data fast transmission,

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A FAST SPEECH DATA FILES TRANSMISSION BY ASCII CHARACTERS IN-CODING

SHORT NOTE

This note shortly describes the system of a fast transmission of oral data files that are converted into ASCII characters. These files are using lower capacity the transmission capacity of the communication virtual data links.

1. INTRODUCTION

Nowadays the sharing data bases, knowledge and experience with other scientific groups are much needed service in every research discipline, among them the medicine. A fast and effective research results exchange by communication facilities, among distributed medical centres is one of the fundamental conditions of the development centres in medical research units.

This short note describes the idea of the method of an interactive knowledge sharing using the transmission facilities of the Internet services for multimedia data files with a speech data units' distribution. The question still can be put - is it possible converting the data files for transmission on-line the audio and image signals, via standard telecommunication cables? The paper shows one of the possible solutions by ASCII characters coding.

2. THE CODING METHOD FOR A SPEECH DATA UNITS

It has been assumed that audio and video data are recorded separately in two independent streams of the data. The method requires specific hardware units, based on a DSP processor [1,2], usually built into the computer sound and video extension cards, for fast audio and video data processing. They are cooperating with software units for speech recognition. The application for the data encoding and for a speech synthesis are usually

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integrated with the coding units. The block scheme in Fig.1 presents component of this interface structure.

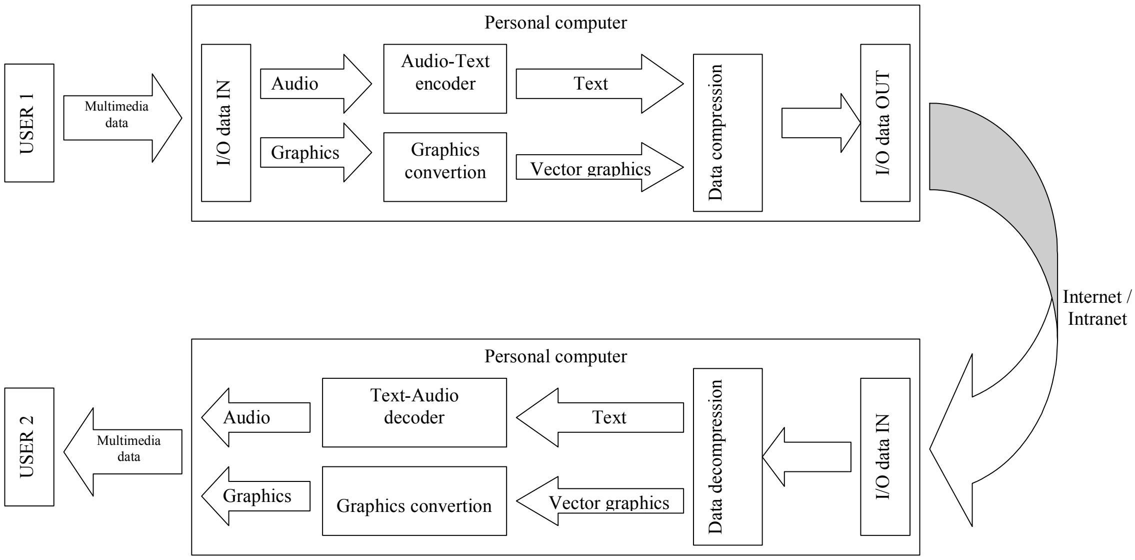

Fig. 1. The block scheme of the data transmitter / receiver for two users

The source audio data stream is recorded then converted into the ASCII strings or characters, by the speech analyser. The comparison results presented in a table 1 clearly proves that an every speech file, compressed in any well known (classical) algorithm provides the user with much bigger product then the file obtained by the proposed algorithm.

Using this ASCII strings compression the user can obtained the very effective transmission rate of the data. Compare in the Table 1 the rows assigned by AUDIO markers and the rows with TEXT markers, where explains encouraging compression results for often used audio formats (as: wav, mp3), zip compression and not compressed text files.

The experiments done for audio records lasted 4s, the compression of ASCII text file was not needed, what is more the necessary headings will take to much space that deny the compression goals (the compressed short text files can be longer then source files).

Table 1. Comparison of sizes of text and audio file formats.

Type of file	Audio quality				File size [B]		
	Frequency [kHz]	Sampling [b]	Mode	Bit rate [kb/s]	wav	mp3	zip
AUDIO	44	16	Stereo	172	799.802	108.015	71.347
	44	16	Mono	172	399.930	35.517	60.126
	32	16	Mono	63	290.180	17.827	29.907
	22	8	Mono	22	10.0022	13.250	45.118
	11	8	Mono	11	50.040	4.201	25.680
	8	8	Mono	8	36.326	2.851	1.9618
TEXT	ASCII text file				71		

This way overworked data is transmitted through the standard telecommunication lines. The audio data stream was reduced remarkable. This method combined with a vector format for graphics, provide the user with the satisfying data transfer system that makes the audio/video teleconferencing via low capacity transmission cables possible (Fig.2).

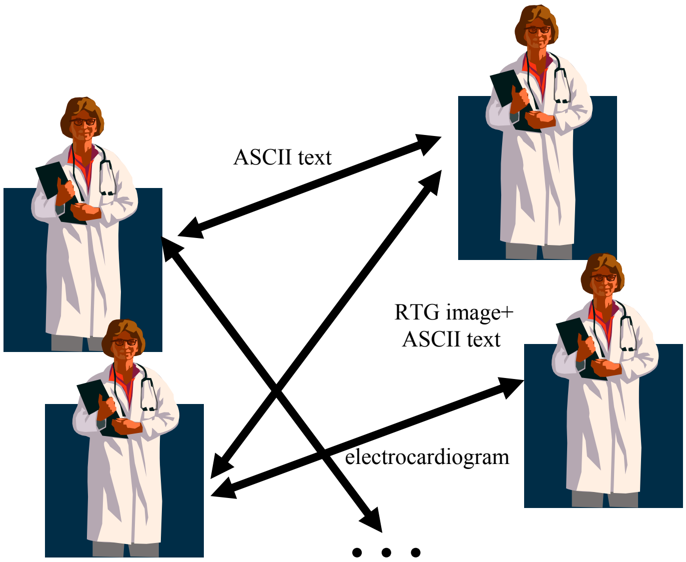

Fig. 2. Multimedia data exchange flow chart.

Anyhow, the implied on-line data processing requires a effective enough processor capacity in allocated terminals. The transmitted voice is not distorted in the end terminal, as it goes in time of its exposition, then conversation into the data stream. The quality of decoded sound depends on scope of the speech synthesiser only. Additionally, in the system there is a possibility of graphics transmission that complements the communication data units.

The static images illustrating the speech (discussion), as RTG images, CT images and so on are converted transferred as vector image units. They are interlaced with the converted audio data files.

This way the data transmission rates will increase remarkable by a fast and effective compression method. The transmission process will be filled up by the short video units that complete the transmitted data.

3. CONCLUSIONS

A fast verbal communication complemented with concurrent transmission of graphics using data link connection based on low capacity transmission channels occurs very encouraging not only for medical knowledge databases content sharing but also for remote decision making diagnosis. The discussed method allows using not expensive data transmission systems as they are available in majority of hospital centres.

Taking into account that doctors are not experienced enough in a keyboard handling with, in on line internet terminals usage (for the discussion), the presented method of communication simplify the information exchange between to or more partners. The natural interfaces offered this way will encourage using the computer communication technologies wider then it is used today.

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