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Computer speech phenomena: multitrack method of recording EVP

Introduction. The study of electronic voice phenomenon (EVP), opened in 20th century by Friedrich Jurgenson,

has its history at least 50 years. In general, existing methods of EVP recordings can be divided into 3 classes:

microphone technique (Jurgenson), radio methods (Jurgenson, Raudive, Franz Seidl, Marcello Bacci, etc.) and

recording by use of special devices (George Meek, Hans Otto Koenig , Jules and Maggie Harsh-Fischbach, etc.) Unfortunately, the microphone and radio methods are often very limited, and special devices are difficult for

most users and very dependent on the human factor (the operator). The aim of researchers seeking to date, in

our opinion, should be finding compromise between the serviceability of the method and its availability. One of

these variants, opened by our group in 2008, we want to propose in this article. EVP recording in this version

requires:

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1. Your computer or laptop running Windows XP,

2. Sound card with built-in function «Mono Mix» or «What I Hear»

3. The program Cool Edit Pro, or its modern version - Adobe Audition.

4. Microphone, if necessary, connected to the mic input sound card through the amplifier.

Preparation. First, you need to create a sampler in the form of "canned" speech from the mixture in a foreign

language. Any recording with a mixture of four or five foreign radio stations, carefully checked for possible coincidences with the native speech can be suitable. If any such parts are found, they should be cut using a sound editor. These "cans" you may save as mp3 files on your computer in a place suitable for rapid access.

Then you should open the Control Panel, find the menu "sounds and audio devices" and choose the option of

internal sound card mixer «Mono (stereo) Mix» or its equivalent - «What I Hear». The volume of internal mixer

must be pointed to minimum, common volume control and microphone volume input - to the maximum. Connect

electret microphone (with its own power supply) to mic input. This microphone is used for asking questions and

it also provides low-level noise carrier, which, as we found, is necessary for the emergence of the effect. If the

signal level is insufficient, it is necessary to use a portable microphone amplifier.

Procedure of recording. Using the standard program Windows Media Player, mp3-file with "cans" is run for playing. It's necessary to adjust the volume of Windows Media Player (in our experiments it was the level of 15-20 percent). You should also adjust the volume of internal sound card mixer. Roughly, the volume of "cans"

should be such, that during recording it has about 1/4 of maximum visual amplitude (the amplitude of the signal,

at which it ceases to rise and goes to the limit). To provide the "other side" the necessary variety of phonemes,

"cans" can be played not from the beginning, but from another point. After playing is started, run the program

Cool Edit, and press the upper left corner button (switching to multitrack mode). To maintain **mono multitrack recording with realtime overlay,** set Track Record -> 16 bit mono in the menu «Options> Settings> Multitrack», as well the option «Sample Accurate Sync» in the menu «Effects». Choose for yourself

the length of track (the best time of recording of each track - 20 - 40 seconds). Press the red button 📧 on the

left side of the first track and start the button («Record») at the bottom. After start of the recording of the

first track, ask your question into the microphone. To give the "other side" some time to respond, the length of

the question should not be more than 10 seconds. After recording the first track press the stop button 💻 , then

press the key on the first track 📧 and press the same button in front of the second one. Then press the record

button again. If you did everything correctly, you should see, that the signal on the second track is overlaid

in real time by the already recorded signal on the first track. Length of recording on the second track is scheduled to coincide with the length of the first. Press the stop button \blacksquare , go to the third track, and so on.

This procedure must be repeated 4 times. After four-stage overlaying of the re-recorded speech signal amplitude becomes maximal. The result should look something like this:



Note 1. If you want to choose the number of tracks, other than four, you must select a different initial volume of "cans"

Note 2. Despite of the author's commitment to elementary style of presentation, it is not an exhaustive guide to

the use of software Cool Edit Pro and Adobe Audition. For complete information we strongly recommend you to

consult Help menu or the relevant literature.

Note 3. We strongly recommend you to disable your speakers (headphones) till the end of recording.

Listening and Processing. Since on the fourth track your question via the microphone is not always heard clearly, we recommend you first call the first track (by the double clicking on it by the left mouse button), listen to

your question and remember the time where it ends on the timeline. Your answer (EVP) is located **on the fourth track**. Call it by double clicking the left mouse button on the track or by double clicking on the name of

the track in the left menu (see photo). At the first stage you can listen to this track without any processing. Replies you desire very often go directly after the question and clearly stand out against the general background

of speech. If you believe, that you have found an answer, you can select a part of recording with the mouse and

subject it to processing «Noise reduction» (see below). In the case of any mistake you can always return to previous situation by clicking the key ("back") on top. If the answer is really received, you can copy it to separate file (Edit> Copy to New) and save (File> Save As). Once you have reviewed the entire track, we move

to secondary processing. It allows you to allocate the available information not audible during normal listening.

This processing is fulfilled by means of the function Effects> Noise Reduction. Below we propose two possible

settings: you can try each of them and choose the best. After copying the values shown in the picture You should

select by your mouse the whole track, call the function "Noise Reduction" and take the noise profile (Get Profile

from Selection). Then press OK. Once the program has made the processing, listen to the track (this time - processed) again.

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Noise Reduction Level Low Noise Reduction Settings FFT Size 4096 points Remove Noise	Reduce by Precision Factor	▶ 15 High 40 dB 77 1	Less	

The 1st variant of processing.



The 2nd variant of processing.

Note 4. Before processing of the tracks by Noise Reduction, you can cut off all the frequencies above 4 kHz using the FFT filter, or a graphic equalizer (menu «Filters»)

Possible explanations. The most "naive" explanation of experimentally proved working of this method is that

the man's subconsciousness "catches" some fragments of speech, and gives them a meaningful interpretation.

It is impossible not to recognize, that this really happens sometimes, especially if you want to "hear" hidden meaning of each fragment of recording. However, as follows from results of hundreds of experiments, this explanation can not be universal and persuasive because:

1. The real answer (which often represents a single word or sentence that gives an answer or a comment to experimenter's question) stands out very clearly against the overall speech signal, can be filtered and consistently identified by experimenters.

2. There is a dependence of the number of responses received on the state and mood of the operator: in certain

conditions (inappropriate time, operator's fatigue) there are no responds at all, and it is confirmed by bystanders.

3. Finally, in violation of certain conditions, the effect is modest or disappears altogether.

So, let us consider other explanations:

1. Influence on the microphone signal by entities from the "other side", which is chosen in such a way as to enhance or suppress individual phonemes during recording "stochastic" phonemic material.

2. Similar effect on the sound card.

3. Manipulation of error in the sampling (digitization) of the analog signal on the sound card, the accumulation of

errors during successive re-recordings through the internal sound card mixer.

4. Influence on the PC processor during signal processing (it is the last stage - application of the function Noise

Reduction).

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We draw a parallel here with the video-method of Klaus Schreiber, who sought the full manifestation of his paranormal images through successive iterations - i.e. series of re-dubbing a picture from the screen on the film (see [1])

Comparison with the Stefan Bion's program EVP Maker ([2]).

Similar to the program EVP Maker, quasi-random verbal factor is used as the raw material for the formation

the responses, but the difference is that the formation of the signal is not discrete, but continuous, and moreover.

in our method emergence of quite distinguishable separate EVPs takes place. These EVPs are quite characteristic and suitable for filtering.

Advantages of the method.

1. Independence from the local broadcasting and interference in it, unlike using psychophone and radiomethods.

2. Exclusion of the argument of "random radio broadcasts" (signal is not broadcasted through the air, it's never

going beyond the sound card)

3. High frequency and quality of the responses.

4. Widespread availability of the method and its performance (distinct meaningful answers come in almost every session).

The disadvantages of the method are:

1. In some cases - the possibility of subjective interpretation of what we heard.

2. In the case when our ITC signal is superimposed on the background speech with similar characteristics - the

difficulty of filtration of the desired signal.

3. Difficulty of the method for experimenters who are not familiar with the applicable software.

Additional factors. It was discovered during experiments, that additional factors, such as quartz crystals, ultraviolet lights, metronome, rhythmically flashing fluorescent lamp, have a positive impact on the process of

EVP recording, but they are not necessary.

Instead of re-recording through the internal mixer, you can use a combination of FM - transceiver which broadcasts "canned" speech to the radio (tuned to unoccupied FM radio frequency, connected to the sound card through mic input.

Parallels. The phenomenon of EVP in a computer without any external microphone signal was also opened by

Brazilian researcher Sonia Rinaldi ([3])

References.

1. Hildegard Schaefer. Bridge between the Terrestrial and Beyond. http://www.worlditc.org/c 04 s bridge content.htm

2. http://stefanbion.de/evpmaker/index.htm

3. IPATI Bulletin № 23.