconds. Notice the dB level on the sound level meter display. The level should be equal to the value provided in the calibrator specs (94 dB for the Sound Level Meter Calibrator Brüel&Kjaer 4231 used at the NCVS) and stable throughout the tone duration.
11) Remove the calibrator from the sound level meter and announce the dB level measured by the sound level meter in the sound level meter microphone.
12) Check the 30 cm distance between the subject’s lips and the sound level meter and have the subject say sustained /a/ at comfortable pitch and loudness, as steady as possible.
13) This concludes the two-step calibration. Stop the DAT recording.

**Setup for the Recording C2:**

14) Explain the procedure which will follow to the subject.
15) Make sure that the accelerometer bias circuit (1.5 V power box in Figure 4.1) has a fresh 1.5 V battery in.
16) Connect the accelerometer bias circuit (power supply) to the 2nd channel of the DAT recorder (accelerometer signal replacing the SLM signal).
17) Make sure the attenuator is connected to the accelerometer cable.
18) Connect the free end of the attenuator to the Mini XLR signal splitter.
19) Connect the first free end of the splitter to the Pocket PC and the other free end to the accelerometer bias circuit, so that the same accelerometer signal goes both to the Pocket PC (straight) as well as to the DAT (through the bias circuit). (The attenuator is between the accelerometer and the splitter).
20) Display the pocket PC Recorder program by pushing the button on the upper left side of the pocket PC (the button with the tape icon).
Recording C2: SAL versus SPL in speech

21) Start simultaneously the recordings on both the DAT recorder and the pocket PC (PPC).
22) Flick the accelerometer several times in order to produce sharp impulses in the accelerometer signal (these will be used later to synchronize the beginning of the PPC and DAT recording).
23) While the recordings are running, replace the accelerometer signal on the DAT again with the SLM signal.
24) While the recordings are running, disconnect the splitter from the accelerometer and connect the accelerometer through the attenuator directly to the PPC.
25) Let the subject read the Rainbow passage in normal, soft and loud voice.
26) Let the subject read the Marvin Williams passage in normal, soft and loud voice.
27) Let the subject do the series of soft phonation tests (sustained /i/ at comfortable pitch as soft as possible, soft glide from low to high pitch, high-pitched /hi-hi-hi-hi-hi/, “Happy Birthday” singing softly at high pitch).
28) With the recordings still running, put the splitter back between the attenuator and the pocket PC (the accelerometer signal goes through the attenuator to the splitter, one splitter output goes to the PPC, the second one goes to the 1.5 V power box).
29) Disconnect the sound level meter cable from the 2nd channel of the DAT recorder and replace it with the cable going from the 1.5 V power box (so that the accelerometer signal is recorded on the 2nd DAT channel, again replacing the SLM signal).
30) Flick the accelerometer several times to produce sharp impulses in the accelerometer signal (these will be used later to synchronize the ends of the PPC and DAT recordings).
31) Stop the DAT and PPC recordings.

Recording C3: Soft phonation tests and sample speech

32) Remove the splitter from the path between the attenuator and the Pocket PC (the accelerometer signal goes through the attenuator directly to the PPC).
33) Start recording on the PPC with the sound recorder.
34) Have the subject do again the series of the four soft phonation tests as in step 27 so that you can determine the softest phonation level from that recording later on.
35) Have the subject say a few sentences (e.g., reading a few sentences from the Marvin Williams passage) so you can determine the lowest possible frequency.
36) Stop the pocket PC recording.
37) This concludes the recording session of the calibration.

7. Analysis of the Recording C3: Determining the phonation threshold and the frequency range of the subject

In this section there are two optional methods to determine the phonation thresholds and frequency range for a subject. The first option is more detailed and should be referred to when
the user has questions. The second option uses the same methodology as the first option but eliminates the need of special software for narrow-band spectrographic analysis as everything is analyzed within the MATLAB environment. The second option is based on the methodology of the first option but it is more streamlined for determining the desired values. The second option can be used exclusively; the details of the underlying analysis are described only within the first option, however.

7.1. Pocket PC Settings:

**OPTION 1** (requires a third-party sound processing software)

1) Download the recordings from the pocket PC (wav files) to the PC and open the **Recording C3** in a sound processing software (e.g., Kay Elemetrics CSL or Multispeech or Voce Vista).

2) Perform a narrow-band spectrographic analysis of the recording to find out the frequency range of the subject (Figure 7.1).

Figure 7.1 shows the pocket-PC-recorded accelerometer signal of a male subject (MT07). The narrow-band spectrogram at the bottom allows for monitoring the voice frequency changes. The phonations begin with the soft sustained /i/ at the frequency of ca. 135 Hz, followed by the soft pitch glide starting at ca. 95 Hz and ending at ca. 275 Hz. Then there is the soft repetition /hee-hee-hee-hee-hee/ at ca. 280 Hz. The last part of the recording is the reading of the Marvin Williams passage in a comfortable voice. The lowest frequency achieved during the reading was around 85 Hz (determined manually by using a cursor in the Multispeech software).

![Figure 7.1. The Multispeech display of the Recording C3 recorded with the pocket PC. Top: the accelerometer signal. Bottom: the narrow-band spectrogram.](image-url)
Next, the same signal is analyzed with a Matlab script (*Jan_FFTpeak_picking23.m*), which has been set up to perform the analysis of the skin acceleration levels (SALs), fundamental frequencies (F0s) and the spectral centroids in about the same manner as the NCVS Dosimeter software does in the pocket PC.

To analyze the signal in Matlab, one should put the script *Jan_FFTpeak_picking23.m* in the same folder as the wav file for analysis. The script should be open from the Matlab program. When opened, the section “Parameters Setup” will be seen below the initial command lines. (In Matlab, the comments follow after the “%” sign and are in green color. The Matlab code is in black color. The parameters that should be given for each of the subjects separately were distinguished for the purpose of this memo by orange color.) The “Parameters Setup” section is reproduced below:

```
%SECTION PARAMETERS SETUP (added Feb 26, 2003):
filename='MT07_ppc_12_12_2003_9_07am_softR.wav'; %added Jan 20, 2003

%Read in wav file
[speech, fs]=wavread(filename);
speech=row(speech); %make sure it is a row vector

%Set start and stop time
start_time=0.00001; %start at the beginning
stop_time=16; %stop at the very end

%Set Soft Phonation threshold
spl_thresh=65; %dB

%Set the limits of the fundamental frequency in Hz
start_freq=80; %lowest F0 [Hz]
stop_freq=600; %highest F0 [Hz] maximum: 1000 Hz

%Correction when harmonic frequencies are extracted instead of the fundamental:
fo_correction=0; %0 - without correction; 1 - with correction

%Set the frame length in seconds
frame_len_s=0.03; %frame length in seconds [0.030 used as default value]
%for the frame length longer than 32 ms,...
%...only 3 middle buffers instead of 5 would be sufficient

%Set the lower limit for artifact correction
artifact_freq=105; %[Hz] recommended value: 105
%all of the spectral components with frequencies below this limit
%will be calculated for artifact recognition - if the energy below this
%limit is larger than the energy in the voicing range, the signal
%will be considered unvoiced
%if the artifact_freq is set lower than the lowest expected F0,
%then its value will automatically be replaced by the startfreq value
```

The following parameters shall be set-up within the section above:
The details for setting-up these parameters are described below.

File name for analysis:

First, the name of the file for analysis needs to be specified (parameter filename). The file name used here as an example will be 'MT07_ppc_12_12_2003_9_07am_softR.wav', which is the Recording C3 (pocket PC recording of the accelerometer signal of the soft phonations) for the male subject with code number MT07. (As the name reveals, the recording was done with the subject MT07 on 12/12/2003 at 9:07am).

Start and stop time:

Next, it should be decided whether the whole or only part of the signal will be analyzed. If one wants to analyze only part of the recording, the start_time and the stop_time would need to be given in seconds (for instance, for analysis of only the soft phonations in Recording C3 marked between the two blue lines in Figure 7.1, which occurs within the time interval from 1 to 16 seconds, the start_time would be set to the value of 1 and the stop_time to the value of 16).

Here, we will initially analyze the whole recording; thus we set the start_time=0.00001 and the stop_time=length(speech)/fs and comment out the other values for these parameters.

\[
\begin{align*}
\text{start\_time} & = 0.00001; & \text{\%start at the beginning} \\
\text{\%start\_time} & = 1; & \text{\%enter the start time in seconds} \\
\text{\%stop\_time} & = 16; & \text{\%enter the end time in seconds} \\
\text{stop\_time} & = \text{length(speech)}/fs; & \text{\%stop at the very end}
\end{align*}
\]

SPL threshold:

The next parameter is the phonation threshold spl_thresh. This is the parameter that needs to be determined. For initial analysis, however, it needs to be set to some preliminary value. A good initial choice is 65 dB, which was found to be about 10 dB above the electronic noise floor of the pocket PC when the accelerometer and attenuator are connected [5].

F0 range:

The narrow band spectral analysis done previously revealed the lowest F0 of about 85 Hz and the highest F0 of 280 Hz for the subject MT07. On the basis of this, the value for the low limit was chosen here to be 80 Hz (start_freq=80). The upper F0 limit (stop_freq) is less
critical for the correct F0 extraction than the lower F0 limit within this program and the default value of 1000 Hz was empirically found to work well for most of the cases. (The value of 1000 Hz is the maximal value that can be properly handled by the software). In the case of the male subject MT07, however, the frequencies of 1000 Hz were unlikely to be reached, so the upper limit was chosen to be 600 Hz (stop_freq=600).

F0 correction:

In some cases, the software could detect upper harmonic frequencies instead of the fundamental. If that happens, a simple F0 correction algorithm has been incorporated in the program to correct for these artifacts. Such artifacts are, however, rare when the accelerometer is attached properly to the skin as is usually the case with recordings done within calibration session. The recommended setting is fo_correction=0 (0 means correction disabled, 1 means correction enabled).

Frame duration:

The frame duration was chosen by the authors to be 0.03 s (frame_len_s=0.03). That means that the SAL, F0 and spectral centroid are determined every 30 ms. The value of 30 ms is small enough for detecting short unvoiced segments in speech, while simultaneously being still manageable by the processor of the pocket PC. The value of 0.03 should not be changed unless there is a specific reason for it.

Unvoiced artifact elimination:

When accidentally hitting the accelerometer (or during swallowing, running, etc.) the dB level of the accelerometer signal can exceed the dB level of the softest phonations. In order to correct for these artifacts, an algorithm has been implemented in the software, which compares the energies within the vocal frequency range to the energies outside the vocal frequency range. If the energy outside of the vocal frequency range is higher than the energy within the vocal frequency range, the signal is recognized as an artifact even when it exceeds the dB threshold. The vocal frequency range needs to be set up for each subject, however.

Generally when voiced, most of the accelerometer signal energy is expected to be within the range between the subject’s lowest fundamental frequency and 1000 Hz (the skin functions as a low-pass filter and thus little vocal energy is present in the harmonic frequencies above 1000 Hz [1]). The program automatically sets up the upper vocal limit to 1000 Hz. The lower vocal limit shall be determined by the user. This limit will be selected as the higher of the two values start_freq and artifact_freq. The recommended value for the artifact_freq (i.e., artifact frequency), which was empirically found to work well, is 105 Hz. However, in certain deep male voices in which the F0 is below 105 Hz and simultaneously the harmonic spectrum is poor, this value can cause some of the voiced passages to be incorrectly determined as unvoiced. In these cases, the artifact frequency value could be lowered below 105 Hz (the artifact elimination can be expected to be less effective in these cases, however). For the initial analysis, the artifact frequency will be set here to the recommended value of 105 Hz (artifact_freq=105).
Figure 7.2 shows the results of the analysis performed by the `Jan_FFTpeak_picking23.m` script with the parameters set to the values mentioned above. The first window gives the real signal, the second window offers the skin acceleration levels in dB, the third window shows the analyzed fundamental frequency, the fourth gives the spectral centroid (i.e., frequency energy center, FEC) and the last window is the voicing detection parameter $k_v$ (which has the value of 1 for voiced and 0 for unvoiced frames).

On the basis of this analysis, the phonation threshold can be determined. The phonation threshold shall be set about 5 dB lower than the level of the softest voice the subject is capable producing, and at least 5 dB above the noise level. If these conditions conflict the threshold is set as a compromise between these values.

The purple horizontal line in Figure 7.2b shows the level of 65 dB, which appears appropriate as a phonation threshold for this person and does not need to be changed. The F0 values in Figure 7.2c (green) correspond to those seen in the narrow-band spectrogram in Figure 7.1 and suggest that the frequency extraction is correct.

The last parameter that should be checked is the artifact frequency. For that, it is useful to look more closely at the part of the subject’s speech, which contains the lowest frequencies that could potentially be problematic. The speech section is between 17s and 25s of the recording. Let us set the Parameter Setup section of the Matlab script to
and run the script again. The script gives the Figure 7.3. Notice, that in some instances, the voiced signal is incorrectly considered unvoiced (see the arrows in Figure 7.3). This can be seen especially in the bottom window where the value goes to 0 within a voiced segment. This suggests that the artifact frequency 105 Hz is set too high and needs to be lowered.

Replacement of the value 105 Hz with the value of 90 Hz ($artifact\_freq=90$) and running the script again gives corrected results, which do not show the problem anymore (Figure 7.4).
OPTION 2

1) Download the recordings from the pocket PC (wav files) to the PC

2) Place files within the folder with Set 1 and Set 2 MATLAB scripts (i.e., those given on page 4-5) and from MATLAB, start the GetThreshold.m script

The MATLAB script GetThreshold.m asks for a file to analyze (Figure 7.5a). The file is then loaded automatically into memory and is played at twice the normal speed (for monitoring purposes). Figure 7.4 steps the options of the script. The final result is the list of values needed by the NCVS Dosimeter.

Figure 7.4:
The same display as Figure 7.3, but this time the voicing detection was corrected by setting the artifact frequency to a lower value.

Figure 7.5a:
Choose a file to analyze
Figure 7.5b: Select a region of the file to analyze

Figure 7.5c: Confirm selection
Figure 7.5d: From spectrogram, choose near highest fundamental frequency then the lowest fundamental frequency.

Figure 7.5e: From previous choices, a spectrally zoomed image is shown, and the highest and lowest fundamental frequencies can be found.
Figure 7.5f:
From the choices, some initial values are shown. Press ‘OK’ to begin trial.

Figure 7.5g:
Confirm the correctness of the analysis with the values from figure 7.5f.

Figure 7.5h:
Values are printed in the MATLAB window.

Figure 7.5. Finding values to customize the NCVS Dosimeter to an individual.
8. The NCVS Dosimeter Setup:

The NCVS dosimeter software is packed in a zip file (*Dosimeter2.7final.zip*). The latest version of the software is the version 2.7, which has been tested extensively and found reliable with the Pocket PC COMPAQ iPAQ Model 3765.

In order to install the NCVS Dosimeter software into the pocket PC, one needs to install MS eMbedded Visual Tools, Version Edition 3.0-2002 software on the personal computer first. The software can be freely downloaded from [http://msdn.microsoft.com/downloads/default.aspx](http://msdn.microsoft.com/downloads/default.aspx). The authors have successfully used the MS eMbedded Visual Tools version 3.0 for Windows 2000.

When the MS eMbedded Visual Tools software is installed on the PC, the *Dosimeter2.7final.zip* file needs to be unzipped. The unzipping procedure creates a folder containing all the necessary NCVS Dosimeter files. The NCVS Dosimeter project can be opened by clicking on the file *Dosimeter.vcw* (which is the project workspace file). The project opens
within the MS eMbedded Visual Tools (Figure 8.1). Then, in the window on the left, the file "DosiWave.h" (under Header Files) shall be selected by clicking on it.

After that, in the largest window on the right, the following section should be located:

```c
//Constants for the Dosimeter application, modify these for any changes needed
#define DOSI_USERID "MT07"
    //user id, MUST be 4 letters or numbers, CRITICAL or the Binary File read will fail when
    // analyzing data.,
    //1/9/03: Changed from 3 characters to 4 characters
#define DOSI_VER "2.7"  //updated 8/21/03 -- MUST BE 3 CHARACTERS ONLY!!!! for binary log
#define SPL_THRESH 65.0  //THE LEVEL THAT DETERMINES THAT SPLS BELOW THIS ARE JUST NOISE, NOT
    //voicing
#define START_FREQ 85 //Lowest F0 (HZ)
#define STOP_FREQ 600 //Highest F0 (HZ)
#define DOSI_GAIN 210000 // 332569 is used for new PPCs 7/2003, 210000 for NCVS PPCs
    //reinstated to use 5/31/02
#define ARTIFACT_FREQ 90 //recommended value is 105, lower limit for artifact correction
```

This section allows the user to adjust

- **the user ID** (DOSI_USERID "MT07"). It shall be a four letter code, similar to the MT07 (meaning 7th male test subject) used here

- **threshold SPL** (SPL_THRESH 65.0). This is the phonation threshold dB value determined earlier by analyzing the calibration *Recording C3* (set to 65 dB). (Although called SPL threshold here, it is in fact the SAL threshold).

- **low limit of F0** (START_FREQ 85)

- **upper limit of F0** (STOP_FREQ 600) Again these are the values determined previously by analyzing the calibration *Recording C3* (set here to 80 Hz and 600 Hz). It is useful to realize that the upper F0 limit shall consider the highest F0 the subject is capable of achieving – that limit could be higher than the highest F0s achieved during the calibration procedure (i.e., in the case of the subject MT07, the highest frequency seen was 280 Hz, but as the very upper extreme was not really tested, the upper limit was set to 600 Hz, to be safe).

- **the value** DOSI_GAIN 210000. This determines the constant used to measure the skin acceleration level in dB. The value 210000 was determined empirically for the Pocket PC COMPAQ iPAQ Model 3765 by having a test person produce phonation at comfortable pitch and loudness and making the skin acceleration level registered by the pocket PC equal to the absolute SPL@30 cm in dB re 20 µPa measured by the sound level meter. The value 210000 is used also in the Matlab code used for determining the SPL threshold and shall not be changed.

- **the artifact frequency** (ARTIFACT_FREQ 90). This should be equal to the artifact frequency determined by analyzing the calibration *Recording C3*. The recommended
default value is 105 Hz, but as seen earlier, the value for the subject MT07 needs to be set lower, to 90 Hz.

The rest of the parameters are not expected to be changed.

When all the parameters are set correctly, the exclamation mark in the menu (indicated by the black arrow in Figure 8.1 within the eMbedded application) shall be clicked on. This compiles the program and downloads it into the Pocket PC (the pocket PC needs to be connected to the PC at this time). The NCVS Dosimeter application will appear on the pocket PC display when done (Figure 8.2). At that point the NCVS Dosimeter is setup for use by the calibrated subject (MT07, in this case).

After the adjustment took place, it is useful to write down all information on the settings and equipment used and save it in an information file (for instance MT07_info.txt):

Calibration: subject MT07  
December 12, 2003, 9:00 am

Accelerometer: A15  
Attenuator: T12  
Pocket PC: D16

Note: The jugular notch is quite deep in this subject, which could potentially cause attachment problems

--------------
Dosimeter adjustment:  
SAL threshold: 65 dB  
F0 lower: 80 Hz  
F0 upper: 600 Hz  
Artifact freq: 90 Hz

--------------
Equipment handed out:  
Accelerometer: A15  
Attenuator: T12  
Dosimeter #D16  
Dosimeter #D19  
Battery pack B02
9. Determining the SAL Versus SPL Relationship

In order to estimate the sound pressure levels of speech from the skin vibration of the neck, the correlation between the SPL and SAL should be determined. The detailed description of how well the SPL and SAL values are correlated and of how accurately the SPL can be estimated from the SAL values was provided elsewhere [9] and will not be repeated here. The purpose of this section is to describe the analysis procedure step by step so that it can easily be repeated.

Before starting the analysis, the calibration Recordings C1 and C2 shall be transferred to a PC and saved as two WAV files. [The recordings can be transferred to a PC digitally, e.g., using the digital interface of the Kay Elemetrics Computerized Speech Lab (CSL) module]. The following files are needed for the analysis:

- A stereo WAV file of the calibration Recording C1 (the two-step calibration procedure) originally recorded on the DAT. The head-mounted microphone signal shall be in Channel 1 and the sound level meter in Channel 2. (An example of a C1 recording MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV of a subject MT07 is available together with the scripts).

- A stereo WAV file of the calibration Recording C2 (containing synchronization impulses at the beginning and the end plus the readings of the Rainbow and Marvin Williams passages in comfortable, soft and loud voice) originally recorded on the DAT. The head-mounted microphone shall be in Channel 1. Channel 2 will contain the accelerometer synchronization impulses at the beginning and the end of the recording and the sound level meter microphone signal in between (MT07_12_12_2003_9_06am_dataset.wav).

- A mono WAV file of the calibration Recording C2 originally recorded on the pocked PC. The recording contains the accelerometer signal of the readings of the Rainbow and Marvin Williams passages and the synchronization impulses at the beginning and at the end of the recording.
The following Matlab scripts are used for the analysis:

- `Script1a_SALxSPLcalibrationR4.m`
- `Script1b_SALxSPLcalibrationR4.m`
- `Script2_SALxSPLcalibrationR4.m`
- `Script3_SALxSPLcalibrationR4.m`
- `Script4_SALxSPLcalibrationR4.m`
- `Script5_SALxSPLcalibrationR4.m`

These scripts should be placed in the same directory as the wav files above. They need to be opened within the Matlab program. Their use is described below.

### 9.1. The `Script1a_SALxSPLcalibrationR4.m`:

This script is used to display the calibration **Recording C1**. The display is used to find out the start and stop times for

1) the stable SLM calibration tone (about 1-2 seconds duration)
2) the most stable part of the sustained /a/ phonation (1-2 seconds duration).

These two parts of the recording need to be selected in order to calibrate the head-mounted microphone signal to measure the absolute SPL@30 cm (in dB re 20 µPa).

The script should be opened within a Matlab program. The section for entering information is found at the beginning of the script:

```matlab
%------------------------------------
%Section for Entering information:
[x,fs_csl]=wavread('MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV');
%------------------------------------
```

In this section, the name of the WAV file shall be specified. The MT07 example C1 recording file `MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV` has been written here (highlighted in red color).

Running the script produces the following figure (Figure 9.1):
Figure 9.1:
Display of the calibration recording C1 containing the stable calibrator tone and the sustained /a/.
Top: the head-mounted microphone signal.
Bottom: the sound level meter signal.
(MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV)

The plot on the top displays the head-mounted microphone signal and the plot on the bottom displays the sound level meter (SLM) signal. The first event to locate is the steady calibrator tone, which can be found in the bottom plot (here starting at 15 s and ending at around 20.5 s). The second event is the sustained /a/ portion, which is seen in both plots (here located between 30 and 35 s). The goal is to find the steadiest parts of these two events (of ca.1-2 s duration), which will be used for the subsequent calculations.

The calibration tone is very steady between 15-20 s, so any 2-second interval can be selected from this portion. In order to find out the steadiest portion of the sustained /a/, it is useful to zoom in on that portion. Figure 9.2 shows the head mounted signal and the sound level meter signal of the sustained /a/ within the time interval of 30-35 s. The steadiest portion appears to be between 31.5 and 32.5 seconds.

Figure 9.2:
The portion of the calibration recording C1 containing only the sustained /a/.
9.2. The Script1b_SALxSPLcalibrationR4.m:

This script calculates and stores the calibration gains, which adjust the level of the head-mounted microphone signal to be equal to the absolute SPL levels at 30 cm distance. The information that shall be entered can be found at the beginning of the script:

```matlab
%---------------------------------
%Section for Entering information:
%
Enter subject code: (e.g., F002)  subject = 'MT07';

%Enter the filename containing calibration wav_filename = 'MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV';

%Give the sound level value of the SLM calibration signal: cal_dbvalue=94.0;   [%dB]

%Give the sound level meter distance in cm: SLM_distance = 30;   [%cm]

%Specify the gain used by the dosimeter software: gain=2.1e5;

%Enter the start and stop time of the stable SLM calibration tone (duration of 1-2 s): start_slm_cal = 18;   [%s] stop_slm_cal = 20;   [%s]

%Enter the start and stop time of the stable portion of sustained /a/ (duration of 1-2 s): start_sust = 31.5;   [%s] stop_sust = 32.5;   [%s]
%End section for entering information: %-------------------------------------
```

The example above provides the setting for the subject’s MT07 calibration Recording C1 (MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV), based on the information obtained before, i.e.

- The subject code (MT07)
- The file name of Recording C1 (the same one as analyzed with the previous script 1a, i.e., MT07_12_12_2003_09_00am_SLMcal_sust_a.WAV)
- The level of the steady calibration tone produced with the calibrator (94 dB)
- The distance of the sound level meter from the mouth of the subject (30 cm)
- The gain used by the NCVS Dosimeter software (as shown previously in the Dosimeter Setup section, the value is 2.1e5, or 210000. This value should not be changed as long as the Pocket PC COMPAQ iPAQ Model 3765 is used.)
- The start and stop time of the stable calibration tone (18 and 20 s)
- The start and stop time of the steadiest portion of the sustained /a/ phonation (31.5 and 32.5 s)

Running the script gives the Figure 9.3, which shows the selected section of the calibrator signal (top), the selected section of the sustained /a/ registered by the sound level meter (middle) and by
the head-mounted microphone (bottom). The calculated calibration values are stored in a file `MT07_calR4.mat`, which will be used by the subsequent scripts for analysis of the calibration recordings.

9.3. The *Script2_SALxSPLcalibrationR4.m*:

This script performs the synchronization of the pocket PC and DAT recordings (C2) based on the synchronization impulses recorded at the beginning and at the end. Furthermore, the script analyzes the pocket-PC signal (i.e., the accelerometer signal) and obtains and stores the skin acceleration levels, F0 and spectral centroid values in the 30-ms time intervals. It also distinguishes the voiced and unvoiced speech segments. Before the analysis can be done, the script requires entering certain pieces of information, which can be found in the following *Section for entering information*:

```matlab
%---------------------------------
%Section for Entering information:
%Enter subject code: (e.g., F002)
subsject = 'MT07';

%Enter the name of the PPC file with the Rainbow and Marvin Williams passages: (e.g., 'Recording1.wav')
filename_ppc_speech = 'MT07_ppc_12_12_2003_9_06am_dataset.wav'

%Enter the name of the CSL file with the Rainbow and Marvin Williams passages: (e.g., 'F002_dataset.WAV')
filename_csl_speech = 'MT07_12_12_2003_9_06am_dataset.wav'
```
% Soft Phonation threshold (for accelerometer signal)
sal_thresh=65; %dB

% Set the limits of the fundamental frequency in Hz
start_freq=80; % lowest F0 [Hz]
stop_freq=600; % highest F0 [Hz]

% Specify whether to perform isolated voiced frame elimination
% (if yes an isolated voiced frame will be made unvoiced)
isol_correction=1; % 0 - no elimination; 1 - with elimination

% Correction when harmonic frequencies are extracted instead of the fundamental:
fo_correction=0; % 0 - without correction; 1 - with correction

% Set the frame length in seconds
FrameDur=0.03; % frame length in seconds [0.030 used as default value]
% (smallest length allowed is 0.019 ms, for 5 buffers)
% if the frame length would be larger than 32 ms,...
% ...only 3 middle buffers instead of 5 would be sufficient

% Set the lower limit for artifact correction (e.g., due to movement-induced accelerometer noise)
artifact_freq=90; % [Hz] recommended value: 105
% energies of all the spectral components with frequencies below this limit
% will be calculated for artifact recognition - if the energy below this
% limit will be larger than the energy in the voicing range the signal
% will be considered unvoiced
% if the artifact_freq is set lower than lowest expected F0, then its value will
% automatically be replaced by the start_freq value

% Use CSL/Multispeech and load the PPC file and CSL file in two windows above each other
% The CSL file - display the 2nd channel which contains the CSL synchronization impulses
% Enter instants of three selected synchronization pulses at the beginning:
% PPC signal:
ppc_start_pulse_1 = 8.19795;
ppc_start_pulse_2 = 7.80731;
ppc_start_pulse_3 = 7.40452;
% CSL signal (2nd Channel):
csl_start_pulse_1 = 8.12337;
csl_start_pulse_2 = 7.73275;
csl_start_pulse_3 = 7.32989;

% Enter instants of three selected synchronization pulses at the end:
% PPC signal:
ppc_end_pulse_1 = 191.53363;
ppc_end_pulse_2 = 191.86798;
ppc_end_pulse_3 = 193.28002;
% CSL signal (2nd Channel):
csl_end_pulse_1 = 191.47255;
csl_end_pulse_2 = 191.80688;
csl_end_pulse_3 = 193.21906;

% Enter beginning of the Rainbow passage in the CSL recording
csl_start_Rainbow = 27; % [s]

% Enter end of the Marvin Williams passage in the CSL recording
csl_end_Marvin = 152; % [s]

% End section for entering information:
The information required is:

- The subject code (MT07).
- The file name of the Recording C2 recorded on the pocket PC (e.g., MT07_ppc_12_12_2003_9_06am_dataset.wav).
- The file name of the Recording C2 recorded on the DAT (e.g., MT07_12_12_2003_9_06am_dataset.wav).
- The soft phonation threshold (this is the dB level determined previously in Section 7 from the softest phonations, i.e., 65 dB).
- The lower and upper limits for F0 extraction (these were also determined previously in Section 7 to be 80 and 600 Hz for this subject).
- The parameter isol_correction is used for elimination of isolated voiced frames. A voicing segment is expected to have duration longer than 30 ms (such a short segment would mean, e.g., only three vibratory cycles of the vocal folds at 100 Hz). In case of detecting only such a short segment, it is most likely an artifact. By entering isol_correction=1, the algorithm eliminates such artifacts; by entering isol_correction=0, the algorithm will not perform elimination of these segments.
- The function of the parameter fo_correction has already been described in Section 7; it can be used when harmonic frequencies are extracted instead of the fundamental frequency (fo_correction=1). The default configuration disables the correction (fo_correction=0) as these errors are found rather rarely if the accelerometer is connected properly to the skin of the neck.
- The frame length of 0.03 s (FrameDur=0.03) specifies the 30 ms frames as discussed previously. The value of 0.03 s is not expected to be changed under usual circumstances.
- The artifact frequency has also been described in Section 7. The appropriate value for the subject MT07 was found to be 90 Hz (artifact_freq=90). The default value is 105 Hz.
- The instances of the synchronization impulses at the beginning and end of the recording, as well as the times of the beginning of the reading of the Rainbow passage and the end of the reading of the Marvin Williams passage, need to be specified. The process how to do it will be described next.

Note: The DAT recording is called in the script “the CSL recording” as it was transferred into the PC using the CSL system (by Kay Elemetrics).

Determining the instances of the synchronization impulses:

The instances of the synchronization impulses are best determined in a sound-processing software. Here, the software Multispeech (by Kay Elemetrics Corp) will be used as an example.

- Open the Multispeech software and load the pocket-PC-recorded speech waveform of the Recording C2 (e.g., MT07_ppc_12_12_2003_9_06am_dataset.wav) in the A window and the
DAT recorded speech waveform (e.g., MT07_12_12_2003_9_06am_dataset.wav) in the B window.

- In B window, display the 2nd channel, which contains the synchronization impulses.

The synchronization impulses are well visible in the pocket-PC signal in the top window of Figure 9.4 (the red signal at the beginning and at the end).

- Zoom in on the beginning of the recordings to see the individual synchronization impulses (View/Selected Data).
- Scale the selected part of the DAT recording (bottom plot) by a factor of ca. 10-50 to amplify the synchronization impulses.
- Find out how the synchronization impulses in the top plot correspond to those in the bottom plot.
- Select one particular impulse (for instance, the last one, as done in Figure 9.4) and zoom on it in both the plots.
- Determine the exact time instance of the impulse in the two recordings (Figure 9.5).
- Repeat the procedure and determine the exact time instances for two more corresponding impulses from the beginning of the recording.
- In a similar way, determine the time instances for three impulses from the end of the recording.
- Enter the time instances in the script.

Note: Use the well defined impulses. The most important part of the impulse is the onset rather than the peak (which could be clipped). If the onset is not abrupt enough, try to find another impulse that is abrupt. In this way the synchronization can be done more accurately. Three impulses (rather than one) are used, so that the errors of manual readings can be averaged out.
Figure 9.4:
The calibration recording C2 displayed in Multispeech. Top – the signal recorded with the pocket PC. There are the three readings of the Rainbow passage (normal, soft and loud) followed by the three readings of the Marvin Williams passage (normal, soft and loud). Normally, soft phonation tests would follow after the Marvin Williams passage, these were omitted here, however, for the sake of a smaller file size. Bottom – the 2nd channel of the signal recorded on the DAT recorder. The blue lines show the end of the intervals selected for displaying the synchronization impulses at the beginning.

Figure 9.5:
The section showing the synchronization impulses at the beginning of the recording as recorded in the pocket PC signal (top) and in the DAT signal (bottom). The last synchronization impulse is selected in both the plots (the impulse between the two blue lines).
Figure 9.6: The detail of the last synchronization impulse in the pocket-PC signal (top) and in the DAT signal (bottom). The cursor line is positioned at the instant of the beginning of the impulse to find out the time at which the impulse happened in the two recordings. The corresponding times are 8.19795 s (top) and 8.12337 s (bottom).

Determining the instances of the beginning of the Rainbow passage and of the end of the Marvin Williams passage:

The last two numbers that shall be entered in the `Script2_SALxSPLcalibrationR4.m` before it can be run is 1) the time instance of the beginning of the Rainbow passage reading and 2) the instance at which the Marvin Williams passage readings ended. These two numbers are only approximate and will be refined later on. The script expects the values to be obtained from the digitized DAT recording (bottom plot of the Figure 9.4), not the pocket PC recording. Leave about 1-2 seconds of silence before and after the readings.

Running the script and checking the synchronization accuracy:

After all the values have been entered, the script can be run. The script will analyze the Pocket PC recording and produce a figure (Figure 9.7) displaying the running values of the SAL, F0 and spectral centroid. Only the section of the Rainbow and Marvin Williams passage readings will be selected and analyzed (the time interval determined by the two last values entered in the
Section for Entering information of the script. The script stores the analyzed values in the file MT07_calR4.mat.

Figure 9.7: Matlab results after running the Script2_SALxSPLcalibrationR4.m: Analysis of the normal, soft and loud readings of the Rainbow passage, followed by the normal, soft and loud readings of the Marvin Williams passage. The top plot shows the skin acceleration levels (SAL) in dB (only the levels of the voiced segments are displayed, the level of the unvoiced segments was made equal to zero). Notice the level differences between the normal, soft and loud readings. The bottom plot shows the fundamental frequencies (blue dots) and spectral centroid (FEC, green dots) values obtained from the accelerometer signal. Note that the FEC values are highest in loud voice and lowest in soft voice. A small number of artifacts can be seen in the F0 values (the blue outliers). The resolution of the F0 and FEC values is ca. 10 Hz.

The figure produced by the script (Figure 9.7) will be used in the following script to locate the segments of the normal, soft and loud readings of the Rainbow passage and of the Marvin Williams passage.

The script also calculates the accuracy of the synchronization. The difference between the time instance of the same impulse in the pocket PC signal and in the digitized DAT signal is calculated for the three impulses at the beginning as well as for the three selected impulses at the end of the recording. The variability of these three differences provides information on the accuracy of the synchronization. The program takes the minimum and maximum of the three differences, calculates the difference between these and displays this value in the main Matlab window. The value should be smaller than 1 ms in order to relate the pocket PC and DAT recordings accurately.
In the case of the data entered for the subject MT01, the program displays the following information in the Matlab window:

```
filename_ppc_speech =
MT07_ppc_12_12_2003_9_06am_dataset.wav
filename_csl_speech =
MT07_12_12_2003_9_06am_dataset.wav
accuracy_start_sync =
7.0000e-005
accuracy_end_sync =
1.4000e-004
```

This says that the accuracy of the synchronization at the beginning is 0.07 ms (7e-5 s) and at the end it is 0.14 ms (1.4e-4). Both of the values are smaller than the 1 ms, which means the synchronization is reasonably accurate.

### 9.4. The Script3_SALxSPLcalibrationR4.m:

This script locates the normal, soft and loud readings of the Rainbow and Marvin Williams passages and calculates the running absolute SPL values (in dB@30 cm re 20 µPa) from the head-mounted microphone signal based on the calibration gains calculated by the Script 1b.

Before running the script, the subject code number (e.g., MT07), and the beginning and end of each of the readings of the Rainbow and Marvin Williams passages shall be entered in the section for entering information:

```matlab
%---------------------------------
%Section for Entering information:
%Enter subject code: (e.g., F002)
subject = 'MT07';

%Enter times for the normal, soft and loud Rainbow passage
start_normal_Rainbow = 28; %[s]
end_normal_Rainbow = 51; %[s]
start_soft_Rainbow = 52; %[s]
end_soft_Rainbow = 76; %[s]
start_loud_Rainbow = 77; %[s]
end_loud_Rainbow = 100; %[s]

%Enter times for the normal, soft and loud Marvin Williams passage
start_normal_Marvin = 102; %[s]
end_normal_Marvin = 117; %[s]
start_soft_Marvin = 119; %[s]
end_soft_Marvin = 135; %[s]
start_loud_Marvin = 135; %[s]
end_loud_Marvin = 151; %[s]

%End section for entering information:
%----------------------------------
```
The time instances of the beginning and end of each of the readings can be determined from the Figure 9.7. The zoom in tool is helpful for this purpose (the arrow in Figure 9.7). The closest whole-second instant before the beginning and after the end of the reading are selected (no decimal places – see Figure 9.8 for an example). Running the script after the values are entered produces Figure 9.9 displaying the running absolute SPL values for all the readings.

Figure 9.8.
The zoomed-in portion of the bottom plot of the Figure 14 showing the end of the comfortable reading and the beginning of the soft reading of the Rainbow passage. This view is used for determining the time instances to be entered in the script 3. The closest whole-second instant (no decimal places) after the end of the reading and before the beginning of the reading is selected. In this case, the values to be selected are
\[ \text{end\_normal\_Rainbow} = 51 \] and
\[ \text{start\_soft\_Rainbow} = 52 \].

Figure 9.9:
The absolute SPL values at 30 cm [dB re 20 µPa] calculated from the head-mounted microphone signal.

\((MT07_12_12_2003_9_06am\_dataset.wav)\)
9.5. The *Script4_SALxSPLcalibrationR4.m*:

The *Script4_SALxSPLcalibrationR4.m* relates the skin acceleration levels (SAL) registered with the accelerometer to the absolute sound pressure levels registered with the head-mounted microphone for all the speech segments. The *Section for Entering information* of the *Script4_SALxSPLcalibrationR4.m* requires entering only the subject code (e.g., MT07), all the other information is taken from the values entered before and calculated by the previous scripts.

```matlab
%---------------------------------
%Section for Entering information:
%Enter subject code: (e.g., F002)
subject = 'MT07';
%End section for entering information:
%-------------------------------------
```

The scripts produce the total of nine figures, which are shown below. The first figure shows the SAL and SPL values of all the readings being analyzed (Figure 9.10). The second figure relates all the corresponding SAL-SPL pairs (Figure 9.11a). The third figure shows the same data as the previous figure but it shows only the voiced values and eliminates all unvoiced values (Figure 9.11b). The fourth figure (Figure 9.11c) eliminates the first and last value of each voicing segment (i.e., voicing onsets and offsets) from the previous figure. In some subjects, the elimination of the onsets and offsets has been found helpful in eliminating outliers. The fifth figure shows the data from the previous figure together with their best-fit relationships (Figure 9.11d). The first order (linear) and second order (quadratic) relationships are displayed here.
together with their 95% confidence intervals and the equations of these relationships are given in the figure. (Notice that the negative coefficients show up in the equation having \(-\) sign in front of it, so that the 1st order equation is given as, e.g., \(y = 1.3161x + 39.3426\), instead of \(y = 1.3161x - 39.3426\). The \(y\) stands for SPL and \(x\) for SAL)

\[ y = 1.3161x ± 39.3426 \]

![Figure 9.11: The SAL versus SPL relationship obtained for the combined comfortable, soft and loud readings of the Rainbow and Marvin Williams passage in Subject MT07: a) all the values, voiced as well as unvoiced; b) only the voiced values; c) the voiced values without the onsets and offsets; d) the first order (red) and second order (green) best fit relationships superimposed on the data from c).](image)
The sixth figure selects only the Rainbow passage readings and shows the SAL and SPL values for these (Figure 19.12a). The seventh figure (Figure 9.12b) gives the SAL versus SPL relationship for the Rainbow passage (normal, soft and loud readings combined). The eight and ninth figures (Figure 9.13a and 9.13b) are the same displays as Figure 9.12 but for the Marvin Williams passage instead of the Rainbow passage.

For the dose calculations, the SAL versus SPL relationship for the Rainbow passage (as given in Figure 9.12b) has been chosen as the most representative one [9]. The second order relationship has been found slightly more accurate than the first order (although not by much). Thus, for this particular subject (MT07), the coefficients from the relationship

\[ SPL = 0.011781 \text{SAL}^2 - 0.56173 \text{SAL} + 35.0137 \]

would be entered in the script for extracting the data collected by the NCVS Dosimeter (e.g., dosi2_7_read5.m) so that the SALs registered by the accelerometer can be converted into SPLs.

Figures 9.11d, 9.12b and 9.13b can be copied and pasted into a word document for archival purposes (as done in the NCVS, where the file TeacherCalibrationsAt30cm.doc has been created for this purpose).
9.6. The Script5_SALxSPL calibration R4.m:

This is the last script, which is used to calculate mean and equivalent SPLs, mean F0s and spectral centroids for each of the readings (e.g., normal, soft, loud) of the Rainbow and Marvin Williams passages. The section for entering information requires entering only the subject code; everything else is obtained from the values calculated by the previous scripts (which are stored in the file 'MT07_calR4.mat').

The script displays Figure 9.14 and exports the parameters and calculated data (310 values altogether) into a Lotus WK1 spreadsheet (in this case, it automatically creates the file MT07_calResults.wk1 for this purpose). It is convenient then to import the data into an MS Excel spreadsheet (e.g., the file CalibrationResults.xls has been created for statistical analysis of the data from different measured subjects. These data were used for investigating the accuracy of the SPL estimation from skin vibration [9]). The data exported by the script are listed below:
Figure 9.14: The vocal doses calculated from the Rainbow and Marvin Williams passage readings and the values used for calculating the vocal doses.

- **a)** the absolute values of SPL@30 cm. The red line shows the mean SPL for the six different readings. The green line shows the equivalent SPL@30 cm for these readings;
- **b)** the extracted fundamental frequencies F0. The red line shows the mean F0 for the six different readings;
- **c)** the voicing-unvoicing parameter kv;
- **d)** the Time Dose;
- **e)** the Cycle Dose;
- **f)** the Distance Dose.

% Create results file:
% This command writes the data in an excel spreadsheet
% wkwrite(xlsfile,datalog);
res(1)=sal_thresh;      % [dB] SAL threshold
res(2)=SLM_distance;   % [cm] Distance of the sound level meter from the mouth
res(3)=FrameDur;       % [s] Frame duration
res(4)=start_freq;     % [Hz] Lowest frequency limit for F0 extraction
res(5)=stop_freq;      % [Hz] Highest frequency limit for F0 extraction
res(6)=fo_correction;  % [1=yes, 0=no] Correction for F0 artifacts
res(7)=isol_correction; % [1=yes, 0=no] Correction for isolated frames
res(8)=accuracy_start_sync; % [s] Accuracy of the synchronization at the beginning
res(9)=accuracy_end_sync;  % [s] Accuracy of the synchronization at the end
res(10)=P1(1);        % 1st coefficient of the 1st order fit for all data
res(11)=P1(2);        % 2nd coefficient of the 1st order fit for all data
res(12)=deltal_ave;   % [+-dB] 95% confidence interval: average value for the 1st order fit
res(13)=P2(1);        % 1st coefficient of the 2nd order fit for all data
res(14)=P2(2);        % 2nd coefficient of the 2nd order fit for all data
res(15)=P2(3);        % 3rd coefficient of the 2nd order fit for all data
res(16) = delta2_ave;           % [+-dB] 95% confidence interval: average value for the 2nd order fit
res(17) = P1_Rainbow(1);       % 1st coefficient of the 1st order fit for Rainbow passage
                                % [SPL = P1(1) * SAL + P1(2)]
res(18) = P1_Rainbow(2);       % 2nd coefficient of the 1st order fit for Rainbow passage
                                % [SPL = P1(1) * SAL + P1(2)]
res(19) = delta1_Rainbow_ave;   % [+-dB] 95% confidence interval: average value for the 1st order fit
res(20) = P2_Rainbow(1);       % 1st coefficient of the 2nd order fit for Rainbow passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(21) = P2_Rainbow(2);       % 2nd coefficient of the 2nd order fit for Rainbow passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(22) = P2_Rainbow(3);       % 3rd coefficient of the 2nd order fit for Rainbow passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(23) = delta2_Rainbow_ave;   % [+-dB] 95% confidence interval: average value for the 2nd order fit
res(24) = P1_Marvin(1);        % 1st coefficient of the 1st order fit for Marvin passage
                                % [SPL = P1(1) * SAL + P1(2)]
res(25) = P1_Marvin(2);        % 2nd coefficient of the 1st order fit for Marvin passage
                                % [SPL = P1(1) * SAL + P1(2)]
res(26) = delta1_Marvin_ave;    % [+-dB] 95% confidence interval: average value for the 1st order fit
res(27) = P2_Marvin(1);        % 1st coefficient of the 2nd order fit for Marvin passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(28) = P2_Marvin(2);        % 2nd coefficient of the 2nd order fit for Marvin passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(29) = P2_Marvin(3);        % 3rd coefficient of the 2nd order fit for Marvin passage
                                % [SPL = P2(1) * SAL^2 + P2(2) * SAL + P2(3)]
res(30) = delta2_Marvin_ave;    % [+-dB] 95% confidence interval: average value for the 2nd order fit
% The whole Rainbow passage
res(31) = fo_mean_Rainbow;      % mean F0 for Rainbow passage
res(32) = FEC2_total_Rainbow;   % Spectral center of gravity for the Rainbow passage
                                % (value from total energy of accelerometer)
res(33) = FEC2_mean_Rainbow;    % Spectral center of gravity for the Rainbow passage —
                                % mean value from all voiced frames (accelerometer)
res(34) = sal_mean_Rainbow;     % Mean SAL for Rainbow passage
res(35) = sal_eq_voiced_Rainbow; % equivalent SAL for Rainbow passage (only voiced elements)
res(36) = spl_mean_Rainbow;     % Mean SPL for Rainbow passage (from head mounted mic)
res(37) = spl_mean_Rainbow_fitM1;  % Mean SPL for Rainbow passage (from SAL 1st order fit of Marvin)
res(38) = spl_mean_Rainbow_fitM2;  % Mean SPL for Rainbow passage (from SAL 2nd order fit of Marvin)
res(39) = spl_mean_Rainbow_fitR1;  % Mean SPL for Rainbow passage (from SAL 1st order fit of Rainbow)
res(40) = spl_mean_Rainbow_fitR2;  % Mean SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)
res(41) = spl_eq_Rainbow;       % Equivalent SPL for Rainbow passage (from head mounted mic)
res(42) = spl_eq_Rainbow_fitM1;  % Equivalent SPL for Rainbow passage (from SAL 1st order fit of Marvin)
res(43) = spl_eq_Rainbow_fitM2;  % Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Marvin)
res(44) = spl_eq_Rainbow_fitR1;  % Equivalent SPL for Rainbow passage (from SAL 1st order fit of Rainbow)
res(45) = spl_eq_Rainbow_fitR2;  % Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)
res(46)=spl_eq_voiced_Rainbow; %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from head mounted mic)
res(47)=spl_eq_voiced_Rainbow_fitM1; %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(48)=spl_eq_voiced_Rainbow_fitM2; %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(49)=spl_eq_voiced_Rainbow_fitR1; %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(50)=spl_eq_voiced_Rainbow_fitR2; %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(51)=Tm_Rainbow; %Measurement time [s] for the Rainbow passage
res(52)=Dt_Rainbow; %Time dose [s] for the Rainbow passage
res(53)=Tperc_Rainbow; %Voicing percentage [%] for the Rainbow passage
res(54)=Dc_cum_Rainbow(end); %Cycle dose [cycles] for the Rainbow passage
res(55)=Dd_cum_Rainbow(end); %Distance dose [m] for the Rainbow passage
res(56)=Dd_cum_Rainbow_fitM1(end); %Distance dose [m] for the Rainbow passage (1st order fit Marvin)
res(57)=Dd_cum_Rainbow_fitM2(end); %Distance dose [m] for the Rainbow passage (2nd order fit Marvin)
res(58)=Dd_cum_Rainbow_fitR1(end); %Distance dose [m] for the Rainbow passage (1st order fit Rainbow)
res(59)=Dd_cum_Rainbow_fitR2(end); %Distance dose [m] for the Rainbow passage (2nd order fit Rainbow)
res(60)=Dc_cum_Rainbow(end)/Dt_Rainbow; %Cycle dose per second [cps] for the Rainbow passage
res(61)=Dd_cum_Rainbow(end)/Dt_Rainbow; %Distance dose per second [m/s] for the Rainbow passage
res(62)=Dd_cum_Rainbow_fitM1(end)/Dt_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (1st order fit Marvin)
res(63)=Dd_cum_Rainbow_fitM2(end)/Dt_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (2nd order fit Marvin)
res(64)=Dd_cum_Rainbow_fitR1(end)/Dt_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (1st order fit Rainbow)
res(65)=Dd_cum_Rainbow_fitR2(end)/Dt_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (2nd order fit Rainbow)

%Rainbow passage normal loudness
res(66)=fo_mean_normal_Rainbow; %mean F0 for Rainbow passage
res(67)=FEC2_total_normal_Rainbow; %Spectral center of gravity for the Rainbow passage (value from total energy of accelerometer)
res(68)=FEC2_mean_normal_Rainbow; %Spectral center of gravity for the Rainbow passage - mean value from all voiced frames (accelerometer)
res(69)=sal_mean_normal_Rainbow; %Mean SAL for Rainbow passage
res(70)=sal_eq_voiced_normal_Rainbow; %equivalent SAL for Rainbow passage (only voiced elements)
res(71)=spl_mean_normal_Rainbow; %Mean SPL for Rainbow passage (from head mounted mic)
res(72)=spl_mean_normal_Rainbow_fitM1; %Mean SPL for Rainbow passage (from SAL 1st order fit of Marvin)
res(73)=spl_mean_normal_Rainbow_fitM2; %Mean SPL for Rainbow passage (from SAL 2nd order fit of Marvin)
res(74)=spl_mean_normal_Rainbow_fitR1; %Mean SPL for Rainbow passage (from SAL 1st order fit of Rainbow)
res(75)=spl_mean_normal_Rainbow_fitR2; %Mean SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)
res(76)=spl_eq_normal_Rainbow; %Equivalent SPL for Rainbow passage (from head mounted mic)
res(77)=spl_eq_normal_Rainbow_fitM1; %Equivalent SPL for Rainbow passage (from SAL 1st order fit of Marvin)
res(78)=spl_eq_normal_Rainbow_fitM2;   %Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Marvin)
res(79)=spl_eq_normal_Rainbow_fitR1;   %Equivalent SPL for Rainbow passage (from SAL 1st order fit of Rainbow)
res(80)=spl_eq_normal_Rainbow_fitR2;   %Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)
res(81)=spl_eq_voiced_normal_Rainbow;  %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from head mounted mic)
res(82)=spl_eq_voiced_normal_Rainbow_fitM1;    %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(83)=spl_eq_voiced_normal_Rainbow_fitM2;    %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(84)=spl_eq_voiced_normal_Rainbow_fitR1;    %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(85)=spl_eq_voiced_normal_Rainbow_fitR2;    %Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(86)=Tm_normal_Rainbow;             %Measurement time [s] for the Rainbow passage
res(87)=Dt_normal_Rainbow;             %Time dose [s] for the Rainbow passage
res(88)=Tperc_normal_Rainbow;          %Voicing percentage [%] for the Rainbow passage
res(89)=Dc_cum_normal_Rainbow(end);    %Cycle dose [cycles] for the Rainbow passage
res(90)=Dd_cum_normal_Rainbow(end);    %Distance dose [m] for the Rainbow passage
res(91)=Dd_cum_normal_Rainbow_fitM1(end); %Distance dose [m] for the Rainbow passage (1st order fit Marvin)
res(92)=Dd_cum_normal_Rainbow_fitM2(end); %Distance dose [m] for the Rainbow passage (2nd order fit Marvin)
res(93)=Dd_cum_normal_Rainbow_fitR1(end); %Distance dose [m] for the Rainbow passage (1st order fit Rainbow)
res(94)=Dd_cum_normal_Rainbow_fitR2(end); %Distance dose [m] for the Rainbow passage (2nd order fit Rainbow)
res(95)=Dd_cum_normal_Rainbow=end)/Dt_normal_Rainbow;    %Cycle dose per second [cps] for the Rainbow passage
res(96)=Dd_cum_normal_Rainbow=end)/Dt_normal_Rainbow;    %Distance dose per second [m/s] for the Rainbow passage
res(97)=Dd_cum_normal_Rainbow_fitM1=end)/Dt_normal_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (1st order fit Marvin)
res(98)=Dd_cum_normal_Rainbow_fitM2=end)/Dt_normal_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (2nd order fit Marvin)
res(99)=Dd_cum_normal_Rainbow_fitR1=end)/Dt_normal_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (1st order fit Rainbow)
res(100)=Dd_cum_normal_Rainbow_fitR2=end)/Dt_normal_Rainbow; %Distance dose per second [m/s] for the Rainbow passage (2nd order fit Rainbow)

%Rainbow passage soft
res(101)=fo_mean_soft_Rainbow;         %mean F0 for Rainbow passage
res(102)=FEC2_total_soft_Rainbow;      %Spectral center of gravity for the Rainbow passage (value from total energy of accelerometer)
res(103)=FEC2_mean_soft_Rainbow;      %Spectral center of gravity for the Rainbow passage - mean value from all voiced frames (accelerometer)
res(104)=sal_mean_soft_Rainbow;        %Mean SAL for Rainbow passage
res(105)=sal_eq_voiced_soft_Rainbow;   %Equivalent SAL for Rainbow passage (only voiced elements)
res(106)=spl_mean_soft_Rainbow;         %Mean SPL for Rainbow passage (from head mounted mic)
res(107)=spl_mean_soft_Rainbow_fitM1;  %Mean SPL for Rainbow passage (from SAL 1st order fit of Marvin)
res(108)=spl_mean_soft_Rainbow_fitM2;  %Mean SPL for Rainbow passage (from SAL 2nd order fit of Marvin)
res(109)=spl_mean_soft_Rainbow_fitR1;  %Mean SPL for Rainbow passage (from SAL 1st order fit of Rainbow)
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<td>res(110)=spl_mean_soft_Rainbow_fitR2; Mean SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)</td>
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<td>111</td>
<td>res(111)=spl_eq_soft_Rainbow; Equivalent SPL for Rainbow passage (from head mounted mic)</td>
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<tr>
<td>112</td>
<td>res(112)=spl_eq_soft_Rainbow_fitM1; Equivalent SPL for Rainbow passage (from SAL 1st order fit of Marvin)</td>
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<td>113</td>
<td>res(113)=spl_eq_soft_Rainbow_fitM2; Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Marvin)</td>
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<td>114</td>
<td>res(114)=spl_eq_soft_Rainbow_fitR1; Equivalent SPL for Rainbow passage (from SAL 1st order fit of Rainbow)</td>
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<td>115</td>
<td>res(115)=spl_eq_soft_Rainbow_fitR2; Equivalent SPL for Rainbow passage (from SAL 2nd order fit of Rainbow)</td>
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<tr>
<td>116</td>
<td>res(116)=spl_eq_voiced_soft_Rainbow; Equivalent SPL for Rainbow passage unvoiced frames eliminated (from head mounted mic)</td>
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<tr>
<td>117</td>
<td>res(117)=spl_eq_voiced_soft_Rainbow_fitM1; Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)</td>
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<td>118</td>
<td>res(118)=spl_eq_voiced_soft_Rainbow_fitM2; Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)</td>
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<tr>
<td>119</td>
<td>res(119)=spl_eq_voiced_soft_Rainbow_fitR1; Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)</td>
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<tr>
<td>120</td>
<td>res(120)=spl_eq_voiced_soft_Rainbow_fitR2; Equivalent SPL for Rainbow passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)</td>
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<td>121</td>
<td>res(121)=Tm_soft_Rainbow; Measurement time [s] for the Rainbow passage</td>
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<td>122</td>
<td>res(122)=Dt_soft_Rainbow; Time dose [s] for the Rainbow passage</td>
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<tr>
<td>123</td>
<td>res(123)=Tperc_soft_Rainbow; Voicing percentage [%] for the Rainbow passage</td>
</tr>
<tr>
<td>124</td>
<td>res(124)=Dc_cum_soft_Rainbow(end); Cycle dose [cycles] for the Rainbow passage</td>
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<tr>
<td>125</td>
<td>res(125)=Dd_cum_soft_Rainbow(end); Distance dose [m] for the Rainbow passage</td>
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<tr>
<td>126</td>
<td>res(126)=Dd_cum_soft_Rainbow_fitM1(end); Distance dose [m] for the Rainbow passage (1st order fit Marvin)</td>
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<tr>
<td>127</td>
<td>res(127)=Dd_cum_soft_Rainbow_fitM2(end); Distance dose [m] for the Rainbow passage (2nd order fit Marvin)</td>
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<tr>
<td>128</td>
<td>res(128)=Dd_cum_soft_Rainbow_fitR1(end); Distance dose [m] for the Rainbow passage (1st order fit Rainbow)</td>
</tr>
<tr>
<td>129</td>
<td>res(129)=Dd_cum_soft_Rainbow_fitR2(end); Distance dose [m] for the Rainbow passage (2nd order fit Rainbow)</td>
</tr>
<tr>
<td>130</td>
<td>res(130)=Dc_cum_soft_Rainbow(end)/Dt_soft_Rainbow; Cycle dose per second [cps] for the Rainbow passage</td>
</tr>
<tr>
<td>131</td>
<td>res(131)=Dd_cum_soft_Rainbow(end)/Dt_soft_Rainbow; Distance dose per second [m/s] for the Rainbow passage</td>
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<tr>
<td>132</td>
<td>res(132)=Dd_cum_soft_Rainbow_fitM1(end)/Dt_soft_Rainbow; Distance dose per second [m/s] for the Rainbow passage (1st order fit Marvin)</td>
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<td>133</td>
<td>res(133)=Dd_cum_soft_Rainbow_fitM2(end)/Dt_soft_Rainbow; Distance dose per second [m/s] for the Rainbow passage (2nd order fit Marvin)</td>
</tr>
<tr>
<td>134</td>
<td>res(134)=Dd_cum_soft_Rainbow_fitR1(end)/Dt_soft_Rainbow; Distance dose per second [m/s] for the Rainbow passage (1st order fit Rainbow)</td>
</tr>
<tr>
<td>135</td>
<td>res(135)=Dd_cum_soft_Rainbow_fitR2(end)/Dt_soft_Rainbow; Distance dose per second [m/s] for the Rainbow passage (2nd order fit Rainbow)</td>
</tr>
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</table>

%Rainbow passage loud

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<td>res(136)=fo_mean_loud_Rainbow; Mean F0 for Rainbow passage</td>
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<td>137</td>
<td>res(137)=FEC2_total_loud_Rainbow; Spectral center of gravity for the Rainbow passage (value from total energy of accelerometer)</td>
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<tr>
<td>138</td>
<td>res(138)=FEC2_mean_loud_Rainbow; Spectral center of gravity for the Rainbow passage - mean value from all voiced frames (accelerometer)</td>
</tr>
<tr>
<td>139</td>
<td>res(139)=sal_mean_loud_Rainbow; Mean SAL for Rainbow passage</td>
</tr>
<tr>
<td>140</td>
<td>res(140)=sal_eq_voiced_loud_Rainbow; equivalent SAL for Rainbow passage (only voiced elements)</td>
</tr>
<tr>
<td>141</td>
<td>res(141)=spl_mean_loud_Rainbow; Mic Mean SPL for Rainbow passage (from head mounted mic)</td>
</tr>
</tbody>
</table>
res(142)=spla loud Rainbow_fitM1; %Mean SPL for Rainbow passage (from SAL 1st
order fit of Marvin)
res(143)=spla loud Rainbow_fitM2; %Mean SPL for Rainbow passage (from SAL 2nd
order fit of Marvin)
res(144)=spla loud Rainbow_fitR1; %Mean SPL for Rainbow passage (from SAL 1st
order fit of Rainbow)
res(145)=spla loud Rainbow_fitR2; %Mean SPL for Rainbow passage (from SAL 2nd
order fit of Rainbow)
res(146)=spla eq loud Rainbow; %Equivalent SPL for Rainbow passage (from head
mounted mic)
res(147)=spla eq loud Rainbow_fitM1; %Equivalent SPL for Rainbow passage (from SAL 1st
order fit of Marvin)
res(148)=spla eq loud Rainbow_fitM2; %Equivalent SPL for Rainbow passage (from SAL 2nd
order fit of Marvin)
res(149)=spla eq loud Rainbow_fitR1; %Equivalent SPL for Rainbow passage (from SAL 1st
order fit of Rainbow)
res(150)=spla eq loud Rainbow_fitR2; %Equivalent SPL for Rainbow passage (from SAL 2nd
order fit of Rainbow)
res(151)=spla eq voiced loud Rainbow; %Equivalent SPL for Rainbow passage unvoiced
frames eliminated (from head mounted mic)
res(152)=spla eq voiced loud Rainbow_fitM1; %Equivalent SPL for Rainbow passage
unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(153)=spla eq voiced loud Rainbow_fitM2; %Equivalent SPL for Rainbow passage
unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(154)=spla eq voiced loud Rainbow_fitR1; %Equivalent SPL for Rainbow passage
unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(155)=spla eq voiced loud Rainbow_fitR2; %Equivalent SPL for Rainbow passage
unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(156)=Tm loud Rainbow; %Measurement time [s] for the Rainbow passage
res(157)=Dt loud Rainbow; %Time dose [s] for the Rainbow passage
res(158)=tperc loud Rainbow; %Voicing percentage [%] for the Rainbow passage
res(159)=Dc_cum loud Rainbow(end); %Cycle dose [cycles] for the Rainbow passage
res(160)=Dd_cum loud Rainbow(end); %Distance dose [m] for the Rainbow passage
res(161)=Dd_cum loud Rainbow_fitM1(end); %Distance dose [m] for the Rainbow passage
(1st order fit Marvin)
res(162)=Dd_cum loud Rainbow_fitM2(end); %Distance dose [m] for the Rainbow passage
(2nd order fit Marvin)
res(163)=Dd_cum loud Rainbow_fitR1(end); %Distance dose [m] for the Rainbow passage
(1st order fit Rainbow)
res(164)=Dd_cum loud Rainbow_fitR2(end); %Distance dose [m] for the Rainbow passage
(2nd order fit Rainbow)
res(165)=Dc_cum loud Rainbow(end)/Dt loud Rainbow; %Cycle dose per second [cps] for
the Rainbow passage
res(166)=Dd_cum loud Rainbow(end)/Dt loud Rainbow; %Distance dose per second [m/s]
for the Rainbow passage
res(167)=Dd_cum loud Rainbow_fitM1(end)/Dt loud Rainbow; %Distance dose per second
[m/s] for the Rainbow passage (1st order fit Marvin)
res(168)=Dd_cum loud Rainbow_fitM2(end)/Dt loud Rainbow; %Distance dose per second
[m/s] for the Rainbow passage (2nd order fit Marvin)
res(169)=Dd_cum loud Rainbow_fitR1(end)/Dt loud Rainbow; %Distance dose per second
[m/s] for the Rainbow passage (1st order fit Rainbow)
res(170)=Dd_cum loud Rainbow_fitR2(end)/Dt loud Rainbow; %Distance dose per second
[m/s] for the Rainbow passage (2nd order fit Rainbow)
%The whole Marvin passage
res(171)=fo_mean Marvin; %mean F0 for Marvin passage
res(172)=FEC2_total Marvin; %Spectral center of gravity for the Marvin passage
(value from total energy of accelerometer)
res(173)=FEC2 mean Marvin; %Spectral center of gravity for the Marvin passage -
mean value from all voiced frames (accelerometer)
res(174)=sal_mean_Marvin;  %Mean SAL for Marvin passage
res(175)=sal_eq_voiced_Marvin;  %equivalent SAL for Marvin passage (only voiced elements)
res(176)=spl_mean_Marvin;  %Mean SPL for Marvin passage (from head mounted mic)
res(177)=spl_mean_Marvin_fitM1;  %Mean SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(178)=spl_mean_Marvin_fitM2;  %Mean SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(179)=spl_mean_Marvin_fitR1;  %Mean SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(180)=spl_mean_Marvin_fitR2;  %Mean SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(181)=spl_eq_Marvin;  %Equivalent SPL for Marvin passage (from head mounted mic)
res(182)=spl_eq_Marvin_fitM1;  %Equivalent SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(183)=spl_eq_Marvin_fitM2;  %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(184)=spl_eq_Marvin_fitR1;  %Equivalent SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(185)=spl_eq_Marvin_fitR2;  %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(186)=spl_eq_voiced_Marvin;  %Equivalent SPL for Marvin passage unvoiced frames eliminated (from head mounted mic)
res(187)=spl_eq_voiced_Marvin_fitM1;  %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(188)=spl_eq_voiced_Marvin_fitM2;  %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(189)=spl_eq_voiced_Marvin_fitR1;  %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(190)=spl_eq_voiced_Marvin_fitR2;  %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(191)=Tm_Marvin;  %Measurement time [s] for the Marvin passage
res(192)=Dt_Marvin;  %Time dose [s] for the Marvin passage
res(193)=Tperc_Marvin;  %Voicing percentage [%] for the Marvin passage
res(194)=Dc_cum_Marvin(end);  %Cycle dose [cycles] for the Marvin passage
res(195)=Dd_cum_Marvin(end);  %Distance dose [m] for the Marvin passage
res(196)=Dd_cum_Marvin_fitM1(end);  %Distance dose [m] for the Marvin passage (1st order fit Marvin)
res(197)=Dd_cum_Marvin_fitM2(end);  %Distance dose [m] for the Marvin passage (2nd order fit Marvin)
res(198)=Dd_cum_Marvin_fitR1(end);  %Distance dose [m] for the Marvin passage (1st order fit Rainbow)
res(199)=Dd_cum_Marvin_fitR2(end);  %Distance dose [m] for the Marvin passage (2nd order fit Rainbow)
res(200)=Dc_cum_Marvin(end)/Dt_Marvin;  %Cycle dose per second [cps] for the Marvin passage
res(201)=Dd_cum_Marvin(end)/Dt_Marvin;  %Distance dose per second [m/s] for the Marvin passage
res(202)=Dd_cum_Marvin_fitM1(end)/Dt_Marvin;  %Distance dose per second [m/s] for the Marvin passage (1st order fit Marvin)
res(203)=Dd_cum_Marvin_fitM2(end)/Dt_Marvin;  %Distance dose per second [m/s] for the Marvin passage (2nd order fit Marvin)
res(204)=Dd_cum_Marvin_fitR1(end)/Dt_Marvin;  %Distance dose per second [m/s] for the Marvin passage (1st order fit Rainbow)
res(205)=Dd_cum_Marvin_fitR2(end)/Dt_Marvin;  %Distance dose per second [m/s] for the Marvin passage (2nd order fit Rainbow)

%Marvin passage normal loudness
res(206)=fo_mean_normal_Marvin;  %mean F0 for Marvin passage
| res(207)=FEC2_total_normal_Marvin; | Spectral center of gravity for the Marvin passage (value from total energy of accelerometer) |
| res(208)=FEC2_mean_normal_Marvin; | Spectral center of gravity for the Marvin passage - mean value from all voiced frames (accelerometer) |
| res(209)=sal_mean_normal_Marvin; | Mean SAL for Marvin passage |
| res(210)=sal_eq_voiced_normal_Marvin; | Equivalent SAL for Marvin passage (only voiced elements) |
| res(211)=spl_mean_normal_Marvin; | Mean SPL for Marvin passage (from head mounted mic) |
| res(212)=spl_mean_normal_Marvin_fitM1; | Mean SPL for Marvin passage (from SAL 1st order fit of Marvin) |
| res(213)=spl_mean_normal_Marvin_fitM2; | Mean SPL for Marvin passage (from SAL 2nd order fit of Marvin) |
| res(214)=spl_mean_normal_Marvin_fitR1; | Mean SPL for Marvin passage (from SAL 1st order fit of Rainbow) |
| res(215)=spl_mean_normal_Marvin_fitR2; | Mean SPL for Marvin passage (from SAL 2nd order fit of Rainbow) |
| res(216)=spl_eq_normal_Marvin; | Equivalent SPL for Marvin passage (from head mounted mic) |
| res(217)=spl_eq_normal_Marvin_fitM1; | Equivalent SPL for Marvin passage (from SAL 1st order fit of Marvin) |
| res(218)=spl_eq_normal_Marvin_fitM2; | Equivalent SPL for Marvin passage (from SAL 2nd order fit of Marvin) |
| res(219)=spl_eq_normal_Marvin_fitR1; | Equivalent SPL for Marvin passage (from SAL 1st order fit of Rainbow) |
| res(220)=spl_eq_normal_Marvin_fitR2; | Equivalent SPL for Marvin passage (from SAL 2nd order fit of Rainbow) |
| res(221)=spl_eq_voiced_normal_Marvin; | Equivalent SPL for Marvin passage unvoiced frames eliminated (from head mounted mic) |
| res(222)=spl_eq_voiced_normal_Marvin_fitM1; | Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Marvin) |
| res(223)=spl_eq_voiced_normal_Marvin_fitM2; | Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin) |
| res(224)=spl_eq_voiced_normal_Marvin_fitR1; | Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow) |
| res(225)=spl_eq_voiced_normal_Marvin_fitR2; | Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow) |
| res(226)=Tm_normal_Marvin; | Measurement time [s] for the Marvin passage |
| res(227)=Dt_normal_Marvin; | Time dose [s] for the Marvin passage |
| res(228)=Tperc_normal_Marvin; | Voicing percentage [%] for the Marvin passage |
| res(229)=Dc_cum_normal_Marvin(end); | Cycle dose [cycles] for the Marvin passage |
| res(230)=Dd_cum_normal_Marvin(end); | Distance dose [m] for the Marvin passage |
| res(231)=Dd_cum_normal_Marvin_fitM1(end); | Distance dose [m] for the Marvin passage (1st order fit Marvin) |
| res(232)=Dd_cum_normal_Marvin_fitM2(end); | Distance dose [m] for the Marvin passage (2nd order fit Marvin) |
| res(233)=Dd_cum_normal_Marvin_fitR1(end); | Distance dose [m] for the Marvin passage (1st order fit Rainbow) |
| res(234)=Dd_cum_normal_Marvin_fitR2(end); | Distance dose [m] for the Marvin passage (2nd order fit Rainbow) |
| res(235)=Dc_cum_normal_Marvin(end)/Dt_normal_Marvin; | Cycle dose per second [cps] for the Marvin passage |
| res(236)=Dd_cum_normal_Marvin(end)/Dt_normal_Marvin; | Distance dose per second [m/s] for the Marvin passage |
| res(237)=Dd_cum_normal_Marvin_fitM1(end)/Dt_normal_Marvin; | Distance dose per second [m/s] for the Marvin passage (1st order fit Marvin) |
| res(238)=Dd_cum_normal_Marvin_fitM2(end)/Dt_normal_Marvin; | Distance dose per second [m/s] for the Marvin passage (2nd order fit Marvin) |
| res(239)=Dd_cum_normal_Marvin_fitR1(end)/Dt_normal_Marvin; | Distance dose per second [m/s] for the Marvin passage (1st order fit Rainbow) |
res(240)=Dd_cum_normal_Marvin_fitR2(end)/Dt_normal_Marvin; %Distance dose per second [m/s] for the Marvin passage (2nd order fit Rainbow)

%Marvin passage soft
res(241)=fo_mean_soft_Marvin; %mean F0 for Marvin passage
res(242)=FEC2_total_soft_Marvin; %Spectral center of gravity for the Marvin passage (value from total energy of accelerometer)
res(243)=FEC2_mean_soft_Marvin; %Spectral center of gravity for the Marvin passage - mean value from all voiced frames (accelerometer)
res(244)=sal_mean_soft_Marvin; %Mean SAL for Marvin passage
res(245)=sal_eq_voiced_soft_Marvin; %equivalent SAL for Marvin passage (only voiced elements)
res(246)=spl_mean_soft_Marvin; %Mean SPL for Marvin passage (from head mounted mic)
res(247)=spl_mean_soft_Marvin_fitM1; %Mean SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(248)=spl_mean_soft_Marvin_fitM2; %Mean SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(249)=spl_mean_soft_Marvin_fitR1; %Mean SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(250)=spl_mean_soft_Marvin_fitR2; %Mean SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(251)=spl_eq_soft_Marvin; %Equivalent SPL for Marvin passage (from head mounted mic)
res(252)=spl_eq_soft_Marvin_fitM1; %Equivalent SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(253)=spl_eq_soft_Marvin_fitM2; %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(254)=spl_eq_soft_Marvin_fitR1; %Equivalent SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(255)=spl_eq_soft_Marvin_fitR2; %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(256)=spl_eq_voiced_soft_Marvin; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from head mounted mic)
res(257)=spl_eq_voiced_soft_Marvin_fitM1; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(258)=spl_eq_voiced_soft_Marvin_fitM2; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(259)=spl_eq_voiced_soft_Marvin_fitR1; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(260)=spl_eq_voiced_soft_Marvin_fitR2; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(261)=Tm_soft_Marvin; %Measurement time [s] for the Marvin passage
res(262)=Dt_soft_Marvin; %Time dose [s] for the Marvin passage
res(263)=Tperc_soft_Marvin; %Voicing percentage [%] for the Marvin passage
res(264)=Dc_cum_soft_Marvin(end); %Cycle dose [cycles] for the Marvin passage
res(265)=Dd_cum_soft_Marvin(end); %Distance dose [m] for the Marvin passage
res(266)=Dd_cum_soft_Marvin_fitM1(end); %Distance dose [m] for the Marvin passage (1st order fit Marvin)
res(267)=Dd_cum_soft_Marvin_fitM2(end); %Distance dose [m] for the Marvin passage (2nd order fit Marvin)
res(268)=Dd_cum_soft_Marvin_fitR1(end); %Distance dose [m] for the Marvin passage (1st order fit Rainbow)
res(269)=Dd_cum_soft_Marvin_fitR2(end); %Distance dose [m] for the Marvin passage (2nd order fit Rainbow)
res(270)=Dc_cum_soft_Marvin(end)/Dt_soft_Marvin; %Cycle dose per second [cps] for the Marvin passage
res(271)=Dd_cum_soft_Marvin(end)/Dt_soft_Marvin; %Distance dose per second [m/s] for the Marvin passage
res(272)=Dd_cum_soft_Marvin_fitM1(end)/Dt_soft_Marvin; %Distance dose per second [m/s] for the Marvin passage (1st order fit Marvin)
res(273)=Dd_cum_soft_Marvin_fitM2(end)/Dt_soft_Marvin; %Distance dose per second [m/s] for the Marvin passage (2nd order fit Marvin)
res(274)=Dd_cum_soft_Marvin_fitR1(end)/Dt_soft_Marvin; %Distance dose per second [m/s] for the Marvin passage (1st order fit Rainbow)
res(275)=Dd_cum_soft_Marvin_fitR2(end)/Dt_soft_Marvin; %Distance dose per second [m/s] for the Marvin passage (2nd order fit Rainbow)

%Marvin passage loud
res(276)=fo_mean_loud_Marvin; %mean F0 for Marvin passage
res(277)=FEC2_total_loud_Marvin; %Spectral center of gravity for the Marvin passage (value from total energy of accelerometer)
res(278)=FEC2_mean_loud_Marvin; %Spectral center of gravity for the Marvin passage - mean value from all voiced frames (accelerometer)
res(279)=sal_mean_loud_Marvin; %Mean SAL for Marvin passage
res(280)=sal_eq_voiced_loud_Marvin; %Spectral center of gravity for the Marvin passage - mean value from all voiced frames (accelerometer)
res(281)=spl_mean_loud_Marvin; %Mean SPL for Marvin passage (from head mounted mic)
res(282)=spl_mean_loud_Marvin_fitM1; %Mean SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(283)=spl_mean_loud_Marvin_fitM2; %Mean SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(284)=spl_mean_loud_Marvin_fitR1; %Mean SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(285)=spl_mean_loud_Marvin_fitR2; %Mean SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(286)=spl_eq_loud_Marvin; %Equivalent SPL for Marvin passage (from head mounted mic)
res(287)=spl_eq_loud_Marvin_fitM1; %Equivalent SPL for Marvin passage (from SAL 1st order fit of Marvin)
res(288)=spl_eq_loud_Marvin_fitM2; %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Marvin)
res(289)=spl_eq_loud_Marvin_fitR1; %Equivalent SPL for Marvin passage (from SAL 1st order fit of Rainbow)
res(290)=spl_eq_loud_Marvin_fitR2; %Equivalent SPL for Marvin passage (from SAL 2nd order fit of Rainbow)
res(291)=spl_eq_voiced_loud_Marvin; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from head mounted mic)
res(292)=spl_eq_voiced_loud_Marvin_fitM1; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Marvin)
res(293)=spl_eq_voiced_loud_Marvin_fitM2; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Marvin)
res(294)=spl_eq_voiced_loud_Marvin_fitR1; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 1st order fit of Rainbow)
res(295)=spl_eq_voiced_loud_Marvin_fitR2; %Equivalent SPL for Marvin passage unvoiced frames eliminated (from SAL 2nd order fit of Rainbow)
res(296)=Tm_loud_Marvin; %Measurement time [s] for the Marvin passage
res(297)=Dt_loud_Marvin; %Time dose [s] for the Marvin passage
res(298)=Tperc_loud_Marvin; %Voicing percentage [%] for the Marvin passage
res(299)=Dc_cum_loud_Marvin(end); %Cycle dose [cycles] for the Marvin passage
res(300)=Dd_cum_loud_Marvin(end); %Distance dose [m] for the Marvin passage (1st order fit Marvin)
res(301)=Dd_cum_loud_Marvin_fitM1(end); %Distance dose [m] for the Marvin passage (2nd order fit Marvin)
res(302)=Dd_cum_loud_Marvin_fitM2(end); %Distance dose [m] for the Marvin passage (2nd order fit Marvin)
res(303)=Dd_cum_loud_Marvin_fitR1(end); %Distance dose [m] for the Marvin passage (1st order fit Rainbow)
res(304)=Dd_cum_loud_Marvin_fitR2(end);  %Distance dose [m] for the Marvin passage (2nd order fit Rainbow)
res(305)=Dc_cum_loud_Marvin(end)/Dt_loud_Marvin;    %Cycle dose per second [cps] for the Marvin passage
res(306)=Dd_cum_loud_Marvin(end)/Dt_loud_Marvin;    %Distance dose per second [m/s] for the Marvin passage
res(307)=Dd_cum_loud_Marvin_fitM1(end)/Dt_loud_Marvin;    %Distance dose per second [m/s] for the Marvin passage (1st order fit Marvin)
res(308)=Dd_cum_loud_Marvin_fitM2(end)/Dt_loud_Marvin;    %Distance dose per second [m/s] for the Marvin passage (2nd order fit Marvin)
res(309)=Dd_cum_loud_Marvin_fitR1(end)/Dt_loud_Marvin;    %Distance dose per second [m/s] for the Marvin passage (1st order fit Rainbow)
res(310)=Dd_cum_loud_Marvin_fitR2(end)/Dt_loud_Marvin;    %Distance dose per second [m/s] for the Marvin passage (2nd order fit Rainbow)

Literature

[9]  Švec JG, Titze IR, Popolo PS. Estimation of sound pressure levels of speech from skin vibration of the neck (in review).

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Revisions

1.0 Jan Švec: Main document (Feb. 2004)
1.1 Eric Hunter: general edits (Mar. 2004)
2.0 Eric Hunter: Section 7 revamped with new tools to streamline work (Mar. 2004)
2.2 Eric Hunter: general edits, finalization (Apr. 2004)
2.3 Jan Švec: minor typesetting corrections (Apr. 2004)
2.4 Eric Hunter: dosimeter to be referred to as NCVS dosimeter, agreements of use (Apr. 2004)