# MeasureTransferFunction 1.0 User Manual

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## 1 Introduction

The software MeasureTransferFunction (MTF) allows the convenient measurement of the transfer functions of acoustic systems. It uses logarithmic sine sweeps for the broadband acoustic excitation of the system under test and supports the measurement of two sweep responses at different positions. The measurement principle is based on the method by Farina [2], which is explained in more detail in Sec. 2. The main strength of the method is that it allows finding the *linear* impulse response of the system even when individual system components (e.g., the loudspeaker) generate harmonic distortions. This means also that low-cost (consumer) audio hardware can be used to perform the measurements. The software requires an audio interface that supports a sampling rate of 96 kHz and 24 bit quantization for the simultaneous playback and capturing of audio data. An example for such an interface is the USB interface Aureon Xfire 8.0HD for about 100 Euros.

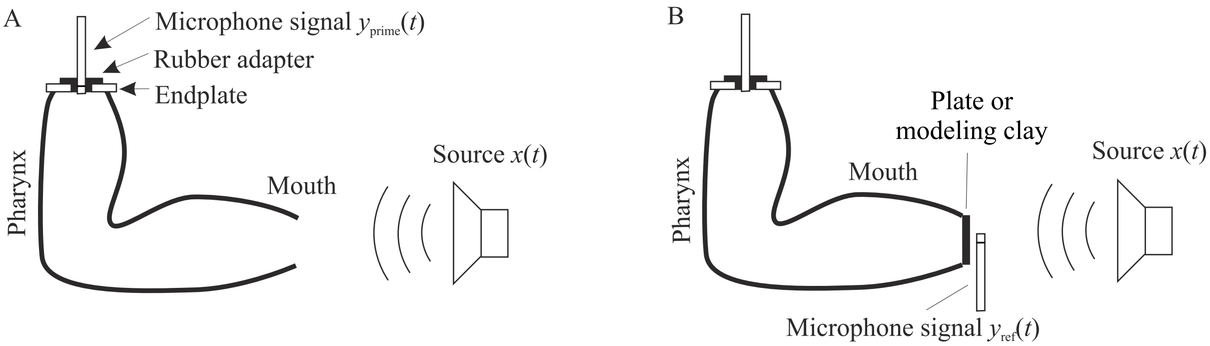


Figure 1: Measurement setup for determining the volume velocity transfer function of physical vocal tract models according to Fleischer et al. [3].

The primary motivation for the development of this software was the measurement of the volume velocity transfer functions between the glottis and the lips of 3D-printed physical models of the vocal tract. In Fleischer et al. [3] we presented a method how this transfer function can be precisely measured while avoiding the need for a broadband volume velocity source at the glottis, which is hard to create. The measurement procedure is shown in Fig. 1. First, a wideband sweep $x(t)$ is emitted by a loudspeaker in front of the vocal tract model into the open "mouth" of the model, where $t$ is the time. The response is measured as the microphone signal $y_{\text{prime}}(t)$ at the position of the glottis within the model as shown in Fig. 1A. This *primary* signal contains the resonances of the model, but is also affected by the radiation impedance of the model and the frequency response of the loudspeaker and the environment. To obtain the pure volume velocity transfer function of the model, a second (reference) measurement is performed according to Fig. 1B. For this measurement, the mouth of the model is first closed with a plate or with modeling clay. With the loudspeaker emitting the same source signal as before, the reference signal $y_{\text{ref}}(t)$ is then captured immediately in front of the closed mouth. The volume velocity

transfer function $H(\omega)$ of the model is finally the ratio of the spectra of both response signals, i.e., $H(\omega) = Y_{\text{prime}}(\omega)/Y_{\text{ref}}(\omega)$, where $\omega$ is the angular frequency. Please refer to [3] for more details. However, the software MTF is not limited to the use case above but is also applicable, for example, to the characterization of loudspeakers or other audio components and to room acoustics measurements.
MTF has been developed for the Windows platform and is internally using the Windows API to control the audio interface. For the graphical user interface, the library wxWidgets 3.1.0 `www.wxwidgets.org` was used. The project was created with MS Visual Studio 2013. If you want to compile the sources with the provided project file, please compile with the "x64 Release" configuration (the other configurations have no valid settings). The software was tested with Windows 7 and Windows 10. However, there is no guarantee that the software is bug-free, so please feel free to report any problems to `peter.birkholz@vocaltractlab.de`. The software is free of charge and open source. Feel free to use it for whatever you want or modify it as you like.

## 2 Theoretical background

This section provides only a brief overview of the background of the method. Please refer to [1,2] for more details. The measurement method uses logarithmic sine sweeps to excite the acoustic system of interest, because they allow the extraction of the linear impulse response of the system even in the presence of harmonic distortions. The excitation signal $x(t)$ is

$$x(t) = x_0 \sin\left[\frac{\omega_1 T}{\ln(\omega_2/\omega_1)}\left(e^{\frac{t}{T}\ln(\omega_2/\omega_1)} - 1\right)\right],$$

where $x_0$ is the amplitude, $\omega_1 = 2\pi f_1$ and $\omega_2 = 2\pi f_2$ are the start and end frequencies of the sweep, respectively, and $T$ is the sweep duration. The energy of this signal is concentrated between $\omega_1$ and $\omega_2$ and has a spectral decay of 3 dB/oct. As a sudden onset and offset of the sweep would cause spectral distortions at the edges of its power band, a fading-in and a fading-out are normally used at the beginning and end of $x(t)$. To determine the acoustic transfer function between the source that emits $x(t)$ and a microphone that captures the acoustic system response $y(t)$, $y(t)$ has to be convolved with the "inverse" of $x(t)$ (inverse in terms of the spectrum) to obtain the impulse response $h(t)$ of the system. The inverse signal $x_{\text{inv}}(t)$ is

$$x_{\text{inv}}(t) = x(T-t) \cdot e^{-\frac{t}{T}\ln(\omega_2/\omega_1)}.$$

Hence, the inverse signal is the time reversed $x(t)$ multiplied by an exponential decay. The system impulse response is hence $h(t) = y(t) * x_{\text{inv}}(t)$, where the operator (*) denotes convolution. Taking the Fourier Transform of $h(t)$ yields the transfer function $H(\omega)$ of the system. The beneficial property of this method is that the signal $h(t)$ contains not only the linear impulse response, but in addition separate impulse responses for each harmonic distortion order. All these impulse responses are well separated along the time axis of the signal $h(t)$ with the impulse responses corresponding to the harmonic distortions appearing *before* the linear impulse response. When only the linear impulse response in $h(t)$ is (manually) selected and the rest of the signal is set to zero (suppression of the distortions), the Fourier Transform gives the corresponding desired *linear* transfer function of the system of interest.

## 3 Usage

The graphical user interface of MTF is shown in Fig. 2. It consists of a control panel on the left side, and three displays for signals on the right side. To setup and perform a measurement, please follow the steps below.

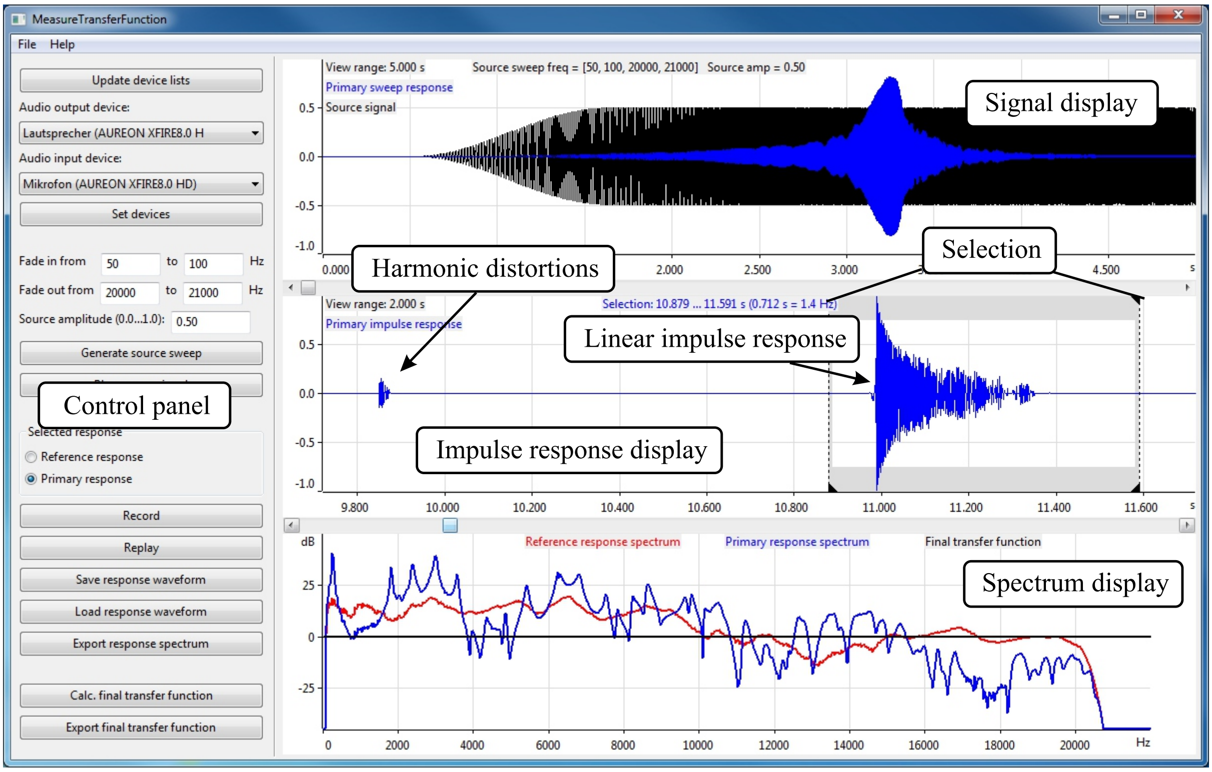


Figure 2: Program window.

1. Select the audio devices for playback and recording using the dropdown lists "Audio output device" and "Audio input device" in the control panel. After selecting the devices press the button "Set devices". If there are any initialization errors of the devices they are shown in the console window. MTF tries to initialize both the input and output devices with a sampling rate of 96 kHz and 24 bit quantization. The sound levels for playback and recording must be set in your operating system. We suggest to set a volume of 100% for the speaker, and adjust the volume of the microphone so that no clipping occurs during the measurements.

2. Generate the logarithmic sine sweep that will be used as the source signal by providing the relevant frequencies in the text boxes for fade-in and fade-out in the control panel and press the button "Generate source sweep". Similar to [1], the sweep is specified by four frequency values: the start and end frequencies of the fade-in phase ($f_1$ and $f_2$, respectively), and the start and end frequencies of the fade-out phase ($f_3$ and $f_4$, respectively). The "usable" frequency range is between the fade-in and fade-out phases, i.e., from $f_2$ to $f_3$. The frequencies $[f_1, f_2, f_3, f_4]$ for the current sweep are given in the signal display. You can listen to the generated sweep signal (without simultaneous recording) with the button "Play source signal". The waveform of the source signal is shown as the black signal in the signal display. To see the spectrum of the source signal select "Show source spectrum" from the context menu in the spectrum display. Here you can also display the inverse source signal and the product of the source signal and its inverse. The product spectrum should have a flat magnitude between $f_2$ and $f_3$. If the frequency ranges for fade-in or fade-out are too small, there may be some ringing at the edges of the power band magnitude, which should be avoided.

3. Select the response signal that you want to record (either "Reference response" or "Primary response").

4. Press the button “Record” to play the source signal and simultaneously record the selected response signal. After recording, the waveform of the response is shown in the signal display (in blue for the primary response and in red for the reference response). You can replay this waveform with the button “Replay”. The impulse response, i.e., the response signal convolved with the inverse of the source signal, is shown in the impulse response display. This signal is automatically peak-normalized, i.e., the maximum value of the signal is 1.0. If the actual impulse response is outside the viewport of the display, select “Go to impulse response” from the context menu or use the scrollbar to move there. You can also use the keys Ctrl + ← and Ctrl + → to scroll through the signal. The (linear) impulse response starts at a time of about 11 s (which corresponds to the length of the source signal and is a consequence of the de-convolution). For a better visibility of the impulse responses the samples are displayed on a logarithmic scale by default. You can change to a linear scale via the context menu. The spectrum display shows the magnitude spectrum of the impulse response. Use the buttons “Save response waveform” and “Load response waveform” to save or load response waveforms as WAV files.

5. Select the *linear* impulse response in the impulse response display and calculate its spectrum. As discussed in Sec. 2, the de-convolved response signal contains not only the linear impulse response, but also the impulse responses of different harmonic distortion orders, all of which occur earlier than the linear impulse response. The impulse response display in Fig. 2 shows an example where the linear impulse response has been selected, and where the impulse response of the first harmonic distortion order is visible further left. Use the context menu in the impulse response display to set the selection (“Set selection start” and “Set selection end”) and to calculate the spectrum from the selected signal (“Selection to spectrum”). The response spectrum is updated accordingly in the spectrum display.

6. If your setup has only one measurement point, you can now export the spectrum of the linear impulse response as a TXT file with the button “Export response spectrum”. The spectrum is saved with a resolution of about 1 Hz. The TXT file can be imported into other programs like Matlab or Excel for further analysis.

7. If you have two measurement points (as in the example in the introduction), you can now select the other response (“Reference” or ”Primary”) and repeat the steps from step 4.

8. If you have calculated two response spectra, i.e., a primary response spectrum $A(\omega)$ and a secondary response spectrum $B(\omega)$, you can use the button “Calc. final transfer function” to calculate the transfer function $H(\omega) = A(\omega)/B(\omega)$. For this calculation, the peak-normalization of the impulse responses performed in step 4 is compensated for. Before the calculation of $H(\omega)$, a dialog appears where you can enter a dB value by which the level of the transfer function will be shifted. Use this to compensate for any differences of the sensitivity of the used microphones, for example. You can export $H(\omega)$ as a TXT file with the button “Export final transfer function”.

## Acknowledgments

I thank Steffen Kürbis for contributing to the source code and testing the software, Simon Stone for proof-reading this manual, and Adam Stark for making available his ’AudioFile’ library, which is used by this software.

## References

[1] Bertrand Delvaux and David Howard. A new method to explore the spectral impact of the piriform fossae on the singing voice: benchmarking using MRI-based 3D-printed vocal tracts. *PLoS One*, 9(7):e102680, 2014.

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[3] Mario Fleischer, Alexander Mainka, Steffen Kürbis, and Peter Birkholz. How to precisely measure the volume velocity transfer function of physical vocal tract models by external excitation. *PLoS One*, 13(3):1–16, 03 2018.