Digital Signal Processing for Knowledge Based Sonotubometry of Eustachian Tube Function

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Abstract: With the advances of electronics and software technologies in the last decade, an important new direction in sonotubometry has been created concerning an assessment of the Eustachian tube (ET) function under physiological conditions. Despite the fact that the sonotubometry technique has gradual been improved in the last twenty five years, it is not yet used systematically to evaluate Eustachian tube ventilatory function, because its reproducibility, history, mapping to the context (patient, clinic data, medication, a.o.) validation and value for clinical practice have not yet been consolidated and integrated in a service oriented, knowledge-based system that nowadays can use advanced modelling, segmentation, search, data analytics and prediction tools. The utilisation of Digital Signal Processing techniques for sound acquisition and low-level processing, noise rejection, attenuation monitoring and digital feature-based representation of the sound records create the premises to confirm the validity and reproducibility of sonotubometry as test method. This is one of the primary contributions of the research reported in this paper. Another objective is the integration of the components necessary to build a knowledge-based system used in context-driven Eustachian tube function evaluation and decision support for otological diagnostic: feature-based modelling of the digitized sound records; context-driven mapping of sound records in a knowledge base with multiple tagging; iterative learning process; data analytics; predictive analysis for decision optimization. The pilot implementation of the system and experiments are provided.

Keywords: Digital Signal Processing, Eustachian tube measurement, sonotubometry, sound features, KB tagging, predictive analysis, decision support, otological diagnostic.

1. INTRODUCTION

With the advances of electronics and software technologies in the last decade, an important new direction in sonotubometry has been created concerning an assessment of the Eustachian tube (ET) function under physiological conditions. The ET is the connection between the nasopharyngeal space and the middle; it assures three important functions with respect to the middle ear cavity: protection, clearance and ventilation (Van der Avoort, 2007). Disturbance of one of the ET functions can result from a multitude of factors and can lead to different disorders (Prades *et al.,* 1998).

Measurements of the Eustachian tube function have been subject of many clinical studies, due to its importance for clinical applications and practice. Because physiology and the effects of ET dysfunctions on the pathogenesis of middle ear diseases are still not totally understood, different types of investigation methods: endoscopic, radiologic, manometric, tympanometric and sonometric have been developed for the assessment of ET function (Di Martino *et al.,* 2004; Honjo, 1988; McBride *et al.*, 1988). Most of the methods used in clinical investigation evaluate pressure variations in the middle ear; as a drawback, such techniques cannot be applied in both patients having intact and perforated eardrums.

Sonotubometry measures the active ventilator function of the Eustachian tube using sound (Virtanen, 1978; Okubo *et al.,* 1987; Palva *et al.,* 1987). A constant acoustic stimulus is applied to the nasopharyngeal orifice (the nostril), while a low noise microphone in the external auditory canal records the sound pressure level directed through the ET and middle ear. Special manoeuvres performed by the patient, like dry swallowing, provoke the ET opening during a brief period of time; an increase in sound level will be measured in the external auditory canal. Thus, the active ventilatory function of the ET can be evaluated non-invasively and the measurements take place under physiological conditions, without the need to use high pressures or tympanic membrane perforation (Van der Avoort *et al.,* 2005).

In addition, sonotubometry is cost effective, painless and easy to perform in both adults and children; it is considered to have a high potential value as complex analysis and diagnostic tool for patients having possible ET pathology.

Various approaches are reported, especially concerning the type of stimulus sound: pure-tone and broadband signals (Murti *et al.,* 1980) with frequencies higher than 6 KHz are considered the most efficient, because noise pollution determined by pharyngeal activity occurs below 5 KHz (Munro *et al.,* 1999).

Despite the fact that the sonotubometry technique has gradual been improved in the last twenty five years, it is not yet used systematically to evaluate Eustachian tube ventilatory function, because its reproducibility, history, mapping to the context (patient, clinic data, medication, a.o.) validation and value for clinical practice have not yet been consolidated, and integrated in a service oriented, knowledge-based system that nowadays can use advanced modelling, segmentation, search, data analytics and prediction tools.

The utilisation of Digital Signal Processing (DSP) techniques in the stage of sound acquisition and low-level processing, noise rejection, attenuation monitoring and digital feature-based representation of the sound records create the premises to confirm the validity and reproducibility of sonotubometry as test method. This is one of the primary contributions of the research reported in this paper.

Another objective of the present research is the integration of the components necessary to build a knowledge-based system used as context-driven Eustachian tube function evaluation and decision support for otological diagnostic: feature-based modelling of the digitized sound records; context-driven mapping of sound records in a Knowledge base with multiple tagging; iterative learning process; data analytics; predictive analysis for decision optimization.

The paper is organized as follows: Section 2 describes the related work in the area of DSP, and computer-based high level processing of the signal acquired from professional sound transmission and recording devices. Section 3 presents the DSP solution that has been developed for sonotubometry measurement. The data acquisition and low level processing system, as well as the solution for digital data filtering are in detail discussed. Section 4 presents the complete information system integrating low- and high-level applications and tools performing the knowledge-based sonotubometry of the Eustachian tube function. The conclusions and perspectives of future work are formulated in Section 5. Experimental results on DSP of sound stimuli and sound feature description are also provided.

2. RELATED WORK

Sonotubometry and Eustachian function evaluation under physiological circumstances have been studied by researchers in both dedicated and complementary approaches, partially using DSP, but very few of them reporting high level (sound) data processing techniques. (Di Martino *et al.,* 2007) evaluate the ET function by sonotubometry with 8 KHz pure-tone signals processed with a small band pass filter around 8 KHz by a custom-made device in normal subjects. The recorded events were analysed in a computer. All statistical analyses were performed with SAS ver. 8.02. The solution evaluates only mean sound intensities and durations by averaging repeated measurements for each manoeuvre of each proband, and report only detailed statistics for four ET opening acts (dry and liquid swallowing, yawning and Valsalva), without any evaluation of measurement reliability. Other research (Antweiler *et al.,* 2006a; Antweiler *et al.,* 2006b) report a real-time system where the Eustachian tube is treated as a linear transmission element; its impulse response and the related transfer function are obtained by a perfect sequence excitation (PSEQ) of the nose-ear ensemble and a subsequent system identification. A polyvinyl chloride (PVC) hardware model copying the human anatomy was used to compare results obtained with the related virtual model. For communication with the measurement hardware, the system uses a PC-based platform for multichannel real time DSP. The authors focus on the match between the reference PVC model and its virtual model. Alternatively, a nose/ear transfer function of a test person was computed, and the related virtual model calculated. After optimization, the virtual ET model featured a reasonable match to the given tube of the PVC model. While the virtual model is considered to visualize artificially the ET activity as a kind of "acoustic tube endoscopy", there are no references to further qualitative evaluation of the ET function or to the definition of new features; there is no context evaluation allowing feature labelling for medical diagnostic.

In (Asenov *et al.,* 2010), a new kind of sound stimulus was used – PSEQ – periodic random noise signals with an ideally flat spectrum. The goal was to investigate the function of impaired ET under physiological conditions. The sonotubometry tests with PSEQ were able to detect ET openings in both normal and pathological ears and to evaluate comparatively the average amplitude and duration of the openings. While this technology is complementary to our solution, it does not address the overall requirements for the otological diagnostic support.

The objective of the research reported in (Van der Avoort, 2009) is to test the outcome of sonotubometric measurement in children with otitis media with effusion (OME) before and after insertion of ventilation tubes. Fewer incidences of the opening of Eustachian tube were recorded in the measurements before insertion of ventilation tubes compared with after insertion. Although this research performs a detailed context analysis (patient, dysfunctions, medical actions – grommet insertion, evolution in time), the only feature analysed is the number of ET openings in a standard time interval, which limits the performances of the ET function assessment. A pilot study about synchronous endoscopy and sonotubometry of the Eustachian tube is done in (Handzel *et al.,* 2012), exploring its advantages compared with the performance of these tests independently. The study reports the following features for ET opening with dilatory efforts: fraction open [%]; duration [seconds]; sound intensity [dB]; and no. of efforts [#] for two types of manoeuvres provoking the opening: swallow and yawn. The classical test procedure uses pure-tone stimulus signals and configurable band pass filters; a Graphical User Interface was developed to detect discordances between the endoscopic view and the recording of the sonotubometer. This study is however limited by the small number of subjects and does not provide, by help of software tools, the capabilities of cluster analysis and pattern recognition in the sound feature space.

The solution proposed in this paper differentiates from the above mentioned work by integrating the sonotubometry sequences of digitized ET opening shape patterns and feature set in a knowledge base obtained through iterative learning.

Due to its multiple-criteria, context-related labelling, the resulting database can be efficiently addressed for decision support for otological diagnostic.

3. DSP FOR SONOTUBOMETRY MEASUREMENT

The sonotubometry method is used in order to evaluate the activity and function of the Eustachian tube in physiological conditions. In this investigation method a sound of certain intensity is applied to the nasopharyngeal orifice, by help of a standard loudspeaker of Insert Earphone type. This acoustic stimulus is conducted through the ET to the middle ear, and reaches the external ear canal when the ET opens for brief time periods; the opening will be provoked by mechanical acts such as: yawning, swallowing and Valsalva manoeuvres, see Fig. 1.

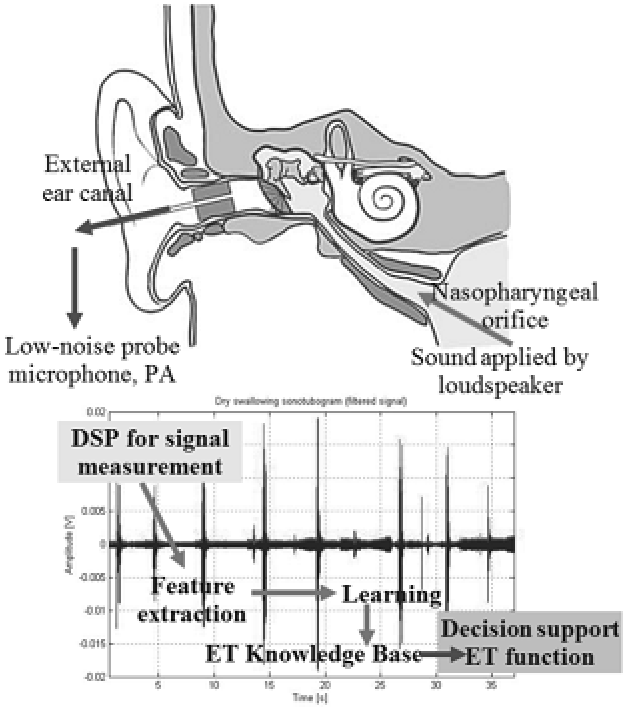

Fig. 1. The sonotubometry method and DSP measurement.

During ET opening, more sound will be directed, which will result in variable higher sound intensities to be captured in the ear's external auditory canal; this signal was acquired by help of a low noise probe microphone, amplified and then transferred for further processing, in order to evaluate the function of the Eustachian tube. A data acquisition system has been designed for this purpose.

3.1 The data acquisition and low level processing system

Digital Signal Processing (DSP) is used for sound acquisition and low level acoustic signal processing; the computer-based system has been designed to meet the requirements:

- Use pure-tone signals at a frequency between 6-8 KHz and acoustic intensity of 100 dB - sufficiently high to compensate for attenuations caused by bad positioning or dislocation of the probes in the nose or the ear, and compression of nostrils.
- Avoid the disturbing noise determined by pharyngeal activity; these signal components occur up to 5 KHz.
- Eliminate as much as possible noise pollution caused by the ambient environment (high frequency components), and disambiguate the response to the acoustic stimulus from the high amplitude noise caused by the mechanical activity of muscles, bones, epiglottis located in the close vicinity of the ET (low frequency sound components).
- Create the premises to automatically detect the start of an ET opening in a series of provoked openings, of any time length.
- Register changes in the sound intensity of the received stimulus over the ET opening periods, and visualise the "opening shape" of the tube for types of manoeuvres, such as: swallowing, yawning, etc.
- Identify 5-10 dB increases of sound intensity during manoeuvres, to classify changes in recorded acoustic signal intensity as valid ET openings.
- Configurability of the frequency, amplitude and sound intensity of the pure-tone stimulus, and of the sampling frequency of the data acquisition process.
- Create the premises for feature-based description of the electric representation of the "ET opening shape", and allow the extraction of these sound features and storage in a dedicated data base for further processing.
- Integrate a graphical, interactive user interface (GIUI) allowing the management of the sound records in terms of: duration, disambiguation, feature definition, context specification and proband data.

The computer-based sonotubometry measurement system using DSP techniques is shown in Fig. 2.

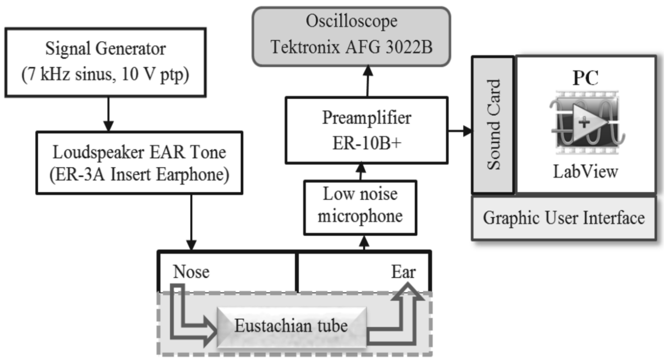

Fig. 2. Computer-based sonotubometry platform with DSP.

For the sonotubometry medical investigation experiment, a signal generator introduces in the system a sinusoidal signal of 7 KHz frequency and 10 V amplitude peak to peak (ptp). This signal is fed to an ER-3A 50 Ohm Insert Earphones loudspeaker acting as sound source for the ET. The device provides external noise exclusion of 30+ dB and has the capacity to eliminate collapsed nasopharyngeal orifice errors; the maximum peak voltage for 1% duty cycle is 20 V at 50 Ohm.

The ER-3A Insert Earphone conveys the stimulus sound to the ET via the nasopharyngeal orifice by help of a standard-length sound tube and a foam eartip. Foam eartips developed for ER-3A insert earphones have constant dimensions to ensure proper calibration and test accuracy; for example, the length respectively the diameter from the end of the eartip to the connection at the end of the earphone tube have been kept at the values of 22 mm and 1.93 mm. To accommodate differences in nostril size of the tested children, there have been used two types of foam eartips: small and medium. Inter pharyngeal attenuation is improved with deep insertion of the foam eartip in the nostril; after insertion, the foam expands and seals acoustically the nose canal.

During the set up phase of the system, the reduction of the background noise that might mask ET conduction test signals and influence threshold determinations was estimated. Ambient noise reduction with this type of earphone typically exceeded 30 dB in the frequency domain of 6-8 KHz which was considered. Audiometric air conduction threshold testing down to 0 dB Hearing Level (HL) could be performed in the presence of a background noise level not exceeding 45 dB.

The sonotubometry platform was designed to capture the sound conveyed through the ET from the external ear canal, to which a low-noise microphone system ER-10B+ was coupled (www.etymotic). The ER-10B+ system includes: (i) a microphone that uses foam eartips to accommodate most infant size ear canals; (ii) a preamplifier (PA) powered by two 9V alkaline batteries, that features a sensitivity of 50 mV/Pascal; (iii) standard front tubes of .95 mm OD x .58 mm ID x 76 mm length. The PA is switchable for 0, 20 or 40 dB additional low-noise gain. The microphone has an equalized flat frequency response from the small-diameter .95 mm OD probe tube; it uses stainless steel sound channels to reduce sound source crosstalk inside the microphone device.

The sound captured by the microphone from the external ear canal and amplified by the PA is input to the PC sound card installed on the PCI bus of a Lenovo ThinkPad T540P laptop with Intel®Core i7-4700 MQ 2.4 GHz processor, 8 GB RAM and 500 GB SSD. The PC sound card performs the signal acquisition process at high sample rate of 9600 samples / second. Thus it is possible to perform the acquisition of the acoustic signal having the component of 7 kHz corresponding to the stimulus sound.

The following configuration parameters have been set up for the sound card:

- One channel used;
- 16 bits resolution of quantization;
- Configurable acquisition time (standard 30 sec);
- Signal sample rate of 40 kHz.

The set up for sound acquisition was done in a dedicated controller which was created in the LabView development environment, as represented in Fig. 3.

After waiting for the completion of the acquisition time, the sound data read from the low-noise microphone system ER-10B+ is stored into a file in order to be filtered and further analysed using external MATLAB tools.

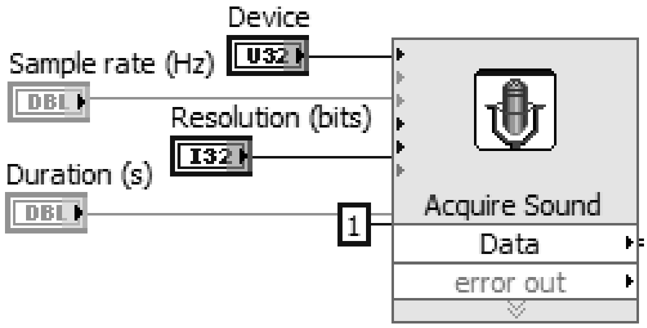

Fig. 3. LabView configuration for 7 KHz sound acquisition.

3.2 Digital data filtering

It is necessary to filter the acoustic signal received from the microphone during the above described experiment which aims at observing the highest possibly intensity of the stimulus signal when the Eustachian tube (ET) opens during the swallowing act. In addition to the 7-8 kHz stimulus signal passing unaltered through the ET during its opening, other components of lower frequencies are present in the received signal, being superposed on the useful one; these lower frequency components are generated by the mechanical activity of other noise generating sources located in the ET's vicinity: tongue movement, muscle and bone activity, epiglottis covering the windpipe, etc.

Filtering is also required to eliminate the noise pollution of higher frequency than 7-8 KHz, caused by the working environment. Fig. 4 shows the spectral distribution before the filtering procedure.

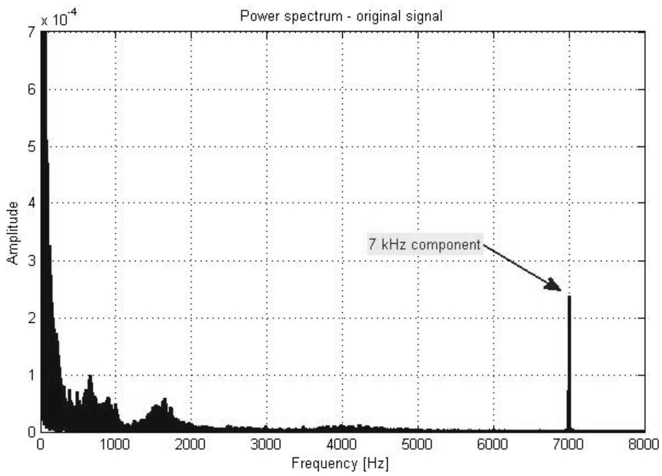

Fig. 4. Power distribution of the original signal acquired from the microphone, for a 7 KHz stimulus signal.

After filtering, only the 7 kHz nonzero component of interest will be kept and analysed.

In order to analyse only the amplitude of the stimulus signal received during ET opening and thus create the premises for correctly estimating its activity, two filtering methods have been considered.

The first method uses conventional filters such as band pass, high pass or Chebyshev filters. In this case, the following unwanted phenomena appear: (i) the resulting filtered signals include the impulse response of the filter which is overlapped on the useful signal; (ii) depending on the filter complexity (the filter's order) a considerable delay in obtaining the stimulus response can be generated. Therefore, the results obtained with this method did not produce the best results.

The second method does not consider the data analysis of the ET activity as an online procedure, and consequently uses directly the frequency representation of the signal, to filter out offline the low components induced by collateral mechanical activities and the high frequency caused by the environment. The filtering process involves the following steps:

- Converting the acquired acoustic signal from time representation vector to frequency representation, by applying a Fourier transform procedure. The MATLAB function **fft** (y) for discrete Fourier transform of a vector of length *N* has been used (1):

$$X(k)=\sum_{j=1}^{N} x(j)\omega_N^{(j-1)(k-1)} \tag{1}$$

- Removing the components outside of the band pass domain;
- Computing the inverse transformation, from frequency domain to time domain. The inverse function in MATLAB is **ifft** (y, *N*) which computes inverse fast Fourier transform of a vector of length *N* (2):

$$x(j)=\sum_{k=1}^{N} X(k)\omega_N^{-(j-1)(k-1)} \tag{2}$$

where $\omega_N = e^{\frac{-2\pi i}{N}} \in C$ and $i^2 = -1$.

Schematically, this procedure can be represented as:

$y(j)\xrightarrow{fft(y)}Y(k)\xrightarrow{filtering[a,b]Hz}Y_f(k)\xrightarrow{ifft(y)}y_f(j)$, where j is a time moment, k is a normed frequency and $y_f(j)$ is the filtered version of $y(j)$.

The result of filtering using the second method based on dual Fourier transforms according to the MATLAB code below is given in Fig. 5.

In order to observe accurately the evolution of the signal's amplitude and define features related to the activity of the Eustachian tube, the envelope of the filtered signal is also computed.

```
t=0:1/Fs:upper((length(y)-1)/Fs);
%puncte fft
N=length(y);

double_plot=figure();
subplot(2,1,1);
plot(t,y);
title 'Time representation';
xlabel 'Time [s]';
ylabel 'Amplitude [V]';
grid on

yfft=fft(y, N);
%frequency array
f=Fs/2*linspace(0,1,N/2+1);

%band pass boundaries
a=6.5e3;
b=7.5e3;

%equivalent sample number
sample_a=N/Fs*a+1;
sample_b=N/Fs*b+1;

%filtered frequencies
yfft_f=zeros(N,1);
yfft_f(sample_a: sample_b)=yfft(sample_a:
sample_b);
yfft_f( N-sample_b:N-sample_a)=yfft(N-
sample_b:  N-sample_a)

y_f=ifft(yfft_f);

figure(double_plot)
subplot(2,1,2);
plot(t, y_f);
title 'Filtered signal'
ylabel 'Amplitudde [v]'
xlabel 'Time [s]'

env=abs(hilbert(y_f));

hold on
plot(t, env,'r');
```

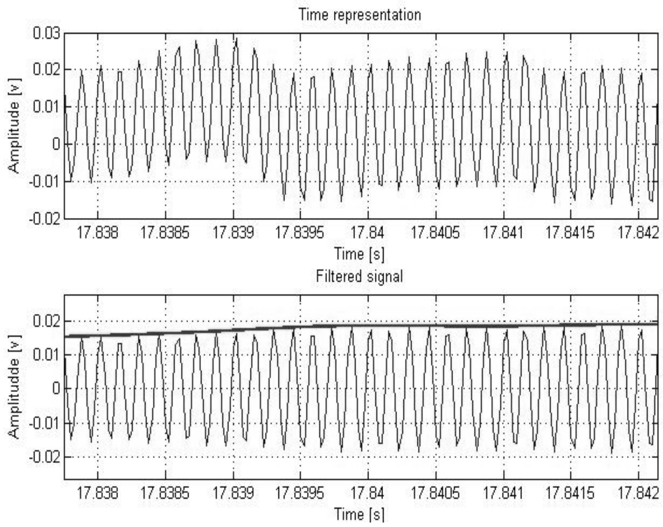

Fig. 5. Detailed view of the original signal (having a signal of lower frequency superimposed) and the filtered signal using the dual FFT transform. The envelope of the filtered signal is represented in red in the same plot.

4. KNOWLEDGE-BASED SONOTUBOMETRY OF THE ET FUNCTION

The computer-based sonotubometry platform using DSP techniques for low-level processing of the acoustic signals is further developed to provide two major functionalities:

- Knowledge Base Learning: this KB contains a number of features extracted from the ET signal records and patterns of ET opening shapes, created in the low-level sonotubometry processing phase.
- Decision Support: once the KB created, it can be used for two important medical activities: (1) Evaluation of the patients' dynamic Eustachian tube function under physiological conditions, and (2) Real time otological diagnostic.

The high-level, knowledge-based sonotubometry processing system is shown in Fig. 6.

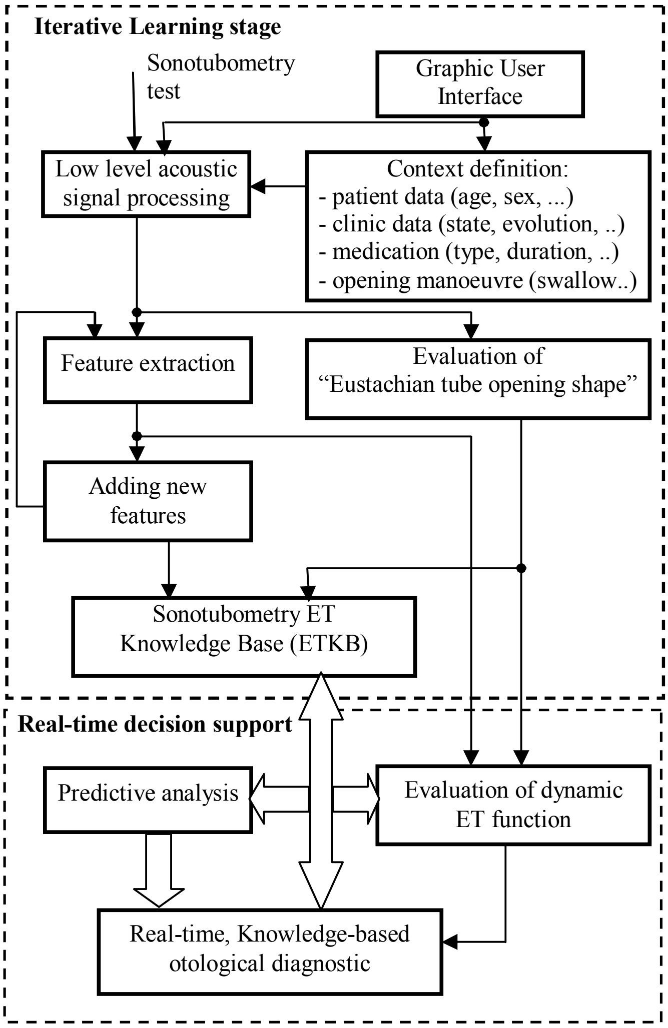

Fig. 6. High level acoustic signal processing for KB decision support to dynamic ET function evaluation and diagnostic.

Although the system operates in real time, the parameters for sound acquisition and low level processing, feature definition and context specification are input to the system using a dedicated Graphical User Interface. This interface is also used to map patient data to new ET opening records during the iterative learning stage, and select segmentation criteria in the KB search stage (Borangiu *et al.,* 2012).

The set of ET features, extracted from filtered sound records and the ET opening shape described as a digital signature are iteratively created during a learning process. Features may be added during this process, and new correlations defined with respect to the operational context: patient data, clinic history, evolution of the health state, effects of medication related to the above mentioned items or particularities of provoked ET openings.

This context can be (re)defined from the Graphical User Interface, which is also used as a multi-criteria searching tool in the ET database. During the learning stage, the main goal is to assure the diversity for ET feature analysis.

The sonotubometry ETKB is further used to assess the ET functions (e.g., ventilation parameters, opening specific, a.o.) of patients, which must also execute the above described sonotubometry test. Thus, in the current development stage, the self-extending ETKB speeds up this evaluation process. The KB otological diagnostic is the highest level evaluation process, and will use predictive analysis tools from the IBM SPSS suite, such as multi-criteria statistics evaluation and modelling.

Currently, there have been defined a number of features that are extracted from the filtered acoustic record corresponding to an Eustachian tube opening and numerically evaluated during the learning stage (these features reflect on individual basis or together important functions of the ET impacting the pathogenesis of middle ear, the pressure equilibration, ventilation and drainage of the middle ear and providing evidence for a larger area of clinical applications):

- Mean duration of Eustachian tube opening, defined as the time period after the moment the ET effectively opens (marked by the increase of at least 1mV of the acoustic signal taken with the ER-10 B+ microphone system and digitally filtered) when the intensity of the acoustic signal increases by 5 dB;
- Median increase of sound intensity during Eustachian tube opening;
- Mean value of sound intensity of the acquired signal during Eustachian tube opening;
- Maximum sound intensity during Eustachian tube opening;
- Number of modes present in the pattern of a single Eustachian tube opening shape.

Fig. 7 shows some of these features that can be selected via the system's GUI during the learning stage, being associated to the filtered acoustic signal during a single ET opening. Other features were defined for the ensemble of ET openings during a sonotubometry experiment, such as: (i) number of ET effective openings (from a total number of provoked actions caused by different types of manoeuvres such as swallowing, yawning, a.o.) in a selected time interval; (ii) percentage of opening with respect to an effective one [%]. An effective ET opening is identified by a sound intensity greater than a predefined threshold (expressed in voltage units or in sound intensity units).

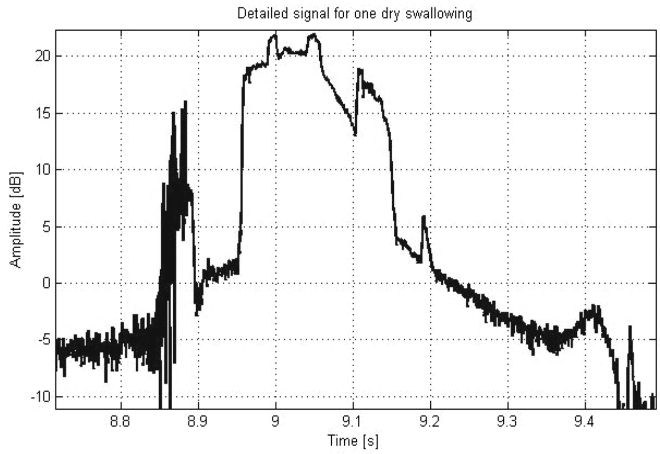

Fig. 7. Features tagging the acoustic signal during an ET opening (sound record obtained from 7 KHz, 10 V pure tone).

As it has been presented in the Introduction and Related Work sections of this paper, the number of effective openings in a sonotubometry sequence is one most important feature.

Fig. 8 presents such a global ET feature at experiment level; the time diagram has been obtained by double FFT filtering as described in Section 3. The envelope of the acoustic signal obtained after filtering is marked in red in the figure.

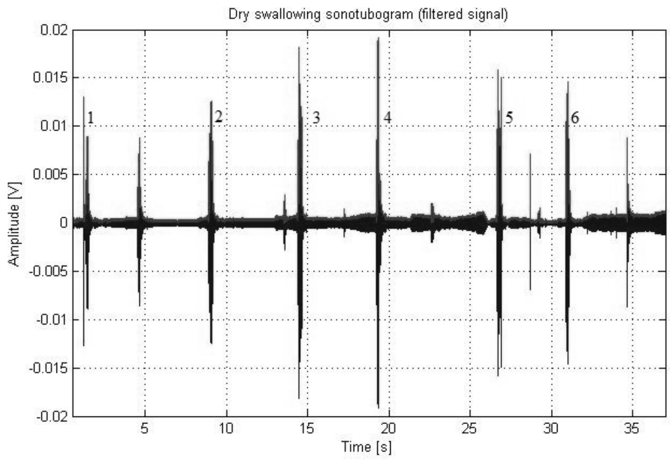

Fig. 8. Global feature at experiment level: number of real ET openings (6 signal amplitude greater than 1 mV) in a testing time interval (35 seconds).

The digital signature of an "ET opening shape" is obtained by temporal sampling and signal quantification (see Fig. 9); the number of modes which are present in the pattern of such a signature can be obtained by simple numerical computation.

In the learning phase of the sonotubometry computer-based system, associations are made between the feature values and the context which created the values (patient, clinic data, etc). Because of the multiple-context tagging of the ETKB, this database can be used as an efficient decision support tool for a broader set of clinical applications in otological diagnostic.

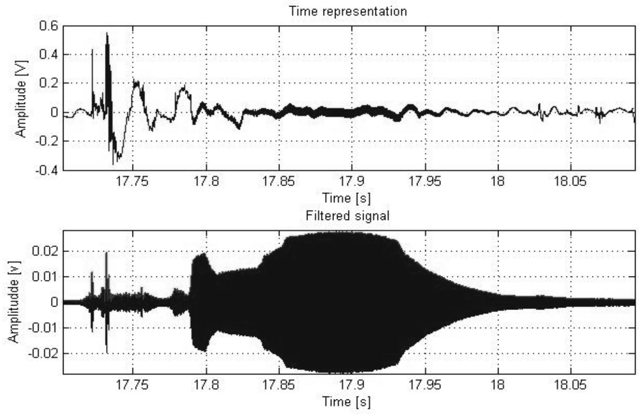

Fig. 9. Digital signature of an individual "ET opening shape".

5. CONCLUSIONS AND FUTURE WORK

This paper presents a novel mechanism for real time decision support for the evaluation of the dynamic Eustachian tube function and for otological diagnostic. The mechanism is constructed as an iterative learning process that produces a multi-tagged knowledge base which can be consulted in real time to evaluate by sonotubometry a patient's Eustachian tube function. This ET Knowledge Base can be exploited for otological diagnostic by help of data analytics and predictive analysis tool: statistics, modelling and decision support from the SPSS suite.

Conceptually the mechanism extracts sound features from acoustic signal records acquired from the external ear canal; these signals are filtered, sampled, quantized and treated in a Digital Signal Processing system using LabView as programming environment and MATLAB for mathematical computing.

The paper reports mainly the research that has been carried out for the accurate representation of the acoustic stimulus conducted through the ET to the middle ear, and captured from the external ear canal when the ET opens for brief time periods. The research describes also the extraction of sound features both from patterns corresponding to single effective ET openings provoked by mechanical acts such as: yawning, swallowing and Valsalva manoeuvres, and from global sound patterns generated by the entire sonotubometry experiment.

The experimental results presented in sections 3 and 4 show that Digital Signal Processing, using directly the frequency representation of the signal, effectively filters out off line the low components induced by collateral mechanical activities and the high frequency caused by the environment. This approach produces better results than the conventional filters such as band pass, high pass or Chebyshev filters, which are reported by most of authors.

The computer-based sonotubometry platform using DSP with LabView in the low-level acoustic signal processing stage is developed for medical investigation of the ET ventilation function at children. The specialised equipment for stimulus generation and reception uses Etymotic Research components of high performance: ER-3A Insert Earphone for probe sound generation, and the ER 10B+ low noise, sensitive microphone system.

A set of 24 sonotubometry tests, where the ET opening was provoked by dry- and liquid swallowing manoeuvres has been performed on healthy patients. The results confirm the viability of system design and the low-level DSP functions that have been created for accurate signal filtering, sound feature extraction, and digital representation of the temporal signature of Eustachian tube opening shapes. Fig. 10 shows the signal envelope [dB] of the filtered stimulus in Fig. 8 over the test period of 35 seconds.

The potential for assisting clinical applications in otological areas with the proposed system is high due to the fact that usage patterns for the evaluation of Eustachian tube functions and real time decision support for diagnostic and treatment are available from a multiple tagged Knowledge base, ETKB. Compared to other approaches for ET investigation, the solution proposed in this paper is superior in two regards: the reconfigurability and context-driven enrichment of sound features in the iterative learning stage of the application, and secondly, the reliable decision support obtained using high level data analytics and prediction tools.

The physical computer-based platform including the Etymotic Research specialized sound generation and transmission equipment for sonotubometry experiments is shown in Fig. 11.

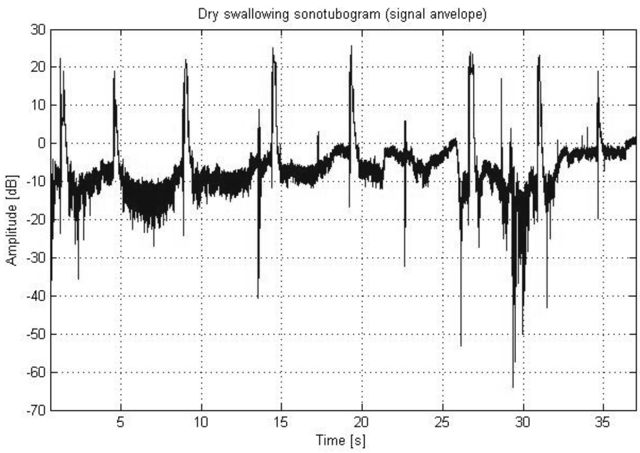

Fig. 10. Envelope of the filtered signal in Fig. 8, obtained in a 35 seconds dry swallowing sonotubogram session.

Future work will extend the sonotubometry Knowledge base (ETKB) by defining new sound features and Eustachian tube opening shape patterns through iterative learning and developing a multi criteria, context driven search mechanism for the qualitative evaluation of ET activity and dysfunctions. A configurable segmentation tool and predictive modelling technique will be integrated in the statistics processing of the ETKB, for decision support to otological diagnostic.

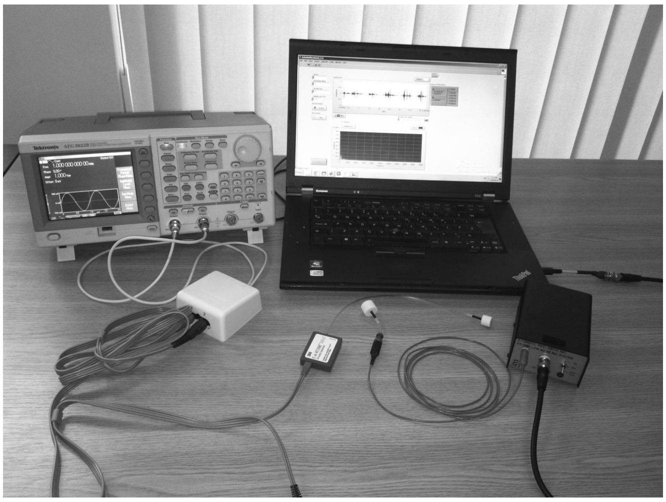

Fig. 11. The computer platform for sonotubometry tests.

REFERENCES

Antweiler, C., Telle, A., Di Martino, E. and Vary, P. (2006a). A New Otological Diagnostic System Providing a Virtual Tube Model, *Proc. of the ISCAS IEEE Conf.*, 21-24.

Antweiler, C., Vary, P. and Di Martino, E. (2006b). Virtual Time-Variant Model of the Eustachian Tube, *Proc. of the IEEE ISCAS Conference*, 5559-5562.

Asenov, D.R., Nath, V., Telle, A., Antweiler, C., Walther, L.E., Vary, P. and Di Martino, E. (2010). Sonotubometry with perfect sequences: First results in pathological ears, *Acta Oto-Laryngologica*, 130, 1242-1248.

Borangiu, A., Iacob, I. and Vlad, A. (2012). Cell-based processing of video and medical data flows from ambulance network, in *Proc. of Int. Conf. FIHS-Fostering Innovation in Healthcare Services*, Carol Davila Publishing House, Bucharest, 1-6.

Di Martino, E., Thaden, R., Krombach. G.A. and Westhofen, M. (2004). Eustachian tube function tests current knowledge, HNO, 52, 1029-1040.

Di Martino, E., Thaden, R., Antweiler, Christiane, Reineke, T., Westhofen, M., Veckschebe, J, Vorländer M. and Vary, P. (2007). Evaluation of Eustachian tube function by sonotubometry: results and reliability of 8 KHz signals in normal subjects, *Eur. Arch. Otorhinolaryngol.*, 264, 231-236.

Handzel, O., Poe, D. and Marchbanks, R.J. (2012). Synchronous Endoscopy & Sonotubometry of the Eustachian Tube: A Pilot Study, *Otology and Neurology*, 33, 184-191.

Honjo, I. (1988). Evaluation of static and dynamic functions of the Eustachian tube, in Honjo, I. (Ed.) *The Eustachian tube in middle ear diseases*, Springer, Tokyo, S25-S38.

McBride T.P., Dekray, C., Cunningham, M. and Doyle, W. (1988). Evaluation of non-invasive Eustachian tube function tests in normal adults, Laryngoscope, 98, 655-658.

Munro, K.J. Benton, C.L. and Marchbanks, R.J. (1999). Sonotubometry findings in children at high risk from middle ear effusion, *Clinic. Otolaryngol*., 24, 223-227.

Murti, K.G., Stern, R., Cantekin, E. and Bluestone, C.D. (1980). Sonometric evaluation of Eustachian tube function using broadband stimuli, *Ann. Otol*., 89, 68, 178-184.

Okubo, J., Watanabe, I., Shibusawa, M., Ishikawa, N., Ishida, H. and Teramura, K. (1987). Sonotubometric measurement of the Eustachian tube function by means of band noise; a clinical view of the acoustic measurement of the Eustachian tube, *ORL J. Otorhinolaryngol. Relat. Spec*., 49(5), 242-245.

Palva, T., Marttila, T. and Jauhiainen, T. (1987). Comparison of pure tones & noise stimuli in sonotubometry, *Acta Otolaryngol*. (Stockholm), 103, 212-216.

Prades, J.M., Dumollard, J.M., Calloch, F., Merzougui, N. and Martin, C. (1998). Descriptive anatomy of the human auditory tube, *Surg. Radiol. Anat*., 20, 335-340.

Van der Avoort, S.J.C., Van Heerbeek, N., Zielhuis, G.A. and Cremers, W.R.J. (2005). Sonotubometry - Eustachian tube ventilatory function test; a review, *Otology and Neurology*, 26(3), 583-543.

Van der Avoort, S.J.C. (2007). *Sonotubometry. Measurement of the Eustachian tube function*, Ph.D. Thesis, Radboud University of Nijmegen, The Netherlands, 11.

Van der Avoort, S.J.C., Van Heerbeeck, N., Zielhuis, G.A. and Cremers, C. (2009). Sonotubometry in Children with Otitis Media with Effusion before and after Insertion of Ventilation Tubes, *Arch. Otolaryngol. Head, Neck. Surg.*, 135(5), 448-452.

Virtanen, H. (1978). Sonotubometry, an acoustical method for objective measurement of auditory tube opening, *Acta Otolaryngol*., 86, 93-103.

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